## Communications System Toolbox ${ }^{\text {TM }}$

 ReferenceR2013a

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## Communications System Toolbox ${ }^{\text {TM }}$ Reference

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# Functions - Alphabetical List 

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Functions - Alphabetical
List

## algdeintrlv

Purpose

Syntax

Description

## Examples

Restore ordering of symbols using algebraically derived permutation table

```
deintrlvd = algdeintrlv(data,num,'takeshita-costello',k,h)
deintrlvd = algdeintrlv(data,num,'welch-costas',alph)
```

deintrlvd = algdeintrlv(data, num, 'takeshita-costello', k, h) restores the original ordering of the elements in data using a permutation table that is algebraically derived using the Takeshita-Costello method. num is the number of elements in data if data is a vector, or the number of rows of data if data is a matrix with multiple columns. In the Takeshita-Costello method, num must be a power of 2 . The multiplicative factor, $k$, must be an odd integer less than num, and the cyclic shift, $h$, must be a nonnegative integer less than num. If data is a matrix with multiple rows and columns, the function processes the columns independently.
deintrlvd = algdeintrlv(data, num, 'welch-costas',alph) uses the Welch-Costas method. In the Welch-Costas method, num +1 must be a prime number. alph is an integer between 1 and num that represents a primitive element of the finite field GF(num +1 ).

To use this function as an inverse of the algintrlv function, use the same inputs in both functions, except for the data input. In that case, the two functions are inverses in the sense that applying algintrlv followed by algdeintrlv leaves data unchanged.

The code below uses the Takeshita-Costello method of algintrlv and algdeintrlv.

```
num = 16; % Power of 2
ncols = 3; % Number of columns of data to interleave
data = rand(num,ncols); % Random data to interleave
k = 3;
h = 4;
intdata = algintrlv(data,num,'takeshita-costello',k,h);
deintdata = algdeintrlv(intdata,num,'takeshita-costello',k,h);
```

References [1] Heegard, Chris, and Stephen B. Wicker, Turbo Coding, Boston, Kluwer Academic Publishers, 1999.
[2] Takeshita, O. Y., and D. J. Costello, Jr., "New Classes Of Algebraic Interleavers for Turbo-Codes," Proc. 1998 IEEE International Symposium on Information Theory, Boston, Aug. 16-21, 1998. p. 419.
See Also ..... algintrlv
How To - "Interleaving"

## algintrlv

Purpose
Reorder symbols using algebraically derived permutation table

Syntax

Description

Examples

```
intrlvd = algintrlv(data,num,'takeshita-costello',k,h)
```

intrlvd = algintrlv(data, num, 'welch-costas', alph)
intrlvd = algintrlv(data, num, 'takeshita-costello', k, h) rearranges the elements in data using a permutation table that is algebraically derived using the Takeshita-Costello method. num is the number of elements in data if data is a vector, or the number of rows of data if data is a matrix with multiple columns. In the Takeshita-Costello method, num must be a power of 2. The multiplicative factor, k , must be an odd integer less than num, and the cyclic shift, $h$, must be a nonnegative integer less than num. If data is a matrix with multiple rows and columns, the function processes the columns independently.
intrlvd = algintrlv(data, num, 'welch-costas', alph) uses the Welch-Costas method. In the Welch-Costas method, num+1 must be a prime number. alph is an integer between 1 and num that represents a primitive element of the finite field GF(num +1 ). This means that every nonzero element of GF(num+1) can be expressed as alph raised to some integer power.

This example illustrates how to use the Welch-Costas method of algebraic interleaving.

1 Define num and the data to interleave.

```
num = 10; % Integer such that num+1 is prime
ncols = 3; % Number of columns of data to interleave
data = randi([0 num-1], num, ncols); % Random data to interleave
```

2 Find primitive polynomials of the finite field GF(num+1). The gfprimfd function represents each primitive polynomial as a row containing the coefficients in order of ascending powers.

```
pr = gfprimfd(1,'all',num+1) % Primitive polynomials of GF(num+1)
pr =
```

| 3 | 1 |
| :--- | :--- |
| 4 | 1 |
| 5 | 1 |
| 9 | 1 |

3 Notice from the output that pr has two columns and that the second column consists solely of 1 s . In other words, each primitive polynomial is a monic degree-one polynomial. This is because num +1 is prime. As a result, to find the primitive element that is a root of each primitive polynomial, find a root of the polynomial by subtracting the first column of pr from num +1 .

```
primel = (num+1)-pr(:,1) % Primitive elements of GF(num+1)
primel =
```

8
7
6
2

4 Now define alph as one of the elements of primel and use algintrlv.

```
alph = primel(1); % Choose one primitive element.
intrlvd = algintrlv(data,num,'Welch-Costas',alph); % Interleave.
```


## Algorithms

- A Takeshita-Costello interleaver uses a length-num cycle vector whose nth element is $\bmod \left(k^{*}(n-1) * n / 2\right.$, num) for integers $n$ between 1 and num. The function creates a permutation vector by listing, for each element of the cycle vector in ascending order, one plus the element's successor. The interleaver's actual permutation table is the result of shifting the elements of the permutation vector left by $h$. (The function performs all computations on numbers and indices modulo num.)
- A Welch-Costas interleaver uses a permutation that maps an integer $K$ to $\bmod \left(A^{K}, n u m+1\right)-1$.


## algintrlv

$\begin{array}{ll}\text { References } & \text { [1] Heegard, Chris, and Stephen B. Wicker, Turbo Coding, Boston, } \\ & \text { Kluwer Academic Publishers, 1999. }\end{array}$
[2] Takeshita, O. Y., and D. J. Costello, Jr., "New Classes Of Algebraic Interleavers for Turbo-Codes," Proc. 1998 IEEE International
Symposium on Information Theory, Boston, Aug. 16-21, 1998. p. 419.

## See Also <br> algdeintrlv

How To . "Interleaving"

## Purpose

Syntax
[ $\mathrm{Xa}, \mathrm{Ya}$ ] = alignsignals( $\mathrm{X}, \mathrm{Y}$ )
[ $\mathrm{Xa}, \mathrm{Ya}$ ] = alignsignals( $\mathrm{X}, \mathrm{Y}$, maxlag)
[Xa,Ya] = alignsignals(X,Y,maxlag,'truncate')
[Xa,Ya, D] = alignsignals( __ )
Align two signals by delaying earliest signal
[ $\mathrm{Xa}, \mathrm{Ya}$ ] = alignsignals( $\mathrm{X}, \mathrm{Y}$ ) estimates the delay $D$ between the two input signals, $X$ and $Y$, and returns the aligned signals, $X a$ and $Y a$.

- If $Y$ is delayed with respect to $X$, then $D$ is positive, and $X$ is delayed by $D$ samples.
- If $Y$ is advanced with respect to $X$, then $D$ is negative, and $Y$ is delayed by $-D$ samples.

Delays in X and Y can be introduced by prepending zeros.
[ $\mathrm{Xa}, \mathrm{Ya}$ ] = alignsignals( $\mathrm{X}, \mathrm{Y}$, maxlag) uses maxlag as the maximum window size to find the estimated delay $D$ between the two input signals, X and Y . It returns the aligned signals, Xa and Ya .
[Xa, Ya ] = alignsignals(X,Y,maxlag,'truncate') keeps the lengths of the aligned signals, $X a$ and $Y a$, the same as those of the input signals, $X$ and $Y$, respectively.

- If the estimated delay $D$ is positive, then $D$ zeros are prepended to $X$ and the last $D$ samples of $X$ are truncated.
- If the estimated delay $D$ is negative, then $-D$ zeros are prepended to $Y$ and the last $-D$ samples of $Y$ are truncated.


## alignsignals

Notes X and Y are row or column vectors of length $L X$ and $L Y$, respectively.

- If $D \geq L X$, then Xa consists of $L X$ zeros. All samples of X are lost.
- If $-D \geq L Y$, then $Y a$ consists of $L Y$ zeros. All samples of $Y$ are lost.

To avoid assigning a specific value to maxlag when using the 'truncate' option, set maxlag to [].
[ $\mathrm{Xa}, \mathrm{Ya}, \mathrm{D}]=$ alignsignals ( _ _ ) returns the estimated delay D . This syntax can include any of the input arguments used in previous syntaxes.

## Input Arguments

## X - First input signal

vector of numeric values
First input signal, specified as a numeric vector of length $L X$.
Example: [1, 2, 3]

## Data Types

single | double | int8 | int16 | int32 | int64 | uint8 |
uint16 | uint32 | uint64
Complex Number Support: Yes

## Y - Second input signal

vector of numeric values
Second input signal, specified as a numeric vector of length $L Y$.
Example: [0, 0, 1, 2, 3]

```
Data Types
single | double | int8 | int16 | int32 | int64 | uint8 |
uint16 | uint32 | uint64
```

Complex Number Support: Yes

## maxlag - Maximum window size or lag <br> scalar integer | []

Maximum window size, or lag, specified as an integer-valued scalar. By default, maxlag is equal to max (length $(X)$, length $(Y))-1$. If maxlag is input as [ ], it is replaced by the default value. If maxlag is negative, it is replaced by its absolute value. If maxlag is not integer valued, or is complex, Inf, or NaN, then alignsignals returns an error.

## Example: 2

```
Data Types
single | double | int8 | int16 | int32 | int64 | uint8 |
uint16 | uint32 | uint64
```

Default: max(length $(X)$, length $(Y))-1$

## Output Arguments

## Xa-Aligned first signal <br> vector of numeric values

Aligned first signal, returned as a numeric vector that is aligned with the second output argument Ya . If input argument X is a row vector, then $X a$ will also be a row vector. If input argument $X$ is a column vector, then $X a$ will also be a column vector. If you specify the 'truncate' option and the estimated delay $D$ is positive, then Xa is equivalent to the input signal X with $D$ zeros prepended to it and its last $D$ samples truncated.

## Ya-Aligned second signal

vector of numeric values
Aligned second signal, returned as a numeric vector that is aligned with the first output argument Xa . If input argument Y is a row vector, then Ya is also a row vector. If input argument Y is a column vector, then Ya is also a column vector. If you specify the 'truncate' option and the estimated delay $D$ is negative, then Ya is equivalent to the input signal $Y$ with $-D$ zeros prepended to it and its last $-D$ samples truncated.

## D - Estimated delay between input signals

## alignsignals

## scalar integer

Estimated delay between input signals, returned as a scalar integer. This integer represents the number of samples by which the two input signals, $X$ and $Y$ are offset.

- If $Y$ is delayed with respect to $X$, then $D$ is positive and $X$ is delayed by D samples.
- If $Y$ is advanced with respect to $X$, then $D$ is negative and $Y$ is delayed by -D samples.
- If $X$ and $Y$ are already aligned, then $D$ is zero and neither $X$ nor $Y$ are delayed.

If you specify a value for the input argument maxlag, then $D$ must be less than or equal to maxlag.

## Examples Aligning two signals where the second signal lags by two samples

Align signal $X$ when $Y$ is delayed with respect to $X$ by two samples.
Create two signals, $X$ and $Y$. $Y$ is exactly the same as $X$, except $Y$ has two leading zeros. Align the two signals.

```
X = [1 2 3];
Y = [0 0 1 2 3 3];
MAXLAG = 2;
[Xa Ya D] = alignsignals(X, Y, MAXLAG)
```

The resulting values are:

| $X a=$ |  |  |  |  |
| :--- | :--- | :--- | :--- | :--- | :--- |
|  |  |  |  |  |
| $Y a$ | 0 | 1 | 2 | 3 |
| $D=$ | 0 | 1 | 2 | 3 |
| 0 |  |  |  |  |

## Aligning two signals where the first signal lags by three samples

Align signal $Y$ with respect to $X$ by advancing it three samples.
Create two signals, $X$ and $Y$. $X$ is exactly the same as $Y$, except $X$ has three leading zeros and one additional following zero. Align the two signals.

```
X = [0 0 0 1 2 3 0 0]';
Y = [1 2 3 0]';
[Xa Ya] = alignsignals(X, Y)
```

The resulting values are:

```
Xa =
    0
    0
    0
    1
    2
    3
    O
    0
Ya =
    0
    0
    0
    1
    2
    3
    0
```


## Aligning two signals where the second signal is noisy

Align signal $Y$ with respect to $X$, despite the fact that $Y$ is a noisy signal.
Create two signals, X and Y . Y is exactly the same as X with some noise added to it. Align the two signals.
$x=\left[\begin{array}{lllll}0 & 0 & 1 & 2 & 3\end{array}\right]$;

## alignsignals

```
Y = [0.02 0.12 1.08 2.21 2.95 -0.09];
[Xa Ya D] = alignsignals(X, Y)
```

You do not need to change the input signals to produce the output signals. The delay D is zero. The resulting values are:

```
Xa =
    0
Ya =
    0.0200
    0.1200
    1.0800 2.2100
    2.9500
D =
    0
```


## Aligning two signals where the second signal is a periodic repetition of the first signal

Align signal $Y$ with respect to $X$, despite the fact that $Y$ is a periodic repetition of $X$. Return the smallest possible delay.

Create two signals, X and Y . Y is exactly the same as X with some noise added to it. Align the two signals.

```
X = [llllll
Y = [1 2 3 0 0 0 0 1 2 3 0 0];
[Xa Ya D] = alignsignals(X, Y)
```

The resulting values are:


## Aligning two signals using the 'truncate' option

Invoke the 'truncate' option when calling the alignsignals function.

Create two signals, $X$ and $Y$. $Y$ is exactly the same as $X$, except $Y$ has two leading zeros. Align the two signals, applying the 'truncate' directive.

```
X = [llll
Y = [0 0 1 2 3];
[Xa Ya D] = alignsignals(X, Y, [], 'truncate');
```

Observe that the output signal Xa has a length of 3 , the same length as input signal X . The resulting values are:

```
Xa =
Ya = llllll
D =
    2
```

In the case where using the 'truncate' option ends up truncating all the original data of x , a warning is issued. To make alignsignals issue such a warning, run the following example.

```
X = [llll
Y = [0 0 0 0 1 2 3];
[Xa Ya D] = alignsignals(X, Y, [], 'truncate')
```

The resulting warning is:

```
Warning: All original data in the first input X has been
truncated because the length of X is smaller than the
estimated delay D: to avoid truncating this data do not use
the 'trunc' option.
> In alignsignals at 136
```

The resulting values are:

```
Xa =
    \(0 \quad 0 \quad 0\)
\(Y a=\)
    \(\begin{array}{lllllll}0 & 0 & 0 & 0 & 1 & 2 & 3\end{array}\)
```


## alignsignals

D =
4

## Algorithms

- You can find the theory on delay estimation in the specification of the finddelay function (see "Algorithms" on page 1-264).
- The alignsignals function uses the estimated delay $D$ to delay the earliest signal such that the two signals have the same starting point.
- As specified for the finddelay function, the pair of signals need not be exact delayed copies of each other. However, the signals can be successfully aligned only if there is sufficient correlation between them.

For more information on estimating covariance and correlation functions, see [1].

## References

[1] Orfanidis, S.J., Optimum Signal Processing. An Introduction. 2nd Edition, Prentice-Hall, Englewood Cliffs, NJ, 1996.

## See Also finddelay

Related - "Use the Find Delay and Align Signals Blocks"
Examples
Concepts •"Delays"

| Purpose | Amplitude demodulation |
| :--- | :--- |
| Syntax | $z=\operatorname{amdemod}(y, F c, F s)$ |
|  | $z=\operatorname{amdemod}(y, F c, F s$, ini_phase $)$ |
|  | $z=\operatorname{amdemod}(y, F c, F s$, ini_phase, carramp $)$ |
| $z$ | $=\operatorname{amdemod}\left(y, F c, F s, i n i \_\right.$phase, carramp, num, den $)$ |

Description
$z=\operatorname{amdemod}(y, F c, F s)$ demodulates the amplitude modulated signal y from a carrier signal with frequency Fc (Hz). The carrier signal and y have sample frequency Fs (Hz). The modulated signal y has zero initial phase and zero carrier amplitude, so it represents suppressed carrier modulation. The demodulation process uses the lowpass filter specified by [num,den] = butter(5,Fc*2/Fs).

Note The Fc and Fs arguments must satisfy Fs $>2(F c+B W)$, where BW is the bandwidth of the original signal that was modulated.
z = amdemod(y,Fc,Fs,ini_phase) specifies the initial phase of the modulated signal in radians.
z = amdemod(y,Fc,Fs,ini_phase,carramp) demodulates a signal that was created via transmitted carrier modulation instead of suppressed carrier modulation. carramp is the carrier amplitude of the modulated signal.
z = amdemod(y,Fc,Fs,ini_phase,carramp,num,den) specifies the numerator and denominator of the lowpass filter used in the demodulation.

## Examples The code below illustrates the use of a nondefault filter.

```
t = .01;
Fc = 10000; Fs = 80000;
t = [0:1/Fs:0.01]';
s = sin(2*pi*300*t)+2*sin(2*pi*600*t); % Original signal
```

```
[num,den] = butter(10,Fc*2/Fs); % Lowpass filter
y1 = ammod(s,Fc,Fs); % Modulate.
s1 = amdemod(y1,Fc,Fs,0,0,num,den); % Demodulate.
```


## See Also

ammod | ssbdemod | fmdemod | pmdemod

How To<br>- "Digital Modulation"

## Purpose Amplitude modulation

Syntax $\quad$| $y$ | $=\operatorname{ammod}(x, F c, F s)$ |
| ---: | :--- |
| $y$ | $=\operatorname{ammod}(x, F c, F s$, ini_phase $)$ |
| $y$ | $=\operatorname{ammod}(x, F c, F s$, ini_phase, carramp $)$ |

## Description

## Examples

$y=\operatorname{ammod}(x, F c, F s)$ uses the message signal $x$ to modulate a carrier signal with frequency $\mathrm{Fc}(\mathrm{Hz})$ using amplitude modulation. The carrier signal and $x$ have sample frequency $\mathrm{Fs}(\mathrm{Hz})$. The modulated signal has zero initial phase and zero carrier amplitude, so the result is suppressed-carrier modulation.

Note The $x, F c$, and Fs input arguments must satisfy Fs $>2(\mathrm{Fc}+\mathrm{BW})$, where BW is the bandwidth of the modulating signal $x$.
$y=\operatorname{ammod}\left(x, F c, F s, i n i \_p h a s e\right)$ specifies the initial phase in the modulated signal y in radians.
y = ammod(x,Fc,Fs,ini_phase,carramp) performs transmitted-carrier modulation instead of suppressed-carrier modulation. The carrier amplitude is carramp.

The example below compares double-sideband and single-sideband amplitude modulation.

```
% Sample the signal }100\mathrm{ times per second, for 2 seconds.
Fs = 100;
t = [0:2*Fs+1]'/Fs;
Fc = 10; % Carrier frequency
x = sin(2*pi*t); % Sinusoidal signal
% Modulate x using single- and double-sideband AM.
ydouble = ammod(x,Fc,Fs);
ysingle = ssbmod(x,Fc,Fs);
```

```
% Compute spectra of both modulated signals.
zdouble = fft(ydouble);
zdouble = abs(zdouble(1:length(zdouble)/2+1));
frqdouble = [0:length(zdouble)-1]*Fs/length(zdouble)/2;
zsingle = fft(ysingle);
zsingle = abs(zsingle(1:length(zsingle)/2+1));
frqsingle = [0:length(zsingle)-1]*Fs/length(zsingle)/2;
% Plot spectra of both modulated signals.
figure;
subplot(2,1,1); plot(frqdouble,zdouble);
title('Spectrum of double-sideband signal');
subplot(2,1,2); plot(frqsingle,zsingle);
title('Spectrum of single-sideband signal');
```



See Also<br>amdemod \| ssbmod \| fmmod \| pmmod<br>How To . "Digital Modulation"

| Purpose | Decode binary code using arithmetic decoding |
| :---: | :---: |
| Syntax | dseq = arithdeco(code, counts, len) |
| Description | dseq = arithdeco(code, counts,len) decodes the binary arithmetic code in the vector code to recover the corresponding sequence of len symbols. The vector counts represents the source's statistics by listing the number of times each symbol of the source's alphabet occurs in a test data set. This function assumes that the data in code was produced by the arithenco function. |
| Examples | This example is similar to the example on the arithenco reference page, except that it uses arithdeco to recover the original sequence. ```counts = [99 1]; % A one occurs 99% of the time. len = 1000; seq = randsrc(1,len,[1 2; .99 .01]); % Random sequence code = arithenco(seq,counts); dseq = arithdeco(code,counts,length(seq)); % Decode. isequal(seq,dseq) % Check that dseq matches the original seq.``` |
|  | The output is ans = |
|  | 1 |
| Algorithms | This function uses the algorithm described in [1]. |
| References | [1] Sayood, Khalid, Introduction to Data Compression, San Francisco, Morgan Kaufmann, 2000. |
| See Also | arithenco |
| How To | - "Arithmetic Coding" |

Purpose Encode sequence of symbols using arithmetic coding

Description

Examples

This example illustrates the compression that arithmetic coding can accomplish in some situations. A source has a two-symbol alphabet and produces a test data set in which $99 \%$ of the symbols are 1s. Encoding 1000 symbols from this source produces a code vector having many fewer than 1000 elements. The actual number of elements in code varies, depending on the particular random sequence contained in seq.

```
counts = [99 1]; % A one occurs 99% of the time.
len = 1000;
seq = randsrc(1,len,[1 2; .99 .01]); % Random sequence
code = arithenco(seq,counts);
s = size(code) % length of code is only 8.3% of length of seq.
```

Algorithms

## References <br> [1] Sayood, Khalid, Introduction to Data Compression, San Francisco, Morgan Kaufmann, 2000.

See Also
This function uses the algorithm described in [1].

The output is
s =

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How To . "Arithmetic Coding"

Purpose Add white Gaussian noise to signal

```
Syntax
y = awgn(x,snr)
y = awgn(x,snr,sigpower)
y = awgn(x,snr,'measured')
y = awgn(x,snr,sigpower,s)
y = awgn(x,snr,'measured',state)
y = awgn(...,powertype)
```


## Description

$y=a w g n(x, s n r)$ adds white Gaussian noise to the vector signal $x$. The scalar snr specifies the signal-to-noise ratio per sample, in dB. If $x$ is complex, awgn adds complex noise. This syntax assumes that the power of $x$ is 0 dBW .
$y=\operatorname{awgn}(x, s n r$, sigpower $)$ is the same as the syntax above, except that sigpower is the power of $x$ in $d B W$.
$y=\operatorname{awgn}(x, s n r, ' m e a s u r e d ')$ is the same as $y=a w g n(x, s n r)$, except that awgn measures the power of $x$ before adding noise.
$y=a w g n(x, s n r$, sigpower,s) uses s, which is a random stream handle, to generate random noise samples with randn. If $s$ is an integer, then resets the state of randn to s. The latter usage is obsolete and may be removed in a future release. If you want to generate repeateable noise samples, then provide the handle of a random stream or use reset method on the default random stream.
$y=\operatorname{awgn}(x, s n r$, 'measured',state) is the same as $y=$ awgn ( $x, \operatorname{snr}$, 'measured'), except that awgn first resets the state of normal random number generator randn to the integer state.

Note This usage is deprecated and may be removed in a future release. Instead of state, use s, as in the previous example.
$\mathrm{y}=\operatorname{awgn}(. . .$, powertype $)$ is the same as the previous syntaxes, except that the string powertype specifies the units of snr and sigpower. Choices for powertype are 'db' and 'linear'. If powertype
'db ', then snr is measured in dB and sigpower is measured in dBW . If powertype is 'linear', snr is measured as a ratio and sigpower is measured in watts.

## Relationship Among SNR, $\mathbf{E}_{\mathbf{s}} / \mathbf{N}_{\mathbf{0}}$, and $\mathbf{E}_{\mathrm{b}} / \mathbf{N}_{\mathbf{0}}$

For the relationships between SNR and other measures of the relative power of the noise, see "AWGN Channel Noise Level".

## Examples

The commands below add white Gaussian noise to a sawtooth signal. It then plots the original and noisy signals.

```
t = 0:.1:10;
x = sawtooth(t); % Create sawtooth signal.
y = awgn(x,10,'measured'); % Add white Gaussian noise.
plot(t,x,t,y) % Plot both signals.
legend('Original signal','Signal with AWGN');
```



The scattereyedemo also illustrates the use of the awgn function.

## See Also

## Purpose BCH decoder

```
Syntax decoded = bchdec(code,n,k)
decoded = bchdec(...,paritypos)
[decoded,cnumerr] = bchdec(...)
[decoded,cnumerr,ccode] = bchdec(...)
```

decoded $=b c h d e c(c o d e, n, k)$ attempts to decode the received signal in code using an $[\mathrm{n}, \mathrm{k}] \mathrm{BCH}$ decoder with the narrow-sense generator polynomial. code is a Galois array of symbols over GF(2). Each n-element row of code represents a corrupted systematic codeword, where the parity symbols are at the end and the leftmost symbol is the most significant symbol.

In the Galois array decoded, each row represents the attempt at decoding the corresponding row in code. A decoding failure occurs if bchdec detects more than $t$ errors in a row of code, where $t$ is the number of correctable errors as reported by bchgenpoly. In the case of a decoding failure, bchdec forms the corresponding row of decoded by merely removing $n-k$ symbols from the end of the row of code.
decoded $=$ bchdec(...,paritypos) specifies whether the parity symbols in code were appended or prepended to the message in the coding operation. The string paritypos can be either 'end' or 'beginning'. The default is 'end'. If paritypos is 'beginning', then a decoding failure causes bchdec to remove $n-k$ symbols from the beginning rather than the end of the row.
[decoded, cnumerr] = bchdec(...) returns a column vector cnumerr, each element of which is the number of corrected errors in the corresponding row of code. A value of -1 in cnumerr indicates a decoding failure in that row in code.
[decoded, cnumerr, ccode] = bchdec(...) returns ccode, the corrected version of code. The Galois array ccode has the same format as code. If a decoding failure occurs in a certain row of code, the corresponding row in ccode contains that row unchanged.

## Results of Error Correction

BCH decoders correct up to a certain number of errors, specified by the user. If the input contains more errors than the decoder is meant to correct, the decoder will most likely not output the correct codeword.

The chance of a BCH decoder decoding a corrupted input to the correct codeword depends on the number of errors in the input and the number of errors the decoder is meant to correct.
For example, when a single-error-correcting BCH decoder is given input with two errors, it actually decodes it to a different codeword. When a double-error-correcting BCH decoder is given input with three errors, then it only sometimes decodes it to a valid codeword.

The following code illustrates this phenomenon for a single-error-correcting BCH decoder given input with two errors.

```
n = 63; k = 57;
s = RandStream('swb2712', 'Seed', 9973);
msg = gf(randi(s,[0 1],1,k));
code = bchenc(msg, n, k);
% Add 2 errors
cnumerr2 = zeros(nchoosek(n,2),1);
nErrs = zeros(nchoosek(n,2),1);
cnumerrIdx = 1;
for idx1 = 1 : n-1
    sprintf('idx1 for 2 errors = %d', idx1)
    for idx2 = idx1+1 : n
        errors = zeros(1,n);
        errors(idx1) = 1;
        errors(idx2) = 1;
        erroredCode = code + gf(errors);
        [decoded2, cnumerr2(cnumerrIdx)]...
            = bchdec(erroredCode, n, k);
        % If bchdec thinks it corrected only one error,
        % then encode the decoded message. Check that
        % the re-encoded message differs from the errored
        % message in only one coordinate.
```


## bchdec

```
        if cnumerr2(cnumerrIdx) == 1
            code2 = bchenc(decoded2, n, k);
                        nErrs(cnumerrIdx) = biterr(double(erroredCode.x),...
                double(code2.x));
            end
                cnumerrIdx = cnumerrIdx + 1;
        end
end
% Plot the computed number of errors, based on the difference
% between the double-errored codeword and the codeword that was
% re-encoded from the initial decoding.
plot(nErrs)
title(['Number of Actual Errors between Errored Codeword and' ...
    'Re-encoded Codeword'])
```

The resulting plot shows that all inputs with two errors are decoded to a codeword that differs in exactly one position.


## Examples

The script below encodes a (random) message, simulates the addition of noise to the code, and then decodes the message.

```
m = 4; n = 2^m-1; % Codeword length
k = 5; % Message length
nwords = 10; % Number of words to encode
msg = gf(randi([0 1],nwords,k));
% Find t, the error-correction capability.
[genpoly,t] = bchgenpoly(n,k);
% Define t2, the number of errors to add in this example.
t2 = t;
% Encode the message.
code = bchenc(msg,n,k);
% Corrupt up to t2 bits in each codeword.
noisycode = code + randerr(nwords,n,1:t2);
```


## bchdec

```
% Decode the noisy code.
[newmsg,err,ccode] = bchdec(noisycode,n,k);
if ccode==code
    disp('All errors were corrected.')
end
if newmsg==msg
    disp('The message was recovered perfectly.')
end
```

In this case, all errors are corrected and the message is recovered perfectly. However, if you change the definition of t2 to
$t 2=t+1$;
then some codewords will contain more than $t$ errors. This is too many errors, and some are not corrected.
AlgorithmsLimitationsReferencesSee Also
How To
bchdec uses the Berlekamp-Massey decoding algorithm. For information about this algorithm, see the works listed in "References" on page 1-28.

The maximum allowable value of n is 65535 .
[1] Wicker, Stephen B., Error Control Systems for Digital Communication and Storage, Upper Saddle River, NJ, Prentice Hall, 1995.
[2] Berlekamp, Elwyn R., Algebraic Coding Theory, New York, McGraw-Hill, 1968.

See Also bchenc \| bchgenpoly

- "Block Codes"


## Purpose

BCH encoder

## Syntax

code $=$ bchenc(msg, n,k)
code $=$ bchenc(..., paritypos)
Description
code $=$ bchenc (msg, $\mathrm{n}, \mathrm{k}$ ) encodes the message in msg using an $[\mathrm{n}, \mathrm{k}]$ BCH encoder with the narrow-sense generator polynomial. msg is a Galois array of symbols over GF(2). Each k-element row of msg represents a message word, where the leftmost symbol is the most significant symbol. Parity symbols are at the end of each word in the output Galois array code.
code $=$ bchenc(...,paritypos) specifies whether bchenc appends or prepends the parity symbols to the input message to form code. The string paritypos can be either 'end' or 'beginning'. The default is 'end'.

The tables below list valid $[n, k]$ pairs for small values of $n$, as well as the corresponding values of the error-correction capability, t .

| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :---: | :---: | :---: |
| 7 | 4 | 1 |


| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :--- | :---: | :--- |
| 15 | 11 | 1 |
| 15 | 7 | 2 |
| 15 | 5 | 3 |

## bchenc

| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :---: | :---: | :---: |
| 31 | 26 | 1 |
| 31 | 21 | 2 |
| 31 | 16 | 3 |
| 31 | 11 | 5 |
| 31 | 6 | 7 |


| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :--- | :---: | :---: |
| 63 | 57 | 1 |
| 63 | 51 | 2 |
| 63 | 45 | 3 |
| 63 | 39 | 4 |
| 63 | 36 | 5 |
| 63 | 30 | 6 |
| 63 | 24 | 7 |
| 63 | 18 | 10 |
| 63 | 16 | 11 |
| 63 | 10 | 13 |
| 63 | 7 | 15 |


| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :---: | :---: | :---: |
| 127 | 120 | 1 |
| 127 | 113 | 2 |


| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :---: | :---: | :---: |
| 127 | 106 | 3 |
| 127 | 99 | 4 |
| 127 | 92 | 5 |
| 127 | 85 | 6 |
| 127 | 78 | 7 |
| 127 | 71 | 9 |
| 127 | 64 | 10 |
| 127 | 57 | 11 |
| 127 | 50 | 13 |
| 127 | 43 | 14 |
| 127 | 36 | 15 |
| 127 | 29 | 21 |
| 127 | 22 | 23 |
| 127 | 15 | 27 |


| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :---: | :---: | :---: |
| 255 | 247 | 1 |
| 255 | 239 | 2 |
| 255 | 231 | 3 |
| 255 | 223 | 4 |
| 255 | 215 | 5 |
| 255 | 207 | 6 |

## bchenc

| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :---: | :---: | :---: |
| 255 | 199 | 7 |
| 255 | 191 | 8 |
| 255 | 187 | 9 |
| 255 | 179 | 10 |
| 255 | 171 | 11 |
| 255 | 163 | 12 |
| 255 | 155 | 13 |
| 255 | 147 | 14 |
| 255 | 139 | 15 |
| 255 | 131 | 18 |
| 255 | 123 | 19 |
| 255 | 115 | 21 |
| 255 | 107 | 22 |
| 255 | 99 | 23 |
| 255 | 91 | 25 |
| 255 | 87 | 26 |
| 255 | 79 | 27 |
| 255 | 71 | 29 |
| 255 | 63 | 30 |
| 255 | 55 | 31 |
| 255 | 47 | 42 |
| 255 | 45 | 43 |
| 255 | 29 | 45 |
| 255 |  | 55 |
| 255 | 21 |  |
|  |  |  |


| $\mathbf{n}$ | $\mathbf{k}$ | $\boldsymbol{t}$ |
| :---: | :---: | :---: |
| 255 | 13 | 59 |
| 255 | 9 | 63 |


| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :---: | :---: | :---: |
| 511 | 502 | 1 |
| 511 | 493 | 2 |
| 511 | 484 | 3 |
| 511 | 475 | 4 |
| 511 | 466 | 5 |
| 511 | 457 | 6 |
| 511 | 448 | 7 |
| 511 | 439 | 8 |
| 511 | 430 | 9 |
| 511 | 421 | 10 |
| 511 | 412 | 11 |
| 511 | 403 | 12 |
| 511 | 394 | 13 |
| 511 | 385 | 14 |
| 511 | 376 | 15 |
| 511 | 367 | 16 |
| 511 | 358 | 18 |
| 511 | 349 | 19 |
| 511 | 340 | 20 |

## bchenc

| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :---: | :---: | :---: |
| 511 | 331 | 21 |
| 511 | 322 | 22 |
| 511 | 313 | 23 |
| 511 | 304 | 25 |
| 511 | 295 | 26 |
| 511 | 286 | 27 |
| 511 | 277 | 28 |
| 511 | 268 | 29 |
| 511 | 259 | 30 |
| 511 | 250 | 31 |
| 511 | 241 | 36 |
| 511 | 238 | 37 |
| 511 | 229 | 38 |
| 511 | 220 | 39 |
| 511 | 211 | 41 |
| 511 | 202 | 42 |
| 511 | 193 | 43 |
| 511 | 184 | 45 |
| 511 | 175 | 46 |
| 511 | 166 | 47 |
| 511 | 157 | 51 |
| 511 | 148 | 53 |
| 511 | 139 | 55 |
| 511 | 130 | 58 |
| 511 |  |  |
|  |  |  |


| $\mathbf{n}$ | $\mathbf{k}$ | $\mathbf{t}$ |
| :---: | :---: | :---: |
| 511 | 112 | 59 |
| 511 | 103 | 61 |
| 511 | 94 | 62 |
| 511 | 85 | 63 |
| 511 | 76 | 85 |
| 511 | 67 | 87 |
| 511 | 58 | 91 |
| 511 | 49 | 93 |
| 511 | 40 | 95 |
| 511 | 31 | 109 |
| 511 | 28 | 111 |
| 11 | 19 | 119 |

Examples
See the example on the reference page for the function bchdec.
Limitations The maximum allowable value of n is 65535 .
See Also bchdec | bchgenpoly | bchnumerr
How To

- "Block Codes"


## bchgenpoly

```
Purpose Generator polynomial of BCH code
Syntax genpoly = bchgenpoly ( \(\mathrm{n}, \mathrm{k}\) )
genpoly = bchgenpoly(n,k,prim_poly)
genpoly = bchgenpoly(n,k,prim_poly,outputFormat)
[genpoly,t] = bchgenpoly(...)
```


## Description

genpoly = bchgenpoly( $\mathrm{n}, \mathrm{k}$ ) returns the narrow-sense generator polynomial of a BCH code with codeword length n and message length $k$. The codeword length $n$ must have the form $2^{\mathrm{m}}-1$ for some integer m between 3 and 16. The output genpoly is a Galois row vector that represents the coefficients of the generator polynomial in order of descending powers. The narrow-sense generator polynomial is LCM $\left[m \_1(\mathrm{x}), \mathrm{m} \_2(\mathrm{x}), \ldots, \mathrm{m} \_2 \mathrm{t}(\mathrm{x})\right]$, where:

- LCM represents the least common multiple,
- m_i(x) represents the minimum polynomial corresponding to $a^{i}, \alpha$ is a root of the default primitive polynomial for the field $G F(n+1)$,
- and t represents the error-correcting capability of the code.

Note Although the bchgenpoly function performs intermediate computations in $\mathrm{GF}(\mathrm{n}+1)$, the final polynomial has binary coefficients. The output from bchgenpoly is a Galois vector in GF(2) rather than in $G F(n+1)$.
genpoly = bchgenpoly(n,k,prim_poly) is the same as the syntax above, except that prim_poly specifies the primitive polynomial for $\mathrm{GF}(\mathrm{n}+1)$ that has Alpha as a root. prim_poly is an integer whose binary representation indicates the coefficients of the primitive polynomial in order of descending powers. To use the default primitive polynomial for GF( $n+1$ ), set prim_poly to [].
genpoly = bchgenpoly( $n, k$, prim_poly,outputFormat) is the same as the previous syntax, except that outputFormat specifies output data type. The value of outputFormat can be ' $g f$ ' or 'double' corresponding

```
to Galois field and double data types respectively. The default value of outputFormat is 'gf'.
[genpoly,t] = bchgenpoly(...) returns t, the error-correction
capability of the code.
```


## Examples

```
The results below show that a \([15,11] \mathrm{BCH}\) code can correct one error and has a generator polynomial \(\mathrm{X}^{4}+\mathrm{X}+1\).
m = 4;
\(\mathrm{n}=2^{\wedge} \mathrm{m}-1\); \% Codeword length
k = 11; \% Message length
\% Get generator polynomial and error-correction capability. [genpoly,t] = bchgenpoly( \(n, k\) )
The output is
genpoly \(=\) GF (2) array.
Array elements =
\(\begin{array}{lllll}1 & 0 & 0 & 1 & 1\end{array}\)
\(\mathrm{t}=\)
1
Limitations The maximum allowable value of n is 511 .
References [1] Peterson, W. Wesley, and E. J. Weldon, Jr., Error-Correcting Codes, 2nd ed., Cambridge, MA, MIT Press, 1972.
See Also bchenc | bchdec | bchnumerr
How To . "Block Codes"
```


## bchnumerr

Purpose Number of correctable errors for BCH code

```
Syntax
    T = bchnumerr(N)
    T = bchnumerr(N, K)
```

$\mathrm{T}=\mathrm{bchnumerr}(\mathrm{N})$ returns all the possible combinations of message length, $K$, and number of correctable errors, $t$, for a BCH code of codeword length, N . N must have the form $2^{\mathrm{m}}-1$ for some integer, m , between 3 and 16. T is a matrix with three columns. The first column lists N , the second column lists K , and the third column lists t .
$\mathrm{T}=\mathrm{bchnumerr}(\mathrm{N}, \mathrm{K})$ returns the number of correctable errors, t , for an ( $\mathrm{N}, \mathrm{K}$ ) BCH code.

## See Also bchenc | bchdec | bchgenpoly

## Purpose

Bit error rate (BER) for uncoded AWGN channels
Syntax

```
ber = berawgn(EbNo,'pam',M)
ber = berawgn(EbNo,'qam',M)
ber = berawgn(EbNo,'psk',M,dataenc)
ber = berawgn(EbNo,'oqpsk',dataenc)
ber = berawgn(EbNo,'dpsk',M)
ber = berawgn(EbNo,'fsk',M,coherence)
ber = berawgn(EbNo,'fsk',2,coherence,rho)
ber = berawgn(EbNo,'msk',precoding)
ber = berawgn(EbNo,'msk',precoding,coherence)
berlb = berawgn(EbNo,'cpfsk',M,modindex,kmin)
[BER,SER] = berawgn(EbNo, ...)
```


## Alternatives As an alternative to the berawgn function, invoke the BERTool GUI

 (bertool), and use the Theoretical tab.
## Description For All Syntaxes

The berawgn function returns the BER of various modulation schemes over an additive white Gaussian noise (AWGN) channel. The first input argument, EbNo, is the ratio of bit energy to noise power spectral density, in dB. If EbNo is a vector, the output ber is a vector of the same size, whose elements correspond to the different $\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}$ levels. The supported modulation schemes, which correspond to the second input argument to the function, are in the following table.

| Modulation Scheme | Second Input Argument |
| :--- | :--- |
| Phase shift keying (PSK) | 'psk' |
| Offset quaternary phase shift <br> keying (OQPSK) | 'oqpsk' |
| Differential phase shift keying <br> (DPSK) | 'dpsk' |
| Pulse amplitude modulation <br> (PAM) | 'pam' |


| Modulation Scheme | Second Input Argument |
| :--- | :--- |
| Quadrature amplitude <br> modulation (QAM) | 'qam' |
| Frequency shift keying (FSK) | 'fsk' |
| Minimum shift keying (MSK) | 'msk' |
| Continuous phase frequency shift <br> keying (CPFSK) | 'cpfsk' |

Most syntaxes also have an $M$ input that specifies the alphabet size for the modulation. M must have the form $2^{\mathrm{k}}$ for some positive integer k. For all cases, the function assumes the use of a Gray-coded signal constellation.

## For Specific Syntaxes

ber = berawgn(EbNo, 'pam' ${ }^{\text {M }}$ ) returns the BER of uncoded PAM over an AWGN channel with coherent demodulation.
ber = berawgn(EbNo, 'qam' ${ }^{\text {, M }}$ ) returns the BER of uncoded QAM over an AWGN channel with coherent demodulation. The alphabet size, M, must be at least 4 . When $k=\log _{2} M$ is odd, a rectangular constellation of size $M=I \times J$ is used, where $I=2^{\frac{k-1}{2}}$ and $J=2^{\frac{k+1}{2}}$. ber $=$ berawgn(EbNo, 'psk', M, dataenc) returns the BER of coherently detected uncoded PSK over an AWGN channel. dataenc is either 'diff' for differential data encoding or 'nondiff' for nondifferential data encoding. If dataenc is 'diff', M must be no greater than 4.
ber = berawgn(EbNo,'oqpsk',dataenc) returns the BER of coherently detected offset-QPSK over an uncoded AWGN channel.
ber = berawgn(EbNo,'dpsk', M) returns the BER of uncoded DPSK modulation over an AWGN channel.
ber $=$ berawgn(EbNo, 'fsk', m, coherence) returns the BER of orthogonal uncoded FSK modulation over an AWGN channel. coherence is either 'coherent' for coherent demodulation or
'noncoherent' for noncoherent demodulation. M must be no greater than 64 for 'noncoherent'.
ber $=$ berawgn(EbNo,'fsk',2, coherence, rho) returns the BER for binary nonorthogonal FSK over an uncoded AWGN channel, where rho is the complex correlation coefficient. See "Nonorthogonal 2-FSK with Coherent Detection" for the definition of the complex correlation coefficient and how to compute it for nonorthogonal BFSK.
ber = berawgn(EbNo, 'msk' , precoding) returns the BER of coherently detected MSK modulation over an uncoded AWGN channel. Setting precoding to 'off' returns results for conventional MSK while setting precoding to 'on' returns results for precoded MSK.
ber = berawgn(EbNo, 'msk', precoding, coherence) specifies whether the detection is coherent or noncoherent.
berlb = berawgn(EbNo,'cpfsk', M, modindex,kmin) returns a lower bound on the BER of uncoded CPFSK modulation over an AWGN channel. modindex is the modulation index, a positive real number. kmin is the number of paths having the minimum distance; if this number is unknown, you can assume a value of 1 .
[BER,SER] = berawgn(EbNo, ...) returns both the BER and SER.

## Limitations

References

The numerical accuracy of this function's output is limited by approximations related to the numerical implementation of the expressions.

You can generally rely on the first couple of significant digits of the function's output.
[1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg, Digital Phase Modulation, New York, Plenum Press, 1986.
[2] Cho, K., and Yoon, D., "On the general BER expression of one- and two-dimensional amplitude modulations", IEEE Trans. Commun., Vol. 50, Number 7, pp. 1074-1080, 2002.
[3] Lee, P. J., "Computation of the bit error rate of coherent M-ary PSK with Gray code bit mapping", IEEE Trans. Commun., Vol. COM-34, Number 5, pp. 488-491, 1986.
[4] Proakis, J. G., Digital Communications, 4th ed., McGraw-Hill, 2001.
[5] Simon, M. K, Hinedi, S. M., and Lindsey, W. C., Digital Communication Techniques - Signal Design and Detection, Prentice-Hall, 1995.
[6] Simon, M. K, "On the bit-error probability of differentially encoded QPSK and offset QPSK in the presence of carrier synchronization", IEEE Trans. Commun., Vol. 54, pp. 806-812, 2006.
[7] Lindsey, W. C., and Simon, M. K, Telecommunication Systems Engineering, Englewood Cliffs, N.J., Prentice-Hall, 1973.

See Also
bercoding | berfading | bersync
How To

- "Theoretical Results"
- Analytical Expressions Used in berawgn


## Purpose

Bit error rate (BER) for coded AWGN channels

```
Syntax
berub = bercoding(EbNo, 'conv',decision, coderate, dspec)
berub = bercoding(EbNo,'block','hard', n,k,dmin)
berub \(=\) bercoding(EbNo,'block','soft', \(n, k, d m i n)\)
berapprox = bercoding(EbNo,'Hamming', 'hard', n)
berub = bercoding(EbNo,'Golay','hard',24)
berapprox \(=\) bercoding (EbNo, 'RS', 'hard', \(n, k\) )
```

Alternatives

Description

As an alternative to the bercoding function, invoke the BERTool GUI (bertool) and use the Theoretical tab.
berub = bercoding(EbNo,'conv',decision,coderate, dspec) returns an upper bound or approximation on the BER of a binary convolutional code with coherent phase shift keying (PSK) modulation over an additive white Gaussian noise (AWGN) channel. EbNo is the ratio of bit energy to noise power spectral density, in dB. If EbNo is a vector, berub is a vector of the same size, whose elements correspond to the different $\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}$ levels. To specify hard-decision decoding, set decision to 'hard'; to specify soft-decision decoding, set decision to 'soft'. The convolutional code has code rate equal to coderate. The dspec input is a structure that contains information about the code's distance spectrum:

- dspec.dfree is the minimum free distance of the code.
- dspec.weight is the weight spectrum of the code.

To find distance spectra for some sample codes, use the distspec function or see [5] and [3].

Note The results for binary PSK and quaternary PSK modulation are the same. This function does not support M-ary PSK when M is other than 2 or 4 .
berub $=$ bercoding(EbNo,'block','hard', $\mathrm{n}, \mathrm{k}, \mathrm{dmin})$ returns an upper bound on the BER of an [ $n, k]$ binary block code with hard-decision decoding and coherent BPSK or QPSK modulation. dmin is the minimum distance of the code.
berub $=$ bercoding(EbNo, 'block', 'soft', $\mathrm{n}, \mathrm{k}, \mathrm{dmin})$ returns an upper bound on the BER of an [n,k] binary block code with soft-decision decoding and coherent BPSK or QPSK modulation. dmin is the minimum distance of the code.
berapprox = bercoding(EbNo, 'Hamming', 'hard', $n$ ) returns an approximation of the BER of a Hamming code using hard-decision decoding and coherent BPSK modulation. (For a Hamming code, if $n$ is known, then k can be computed directly from n .)
berub = bercoding(EbNo,'Golay','hard',24) returns an upper bound of the BER of a Golay code using hard-decision decoding and coherent BPSK modulation. Support for Golay currently is only for $\mathrm{n}=24$. In accordance with [3], the Golay coding upper bound assumes only the correction of 3 -error patterns. Even though it is theoretically possible to correct approximately $19 \%$ of 4 -error patterns, most decoders in practice do not have this capability.
berapprox $=$ bercoding(EbNo,'RS', 'hard', $n, k$ ) returns an approximation of the BER of ( $\mathrm{n}, \mathrm{k}$ ) Reed-Solomon code using hard-decision decoding and coherent BPSK modulation.

Examples An example using this function for a convolutional code is in "Plotting Theoretical Error Rates".

The following example finds an upper bound on the theoretical BER of a block code. It also uses the berfit function to perform curve fitting.

```
n = 23; k = 12; % Lengths of codewords and messages
dmin = 7; % Minimum distance
EbNo = 1:10;
ber_block = bercoding(EbNo,'block','hard',n,k,dmin);
berfit(EbNo,ber_block) % Plot BER points and fitted curve.
ylabel('Bit Error Probability');
```

title('BER Upper Bound vs. Eb/No, with Best Curve Fit');


## Limitations

## References

The numerical accuracy of this function's output is limited by

- Approximations in the analysis leading to the closed-form expressions that the function uses
- Approximations related to the numerical implementation of the expressions

You can generally rely on the first couple of significant digits of the function's output.
[1] Proakis, J. G., Digital Communications, 4th ed., New York, McGraw-Hill, 2001.

## bercoding

[2] Frenger, P., P. Orten, and T. Ottosson, "Convolutional Codes with Optimum Distance Spectrum," IEEE Communications Letters, Vol. 3, No. 11, Nov. 1999, pp. 317-319.
[3] Odenwalder, J. P., Error Control Coding Handbook, Final Report, LINKABIT Corporation, San Diego, CA, 1976.
[4] Sklar, B., Digital Communications, 2nd ed., Prentice Hall, 2001.
[5] Ziemer, R. E., and R. L., Peterson, Introduction to Digital Communication, 2nd ed., Prentice Hall, 2001.

See Also<br>berawgn | berfading | bersync | distspec<br>How To<br>- "Theoretical Performance Results"<br>- Analytical Expressions Used in bercoding and BERTool

## Purpose

Bit error rate (BER) and confidence interval of Monte Carlo simulation

## Syntax

Description

## Examples

```
[ber,interval] = berconfint(nerrs,ntrials)
```

[ber,interval] = berconfint(nerrs,ntrials,level)
[ber, interval] = berconfint(nerrs,ntrials) returns the error probability estimate ber and the $95 \%$ confidence interval interval for a Monte Carlo simulation of ntrials trials with nerrs errors. interval is a two-element vector that lists the endpoints of the interval. If the errors and trials are measured in bits, the error probability is the bit error rate (BER); if the errors and trials are measured in symbols, the error probability is the symbol error rate (SER).
[ber,interval] = berconfint(nerrs,ntrials,level) specifies the confidence level as a real number between 0 and 1 .

If a simulation of a communication system results in 100 bit errors in $10^{6}$ trials, the BER (bit error rate) for that simulation is the quotient $10^{-4}$. The command below finds the $95 \%$ confidence interval for the BER of the system.

```
nerrs = 100; % Number of bit errors in simulation
ntrials = 10^6; % Number of trials in simulation
level = 0.95; % Confidence level
[ber,interval] = berconfint(nerrs,ntrials,level)
```

The output below shows that, with $95 \%$ confidence, the BER for the system is between 0.0000814 and 0.0001216 .

```
ber =
```

    1.0000e-004
    interval =
1.0e-003 *

## berconfint

0.0814 ..... 0.1216
For an example that uses the output of berconfint to plot error bars on a BER plot, see "Example: Curve Fitting for an Error Rate Plot"
References [1] Jeruchim, Michel C., Philip Balaban, and K. Sam Shanmugan, Simulation of Communication Systems, Second Edition, New York, Kluwer Academic/Plenum, 2000.
See Also ..... binofit | mle

Purpose
Bit error rate (BER) for Rayleigh and Rician fading channels
Syntax

```
ber = berfading(EbNo,'pam',M,divorder)
ber = berfading(EbNo,'qam',M,divorder)
ber = berfading(EbNo,'psk',M,divorder)
ber = berfading(EbNo,'depsk',M,divorder)
ber = berfading(EbNo,'oqpsk',divorder)
ber = berfading(EbNo,'dpsk',M,divorder)
ber = berfading(EbNo,'fsk',M,divorder,coherence)
ber = berfading(EbNo,'fsk',2,divorder,coherence,rho)
ber = berfading(EbNo,...,K)
ber = berfading(EbNo,'psk',2,1,K,phaserr)
[BER,SER] = berfading(EbNo, ...)
```


## Alternatives As an alternative to the berfading function, invoke the BERTool GUI

 (bertool), and use the Theoretical tab.
## Description For All Syntaxes

The first input argument, EbNo, is the ratio of bit energy to noise power spectral density, in dB . If EbNo is a vector, the output ber is a vector of the same size, whose elements correspond to the different $\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}$ levels.
Most syntaxes also have an Minput that specifies the alphabet size for the modulation. $M$ must have the form $2^{k}$ for some positive integer $k$.
berfading uses expressions that assume Gray coding. If you use binary coding, the results may differ.

For cases where diversity is used, the $\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}$ on each diversity branch is EbNo/divorder, where divorder is the diversity order (the number of diversity branches) and is a positive integer.

## For Specific Syntaxes

ber = berfading(EbNo, 'pam' , M, divorder) returns the BER for PAM over an uncoded Rayleigh fading channel with coherent demodulation.
ber = berfading(EbNo, 'qam', M, divorder) returns the BER for QAM over an uncoded Rayleigh fading channel with coherent demodulation.

## berfading

The alphabet size, $M$, must be at least 4 . When $k=\log _{2} M$ is odd, a rectangular constellation of size $M=I \times J$ is used, where $I=2^{\frac{k-1}{2}}$ and $J=2^{\frac{k+1}{2}}$.
ber = berfading(EbNo,'psk', m, divorder) returns the BER for coherently detected PSK over an uncoded Rayleigh fading channel.
ber = berfading(EbNo, 'depsk', M, divorder) returns the BER for coherently detected PSK with differential data encoding over an uncoded Rayleigh fading channel. Only $M=2$ is currently supported. ber = berfading(EbNo, 'oqpsk', divorder) returns the BER of coherently detected offset-QPSK over an uncoded Rayleigh fading channel.
ber = berfading(EbNo,'dpsk', m,divorder) returns the BER for DPSK over an uncoded Rayleigh fading channel. For DPSK, it is assumed that the fading is slow enough that two consecutive symbols are affected by the same fading coefficient.
ber = berfading(EbNo,'fsk',M, divorder, coherence) returns the BER for orthogonal FSK over an uncoded Rayleigh fading channel. coherence should be 'coherent' for coherent detection, or 'noncoherent' for noncoherent detection.
ber = berfading(EbNo,'fsk',2,divorder, coherence, rho) returns the BER for binary nonorthogonal FSK over an uncoded Rayleigh fading channel. rho is the complex correlation coefficient. See "Nonorthogonal 2-FSK with Coherent Detection" for the definition of the complex correlation coefficient and how to compute it for nonorthogonal BFSK.
ber $=$ berfading(EbNo, ...,K) returns the BER over an uncoded Rician fading channel, where $K$ is the ratio of specular to diffuse energy in linear scale. For the case of 'fsk', rho must be specified before K. ber $=$ berfading(EbNo, 'psk',2,1,K, phaserr) returns the BER of BPSK over an uncoded Rician fading channel with imperfect phase
synchronization. phaserr is the standard deviation of the reference carrier phase error in radians.
[BER,SER] = berfading(EbNo, ...) returns both the BER and SER.

## Examples

The following example computes and plots the BER for uncoded DQPSK (differential quaternary phase shift keying) modulation over a flat Rayleigh fading channel for several diversity order values.

```
EbNo = 8:2:20;
M = 16; % Use 16 QAM
L = 1; % Start without diversity
ber = berfading(EbNo,'qam',M,L);
semilogy(EbNo,ber);
text(18.5, 0.02, sprintf('L=%d', L))
hold on
% Loop over diversity order, L, 2 to 20
for L=2:20
    ber = berfading(EbNo,'qam',M,L);
    semilogy(EbNo,ber);
end
text(18.5, 1e-11, sprintf('L=%d', L))
title('QAM over fading channel with diversity order 1 to 20')
xlabel('E_b/N_o (dB)')
ylabel('BER')
grid on
```


## berfading



Limitations

References

The numerical accuracy of this function's output is limited by approximations related to the numerical implementation of the expressions

You can generally rely on the first couple of significant digits of the function's output.
[1] Proakis, John G., Digital Communications, 4th ed., New York, McGraw-Hill, 2001.
[2] Modestino, James W., and Mui, Shou Y., Convolutional code performance in the Rician fading channel, IEEE Trans. Commun., 1976.
[3] Cho, K., and Yoon, D., "On the general BER expression of one- and two-dimensional amplitude modulations", IEEE Trans. Commun., Vol. 50, Number 7, pp. 1074-1080, 2002.
[4] Lee, P. J., "Computation of the bit error rate of coherent M-ary PSK with Gray code bit mapping", IEEE Trans. Commun., Vol. COM-34, Number 5, pp. 488-491, 1986.
[5] Lindsey, W. C., "Error probabilities for Rician fading multichannel reception of binary and N-ary signals", IEEE Trans. Inform. Theory, Vol. IT-10, pp. 339-350, 1964.
[6] Simon, M. K , Hinedi, S. M., and Lindsey, W. C., Digital Communication Techniques - Signal Design and Detection, Prentice-Hall, 1995.
[7] Simon, M. K., and Alouini, M. S., Digital Communication over Fading Channels - A Unified Approach to Performance Analysis, 1st ed., Wiley, 2000.
[8] Simon, M. K , "On the bit-error probability of differentially encoded QPSK and offset QPSK in the presence of carrier synchronization", IEEE Trans. Commun., Vol. 54, pp. 806-812, 2006.

See Also<br>berawgn | bercoding | bersync<br>How To<br>- "Theoretical Performance Results"<br>- Analytical Expressions Used in berfading

```
Purpose Fit curve to nonsmooth empirical bit error rate (BER) data
Syntax fitber = berfit(empEbNo,empber)
fitber = berfit(empEbNo,empber,fitEbNo)
fitber = berfit(empEbNo,empber,fitEbNo,options)
fitber = berfit(empEbNo,empber,fitEbNo,options,fittype)
[fitber,fitprops] = berfit(...)
berfit(...)
berfit(empEbNo,empber,fitEbNo,options,'all')
Description
fitber = berfit(empEbNo,empber) fits a curve to the empirical BER data in the vector empber and returns a vector of fitted bit error rate (BER) points. The values in empber and fitber correspond to the \(\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}\) values, in dB , given by empEbNo. The vector empEbNo must be in ascending order and must have at least four elements.
```

Note The berfit function is intended for curve fitting or interpolation, not extrapolation. Extrapolating BER data beyond an order of magnitude below the smallest empirical BER value is inherently unreliable.
fitber = berfit(empEbNo, empber,fitEbNo) fits a curve to the empirical BER data in the vector empber corresponding to the $\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}$ values, in $d B$, given by empEbNo. The function then evaluates the curve at the $\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}$ values, in dB , given by fitEbNo and returns the fitted BER points. The length of fitEbNo must equal or exceed that of empEbNo.
fitber = berfit(empEbNo,empber,fitEbNo,options) uses the structure options to override the default options used for optimization. These options are the ones used by the fminsearch function. You can create the options structure using the optimset function. Particularly relevant fields are described in the table below.

| Field | Description |
| :--- | :--- |
| options.Display | Level of display: 'off' (default) <br> displays no output; 'iter' <br> displays output at each iteration; <br> 'final' displays only the final <br> output; 'notify' displays output <br> only if the function does not <br> converge. |
| options.MaxFunEvals | Maximum number of function <br> evaluations before optimization <br> ceases. The default is $10^{4}$. |
| options.MaxIter | Maximum number of iterations <br> before optimization ceases. The <br> default is $10^{4}$. |
| options. TolFun | Termination tolerance on the <br> closed-form function used to <br> generate the fit. The default is <br> $10^{-4}$. |
| options. TolX | Termination tolerance on <br> the coefficient values of the <br> closed-form function used to <br> generate the fit. The default is <br> $10^{-4}$. |

fitber = berfit(empEbNo,empber,fitEbNo,options,fittype) specifies which closed-form function berfit uses to fit the empirical data, from the possible fits listed in "Algorithms" on page 1-57 below. fittype can be 'exp', 'exp+const', 'polyRatio', or 'doubleExp+const'. To avoid overriding default optimization options, use options = [].
[fitber,fitprops] = berfit(...) returns the MATLAB structure fitprops, which describes the results of the curve fit. Its fields are described in the table below.

## berfit

| Field | Description |
| :--- | :--- |
| fitprops.fitType | The closed-form function type <br> used to generate the fit: 'exp', <br> 'exp+const', 'polyRatio', or <br> 'doubleExp+const'. |
| fitprops.coeffs | The coefficients used to generate <br> the fit. If the function cannot <br> find a valid fit, fitprops.coeffs <br> is an empty vector. |
| fitprops.sumSqErr | The sum squared error between <br> the log of the fitted BER points <br> and the log of the empirical BER <br> points. |
| fitprops.exitState | The exit condition of berfit: <br> 'The curve fit converged <br> to a solution. ', The <br> maximum number of function <br> evaluations was exceeded. ', <br> or 'No desirable fit was <br> found'. |
| fitprops.funcCount | The number of function <br> evaluations used in minimizing <br> the sum squared error function. |
| fitprops.iterations | The number of iterations taken <br> in minimizing the sum squared <br> error function. This is not <br> necessarily equal to the number <br> of function evaluations. |

berfit(...) plots the empirical and fitted BER data.
berfit(empEbNo,empber, fitEbNo, options, 'all') plots the empirical and fitted BER data from all the possible fits, listed in the "Algorithms"
on page 1-57 below, that return a valid fit. To avoid overriding default options, use options = [].

Note A valid fit must be

- real-valued
- monotonically decreasing
- greater than or equal to 0 and less than or equal to 0.5

If a fit does not confirm to this criteria, it is rejected.

## Algorithms

The berfit function fits the BER data using unconstrained nonlinear optimization via the fminsearch function. The closed-form functions that berfit considers are listed in the table below, where $x$ is the $\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}$ in linear terms (not dB) and $f$ is the estimated BER. These functions were empirically found to provide close fits in a wide variety of situations, including exponentially decaying BERs, linearly varying BERs, and BER curves with error rate floors.

| Value of fittype | Functional Expression |
| :--- | :--- |
| 'exp' $^{\prime}$ | $f(x)=\frac{a_{1} \exp \left\{-\left[\left(x-a_{2}\right)^{a_{3}}\right]\right\}}{a_{4}}$ |
| 'exp+const' | $f(x)=\frac{a_{1} \exp \left[-\left(x-a_{2}\right)^{a_{3}}\right]}{a_{4}}+a_{5}$ |
|  |  |

## berfit

| Value of fittype | Functional Expression |
| :--- | :---: |
| 'polyRatio' | $f(x)=\frac{a_{1} x^{2}+a_{2} x+a_{3}}{x^{3}+a_{4} x^{2}+a_{5} x+a_{6}}$ |
| 'doubleExp+const' | $\frac{a_{1} \exp \left[-\left(x-a_{2}\right)^{a_{3}}\right]}{a_{4}}$ |
|  | $+\frac{a_{5} \exp \left[-\left(x-a_{6}\right)^{a_{7}}\right]}{a_{8}}+a_{9}$ |

The sum squared error function that fminsearch attempts to minimize is

$$
F=\sum[\log (\text { empirical BER })-\log (\text { fitted BER })]^{2}
$$

where the fitted BER points are the values in fitber and the sum is over the $\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}$ points given in empEbNo. It is important to use the log of the BER values rather than the BER values themselves so that the high-BER regions do not dominate the objective function inappropriately.

## Examples

The examples below illustrate the syntax of the function, but they use hard-coded or theoretical BER data for simplicity. For an example that uses empirical BER data from a simulation, see "Example: Curve Fitting for an Error Rate Plot".
The code below plots the best fit for a sample set of data.

```
EbNo = 0:13;
berdata = [.2 . 15 . 13 . 12 . 08 . 09 . 08 . 07 . 06 . 04 . 03 . 02 . 01 .004];
berfit(EbNo,berdata); % Plot the best fit.
```



The curve connects the points created by evaluating the fit expression at the values in EbNo. To make the curve look smoother, use a syntax like berfit(EbNo,berdata, $0: 0.2: 13]$ ). This alternative syntax uses more points when plotting the curve, but it does not change the fit expression.

The next example demonstrates a fit for a BER curve with an error floor. We generate the empirical BER array by simulating a channel with a null $\left(\mathrm{ch}=\left[\begin{array}{ll}0.5 & 0.47\end{array}\right]\right)$ with BPSK modulation and linear MMSE equalizer at the receiver. We run the berfit with the 'all' option. The 'doubleExp+const' fit does not provide a valid fit, and the 'exp' fit type does not work well for this data. The 'exp+const' and 'polyRatio' fits closely match the simulated data.

```
EbNo = -10:3:15;
empBER = [0.3361 0.3076 0.2470 0.1878 0.1212 0.0845 0.0650 0.0540 0.0474];
```

figure; berfit(EbNo, empBER, [], [], 'all');


The following code illustrates the use of the options input structure as well as the fitprops output structure. The 'notify' value for the display level causes the function to produce output when one of the attempted fits does not converge. The exitState field of the output structure also indicates which fit converges and which fit does not.

```
M = 4; EbNo = 3:10;
berdata = berfading(EbNo,'psk',M,2); % Compute theoretical BER.
noisydata = berdata.*[.93 . 92 1 . 59 . 08 . 15 .01 .01];
% Say when fit fails to converge.
options = optimset('display','notify');
disp('*** Trying exponential fit.') % Poor fit
[fitber1,fitprops1] = berfit(EbNo,noisydata,EbNo,...
```

```
    options,'exp')
disp('*** Trying polynomial ratio fit.') % Good fit
[fitber2,fitprops2] = berfit(EbNo,noisydata,EbNo,...
    options,'polyRatio')
```

The output is as follows:

```
*** Trying exponential fit.
```

Exiting: Maximum number of function evaluations has been exceeded - increase MaxFunEvals option. Current function value: 2.729948

## fitber1 =

$0.0766 \quad 0.0423$
0.0205
0.0086
0.0030
0.0009
0.0002
0.0000
fitprops1 =
fitType: 'exp'
coeffs: [4×1 double]
sumSqErr: 2.7299
exitState: 'The maximum number of function evaluations... has been exceeded'
funcCount: 10000
iterations: 6177
*** Trying polynomial ratio fit.
fitber2 =
$\begin{array}{llllllll}0.0931 & 0.0476 & 0.0220 & 0.0090 & 0.0031 & 0.0008 & 0.0001 & 0.0001\end{array}$

## berfit

```
fitprops2 =
    fitType: 'polyRatio'
        coeffs: [6x1 double]
        sumSqErr: 2.0578
        exitState: 'The curve fit converged to a solution'
        funcCount: 580
        iterations: 344
```

    References For a general description of unconstrained nonlinear optimization, see
        the following work.
    [1] Chapra, Steven C., and Raymond P. Canale, Numerical Methods for Engineers, Fourth Edition, New York, McGraw-Hill, 2002.

## See Also <br> fminsearch | optimset

## Purpose

Bit error rate (BER) for imperfect synchronization
Syntax
ber $=$ bersync(EbNo,timerr,'timing')
ber = bersync(EbNo,phaserr,'carrier')

## Examples The code below computes the BER of coherent BPSK modulation over an

 AWGN channel with imperfect timing. The example varies both EbNo and timerr. (When timerr assumes the final value of zero, the bersync command produces the same result as berawgn(EbNo, 'psk',2).)```
EbNo = [4 8 12];
timerr = [0.2 0.07 0];
ber = zeros(length(timerr), length(EbNo));
for ii = 1:length(timerr)
    ber(ii,:) = bersync(EbNo, timerr(ii),'timerr');
end
% Display result using scientific notation.
format short e; ber
```

```
format; % Switch back to default notation format.
```

The output is below, where each row corresponds to a different value of timerr and each column corresponds to a different value of EbNo.

```
ber =
    5.2073e-002 2.0536e-002 1.1160e-002
    1.8948e-002 7.9757e-004 4.9008e-006
    1.2501e-002 1.9091e-004 9.0060e-009
```


## Limitations

## Algorithms

This function uses formulas from [3].
When the last input is 'timing', the function computes

$$
\frac{1}{4 \pi \sigma} \int_{-\infty}^{\infty} \exp \left(-\frac{\xi^{2}}{2 \sigma^{2}}\right) \int_{\sqrt{2 R}(1-2|\xi|)}^{\infty} \exp \left(-\frac{x^{2}}{2}\right) d x d \xi+\frac{1}{2 \sqrt{2 \pi}} \int_{\sqrt{2 R}}^{\infty} \exp \left(-\frac{x^{2}}{2}\right) d x
$$

where $\sigma$ is the timerr input and $R$ is the value of EbNo converted from dB to a linear scale.

When the last input is 'carrier', the function computes

$$
\frac{1}{\pi \sigma} \int_{0}^{\infty} \exp \left(-\frac{\phi^{2}}{2 \sigma^{2}}\right) \int_{\sqrt{2 R} \cos \phi}^{\infty} \exp \left(-\frac{y^{2}}{2}\right) d y d \phi
$$

where $\sigma$ is the phaserr input and $R$ is the value of EbNo converted from dB to a linear scale.

References [1] Jeruchim, Michel C., Philip Balaban, and K. Sam Shanmugan, Simulation of Communication Systems, Second Edition, New York, Kluwer Academic/Plenum, 2000.<br>[2] Sklar, Bernard, Digital Communications: Fundamentals and Applications, Second Edition, Upper Saddle River, NJ, Prentice-Hall, 2001.<br>[3] Stiffler, J. J., Theory of Synchronous Communications, Englewood Cliffs, NJ, Prentice-Hall, 1971.

## See Also berawgn | bercoding | berfading

How To . "Theoretical Results"

## bertool

Purpose Open bit error rate analysis GUI (BERTool)

## Syntax <br> bertool

Description bertool launches the Bit Error Rate Analysis Tool (BERTool). BERTool is a graphical user interface (GUI) that enables you to analyze BER performance of communications systems. Performance analysis is done via simulation-based, semianalytic, or theoretical approach. See "BERTool" to learn more.

## Purpose Convert binary vectors to decimal numbers

Syntax

d = bi2de(b)
d = bi2de(b,flg)
$d=b i 2 d e(b, p)$
d $=$ bi2de(b, p,flg)
$d=$ bi2de (b) converts a binary row vector $b$ to a nonnegative decimal integer. If $b$ is a matrix, each row is interpreted separately as a binary number. In this case, the output $d$ is a column vector, each element of which is the decimal representation of the corresponding row of $b$.

Note By default, bi2de interprets the first column of $b$ as the lowest-order digit.
$\mathrm{d}=\mathrm{bi} 2 \mathrm{de}(\mathrm{b}, f l g)$ is the same as the syntax above, except that $f l g$ is a string that determines whether the first column of $b$ contains the lowest-order or highest-order digits. Possible values for $f l g$ are 'right-msb' and 'left-msb'. The value 'right-msb' produces the default behavior.
$d=b i 2 d e(b, p)$ converts a base-p row vector $b$ to a nonnegative decimal integer, where $p$ is an integer greater than or equal to 2 . The first column of b is the lowest base-p digit. If b is a matrix, the output $d$ is a nonnegative decimal vector, each row of which is the decimal form of the corresponding row of $b$.
$\mathrm{d}=\mathrm{bi} 2 \mathrm{de}(\mathrm{b}, \mathrm{p}, f 1 \mathrm{l})$ is the same as the syntax above, except that $f l g$ is a string that determines whether the first column of $b$ contains the lowest-order or highest-order digits. Possible values for $f l g$ are 'right-msb' and 'left-msb'. The value 'right-msb' produces the default behavior.

Examples Generate a matrix that contains binary representations of five random numbers between 0 and 15, and then convert all five numbers to decimal integers.

```
b = randi([0 1],5,4); % Generate a 5-by-4 random binary matrix.
de = bi2de(b);
disp(' Dec Binary')
disp(' ----- -----------------')
disp([de, b])
```

Sample output is below. Your results might vary because the numbers are random.

| Dec |  | Binary |  |  |
| :---: | :---: | :---: | :---: | :---: |
| 13 | 1 | 0 | 1 | 1 |
| 7 | 1 | 1 | 1 | 0 |
| 15 | 1 | 1 | 1 | 1 |
| 4 | 0 | 0 | 1 | 0 |
| 9 | 1 | 0 | 0 | 1 |

Convert a base-five number into its decimal counterpart, using the leftmost base-five digit (4 in this case) as the most significant digit. This example reflects the fact that $4\left(5^{3}\right)+2\left(5^{2}\right)+5^{0}=551$.
d = bi2de([4 2001$\left.], 5, ' l e f t-m s b^{\prime}\right)$
The output is
d $=$
551

## See Also

de2bi

## bin2gray

## Purpose

Convert positive integers into corresponding Gray-encoded integers
Syntax
$y=$ bin2gray(x, modulation, $M$ )
[y,map] = bin2gray(x,modulation, M)
$y=$ bin2gray ( $x$, modulation,$M$ ) generates a Gray-encoded vector or matrix output $y$ with the same dimensions as its input parameter $x . x$ can be a scalar, vector, or matrix. modulation is the modulation type and must be a string equal to 'qam', 'pam', 'fsk', 'dpsk', or 'psk'. M is the modulation order that can be an integer power of 2 .
[y,map] = bin2gray(x,modulation,M) generates a Gray-encoded output $y$ with its respective Gray-encoded constellation map, map.

You can use map output to label a Gray-encoded constellation. The map output gives the Gray encoded labels for the corresponding modulation. See the example below.

Note If you are converting binary coded data to Gray-coded data and modulating the result immediately afterwards, you should use the appropriate modulation object or function with the 'Gray ' option, instead of BIN2GRAY.

## Examples

Gray encode a vector $x$ with a 16-QAM Gray encoded constellation and plot its map.

```
% To Gray encode a vector x with a 16-QAM Gray encoded
% constellation and return its map, use:
x=randi([0 15],1,100);
[y,map] = bin2gray(x,'qam',16);
% Obtain the symbols for 16-QAM
hMod = modem.qammod('M', 16);
symbols = hMod.Constellation;
% Plot the constellation
scatterplot(symbols);
set(get(gca,'Children'),'Marker','d','MarkerFaceColor',...
```


## bin2gray

```
'auto'); hold on;
% Label the constellation points according
% to the Gray mapping
for jj=1:16
text(real(symbols(jj))-0.15,imag(symbols(jj))+0.15,\ldots
dec2base(map(jj),2,4));
end
set(gca,'yTick',(-4:2:4),'xTick',(-4:2:4),...
'XLim',[-4 4],'YLim',...
[-4 4],'Box','on','YGrid','on', 'XGrid','on');
```

The example code generates the following plot, which shows the 16 QAM constellation with Gray-encoded labeling.


## See Also

gray2bin

How To - Gray Encoding a Modulated Signal

## biterr

Purpose Compute number of bit errors and bit error rate (BER)

```
Syntax
[number,ratio] = biterr(x,y)
[number, ratio] = biterr(x,y,k)
[number, ratio] = biterr(x,y,k,flg)
[number, ratio,individual] = biterr(...)
```


## Description

## For All Syntaxes

The biterr function compares unsigned binary representations of elements in $x$ with those in $y$. The schematics below illustrate how the shapes of x and y determine which elements biterr compares.


Each element of $x$ and $y$ must be a nonnegative decimal integer; biterr converts each element into its natural unsigned binary representation. number is a scalar or vector that indicates the number of bits that differ. ratio is number divided by the total number of bits. The total number of bits, the size of number, and the elements that biterr compares are determined by the dimensions of $x$ and $y$ and by the optional parameters.

## For Specific Syntaxes

[number, ratio] $=$ biterr $(x, y)$ compares the elements in $x$ and $y$. If the largest among all elements of $x$ and $y$ has exactly $k$ bits in its simplest binary representation, the total number of bits is k times the number of entries in the smaller input. The sizes of x and y determine which elements are compared:

- If $x$ and $y$ are matrices of the same dimensions, then biterr compares $x$ and $y$ element by element. number is a scalar. See schematic (a) in the preceding figure.
- If one is a row (respectively, column) vector and the other is a two-dimensional matrix, then biterr compares the vector element by element with each row (resp., column) of the matrix. The length of the vector must equal the number of columns (resp., rows) in the matrix. number is a column (resp., row) vector whose mth entry indicates the number of bits that differ when comparing the vector with the mth row (resp., column) of the matrix. See schematics (b) and (c) in the figure.
[number, ratio] = biterr $(x, y, k)$ is the same as the first syntax, except that it considers each entry in $x$ and $y$ to have $k$ bits. The total number of bits is $k$ times the number of entries of the smaller of $x$ and $y$. An error occurs if the binary representation of an element of $x$ or $y$ would require more than k digits.
[number, ratio] = biterr $(x, y, k, f l g)$ is similar to the previous syntaxes, except that $f l g$ can override the defaults that govern which elements biterr compares and how biterr computes the outputs. The possible values of $f l g$ are 'row-wise', 'column-wise', and 'overall'. The table below describes the differences that result from various combinations of inputs. As always, ratio is number divided by the total number of bits. If you do not provide $k$ as an input argument, the function defines it internally as the number of bits in the simplest binary representation of the largest among all elements of $x$ and $y$.


## biterr

## Comparing a Two-Dimensional Matrix $x$ with Another Input y

| Shape of y | flg | Type of <br> Comparison | number | Total <br> Number <br> of Bits |
| :--- | :--- | :--- | :--- | :--- |
| 2-D matrix | 'overall' <br> (default) | Element by <br> element | Total <br> number <br> of bit <br> errors | k times <br> number of <br> entries of y |
|  | 'row-wise' | mth row of $x$ <br> vs. mth row <br> of $y$ | Column <br> vector <br> whose <br> entries <br> count bit <br> errors in <br> each row | k times <br> number of <br> entries of y |
|  | 'column-wise' | mth column <br> of x vs. mth <br> column of $y$ | Row <br> vector <br> whose <br> entries <br> count bit <br> errors <br> in each <br> column | k times <br> number of <br> entries of $y$ |

## Comparing a Two-Dimensional Matrix x with Another Input y (Continued)

| Shape of $y$ | flg | Type of Comparison | number | Total Number of Bits |
| :---: | :---: | :---: | :---: | :---: |
| Row vector | 'overall' | $y$ vs. each row of $x$ | Total number of bit errors | k times number of entries of $x$ |
|  | 'row-wise' <br> (default) | $y$ vs. each row of $x$ | Column vector whose entries count bit errors in each row of $x$ | k times size of $y$ |
| Column vector | 'overall' | y vs. each column of $x$ | Total number of bit errors | k times number of entries of $x$ |
|  | 'column-wise' (default) | $y$ vs. each column of $x$ | Row <br> vector <br> whose entries count bit errors in each column of x | k times size of $y$ |

[number, ratio,individual] = biterr(...) returns a matrix individual whose dimensions are those of the larger of $x$ and $y$. Each

## biterr

entry of individual corresponds to a comparison between a pair of elements of $x$ and $y$, and specifies the number of bits by which the elements in the pair differ.

## Examples

## Example 1

The commands below compare the column vector $[0 ; 0 ; 0]$ to each column of a random binary matrix. The output is the number, proportion, and locations of 1 s in the matrix. In this case, individual is the same as the random matrix.

```
format rat;
[number,ratio,individual] = biterr([0;0;0],randi([0 1],3,5))
The output is
number =
    2 0 0 % 3
ratio =
    2/3
                                0
                                0
                                1
individual =
\begin{tabular}{lllll}
1 & 0 & 0 & 1 & 0 \\
1 & 0 & 0 & 1 & 0 \\
0 & 0 & 0 & 1 & 1
\end{tabular}
```


## Example 2

The commands below illustrate the use of $f l g$ to override the default row-by-row comparison. number and ratio are scalars, and individual has the same dimensions as the larger of the first two arguments of biterr.

```
format rat;
[number2,ratio2,individual2] = biterr([1 2; 3 4],[1 3],3,'overall')
The output is
```

```
number2 =
```

number2 =
5
ratio2 =
5/12
individual2 =
0 1
1 3

```

\section*{Example 3}
```

The script below adds errors to $10 \%$ of the elements in a matrix. Each entry in the matrix is a two-bit number in decimal form. The script computes the bit error rate using biterr and the symbol error rate using symerr.

```
```

x = randi([0 3],100); % Original signal

```
x = randi([0 3],100); % Original signal
% Create errors to add to ten percent of the elements of x.
% Create errors to add to ten percent of the elements of x.
% Errors can be either 1, 2, or 3 (not zero).
% Errors can be either 1, 2, or 3 (not zero).
errorplace = (rand(100,100) > .9); % Where to put errors
errorplace = (rand(100,100) > .9); % Where to put errors
errorvalue = randi(3,100); % Value of the errors
errorvalue = randi(3,100); % Value of the errors
errors = errorplace.*errorvalue;
errors = errorplace.*errorvalue;
y = rem(x+errors,4); % Signal with errors added, mod 4
y = rem(x+errors,4); % Signal with errors added, mod 4
format short
format short
[num_bit,ratio_bit] = biterr(x,y,2)
[num_bit,ratio_bit] = biterr(x,y,2)
[num_sym,ratio_sym] = symerr(x,y)
```

[num_sym,ratio_sym] = symerr(x,y)

```

\section*{biterr}
```

Sample output is below. ratio_sym is close to the target value of 0.10 . Your results might vary because the example uses random numbers.
num_bit =
1 3 0 4
ratio_bit =
0.0652
num_sym =
981
ratio_sym =
0.0981

```

\section*{Example 4}

The following example uses logical input arguments.
```

SNR = 3; frameLen = 100;
x = randi([0 1], frameLen, 1);
y = awgn(2*x-1, SNR);
z = y > 0;
biterr(x, z)

```

\section*{Example 5}

The following example uses logical input arguments.
```

SNR = 5; frameLen = 100;
x = rand(100, 1) > 0.5;
y = awgn(2*x-1, SNR);

```
\[
\begin{aligned}
& z=y>0 ; \\
& \operatorname{biterr}(x, z)
\end{aligned}
\]

See Also symerr

Purpose Model binary symmetric channel
Syntax
Description
ndata \(=\) bsc(data, \(p\) )
ndata \(=\) bsc(data, \(p, s)\)
ndata = bsc(data, p,state)
[ndata,err] = bsc(...)
ndata \(=\mathrm{bsc}(\) data, p\()\) passes the binary input signal data through a binary symmetric channel with error probability p. The channel introduces a bit error with probability \(p\), processing each element of data independently. data must be an array of binary numbers or a Galois array in GF(2). p must be a scalar between 0 and 1.ndata \(=\) bsc (data, \(p, s\) ) causes rand to use the random stream \(s\). s is any valid random stream. See RandStream for more details.
ndata \(=\) bsc (data, \(p\), state) resets the state of the uniform random number generator rand to the integer state.

Note This usage is deprecated and may be removed in a future release. Instead of state, use s, as in the previous example.
[ndata,err] = bsc(...) returns an array, err, containing the channel errors.

This function uses, by default, the Mersenne Twister algorithm by Nishimura and Matsumoto.

Note Using the state parameter causes this function to switch random generators to use the 'state' algorithm of the rand function.

See rand for details on the generator algorithm.
Examples
pcterrs =0.1509Another example using this function is in "Design a Rate 2/3Feedforward Encoder Using Simulink \({ }^{\mathbb{®}}\) ".
See Also rand | awgn
How To - "Binary Symmetric Channels"
\begin{tabular}{ll} 
Purpose & Construct constant modulus algorithm (CMA) object \\
Syntax & \begin{tabular}{l} 
alg = cma(stepsize) \\
alg \(=\) cma(stepsize, leakagefactor)
\end{tabular} \\
Description & \begin{tabular}{l} 
The cma function creates an adaptive algorithm object that you can use \\
with the lineareq function or dfe function to create an equalizer object. \\
You can then use the equalizer object with the equalize function to \\
equalize a signal. To learn more about the process for equalizing a \\
signal, see "Adaptive Algorithms".
\end{tabular}
\end{tabular}

Note After you use either lineareq or dfe to create a CMA equalizer object, you should initialize the equalizer object's Weights property with a nonzero vector. Typically, CMA is used with differential modulation; otherwise, the initial weights are very important. A typical vector of initial weights has a 1 corresponding to the center tap and 0 s elsewhere.
alg = cma(stepsize) constructs an adaptive algorithm object based on the constant modulus algorithm (CMA) with a step size of stepsize.
alg = cma(stepsize, leakagefactor) sets the leakage factor of the CMA. leakagefactor must be between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, while a value of 0 corresponds to a memoryless update algorithm.

\section*{Properties}

The table below describes the properties of the CMA adaptive algorithm object. To learn how to view or change the values of an adaptive algorithm object, see "Access Properties of an Adaptive Algorithm".
\begin{tabular}{l|l}
\hline Property & Description \\
\hline AlgType & \begin{tabular}{l} 
Fixed value, 'Constant \\
Modulus'
\end{tabular} \\
\hline StepSize & \begin{tabular}{l} 
CMA step size parameter, a \\
nonnegative real number
\end{tabular} \\
\hline LeakageFactor & \begin{tabular}{l} 
CMA leakage factor, a real \\
number between 0 and 1
\end{tabular} \\
\hline
\end{tabular}

\footnotetext{
Algorithms
Referenceslms | signlms | normlms | varlms | rls | lineareq | dfe | equalize
How To

- "Equalization"

Referring to the schematics in "Equalizer Structure", define \(w\) as the vector of all weights \(w_{\mathrm{i}}\) and define \(u\) as the vector of all inputs \(u_{\mathrm{i}}\). Based on the current set of weights, \(w\), this adaptive algorithm creates the new set of weights given by
\[
\text { (LeakageFactor) w + (StepSize) } \mathrm{u}^{*} \mathrm{e}
\]
where the * operator denotes the complex conjugate.
[1] Haykin, Simon, Adaptive Filter Theory, Third Ed., Upper Saddle River, NJ, Prentice-Hall, 1996.
[2] Johnson, Richard C., Jr., Philip Schniter, Thomas. J. Endres, et al., "Blind Equalization Using the Constant Modulus Criterion: A Review," Proceedings of the IEEE, Vol. 86, October 1998, pp. 1927-1950.

\section*{See Also \\ See Also}
}

Purpose Library link information for Communications System Toolbox blocks
Syntax \begin{tabular}{l} 
comm_links \\
comm_links(sys) \\
comm_links(sys,color)
\end{tabular}

\section*{Description}

See Also
comm_links returns a structure with two elements. Each element contains a cell array of strings containing names of library blocks in the current system. The blocks are grouped into two categories: obsolete and current. Blocks at all levels of the model are analyzed.
comm_links(sys) works as above on the named system sys, instead of the current system.
comm_links(sys,color) additionally colors all obsolete blocks according to the specified color. color is one of the following strings: 'blue', 'green', 'red', 'cyan', 'magenta', 'yellow', or 'black'.
Obsolete blocks are blocks that are no longer supported. They might or might not work properly.

Current blocks are supported and represent the latest block functionality.
liblinks | commliblist
Purpose Open main Communications System Toolbox block library
Syntax ..... commlib
Description commlib opens the latest version of the Communications System Toolbox \({ }^{\mathrm{TM}}\) block library.
See Also ..... dsplib

Purpose Package of communications scope classes

\section*{Syntax \(\quad h=\) commscope. \(<\) type \(>(\ldots)\)}

Description \(\quad h=\) commscope.<type>(...) returns a communications scope object \(h\) of type type.

Type help commscope to get a complete list of available types.
Each type of communications scope object is equipped with functions for simulation and visualization. Type help commscope. <type> to get the complete help on a specific communications scope object (e.g., help commscope.eyediagram).

\section*{See Also}
commscope.eyediagram

\section*{Purpose}

Eye diagram analysis
Syntax
h = commscope.eyediagram
h = commscope.eyediagram(property1, value1,...)
Description
\(\mathrm{h}=\) commscope.eyediagram constructs an eye diagram object, h , with
default properties. This syntax is equivalent to:
```

H = commscope.eyediagram('SamplingFrequency', 10000, ...
'SamplesPerSymbol', 100, ...
'SymbolsPerTrace', 2, ...
'MinimumAmplitude', -1, ...
'MaximumAmplitude', 1, ...
'AmplitudeResolution', 0.0100, ...
'MeasurementDelay', 0, ...
'PlotType', '2D Color', ...
'PlotTimeOffset', 0, ...
'PlotPDFRange', [0 1], ...
'ColorScale', 'linear', ...
'RefreshPlot', 'on');

```
h = commscope.eyediagram(property1, value1,...) constructs an
eye diagram object, \(h\), with properties as specified by property/value
pairs.

The eye diagram object creates a series of vertical histograms from zero to \(T\) seconds, at \(T_{s}\) second intervals, where \(T\) is a multiple of the symbol duration of the input signal and \(T_{s}\) is the sampling time. A vertical histogram is defined as the histogram of the amplitude of the input signal at a given time. The histogram information is used to obtain an approximation to the probability density function (PDF) of the input amplitude distribution. The histogram data is used to generate '2D Color' plots, where the color indicates the value of the PDF, and '3D Color' plots. The '2D Line' plot is obtained by constructing an eye diagram from the last \(n\) traces stored in the object, where a trace is defined as the segment of the input signal for a \(T\) second interval.

You can change the plot type by setting the PlotType property. The following plots are examples of each type.


\section*{2D-Color Eye Diagram}


3D-Color Eye Diagram


\section*{2D-Line Eye Diagram}

To see a detailed demonstration of this object's use, type showdemo scattereyedemo; at the command line.

\section*{Properties}

An eye diagram scope object has the properties shown on the following table. All properties are writable except for the ones explicitly noted otherwise.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline Type & \begin{tabular}{l} 
Type of scope object ('Eye Diagram '). \\
This property is not writable.
\end{tabular} \\
\hline SamplingFrequency & \begin{tabular}{l} 
Sampling frequency of the input signal \\
in hertz.
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline SamplesPerSymbol & \begin{tabular}{l} 
Number of samples used to \\
represent a symbol. An increase \\
in SamplesPerSymbol improves the \\
resolution of an eye diagram.
\end{tabular} \\
\hline SymbolRate & \begin{tabular}{l} 
The symbol rate of the input signal. \\
This property is not writable \\
and is automatically computed \\
based on SamplingFrequency and \\
SamplesPerSymbol.
\end{tabular} \\
\hline SymbolsPerTrace & \begin{tabular}{l} 
The number of symbols spanned on the \\
time axis of the eye diagram scope.
\end{tabular} \\
\hline MinimumAmplitude & \begin{tabular}{l} 
Minimum amplitude of the input signal. \\
Signal values less than this value \\
are ignored both for plotting and for \\
measurement computation.
\end{tabular} \\
\hline MaximumAmplitude & \begin{tabular}{l} 
Maximum amplitude of the input signal. \\
Signal values greater than this value \\
are ignored both for plotting and for \\
measurement computation.
\end{tabular} \\
\hline AmplitudeResolution & \begin{tabular}{l} 
The resolution of the amplitude \\
axis. The amplitude axis is \\
created from MinimumAmplitude \\
to MaximumAmplitude with \\
AmplitudeResolution steps.
\end{tabular} \\
\hline MeasurementDelay & \begin{tabular}{l} 
The time in seconds the scope waits \\
before starting to collect data.
\end{tabular} \\
\hline & \begin{tabular}{l} 
Man
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline PlotType & \begin{tabular}{l} 
Type of the eye diagram plot. The \\
choices are '2D Color ' (two dimensional \\
eye diagram, where color intensity \\
represents the probability density \\
function values), '3D Color' (three \\
dimensional eye diagram, where the \\
z-axis represents the probability density \\
function values), and '2D Line' (two \\
dimensional eye diagram, where each \\
trace is represented by a line).
\end{tabular} \\
\hline NumberOfStoredTraces & \begin{tabular}{l} 
The number of traces stored to display \\
the eye diagram in '2D Line ' mode.
\end{tabular} \\
\hline PlotTimeOffset & \begin{tabular}{l} 
The plot time offset input values must \\
reside in the closed interval [-Tsym \\
Tsym], where Tsym is the symbol \\
duration. Since the eye diagram is \\
periodic, if the value you enter is out of \\
range, it wraps to a position on the eye \\
diagram that is within range.
\end{tabular} \\
\hline RefreshPlot & \begin{tabular}{l} 
The switch that controls the plot refresh \\
style. The choices are 'on' (the eye \\
diagram plot is refreshed every time the \\
update method is called) and 'off' (the \\
eye diagram plot is not refreshed when \\
the update method is called).
\end{tabular} \\
\hline PolorScale & \begin{tabular}{l} 
The range of the PDF values that will be \\
displayed in the '2D Color' mode. The \\
PDF values outside the range are set to \\
a constant mask color.
\end{tabular} \\
\hline PlotPDFRange & \begin{tabular}{l} 
The scale used to represent the color, \\
the z-axis, or both. The choices are \\
'linear' (linear scale) and 'log' (base \\
ten logarithmic scale).
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline SamplesProcessed & \begin{tabular}{l} 
The number of samples processed by the \\
eye diagram object. This value does not \\
include the discarded samples during the \\
MeasurementDelay period. This property \\
is not writable.
\end{tabular} \\
\hline OperationMode & \begin{tabular}{l} 
When the operation mode is complex \\
signal, the eye diagram collects and plots \\
data on both the in-phase component \\
and the quadrature component. When \\
the operation mode is real signal, the eye \\
diagram collects and plots real signal \\
data.
\end{tabular} \\
\hline Measurements & \begin{tabular}{l} 
An eye diagram can display various types \\
of measurements. All measurements \\
are done on both the in-phase and \\
quadrature signal, unless otherwise \\
stated. For more information, see the \\
Measurements section.
\end{tabular} \\
\hline
\end{tabular}

The resolution of the eye diagram in ' 2 D Color' and '3D Color' modes can be increased by increasing SamplingFrequency, decreasing AmplitudeResolution, or both.

Changing MinimumAmplitude, MaximumAmplitude, AmplitudeResolution, SamplesPerSymbol, SymbolsPerTrace, and MeasurementDelay resets the measurements and updates the eye diagram.

\section*{Methods}

An eye diagram object is equipped with seven methods for inspection, object management, and visualization.

\section*{update}

This method updates the eye diagram object data.
update ( \(h, x\) ) updates the collected data of the eye diagram object \(h\) with the input \(x\).

If the RefreshPlot property is set to 'on', the update method also refreshes the eye diagram figure.

The following example shows this method's use:
```

% Create an eye diagram scope object
h = commscope.eyediagram('RefreshPlot', 'off')
% Prepare a noisy sinusoidal as input
hChan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)',...
'SNR', 20);
x = step(hChan,0.5*sin(2*pi*(0:1/100:10))+j*cos(2*pi*(0:1/100:10)));
% update the eyediagram
update(h, x);
% Check the number of proccessed samples
h.SamplesProcessed

```

\section*{plot}

This method displays the eye diagram figure.
The plot method has three usage cases:
plot (h) plots the eye diagram for the eye diagram object h with the current colormap or the default linespec.
plot (h, cmap), when used with the plottype set to '2D Color' or '3D Color ', plots the eye diagram for the object \(h\), and sets the colormap to cmap.
plot(h,linespec), when used with the plottype set to '2D Line', plots the eye diagram for the object h using linespec as the line specification. See the help for plot for valid linespecs.

The following example shows this method's use:
```

% Create an eye diagram scope object
h = commscope.eyediagram;
% Prepare a noisy sinusoid as input

```
```

hChan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)'
'SNR', 20);
x = step(hChan, 0.5*sin(2*pi**(0:1/100:10))+ j*0.5*cos(2*pi*(0:1/100:10
% Update the eye diagram
update(h, x);
% Display the eye diagram figure
plot(h)
% Display the eye diagram figure with jet colormap
plot(h, jet(64))
% Display 2D Line eye diagram with red dashed lines
h.PlotType = '2D Line';
plot(h, 'r--')

```

\section*{exportdata}

This method exports the eye diagram data.
[VERHIST EYEL HORHISTX HORHISTRF] = EXPORTDATA(H) Exports the eye diagram data collected by the eyediagram object \(H\).

VERHIST is a matrix that holds the vertical histogram, which is also used to plot '2D Color' and '3D Color' eye diagrams.
EYEL is a matrix that holds the data used to plot 2D Line eye diagram. Each row of the EYEC holds one trace of the input signal.

HORHISTX is a matrix that holds the crossing point histogram data collected for the values defined by the CrossingAmplitudes property of the MeasurementSetup object. HORHISTX(i, :) represents the histogram for CrossingAmplitudes(i).

HORHISTRF is a matrix that holds the crossing point histograms for rise and fall time levels. HORHISTRF(i,:) represents the histogram for AmplitudeThreshold(i).

The following example shows this method's use:
```

% Create an eye diagram scope object
h = commscope.eyediagram('RefreshPlot', 'off');

```
```

% Prepare a noisy sinusoidal as input
hChan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)',...
'SNR', 20);
x = step(hChan, 0.5*sin(2*pi*(0:1/100:10))+ j*0.5*cos(2*pi*(0:1/100:10)))
% Update the eyediagram
update(h, x);
% Export the data
[eyec eyel horhistx horhistrf] = exportdata(h);
% Plot line data
t=0:1/h.SamplingFrequency:h.SymbolsPerTrace/h.SymbolRate;
plot(t, real(eyel)); xlabel('time (s)');...
ylabel('Amplitude (AU)'); grid on;
% Plot 2D Color data
t=0:1/h.SamplingFrequency:h.SymbolsPerTrace/h.SymbolRate;
a=h.MinimumAmplitude:h.AmplitudeResolution:h.MaximumAmplitude;
imagesc(t,a,eyec); xlabel('time (s)'); ylabel('Amplitude (AU)');
reset

```

This method resets the eye diagram object.
reset ( h ) resets the eye diagram object h . Resetting h clears all the collected data.

The following example shows this method's use:
```

% Create an eye diagram scope object
h = commscope.eyediagram('RefreshPlot', 'off');
% Prepare a noisy sinusoidal as input
hChan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)',...
'SNR', 20);
x = step(hChan, 0.5*sin(2*pi*(0:1/100:10))+ j*0.5*cos(2*pi*(0:1/100:10)))
update(h, x); % update the eyediagram
h.SamplesProcessed % Check the number of proccessed samples
reset(h); % reset the object
h.SamplesProcessed %Check the number of proccessed samples
copy

```

This method copies the eye diagram object.
\(h=\) copy (ref_obj) creates a new eye diagram object \(h\) and copies the properties of object \(h\) from properties of ref_obj.
The following example shows this method's use:
```

% Create an eye diagram scope object
h = commscope.eyediagram('MinimumAmplitude', -3, ...
'MaximumAmplitude', 3);
disp(h); % display object properties
h1 = copy(h)
disp

```

This method displays properties of the eye diagram object.
disp(h) displays relevant properties of eye diagram object \(h\).
If a property is not relevant to the object's configuration, it is not displayed. For example, for a commscope. eyediagram object, the ColorScale property is not relevant when PlotType property is set to '2D Line'. In this case the ColorScale property is not displayed.

The following is an example of its use:
```

% Create an eye diagram scope object
h = commscope.eyediagram;
% Display object properties
disp(h);
h = commscope.eyediagram('PlotType', '2D Line')
close

```

This method closes the eye diagram object figure.
close ( \(h\) ) closes the figure of the eye diagram object \(h\).
The following example shows this method's use:
```

% Create an eye diagram scope object
h = commscope.eyediagram;
% Call the plot method to display the scope
plot(h);
% Wait for 1 seconds

```
```

pause(1)
% Close the scope
close(h)

```

\section*{analyze}

This methods executes eye diagram measurements. analyze(h) executes the eye diagram measurements on the collected data of the eye diagram scope object \(h\). The results of the measurements are stored in the Measurements property of \(h\). See "Measurements" on page 1-98 for more information.

In some cases, the analyze method cannot determine a measurement value. If this problem occurs, verify that your settings for measurement setup values or the eye diagram are valid.

Measurements You can obtain the following measurements on an eye diagram:
- Amplitude Measurements
- Eye Amplitude
- Eye Crossing Amplitude
- Eye Crossing Percentage
- Eye Height
- Eye Level
- Eye SNR
- Quality Factor
- Vertical Eye Opening
- Time Measurements
- Deterministic Jitter
- Eye Crossing Time
- Eye Delay
- Eye Fall Time
- Eye Rise Time
- Eye Width
- Horizontal Eye Opening
- Peak-to-Peak Jitter
- Random Jitter
- RMS Jitter
- Total Jitter

Measurements assume that the eye diagram object has valid data. A valid eye diagram has two distinct eye crossing points and two distinct eye levels.
The deterministic jitter, horizontal eye opening, quality factor, random jitter, and vertical eye opening measurements utilize a dual-Driac algorithm. Jitter is the deviation of a signal's timing event from its intended (ideal) occurrence in time [1]. Jitter can be represented with a dual-Driac model. A dual-Driac model assumes that the jitter has two components: deterministic jitter (DJ) and random jitter (RJ). The DJ PDF comprises two delta functions, one at \(\mu_{\mathrm{L}}\) and one at \(\mu_{\mathrm{R}}\). The RJ PDF is assumed to be Gaussian with zero mean and variance \(\sigma\).

The Total Jitter (TJJ) PDF is the convolution of these two PDFs, which is composed of two Gaussian curves with variance \(\sigma\) and mean values \(\mu_{L}\) and \(\mu_{R}\). See the following figure.


The dual-Dirac model is described in [5] in more detail. The amplitude of the two Dirac functions may not be the same. In such a case, the analyze method estimates these amplitudes, \(\rho_{\mathrm{L}}\) and \(\rho_{\mathrm{R}}\).

\section*{Amplitude Measurements}

You can use the vertical histogram to obtain a variety of amplitude measurements. For complex signals, measurements are done on both in-phase and the quadrature components, unless otherwise specified.

Note For amplitude measurements, at least one bin per vertical histogram must reach 10 hits before the measurement is taken, ensuring higher accuracy.

\section*{Eye Amplitude (EyeAmplitude)}

Eye Amplitude, measured in Amplitude Units (AU), is defined as the distance between two neighboring eye levels. For an NRZ signal, there are only two levels: the high level (level 1 in figure) and the low level (level 0 in figure). The eye amplitude is the difference of these two values, as shown in figure [3].


\section*{Eye Crossing Amplitude (EyeCrossingLevel)}

Eye crossing amplitudes are the amplitude levels at which the eye crossings occur, measured in Amplitude Units (AU). The analyze
method calculates this value using the mean value of the vertical histogram at the crossing times [3]. See the following figure.


The next figure shows the vertical histogram at the first eye crossing time.


\section*{Eye Crossing Percentage (EyeOpeningVer)}

Eye Crossing Percentage is the location of the eye crossing levels as a percentage of the eye amplitude.

\section*{Eye Height (EyeHeight)}

Eye Height, measured in Amplitude Units (AU), is defined as the 30 distance between two neighboring eye levels.

For an NRZ signal, there are only two levels: the high level (level 1 in figure) and the low level (level 0 in figure). The eye height is the difference of the two \(3 \sigma\) points, as shown in the next figure. The \(3 \sigma\) point is defined as the point that is three standard deviations away from the mean value of a PDF.


\section*{Eye Level (EyeLevel)}

Eye Level is the amplitude level used to represent data bits, measured in Amplitude Units (AU).

For an ideal NRZ signal, there are two eye levels: +A and -A. The analyze method calculates eye levels by estimating the mean value of the vertical histogram in a window around the EyeDelay, which is also the \(50 \%\) point between eye crossing times [3]. The width of
this window is determined by the EyeLevelBoundary property of the eyemeasurementsetup object, shown in the next figure.


The analyze method calculates the mean value of all the vertical histograms within the eye level boundaries. The mean vertical histogram appears in the following figure. There are two distinct PDFs, one for each eye level. The mean values of the individual histograms are the eye levels as shown in this figure.


\section*{Eye SNR (EyeSNR)}

Eye signal-to-noise ratio is defined as the ratio of the eye amplitude to the sum of the standard deviations of the two eye levels. It can be expressed as:
\[
\mathrm{SNR}=\frac{L_{1}-L_{0}}{\sigma_{1}+\sigma_{0}}
\]
where \(L_{1}\) and \(L_{0}\) represent eye level 1 and 0 , respectively, and \(\sigma_{1}\) and \(\sigma_{2}\) are the standard deviation of eye level 1 and 0 , respectively.

For an NRZ signal, eye level 1 corresponds to the high level, and the eye level 0 corresponds to low level.

\section*{Quality Factor (QualityFactor)}

The analyze method calculates Quality Factor the same way as the eye SNR. However, instead of using the mean and standard deviation values of the vertical histogram for \(\mathrm{L}_{1}\) and \(\sigma_{1}\), the analyze method uses the mean and standard deviation values estimated using the dual-Dirac method. [2] See dual-Dirac section for more detail.

\section*{Vertical Eye Opening (EyeOpeningVer)}

Vertial Eye Opening is defined as the vertical distance between two points on the vertical histogram at EyeDelay that corresponds to the BER value defined by the BERThreshold property of the eyemeasurementsetup object. The analyze method calculates this measurement taking into account the random and deterministic components using a dual-Dirac model [5] (see the Dual Dirac Section). A typical BER value for the eye opening measurements is \(10^{-12}\), which approximately corresponds to the \(7 \sigma\) point assuming a Gaussian distribution.

\section*{Time Measurements}

You can use the horizontal histogram of an eye diagram to obtain a variety of timing measurements.

Note For time measurements, at least one bin per horizontal histogram must reach 10 hits before the measurement is taken.

\section*{Deterministic Jitter (JitterDeterministic)}

Deterministic Jitter is the deterministic component of the jitter. You calculate it using the tail mean value, which is estimated using the dual-Dirac method as follows [5]:

DJ \(=\mu_{L}-\mu_{R}\)
where \(\mu_{L}\) and \(\mu_{R}\) are the mean values returned by the dual-Dirac algorithm.

\section*{Eye Crossing Time (EyeCrossingTime)}

Eye crossing times are calculated as the mean of the horizontal histogram for each crossing point, around the reference amplitude level. This value is measured in seconds. The mean value of all the horizontal PDFs is calculated in a region defined by the CrossingBandWith property of the eyemeasurementsetup object.
The region is from \(-A_{\text {total }} * B W\) to \(+A_{\text {total }} * B W\), where \(A_{\text {total }}\) is the total amplitude range of the eye diagram (i.e., \(A_{\text {total }}=A_{\text {max }}-A_{\min }\) ) and \(B W\) is the crossing band width, shown in the following figure.


The following figure shows the average PDF in this region. Because this example assumes two symbols per trace, there are two crossing points.


Note When an eye crossing time measurement falls within the [-0.5/Fs, 0 ) seconds interval, the time measurement wraps to the end of the eye diagram, i.e., the measurement wraps by \(2 * \mathrm{Ts}\) seconds (where Ts is the symbol time). For a complex signal case, the analyze method issues a warning if the crossing time measurement of the in-phase branch wraps while that of the quadrature branch does not (or vice versa).

To avoid the time-wrapping or a warning, add a half-symbol duration delay to the current value in the MeasurementDelay property of the eye diagram object. This additional delay repositions the eye in the approximate center of the scope.

\section*{Eye Delay (EyeDelay)}

Eye Delay is the distance from the midpoint of the eye to the time origin, measured in seconds. The analyze method calculates this distance using the crossing time. For a symmetric signal, EyeDelay is also the best sampling point.


\section*{Eye Fall Time (EyeFallTime)}

Eye Fall Time is the mean time between the high and low threshold values defined by the AmplitudeThreshold property of the eyemeasurementsetup object. The previous figure shows the fall time calculated from \(10 \%\) to \(90 \%\) of the eye amplitude.

\section*{Eye Rise Time (EyeRiseTime)}

Eye Rise Time is the mean time between the low and high threshold values defined by the AmplitudeThreshold property of the eyemeasurementsetup object. The following figure shows the rise time calculated from \(10 \%\) to \(90 \%\) of the eye amplitude.


\section*{Eye Width (EyeWidth)}

Eye Width is the horizontal distance between two points that are three standard deviations ( \(3 \sigma\) ) from the mean eye crossing times, towards the center of the eye. The value for Eye Width measurements is seconds.


\section*{Horizontal Eye Opening (EyeOpeningHor)}

Horizontal Eye Opening is the horizontal distance between two points on the horizontal histogram that correspond to the \(B E R\) value defined by the BERThreshold property of the eyemeasurementsetup object. The measurement is take at the amplitude value defined by the ReferenceAmplitude property of the eyemeasurementsetup object. It is calculated taking into account the random and deterministic components using a dual-Dirac model [5] (see the Dual Dirac Section).

A typical \(B E R\) value for the eye opening measurements is \(10^{-12}\), which approximately corresponds to the \(7 \sigma\) point assuming a Gaussian distribution.

\section*{Peak-to-Peak Jitter (JitterP2P)}

Peak-To-Peak Jitter is the difference between the extreme data points of the histogram.

\section*{Random Jitter (JitterRandom)}

Random Jitter is defined as the Gaussian unbounded component of the jitter. The analyze method calculates it using the tail standard deviation estimated using the dual-Dirac method as follows [5]:
\(R J=\left(Q_{\mathrm{L}}+Q_{\mathrm{R}}\right) * \sigma\)
where
\(Q_{L}=\sqrt{2} * \operatorname{erfc}^{-1}\left(\frac{2 * B E R}{\rho_{L}}\right)\)
and
\(Q_{R}=\sqrt{2} * e r f c^{-1}\left(\frac{2 * B E R}{}\right)\)
\(B E R\) is the bit error PRio at which the random jitter is calculated. It is defined with the BERThreshold property of the eyemeasuremensetup object.

\section*{RMS Jitter (JitterRMS)}

RMS Jitter is the standard deviation of the jitter calculated from the horizontal histogram.

\section*{Total Jitter (JitterTotal)}

Total Jitter is the sum of the random jitter and the deterministic jitter [5].

Measurement A number of set-up parameters control eye diagram measurements. Setup Parameters This section describes these set-up parameters and the measurements they affect.

\section*{Eye Level Boundaries}

Eye Level Boundaries are defined as a percentage of the symbol duration. The analyze method calculates the eye levels by averaging the vertical histogram within a given time interval defined by the eye level boundaries. A common value you can use for NRZ signals is \(40 \%\) to \(60 \%\). For RZ signals, a narrower band of \(5 \%\) is more appropriate. The
default setting for Eye level Boundaries is a 2-by-1 vector where the first element is the lower boundary and the second element is the upper boundary. When the eye level boundary changes, the object recalculates this value.

\section*{Reference Amplitude}

Reference Amplitude is the boundary value at which point the signal crosses from one signal level to another. Reference amplitude represents the decision boundary of the modulation scheme. This value is used to perform jitter measurements. The default setting for Reference Amplitude is a 2 -by- 1 double vector where the first element is the lower boundary and the second element is the upper boundary. Setting the reference amplitude resets the eye diagram.

The crossing instants of the input signal are detected and recorded as crossing times. A common value you can use for NRZ signals is 0 . For RZ signals, you can use the mean value of 1 and 0 levels. Reference amplitude is stored in a 2 -by-N matrix, where the first row is the in-phase values and second row is the quadrature values. See Eye Crossing Time for more information.

\section*{Crossing Bandwidth}

Crossing Bandwidth is the amplitude band used to measure the crossing times of the eye diagram. Crossing Bandwidth represents a percentage of the amplitude span of the eye diagram, typically \(5 \%\). See Eye Crossing Time for more information. The default setting for Crossing Bandwidth is 0.0500 .

\section*{Bit Error Rate Threshold}

The eye opening measurements, random, and total jitter measurements are performed at a given BER value. This BER value defines the BER threshold. A typical value is \(1 \mathrm{e}^{-12}\). The default setting for Bit Error Threshold is \(1.0000 \mathrm{e}^{-12}\). When the bit error rate threshold changes, the object recalculates this value.

\section*{Amplitude Threshold}

The rise time of the signal is defined as the time required for the signal to travel from the lower amplitude threshold to the upper amplitude threshold. The fall time, measured from the upper amplitude threshold to the lower amplitude threshold, is defined as a percentage of the eye amplitude. The default setting is \(10 \%\) for the lower threshold and \(90 \%\) for the upper threshold. Setting the amplitude threshold resets the eye diagram. See Eye Rise Time and Eye Fall Time for more information.

\section*{Jitter Hysteresis}

You can use the JitterHysteresis property of the eyemeasurementsetup object to remove the effect of noise from the horizontal histogram estimation. The default value for Jitter Hysteresis is zero. Setting the jitter hysteresis value resets the eye diagram.
If channel noise impairs the signal being tested, as shown in the following figure, the signal may seem like it crosses the reference amplitude level multiple times during a single \(0-1\) or 1-0 transition.


See the zoomed-in image for more detail.


To eliminate the effect of noise, define a hysteresis region between two threshold values: \(\mathrm{A}_{\text {ref }}+\Delta \mathrm{A}\) and \(\mathrm{A}_{\text {ref }}-\Delta \mathrm{A}\), where \(\mathrm{A}_{\text {ref }}\) is the reference amplitude value and \(\Delta \mathrm{A}\) is the jitter hysteresis value. If the signal crosses both threshold values, level crossing is declared. Then, linear interpolation calculates the crossing point in the horizontal histogram estimation.

\section*{Examples}
```

% Construct an eye diagram object for signals in the range
% of [-3 3]
h = commscope.eyediagram('MinimumAmplitude', -3, ...
'MaximumAmplitude', 3)
% Construct an eye diagram object for a signal with
% 1e-3 seconds of transient time
h = commscope.eyediagram('MeasurementDelay', 1e-3)
% Construct an eye diagram object for '2D Line' plot type
% with 100 traces to display
h = commscope.eyediagram('PlotType', '2D Line', ...
'NumberOfStoredTraces', 100)

```

\footnotetext{
References [1] Nelson Ou, et al, Models for the Design and Test of Gbps-Speed Serial Interconnects,IEEE Design \& Test of Computers, pp. 302-313, July-August 2004.
[2] HP E4543A Q Factor and Eye Contours Application Software, Operating Manual, http://agilent.com
[3] Agilent 71501D Eye-Diagram Analysis, User's Guide, http://www.agilent.com
[4] 4] Guy Foster, Measurement Brief: Examining Sampling Scope Jitter Histograms, White Paper, SyntheSys Research, Inc., July 2005.
[5] Jitter Analysis: The dual-Dirac Model, RJ/DJ, and Q-Scale, White Paper, Agilent Technologies, December 2004, http://www.agilent.com

See Also
commscope
}

\section*{Purpose}

Create Scatter Plot scope
Syntax

Description
h = commscope.ScatterPlot
h = commscope.ScatterPlot('PropertyName',PropertyValue,...)
commscope. ScatterPlot collects data and displays results in a Figure window. You can create a scatter plot using a default configuration or by defining properties.
\(\mathrm{h}=\) commscope.ScatterPlot returns a scatter plot scope, \(h\).
h = commscope.ScatterPlot('PropertyName',PropertyValue,...) returns a scatter plot scope, \(h\), with property values set to PropertyValues. See the Properties section of this help page for valid PropertyNames.

\section*{Properties}

A ScatterPlot object has the properties shown on the following table. All properties are writable except for the ones explicitly noted otherwise.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline Type & \begin{tabular}{l} 
'Scatter Plot'. This is a read-only \\
property.
\end{tabular} \\
\hline SamplingFrequency & \begin{tabular}{l} 
Sampling frequency of the input \\
signal in Hz.
\end{tabular} \\
\hline SamplesPerSymbol & \begin{tabular}{l} 
Number of samples used to \\
represent a symbol.
\end{tabular} \\
\hline SymbolRate & \begin{tabular}{l} 
The symbol rate of the input \\
signal. This property is read-only \\
and is automatically computed \\
based on SamplingFrequency \\
and SamplesPerSymbol.
\end{tabular} \\
\hline MeasurementDelay & \begin{tabular}{l} 
The time in seconds the scope \\
will wait before starting to collect \\
data.
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{|c|c|}
\hline Property & Description \\
\hline SamplingOffset & The number of samples skipped at each sampling point relative to the MeasurementDelay. \\
\hline Constellation & Expected constellation of the input signal. \\
\hline RefreshPlot & \begin{tabular}{l}
The switch that controls the plot refresh style. The choices are: \\
- 'on' - The scatter plot refreshes every time the update method is called. \\
- 'off' - The scatter plot does not refresh when the update method is called.
\end{tabular} \\
\hline SamplesProcessed & The number of samples processed by the scope. This value does not include the discarded samples during the MeasurementDelay period. This property is read-only. \\
\hline PlotSettings & \begin{tabular}{l}
Plot settings control the scatter plot figure. \\
- SymbolStyle - Line style of symbols \\
- SignalTrajectory - The switch to control the visibility of the signal trajectory. The choices are 'on' or 'off'. \\
- SignalTrajectoryStyle - Line style of signal trajectory \\
- Constellation - The switch to control the visibility of
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline \multicolumn{1}{l}{\begin{tabular}{l} 
the constellation points. The \\
choices are 'on' or 'off'. \\
\\
\\
\\
\\
\\
\\
\\
- \begin{tabular}{l} 
ConstellationStyle - Line style \\
of signal trajectory \\
- Grid - The switch to control \\
the visibility of the grid. The \\
choices are 'on' or 'off'.
\end{tabular} \\
\hline
\end{tabular}}
\end{tabular}

\section*{Methods}

A Scatter Plot has the following methods.

\section*{autoscale}

This method automatically scales the plot figure so its entire contents displays.

\section*{close}

This method closes the scatter plot figure.

\section*{disp}

This method displays the scatter plot properties.

\section*{plot}

This method creates a scatter plot figure. If a figure exists, this method updates the figure's contents.
plot (h) plots a scatter plot figure using default settings.

\section*{reset}

This method resets the collected data of the scatter plot object.
reset ( h ) resets the collected data of the scatter plot object h . Resetting \(h\) also clears the plot and NumberOfSymbols.

\section*{update}

This method updates the collected data of the scatter plot.
update ( \(\mathrm{h}, \mathrm{r}\) ) updates the collected data of the scatter plot, where \(h\) is the handle of the scatter plot object and \(r\) is the complex input data under test. This method updates the collected data and the plot (if RefreshPlot is true).

How To - Viewing Signals Using Scatter Plots

\section*{Purpose}

Construct pattern generator object

\section*{Syntax}

Description

\section*{Properties}
h = commsrc.pattern syntax is equivalent to: object can also inject jitter into the modulated signal.
\(\mathrm{h}=\) commsrc.pattern constructs a pattern generator object, \(h\). This
```

h = commsrc.pattern('SamplingFrequency', 10000, ...
'SamplesPerSymbol', 100, ...
'PulseType', 'NRZ', ...
'OutputLevels', [-1 1], ...
'RiseTime', 0, ...
'FallTime', 0, ...
'DataPattern', 'PRBS7', ...
'Jitter', commsrc.combinedjitter)

```

The pattern generator object produces modulated data patterns. This

A pattern generator object has the properties shown on the following table. You can edit all properties, except those explicitly noted otherwise.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline Type & \begin{tabular}{l} 
Type of pattern generator object \\
('Pattern Generator'). This property \\
is not writable.
\end{tabular} \\
\hline SamplingFrequency & \begin{tabular}{l} 
Sampling frequency of the input signal \\
in hertz.
\end{tabular} \\
\hline SymbolRate & \begin{tabular}{l} 
The symbol rate of the input \\
signal. This property depend \\
upon the SamplingFequency and \\
SamplesPerSymbol properties. This \\
property is not writable.
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{|c|c|}
\hline Property & Description \\
\hline SamplesPerSymbol & The number of samples representing a symbol. SamplesPerSymbol must be an integer. This property affects SymbolRate. \\
\hline PulseType & The type of pulse the object generates. Pulse types available: return-to-zero (RZ) and nonreturn-to-zero (NRZ). The initial condition for an NRZ pulse is 0 . \\
\hline OutputLevels & Amplitude levels that correspond to the symbol indices. For an NRZ pulse, this is a 1 -by- 2 vector. The first element of the 1-by-2 vector corresponds to the 0th symbol (data bit value 0 ). The second element corresponds to the 1st symbol (data bit value 1). For an RZ pulse, this is a scalar and the value corresponds to the data bit value 1 . \\
\hline DutyCycle & The duty cycle of the pulse the object generates. Displays calculated duty cycle based on pulse parameters. This property is not writable. \\
\hline RiseTime & Specifies \(10 \%\) to \(90 \%\) rise time of the pulse in seconds. \\
\hline PulseDuration & Pulse duration in seconds defined by IEEE STD 181 standard. (See the Return-to-Zero (RZ) Signal Conversion: Ideal Pulse to STD-181 figure in the Methods section.) Setting PulseType to return-to-zero enables this property. \\
\hline FallTime & Specifies \(10 \%\) to \(90 \%\) fall time of the pulse in seconds. \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline DataPattern & \begin{tabular}{l} 
The bit sequence the object uses. The \\
following patterns are available: PRBS5 \\
to PRBS15, PRBS23, PRBS31, and User \\
Defined.
\end{tabular} \\
\hline UserDataPattern & \begin{tabular}{l} 
User-defined bit pattern consisting of a \\
vector of ones and zeroes. Setting data \\
pattern to user defined enables this \\
property.
\end{tabular} \\
\hline Jitter & \begin{tabular}{l} 
Specifies jitter characteristics. Use this \\
property to configure Random, Periodic \\
and Dual Dirac Jitter.
\end{tabular} \\
\hline
\end{tabular}

\section*{Methods}

A pattern generator object has five methods, as described in this section.

\section*{generate}

This method outputs a frame worth of modulated and interpolated symbols. It has one input argument, which is the number of symbols in a frame. Its output is a double-column vector. You can call this method using the following syntax
\(\mathrm{x}=\) generate(h, N )
where \(h\) is the handle to the object, \(N\) is the number of output symbols, and \(x\) is a double-column vector.

\section*{reset}

This method resets the pattern generator to its default state. The property values do not reset unless they relate to the state of the object. This method has no input arguments.

\section*{idealtostd 181}

This method converts the ideal pulse specifications to IEEE STD-181 specifications: \(0 \%\) to \(100 \%\) rise time (TR) and fall time (TF) convert to \(10 \%\) to \(90 \%\) rise and fall times with a \(50 \%\) pulse width duration, as
shown in the following figure. This method also sets the appropriate properties.
idealtostd181( \(\left.\mathrm{t}_{\mathrm{R}}, \mathrm{t}_{\mathrm{F}}, \mathrm{PW}\right)\)


IEEE STD-181 Return-to-Zero (RZ) Signal Parameters

\section*{std 181 toideal}

This method converts the IEEE STD-181 pulse specifications, stored in the pattern generator, to ideal pulse specifications. This method converts the \(10 \%\) to \(90 \%\) rise and fall times to \(0 \%\) to \(100 \%\) rise and fall times ( \(T R\) and \(T F\) ). It also converts the \(50 \%\) pulse duration to pulse width (as shown in the following figure). Use the property values for IEEE STD-181 specifications
```

[tr tf pw] = stdstd181toideal(h)

```
where \(h\) is the pattern generator object handle and \(t_{R}\) is 0 to \(100 \%\) rise time.


Ideal Pulse Non-Return-to-Zero (NRZ) Signal Parameters

\section*{computedcd}

Computes the duty cycle distortion, DCD, of the pulse defined by the pattern generator object \(h\).

DCD represents the ratio of the pulse on duration to the pulse off duration. For an NRZ pulse, on duration is the duration the pulse spends above the symbol boundary level. Off duration is the duration the pulse spends below zero.
```

dcd = computedcd(h)

```

The software calculates DCD given \(t_{R}, t_{F}, T_{\text {sym }}\). This formula assumes that the symbol boundary level is zero.
\(T_{h}=\left(A_{h}-A_{l}\right) * \frac{t_{R}}{A_{l}}+\left(A_{h}-A_{\nu}\right) * \frac{t_{F}}{A_{l}}+\mathrm{PW}_{+}\)
\(T_{l}=\left(A_{h}-A_{l}\right) * \frac{t_{R}}{A_{l}}+\left(A_{h}-A_{l}\right) * \frac{t_{F}}{A_{l}}+\mathrm{PW}\).
\(\mathrm{DCD}=\frac{T_{h}}{T_{l}}\)

Where \(T_{h}\) is the duration of the high signal, \(T_{l}\) is the duration of the low signal, and DCD represents the ratio of the duration of the high signal to the low signal.

\section*{Purpose}

Create PN sequence generator package

\section*{Syntax}
\(\mathrm{h}=\) commsrc.pn
h = commsrc.pn(property1,value1,...)
\(\mathrm{h}=\) commsrc.pn creates a default PN sequence generator object \(h\), and is equivalent to the following:
```

H = COMMSRC.PN('GenPoly',
[1 0 0 0 0 1 1], ...
'InitialStates',
[0 0 0 0 0 1], ...
'CurrentStates',
'Mask'
[0 0 0 0 0 1], ...
[0 0 0 0 0 1], ...
'NumBitsOut',
1)

```
or
```

H = COMMSRC.PN('GenPoly', $\quad\left[\begin{array}{llllll}1 & 0 & 0 & 0 & 0 & 1 \\ 1\end{array}\right], \ldots$
'InitialStates', [0 00001 1], ...
'CurrentStates', $[000001]$, ...
'Shift', 0
'NumBitsOut', 1)

```
h = commsrc.pn(property1, value1,...) creates a PN sequence generator object, \(h\), with properties you specify as property/value pairs.

\section*{Properties}

A PN sequence generator has the properties shown on the following table. All properties are writable except for the ones explicitly noted otherwise.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline GenPoly & \begin{tabular}{l} 
Generator polynomial vector \\
array of bits; must be descending \\
order
\end{tabular} \\
\hline InitialStates & \begin{tabular}{l} 
Vector array (with length of the \\
generator polynomial order) of \\
initial shift register values (in \\
bits)
\end{tabular} \\
\hline CurrentStates & \begin{tabular}{l} 
Vector array (with length of the \\
generator polynomial order) of \\
present shift register values (in \\
bits)
\end{tabular} \\
\hline NumBitsOut & \begin{tabular}{l} 
Number of bits to output at each \\
generate method invocation
\end{tabular} \\
\hline Mask or Shift & \begin{tabular}{l} 
A mask vector of binary 0 and 1 \\
values is used to specify which \\
shift register state bits are XORed \\
to produce the resulting output \\
bit value. \\
Alternatively, a scalar shift \\
value may be used to specify an \\
equivalent shift (either a delay or \\
advance) in the output sequence.
\end{tabular} \\
\hline
\end{tabular}

The 'GenPoly' property values specify the shift register connections. Enter these values as either a binary vector or a vector of exponents of the nonzero terms of the generator polynomial in descending order of powers. For the binary vector representation, the first and last elements of the vector must be 1 . For the descending-ordered polynomial representation, the last element of the vector must be 0 . For more information and examples, see the LFSR SSRG Details section of this page.

\footnotetext{
Methods
A PN sequence generator is equipped with the following methods.
}
\begin{tabular}{ll} 
& generate \\
& Generate [NumBitsOut x 1] PN sequence generator values \\
& reset \\
& Set the CurrentStates values to the InitialStates values \\
& getshift \\
& Get the actual or equivalent shift property value \\
& getmask \\
& Get the actual or equivalent Mask property value \\
& copy \\
& Make an independent copy of a commsrc.pn object \\
& disp \\
& Display PN sequence generator object properties
\end{tabular}


All \(r\) registers in the generator update their values at each time step according to the value of the incoming arrow to the shift register. The adders perform addition modulo 2 . The shift register is described by the 'GenPoly' property (generator polynomial), which is a primitive binary polynomial in \(z, \mathrm{~g}_{\mathrm{r}} \mathrm{r}^{\mathrm{r}}+\mathrm{g}_{\mathrm{r}-1} \mathrm{z}^{\mathrm{r}-1}+\mathrm{g}_{\mathrm{r}-2} \mathrm{z}^{\mathrm{r}-2}+\ldots+\mathrm{g}_{0}\). The coefficient \(\mathrm{g}_{\mathrm{k}}\) is 1 if there is a connection from the kth register, as labeled in the preceding diagram, to the adder. The leading term \(\mathrm{g}_{\mathrm{r}}\) and the constant term \(\mathrm{g}_{0}\) of the 'GenPoly' property must be 1 because the polynomial must be primitive.

You can specify the Generator polynomial parameter using either of these formats:
- A vector that lists the coefficients of the polynomial in descending order of powers. The first and last entries must be 1 . Note that the length of this vector is one more than the degree of the generator polynomial.
- A vector containing the exponents of \(z\) for the nonzero terms of the polynomial in descending order of powers. The last entry must be 0 .

For example, \(\left[\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0\end{array} 1\right]\) and [8 200\(]\) represent the same polynomial, \(p(z)=z^{8}+z^{2}+1\).
The Initial states parameter is a vector specifying the initial values of the registers. The Initial states parameter must satisfy these criteria:
- All elements of the Initial states vector must be binary numbers.
- The length of the Initial states vector must equal the degree of the generator polynomial.

Note At least one element of the Initial states vector must be nonzero in order for the block to generate a nonzero sequence. That is, the initial state of at least one of the registers must be nonzero.

For example, the following table indicates two sets of parameter values that correspond to a generator polynomial of \(p(z)=z^{8}+z^{2}+1\).
\begin{tabular}{l|l|l|l}
\hline Quantity & Example 1 & & Example 2 \\
\hline \begin{tabular}{l} 
Generator \\
polynomial
\end{tabular} & \begin{tabular}{l}
\(g 1=\left[\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0 \\
1\end{array}\right]\)
\end{tabular} \(\mathrm{g} 2=\left[\begin{array}{llll}8 & 2 & 0\end{array}\right]\) \\
\hline \begin{tabular}{l} 
Degree of \\
generator \\
polynomial
\end{tabular} & 8, which is length \((g 1)-1\) & 8 \\
\hline \begin{tabular}{l} 
Initial \\
states
\end{tabular} & {\(\left[\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0\end{array}\right]\)} & {\(\left[\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0\end{array}\right]\)} \\
\hline
\end{tabular}

Output mask vector (or scalar shift value) shifts the starting point of the output sequence. With the default setting for this parameter, the only connection is along the arrow labeled \(m_{0}\), which corresponds to a shift of 0 . The parameter is described in greater detail below.

You can shift the starting point of the PN sequence with Output mask vector (or scalar shift value). You can specify the parameter in either of two ways:
- An integer representing the length of the shift
- A binary vector, called the mask vector, whose length is equal to the degree of the generator polynomial

The difference between the block's output when you set Output mask vector (or scalar shift value) to 0 , versus a positive integer \(d\), is shown in the following table.
\begin{tabular}{lllllll} 
& \(\mathbf{T}=\mathbf{0}\) & \(\mathbf{T}=\mathbf{1}\) & \(\mathbf{T}=\mathbf{2}\) & \(\ldots\) & \(\mathbf{T}=\mathbf{d}\) & \begin{tabular}{c}
\(\mathbf{T}=\) \\
\(\mathbf{d}+\mathbf{1}\)
\end{tabular} \\
Shift = 0 & \(x_{0}\) & \(x_{1}\) & \(x_{2}\) & \(\ldots\) & \(x_{\mathrm{d}}\) & \(x_{\mathrm{d}+1}\) \\
Shift = d & \(x_{\mathrm{d}}\) & \(x_{\mathrm{d}+1}\) & \(x_{\mathrm{d}+2}\) & \(\ldots\) & \(x_{2 \mathrm{~d}}\) & \(x_{2 \mathrm{~d}+1}\)
\end{tabular}

Alternatively, you can set Output mask vector (or scalar shift value) to a binary vector, corresponding to a polynomial in \(z, \mathrm{~m}_{\mathrm{r}-\mathrm{z}} \mathrm{z}^{\mathrm{r}-1}\) \(+\mathrm{m}_{\mathrm{r}-2} \mathrm{z}^{\mathrm{r}-2}+\ldots+\mathrm{m}_{1} \mathrm{z}+\mathrm{m}_{0}\), of degree at most \(\mathrm{r}-1\). The mask vector corresponding to a shift of \(d\) is the vector that represents \(m(z)=z^{d}\) modulo \(\mathrm{g}(z)\), where \(\mathrm{g}(z)\) is the generator polynomial. For example, if the degree of the generator polynomial is 4 , then the mask vector corresponding to \(d=2\) is \(\left[\begin{array}{lll}0 & 1 & 0\end{array} 0\right.\) ], which represents the polynomial \(m(z)=z^{2}\). The preceding schematic diagram shows how Output mask vector (or scalar shift value) is implemented when you specify it as a mask vector. The default setting for Output mask vector (or scalar shift value) is 0 . You can calculate the mask vector using the Communications System Toolbox function shift2mask.

\section*{Sequences of Maximum Length}

If you want to generate a sequence of the maximum possible length for a fixed degree, \(r\), of the generator polynomial, you can set Generator polynomial to a value from the following table. See Proakis, John G., Digital Communications, Third edition, New York, McGraw Hill, 1995 for more information about the shift-register configurations that these polynomials represent.
\begin{tabular}{|c|c|c|c|}
\hline r & Generator Polynomial & r & Generator Polynomial \\
\hline 2 & \(\left[\begin{array}{lll}2 & 1 & 0\end{array}\right]\) & 21 & [21 190] \\
\hline 3 & \(\left[\begin{array}{lll}3 & 2 & 0\end{array}\right]\) & 22 & \(\left[\begin{array}{llll}22 & 21 & 0\end{array}\right]\) \\
\hline 4 & \(\left[\begin{array}{lll}4 & 3 & 0\end{array}\right]\) & 23 & \(\left[\begin{array}{lll}23 & 18 & 0\end{array}\right]\) \\
\hline 5 & \(\left[\begin{array}{lll}5 & 3 & 0\end{array}\right]\) & 24 & \(\left[\begin{array}{lllll}24 & 23 & 22 & 17 & 0\end{array}\right]\) \\
\hline 6 & \(\left[\begin{array}{lll}6 & 5 & 0\end{array}\right]\) & 25 & \(\left[\begin{array}{lll}25 & 22 & 0\end{array}\right]\) \\
\hline 7 & \(\left[\begin{array}{lll}7 & 6 & 0\end{array}\right]\) & 26 & \(\left[\begin{array}{lllll}26 & 25 & 24 & 20 & 0\end{array}\right]\) \\
\hline 8 & [ \(\left.\begin{array}{llllll}8 & 6 & 5 & 4 & 0\end{array}\right]\) & 27 & \(\left[\begin{array}{lllll}27 & 26 & 25 & 22 & 0\end{array}\right]\) \\
\hline 9 & \(\left[\begin{array}{lll}9 & 5 & 0\end{array}\right]\) & 28 & [28 25 0] \\
\hline 10 & [10 7 0] & 29 & \(\left[\begin{array}{lll}29 & 27 & 0\end{array}\right]\) \\
\hline 11 & \(\left[\begin{array}{lll}11 & 9 & 0\end{array}\right]\) & 30 & \(\left[\begin{array}{lllll}30 & 29 & 28 & 7 & 0\end{array}\right]\) \\
\hline 12 & \(\left[\begin{array}{lllll}12 & 11 & 8 & 6 & 0\end{array}\right]\) & 31 & \(\left[\begin{array}{lll}31 & 28 & 0\end{array}\right]\) \\
\hline 13 & \(\left[\begin{array}{llllll}13 & 12 & 10 & 9 & 0\end{array}\right]\) & 32 & \(\left[\begin{array}{llllll}32 & 31 & 30 & 10 & 0\end{array}\right]\) \\
\hline 14 & \(\left[\begin{array}{llllll}14 & 13 & 8 & 4 & 0\end{array}\right]\) & 33 & \(\left[\begin{array}{llll}33 & 20 & 0\end{array}\right]\) \\
\hline 15 & [ \(\left.\begin{array}{llll}15 & 14 & 0\end{array}\right]\) & 34 & \(\left[\begin{array}{llllll}34 & 15 & 14 & 1 & 0\end{array}\right]\) \\
\hline 16 & \(\left[\begin{array}{lllll}16 & 15 & 13 & 4 & 0\end{array}\right]\) & 35 & [35 2 0] \\
\hline 17 & \(\left[\begin{array}{llll}17 & 14 & 0\end{array}\right]\) & 36 & \(\left[\begin{array}{llll}36 & 11 & 0\end{array}\right]\) \\
\hline 18 & \(\left[\begin{array}{llll}18 & 11 & 0\end{array}\right]\) & 37 & [37 121010200\(]\) \\
\hline 19 & \(\left[\begin{array}{llllll}19 & 18 & 17 & 14 & 0\end{array}\right]\) & 38 & [38665lll \\
\hline 20 & \(\left[\begin{array}{llll}20 & 17 & 0\end{array}\right]\) & 39 & [39 800 \\
\hline 40 & \(\left[\begin{array}{lllll}40 & 5 & 4 & 3 & 0\end{array}\right]\) & 47 & [47 14 0] \\
\hline 41 & \(\left[\begin{array}{llll}41 & 3 & 0\end{array}\right]\) & 48 & [48 2827100\(]\) \\
\hline 42 & \(\left[\begin{array}{lllll}42 & 23 & 22 & 1 & 0\end{array}\right]\) & 49 & [49 9 0] \\
\hline 43 & \(\left[\begin{array}{lllll}43 & 6 & 4 & 3 & 0\end{array}\right]\) & 50 & [50 433200\(]\) \\
\hline 44 & \(\left[\begin{array}{lllll}44 & 6 & 5 & 2 & 0\end{array}\right]\) & 51 & [51 5131100\(]\) \\
\hline
\end{tabular}
\begin{tabular}{l|l|l|l}
\hline \(\mathbf{r}\) & \begin{tabular}{l} 
Generator \\
Polynomial
\end{tabular} & \(\mathbf{r}\) & Generator Polynomial \\
\hline 45 & {\(\left[\begin{array}{lllll}45 & 4 & 3 & 1 & 0\end{array}\right]\)} & 52 & {\(\left[\begin{array}{llll}5 & 3 & 0\end{array}\right]\)} \\
\hline 46 & {\(\left[\begin{array}{llllll}46 & 21 & 10 & 1 & 0\end{array}\right]\)} & 53 & {\(\left[\begin{array}{llll}5 & 6 & 2 & 1\end{array}\right]\)} \\
\hline
\end{tabular}

\section*{Examples Setting up the PN sequence generator}


This figure defines a PN sequence generator with a generator polynomial \(p(z)=z^{6}+z+1\). You can set up the PN sequence generator by typing the following at the MATLAB command line:
```

h1 = commsrc.pn('GenPoly', [1 0 0 0 0 1 1], 'Mask', [1 1 0 1 0 1]);
h2 = commsrc.pn('GenPoly', [1 0 0 0 0 1 1], 'Shift', 22);
mask2shift ([1 0 0 0 0 1 1],[1 1 0 1 0 1])

```

The output of the example is given below:
ans =
22
Alternatively, you can input GenPoly as the exponents of \(z\) for the nonzero terms of the polynomial in descending order of powers:
h = commsrc.pn('GenPoly', [ \(\left.\begin{array}{lll}6 & 1 & 0\end{array}\right]\), 'Mask', \(\left[\begin{array}{llllll}1 & 1 & 0 & 1 & 0 & 1\end{array}\right]\)

\section*{General Use of commsrc.pn}

The following is an example of typical usage:
```

% Construct a PN object
h = commsrc.pn('Shift', 0);
% Output 10 PN bits
set(h, 'NumBitsOut', 10);
generate(h)
% Output 10 more PN bits
generate(h)
% Reset (to the initial shift register state values)
reset(h);
% Output 4 PN bits
set(h, 'NumBitsOut', 4);
generate(h)

```

\section*{Behavior of a Copied commsrc.pn Object}

When a commsrc.pn object is copied, its states are also copied. The subsequent outputs, therefore, from the copied object are likely to be different from the initial outputs from the original object. The following code illustrates this behavior:
```

h = commsrc.pn('Shift', 0);
set(h, 'NumBitsOut', 5);
generate(h)

```
h generates the sequence:

However, if h is copied to g , and g is made to generate a sequence:
g=copy (h);
generate(g)
the generated sequence is different from that initially generated from h :

0
1
0
0
0

This difference occurs because the state of h having generated 5 bits was copied to g . If g is reset:
reset(g);
generate(g)
then it generates the same sequence that h did:

1
0
0
0
0

\section*{See Also}
mask2shift | shift2mask

\section*{Purpose}

\section*{Syntax}

Description

Default Simulink model settings for Communications System Toolbox software
```

commstartup

```
commstartup changes the default Simulink model settings to values more appropriate for the simulation of communication systems. The changes apply to new models that you create later in the MATLAB \({ }^{\circledR}\) session, but not to previously created models.

Note The DSP System Toolbox \({ }^{\text {TM }}\) application includes a similar dspstartup script, which assigns different model settings. For modeling communication systems, you should use commstartup alone.

To install the communications-related model settings each time you start MATLAB, invoke commstartup from your startup.m file.

To be more specific, the settings in commstartup cause models to:
- Use the variable-step discrete solver in single-tasking mode
- Use starting and ending times of 0 and Inf, respectively
- Avoid producing a warning or error message for inherited sample times in source blocks
- Set the Simulink Boolean logic signals parameter to Off
- Avoid saving output or time information to the workspace
- Produce an error upon detecting an algebraic loop
- Inline parameters if you use the Model Reference feature of Simulink

If your communications model does not work well with these default settings, you can change each of the individual settings as the model requires.

See Also startup

\section*{Purpose Create error rate test console}

Syntax
Description

\section*{Properties}
h = commtest.ErrorRate
h = commtest.ErrorRate(sys)
h = commtest.ErrorRate(sys,'PropertyName',PropertyValue,...)
h = commtest.ErrorRate('PropertyName',PropertyValue,...)
\(\mathrm{h}=\) commtest.ErrorRate returns an error rate test console, h. The error rate test console runs simulations of a system under test to obtain error rates.
h = commtest.ErrorRate(sys) returns an error rate test console, error rate test console, \(h\), with each specified property set to the \(h\), with an attached system under test, SYS.
h =
commtest.ErrorRate(sys,'PropertyName', PropertyValue,....) returns an error rate test console, \(h\), with an attached system under test, sys. Each specified property, 'PropertyName', is set to the specified value, PropertyValue.
h = commtest.ErrorRate('PropertyName',PropertyValue,...) returns an error rate test console, h, with each specified property 'PropertyName', set to the specified value, PropertyValue.

The error rate test console object has the properties in the following table. Setting any property resets the object. A property that is irrelevant is one that you can set, but its value does not affect measurements. Similarly, you cannot display irrelevant properties using the disp method. You can write to all properties, except for the ones explicitly noted otherwise.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline Description & 'Error Rate Test Console'. Read-only. \\
\hline SystemUnderTestName & System under test name. Read-only. \\
\hline
\end{tabular}
\begin{tabular}{|c|c|}
\hline Property & Description \\
\hline FrameLength & \begin{tabular}{l}
Specify the length of the transmission frame at each iteration. This property becomes relevant only when the system under test registers a valid test input. \\
- If the system under test registers a NumTransmissions test input and calls its get Input method, the error rate test console returns the value stored in FrameLength. Using an internal data source, the system under test uses this value to generate a transmission frame of the specified length. \\
- If the system under test registers a DiscreteRandomSource test input and calls its get Input method, the test console generates and returns a frame of symbols. The length of the frame of symbols matches the FrameLength property. This property defaults to 500 .
\end{tabular} \\
\hline IterationMode & \begin{tabular}{l}
Specify how the object determines simulation points. \\
- If set to Combinatorial, the object performs simulations for all possible combinations of registered test parameter sweep values. \\
- If set to Indexed, the object performs simulations for all indexed sweep value sets. The \(i^{\text {th }}\) sweep value set consists of the \(i^{\text {th }}\) element of every sweep value vector for each registered test parameter. All sweep value vectors must have equal length, except for values that are unit length. \\
Note that for the following sweep parameter settings: \\
- Parameter \(1=\left[a_{1} a_{2}\right]\) \\
- Parameter2 \(=\left[\mathrm{b}_{1} \mathrm{~b}_{2}\right]\) \\
- Parameter3 \(=\left[\mathrm{c}_{1}\right]\) \\
In Indexed Mode, the test console performs simulations for the following sweep parameter sets:
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{|c|c|}
\hline Property & Description \\
\hline & \begin{tabular}{l}
\[
\begin{aligned}
& \left(\mathrm{a}_{1}, \mathrm{~b}_{1}, \mathrm{c}_{1}\right) \\
& \left(\mathrm{a}_{2}, \mathrm{~b}_{2}, \mathrm{c}_{1}\right)
\end{aligned}
\] \\
In Combinatorial Mode, the test console performs simulations for the following sweep parameter sets:
\[
\begin{aligned}
& \left(a_{1}, b_{1}, c_{1}\right) \\
& \left(a_{1}, b_{2}, c_{1}\right) \\
& \left(a_{2}, b_{1}, c_{1}\right) \\
& \left(a_{2}, b_{2}, c_{1}\right)
\end{aligned}
\]
\end{tabular} \\
\hline SystemResetMode & \begin{tabular}{l}
Specify the stage of a simulation run at which the system resets. \\
- Setting to Reset at new simulation point resets the system under test at the beginning of a new simulation point. \\
- Setting to Reset at every iteration resets the system under test at every iteration.
\end{tabular} \\
\hline SimulationLimitOption & \begin{tabular}{l}
Specify how to stop the simulation for each sweep parameter point. \\
- If set to Number of transmissions the simulation for a sweep parameter point stops when the number of transmissions equals the value for MaxNumTransmissions. \\
- Set TransmissionCountTestPoint to the name of the registered test point containing the transmission count you are comparing to MaxNumTransmissions. \\
- If set to Number of errors the simulation for a sweep parameter point stops when the number of errors equals the value for MinNumErrors. \\
- Set the ErrorCountTestPoint to the name of the registered test point containing the error count you are comparing to the MinNumErrors.
\end{tabular} \\
\hline
\end{tabular}

\section*{Property \(\quad\) Description}
- Setting to Number of errors or transmissions stops the simulation for a sweep parameter point when meeting one of two conditions.
- The simulation stops when the number of transmissions equals the value for MaxNumTransmissions.
- The simulation stops when obtaining the number of errors matching NumErrors.
- Setting this property to Number of errors and transmissions stops the simulation for a sweep parameter point when meeting the following condition.
- The simulation stops when the number of transmissions and the number errors have at least reached the values in MinNumTransmissions and MinNumErrors.

Set TransmissionCountTestPoint to the name of the registered test point that contains the transmission count you are comparing to the MaxNumTransmissions property.

To control the simulation length, set ErrorCountTestPoint to the name of the registered test point containing the error count you are comparing to MinNumErrors.

Call the info method of the error rate test console to see the valid registered test point names.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline MaxNumTransmissions & \begin{tabular}{l} 
Specify the maximum number of transmissions the \\
object counts before stopping the simulation for \\
a sweep parameter point. This property becomes \\
relevant only when SimulationLimitOption is \\
Number of transmissions or Number of errors or \\
transmissions.
\end{tabular} \\
& \begin{tabular}{l} 
- When setting SimulationLimitoption to Number \\
of transmissions the simulation for each sweep \\
parameter point stops when reaching the number of \\
transmissions MaxNumTransmissions specifies.
\end{tabular} \\
- \begin{tabular}{l} 
Setting SimulationLimitOption to Number of errors \\
or transmissions stops the simulation for each sweep \\
parameter point for one of two conditions. \\
- The simulation stops when completing the number of \\
transmissions MaxNumTransmissions specifies. \\
- The simulation stops when obtaining the number of \\
errors MinNumErrors specifies.
\end{tabular} \\
& \begin{tabular}{l} 
The TransmissionCountTestPoint property supplies
\end{tabular} \\
the name of a registered test point containing the count \\
transmission type. Calling the info method of the \\
error rate test console displays the valid registered test \\
points. If this property contains registered test points, \\
the test console runs iterations equal to the value for \\
MaxNumTransmissions for each sweep parameter point. If \\
this property has no registered test parameters, the test \\
console runs the number of iterations equal to the value \\
for MaxNumTransmissions and stops. The value defaults \\
to 1000.
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline MinNumErrors & \begin{tabular}{l} 
Specify the minimum number of errors the object counts \\
before stoping the simulation for a sweep parameter \\
point. This property becomes relevant only when setting \\
the SimulationLimitoption to Number of errors or \\
Number of errors or transmissions.
\end{tabular} \\
- When setting SimulationLimitoption to Number of \\
errors the simulation for each parameter point stops \\
when reaching the number of errors you specify for the \\
MinNumErrors property. \\
- When setting the SimulationLimitoption property to \\
Number of errors or transmissions the simulation \\
for each sweep parameter point stops for one of two \\
conditions. \\
- The simulation stops when reaching the number \\
of errors you specify for the MaxNumTransmissions \\
property. \\
- The simulation stops when reaching the number of \\
errors you specify for the MinNumErrors property.
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline TransmissionCountTestPoint & \begin{tabular}{l} 
Specify and register a test point containing the \\
transmission count that controls the test console \\
simulation stop mechanism. This property becomes \\
relevant only when setting SimulationLimitOption \\
to Number of transmissions, Number of errors \\
or transmissions, or Number of errors and \\
transmissions. In this scenario, if you register a test \\
point, and TransmissionCountTestPoint equals Not set, \\
the value of this property automatically updates to that of \\
the registered test point name. Call the info method to \\
see the valid test point names.
\end{tabular} \\
\hline ErrorCountTestPoint & \begin{tabular}{l} 
Specify and register the name of a test point containing \\
the error count that controls the simulation stop \\
mechanism. This property is only relevant when setting \\
the SimulationLimitOption property to Number of \\
errors, Number of errors or transmissions, or
\end{tabular} \\
Number of errors and transmissions. In this scenario, \\
if you register a test point, and ErrorCountTestPoint \\
equals Not set, the value of this property automatically \\
updates to that of the registered test point name. Call the \\
info method to see the valid test point names.
\end{tabular}

\section*{Methods \\ The error rate test console object has the following methods:}

\section*{run}

Runs a simulation.
Runs the number of error rate simulations you specify for a system under test with a specified set of parameter values. If a Parallel Computing Toolbox \({ }^{\mathrm{TM}}\) license is available and a matlabpool is open, then the object distributes the iterations among the number of workers available.

\section*{getResults}

Returns the simulation results.
\(\mathrm{r}=\) getResults(h) returns the simulation results, \(r\), for the test console, \(h . r\) is an object of the type you specify using testconsole.Results. It contains the simulation data for all the registered test points and methods to parse the data and plot it.

\section*{info}

Returns a report of the current test console settings.
info(h) displays the current test console settings, such as registered test parameters and registered test points.

\section*{reset}

Resets the error rate test console.
reset(h) resets test parameters and test probes and then clears all simulation results of test console, \(h\).

\section*{attachSystem}

Attaches a system to test console.
attachSystem(ho,sys) attaches a valid user-defined system, sys, to the test console, h.

\section*{detachSystem}

Detaches the system from the test console.
detachSystem(h) detaches a system from the test console, h. This method also clears the registered test inputs, test parameters, test probes, and test points.

\section*{setTestParameterSweepValues}

Sets test parameter sweep values.
setTestParameterSweepValues(h,name,sweep) specifies a set of sweep values, 'sweep', for the registered test parameter, 'name', in the test console, h. You only specify sweep values for registered test parameters. sweep must have values within the specified range of the test parameter. It can be a row vector of numeric values,
or a cell array of char values. Display the valid ranges using the getTestParameterValidRanges method.
setTestParameterSweepValues(h,name1,sweep1,name2,sweep2...) simultaneously specifies sweep values for multiple registered test parameters.

\section*{getTestParameterSweepValues}

Returns test parameter sweep values.
getTestParameterSweepValues(h,name) gets the sweep values currently specified for the registered test parameter, name, in the test console, h.

\section*{getTestParameterValidRanges}

Returns the test parameter valid ranges.
getTestParameterValidRanges(h,name) gets the valid ranges for a registered test parameter, name, in the test console, \(h\).

\section*{registerTestPoint}

Registers a test point.
registerTestPoint(h, name, actprobe,expprobe) registers a new test point object, name, to the error rate test console, \(h\). The test point must contain a pair of registered test probes, actprobe, and expprobe. actprobe contains actual data, and expprobe contains expected data. The object compares the data from these probes and obtains error rate values. The error rate calculation uses a default error rate calculator function that simply performs one-to-one comparisons of the data vectors available in the probes.
registerTestPoint(h, name, actprobe, expprobe, handle) adds the handle, handle, to a user-defined error calculation function that compares the data in the probes and then obtains error rate results.

The user-defined error calculation function must comply with the following syntax: [ecnt tent] = functionName(act, exp, udata) where
- ecnt output corresponds to the error count
- tent output is the number of transmissions used to obtain the error count
- act and exp correspond to actual and expected data

The error rate test console sets the inputs to the data available in the pair of test point probes, actprobe, and expprobe.
udata is a data input that the system under test passes to the test console at run time, using the setUserData method. udata contains the data necessary to compute errors, such as delays and data buffers.

The error rate test console passes the data that the system under test logs to the error calculation functions for all the registered test points. Calling the info method returns the names of the registered test points and the error rate calculator functions associated with them. It also returns the names of the registered test probes.

\section*{unregisterTestPoint}

Unregister a test point.
unregisterTestPoint(h,name) removes the test point, name, from the test console, h.

\section*{Examples}
```

% Obtain bit error rate and symbol error rate of an M-PSK system
% for different modulation orders and EbNo values.
% Instantiate an ErrorRate test console. The default error rate
% test console has an M-PSK system attached.
h = commtest.ErrorRate;
% Set sweep values for simulation test parameters
setTestParameterSweepValues(h,'M',2.^[[$$
\begin{array}{llll}{1}&{2}&{3}\end{array}
$$])
setTestParameterSweepValues(h,'EbNo',(-5:5))
% Register test points
registerTestPoint(h,'SymbolErrorRate','TxInputSymbols',...,
'RxOutputSymbols')
registerTestPoint(h,'BitErrorRate','TxInputBits','RxOutputBits')

```
```

% Set simulation stop criteria.
h.TransmissionCountTestPoint = 'SymbolErrorRate';
% Get information about the simulation settings
info(h)
% Run the MPSK simulations
run(h)
% Get the results
R = getResults(h);
% Plot EbNo versus bit error rate for different values of modulation
% order M
R.TestParameter2 = 'M';
plot(R)

```

This example generates a figure similar to the following:


See Also
How To
testconsole.Results
- Running Simulations Using the Error Rate Test Console
- Error Rate Test Console

\section*{Purpose \\ Source code mu-law or A-law compressor or expander}

\section*{Syntax}
```

out = compand(in,param,v)
out = compand(in,Mu,v,'mu/compressor')
out = compand(in,Mu,v,'mu/expander')
out = compand(in,A,v,'A/compressor')
out = compand(in,A,v,'A/expander')

```

\section*{Description}
out = compand(in, param,v) implements a \(\mu\)-law compressor for the input vector in. Mu specifies \(\mu\), and \(v\) is the input signal's maximum magnitude. out has the same dimensions and maximum magnitude as in.
out = compand(in, Mu, v,'mu/compressor') is the same as the syntax above.
out = compand(in, Mu, v,'mu/expander') implements a \(\mu\)-law expander for the input vector in. Mu specifies \(\mu\) and \(v\) is the input signal's maximum magnitude. out has the same dimensions and maximum magnitude as in.
out = compand(in, A, v,'A/compressor') implements an A-law compressor for the input vector in. The scalar A is the A-law parameter, and \(v\) is the input signal's maximum magnitude. out is a vector of the same length and maximum magnitude as in.
out \(=\) compand(in, \(\mathrm{A}, \mathrm{v}\), 'A/expander') implements an A-law expander for the input vector in. The scalar A is the A-law parameter, and \(v\) is the input signal's maximum magnitude. out is a vector of the same length and maximum magnitude as in.

Note The prevailing parameters used in practice are \(\mu=255\) and A = 87.6.

\section*{Examples}

The examples below illustrate the fact that compressors and expanders perform inverse operations.
```

compressed = compand(1:5,87.6,5,'a/compressor')
expanded = compand(compressed,87.6,5,'a/expander')

```

The output is
compressed =
3.5296
4.1629
4.5333
4.7961
5.0000
expanded \(=\)
\[
\begin{array}{lllll}
1.0000 & 2.0000 & 3.0000 & 4.0000 & 5.0000
\end{array}
\]

\section*{Algorithms}

\section*{References}

For a given signal \(x\), the output of the \(\mu\)-law compressor is
\[
y=\frac{V \log (1+\mu|x| / V)}{\log (1+\mu)} \operatorname{sgn}(x)
\]
where \(V\) is the maximum value of the signal \(x, \mu\) is the \(\mu\)-law parameter of the compander, log is the natural logarithm, and sgn is the signum function (sign in MATLAB).

The output of the A-law compressor is
\[
y=\left\{\begin{array}{cl}
\frac{A|x|}{1+\log A} \operatorname{sgn}(x) & \text { for } 0 \leq|x| \leq \frac{V}{A} \\
\frac{V(1+\log (A|x| / V))}{1+\log A} \operatorname{sgn}(x) & \text { for } \frac{V}{A}<|x| \leq V
\end{array}\right.
\]
where \(A\) is the A-law parameter of the compander and the other elements are as in the \(\mu\)-law case.
[1] Sklar, Bernard, Digital Communications: Fundamentals and Applications, Englewood Cliffs, NJ, Prentice-Hall, 1988.

See Also quantiz \(\mid\) dpcmenco \(\mid\) dpcmdeco
How To . "Compand a Signal"

\section*{Purpose \\ Syntax \\ Description}

Restore ordering of symbols using shift registers
```

deintrlved = convdeintrlv(data,nrows,slope)
[deintrlved,state] = convdeintrlv(data,nrows,slope)
[deintrlved,state] = convdeintrlv(data,nrows,slope,
init_state)

```
deintrlved = convdeintrlv(data, nrows,slope) restores the ordering of elements in data by using a set of nrows internal shift registers. The delay value of the kth shift register is (nrows-k)*slope, where \(\mathrm{k}=1,2,3, \ldots\), nrows. Before the function begins to process data, it initializes all shift registers with zeros. If data is a matrix with multiple rows and columns, the function processes the columns independently.
[deintrlved,state] = convdeintrlv(data, nrows,slope) returns a structure that holds the final state of the shift registers. state. value stores any unshifted symbols. state.index is the index of the next register to be shifted.
[deintrlved,state] =
convdeintrlv(data, nrows, slope, init_state) initializes the shift registers with the symbols contained in init_state.value and directs the first input symbol to the shift register referenced by init_state.index. The structure init_state is typically the state output from a previous call to this same function, and is unrelated to the corresponding interleaver.

\section*{Using an Interleaver-Deinterleaver Pair}

To use this function as an inverse of the convintrlv function, use the same nrows and slope inputs in both functions. In that case, the two functions are inverses in the sense that applying convintrlv followed by convdeintrlv leaves data unchanged, after you take their combined delay of nrows* (nrows-1)*slope into account. To learn more about delays of convolutional interleavers, see "Delays of Convolutional Interleavers".
\begin{tabular}{ll} 
Examples & \begin{tabular}{l} 
The example in "Effect of Delays on Recovery of Convolutionally \\
Interleaved Data Using MATLAB" uses convdeintrlv and illustrates \\
how you can handle the delay of the interleaver/deinterleaver pair \\
when recovering data.
\end{tabular} \\
& \begin{tabular}{l} 
The example on the reference page for muxdeintrlv illustrates how to \\
use the state output and init_state input with that function; the \\
process is analogous for this function.
\end{tabular} \\
References & \begin{tabular}{l} 
[1] Heegard, Chris, and Stephen B. Wicker, Turbo Coding, Boston, \\
Kluwer Academic Publishers, 1999.
\end{tabular} \\
See Also & convintrlv | muxdeintrlv \\
How To & . "Interleaving"
\end{tabular}

\section*{Purpose Convolutionally encode binary data}
```

Syntax code = convenc(msg,trellis)
code = convenc(msg,trellis,puncpat)
code = convenc(msg,trellis,...,init_state)
[code,final_state] = convenc(...)

```

\section*{Description}

\section*{Examples}
code \(=\) convenc(msg,trellis) encodes the binary vector msg using the convolutional encoder whose MATLAB trellis structure is trellis. For details about MATLAB trellis structures, see "Trellis Description of a Convolutional Code". Each symbol in msg consists of log2(trellis.numInputSymbols) bits. The vector msg contains one or more symbols. The output vector code contains one or more symbols, each of which consists of log2(trellis. numOutputSymbols) bits.
code \(=\) convenc(msg,trellis, puncpat) is the same as the syntax above, except that it specifies a puncture pattern, puncpat, to allow higher rate encoding. puncpat must be a vector of 1 s and 0 s, where the Os indicate the punctured bits. puncpat must have a length of at least log2(trellis.numOutputSymbols) bits.
code \(=\) convenc(msg,trellis,...,init_state) allows the encoder registers to start at a state specified by init_state. init_state is an integer between 0 and trellis.numStates -1 and must be the last input parameter.
[code,final_state] = convenc(...) encodes the input message and also returns the encoder's state in final_state. final_state has the same format as init_state.

Encodes five two-bit symbols using a rate \(2 / 3\) convolutional code. A schematic of this encoder is on the poly2trellis reference page.
```

s = RandStream.create('mt19937ar', 'seed',123);
prevStream = RandStream.setGlobalStream(s); % Set stream for repeatab
code1 = convenc(randi([0 1],10,1),···
poly2trellis([5 4],[23 35 0; 0 5 13]));
RandStream.setGlobalStream(prevStream); % Restore default stream

```

The following syntax defines the encoder's trellis structure explicitly and then uses convenc to encode 10 one-bit symbols. A schematic of this encoder is in "Trellis Description of a Convolutional Code".
```

trel = struct('numInputSymbols',2,'numOutputSymbols',4,···.
'numStates',4,'nextStates',[0 2;0 2;1 3;1 3],...
'outputs',[0 3;1 2;3 0;2 1]);
code2 = convenc(randi([0 1],10,1),trel);

```

The following syntax illustrates how to use the final state and initial state arguments when invoking convenc repeatedly. Notice that [code3; code4] is the same as the earlier example's output, code1.
```

s = RandStream.create('mt19937ar', 'seed',123);
prevStream = RandStream.setGlobalStream(s); % Set stream for repeatabilit
trel = poly2trellis([5 4],[23 35 0; 0 5 13]);
msg = randi([0 1],10,1);
% Encode part of msg, recording final state for later use.
[code3,fstate] = convenc(msg(1:6),trel);
% Encode the rest of msg, using state as an input argument.
code4 = convenc(msg(7:10),trel,fstate);
RandStream.setGlobalStream(prevStream); % Restore default stream

```

\section*{Examples}

\section*{References}

For some commonly used puncture patterns for specific rates and polynomials, see the last three references.
[1] Clark, G. C. Jr. and J. Bibb Cain., Error-Correction Coding for Digital Communications, New York, Plenum Press, 1981.
[2] Gitlin, Richard D., Jeremiah F. Hayes, and Stephen B. Weinstein, Data Communications Principles, New York, Plenum, 1992.
[3] Yasuda, Y., et. al., "High rate punctured convolutional codes for soft decision Viterbi decoding," IEEE Transactions on Communications, vol. COM-32, No. 3, pp 315-319, Mar. 1984.
[4] Haccoun, D., and G. Begin, "High-rate punctured convolutional codes for Viterbi and sequential decoding," IEEE Transactions on Communications, vol. 37, No. 11, pp 1113-1125, Nov. 1989.
[5] Begin, G., et.al., "Further results on high-rate punctured convolutional codes for Viterbi and sequential decoding," IEEE Transactions on Communications, vol. 38, No. 11, pp 1922-1928, Nov. 1990.

\section*{See Also \\ distspec | vitdec | poly2trellis | istrellis}

How To . "Convolutional Codes"

Purpose Permute symbols using shift registers
```

Syntax intrlved = convintrlv(data,nrows,slope)
[intrlved,state] = convintrlv(data,nrows,slope)
[intrlved,state] = convintrlv(data,nrows,slope,init_state)

```

\section*{Description}

\section*{Examples}
intrlved = convintrlv(data, nrows, slope) permutes the elements in data by using a set of nrows internal shift registers. The delay value of the kth shift register is ( \(\mathrm{k}-1\) ) *slope, where \(\mathrm{k}=1,2,3, \ldots\) nrows. Before the function begins to process data, it initializes all shift registers with zeros. If data is a matrix with multiple rows and columns, the function processes the columns independently.
[intrlved,state] = convintrlv(data, nrows,slope) returns a structure that holds the final state of the shift registers. state. value stores any unshifted symbols. state.index is the index of the next register to be shifted.
[intrlved,state] = convintrlv(data, nrows,slope,init_state) initializes the shift registers with the symbols contained in init_state.value and directs the first input symbol to the shift register referenced by init_state.index. The structure init_state is typically the state output from a previous call to this same function, and is unrelated to the corresponding deinterleaver.

The example below shows that convintrlv is a special case of the more general function muxintrlv. Both functions yield the same numerical results.
```

x = randi([0 1],100,1); % Original data
nrows = 5; % Use 5 shift registers
slope = 3; % Delays are 0, 3, 6, 9, and 12.
y = convintrlv(x,nrows,slope); % Interleaving using convintrlv.
delay = [0:3:12]; % Another way to express set of delays
y1 = muxintrlv(x,delay); % Interleave using muxintrlv.
isequal(y,y1)

```

The output below shows that \(y\), obtained using convintrlv, and y 1 , obtained using muxintrlv, are the same.
ans \(=\)

1

Another example using this function is in "Effect of Delays on Recovery of Convolutionally Interleaved Data Using MATLAB".

The example on the muxdeintrlv reference page illustrates how to use the state output and init_state input with that function; the process is analogous for this function.

\section*{References}

See Also
How To . "Interleaving"
[1] Heegard, Chris, and Stephen B. Wicker, Turbo Coding, Boston, Kluwer Academic Publishers, 1999.
convdeintrlv | muxintrlv | helintrlv

\section*{Purpose Convolution matrix of Galois field vector}

\section*{Syntax \(\quad A=\operatorname{convmtx}(c, n)\)}

Description A convolution matrix is a matrix, formed from a vector, whose inner product with another vector is the convolution of the two vectors.
\(\mathrm{A}=\) convmtx \((\mathrm{c}, \mathrm{n})\) returns a convolution matrix for the Galois vector c. The output A is a Galois array that represents convolution with c in the sense that conv ( \(c, x\) ) equals
- A*x, if \(c\) is a column vector and \(x\) is any Galois column vector of length \(n\). In this case, \(A\) has \(n\) columns and \(m+n-1\) rows.
- \(x * A\), if \(c\) is a row vector and \(x\) is any Galois row vector of length \(n\). In this case, \(A\) has \(n\) rows and \(m+n-1\) columns.

\section*{Examples The code below illustrates the equivalence between using the conv} function and multiplying by the output of convmtx.
```

m = 4;
c = gf([1; 9; 3],m); % Column vector
n = 6;
x = gf(randi([0 2^m-1],n,1),m);
ck1 = isequal(conv(c,x), convmtx(c,n)*x) % True
ck2 = isequal(conv(c',x'), x'*convmtx(c',n)) % True

```

The output is
```

ck1 =

```

1
ck2 =
\(\begin{array}{ll}\text { See Also conv } \\ \text { How To } & \text {. "Signal Processing Operations in Galois Fields" }\end{array}\)

\section*{Purpose Produce cyclotomic cosets for Galois field}

\section*{Syntax \\ cst = cosets(m)}

Description

\section*{Examples}
cst \(=\) cosets (m) produces cyclotomic cosets mod \(2^{\wedge} m-1\). Each element of the cell array cst is a Galois array that represents one cyclotomic coset.

A cyclotomic coset is a set of elements that share the same minimal polynomial. Together, the cyclotomic cosets \(\bmod 2^{\wedge} m-1\) form a partition of the group of nonzero elements of \(\mathrm{GF}\left(\mathbf{2}^{\wedge} \mathrm{m}\right)\). For more details on cyclotomic cosets, see the works listed in "References" on page 1-163.

The commands below find and display the cyclotomic cosets for GF(8). As an example of interpreting the results, \(\mathrm{c}\{2\}\) indicates that \(\mathrm{A}, \mathrm{A}^{2}\), and \(\mathrm{A}^{2}+\mathrm{A}\) share the same minimal polynomial, where A is a primitive element for GF(8).
c = cosets(3);
c \{1\}
c \(\{2\}\) '
c \(\{3\) \}
The output is below.
```

ans = GF(2^3) array. Primitive polynomial = D^3+D+1 (11 decimal)
Array elements =

```

1
ans \(=G F\left(2^{\wedge} 3\right)\) array. Primitive polynomial \(=D^{\wedge} 3+D+1\) (11 decimal)

Array elements =
246
```

ans = GF(2^3) array. Primitive polynomial = D^3+D+1 (11 decimal)
Array elements =
3 5

```

\author{
References [1] Blahut, Richard E., Theory and Practice of Error Control Codes, Reading, MA, Addison-Wesley, 1983, p. 105. \\ [2] Lin, Shu, and Daniel J. Costello, Jr., Error Control Coding: Fundamentals and Applications, Englewood Cliffs, NJ, Prentice-Hall, 1983.
}

\section*{See Also \\ minpol}
\begin{tabular}{|c|c|}
\hline Purpose & Construct CRC detector object \\
\hline \multirow[t]{4}{*}{Syntax} & \(\mathrm{h}=\) crc.detector(polynomial) \\
\hline & \(\mathrm{h}=\mathrm{crc}\). detector (generatorObj) \\
\hline & ```
h= crc.detector(`Polynomial', polynomial, `param1', val1,
etc.)
``` \\
\hline & \(\mathrm{h}=\mathrm{crc}\). detector \\
\hline \multirow[t]{7}{*}{Description} & \(\mathrm{h}=\) crc.detector(polynomial) constructs a CRC detector object H defined by the generator polynomial POLYNOMIAL \\
\hline & \(\mathrm{h}=\mathrm{crc}\).detector (generatorObj) constructs a CRC detector object \(H\) defined by the parameters found in the CRC generator object GENERATOROBJ \\
\hline & \(\mathrm{h}=\) crc.detector(`property1', val1, ...) constructs a CRC detector object H with properties as specified by PROPERTY/VALUE pairs. \\
\hline & \(\mathrm{h}=\mathrm{crc}\).detector constructs a CRC detector object H with default properties. It constructs a CRC-CCITT detector, and is equivalent to: \\
\hline & ```
h= crc.detector('Polynomial', '0x1021', 'InitialState',
'0xFFFF', 'ReflectInput', ...
``` \\
\hline & false, 'ReflectRemainder', false, 'FinalXOR', '0x0000') \\
\hline & Properties \\
\hline
\end{tabular}

The following table describes the properties of a CRC detector object. All properties are writable, except Type.
\begin{tabular}{|c|c|}
\hline Property & Description \\
\hline Type & Specifies the object as a 'CRC Detector'. \\
\hline Polynomial & The generator polynomial that defines connections for a linear feedback shift register. This property can be specified as a binary vector representing descending powers of the polynomial. In this case, the leading ' 1 ' of the polynomial must be included. It can also be specified as a string, prefaced by ' 0 x ', that is a hexadecimal representation of the descending powers of the polynomial. In this case, the leading ' 1 ' of the polynomial is omitted. \\
\hline InitialState & The initial contents of the shift register. This property can be specified as a binary scalar, a binary vector, or as a string, prefaced by ' 0 x ', that is a hexadecimal representation of the binary vector. As a binary vector, its length must be one less than the length of the binary vector representation of the Polynomial. \\
\hline ReflectInput & A Boolean quantity that specifies whether the input data should be flipped on a bytewise basis prior to entering the shift register. \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline ReflectRemainder & \begin{tabular}{l} 
A Boolean quantity that specifies \\
whether the binary output CRC \\
checksum should be flipped \\
around its center after the input \\
data is completely through the \\
shift register.
\end{tabular} \\
\hline FinalXOR & \begin{tabular}{l} 
The value with which the CRC \\
checksum is to be XORed just \\
prior to detecting the input data. \\
This property can be specified as \\
a binary scalar, a binary vector or \\
as a string, prefaced by '0x', that \\
is a hexadecimal representation \\
of the binary vector. As a binary \\
vector, its length must be one \\
less than the length of the binary \\
vector representation of the \\
Polynomial.
\end{tabular} \\
\hline
\end{tabular}

A detect method is used with the object to detect errors in digital transmission.

\section*{CRC Generation Algorithm}

For information pertaining to the CRC generation algorithm, see in the Communications System Toolbox User's Guide.

\section*{Detector Method}
[OUTDATA ERROR] = DETECT(H, INDATA) detects transmission errors in the encoded input message INDATA by regenerating a CRC checksum using the CRC detector object H. The detector then compares the regenerated checksum with the checksum appended to INDATA. The binary-valued INDATA can be either a column vector or a matrix. If it is a matrix, each column is considered to be a separate channel. OUTDATA is identical to the input message INDATA, except that it has the CRC checksum stripped off. ERROR is a 1 xC logical vector
indicating if the encoded message INDATA has errors, where C is the number of channels in INDATA. An ERROR value of 0 indicates no errors, and a value of 1 indicates errors.

\section*{Examples}

The following three examples demonstrate the use of constructing an object. The fourth example demonstrates use of the detect method.
```

% Construct a CRC detector with a polynomial
% defined by x^4+x^3+x^2+x+1:
h = crc.detector([[1 1 1 1 1 1 1])

```

This example generates the following output:
```

h =
Type: CRC Detector
Polynomial: OxF
InitialState: 0x0
ReflectInput: false
ReflectRemainder: false
FinalXOR: OxO
% Construct a CRC detector with a polynomial
% defined by x^3+x+1, with
% zero initial states, and with an all-ones
% final XOR value:
h = crc.detector('Polynomial', [1 0 1 1], ...
'InitialState', [0 O 0], 'FinalXOR', [$$
\begin{array}{lll}{1}&{1}&{1}\end{array}
$$])

```

This example generates the following output:
\(\mathrm{h}=\)
Type: CRC Detector
Polynomial: [ \(\left.\begin{array}{llll}1 & 0 & 1 & 1\end{array}\right]\)
InitialState: [0 0 0]
ReflectInput: false
ReflectRemainder: false
```

% Construct a CRC detector with a polynomial
% defined by x^4+x^3+x^2+x+1,
% all-ones initial states, reflected input, and all-zeros
% final XOR value:
h = crc.detector('Polynomial', 'OxF', 'InitialState', ...
'0xF', 'ReflectInput', true, 'FinalXOR', 'Ox0')

```

This example generates the following output:
\(\mathrm{h}=\)

Type: CRC Detector
Polynomial: OxF InitialState: OxF ReflectInput: true ReflectRemainder: false

FinalXOR: 0x0
\% Create a CRC-16 CRC generator, then use it to generate
\% a checksum for the
\% binary vector represented by the
\% ASCII sequence '123456789'.
\% Introduce an error, then detect it
\% using a CRC-16 CRC detector.
gen = crc.generator('Polynomial', '0x8005', 'ReflectInput', ...
true, 'ReflectRemainder', true);
det = crc.detector('Polynomial', '0x8005', 'ReflectInput', ...
true, 'ReflectRemainder', true);
\% The message below is an ASCII representation
\% of the digits 1-9
msg = reshape(de2bi(49:57, 8, 'left-msb')', 72, 1);
encoded = generate(gen, msg);
encoded(1) = ~encoded(1); \% Introduce an error
[outdata error] = detect(det, encoded); \% Detect the error
noErrors = isequal(msg, outdata) \% Should be 0
error \% Should be 1

This example generates the following output:
noErrors \(=\)
0
error \(=\)
1
See Also
crc.generator

\section*{Purpose Construct CRC generator object}
```

Syntax
h = crc.generator(polynomial)
h = crc.generator(detectorObj)
h = crc.generator(`Polynomial', polynomial, `param1', val1,
etc.)
h = crc.generator

```

\section*{Description}
\(\mathrm{h}=\mathrm{crc}\). generator (polynomial) constructs a CRC generator object H defined by the generator polynomial POLYNOMIAL.
\(\mathrm{h}=\mathrm{crc}\). generator (detectorObj) constructs a CRC generator object H defined by the parameters found in the CRC detector object DETECTOROBJ.
h = crc.generator(`property1', val1, ...) constructs a CRC generator object H with properties as specified by the PROPERTY/VALUE pairs.
\(\mathrm{h}=\mathrm{crc}\). generator constructs a CRC generator object H with default properties. It constructs a CRC-CCITT generator, and is equivalent to: \(\mathrm{h}=\) crc.generator('Polynomial', '0x1021', 'InitialState', '0xFFFF', ...
'ReflectInput', false, 'ReflectRemainder', false, 'FinalXOR', '0x0000').

\section*{Properties}

The following table describes the properties of a CRC generator object. All properties are writable, except Polynomial.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline Polynomial & \begin{tabular}{l} 
The generator polynomial that \\
defines connections for a linear \\
feedback shift register. This \\
property can be specified as \\
a binary vector representing \\
descending powers of the \\
polynomial. In this case, the \\
leading '1' of the polynomial \\
must be included. It can also be \\
specified as a string, prefaced \\
by '0x', that is a hexadecimal \\
representation of the descending \\
powers of the polynomial. In \\
this case, the leading ' '1' of the \\
polynomial is omitted.
\end{tabular} \\
\hline InitialState & \begin{tabular}{l} 
The initial contents of the \\
shift register. This property \\
can be specified as a binary
\end{tabular} \\
scalar, a binary vector, or as a \\
string, prefaced by 'ox', that is a \\
hexadecimal representation of the \\
binary vector. As a binary vector, \\
its length must be one less than \\
the length of the binary vector \\
representation of the Polynomial.
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline ReflectRemainder & \begin{tabular}{l} 
A Boolean quantity that specifies \\
whether the binary output CRC \\
checksum should be flipped \\
around its center after the input \\
data is completely through the \\
shift register.
\end{tabular} \\
\hline FinalXOR & \begin{tabular}{l} 
The value with which the CRC \\
checksum is to be XORed just \\
prior to being appended to \\
the input data. This property \\
can be specified as a binary \\
scalar, a binary vector, or as a \\
string, prefaced by '0x', that is a \\
hexadecimal representation of the \\
binary vector. As a binary vector, \\
its length must be one less than \\
the length of the binary vector \\
representation of the Polynomial.
\end{tabular} \\
\hline
\end{tabular}

\section*{CRC Generation Algorithm}

For information pertaining to the CRC generation algorithm, refer to the "CRC Non-Direct Algorithm" section of the Communications System Toolbox User's Guide.

\section*{Generator Method}
encoded \(=\) generate \((\mathrm{h}, \mathrm{msg})\) generates a CRC checksum for an input message using the CRC generator object H. It appends the checksum to the end of MSG. The binary-valued MSG can be either a column vector or a matrix. If it is a matrix, then each column is considered to be a separate channel.

\section*{Usage Example}

The following examples demonstrate the use of this object.
```

% Construct a CRC generator with a polynomial defined
% by x^4+x^3+x^2+x+1:
h = crc.generator([[1 1 1 1 1 1 1])
% Construct a CRC generator with a polynomial defined
% by x^4+x^3+x^2+x+1, all-ones initial states, reflected
% input, and all-zeros final XOR value:
h = crc.generator('Polynomial', 'OxF', 'InitialState', ...
'0xF', 'ReflectInput', true, 'FinalXOR', '0xO')
% Create a CRC-16 CRC generator, then use it to generate
% a checksum for the
% binary vector represented by the ASCII sequence '123456789'.
gen = crc.generator('Polynomial', '0x8005', ...
'ReflectInput', true, 'ReflectRemainder', true);
% The message below is an ASCII representation of ...
% the digits 1-9
msg = reshape(de2bi(49:57, 8, 'left-msb')', 72, 1);
encoded = generate(gen, msg);
% Construct a CRC generator with a polynomial defined
% by x^3+x+1, with zero initial states,
% and with an all-ones final XOR value:
h = crc.generator('Polynomial', [1 0 1 1], ...
'InitialState', [0 0 0], ...
'FinalXOR', [$$
\begin{array}{lll}{1}&{1}&{1}\end{array}
$$]

```

\author{
See Also \\ crc.detector
}

Purpose
Produce parity-check and generator matrices for cyclic code
Syntax
\(\mathrm{h}=\operatorname{cyclgen}(\mathrm{n}, \mathrm{pol})\)
\(h=c y c l g e n(n, p o l, o p t)\)
[h,g] = cyclgen(...)
[h,g,k] = cyclgen(...)

\section*{Description}

\section*{Examples}

For all syntaxes, the codeword length is \(n\) and the message length is \(k\). A polynomial can generate a cyclic code with codeword length \(n\) and message length \(k\) if and only if the polynomial is a degree-( \(n-k\) ) divisor of \(x^{\wedge} n-1\). (Over the binary field GF(2), \(x^{\wedge} n-1\) is the same as \(x^{\wedge} n+1\).) This implies that \(k\) equals \(n\) minus the degree of the generator polynomial.
\(\mathrm{h}=\operatorname{cyclgen}(\mathrm{n}, \mathrm{pol})\) produces an ( \(\mathrm{n}-\mathrm{k}\) )-by-n parity-check matrix for a systematic binary cyclic code having codeword length \(n\). The row vector pol gives the binary coefficients, in order of ascending powers, of the degree-( \(n-k\) ) generator polynomial.
\(\mathrm{h}=\operatorname{cyclgen}(\mathrm{n}, \mathrm{pol}, o p t)\) is the same as the syntax above, except that the argument opt determines whether the matrix should be associated with a systematic or nonsystematic code. The values for opt are 'system' and 'nonsys'.
\([h, g]=\operatorname{cyclgen}(\ldots)\) is the same as \(h=\operatorname{cyclgen}(\ldots)\), except that it also produces the k-by-n generator matrix g that corresponds to the parity-check matrix \(h\).
\([h, g, k]=\operatorname{cyclgen}(\ldots)\) is the same as \([h, g]=\operatorname{cyclgen}(\ldots)\), except that it also returns the message length \(k\).

The code below produces parity-check and generator matrices for a binary cyclic code with codeword length 7 and message length 4 .
```

pol = cyclpoly(7,4);
[parmat,genmat,k] = cyclgen(7,pol)

```

The output is
parmat \(=\)
\begin{tabular}{lllllll}
1 & 0 & 0 & 1 & 1 & 1 & 0 \\
0 & 1 & 0 & 0 & 1 & 1 & 1 \\
0 & 0 & 1 & 1 & 1 & 0 & 1 \\
& & & & & & \\
genmat \(=\) \\
& & & & & & \\
1 & 0 & 1 & 1 & 0 & 0 & 0 \\
1 & 1 & 1 & 0 & 1 & 0 & 0 \\
1 & 1 & 0 & 0 & 0 & 1 & 0 \\
0 & 1 & 1 & 0 & 0 & 0 & 1
\end{tabular}

In the output below, notice that the parity-check matrix is different from parmat above, because it corresponds to a nonsystematic cyclic code. In particular, parmatn does not have a 3 -by- 3 identity matrix in its leftmost three columns, as parmat does.
```

parmatn = cyclgen(7,cyclpoly(7,4),'nonsys')
parmatn =

```
\begin{tabular}{lllllll}
1 & 1 & 1 & 0 & 1 & 0 & 0 \\
0 & 1 & 1 & 1 & 0 & 1 & 0 \\
0 & 0 & 1 & 1 & 1 & 0 & 1
\end{tabular}

\section*{See Also}
encode | decode | bchgenpoly | cyclpoly
How To . "Block Codes"

Purpose
Produce generator polynomials for cyclic code
Syntax
pol \(=\) cyclpoly( \(n, k\) )
pol \(=\) cyclpoly( \(n, k, o p t)\)

\section*{Description}

For all syntaxes, a polynomial is represented as a row containing the coefficients in order of ascending powers.
pol \(=\) cyclpoly ( \(n, k\) ) returns the row vector representing one nontrivial generator polynomial for a cyclic code having codeword length n and message length k .
pol \(=\) cyclpoly ( \(\mathrm{n}, \mathrm{k}, \mathrm{opt}\) ) searches for one or more nontrivial generator polynomials for cyclic codes having codeword length \(n\) and message length \(k\). The output pol depends on the argument opt as shown in the table below.
\begin{tabular}{l|l|l}
\hline opt & Significance of pol & Format of pol \\
\hline 'min' & \begin{tabular}{l} 
One generator \\
polynomial having \\
the smallest possible \\
weight
\end{tabular} & \begin{tabular}{l} 
Row vector \\
representing the \\
polynomial
\end{tabular} \\
\hline 'max' & \begin{tabular}{l} 
One generator \\
polynomial having \\
the greatest possible \\
weight
\end{tabular} & \begin{tabular}{l} 
Row vector \\
representing the \\
polynomial
\end{tabular} \\
\hline 'all' & \begin{tabular}{l} 
All generator \\
polynomials M
\end{tabular} & \begin{tabular}{l} 
Matrix, each row of \\
which represents one \\
such polynomial
\end{tabular} \\
\hline a positive integer, L & \begin{tabular}{l} 
All generator \\
polynomials having \\
weight L
\end{tabular} & \begin{tabular}{l} 
Matrix, each row of \\
which represents one \\
such polynomial
\end{tabular} \\
\hline
\end{tabular}

The weight of a binary polynomial is the number of nonzero terms it has. If no generator polynomial satisfies the given conditions, the output pol is empty and a warning message is displayed.

\section*{Examples}

The first command below produces representations of three generator polynomials for a [15,4] cyclic code. The second command shows that \(1+x+x^{2}+x^{3}+x^{5}+x^{7}+x^{8}+x^{11}\) is one such polynomial having the largest number of nonzero terms.
c1 = cyclpoly(15,4,'all')
c2 \(=\) cyclpoly (15,4,'max')
The output is
```

c1 =

```

Columns 1 through 10

Columns 11 through 12
\begin{tabular}{ll}
1 & 1 \\
1 & 1 \\
0 & 1
\end{tabular}
c2 \(=\)

Columns 1 through 10
\(\begin{array}{llllllllll}1 & 1 & 1 & 1 & 0 & 1 & 0 & 1 & 1 & 0\end{array}\)

Columns 11 through 12

\section*{\(0 \quad 1\)}

This command shows that no generator polynomial for a [15,4] cyclic code has exactly three nonzero terms.
```

c3 = cyclpoly(15,4,3)

```

Warning: No cyclic generator polynomial satisfies the given constraints. > In cyclpoly at 131
c3 =
[]
\begin{tabular}{ll} 
Algorithms & \begin{tabular}{l} 
If opt is 'min', 'ma \\
converting decimal i \\
gfprimfd returns the \\
appropriate conditio \\
gfprimfd.
\end{tabular} \\
See Also & cyclgen | encode \\
How To & . "Block Codes"
\end{tabular}

\title{
Purpose \\ Convert decimal numbers to binary vectors \\ Syntax \\ b = de2bi(d) \\ b = de2bi(d,n) \\ b = de2bi(d,n,p) \\ b = de2bi(d,[],p) \\ b \(=\operatorname{de2bi}(d, \ldots, f l g)\) \\ Description \\ b = de2bi(d) converts a nonnegative decimal integer d to a binary row vector. If \(d\) is a vector, the output \(b\) is a matrix, each row of which is the binary form of the corresponding element in d . If d is a matrix, de2bi treats it like the vector \(d(:)\).
}

Note By default, de2bi uses the first column of b as the lowest-order digit.
\(b=\operatorname{de2bi}(d, n)\) is the same as \(b=d e 2 b i(d)\), except that its output has n columns, where n is a positive integer. An error occurs if the binary representations would require more than n digits. If necessary, the binary representation of \(d\) is padded with extra zeros.
\(b=d e 2 b i(d, n, p)\) converts a nonnegative decimal integer \(d\) to \(a\) base-p row vector, where \(p\) is an integer greater than or equal to 2 . The first column of b is the lowest base-p digit. b is padded with extra zeros if necessary, so that it has \(n\) columns, where \(n\) is a positive integer. An error occurs if the base-p representations would require more than \(n\) digits. If \(d\) is a nonnegative decimal vector, the output \(b\) is a matrix, each row of which is the (possibly zero-padded) base-p form of the corresponding element in d . If d is a matrix, de2bi treats it like the vector \(\mathrm{d}(:)\).
\(b=d e 2 b i(d,[], p)\) specifies the base \(p\) but not the number of columns.
b = de2bi(d,...,flg) uses the string \(f l g\) to determine whether the first column of \(b\) contains the lowest-order or highest-order
digits. Values for \(f l g\) are 'right-msb' and 'left-msb'. The value 'right-msb' produces the default behavior.

\section*{Examples}

The code below counts to 10 in decimal and binary.
```

d = (1:10)';
b = de2bi(d);
disp(' Dec Binary ')
disp(' ----- -----------------')
disp([d, b])

```

The output is below.
\begin{tabular}{|c|c|c|c|c|}
\hline Dec & & \multicolumn{3}{|l|}{Binary} \\
\hline 1 & 1 & 0 & 0 & 0 \\
\hline 2 & 0 & 1 & 0 & 0 \\
\hline 3 & 1 & 1 & 0 & 0 \\
\hline 4 & 0 & 0 & 1 & 0 \\
\hline 5 & 1 & 0 & 1 & 0 \\
\hline 6 & 0 & 1 & 1 & 0 \\
\hline 7 & 1 & 1 & 1 & 0 \\
\hline 8 & 0 & 0 & 0 & 1 \\
\hline 9 & 1 & 0 & 0 & 1 \\
\hline 10 & 0 & 1 & 0 & 1 \\
\hline
\end{tabular}

The command below shows how de2bi pads its output with zeros.
```

bb = de2bi([3 9],5) % Zero-padding the output
bb =

```
\begin{tabular}{lllll}
1 & 1 & 0 & 0 & 0 \\
1 & 0 & 0 & 1 & 0
\end{tabular}

The commands below show how to convert a decimal integer to base three without specifying the number of columns in the output matrix.

They also show how to place the most significant digit on the left instead of on the right.
t = de2bi(12,[],3) \% Convert 12 to base 3.
tleft \(=\) de2bi(12,[],3,'left-msb') \% Significant digit on left

The output is
t =
\(\begin{array}{lll}0 & 1 & 1\end{array}\)
tleft =
\(1 \quad 1 \quad 0\)

See Also bi2de

Purpose
Block decoder
```

Syntax
msg = decode(code,n,k,'hamming/fmt',prim_poly)
msg = decode(code,n,k,'linear/fmt',genmat,trt)
msg = decode(code,n,k,'cyclic/fmt',genpoly,trt)
msg = decode(code,n,k)
[msg,err] = decode(...)
[msg,err,ccode] = decode(...)
[msg,err,ccode,cerr] = decode(...)

```

\section*{Optional Inputs}
\begin{tabular}{l|l}
\hline Input & Default Value \\
\hline fmt & binary \\
\hline prim_poly & gfprimdf \((\mathrm{m})\) where \(\mathrm{n}=2^{\wedge} \mathrm{m}-1\) \\
\hline genpoly & cyclpoly \((\mathrm{n}, \mathrm{k})\) \\
\hline trt & \begin{tabular}{l} 
Uses syndtable to create \\
the syndrome decoding table \\
associated with the method’s \\
parity-check matrix
\end{tabular} \\
\hline
\end{tabular}

\section*{Description}

\section*{For All Syntaxes}

The decode function aims to recover messages that were encoded using an error-correction coding technique. The technique and the defining parameters must match those that were used to encode the original signal.

The "For All Syntaxes" on page 1-234 section on the encode reference page explains the meanings of \(n\) and \(k\), the possible values of \(f m t\), and the possible formats for code and msg. You should be familiar with the conventions described there before reading the rest of this section. Using the decode function with an input argument code that was not created by the encode function might cause errors.

\section*{For Specific Syntaxes}
msg = decode(code, \(\mathrm{n}, \mathrm{k}\), 'hamming/fmt', prim_poly) decodes code using the Hamming method. For this syntax, \(n\) must have the form \(2^{m}-1\) for some integer \(m\) greater than or equal to 3 , and \(k\) must equal \(n-m\). prim_poly is a row vector that gives the binary coefficients, in order of ascending powers, of the primitive polynomial for GF \(\left(2^{\mathrm{m}}\right)\) that is used in the encoding process. The default value of prim_poly is gfprimdf(m). The decoding table that the function uses to correct a single error in each codeword is syndtable(hammgen(m)).
msg = decode(code, \(\mathrm{n}, \mathrm{k}\), 'linear/fmt', genmat, trt) decodes code, which is a linear block code determined by the \(k\)-by-n generator matrix genmat. genmat is required as input. decode tries to correct errors using the decoding table trt, where trt is a \(2^{\wedge}(n-k)\)-by-n matrix.
msg = decode(code, \(\mathrm{n}, \mathrm{k}\), 'cyclic/fmt', genpoly,trt) decodes the cyclic code code and tries to correct errors using the decoding table trt, where trt is a \(2^{\wedge}(n-k)\)-by-n matrix. genpoly is a row vector that gives the coefficients, in order of ascending powers, of the binary generator polynomial of the code. The default value of genpoly is cyclpoly ( \(n, k\) ). By definition, the generator polynomial for an [ \(n, k\) ] cyclic code must have degree \(n-k\) and must divide \(x^{n}-1\).
```

msg = decode(code,n,k) is the same as
msg = decode(code,n,k,'hamming/binary').

```
[msg,err] = decode(...) returns a column vector err that gives information about error correction. If the code is a convolutional code, err contains the metric calculations used in the decoding decision process. For other types of codes, a nonnegative integer in the rth row of err indicates the number of errors corrected in the rth message word; a negative integer indicates that there are more errors in the rth word than can be corrected.
[msg,err,ccode] = decode(...) returns the corrected code in ccode.
[msg,err,ccode,cerr] = decode(...) returns a column vector cerr whose meaning depends on the format of code:
- If code is a binary vector, a nonnegative integer in the rth row of vec2matcerr indicates the number of errors corrected in the rth codeword; a negative integer indicates that there are more errors in the rth codeword than can be corrected.
- If code is not a binary vector, cerr = err.

\section*{Examples}

On the reference page for encode, some of the example code illustrates the use of the decode function.

The example below illustrates the use of err and cerr when the coding method is not convolutional code and the code is a binary vector. The script encodes two five-bit messages using a cyclic code. Each codeword has 15 bits. Errors are added to the first two bits of the first codeword and the first bit of the second codeword. Then decode is used to recover the original message. As a result, the errors are corrected. err reflects the fact that the first message was recovered after correcting two errors, while the second message was recovered after correcting one error. cerr reflects the fact that the first codeword was decoded after correcting two errors, while the second codeword was decoded after correcting one error.
```

m = 4; n = 2^m-1; % Codeword length is 15.
k = 5; % Message length
msg = ones(10,1); % Two messages, five bits each
code = encode(msg,n,k,'cyclic'); % Encode the message.
% Now place two errors in first word and one error
% in the second word. Create errors by reversing bits.
noisycode = code;
noisycode(1:2) = bitxor(noisycode(1:2),[1 1]');
noisycode(16) = bitxor(noisycode(16),1);
% Decode and try to correct the errors.
[newmsg,err,ccode,cerr] = decode(noisycode,n,k,'cyclic');
disp('Transpose of err is'); disp(err')
disp('Transpose of cerr is'); disp(cerr')

```

The output is below.
```

Single-error patterns loaded in decoding table.
1008 rows remaining.
2-error patterns loaded. }918\mathrm{ rows remaining.
3-error patterns loaded. }648\mathrm{ rows remaining.
4-error patterns loaded. }243\mathrm{ rows remaining.
5-error patterns loaded. 0 rows remaining.
Transpose of err is
2 1
Transpose of cerr is
2 1

```

\section*{Algorithms \\ Depending on the decoding method, decode relies on such lower-level functions as hammgen, syndtable, and cyclgen.}
```

See Also
encode | cyclpoly | syndtable | gen2par

```

How To . "Block Codes"

Purpose Restore ordering of symbols
```

Syntax deintrlvd = deintrlv(data,elements)

```

Description deintrlvd = deintrlv(data, elements) restores the original ordering of the elements of data by acting as an inverse of intrlv. If data is a length- N vector or an N -row matrix, elements is a length N vector that permutes the integers from 1 to N . To use this function as an inverse of the intrlv function, use the same elements input in both functions. In that case, the two functions are inverses in the sense that applying intrlv followed by deintrlv leaves data unchanged.

\section*{Examples}

The code below illustrates the inverse relationship between intrlv and deintrlv.
```

p = randperm(10); % Permutation vector
a = intrlv(10:10:100,p); % Rearrange [10 20 30 ... 100].
b = deintrlv(a,p) % Deinterleave a to restore ordering.

```

The output is
b =
\(\begin{array}{llllllllll}10 & 20 & 30 & 40 & 50 & 60 & 70 & 80 & 90 & 100\end{array}\)

\section*{See Also intrlv}

How To . "Interleaving"

\section*{Purpose \\ Syntax \\ Description}

Construct decision-feedback equalizer object
eqobj = dfe(nfwdweights,nfbkweights,alg)
eqobj = dfe(nfwdweights,nfbkweights,alg,sigconst)
eqobj = dfe(nfwdweights,nfbkweights,alg,sigconst,nsamp)
The dfe function creates an equalizer object that you can use with the
equalize function to equalize a signal. To learn more about the process for equalizing a signal, see "Adaptive Algorithms".
eqobj = dfe(nfwdweights, nfbkweights,alg) constructs a decision feedback equalizer object. The equalizer's feedforward and feedback filters have nfwdweights and nfbkweights symbol-spaced complex weights, respectively, which are initially all zeros. alg describes the adaptive algorithm that the equalizer uses; you should create alg using any of these functions: lms, signlms, normlms, varlms, rls, or cma. The signal constellation of the desired output is [ \(\left.-11 \begin{array}{l}1\end{array}\right]\), which corresponds to binary phase shift keying (BPSK).
eqobj = dfe(nfwdweights, nfbkweights,alg,sigconst) specifies the signal constellation vector of the desired output.
eqobj = dfe(nfwdweights,nfbkweights,alg,sigconst,nsamp) constructs a DFE with a fractionally spaced forward filter. The forward filter has nfwdweights complex weights spaced at T/nsamp, where \(T\) is the symbol period and nsamp is a positive integer. nsamp \(=1\) corresponds to a symbol-spaced forward filter.

\section*{Properties}

The table below describes the properties of the decision feedback equalizer object. To learn how to view or change the values of a decision feedback equalizer object, see "Accessing Properties of an Equalizer".

Note To initialize or reset the equalizer object eqobj, enter reset(eqobj).
\begin{tabular}{l|l}
\hline Property & Description \\
\hline EqType & \begin{tabular}{l} 
Fixed value, 'Decision \\
Feedback Equalizer'
\end{tabular} \\
\hline AlgType & \begin{tabular}{l} 
Name of the adaptive algorithm \\
represented by alg
\end{tabular} \\
\hline nWeights & \begin{tabular}{l} 
Number of weights in the forward \\
filter and the feedback filter, \\
in the format [nfwdweights, \\
nfbkweights ]. The number of \\
weights in the forward filter must \\
be at least 1.
\end{tabular} \\
\hline nSampPerSym & \begin{tabular}{l} 
Number of input samples per \\
symbol (equivalent to nsamp \\
input argument). This value \\
relates to both the equalizer \\
structure (see the use of K in \\
"Decision-Feedback Equalizers") \\
and an assumption about the \\
signal to be equalized.
\end{tabular} \\
\hline RefTap (except for CMA & \begin{tabular}{l} 
Reference tap index, between 1 \\
equalizers) \\
andwdweights. Setting this to \\
a value greater than 1 effectively \\
delays the reference signal with \\
respect to the equalizer's input \\
signal.
\end{tabular} \\
\hline SigConst & \begin{tabular}{l} 
Signal constellation, a vector \\
whose length is typically a power \\
of 2.
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline Weights & \begin{tabular}{l} 
Vector that concatenates the \\
complex coefficients from the \\
forward filter and the feedback \\
filter. This is the set of \(w_{i}\) \\
values in the schematic in \\
"Decision-Feedback Equalizers".
\end{tabular} \\
\hline WeightInputs & \begin{tabular}{l} 
Vector that concatenates the tap \\
weight inputs for the forward \\
filter and the feedback filter. \\
This is the set of u \\
i values in the \\
schematic in "Decision-Feedback \\
Equalizers".
\end{tabular} \\
\hline ResetBeforeFiltering & \begin{tabular}{l} 
If 1, each call to equalize \\
resets the state of eqobj before \\
equalizing. If 0, the equalization \\
process maintains continuity \\
from one call to the next.
\end{tabular} \\
\hline NumSamplesProcessed & \begin{tabular}{l} 
Number of samples the equalizer \\
processed since the last reset.
\end{tabular} \\
When you create or reset eqobj, \\
this property value is 0.
\end{tabular}\(|\)\begin{tabular}{ll} 
Properties specific to the adaptive & \begin{tabular}{l} 
See reference page for the \\
adaptive algorithm function \\
that created alg: lms, signlms, \\
normlms, varlms, rls, or cma.
\end{tabular} \\
\hline algorithm represented by alg
\end{tabular}

\section*{Relationships Among Properties}

If you change nWeights, MATLAB maintains consistency in the equalizer object by adjusting the values of the properties listed below.
\begin{tabular}{l|l}
\hline Property & Adjusted Value \\
\hline Weights & zeros(1, sum(nWeights)) \\
\hline WeightInputs & zeros(1, sum(nWeights)) \\
\hline \begin{tabular}{l} 
StepSize \\
(Variable-step-size LMS \\
equalizers)
\end{tabular} & InitStep*ones (1, sum(nWeights)) \\
\hline \begin{tabular}{l} 
InvCorrMatrix (RLS \\
equalizers)
\end{tabular} & InvCorrInit*eye (sum(nWeights)) \\
\hline
\end{tabular}

An example illustrating relationships among properties is in "Linked Properties of an Equalizer Object".

Apply a decision feedback equalizer (DFE) to an 8-PSK modulated signal

This example shows how to apply a decision feedback equalizer (DFE) to an 8-PSK modulated signal impaired by a frequency selector channel. The DFE uses 400 training symbols.

Set the modulation order to define 8-PSK modulation, and create a PSK modulator System object \({ }^{\mathrm{TM}}\).

M = 8;
hMod = comm.PSKModulator(M);
Create a 1500 -by- 1 column vector of random message symbols.
msg = randi([0 M-1],1500,1);
Modulate the random message signal by calling the step method of the comm. PSKModulator System object.
modmsg \(=\) step(hMod,msg);
Define a frequency selective channel with four taps, and then pass the modulated signal through the channel, introducing channel distortion.
```

chan = [.986; .845; .237; .123+.31i];

```
filtmsg = filter(chan,1,modmsg);

Create a DFE equalizer that has 10 feed forward tabs and five feedback tabs. The equalizer uses the LMS update method with a step size of 0.01.
```

numFFTaps = 10; numFBTaps = 5;
eq1 = dfe(numFFTaps, numFBTaps, lms(0.01));

```

For decision directed operation, the DFE must use the same signal constellation as the transmission scheme. Set the SigConst property to the constellation the modulator System object uses.
```

eq1.SigConst = step(hMod,(0:M-1)')';

```

Equalize the signal to help remove the effects of channel distortion. Use the first 400 symbols to train the equalizer.
```

trainlen = 400;
[symbolest,yd] = equalize(eq1,filtmsg,modmsg(1:trainlen));

```

Plot the received signal, equalizer output after training, and the ideal signal constellation.
```

h = scatterplot(filtmsg,1,trainlen,'bx'); hold on;
scatterplot(symbolest,1,trainlen,'g.',h);
scatterplot(eq1.SigConst,1,0,'k*',h);
legend('Filtered signal','Equalized signal',...
'Ideal signal constellation');
hold off;

```

Demodulate the signal at the equalizer output, and the unequalized signal at the input of the equalizer.
```

hDemod = comm.PSKDemodulator(8);
demodmsg_noeq = step(hDemod,filtmsg);
demodmsg = step(hDemod,yd);

```

Compute the error rates for the two demodulated signals and compare the results.
```

hErrorCalc = comm.ErrorRate;
ser_noEq = step(hErrorCalc, ...
msg(trainlen+1:end), demodmsg_noeq(trainlen+1:end));
reset(hErrorCalc)
ser_Eq = step(hErrorCalc, msg(trainlen+1:end),demodmsg(trainlen+1:end));
disp('Symbol error rates with and without equalizer:')
disp([ser_Eq(1) ser_noEq(1)])

```

The equalizer helps eliminate the distortion introduced by the frequency selective channel, and reduces the error rate.

See Also lms | signlms | normlms | varlms | rls | cma | lineareq | equalize
How To
- "Equalization"

\section*{Purpose}

Discrete Fourier transform matrix in Galois field

\section*{Syntax \\ dm = dftmtx(alph)}
\(\mathrm{dm}=\mathrm{dftmtx}(\mathrm{alph})\) returns a Galois array that represents the discrete Fourier transform operation on a Galois vector, with respect to the Galois scalar alph. The element alph is a primitive nth root of unity in the Galois field \(\operatorname{GF}\left(2^{\mathrm{m}}\right)=\mathrm{GF}(\mathrm{n}+1)\); that is, n must be the smallest positive value of \(k\) for which alph^k equals 1 . The discrete Fourier transform has size \(n\) and dm is an n-by-n array. The array dm represents the transform in the sense that dm times any length-n Galois column vector yields the transform of that vector.

Note The inverse discrete Fourier transform matrix is dftmtx(1/alph).

Examples The example below illustrates the discrete Fourier transform and its inverse, with respect to the element \(g f(3,4)\). The example examines the first \(n\) powers of that element to make sure that only the nth power equals one. Afterward, the example transforms a random Galois vector, undoes the transform, and checks the result.
```

m = 4;
n = 2^m-1;
a = 3;
alph = gf(a,m);
mp = minpol(alph);
if (mp(1)==1 \&\& isprimitive(mp)) % Check that alph has order n.
disp('alph is a primitive nth root of unity.')
dm = dftmtx(alph);
idm = dftmtx(1/alph);
x = gf(randi([0 2^m-1],n,1),m);
y = dm*x; % Transform x.
z = idm*y; % Recover x.
ck = isequal(x,z)

```
end

The output is
alph is a primitive nth root of unity.
ck =

1

\section*{Limitations}

See Also
How To

Algorithms The element dm(a,b) equals alph^((a-1)*(b-1)).
fft | ifft
The Galois field over which this function works must have 256 or fewer elements. In other words, alph must be a primitive nth root of unity in the Galois field \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\), where m is an integer between 1 and 8 .
- "Signal Processing Operations in Galois Fields"

\section*{Purpose}

Compute distance spectrum of convolutional code

\section*{Syntax}

Description
```

spect = distspec(trellis,n)

```
spect \(=\) distspec(trellis)
spect \(=\) distspec(trellis, \(n\) ) computes the free distance and the first \(n\) components of the weight and distance spectra of a linear convolutional code. Because convolutional codes do not have block boundaries, the weight spectrum and distance spectrum are semi-infinite and are most often approximated by the first few components. The input trellis is a valid MATLAB trellis structure, as described in "Trellis Description of a Convolutional Code". The output, spect, is a structure with these fields:
\begin{tabular}{l|l}
\hline Field & Meaning \\
\hline spect.dfree & \begin{tabular}{l} 
Free distance of the code. This is \\
the minimum number of errors in \\
the encoded sequence required to \\
create an error event.
\end{tabular} \\
\hline spect.weight & \begin{tabular}{l} 
A length-n vector that lists the \\
total number of information \\
bit errors in the error events \\
enumerated in spect.event.
\end{tabular} \\
\hline spect.event & \begin{tabular}{l} 
A length-n vector that lists the \\
number of error events for each \\
distance between spect. dfree \\
and spect.dfree \(+\mathrm{n}-1\). The \\
vector represents the first \(n\) \\
components of the distance \\
spectrum.
\end{tabular} \\
\hline
\end{tabular}

\footnotetext{
spect \(=\) distspec(trellis) is the same as spect \(=\) distspec(trellis,1).
}

\section*{distspec}

Examples The example below performs these tasks:
- Computes the distance spectrum for the rate \(2 / 3\) convolutional code that is depicted on the reference page for the poly2trellis function
- Uses the output of distspec as an input to the bercoding function, to find a theoretical upper bound on the bit error rate for a system that uses this code with coherent BPSK modulation
- Plots the upper bound using the berfit function
trellis = poly2trellis([5 4],[23 \(350 ; 05\) 13])
spect = distspec(trellis,4)
berub \(=\) bercoding(1:10,'conv','hard',2/3,spect); \% BER bound berfit(1:10,berub); ylabel('Upper Bound on BER'); \% Plot.

The output and plot are below.
```

trellis =
numInputSymbols: 4
numOutputSymbols: 8
numStates: 128
nextStates: [128x4 double]
outputs: [128x4 double]
spect =
dfree: 5
weight: [$$
\begin{array}{llll}{1}&{6}&{28}&{142]}\end{array}
$$]
event: [1 2 8 25]

```


\section*{Algorithms}

\section*{References}

The function uses a tree search algorithm implemented with a stack, as described in [2].
[1] Bocharova, I. E., and B. D. Kudryashov, "Rational Rate Punctured Convolutional Codes for Soft-Decision Viterbi Decoding," IEEE Transactions on Information Theory, Vol. 43, No. 4, July 1997, pp. 1305-1313.
[2] Cedervall, M., and R. Johannesson, "A Fast Algorithm for Computing Distance Spectrum of Convolutional Codes," IEEE Transactions on Information Theory, Vol. 35, No. 6, Nov. 1989, pp. 1146-1159.
[3] Chang, J., D. Hwang, and M. Lin, "Some Extended Results on the Search for Good Convolutional Codes," IEEE Transactions on Information Theory, Vol. 43, No. 5, Sep. 1997, pp. 1682-1697.

\section*{distspec}
[4] Frenger, P., P. Orten, and T. Ottosson, "Comments and Additions to Recent Papers on New Convolutional Codes," IEEE Transactions on Information Theory, Vol. 47, No. 3, March 2001, pp. 1199-1201.

See Also bercoding | iscatastrophic | istrellis | poly2trellis
\begin{tabular}{ll} 
Purpose & Package of Doppler classes \\
Description & \begin{tabular}{l} 
This package contains the classes that instantiate Doppler objects. \\
These objects are used as values of the DopplerSpectrum property, \\
which is common to both Rayleigh and Rician channel objects.
\end{tabular} \\
Properties & \begin{tabular}{l} 
Every Doppler object has a read-only SpectrumType property. Other \\
properties are specific to each Doppler class.
\end{tabular} \\
and \\
Methods & \begin{tabular}{l} 
Every Doppler object has a copy method, to duplicate itself, and a disp \\
method, to display its properties.
\end{tabular} \\
See Also & \begin{tabular}{l} 
doppler.ajakes | doppler.bell | doppler.bigaussian \\
| doppler.flat | doppler.gaussian | doppler.jakes | \\
doppler.rjakes | doppler.rounded | rayleighchan | ricianchan \\
| stdchan
\end{tabular} \\
How To & \begin{tabular}{l} 
- "Fading Channels"
\end{tabular}
\end{tabular}

\section*{doppler.ajakes}

\section*{Purpose Construct asymmetrical Doppler spectrum object}
```

Syntax dop = doppler.ajakes(freqminmaxajakes)
dop = doppler.ajakes

```

\section*{Description The doppler.ajakes function creates an asymmetrical Jakes} (AJakes) Doppler spectrum object. This object is to be used for the DopplerSpectrum property of a channel object created with the rayleighchan or the ricianchan functions.
dop = doppler.ajakes(freqminmaxajakes), where freqminmaxajakes is a row vector of two finite real numbers between -1 and 1 , creates a Jakes Doppler spectrum that is nonzero only for normalized (by the maximum Doppler shift \(f_{d}\), in Hz ) frequencies \(f_{\text {norm }}\) such that \(-1 \leq f_{\text {min,norm }} \leq f_{\text {norm }} \leq f_{\text {max, norm }} \leq 1\), where \(f_{\text {min,norm }}\) is given by freqminmaxajakes(1) and \(f_{\text {max,norm }}\) is given by freqminmaxajakes(2). The maximum Doppler shift \(f_{d}\) is specified by the MaxDopplerShift property of the channel object. Analytically: \(f_{\text {min,norm }}=f_{\text {min }} / f_{d}\) and \(f_{\max , \text { norm }}=f_{\max } / f_{d}\), where \(f_{\min }\) is the minimum Doppler shift (in hertz) and \(f_{\max }\) is the maximum Doppler shift (in hertz).
When dop is used as the DopplerSpectrum property of a channel object, space freqminmaxajakes(1) and freqminmaxajakes(2) by more than 1/50. Assigning a smaller spacing results in freqminmaxarjakes being reset to the default value of \(\left[\begin{array}{ll}0 & 1\end{array}\right]\).
dop = doppler.ajakes creates an asymmetrical Doppler spectrum object with a default freqminmaxajakes = [0 1]. This syntax is equivalent to constructing a Jakes Doppler spectrum that is nonzero only for positive frequencies.

\section*{doppler.ajakes}

\section*{Properties}

The AJakes Doppler spectrum object contains the following properties.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline SpectrumType & Fixed value, 'AJakes ' \\
\hline FreqMinMaxAJakes & \begin{tabular}{l} 
Vector of minimum and maximum \\
normalized Doppler shifts, two \\
real finite numbers between -1 \\
and 1
\end{tabular} \\
\hline
\end{tabular}

\section*{Theory and Applications}

The Jakes power spectrum is based on the assumption that the angles of arrival at the mobile receiver are uniformly distributed [1]: the
spectrum then covers the frequency range from \(-f_{d}\) to \(f_{d}, f_{d}\) being the maximum Doppler shift. When the angles of arrival are not uniformly distributed, then the Jakes power spectrum does not cover
the full Doppler bandwidth from \(-f_{d}\) to \(f_{d}\). The AJakes Doppler spectrum object covers the case of a power spectrum that is nonzero only for frequencies \(f\) such that \(-f_{d} \leq f_{\min } \leq f \leq f_{\max } \leq f_{d}\). It is an asymmetrical spectrum in the general case, but becomes a symmetrical
spectrum if \(f_{\text {min }}=-f_{\text {max }}\).
The normalized AJakes Doppler power spectrum is given analytically by:
\[
\begin{aligned}
S(f) & =\frac{A_{a}}{\pi f_{d} \sqrt{1-\left(f / f_{d}\right)^{2}}},-f_{d} \leq f_{\min } \leq f \leq f_{\max } \leq f_{d} \\
A_{a} & =\frac{1}{\frac{1}{\pi}\left[\sin ^{-1}\left(\frac{f_{\max }}{f_{d}}\right)-\sin ^{-1}\left(\frac{f_{\min }}{f_{d}}\right)\right]}
\end{aligned}
\]

\section*{doppler.ajakes}
where \(f_{\text {min }}\) and \(f_{\text {max }}\) denote the minimum and maximum frequencies where the spectrum is nonzero. You can determine these values from the probability density function of the angles of arrival.

\section*{Examples}

\section*{References}
[1] Jakes, W. C., Ed., Microwave Mobile Communications, Wiley, 1974.
[2] Lee, W. C. Y., Mobile Communications Engineering: Theory and Applications, 2nd Ed., McGraw-Hill, 1998.
[3] Pätzold, M., Mobile Fading Channels, Wiley, 2002.

\section*{See Also}

How To . "Fading Channels"

\section*{doppler.bell}

Purpose Construct bell-shaped Doppler spectrum object
Syntax doppler.bell
doppler.bell(coeffbell)

\section*{Description}

\section*{Properties}

The bell Doppler spectrum object has the following properties.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline SpectrumType & Fixed value, 'Bell' \\
\hline CoeffBell & \begin{tabular}{l} 
Bell spectrum coefficient, positive \\
real finite scalar.
\end{tabular} \\
\hline
\end{tabular}

\section*{Theory} and Applications

A bell spectrum was proposed in [1] for the Doppler spectrum of indoor MIMO channels, for 802.11n channel modeling.

The normalized bell Doppler spectrum is given analytically by:
\[
S(f)=\frac{C_{b}}{1+A\left(\frac{f}{f_{d}}\right)^{2}}
\]
where

\section*{doppler.bell}
\[
|f| \leq f_{d}
\]
and
\[
C_{b}=\frac{\sqrt{A}}{\pi f_{d}}
\]
\(f_{d}\) represents the maximum Doppler shift specified for the channel object, and \(A\) represents a positive real finite scalar (CoeffBell). The indoor MIMO channel model of IEEE 802.11n [1] uses the following parameter: \(A=9\). Since the channel is modeled as Rician fading with a fixed line-of-sight (LOS) component, a Dirac delta is also present in the Doppler spectrum at \(f=0\).

\section*{Examples}

References

See Also
doppler | doppler.ajakes | doppler.flat | doppler.gaussian | doppler.jakes | doppler.rjakes | doppler.rounded | rayleighchan | ricianchan | stdchan

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\section*{doppler.bigaussian}
\begin{tabular}{|c|c|}
\hline Purpose & Construct bi-Gaussian Doppler spectrum object \\
\hline Syntax & ```
dop = doppler.bigaussian(property1,value1,...)
dop = doppler.bigaussian
``` \\
\hline Description & \begin{tabular}{l}
The doppler.bigaussian function creates a bi-Gaussian Doppler spectrum object to be used for the DopplerSpectrum property of a channel object (created with either the rayleighchan function or the ricianchan function). \\
dop = doppler.bigaussian(property1,value1,...) creates a bi-Gaussian Doppler spectrum object with properties as specified by the property/value pairs. If you do not specify a value for a property, the property is assigned a default value. \\
dop \(=\) doppler.bigaussian creates a bi-Gaussian Doppler spectrum object with default properties. The constructed Doppler spectrum object is equivalent to a single Gaussian Doppler spectrum centered at zero frequency. The equivalent command with property/value pairs is:
```

dop = doppler.bigaussian('SigmaGaussian1', 1/sqrt(2), ...
'SigmaGaussian2', 1/sqrt(2), ...
'CenterFreqGaussian1', 0, ...
'CenterFreqGaussian2', 0, ...
'GainGaussian1', 0.5, ...
'GainGaussian2', 0.5)

```
\end{tabular} \\
\hline
\end{tabular}

\section*{Properties}

The bi-Gaussian Doppler spectrum object contains the following properties.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline SpectrumType & Fixed value, 'BiGaussian ' \\
\hline SigmaGaussian1 & \begin{tabular}{l} 
Normalized standard deviation \\
of first Gaussian function (real \\
positive finite scalar value)
\end{tabular} \\
\hline
\end{tabular}

\section*{doppler.bigaussian}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline SigmaGaussian2 & \begin{tabular}{l} 
Normalized standard deviation \\
of second Gaussian function (real \\
positive finite scalar value)
\end{tabular} \\
\hline CenterFreqGaussian1 & \begin{tabular}{l} 
Normalized center frequency \\
of first Gaussian function (real \\
scalar value between -1 and 1)
\end{tabular} \\
\hline CenterFreqGaussian2 & \begin{tabular}{l} 
Normalized center frequency of \\
second Gaussian function (real \\
scalar value between -1 and 1)
\end{tabular} \\
\hline GainGaussian1 & \begin{tabular}{l} 
Power gain of first Gaussian \\
function (linear scale, real \\
nonnegative finite scalar value)
\end{tabular} \\
\hline GainGaussian2 & \begin{tabular}{l} 
Power gain of second Gaussian \\
function (linear scale, real \\
nonnegative finite scalar value)
\end{tabular} \\
\hline
\end{tabular}

All properties are writable except for the SpectrumType property.
The properties SigmaGaussian1, SigmaGaussian2, GainGaussian1, and GainGaussian2 are normalized by the MaxDopplerShift property of the associated channel object.
Analytically, the normalized standard deviations of the first and second Gaussian functions are determined as \(\sigma_{G 1, \text { norm }}=\sigma_{G 1} / f_{d}\) and \(\sigma_{G 2, \text { norm }}=\sigma_{G 2} / f_{d}\), respectively, where \(\sigma_{G 1}\) and \(\sigma_{G 2}\) are the standard deviations of the first and second Gaussian functions, and
\(f_{d}\) is the maximum Doppler shift, in hertz. Similarly, the normalized center frequencies of the first and second Gaussian functions are determined as \(f_{G 1, \text { norm }}=f_{G 1} / f_{d}\) and \(f_{G 2, \text { norm }}=f_{G 2} / f_{d}\), respectively, where \(f_{G 1}\) and \(f_{G 2}\) are the center frequencies of the first and second Gaussian functions. The properties GainGaussian1 and GainGaussian2

\section*{doppler.bigaussian}
correspond to the power gains \(C_{G 1}\) and \(C_{G 2}\), respectively, of the two Gaussian functions.

\section*{Theory and \\ Applications}

The bi-Gaussian power spectrum consists of two frequency-shifted Gaussian spectra. The COST207 channel models ([1], [2], [3]) specify two distinct bi-Gaussian Doppler spectra, GAUS1 and GAUS2, to be used in modeling long echos for urban and hilly terrain profiles.
The normalized bi-Gaussian Doppler spectrum is given analytically by:
\[
S_{G}(f)=A_{G}\left[\frac{C_{G 1}}{\sqrt{2 \pi \sigma_{G 1}^{2}}} \exp \left(-\frac{\left(f-f_{G 1}\right)^{2}}{2 \sigma_{G 1}^{2}}\right)+\frac{C_{G, 2}}{\sqrt{2 \pi \sigma_{G 2}^{2}}} \exp \left(-\frac{\left(f-f_{G 2}\right)^{2}}{2 \sigma_{G 2}^{2}}\right)\right]
\]
where \(\sigma_{G 1}\) and \(\sigma_{G 2}\) are standard deviations, \(f_{G 1}\) and \(f_{G 2}\) are center
frequencies, \(C_{G 1}\) and \(C_{G 2}\) are power gains, and \(A_{G}=\frac{1}{C_{G 1}+C_{G 2}}\) is a
normalization coefficient.
If either \(f_{G 1}=0\) or \(f_{G 2}=0\), a frequency-shifted Gaussian Doppler spectrum is obtained.

\section*{Examples}

The following MATLAB code first creates a bi-Gaussian Doppler spectrum object with the same parameters as that of a COST 207 GAUS2 Doppler spectrum. It then creates a Rayleigh channel object
with a maximum Doppler shift of \(f_{d}=30\) and assigns the constructed Doppler spectrum object to its DopplerSpectrum property.
```

dop_bigaussian = doppler.bigaussian('SigmaGaussian1', 0.1, ...
'SigmaGaussian2', 0.15, 'CenterFreqGaussian1', 0.7, ...
'CenterFreqGaussian2', -0.4, 'GainGaussian1', 1, ...
'GainGaussian2', 1/10^1.5)
chan = rayleighchan(1e-3, 30);
chan.DopplerSpectrum = dop_bigaussian;

```
ReferencesSee Alsodoppler | doppler.ajakes | doppler.bell | doppler.flat| doppler.gaussian | doppler.jakes | doppler.rjakes |doppler.rounded | rayleighchan | ricianchan | stdchan
How To - "Fading Channels"

\section*{doppler.flat}

Purpose Construct flat Doppler spectrum object
Syntax dop = doppler.flat
Description dop = doppler.flat creates a flat Doppler spectrum object that is to be used for the DopplerSpectrum property of a channel object (created with either the rayleighchan or the ricianchan function). The maximum Doppler shift of the flat Doppler spectrum object is specified by the MaxDopplerShift property of the channel object.

Properties The flat Doppler spectrum object contains only one property, SpectrumType, which is read-only and has a fixed value of 'Flat'.

Theory In a 3-D isotropic scattering environment, where the angles of arrival and
Applications are uniformly distributed in the azimuth and elevation planes, the Doppler spectrum is found theoretically to be flat [2]. A flat Doppler spectrum is also specified in some cases of the ANSI J-STD-008 reference channel models for PCS, for both outdoor (pedestrian) and indoor (commercial) [1] applications.
The normalized flat Doppler power spectrum is given analytically by:
\[
S(f)=\frac{1}{2 f_{d}},|f| \leq f_{d}
\]
where \(f_{d}\) is the maximum Doppler frequency.

\section*{References}
[1] ANSI J-STD-008, Personal Station-Base Station Compatibility Requirements for 1.8 to 2.0 GHz Code Division Multiple Access (CDMA) Personal Communications Systems, March 1995.
[2] Clarke, R. H., and Khoo, W. L., "3-D Mobile Radio Channel Statistics", IEEE Trans. Veh. Technol., Vol. 46, No. 3, pp. 798-799, August 1997.
```

See Also
doppler | doppler.ajakes | doppler.bell | doppler.bigaussian
| doppler.gaussian | doppler.jakes | doppler.rjakes |
doppler.rounded | rayleighchan | ricianchan | stdchan
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```

\section*{doppler.gaussian}
```

Purpose Construct Gaussian Doppler spectrum object
Syntax dop = doppler.gaussian
dop = doppler.gaussian(sigmagaussian)

```

\section*{Properties}

The Gaussian Doppler spectrum object contains the following properties.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline SpectrumType & Fixed value, 'Gaussian' \\
\hline SigmaGaussian & \begin{tabular}{l} 
Normalized standard deviation of \\
the Gaussian Doppler spectrum \\
(a real positive number)
\end{tabular} \\
\hline
\end{tabular}

Theory
and
Applications
The Gaussian power spectrum is considered to be a good model for multipath components with long delays in UHF communications [3]. It is also proposed as a model for the aeronautical channel [2]. A Gaussian Doppler spectrum is also specified in some cases of the ANSI J-STD-008 reference channel models for PCS applications, for both outdoor
(wireless loop) and indoor (residential, office) [1]. The normalized Gaussian Doppler power spectrum is given analytically by:
\[
S_{G}(f)=\frac{1}{\sqrt{2 \pi \sigma_{G}^{2}}} \exp \left(-\frac{f^{2}}{2 \sigma_{G}^{2}}\right)
\]

An alternate representation is [4]:
\[
S_{G}(f)=\frac{1}{f_{c}} \sqrt{\frac{\ln 2}{\pi}} \exp \left(-(\ln 2)\left(\frac{f}{f_{c}}\right)^{2}\right)
\]
where \(f_{c}=\sigma_{G} \sqrt{2 \ln 2}\) is the 3 dB cutoff frequency. If you set \(f_{c}=f_{d} \sqrt{\ln 2}\), where \(f_{d}\) is the maximum Doppler shift, or equivalently \(\sigma_{G}=f_{d} / \sqrt{2}\), the Doppler spread of the Gaussian power spectrum becomes equal to the Doppler spread of the Jakes power spectrum, where Doppler spread is defined as:
\[
\sigma_{D}=\sqrt{\int_{-\infty}^{\infty} f^{2} S(f) d f}
\]

The following code creates a Rayleigh channel object with a maximum Doppler shift of \(f_{d}=10\). It then creates a Gaussian Doppler spectrum object with a normalized standard deviation of \(\sigma_{G, \text { norm }}=0.5\), and assigns it to the DopplerSpectrum property of the channel object.
```

chan = rayleighchan(1/1000,10);
dop_gaussian = doppler.gaussian(0.5);
chan.DopplerSpectrum = dop_gaussian;

```

\section*{doppler.gaussian}
ReferencesSee Also doppler | doppler.ajakes | doppler.bell | doppler.bigaussian |doppler.flat | doppler.jakes | doppler.rjakes | doppler.rounded| rayleighchan | ricianchan | stdchan
How To - "Fading Channels"

\section*{doppler.jakes}

\section*{Purpose Construct Jakes Doppler spectrum object}

\section*{Syntax}

Description

Properties
Theory and Applications
dop \(=\) doppler.jakes creates a Jakes Doppler spectrum object that is to be used for the DopplerSpectrum property of a channel object (created with either the rayleighchan or the ricianchan function). The maximum Doppler shift of the Jakes Doppler spectrum object is specified by the MaxDopplerShift property of the channel object. By default, channel objects are created with a Jakes Doppler spectrum.

The Jakes Doppler spectrum object contains only one property, SpectrumType, which is read-only and has a fixed value of 'Jakes'.

The Jakes Doppler power spectrum model is actually due to Gans [2], who analyzed the Clarke-Gilbert model ([1], [3], and [5]). The Clarke-Gilbert model is also called the classical model.

The Jakes Doppler power spectrum applies to a mobile receiver. It derives from the following assumptions [6]:
- The radio waves propagate horizontally.
- At the mobile receiver, the angles of arrival of the radio waves are uniformly distributed over \([-\pi, \pi]\).
- At the mobile receiver, the antenna is omnidirectional (i.e., the antenna pattern is circular-symmetrical).
The normalized Jakes Doppler power spectrum is given analytically by:
\[
S(f)=\frac{1}{\pi f_{d} \sqrt{1-\left(f / f_{d}\right)^{2}}},|f| \leq f_{d}
\]
where \(f_{d}\) is the maximum Doppler frequency.

\section*{doppler.jakes}
\begin{tabular}{|c|c|}
\hline Examples & \begin{tabular}{l}
Create a Rayleigh channel object with a maximum Doppler shift of fd=10 Hertz. Then, create a Jakes Doppler spectrum object and assigns it to the DopplerSpectrum property of the channel object. \\
chan = rayleighchan(1/1000,10); \\
dop_gaussian = doppler.jakes; \\
chan.DopplerSpectrum = dop_gaussian
\end{tabular} \\
\hline \multirow[t]{6}{*}{References} & [1] Clarke, R. H., "A Statistical Theory of Mobile-Radio Reception," Bell System Technical Journal, Vol. 47, No. 6, pp. 957-1000, July-August 1968. \\
\hline & [2] Gans, M. J., "A Power-Spectral Theory of Propagation in the Mobile-Radio Environment," IEEE Trans. Veh. Technol., Vol. VT-21, No. 1, pp. 27-38, Feb. 1972. \\
\hline & [3] Gilbert, E. N., "Energy Reception for Mobile Radio," Bell System Technical Journal, Vol. 44, No. 8, pp. 1779-1803, Oct. 1965. \\
\hline & [4] Jakes, W. C., Ed. Microwave Mobile Communications, Wiley, 1974. \\
\hline & [5] Lee, W. C. Y., Mobile Communications Engineering: Theory and Applications, 2nd Ed., McGraw-Hill, 1998. \\
\hline & [6] Pätzold, M., Mobile Fading Channels, Wiley, 2002. \\
\hline See Also & ```
doppler | doppler.ajakes | doppler.bell | doppler.bigaussian
    | doppler.flat | doppler.gaussian | doppler.rjakes |
doppler.rounded | rayleighchan | ricianchan | stdchan
``` \\
\hline How To & - "Fading Channels" \\
\hline
\end{tabular}

\section*{Purpose}

Construct restricted Jakes Doppler spectrum object

\section*{Syntax}

Description
dop = doppler.rjakes
dop \(=\) doppler.rjakes(freqminmaxrjakes)
The doppler.rjakes function creates a restricted Jakes (RJakes)

Doppler spectrum object that is used for the DopplerSpectrum property of a channel object (created with either the rayleighchan or the ricianchan function).
dop \(=\) doppler.rjakes creates a Doppler spectrum object equivalent to the Jakes Doppler spectrum. The maximum Doppler shift of the RJakes Doppler spectrum object is specified by the MaxDopplerShift property of the channel object.
dop = doppler.rjakes(freqminmaxrjakes), where freqminmaxrjakes is a row vector of two finite real numbers between 0 and 1, creates a Jakes Doppler spectrum. This spectrum is nonzero only for normalized frequencies (by the maximum Doppler shift,
\(f_{d}\), in Hertz), \(f_{\text {norm }}\), such that \(0 \leq f_{\min , \text { norm }} \leq\left|f_{\text {norm }}\right| \leq f_{\max , \text { norm }} \leq 1\), where \(f_{\text {min,norm }}\) is given by freqminmaxrjakes(1) and \(f_{\text {max, norm }}\) is given by freqminmaxrjakes(2). The maximum Doppler shift \(f_{d}\) is specified by the MaxDopplerShift property of the channel object.
Analytically, \(f_{\min , n o r m}=f_{\min } / f_{d}\) and \(f_{\max , n o r m}=f_{\max } / f_{d}\), where \(f_{\min }\) is the minimum Doppler shift (in Hertz) and \(f_{\text {max }}\) is the maximum Doppler shift (in Hertz).
When dop is used as the DopplerSpectrum property of a channel object, freqminmaxrjakes(1) and freqminmaxrjakes(2) should be spaced by more than \(1 / 50\). Assigning a smaller spacing results in freqminmaxrjakes being reset to the default value of [ 011\(]\).

\section*{doppler.riakes}

Properties
The RJakes Doppler spectrum object contains the following properties.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline SpectrumType & Fixed value, 'RJakes ' \\
\hline FreqMinMaxRJakes & \begin{tabular}{l} 
Vector of minimum and maximum \\
normalized Doppler shifts (two \\
real finite numbers between 0 \\
and 1)
\end{tabular} \\
\hline
\end{tabular}

Theory and
Applications

The Jakes power spectrum is based on the assumption that the angles of arrival at the mobile receiver are uniformly distributed [1], where the spectrum covers the frequency range from \(-f_{d}\) to \(f_{d}, f_{d}\) being the maximum Doppler shift. When the angles of arrival are not uniformly distributed, the Jakes power spectrum does not cover the full Doppler
bandwidth from \(-f_{d}\) to \(f_{d}\). This exception also applies to the case where the antenna pattern is directional. This type of spectrum is known as restricted Jakes [3]. The RJakes Doppler spectrum object covers only the case of a symmetrical power spectrum, which is nonzero
only for frequencies \(f\) such that \(0 \leq f_{\text {min }} \leq|f| \leq f_{\text {max }} \leq f_{d}\).
The normalized RJakes Doppler power spectrum is given analytically by:
\[
S(f)=\frac{A_{r}}{\pi f_{d} \sqrt{1-\left(f / f_{d}\right)^{2}}}, 0 \leq f_{\min } \leq|f| \leq f_{\max } \leq f_{d}
\]
where
\[
A_{r}=\frac{1}{\frac{2}{\pi}\left[\sin ^{-1}\left(\frac{f_{\max }}{f_{d}}\right)-\sin ^{-1}\left(\frac{f_{\min }}{f_{d}}\right)\right]}
\]
\(f_{\min }\) and \(f_{\max }\) denote the minimum and maximum frequencies where the spectrum is nonzero. They can be determined from the probability density function of the angles of arrival.

\section*{Examples}

References

See Also

The following code first creates a Rayleigh channel object with a maximum Doppler shift of \(f_{d}=10\). It then creates an RJakes Doppler object with minimum normalized Doppler shift \(f_{\min , n o r m}=0.14\) and maximum normalized Doppler shift \(f_{\text {max, norm }}=0.9\).

The Doppler object is assigned to the DopplerSpectrum property of the channel object. The channel then has a Doppler spectrum that is nonzero for frequencies \(f\) such that \(0 \leq f_{\text {min }} \leq|f| \leq f_{\text {max }} \leq f_{d}\), where \(f_{\text {min }}=f_{\min , \text { norm }} \times f_{d}=1.4 \mathrm{~Hz}\) and \(f_{\max }=f_{\max , n o r m} \times f_{d}=9 \mathrm{~Hz}\). chan = rayleighchan(1/1000, 10); dop_rjakes = doppler.rjakes([0.14 0.9]); chan.DopplerSpectrum = dop_rjakes; chan.DopplerSpectrum

The output is:
SpectrumType: 'RJakes'
FreqMinMaxRJakes: [0.1400 0.9000]
[1] Jakes, W. C., Ed. Microwave Mobile Communications, Wiley, 1974.
[2] Lee, W. C. Y., Mobile Communications Engineering: Theory and Applications, 2nd Ed., McGraw-Hill, 1998.
[3] Pätzold, M., Mobile Fading Channels, Wiley, 2002.
doppler | doppler.ajakes | doppler.bell | doppler.bigaussian doppler.flat | doppler.gaussian | doppler.jakes doppler. rounded | rayleighchan | ricianchan | stdchan

\section*{doppler.riakes}

\author{
How To . "Fading Channels"
}

\section*{doppler.rounded}

Purpose
Syntax

Description

Construct rounded Doppler spectrum object
```

dop = doppler.rounded
dop = doppler.rounded(coeffrounded)

```

The doppler.rounded function creates a rounded Doppler spectrum object that is used for the DopplerSpectrum property of a channel object (created with either the rayleighchan or the ricianchan function).
dop = doppler.rounded creates a rounded Doppler spectrum object
with default polynomial coefficients \(a_{0}=1, a_{2}=-1.72, a_{4}=0.785\) (see "Theory and Applications" on page 1-221 for the meaning of these coefficients). The maximum Doppler shift \(f_{d}\) (in Hertz) is specified by the MaxDopplerShift property of the channel object.
dop = doppler. rounded(coeffrounded), where coeffrounded is a row vector of three finite real numbers, creates a rounded Doppler spectrum object with polynomial coefficients, \(a_{0}, a_{2}, a_{4}\), given by coeffrounded(1), coeffrounded (2), and coeffrounded(3), respectively.

The rounded Doppler spectrum object contains the following properties.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline SpectrumType & Fixed value, 'Rounded ' \\
\hline CoeffRounded & \begin{tabular}{l} 
Vector of three polynomial \\
coefficients (real finite numbers)
\end{tabular} \\
\hline
\end{tabular}

A rounded spectrum is proposed as an approximation to the measured Doppler spectrum of the scatter component of fixed wireless channels at 2.5 GHz [1]. However, the shape of the spectrum is influenced by the center carrier frequency.

\section*{doppler.rounded}

The normalized rounded Doppler spectrum is given analytically by a polynomial in \(f\) of order four, where only the even powers of \(f\) are retained:
\[
S(f)=C_{r}\left[a_{0}+a_{2}\left(\frac{f}{f_{d}}\right)^{2}+a_{4}\left(\frac{f}{f_{d}}\right)^{4}\right],|f| \leq f_{d}
\]
where
\[
C_{r}=\frac{1}{2 f_{d}\left[a_{0}+\frac{a_{2}}{3}+\frac{a_{4}}{5}\right]}
\]
\(f_{d}\) is the maximum Doppler shift, and \(a_{0}, a_{2}, a_{4}\) are real finite coefficients. The fixed wireless channel model of IEEE 802.16 [1] uses the following parameters: \(a_{0}=1, a_{2}=-1.72\), and \(a_{4}=0.785\). Because the channel is modeled as Rician fading with a fixed line-of-sight (LOS) component, a Dirac delta is also present in the Doppler spectrum at \(f=0\).

\section*{Examples}

References

The following code creates a Rician channel object with a maximum Doppler shift of \(f_{d}=10\). It then creates a rounded Doppler spectrum object with polynomial coefficients \(a_{0}=1.0, a_{2}=-0.5, a_{4}=1.5\), and assigns it to the DopplerSpectrum property of the channel object.
```

chan = ricianchan(1/1000,10,1);
dop_rounded = doppler.rounded([1.0 -0.5 1.5]);
chan.DopplerSpectrum = dop_rounded;

```
[1] IEEE 802.16 Broadband Wireless Access Working Group, "Channel models for fixed wireless applications," IEEE 802.16a-03/01, 2003-06-27.

\section*{doppler.rounded}

See Also \(\begin{aligned} & \text { doppler | doppler.ajakes | doppler.bell | doppler.bigaussian } \\ & \text { | doppler.flat | doppler.gaussian | doppler.jakes | } \\ & \text { doppler.rjakes | rayleighchan | ricianchan | stdchan }\end{aligned}\)
How To \(\quad\). "Fading Channels"
Purpose \begin{tabular}{ll} 
Syntax & \begin{tabular}{l} 
Decode using differential pulse code modulation \\
sig \(=\) dpcmdeco(indx, codebook, predictor) \\
[sig, quanterror] \(=\) dpcmdeco(indx, codebook, predictor)
\end{tabular} \\
& \begin{tabular}{l} 
sig \(=\) dpcmdeco(indx, codebook, predictor) implements differential \\
pulse code demodulation to decode the vector indx. The vector codebook \\
represents the predictive-error quantization codebook. The vector \\
predictor specifies the predictive transfer function. If the transfer \\
function has predictive order M, predictor has length M+1 and an \\
initial entry of 0. To decode correctly, use the same codebook and \\
predictor in dpcmenco and dpcmdeco.
\end{tabular} \\
\begin{tabular}{l} 
See "Represent Partitions", "Represent Codebooks", or the quantiz \\
reference page, for a description of the formats of partition and \\
codebook.
\end{tabular} \\
\begin{tabular}{l} 
[sig, quanterror] = dpcmdeco(indx, codebook, predictor) is the \\
same as the syntax above, except that the vector quanterror is \\
the quantization of the predictive error based on the quantization \\
parameters. quanterror is the same size as sig.
\end{tabular}
\end{tabular}

> Note You can estimate the input parameters codebook, partition, and predictor using the function dpcmopt.

\author{
Examples See "Example: DPCM Encoding and Decoding" and "Example: Comparing Optimized and Nonoptimized DPCM Parameters" for examples that use dpcmdeco. \\ [1] Kondoz, A. M., Digital Speech, Chichester, England, John Wiley \& Sons, 1994. \\ \section*{See Also \\ \\ quantiz | dpcmopt | dpcmenco} \\ How To . "Differential Pulse Code Modulation"
}
\begin{tabular}{ll} 
Purpose & \begin{tabular}{l} 
Encode using differential pulse code modulation
\end{tabular} \\
Syntax & \begin{tabular}{l} 
indx = dpcmenco(sig, codebook, partition, predictor) \\
[indx, quants] = dpcmenco(sig, codebook, partition, predictor)
\end{tabular} \\
Description & \begin{tabular}{l} 
indx = dpcmenco(sig, codebook, partition, predictor) implements \\
differential pulse code modulation to encode the vector sig. partition \\
is a vector whose entries give the endpoints of the partition intervals. \\
codebook, a vector whose length exceeds the length of partition \\
by one, prescribes a value for each partition in the quantization. \\
predictor specifies the predictive transfer function. If the transfer \\
function has predictive order M, predictor has length M+1 and an \\
initial entry of 0. The output vector indx is the quantization index.
\end{tabular} \\
\begin{tabular}{l} 
See "Differential Pulse Code Modulation" for more about the format \\
of predictor. See "Represent Partitions", "Represent Partitions", or \\
the reference page for quantiz in this chapter, for a description of the \\
formats of partition and codebook.
\end{tabular} \\
\begin{tabular}{l} 
[indx, quants ] \(=\) dpcmenco(sig, codebook, partition, predictor) is the \\
same as the syntax above, except that quants contains the quantization \\
of sig based on the quantization parameters. quants is a vector of \\
the same size as sig.
\end{tabular} \\
\hline \begin{tabular}{l} 
Note If predictor is an order-one transfer function, the modulation is
\end{tabular} \\
called a delta modulation.
\end{tabular}

\author{
How To \\ - "Differential Pulse Code Modulation"
}
```

Purpose
Optimize differential pulse code modulation parameters
Syntax predictor = dpcmopt(training_set,ord)
[predictor,codebook,partition] = dpcmopt(training_set,ord,
len)
[predictor,codebook,partition] = dpcmopt(training_set,ord,
ini_cb)
predictor = dpcmopt(training_set,ord) returns a vector representing a predictive transfer function of order ord that is appropriate for the training data in the vector training_set. predictor is a row vector of length ord +1 . See "Represent Predictors" for more about its format.

```

Note dpcmopt optimizes for the data in training_set. For best results, training_set should be similar to the data that you plan to quantize.
[predictor, codebook, partition] = dpcmopt(training_set,ord,len) is the same as the syntax above, except that it also returns corresponding optimized codebook and partition vectors codebook and partition. len is an integer that prescribes the length of codebook. partition is a vector of length len-1. See "Represent Partitions", "Represent Codebooks", or the reference page for quantiz in this chapter, for a description of the formats of partition and codebook.
[predictor, codebook,partition] =
dpcmopt(training_set, ord, ini_cb) is the same as the first syntax, except that it also returns corresponding optimized codebook and partition vectors codebook and partition. ini_cb, a vector of length at least 2 , is the initial guess of the codebook values. The output codebook is a vector of the same length as ini_cb. The output partition is a vector whose length is one less than the length of codebook.

\section*{dpemopt}

\author{
Examples See "Example: Comparing Optimized and Nonoptimized DPCM Parameters" for an example that uses dpcmopt. \\ See Also dpcmenco | dpcmdeco | quantiz | lloyds \\ How To . "Differential Pulse Code Modulation"
}
```

Purpose
Differential phase shift keying demodulation
Syntax $\quad z=\operatorname{dpskdemod}(y, M)$
z = dpskdemod(y,M,phaserot)
z = dpskdemod(y,M,phaserot,symbol_order)

```

\section*{Description}

Examples The example below illustrates the fact that the first output symbol of a differential PSK demodulator is an initial condition rather than useful information.

The output is below.
s1 =

\section*{dpskdemod}
```

    1
    s2 =
0

```

For another example that uses this function, see "Example: Curve Fitting for an Error Rate Plot".

\author{
See Also dpskmod \| pskdemod \| pskmod \\ How To • "Digital Modulation"
}

\section*{Purpose}

Differential phase shift keying modulation

\section*{Syntax}
\(\mathrm{y}=\mathrm{dpskmod}(\mathrm{x}, \mathrm{M})\)
y = dpskmod(x,M,phaserot)
y = dpskmod(x,M,phaserot,symbol_order)
\(\mathrm{y}=\operatorname{dpskmod}(\mathrm{x}, \mathrm{M})\) outputs the complex envelope y of the modulation of the message signal \(x\) using differential phase shift keying modulation. \(M\) is the alphabet size and must be an integer. The message signal must consist of integers between 0 and \(M-1\). If \(x\) is a matrix with multiple rows and columns, the function processes the columns independently.
\(y=d p s k m o d(x, M, p h a s e r o t)\) specifies the phase rotation of the modulation in radians. In this case, the total phase shift per symbol is the sum of phaserot and the phase generated by the differential modulation.
\(\mathrm{y}=\mathrm{dpskmod}(\mathrm{x}, \mathrm{M}\), phaserot, symbol_order) specifies how the function assigns binary words to corresponding integers. If symbol_order is set to 'bin' (default), the function uses a natural binary-coded ordering. If symbol_order is set to 'gray', it uses a Gray-coded ordering.

\section*{Examples}

The example below plots the output of the dpskmod function. The image shows the possible transitions from each symbol in the DPSK signal constellation to the next symbol.
```

s = RandStream.create('mt19937ar', 'seed',131);
prevStream = RandStream.setGlobalStream(s); % seed for repeatability
M = 4; % Use DQPSK in this example, so M is 4.
x = randi([0 M-1],500,1); % Random data
y = dpskmod(x,M,pi/8); % Modulate using a nonzero initial phase.
plot(y) % Plot all points, using lines to connect them.

```

\section*{dpskmod}


For another example that uses this function, see "Example: Curve Fitting for an Error Rate Plot".

\section*{See Also}
dpskdemod | pskmod | pskdemod

\author{
How To \\ - "Digital Modulation"
}
Purpose Low-density parity-check codes from DVB-S. 2 standard
Syntax H = dvbs2ldpc(r)
Description \(H=\operatorname{dvbs} 2 l d p c(r)\) returns the parity-check matrix of the LDPC codewith code rate \(r\) from the DVB-S. 2 standard. \(H\) is a sparse logical matrix.
Possible values for \(r\) are \(1 / 4,1 / 3,2 / 5,1 / 2,3 / 5,2 / 3,3 / 4,4 / 5,5 / 6,8 / 9\), and \(9 / 10\). The block length of the code is 64800 .
The default parity-check matrix (32400-by-64800) corresponds to an irregular LDPC code with the structure shown in the following table.
\begin{tabular}{l|l}
\hline Row & Number of 1s Per Row \\
\hline 1 & 6 \\
\hline 2 to 32400 & 7 \\
\hline
\end{tabular}
\begin{tabular}{|l|l}
\hline Column & Number of 1s Per Column \\
\hline 1 to 12960 & 8 \\
\hline 12961 to 32400 & 3 \\
\hline
\end{tabular}
Columns 32401 to 64800 form a lower triangular matrix. Only the elements on its main diagonal and the subdiagonal immediately below are 1 s . This LDPC code is used in conjunction with a BCH code in the Digital Video Broadcasting standard DVB-S. 2 to achieve a packet error rate below \(10^{-7}\) at about 0.7 dB to 1 dB from the Shannon limit.
```Examples
H = dvbs2ldpc(3/5);
spy(H); % Visualize the location of nonzero elements in H.
henc = comm.LDPCEncoder(H);
hdec = comm.LDPCDecoder(H);
```

How To • spy

Purpose
Block encoder

## Syntax

```
code = encode(msg,n,k,'linear/fmt',genmat)
code = encode(msg,n,k,'cyclic/fmt',genpoly)
code = encode(msg,n,k,'hamming/fmt',prim_poly)
code = encode(msg,n,k)
[code,added] = encode(...)
```


## Optional

Inputs

| Input | Default Value |
| :--- | :--- |
| fmt | binary |
| genpoly | cyclpoly $(n, k)$ |
| prim_poly | gfprimdf $(n-k)$ |

## Description

## For All Syntaxes

The encode function encodes messages using one of the following error-correction coding methods:

- Linear block
- Cyclic
- Hamming

For all of these methods, the codeword length is n and the message length is $k$.
msg, which represents the messages, can have one of several formats. The table below shows which formats are allowed for msg, how the argument fmt should reflect the format of msg, and how the format of the output code depends on these choices. The examples in the table are for $\mathrm{k}=4$. If fmt is not specified as input, its default value is binary.

Note If $2^{\wedge} n$ or $2^{\wedge} k$ is large, use the default binary format instead of the decimal format. This is because the function uses a binary format internally, while the roundoff error associated with converting many bits to large decimal numbers and back might be substantial.

## Information Formats

| Dimension of msg | Value of fmt Argument | Dimension of code |
| :---: | :---: | :---: |
| Binary column or row vector | binary | Binary column or row vector |
| Example: msg = $\left.\begin{array}{lllllllllllll}0 & 1 & 1 & 0, & 1 & 0 & 1 & 1 & 0 & 1\end{array}\right] . '$ |  |  |
| Binary matrix with k columns | binary | Binary matrix with $n$ columns |
|  |  |  |
| Column or row vector of integers in the range [ $0,2^{\wedge} \mathrm{k}-1$ ] | decimal | Column or row vector of integers in the range [0, 2^n-1] |
| Example: msg = [6, 10, 9].' |  |  |

## For Specific Syntaxes

code $=$ encode(msg, n, k, 'linear/fmt', genmat) encodes msg using genmat as the generator matrix for the linear block encoding method. genmat, a k-by-n matrix, is required as input.
code = encode(msg, n, k, 'cyclic/fmt', genpoly) encodes msg and creates a systematic cyclic code. genpoly is a row vector that gives the coefficients, in order of ascending powers, of the binary generator polynomial. The default value of genpoly is cyclpoly ( $n, k$ ). By definition, the generator polynomial for an $[\mathrm{n}, \mathrm{k}]$ cyclic code must have degree $\mathrm{n}-\mathrm{k}$ and must divide $\mathrm{x}^{\mathrm{n}}-1$.
code $=$ encode(msg, n, $k$, 'hamming/fmt',prim_poly) encodes msg using the Hamming encoding method. For this syntax, $n$ must have the form $2^{\mathrm{m}}-1$ for some integer m greater than or equal to 3 , and k must equal $n-m$. prim_poly is a row vector that gives the binary coefficients, in order of ascending powers, of the primitive polynomial for $\mathrm{GF}\left(2^{\mathrm{m}}\right)$ that is used in the encoding process. The default value of prim_poly is the default primitive polynomial gfprimdf(m).
code $=$ encode(msg, $n, k$ ) is the same as code $=$ encode(msg, n, k, 'hamming/binary').
[code,added] = encode(...) returns the additional variable added. added is the number of zeros that were placed at the end of the message matrix before encoding in order for the matrix to have the appropriate shape. "Appropriate" depends on $n$, $k$, the shape of msg, and the encoding method.

Examples The example below illustrates the three different information formats (binary vector, binary matrix, and decimal vector) for Hamming code. The three messages have identical content in different formats; as a result, the three codes that encode creates have identical content in correspondingly different formats.

```
m = 4; n = 2^m-1; % Codeword length = 15
k = 11; % Message length
% Create 100 messages, k bits each.
msg1 = randi([0,1],100*k,1); % As a column vector
msg2 = vec2mat(msg1,k); % As a k-column matrix
msg3 = bi2de(msg2)'; % As a row vector of decimal integers
% Create 100 codewords, n bits each.
code1 = encode(msg1,n,k,'hamming/binary');
code2 = encode(msg2,n,k,'hamming/binary');
code3 = encode(msg3,n,k,'hamming/decimal');
if ( vec2mat(code1,n)==code2 & de2bi(code3',n)==code2 )
    disp('All three formats produced the same content.')
end
```

The output is
All three formats produced the same content.
The next example creates a cyclic code, adds noise, and then decodes the noisy code. It uses the decode function.

```
n = 3; k = 2; % A (3,2) cyclic code
msg = randi([0,1],100,k); % 100 messages, k bits each
code = encode(msg,n,k,'cyclic/binary');
% Add noise.
noisycode = rem(code + randerr(100,n,[0 1;.7 .3]), 2);
newmsg = decode(noisycode,n,k,'cyclic'); % Try to decode.
% Compute error rate for decoding the noisy code.
[number,ratio] = biterr(newmsg,msg);
disp(['The bit error rate is ',num2str(ratio)])
```

The output is below. Your error rate results might vary because the noise is random.

The bit error rate is 0.08
The next example encodes the same message using Hamming and cyclic methods. This example also creates Hamming code with the 'linear' option of the encode command. It then decodes each code and recovers the original message.

```
n = 7; % Codeword length
k = 4; % Message length
m = log2(n+1); % Express n as 2^m-1.
msg = randi([0,2^k-1],100,1); % Column of decimal integers
% Create various codes.
codehamming = encode(msg,n,k,'hamming/decimal');
[parmat,genmat] = hammgen(m);
codehamming2 = encode(msg,n,k,'linear/decimal',genmat);
if codehamming==codehamming2
    disp('The ''linear'' method can create Hamming code.')
```

```
end
codecyclic = encode(msg,n,k,'cyclic/decimal');
% Decode to recover the original message.
decodedhamming = decode(codehamming,n,k,'hamming/decimal');
decodedcyclic = decode(codecyclic,n,k,'cyclic/decimal');
if (decodedhamming==msg & decodedcyclic==msg)
    disp('All decoding worked flawlessly in this noiseless world.')
end
```

The output is
The 'linear' method can create Hamming code. All decoding worked flawlessly in this noiseless world.

## Algorithms

See Also
How To

Depending on the encoding method, encode relies on such lower-level functions as hammgen and cyclgen.

- "Block Codes"

Purpose
Equalize signal using equalizer object
Syntax
$y=$ equalize(eqobj, $x$ )
$y=$ equalize(eqobj, $x$,trainsig)
[y,yd] = equalize(...)
[y,yd,e] = equalize(...)
Description
$y=$ equalize(eqobj, $x$ ) processes the baseband signal vector $x$ with equalizer object eqobj and returns the equalized signal vector $y$. At the end of the process, eqobj contains updated state information such as equalizer weight values and input buffer values. To construct eqobj, use the lineareq or dfe function, as described in "Adaptive Algorithms". The equalize function assumes that the signal x is sampled at nsamp samples per symbol, where nsamp is the value of the nSampPerSym property of eqobj. For adaptive algorithms other than CMA, the equalizer adapts in decision-directed mode using a detector specified by the SigConst property of eqobj. The delay of the equalizer is (eqobj.RefTap-1)/eqobj.nSampPerSym, as described in "Delays from Equalization".

Note that (eqobj.RefTap-1) must be an integer multiple of nSampPerSym. For a fractionally-spaced equalizer, the taps are spaced at fractions of a symbol period. The reference tap pertains to training symbols, and thus, must coincide with a whole number of symbols (i.e., an integer number of samples per symbol). eqobj.RefTap=1 corresponds to the first symbol, eqobj.RefTap=nSampPerSym+1 to the second, and so on. Therefore (eqobj.RefTap-1) must be an integer multiple of nSampPerSym.

If eqobj.ResetBeforeFiltering is 0 , equalize uses the existing state information in eqobj when starting the equalization operation. As a result, equalize (eqobj, [x1 x2]) is equivalent to [equalize(eqobj, x1) equalize(eqobj, x2)]. To reset eqobj manually, apply the reset function to eqobj.
If eqobj.ResetBeforeFiltering is 1 , equalize resets eqobj before starting the equalization operation, overwriting any previous state information in eqobj.
$y=$ equalize(eqobj, $x$, trainsig) initially uses a training sequence to adapt the equalizer. After processing the training sequence, the equalizer adapts in decision-directed mode. The vector length of trainsig must be less than or equal to length (x)-(eqobj.RefTap-1)/eqobj.nSampPerSym.
[y,yd] = equalize(...) returns the vector yd of detected data symbols.
[y,yd,e] = equalize(...) returns the result of the error calculation described in "Error Calculation". For adaptive algorithms other than CMA, e is the vector of errors between y and the reference signal, where the reference signal consists of the training sequence or detected symbols.

\author{
Examples For examples that use this function, see "Equalize Using a Training Sequence in MATLAB", "Example: Equalizing Multiple Times, Varying the Mode", and "Example: Adaptive Equalization Within a Loop". <br> ```
See Also lms | signlms | normlms | varlms | rls | cma | lineareq | dfe <br> How To . "Equalization"

```
}

Purpose
Syntax
Description

Generate eye diagram
eyediagram ( \(x, n\) )
eyediagram(x, n, period)
eyediagram( \(x, n\), period, offset)
eyediagram(x, \(n, p e r i o d, o f f s e t, p l o t s t r i n g)\)
eyediagram(x, \(n\), period, offset, plotstring, \(h\) )
h = eyediagram(...)
eyediagram ( \(x, n\) ) creates an eye diagram for the signal \(x\), plotting \(n\) samples in each trace. \(n\) must be an integer greater than 1 . The labels on the horizontal axis of the diagram range between \(-1 / 2\) and \(1 / 2\). The function assumes that the first value of the signal, and every nth value thereafter, occur at integer times. The interpretation of \(x\) and the number of plots depend on the shape and complexity of \(x\) :
- If \(x\) is a real two-column matrix, eyediagram interprets the first column as in-phase components and the second column as quadrature components. The two components appear in different subplots of a single figure window.
- If x is a complex vector, eyediagram interprets the real part as in-phase components and the imaginary part as quadrature components. The two components appear in different subplots of a single figure window.
- If x is a real vector, eyediagram interprets it as a real signal. The figure window contains a single plot.
eyediagram( \(x, n\), period) is the same as the syntax above, except that the labels on the horizontal axis range between -period/2 and period/2.
eyediagram( \(x, n\), period, offset) is the same as the syntax above, except that the function assumes that the (offset+1)st value of the signal, and every nth value thereafter, occur at times that are integer multiples of period. The variable offset must be a nonnegative integer between 0 and \(\mathrm{n}-1\).
eyediagram( \(x, n\), period, offset, plotstring) is the same as the syntax above, except that plotstring determines the plotting symbol, line type, and color for the plot. plotstring is a string whose format and meaning are the same as in the plot function. The default string is 'b-', which produces a blue solid line.
eyediagram( \(x, n\), period, offset, plotstring, \(h\) ) is the same as the syntax above, except that the eye diagram is in the figure whose handle is \(h\), rather than in a new figure. \(h\) must be a handle to a figure that eyediagram previously generated.

Note You cannot use hold on to plot multiple signals in the same figure.
\(h=\) eyediagram(...) is the same as the earlier syntaxes, except that h is the handle to the figure that contains the eye diagram.

\section*{Examples \\ For an online demonstration, type showdemo scattereyedemo.}

See Also scatterplot | plot
How To . "Eye Diagram Analysis"

\section*{Purpose}

Launch eye diagram scope for eye diagram object H

\section*{Syntax}
eyescope eyescope(h)

Use EyeScope to examine the data in an eye diagram object. EyeScope shows both the eye diagram plot and measurement results in a unified, graphical environment, providing a very efficient means for viewing eye diagram data. There are two ways to call EyeScope:
- eyescope calls an empty scope
- eyescope(h) call EyeScope, displaying object \(h\)

\begin{abstract}
Note You can call EyeScope with an eye diagram object as the input argument. EyeScope uses the inputname function to resolve the caller's work space name for the argument. If the inputname function cannot resolve the caller's work space name, then EyeScope uses a default name. To learn about the cases when EyeScope can not determine the work space name, type help inputname at the MATLAB command line.
\end{abstract}

For more information, see "Eye Diagram Analysis".
Starting
EyeScope
To start EyeScope from the MATLAB command line, type:
eyescope
The following figure shows an EyeScope that does not have an eye diagram object loaded in its memory.


Alternatively, you can start EyeScope so it displays an eye diagram object. To start EyeScope so it displays an eye diagram object, type the following at the MATLAB command line:
```

eyescope(h)

```

Note \(h\) is a handle to an eye diagram object in the workspace.

The
EyeScope
Environment
- "EyeScope Menu Bar" on page 1-245
- "Eye Diagram Object Plot and Plot Controls" on page 1-245
- "Eye Diagram Object Settings Panel" on page 1-247
- "Measurements" on page 1-248

\section*{EyeScope Menu Bar}

EyeScope Menu Bar
The EyeScope menu bar is comprised of four menus: File, Options, View, and Help.
- Use the File menu to control the session management functions, import an eye diagram object into EyeScope, and export an eye diagram plot.
- Use the Options menu to setup the eye diagram scope by selecting which eye diagram settings and measurements EyeScope displays.
- Use the View menu to toggle between Single eye diagram view or Compare measurement results view, and to add or modify a legend for the eye diagram plot.
- The Help menu is used to access help pertaining to the eye diagram object and EyeScope.

\section*{Eye Diagram Object Plot and Plot Controls}

The Eye diagram object plot is the region of the GUI where the eye diagram plot appears.


Eye diagram plot controls are user-configurable settings that specify plot type, color scale, minimum and maximum plot PDF range, and plot time offset for the eye diagram being analyzed. To access the EyeScope plot controls Options > Eye Diagram Plot Controls


Note The value for the Plot time offset parameter can either be entered directly into the text box or set using the slide bar control.

For more information pertaining to the eye diagram properties, refer to the commscope.eyediagram reference page.

\section*{Eye Diagram Object Settings Panel}

The eye diagram object settings panel displays the eye diagram object settings. The default EyeScope configuration displays the following eye diagram object settings:
- Sampling frequency
- Symbol rate
- Eye level boundaries
- BER threshold
- Amplitude threshold

\section*{EyeScope}


To specify which eye diagram object settings display in EyeScope, refer to "Selecting Which Eye Diagram Object Settings To Display" on page \(1-255\). If you select additional eye diagram object settings to display in EyeScope, use the scroll buttons to view all of the settings.

\section*{Measurements}

The Measurements panel displays the eye diagram measurement settings. The default EyeScope configuration displays the following eye diagram object measurements:
- Horizontal Eye Opening
- Random Jitter
- Deterministic Jitter
- Total Jitter
- RMS Jitter
- Peak to Peak Jitter
- Vertical Opening
- Rise Time
- Fall Time
- Eye SNR
\begin{tabular}{|lrr|}
\hline Measurements \\
Measurements & & \\
Eye Crossing Time [1] (us): & Value & \\
Eye Crossing Time [2] (ms): & 11 & \\
Eye Level [1] (mAU): & -907 & \\
Eye Level [2] (mAU): & 906 \\
Eye Amplitude (AU): & 1.81 & \\
Eye SNR: & 9.66 & \\
Horizontal Eye Opening (ms): & 7.16 & \\
Random Jitter (ms): & 1.94 & \\
Deterministic Jitter (us): & 904 & \\
Total Jitter (ms): & 2.84 & V \\
& & \\
\hline
\end{tabular}

To select which eye diagram measurements EyeScope displays, refer to "Selecting Which Eye Diagram Measurements To Display" on page 1-256. If you select additional eye diagram object measurements to display in EyeScope, use the scroll buttons to view all of the settings.

\section*{Using \\ EyeScope}
- "Starting EyeScope with an Argument" on page 1-250
- "Starting a new Session" on page 1-250
- "Opening a Session" on page 1-250
- "Saving a Session" on page 1-251
- "Importing an Eye Diagram Object" on page 1-252
- "Printing to a Figure" on page 1-254
- "Selecting Which Eye Diagram Object Settings To Display" on page 1-255
- "Selecting Which Eye Diagram Measurements To Display" on page 1-256

\section*{EyeScope}

\section*{Starting EyeScope with an Argument}

You can start EyeScope so it is displaying an eye diagram object. To start EyeScope so it is displaying an eye diagram object, type the following at the MATLAB command line:
eyescope(h)

Note \(h\) is a handle to an eye diagram object presently in the workspace.

\section*{Starting a new Session}

Starting a new session purges EyeScope memory, returning EyeScope to an empty plot display. If changes have been made to an open session and you start a new session, you will be prompted to save the open session.

\section*{Opening a Session}

To open session, choose the file name and location of the session file. The file extensions for a session file is .eds, which stands for eye diagram scope. If changes have been made to a session that is presently open and you try to open up a new session, you will be prompted to save the session that is presently open before the new session can start.

To open a session:

\section*{1}

\section*{Click File > Open Session.}

The Select File To Open Window appears.


\section*{2}

Navigate to the EyeScope session file you want, and click Open.

\section*{Saving a Session}

The Save Session selection saves the current session, updating the session file. A session file includes the eye diagram object, eyescope options, and plot control selections.
If you attempt to save a session that you have not previously saved, EyeScope will prompt you for a file name and location. Otherwise, the session is saved to the previously selected file.

To save a session, follow these steps:

\section*{EyeScope}

\section*{Click File > Save Session.}

2
Navigate to the folder where you want to save the EyeScope session file and click Save.

\section*{Importing an Eye Diagram Object}

The Import menu selection imports an eye diagram object from either the workspace or a MAT-file to EyeScope. The imported variable name will be reconstructed to reflect the origin of the eye diagram object, as follows:
- If an object is imported from the workspace, the variable name will be ws_object name, where object name is the name of the original variable.
- If the object is imported from a MATLAB file, then the file name (without the path) precedes the object name.

Importing an object creates a copy of the object, using the naming convention previously described. EyeScope displays the object's contents as configured when the object was imported. EyeScope does not track any object changes made in the workspace (or to the MATLAB file) from which the object was imported.

To import an eye diagram object:

\section*{1}

\section*{Click File > Import Eye Diagram Object}

The Import eye diagram object window appears.


The contents panel of the of the Import eye diagram object window displays all eye diagram objects available in the source location.

\section*{2}

From the Import eye diagram object window, select the source for the object being imported.
- Select From workspace to import an eye diagram object directly from the workspace.
- Select From File to choose an eye diagram object file that was previously saved and click Browse to select the file to be loaded.

3
Click Import.

\section*{Printing to a Figure}

EyeScope allows you to print an eye diagram plot to a separate
MATLAB figure window. From the MATLAB figure window, along with other tasks, you can print, zoom, or edit the plot.
To export an eye diagram figure:
1
Click File > Print to Figure
The MATLAB figure window, containing the exported image, appears.


\section*{EyeScope}

\section*{Selecting Which Eye Diagram Object Settings To Display}

The Eye Diagram Object Settings View allows you to select which object settings display in the eye diagram object settings panel. You make your selections in the Configure eye diagram object settings view window, where a shuttle control allows you to add, remove, or reorder the settings you are displaying.

To add an eye diagram object setting:
1

\section*{Click Options > Eye Diagram Object Settings View}

The Configure eye diagram object settings view window appears.


2
Locate any items to be added in the list of Available items, and left-click to select.

\section*{EyeScope}

Note To select multiple items, you can either press and hold the <Shift> key and left-click or press and hold the <Ctrl> key and left-click.

When you select an item, the Quick help panel displays information about the item. If you select multiple items, Quick help displays information pertaining to the last item you select.

3
Click Add.

Note Using the Move Up orMove Down buttons, you can change the order in which the eye diagrams settings you select appear.

4
Click OK .

\section*{Selecting Which Eye Diagram Measurements To Display}

You can modify the contents of the measurement panel by selecting which eye diagram measurements display in the eye diagram object settings panel. You make your selections in the Configure measurements view window, where a shuttle control allows you to add, remove, or reorder the settings you are including.

Adding An Eye Diagram Measurement Setting

Click Options > Measurements View
The Configure measurements window appears.


2
Locate any items to be added in the list of Available items, and left-click to select.

Note To select multiple items, you can either press and hold the <Shift> key and left-click or press and hold the <Ctrl> key and left-click.

When you select an item, the Quick help panel displays information about the item. If you select multiple items, Quick help displays information pertaining to the last item you select.

3
Click Add.

Note Using the Move Up or Move Down buttons, you can change the order in which the eye diagrams settings you select appear.

4
Click OK .

\section*{Purpose}

Discrete Fourier transform

\section*{Syntax \\ fft(x)}

Description \(\quad f f t(x)\) is the discrete Fourier transform (DFT) of the Galois vector \(x\). If x is in the Galois field \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\), the length of x must be \(2^{\mathrm{m}}-1\).

\section*{Examples}
\(\mathrm{m}=4\);
\(\mathrm{n}=2^{\wedge} \mathrm{m}-1\);
\(x=\) gf(randi([0 2^m-1],n,1),m); \% Random vector \(y=f f t(x)\); \% Transform of \(x\) \(z=i f f t(y) ; \%\) Inverse transform of \(y\) ck \(=\) isequal \((z, x)\) \% Check that ifft(fft(x)) recovers \(x\).

The output is
ck =
1

\section*{Limitations}

Algorithms

See Also
How To

The Galois field over which this function works must have 256 or fewer elements. In other words, \(x\) must be in the Galois field \(\operatorname{GF}\left(2^{m}\right)\), where \(m\) is an integer between 1 and 8 .

If \(x\) is a column vector, fft applies dftmtx to the primitive element of the Galois field and multiplies the resulting matrix by x .
ifft | dftmtx
- "Signal Processing Operations in Galois Fields"

\section*{filter (channel)}
Purpose Filter signal with channel object
Syntax ..... y = filter(chan, \(x\) )
Description \(y=\) filter (chan, \(x\) ) processes the baseband signal vector \(x\) with thechannel object chan. The result is the signal vector \(y\). The final stateof the channel is stored in chan. You can construct chan using eitherrayleighchan or ricianchan. The filter function assumes \(x\) issampled at frequency \(1 /\) ts, where ts equals the InputSamplePeriodproperty of chan.
If chan.ResetBeforeFiltering is 0, filter uses the existing state information in chan when starting the filtering operation. As a result, filter(chan, [x1 x2]) is equivalent to [filter(chan, x 1 ) filter(chan, x 2\()\) ]. To reset chan manually, apply the reset function to chan.
If chan.ResetBeforeFiltering is 1, filter resets chan before starting the filtering operation, overwriting any previous state information in chan.
Examples Examples using this function are in "Use Fading Channels".
References [1] Jeruchim, Michel C., Philip Balaban, and K. Sam Shanmugan, Simulation of Communication Systems, Second Edition, New York, Kluwer Academic/Plenum, 2000.
See Also rayleighchan | ricianchan | reset
How To - "Fading Channels"

\section*{Purpose}

1-D digital filter over Galois field

\section*{Syntax}
\(y=\) filter \((b, a, x)\)
[y,zf] = filter(b,a,x)
\(y=\) filter \((b, a, x)\) filters the data in the vector \(x\) with the filter described by numerator coefficient vector \(b\) and denominator coefficient vector \(a\). The vectors \(b\), \(a\), and \(x\) must be Galois vectors in the same field. If a(1) is not equal to 1, filter normalizes the filter coefficients by a(1). As a result, a(1) must be nonzero.

The filter is a "Direct Form II Transposed" implementation of the standard difference equation below.
```

a(1)*y(n) = b(1)*x(n) + b(2)*x(n-1) + ... + b(nb+1)*x(n-nb) ...
-a(2)*y(n-1) - ... - a(na+1)*y(n-na)

```
\([y, z f]=\) filter \((b, a, x)\) returns the final conditions of the filter delays in the Galois vector \(z f\). The length of the vector \(z f\) is \(\max (\operatorname{size}(\mathrm{a}), \operatorname{size}(\mathrm{b}))-1\).

Examples An example is in "Huffman Coding".

\section*{finddelay}
\begin{tabular}{ll} 
Purpose & Estimate delay(s) between signals \\
Syntax & D \(=\) finddelay \((X, Y)\) \\
& \(D=\) finddelay \((\ldots\), MAXLAG \()\)
\end{tabular}

Description
\(D=\) finddelay \((X, Y)\), where \(X\) and \(Y\) are row or column vectors, returns an estimate of the delay \(D\) between \(X\) and \(Y\), where \(X\) serves as the reference vector. If \(Y\) is delayed with respect to \(X\), then \(D\) is positive. If \(Y\) is advanced with respect to \(X\), then \(D\) is negative. Delays in \(X\) and \(Y\) can be introduced by pre-pending zeros.
\(X\) and \(Y\) need not be exact delayed copies of each other, as finddelay ( \(\mathrm{X}, \mathrm{Y}\) ) returns an estimate of the delay via cross-correlation. However this estimated delay has a useful meaning only if there is sufficient correlation between delayed versions of \(X\) and \(Y\). Also, if several delays are possible, as in the case of periodic signals, the delay with the smallest absolute value is returned. In the case that both a positive and a negative delay with the same absolute value are possible, the positive delay is returned.
\(\mathrm{D}=\mathrm{finddelay}(\mathrm{X}, \mathrm{Y})\), where X is a matrix of size \(M X\)-by- \(N X(M X>1\) and \(N X>1\) ) and \(Y\) is a matrix of size \(M Y\)-by- \(N Y\) ( \(M Y>1\) and \(N Y>1\) ), returns a row vector \(D\) of estimated delays between each column of \(X\) and the corresponding column of \(Y\). With this usage the number of columns of \(X\) must be equal to the number of columns of \(Y\) (i.e., \(N X=N Y\) ).
\(D=\) finddelay (..., MAXLAG), uses MAXLAG as the maximum correlation window size used to find the estimated delay(s) between \(X\) and \(Y\). The usage of MAXLAG is detailed in the table below.

By default, MAXLAG is equal to \(\operatorname{MAX}(L X, L Y)-1\) for two vector inputs (where \(L X\) and \(L Y\) are the lengths of X and Y , respectively), \(\mathrm{MAX}(M X\), \(M Y\) )- 1 for two matrix inputs, and \(\operatorname{MAX}(L X, M Y)-1\) or \(\operatorname{MAX}(M X, L Y)-1\) for one vector input and one matrix input. If MAXLAG is input as [ ], it is replaced by the default value. If any element of MAXLAG is negative, it is replaced by its absolute value. If any element of MAXLAG is not integer-valued, or is complex, Inf, or NaN, then finddelay returns an error.

The calculation of the vector of estimated delays, D , depends on \(\mathrm{X}, \mathrm{Y}\), and MAXLAG as shown in the following table.
\begin{tabular}{|c|c|c|c|}
\hline MAXLAG & X & Y & D is calculated by... \\
\hline Integer-valued scalar & Row or column vector or matrix & Row or column vector or matrix & Cross-correlating the columns of \(X\) and \(Y\) over a range of lags -MAXLAG:MAXLAG. \\
\hline Integer-valued row or column vector & Row or column vector of length \(L X \geq 1\) & \[
\begin{aligned}
& \text { Matrix of size } \\
& M Y \text {-by- } N Y \\
& (M Y>1, N Y>1)
\end{aligned}
\] & Cross-correlating \(X\) and column \(j\) of \(Y\) over a range of lags -MAXLAG \((j): M A X L A G(j)\), for \(j=1: N Y\). \\
\hline Integer-valued row or column vector & \begin{tabular}{l}
Matrix of size \(M X\)-by- \(N X\) \\
( \(M X>1, N X>1\) )
\end{tabular} & Row or column vector of length \(L Y \geq 1\) & Cross-correlating column \(j\) of \(X\) and \(Y\) over a range of lags -MAXLAG( \(j\) ):MAXLAG( \(j\) ), for \(j=1: N X\). \\
\hline Integer-valued row or column vector & \begin{tabular}{l}
Matrix of size \(M X\)-by- \(N X\) \\
( \(M X>1, N X>1\) )
\end{tabular} & Matrix of size MY-by-NY ( \(M Y>1\), \(N Y=N X>1)\) & Cross-correlating column \(j\) of \(X\) and column \(j\) of \(Y\) over a range of lags -MAXLAG( \(j\) ):MAXLAG( \(j\) ), for \(j=1: N Y\). \\
\hline
\end{tabular}

\section*{Treating \(X\) as Multiple Channels}

If you wish to treat a row vector X of length \(L X\) as comprising one sample from \(L X\) different channels, you need to append one or more rows of zeros to \(X\) so that it appears as a matrix. Then each column of \(X\) will be considered a channel.

For example, \(X=\left[\begin{array}{llll}1 & 1 & 1 & 1\end{array}\right]\) is considered a single channel comprising four samples. To treat it as four different channels, each channel comprising one sample, define a new matrix Xm :
\[
\left.\mathrm{Xm}=\begin{array}{cccc}
{\left[\begin{array}{lll}
1 & 1 & 1 \\
1 ;
\end{array}\right.} \\
0 & 0 & 0 & 0
\end{array}\right] ;
\]

Each column of \(X m\) corresponds to a single channel, each one containing the samples 1 and 0 .

\section*{finddelay}

Algorithms

\section*{Examples}

The finddelay function uses the xcorr function of Signal Processing Toolbox to determine the cross-correlation between each pair of signals at all possible lags specified by the user. The normalized cross-correlation between each pair of signals is then calculated. The estimated delay is given by the negative of the lag for which the normalized cross-correlation has the largest absolute value.

If more than one lag leads to the largest absolute value of the cross-correlation, such as in the case of periodic signals, the delay is chosen as the negative of the smallest (in absolute value) of such lags.

Pairs of signals need not be exact delayed copies of each other. However, the estimated delay has a useful meaning only if there is sufficient correlation between at least one pair of the delayed signals.

\section*{\(X\) and \(Y\) Are Vectors, and MAXLAG Is Not Specified}

The following shows \(Y\) being delayed with respect to \(X\) by two samples.
```

X = [llll
Y = [0 0 1 2 3];
D = finddelay(X,Y)

```

The result is \(D=2\).
Here is a case of \(Y\) advanced with respect to \(X\) by three samples.
\(X=\left[\begin{array}{llllllll}0 & 0 & 0 & 1 & 2 & 3 & 0 & 0\end{array}\right]^{\prime} ;\)
\(Y=\left[\begin{array}{lll}1 & 2 & 3\end{array}\right]\) ';
D = finddelay (X, Y)
The result is \(\mathrm{D}=-3\).
The following illustrates a case where Y is aligned with X but is noisy.
```

X = [0 0 1 2 3 0];
Y = [0.02 0.12 1.08 2.21 2.95 -0.09];
D = finddelay(X,Y)

```

The result is \(\mathrm{D}=0\).

If \(Y\) is a periodic version of \(X\), the smallest possible delay is returned.
```

X = [llllll
Y = [1 2 3 0 0 0 0 1 2 3 0 0];
D = finddelay(X,Y)

```

The result is \(\mathrm{D}=-1\).

\section*{\(X\) is a Vector, \(Y\) a Matrix, and MAXLAG Is a Scalar}

MAXLAG is specified as a scalar (same maximum window sizes).
```

X = [0 1 2];
Y = [0 1 0 0;
120 0;
2 0 1 0;
O O 2 1];
MAXLAG = 3;
D = finddelay(X,Y,MAXLAG)

```

The result is \(\mathrm{D}=\left[\begin{array}{llll}0 & -1 & 1 & 1\end{array}\right]\).
\(X\) and \(Y\) Are Matrices, and MAXLAG Is Not Specified
```

X = [l0 1 0 0;
12 0 0;
2 0 1 0;
10 2 1;
O O O 2];
Y = [lllll
1 1 0;
2 2 0 1;
10 0 2;
0 0 0 0];
D = finddelay(X,Y)

```

The result is \(D=\left[\begin{array}{llll}0 & 1 & -2 & -1\end{array}\right]\).

\section*{finddelay}

X and Y Are Matrices, and MAXLAG Is Specified
X = \(\begin{array}{llll}0 & 1 & 0 & 0 ;\end{array}\)
1200 ;
2010 ;
102 1;
\(00021 ;\)
\(Y=\left[\begin{array}{llll}0 & 0 & 1 & 0 ;\end{array}\right.\)
\(1120 ;\)
2201 ;
100 2;
\(00001 ;\)
MAXLAG = [10 1020 20];
\(\mathrm{D}=\) finddelay \((\mathrm{X}, \mathrm{Y}\), MAXLAG \()\)
The result is \(\mathrm{D}=\left[\begin{array}{llll}0 & 1 & -2 & -1\end{array}\right]\).
See Also alignsignals | xcorr
\begin{tabular}{|c|c|}
\hline Purpose & Frequency demodulation \\
\hline Syntax & \begin{tabular}{l}
\(z=f m d e m o d(y, F c, F s, f r e q d e v)\) \\
z = fmdemod(y,Fc,Fs,freqdev,ini_phase)
\end{tabular} \\
\hline Description & \begin{tabular}{l}
\(z=f m d e m o d(y, F c, F s, f r e q d e v)\) demodulates the modulating signal \(z\) from the carrier signal using frequency demodulation. The carrier signal has frequency \(\mathrm{Fc}(\mathrm{Hz})\) and sampling rate \(\mathrm{Fs}(\mathrm{Hz})\), where Fs must be at least \(2 *\) Fc. The freqdev argument is the frequency deviation \((\mathrm{Hz})\) of the modulated signal \(y\). \\
\(z=f m d e m o d\left(y, F c, F s, f r e q d e v, i n i \_p h a s e\right)\) specifies the initial phase of the modulated signal, in radians.
\end{tabular} \\
\hline Examples & An example using fmdemod is on the reference page for fmmod. \\
\hline See Also & fmmod | pmmod | pmdemod \\
\hline How To & . "Digital Modulation" \\
\hline
\end{tabular}

\section*{fmmod}
\begin{tabular}{ll} 
Purpose & Frequency modulation \\
Syntax & \(y=f m m o d(x, F c, F s, f r e q d e v)\) \\
& \(y=f m m o d(x, F c, F s, f r e q d e v\), ini_phase \()\)
\end{tabular}

Description

\section*{Examples}

See Also fmdemod \| ammod \| pmmod
```

Purpose Frequency shift keying demodulation
Syntax $\quad z=$ fskdemod $\left(y, M, f r e q \_s e p, n s a m p\right)$
z = fskdemod(y,M,freq_sep,nsamp,Fs)
z = fskdemod(y,M,freq_sep,nsamp,Fs,symbol_order)

```

\section*{Description}

\section*{Examples}

The example below illustrates FSK modulation and demodulation over an AWGN channel.
```

M = 2; k = log2(M);
EbNo = 5;
Fs = 16; nsamp = 17; freqsep = 8;
msg = randi([0 M-1],5000,1); % Random signal
txsig = fskmod(msg,M,freqsep,nsamp,Fs); % Modulate.
msg_rx = awgn(txsig,EbNo+10*log10(k)-10*log10(nsamp),...
'measured',[],'dB'); % AWGN channel
msg_rrx = fskdemod(msg_rx,M,freqsep,nsamp,Fs); % Demodulate
[num,BER] = biterr(msg,msg_rrx) % Bit error rate
BER_theory = berawgn(EbNo,'fsk',M,'noncoherent') % Theoretical BER

```

\section*{fskdemod}

The output is shown below. Your BER value might vary because the example uses random numbers.
```

BER =
0.1086
BER_theory =
0.1029

```
See Also
fskmod \| pskmod \| pskdemod

How To . "Digital Modulation"
```

Purpose
Frequency shift keying modulation
Syntax $\quad y=$ fskmod $\left(x, M, f r e q \_s e p, n s a m p\right)$
y = fskmod(x, M,freq_sep, nsamp,Fs)
y = fskmod(x,M,freq_sep, nsamp,Fs, phase_cont)
y = FSKMOD(x,M,freq_sep,nsamp,Fs, phase_cont,symbol_order)

```

Description

Examples
\(y=f s k m o d\left(x, M, f r e q \_s e p, n s a m p\right)\) outputs the complex envelope \(y\) of the modulation of the message signal \(x\) using frequency shift keying modulation. M is the alphabet size and must be an integer power of 2. The message signal must consist of integers between 0 and \(\mathrm{M}-1\). freq_sep is the desired separation between successive frequencies in Hz. nsamp denotes the number of samples per symbol in y and must be a positive integer greater than 1 . The sampling rate of y is 1 Hz. By the Nyquist sampling theorem, freq_sep and \(M\) must satisfy \((M-1) * f r e q \_s e p<=1\). If \(x\) is a matrix with multiple rows and columns, the function processes the columns independently.
y = fskmod(x,M,freq_sep, nsamp,Fs) specifies the sampling rate of \(y\) in Hz. Because the Nyquist sampling theorem implies that the maximum frequency must be no larger than \(\mathrm{Fs} / 2\), the inputs must satisfy (M-1)*freq_sep <= Fs.
\(y=\) fskmod \(\left(x, M, f r e q \_s e p, n s a m p, F s\right.\), phase_cont \()\) specifies the phase continuity. Set phase_cont to 'cont' to force phase continuity across symbol boundaries in \(y\), or 'discont' to avoid forcing phase continuity. The default is 'cont'.
y = FSKMOD(x,M,freq_sep,nsamp,Fs,phase_cont,symbol_order) specifies how the function assigns binary words to corresponding integers. If symbol_order is set to 'bin' (default), the function uses a natural binary-coded ordering. If symbol_order is set to 'gray', it uses a Gray-coded ordering.

The example below illustrates the syntax of fskmod using a random signal.
```

M = 4; freqsep = 8; nsamp = 8; Fs = 32;

```
```

x = randi([0 M-1],1000,1); % Random signal
y = fskmod(x,M,freqsep,nsamp,Fs); % Modulate.
ly = length(y);
% Create an FFT plot.
freq = [-Fs/2 : Fs/ly : Fs/2 - Fs/ly];
Syy = 10*log10(fftshift(abs(fft(y))));
plot(freq,Syy)

```


See Also
How To
fskdemod | pskmod \| pskdemod
- "Digital Modulation"

\section*{Purpose}

Convert between parity-check and generator matrices
Syntax

Description

\section*{Examples}
```

parmat = gen2par(genmat)

```
genmat \(=\) gen2par(parmat)
parmat = gen2par(genmat) converts the standard-form binary generator matrix genmat into the corresponding parity-check matrix parmat.
genmat = gen2par(parmat) converts the standard-form binary parity-check matrix parmat into the corresponding generator matrix genmat.

The standard forms of the generator and parity-check matrices for an [ \(\mathrm{n}, \mathrm{k}\) ] binary linear block code are shown in the table below
\begin{tabular}{l|l|l}
\hline Type of Matrix & Standard Form & Dimensions \\
\hline Generator & {\(\left[\mathrm{I}_{\mathrm{k}} \mathrm{P}\right]\) or \(\left[\mathrm{P}_{\mathrm{k}}\right]\)} & k -by-n \\
\hline Parity-check & {\(\left[-\mathrm{P}^{\prime} \mathrm{I}_{\mathrm{n}-\mathrm{k}}\right]\) or \(\left[\mathrm{I}_{\mathrm{n}-\mathrm{k}}-\mathrm{P}^{\prime}\right]\)} & \((\mathrm{n}-\mathrm{k})\)-by-n \\
\hline
\end{tabular}
where \(I_{k}\) is the identity matrix of size \(k\) and the ' symbol indicates matrix transpose. Two standard forms are listed for each type, because different authors use different conventions. For binary codes, the minus signs in the parity-check form listed above are irrelevant; that is, \(-1=1\) in the binary field.

The commands below convert the parity-check matrix for a Hamming code into the corresponding generator matrix and back again.
```

parmat = hammgen(3)
genmat1 = gen2par(parmat)
parmat2 = gen2par(genmat1) % Ans should be the same as parmat above
The output is
parmat =

```
\begin{tabular}{ccccccc}
1 & 0 & 0 & 1 & 0 & 1 & 1 \\
0 & 1 & 0 & 1 & 1 & 1 & 0 \\
0 & 0 & 1 & 0 & 1 & 1 & 1 \\
genmat \(=\) & & & & & & \\
& & & & & & \\
1 & 1 & 0 & 1 & 0 & 0 & 0 \\
0 & 1 & 1 & 0 & 1 & 0 & 0 \\
1 & 1 & 1 & 0 & 0 & 1 & 0 \\
1 & 0 & 1 & 0 & 0 & 0 & 1 \\
parmat2 \(=\) & & & & & & \\
& & & & & & \\
1 & 0 & 0 & 1 & 0 & 1 & 1 \\
0 & 1 & 0 & 1 & 1 & 1 & 0 \\
0 & 0 & 1 & 0 & 1 & 1 & 1
\end{tabular}
See Also
cyclgen | hammgen
How To
- "Block Codes"

\section*{genqamdemod}
Purpose General quadrature amplitude demodulation
Syntax ..... z = genqamdemod(y,const)
Description Warning
This function is obsolete and may be removed in thefuture. We strongly recommend that you use thecomm. GeneralQAMDemodulator System object instead.
\(z=\) genqamdemod(y, const) demodulates the complex envelope \(y\) of a quadrature amplitude modulated signal. The complex vector const specifies the signal mapping. If \(y\) is a matrix with multiple rows, the function processes the columns independently.
Examples The reference page for genqammod has an example that uses genqamdemod.
See Also genqammod | qammod \| qamdemod \| pammod \| pamdemod
How To - "Digital Modulation"

Purpose General quadrature amplitude modulation
Syntax \(\quad y=\operatorname{genqammod}(x\), const \()\)
Description

\section*{Examples}

\section*{Warning} System object instead.
\(y=\) genqammod(x, const) outputs the complex envelope \(y\) of the modulation of the message signal \(x\) using quadrature amplitude modulation. The message signal must consist of integers between 0 mapping. If \(x\) is a matrix with multiple rows, the function processes the columns independently.

The code below plots a signal constellation that has a hexagonal

This function is obsolete and may be removed in the future. We strongly recommend that you use the comm. GeneralQAMModulator and length (const) -1 . The complex vector const specifies the signal structure. It also uses genqammod and genqamdemod to modulate and demodulate a message [3 8 5 10 7] using this constellation.
```

% Describe hexagonal constellation.

```
% Describe hexagonal constellation.
inphase = [1/2 1 1 1/2 1/2 2 2 5/2];
inphase = [1/2 1 1 1/2 1/2 2 2 5/2];
quadr = [00 1 -1 2 -2 1 -1 0];
quadr = [00 1 -1 2 -2 1 -1 0];
inphase = [inphase;-inphase]; inphase = inphase(:);
inphase = [inphase;-inphase]; inphase = inphase(:);
quadr = [quadr;quadr]; quadr = quadr(:);
quadr = [quadr;quadr]; quadr = quadr(:);
const = inphase + j*quadr;
const = inphase + j*quadr;
% Plot constellation.
% Plot constellation.
h = scatterplot(const);
h = scatterplot(const);
% Modulate message using this constellation.
% Modulate message using this constellation.
x = [3 8 8 5 10 7]; % Message signal
x = [3 8 8 5 10 7]; % Message signal
y = genqammod(x,const);
y = genqammod(x,const);
z = genqamdemod(y,const); % Demodulate.
z = genqamdemod(y,const); % Demodulate.
% Plot modulated signal in same figure.
```

% Plot modulated signal in same figure.

```
```

hold on; scatterplot(y,1,0,'ro',h);
legend('Constellation','Modulated signal','Location','NorthWest'); % Include legend.
hold off;

```


Another example using this function is the Gray-coded constellation example in "Examples of Signal Constellation Plots".

\author{
See Also
}
genqamdemod | qammod \| qamdemod | pammod | pamdemod
How To - "Digital Modulation"
\begin{tabular}{|c|c|}
\hline Purpose & Create Galois field array \\
\hline \multirow[t]{3}{*}{Syntax} & \(x \_g f=g f(x, m)\) \\
\hline & x_gf = gf(x,m, prim_poly) \\
\hline & \(x \_g f=g f(x)\) \\
\hline
\end{tabular}

\section*{Description}
\(x \_g f=g f(x, m)\) creates a Galois field array from the matrix \(x\). The Galois field has 2^m elements, where \(m\) is an integer between 1 and 16. The elements of \(x\) must be integers between 0 and \(2^{\wedge} m-1\). The output \(\mathrm{x} \_\mathrm{gf}\) is a variable that MATLAB recognizes as a Galois field array, rather than an array of integers. As a result, when you manipulate x_gf using operators or functions such as + or det, MATLAB works within the Galois field you have specified.

Note To learn how to manipulate x_gf using familiar MATLAB operators and functions, see "Galois Field Computations". To learn how the integers in \(x\) represent elements of \(\mathrm{GF}\left(2^{\wedge} \mathrm{m}\right)\), see "How Integers Correspond to Galois Field Elements".
\(\mathrm{x} \_\mathrm{gf}=\mathrm{gf}\left(\mathrm{x}, \mathrm{m}, \mathrm{prim} \_\right.\)poly) is the same as the previous syntax, except it uses the primitive polynomial prim_poly to define the field. prim_poly is the integer representation of a primitive polynomial. For example, the number 37 represents the polynomial \(D^{\wedge} 5+D^{\wedge} 2+1\) because the binary form of 37 is 100101 . For more information about the primitive polynomial, see "Specifying the Primitive Polynomial".
\(x \_g f=g f(x)\) creates a GF(2) array from the matrix x. Each element of \(x\) must be 0 or 1 .

\section*{Default Primitive Polynomials}

The table below lists the primitive polynomial that gf uses by default for each Galois field GF( \(\left.2^{\wedge} m\right)\). To use a different primitive polynomial, specify prim_poly as an input argument when you invoke gf.
\begin{tabular}{l|l|l}
\hline \(\mathbf{m}\) & \begin{tabular}{l} 
Default Primitive \\
Polynomial
\end{tabular} & \begin{tabular}{l} 
Integer \\
Representation
\end{tabular} \\
\hline 1 & \(\mathrm{D}+1\) & 3 \\
\hline 2 & \(\mathrm{D}^{\wedge} 2+\mathrm{D}+1\) & 7 \\
\hline 3 & \(\mathrm{D}^{\wedge} 3+\mathrm{D}+1\) & 11 \\
\hline 4 & \(\mathrm{D}^{\wedge} 4+\mathrm{D}+1\) & 19 \\
\hline 5 & \(\mathrm{D}^{\wedge} 5+\mathrm{D}^{\wedge} 2+1\) & 37 \\
\hline 6 & \(\mathrm{D}^{\wedge} 6+\mathrm{D}+1\) & 67 \\
\hline 7 & \(\mathrm{D}^{\wedge} 7+\mathrm{D}^{\wedge} 3+1\) & 137 \\
\hline 8 & \begin{tabular}{l}
\(\mathrm{D}^{\wedge} 8+\mathrm{D}^{\wedge} 4+\mathrm{D}^{\wedge} 3+\) \\
\(\mathrm{D}^{\wedge} 2+1\)
\end{tabular} & 285 \\
\hline 9 & \(\mathrm{D}^{\wedge} 9+\mathrm{D}^{\wedge} 4+1\) & 529 \\
\hline 10 & \(\mathrm{D}^{\wedge} 10+\mathrm{D}^{\wedge} 3+1\) & 1033 \\
\hline 11 & \(\mathrm{D}^{\wedge} 11+\mathrm{D}^{\wedge} 2+1\) & 2053 \\
\hline 12 & \begin{tabular}{l}
\(\mathrm{D}^{\wedge} 12+\mathrm{D}^{\wedge} 6+\mathrm{D}^{\wedge} 4+\) \\
\(\mathrm{D}^{2}+1\)
\end{tabular} & 4179 \\
\hline 13 & \begin{tabular}{l}
\(\mathrm{D}^{\wedge} 13+\mathrm{D}^{\wedge} 4+\mathrm{D}^{\wedge} 3+\) \\
\(\mathrm{D}^{2}+1\)
\end{tabular} & 8219 \\
\hline 14 & \begin{tabular}{l}
\(\mathrm{D}^{\wedge} 14+\mathrm{D}^{\wedge} 10+\mathrm{D}^{\wedge} 6\) \\
\(+\mathrm{D}^{2}+1\)
\end{tabular} & 17475 \\
\hline 15 & \(\mathrm{D}^{\wedge} 15+\mathrm{D}^{2}+1\) & 32771 \\
\hline 16 & \begin{tabular}{l}
\(\mathrm{D}^{\wedge} 16+\mathrm{D}^{\wedge} 12+\mathrm{D}^{\wedge} 3\) \\
\(+\mathrm{D}^{2}+1\)
\end{tabular} & 69643 \\
\hline
\end{tabular}

Examples For examples that use gf, see
- "Example: Creating Galois Field Variables"
- "Example: Representing a Primitive Element"
- Other sample code within "Galois Field Computations"
- The Galois field demonstration: type showdemo gfdemo.

\section*{See Also gftable}

How To
- Galois field computations
- "Galois Field Computations"

Purpose Add polynomials over Galois field
Syntax
\(c=\operatorname{gfadd}(a, b)\)
\(c=\operatorname{gfadd}(a, b, p)\)
\(c=\operatorname{gfadd}(a, b, p, l e n)\)
c = gfadd(a,b,field)

\section*{Description}

Note This function performs computations in \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\) where p is prime. To work in \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\), apply the + operator to Galois arrays of equal size. For details, see "Example: Addition and Subtraction".
\(c=\operatorname{gfadd}(a, b)\) adds two GF(2) polynomials, \(a\) and \(b\). If \(a\) and \(b\) are vectors of the same orientation but different lengths, then the shorter vector is zero-padded. If \(a\) and \(b\) are matrices they must be of the same size.
\(c=\operatorname{gfadd}(a, b, p)\) adds two \(G F(p)\) polynomials, where \(p\) is a prime number. \(a, b\), and \(c\) are row vectors that give the coefficients of the corresponding polynomials in order of ascending powers. Each coefficient is between 0 and \(p-1\). If a and \(b\) are matrices of the same size, the function treats each row independently.
\(c=\operatorname{gfadd}(a, b, p, l e n)\) adds row vectors \(a\) and \(b\) as in the previous syntax, except that it returns a row vector of length len. The output c is a truncated or extended representation of the sum. If the row vector corresponding to the sum has fewer than len entries (including zeros), extra zeros are added at the end; if it has more than len entries, entries from the end are removed.
\(c=\operatorname{gfadd}(a, b, f i e l d)\) adds two \(G F\left(p^{m}\right)\) elements, where \(m\) is a positive integer. \(a\) and \(b\) are the exponential format of the two elements, relative to some primitive element of \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\). field is the matrix listing all elements of \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\), arranged relative to the same primitive element. c is the exponential format of the sum, relative to the same primitive element. See "Representing Elements of Galois Fields" for an

\section*{gfadd}
explanation of these formats. If \(a\) and \(b\) are matrices of the same size, the function treats each element independently.

\section*{Examples}

In the code below, sum 5 is the sum of \(2+3 x+x^{2}\) and \(4+2 x+3 x^{2}\) over \(\mathrm{GF}(5)\), and linpart is the degree-one part of sum5.
```

sum5 = gfadd([2 3 1],[4 2 3],5)
linpart = gfadd([2 3 1],[4 2 3],5,2)

```

The output is
sum5 \(=\)
\begin{tabular}{ccc}
1 & 0 & 4 \\
linpart \(=\) & \\
1 & 0
\end{tabular}

The code below shows that \(\mathrm{A}^{2}+\mathrm{A}^{4}=\mathrm{A}^{1}\), where \(A\) is a root of the primitive polynomial \(2+2 x+x^{2}\) for \(\mathrm{GF}(9)\).
```

p = 3; m = 2;
prim_poly = [2 2 1];
field = gftuple([-1:p^m-2]',prim_poly,p);
g = gfadd(2,4,field)

```

The output is
\(\mathrm{g}=\)
1
Other examples are in "Arithmetic in Galois Fields".
See Also
gfsub | gfconv | gfmul | gfdeconv | gfdiv | gftuple

\section*{Purpose Multiply polynomials over Galois field}

Syntax
\(c=\operatorname{gfconv}(a, b)\)
\(c=\operatorname{gfconv}(a, b, p)\)
\(c=\operatorname{gfconv}(a, b, f i e l d)\)

\section*{Description}

Note This function performs computations in GF( \(p^{m}\) ), where p is prime. To work in \(\operatorname{GF}\left(2^{\mathrm{m}}\right)\), use the conv function with Galois arrays. For details, see "Multiplication and Division of Polynomials".

The gfconv function multiplies polynomials over a Galois field. (To multiply elements of a Galois field, use gfmul instead.) Algebraically, multiplying polynomials over a Galois field is equivalent to convolving vectors containing the polynomials' coefficients, where the convolution operation uses arithmetic over the same Galois field.
\(c=\operatorname{gfconv}(a, b)\) multiplies two \(G F(2)\) polynomials, \(a\) and \(b\). The polynomial degree of the resulting \(\mathrm{GF}(2)\) polynomial c equals the degree of a plus the degree of \(b\).
\(c=g f c o n v(a, b, p)\) multiplies two \(G F(p)\) polynomials, where \(p\) is a prime number. \(a, b\), and \(c\) are row vectors that give the coefficients of the corresponding polynomials in order of ascending powers. Each coefficient is between 0 and \(\mathrm{p}-1\).
\(c=\operatorname{gfconv}(a, b, f i e l d)\) multiplies two \(G F\left(p^{m}\right)\) polynomials, where \(p\) is a prime number and \(m\) is a positive integer. \(a, b\), and \(c\) are row vectors that list the exponential formats of the coefficients of the corresponding polynomials, in order of ascending powers. The exponential format is relative to some primitive element of \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\). field is the matrix listing all elements of \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\), arranged relative to the same primitive element. See "Representing Elements of Galois Fields" for an explanation of these formats.

Examples The command below shows that
\[
\left(1+x+x^{4}\right)\left(x+x^{2}\right)=x+2 x^{2}+x^{3}+x^{5}+x^{6}
\]
over GF(3).
gfc \(=\operatorname{gfconv}\left(\left[\begin{array}{lllll}1 & 1 & 0 & 0 & 1\end{array}\right],\left[\begin{array}{lll}0 & 1 & 1\end{array}\right], 3\right)\)
The output is
\(\mathrm{gfc}=\)
\begin{tabular}{lllllll}
0 & 1 & 2 & 1 & 0 & 1 & 1
\end{tabular}

The code below illustrates the identity
\[
\left(x^{r}+x^{s}\right)^{p}=x^{r p}+x^{s p}
\]
for the case in which \(p=7, r=5\), and \(s=3\). (The identity holds when \(p\) is any prime number, and \(r\) and \(s\) are positive integers.)
```

p = 7; r = 5; s = 3;
a = gfrepcov([r s]); % x^r + x^s
% Compute a^p over GF(p).
c = 1;
for ii = 1:p
c = gfconv(c,a,p);
end;
% Check whether c = x^(rp) + x^(sp).
powers = [];
for ii = 1:length(c)
if c(ii)~=0
powers = [powers, ii];
end;
end;
if (powers==[r*p+1 s*p+1] | powers==[s*p+1 r*p+1])

```
```

    disp('The identity is proved for this case of r, s, and p.')
    end

```

See Also gfdeconv \| gfadd \| gfsub \| gfmul | gftuple

Purpose Produce cyclotomic cosets for Galois field
Syntax
c = gfcosets(m)
c = gfcosets(m,p)

\section*{Description}

Note This function performs computations in GF( \(p^{m}\) ), where \(p\) is prime. To work in \(\operatorname{GF}\left(2^{\mathrm{m}}\right)\), use the cosets function.
c = gfcosets(m) produces cyclotomic cosets mod( \(\left.2^{m}-1\right)\). Each row of the output GFCS contains one cyclotomic coset.
\(c=g f \operatorname{cosets}(m, p)\) produces the cyclotomic cosets for GF(p^m), where \(m\) is a positive integer and \(p\) is a prime number.

The output matrix c is structured so that each row represents one coset. The row represents the coset by giving the exponential format of the elements of the coset, relative to the default primitive polynomial for the field. For a description of exponential formats, see "Representing Elements of Galois Fields".

The first column contains the coset leaders. Because the lengths of cosets might vary, entries of NaN are used to fill the extra spaces when necessary to make c rectangular.

A cyclotomic coset is a set of elements that all satisfy the same minimal polynomial. For more details on cyclotomic cosets, see the works listed in "References" on page 1-287.

Examples The command below finds the cyclotomic cosets for GF(9).
\(c=\operatorname{gfcosets}(2,3)\)
The output is
c \(=\)
\[
0 \quad \mathrm{NaN}
\]

13
26
4 NaN
\(5 \quad 7\)

The gfminpol function can check that the elements of, for example, the third row of c indeed belong in the same coset.
\(\mathrm{m}=[\mathrm{gfminpol}(2,2,3) ; \operatorname{gfminpol}(6,2,3)] \%\) Rows are identical.
The output is
\(\mathrm{m}=\)
\begin{tabular}{lll}
1 & 0 & 1 \\
1 & 0 & 1
\end{tabular}

\footnotetext{
References [1] Blahut, Richard E., Theory and Practice of Error Control Codes, Reading, MA, Addison-Wesley, 1983, p. 105.
[2] Lin, Shu, and Daniel J. Costello, Jr., Error Control Coding: Fundamentals and Applications, Englewood Cliffs, NJ, Prentice-Hall, 1983.
}

See Also gfminpol | gfprimdf | gfroots
\begin{tabular}{|c|c|}
\hline Purpose & Divide polynomials over Galois field \\
\hline \multirow[t]{3}{*}{Syntax} & [quot, remd] = gfdeconv(b,a) \\
\hline & [quot, remd] = gfdeconv(b,a, p) \\
\hline & [quot, remd] = gfdeconv(b,a,field) \\
\hline
\end{tabular}

\section*{Description}

Note This function performs computations in GF(p \({ }^{m}\) ), where \(p\) is prime. To work in \(\operatorname{GF}\left(2^{\mathrm{m}}\right)\), use the deconv function with Galois arrays. For details, see "Multiplication and Division of Polynomials".

The gfdeconv function divides polynomials over a Galois field. (To divide elements of a Galois field, use gfdiv instead.) Algebraically, dividing polynomials over a Galois field is equivalent to deconvolving vectors containing the polynomials' coefficients, where the deconvolution operation uses arithmetic over the same Galois field.
[quot, remd] = gfdeconv(b,a) computes the quotient quot and remainder remd of the division of \(b\) by a in \(\operatorname{GF}(2)\).
[quot, remd] = gfdeconv(b,a,p) divides the polynomial \(b\) by the polynomial a over GF(p) and returns the quotient in quot and the remainder in remd. \(p\) is a prime number. \(b, a\), quot, and remd are row vectors that give the coefficients of the corresponding polynomials in order of ascending powers. Each coefficient is between 0 and p-1.
[quot,remd] = gfdeconv(b,a,field) divides the polynomial by the polynomial a over \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\) and returns the quotient in quot and the remainder in remd. Here \(p\) is a prime number and \(m\) is a positive integer. b, a, quot, and remd are row vectors that list the exponential formats of the coefficients of the corresponding polynomials, in order of ascending powers. The exponential format is relative to some primitive element of \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\). field is the matrix listing all elements of \(\mathrm{GF}\left(\mathrm{p}^{m}\right)\), arranged relative to the same primitive element. See "Representing Elements of Galois Fields" for an explanation of these formats.

Examples The code below shows that
\[
\left(x+x^{3}+x^{4}\right) \div(1+x)=1+x^{3} \text { Remainder } 2
\]
in \(\mathrm{GF}(3)\). It also checks the results of the division.
```

p = 3;
b = [0 1 0 1 1]; a = [1 1];
[quot, remd] = gfdeconv(b,a,p)
% Check the result.
bnew = gfadd(gfconv(quot,a,p),remd,p);
if isequal(bnew,b)
disp('Correct.')
end;

```

The output is below.
quot \(=\)
10001
remd \(=\)

2
Correct.
Working over GF(3), the code below outputs those polynomials of the form \(\mathrm{x}^{\mathrm{k}}-1(\mathrm{k}=2,3,4, \ldots, 8)\) that \(1+\mathrm{x}^{2}\) divides evenly.
```

p = 3; m = 2;
a = [110 1]; \% 1+x^2
for $i i=2: p^{\wedge} m-1$
b = gfrepcov(ii) ; \% x^ii
$b(1)=p-1$; \% -1+x^ii
[quot, remd] = gfdeconv(b,a,p);
\% Display -1+x^ii if a divides it evenly.
if remd==0

```

\section*{gfdeconv}
```

        multiple{ii}=b;
        gfpretty(b)
    end
    end

```

The output is below.
\[
\begin{aligned}
& 2+x^{4} \\
& 2+x^{8}
\end{aligned}
\]

In light of the discussion in "Algorithms" on page 1-303 on the gfprimck reference page, along with the irreducibility of \(1+x^{2}\) over GF(3), this output indicates that \(1+x^{2}\) is not primitive for \(\mathrm{GF}(9)\).

\section*{Algorithms The algorithm of gfdeconv is similar to that of the MATLAB function deconv.}

\author{
See Also gfconv | gfadd | gfsub | gfdiv | gftuple
}

\section*{Purpose Divide elements of Galois field}
```

Syntax quot = gfdiv(b,a)
quot = gfdiv(b,a,p)
quot = gfdiv(b,a,field)

```

\section*{Description}

\section*{Examples}

The code below displays lists of multiplicative inverses in GF(5) and GF(25). It uses column vectors as inputs to gfdiv.

\section*{gfdiv}
```

    % Find inverses of nonzero elements of GF(5).
    p = 5;
    b = ones(p-1,1);
    a = [1:p-1];
    quot1 = gfdiv(b,a,p);
disp('Inverses in GF(5):')
disp('element inverse')
disp([a, quot1])
% Find inverses of nonzero elements of GF(25).
m = 2;
field = gftuple([-1:p^m-2]',m,p);
b = zeros(p^m-1,1); % Numerator is zero since 1 = alpha^0.
a = [0:p^m-2]';
quot2 = gfdiv(b,a,field);
disp('Inverses in GF(25), expressed in EXPONENTIAL FORMAT with')
disp('respect to a root of the default primitive polynomial:')
disp('element inverse')
disp([a, quot2])

```

See Also gfmul | gfdeconv \| gfconv \| gftuple

\section*{Purpose}

Filter data using polynomials over prime Galois field

\section*{Syntax}

\section*{Description}
\[
\begin{aligned}
& y=\operatorname{gffilter}(b, a, x) \\
& y=\operatorname{gffilter}(b, a, x, p)
\end{aligned}
\]

Note This function performs computations in GF( \(p^{m}\) ), where \(p\) is prime. To work in \(\operatorname{GF}\left(2^{\mathrm{m}}\right)\), use the filter function with Galois arrays. For details, see "Filtering".
\(y=g f f i l t e r(b, a, x)\) filters the data in vector \(x\) with the filter described by vectors \(b\) and \(a\). The vectors \(b\), \(a\) and \(x\) must be in GF(2), that is, be binary and \(y\) is also in GF(2).
\(y=g f f i l t e r(b, a, x, p)\) filters the data \(x\) using the filter described by vectors \(a\) and \(b\). \(y\) is the filtered data in \(\operatorname{GF}(p)\). \(p\) is a prime number, and all entries of \(a\) and \(b\) are between 0 and \(p-1\).
By definition of the filter, \(y\) solves the difference equation
\[
\begin{aligned}
& a(1) y(n)=b(1) x(n)+b(2) x(n-1)+b(3) x(n-2)+\ldots+b(B+1) x(n-B) \\
&-a(2) y(n-1)-a(3) y(n-2)-\ldots-a(A+1) y(n-A)
\end{aligned}
\]
where
- A+1 is the length of the vector a
- \(B+1\) is the length of the vector \(b\)
- \(n\) varies between 1 and the length of the vector \(x\).

The vector a represents the degree- \(\mathrm{n}_{\mathrm{a}}\) polynomial
\(a(1)+a(2) x+a(3) x^{\wedge} 2+\ldots+a(A+1) x^{\wedge} A\)

\section*{Examples}

The impulse response of a particular filter is given in the code and diagram below.
```

b = [1 0 0 1 0 1 0 1];
a = [1 0 1 1];
y = gffilter(b,a,[1,zeros(1,19)]);
stem(y);
axis([0 20 -.1 1.1])

```


See Also gfconv \| gfadd | filter

\section*{Purpose}

Find particular solution of \(\mathrm{Ax}=\mathrm{b}\) over prime Galois field
Syntax
\(x=g f l i n e q(A, b)\)
\(x=g f l i n e q(A, b, p)\)
[x,vld] = gflineq(...)

\section*{Description}

\section*{Examples}

The code below produces some valid solutions of a linear equation over GF(3).
```

A = [2 0 1;
1 1 0;
1 1 2];
% An example in which the solutions are valid
[x,vld] = gflineq(A,[1;0;0],3)

```

The output is below.
x =
2
1
0
vld =
1
By contrast, the command below finds that the linear equation has no solutions.
[x2,vld2] = gflineq(zeros (3,3),[2;0;0],3)
The output is below.
This linear equation has no solution.
x2 =
[]
vld2 =
0

\section*{Algorithms gflineq uses Gaussian elimination.}

See Also gfadd | gfdiv | gfroots | gfrank | gfconv | conv

\section*{Purpose}

Find minimal polynomial of Galois field element
Syntax
```

pol = gfminpol(k,m)
pol = gfminpol(k,m,p)
pol = gfminpol(k,prim_poly,p)

```

\section*{Description}

Note This function performs computations in GF(p \({ }^{m}\) ), where p is prime. To work in \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\), use the minpol function with Galois arrays. For details, see "Minimal Polynomials".
pol = gfminpol( \(k, m\) ) produces a minimal polynomial for each entry in \(k\). \(k\) must be either a scalar or a column vector. Each entry in k represents an element of \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\) in exponential format. That is, k represents alpha^k, where alpha is a primitive element in \(\operatorname{GF}\left(2^{\mathrm{m}}\right)\). The \(i\) th row of pol represents the minimal polynomial of \(\mathrm{k}(i)\). The coefficients of the minimal polynomial are in the base field GF(2) and listed in order of ascending exponents.
pol \(=\operatorname{gfminpol}(k, m, p)\) finds the minimal polynomial of \(A^{k}\) over \(\mathrm{GF}(\mathrm{p})\), where p is a prime number, m is an integer greater than 1 , and A is a root of the default primitive polynomial for \(\mathrm{GF}\left(\mathrm{p}^{\wedge} \mathrm{m}\right)\). The format of the output is as follows:
- If \(k\) is a nonnegative integer, pol is a row vector that gives the coefficients of the minimal polynomial in order of ascending powers.
- If k is a vector of length len all of whose entries are nonnegative integers, pol is a matrix having len rows; the rth row of pol gives the coefficients of the minimal polynomial of \(\mathrm{A}^{\mathrm{k}(r)}\) in order of ascending powers.
pol = gfminpol(k,prim_poly, \(p\) ) is the same as the first syntax listed, except that A is a root of the primitive polynomial for \(G F\left(\mathrm{p}^{m}\right)\) specified by prim_poly. prim_poly is a row vector that gives the coefficients of the degree-m primitive polynomial in order of ascending powers.

Examples The syntax \(\operatorname{gfminpol}(\mathrm{k}, \mathrm{m}, \mathrm{p})\) is used in the sample code in "Characterization of Polynomials".

See Also gfprimdf | gfcosets | gfroots

\section*{Purpose \\ Multiply elements of Galois field}

Syntax
\(c=\operatorname{gfmul}(a, b, p)\)
\(c=\) gfmul(a,b,field)

\section*{Examples "Arithmetic in Galois Fields" contains examples. Also, the code below} shows that
\[
A^{2} \cdot A^{4}=A^{6}
\]
where \(A\) is a root of the primitive polynomial \(2+2 \mathrm{x}+\mathrm{x}^{2}\) for GF(9).
```

p = 3; m = 2;
prim_poly = [2 2 1];
field = gftuple([-1:p^m-2]',prim_poly,p);

```
\[
a=\operatorname{gfmul}(2,4, f i e l d)
\]

The output is
\(\mathrm{a}=\)
6
See Also gfdiv | gfdeconv \| gfadd | gfsub | gftuple

\section*{Purpose Polynomial in traditional format}

Syntax
gfpretty(a)
gfpretty(a,st)
gfpretty(a,st,n)

\section*{Description}
gfpretty (a) displays a polynomial in a traditional format, using \(X\) as the variable and the entries of the row vector a as the coefficients in order of ascending powers. The polynomial is displayed in order of ascending powers. Terms having a zero coefficient are not displayed.
gfpretty (a,st) is the same as the first syntax listed, except that the content of the string st is used as the variable instead of \(X\).
gfpretty (a, st, \(n\) ) is the same as the first syntax listed, except that the content of the string st is used as the variable instead of \(X\), and each line of the display has width n instead of the default value of 79 .

Note For all syntaxes: If you do not use a fixed-width font, the spacing in the display might not look correct.

\section*{Examples \\ Display statements about the elements of GF(81).}
```

p = 3; m = 4;
ii = randi([1,p^m-2],1,1); % Random exponent for prim element
primpolys = gfprimfd(m,'all',p);
[rows, cols] = size(primpolys);
jj = randi([1,rows],1,1); % Random primitive polynomial
disp('If A is a root of the primitive polynomial')
gfpretty(primpolys(jj,:)) % Polynomial in X
disp('then the element')
gfpretty([zeros(1,ii),1],'A') % The polynomial A^ii
disp('can also be expressed as')
gfpretty(gftuple(ii,m,p),'A') % Polynomial in A

```

Below is a sample of the output.

If \(A\) is a root of the primitive polynomial
\[
2+2 x^{3}+x^{4}
\]
then the element

22
A
can also be expressed as
\[
2+A^{2}+A^{3}
\]

See Also gftuple | gfprimdf

\section*{Purpose}

Check whether polynomial over Galois field is primitive

\section*{Syntax}

\section*{Description}

\section*{Examples}

Algorithms

References
ck = gfprimck(a)
ck = gfprimck(a,p)

Note This function performs computations in GF(p \({ }^{m}\) ), where p is prime. If you are working in \(\operatorname{GF}\left(2^{\mathrm{m}}\right)\), use the isprimitive function. For details, see "Finding Primitive Polynomials".
ck = gfprimck(a) checks whether the degree-m GF(2) polynomial a is a primitive polynomial for \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\), where \(\mathrm{m}=\) length(a) -1 . The output ck is as follows:
- - 1 if a is not an irreducible polynomial
- 0 if a is irreducible but not a primitive polynomial for \(\mathrm{GF}\left(\mathrm{p}^{m}\right)\)
- 1 if a is a primitive polynomial for \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\)
ck = gfprimck(a,p) checks whether the degree-m GF(P) polynomial a is a primitive polynomial for \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\). p is a prime number.

This function considers the zero polynomial to be "not irreducible" and considers all polynomials of degree zero or one to be primitive.
"Characterization of Polynomials" contains examples.
An irreducible polynomial over \(\mathrm{GF}(\mathrm{p})\) of degree at least 2 is primitive if and only if it does not divide \(-1+\mathrm{x}^{\mathrm{k}}\) for any positive integer k smaller than \(\mathrm{p}^{\mathrm{m}}-1\).
[1] Clark, George C. Jr., and J. Bibb Cain, Error-Correction Coding for Digital Communications, New York, Plenum, 1981.
[2] Krogsgaard, K., and T., Karp, Fast Identification of Primitive Polynomials over Galois Fields: Results from a Course Project, ICASSP 2005, Philadelphia, PA, 2004.

\section*{gfprimck}

See Also gfprimfd \| gfprimdf \| gftuple \| gfminpol \| gfadd

\section*{Purpose Provide default primitive polynomials for Galois field}

Syntax
```

pol = gfprimdf(m)
pol = gfprimdf(m,p)

```

Note This function performs computations in GF(p \({ }^{m}\) ), where p is prime. To work in \(\operatorname{GF}\left(2^{m}\right)\), use the primpoly function. For details, see "Finding Primitive Polynomials".
pol = gfprimdf(m) outputs the default primitive polynomial pol in \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\).
pol \(=\operatorname{gfprimdf}(m, p)\) returns the row vector that gives the coefficients, in order of ascending powers, of the default primitive polynomial for \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\). m is a positive integer and p is a prime number.

Examples The command below shows that \(2+\mathrm{x}+\mathrm{x}^{2}\) is the default primitive polynomial for GF( \(5^{2}\) ).
```

pol = gfprimdf(2,5)

```
pol =
    211

The code below displays the default primitive polynomial for each of the fields GF( \(3^{\mathrm{m}}\) ), where m ranges between 3 and 5 .
```

for m = 3:5
gfpretty(gfprimdf(m,3))
end

```

The output is below.
\[
\begin{aligned}
& 2+x+x^{4} \\
& 1+2 x+x^{5}
\end{aligned}
\]

See Also
gfprimck | gfprimfd | gftuple | gfminpol

\section*{Purpose}

Find primitive polynomials for Galois field

\section*{Syntax}
pol = gfprimfd(m,opt, p)

Note This function performs computations in GF(p \({ }^{m}\) ), where p is prime. To work in \(\operatorname{GF}\left(2^{\mathrm{m}}\right)\), use the primpoly function. For details, see "Finding Primitive Polynomials".
- If \(m=1\), pol = [11].
- A polynomial is represented as a row containing the coefficients in order of ascending powers.
pol = gfprimfd(m,opt, \(p\) ) searches for one or more primitive polynomials for \(G F\left(p^{\wedge} m\right)\), where \(p\) is a prime number and \(m\) is a positive integer. If \(m=1\), pol \(=\left[\begin{array}{ll}11\end{array}\right]\). If \(m>1\), the output pol depends on the argument opt as shown in the table below. Each polynomial is represented in pol as a row containing the coefficients in order of ascending powers.
\begin{tabular}{l|l|l}
\hline opt & Significance of pol & Format of pol \\
\hline 'min' & \begin{tabular}{l} 
One primitive \\
polynomial for \\
GF(p^m) having the \\
smallest possible \\
number of nonzero \\
terms
\end{tabular} & \begin{tabular}{l} 
The row vector \\
representing the \\
polynomial
\end{tabular} \\
\hline 'max' & \begin{tabular}{l} 
One primitive \\
polynomial for \\
GF(p^m) having the \\
greatest possible \\
number of nonzero \\
terms
\end{tabular} & \begin{tabular}{l} 
The row vector \\
representing the \\
polynomial
\end{tabular} \\
\hline
\end{tabular}

\section*{gfprimfd}
\begin{tabular}{l|l|l}
\hline opt & Significance of pol & Format of pol \\
\hline 'all' & \begin{tabular}{l} 
All primitive \\
polynomials for \\
GF(p^m)
\end{tabular} & \begin{tabular}{l} 
A matrix, each row of \\
which represents one \\
such polynomial
\end{tabular} \\
\hline A positive integer & \begin{tabular}{l} 
All primitive \\
polynomials for \\
GF(p^m) that have \\
opt nonzero terms
\end{tabular} & \begin{tabular}{l} 
A matrix, each row of \\
which represents one \\
such polynomial
\end{tabular} \\
\hline
\end{tabular}

\section*{Examples}

The code below seeks primitive polynomials for GF(81) having various other properties. Notice that fourterms is empty because no primitive polynomial for GF(81) has exactly four nonzero terms. Also notice that fewterms represents a single polynomial having three terms, while threeterms represents all of the three-term primitive polynomials for GF(81).
```

p = 3; m = 4; % Work in GF(81).
fewterms = gfprimfd(m,'min',p)
threeterms = gfprimfd(m,3,p)
fourterms = gfprimfd(m,4,p)

```

The output is below.
```

fewterms =
2 1 0
threeterms =

| 2 | 1 | 0 | 0 | 1 |
| :--- | :--- | :--- | :--- | :--- |
| 2 | 2 | 0 | 0 | 1 |
| 2 | 0 | 0 | 1 | 1 |
| 2 | 0 | 0 | 2 | 1 |

```
```

No primitive polynomial satisfies the given constraints.
fourterms =

```

\section*{[]}

\section*{Algorithms}

See Also
gfprimfd tests for primitivity using gfprimck. If opt is 'min', 'max', or omitted, polynomials are constructed by converting decimal integers to base \(p\). Based on the decimal ordering, gfprimfd returns the first polynomial it finds that satisfies the appropriate conditions.
gfprimck | gfprimdf | gftuple | gfminpol

Purpose Compute rank of matrix over Galois field
\[
\text { Syntax } \quad r k=\operatorname{gfrank}(A, p)
\]

\section*{Description}

Note This function performs computations in \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\) where p is prime. If you are working in \(\operatorname{GF}\left(2^{\mathrm{m}}\right)\), use the rank function with Galois arrays. For details, see "Computing Ranks".
\(r k=\operatorname{gfrank}(A, p)\) calculates the rank of the matrix \(A\) in \(G F(p)\), where \(p\) is a prime number.

\section*{Algorithms}
gfrank uses an algorithm similar to Gaussian elimination.
Examples In the code below, gfrank says that the matrix \(A\) has less than full rank. This conclusion makes sense because the determinant of A is zero \(\bmod p\).
```

A = [$$
\begin{array}{lll}{1}&{0}&{1;}\end{array}
$$]
2 1 0;
0 1 1];
p = 3;
det_a = det(A); % Ordinary determinant of A
detmodp = rem(det(A),p); % Determinant mod p
rankp = gfrank(A,p);
disp(['Determinant = ',num2str(det_a)])
disp(['Determinant mod p is ',num2str(detmodp)])
disp(['Rank over GF(p) is ',num2str(rankp)])

```

The output is below.
```

Determinant = 3
Determinant mod p is 0
Rank over GF(p) is 2

```

\section*{Purpose}

Convert one binary polynomial representation to another

\section*{Syntax}

Description
```

polystandard = gfrepcov(poly2)

```

Two logical ways to represent polynomials over GF(2) are listed below.

1 [A_0 A_1 A_2 ... A_(m-1)] represents the polynomial
\[
\mathrm{A} \_0+\mathrm{A} \_1 x+\mathrm{A} \_2 x^{2}+\cdots+\mathrm{A} \_(\mathrm{m}-1) x^{m-1}
\]

Each entry A_k is either one or zero.
2 [A_0 A_1 A_2 ... A_(m-1)] represents the polynomial
\[
x^{\mathrm{A} \_0}+x^{\mathrm{A} \_1}+x^{\mathrm{A} \_2}+\cdots+x^{\mathrm{A} \_(\mathrm{m}-1)}
\]

Each entry A_k is a nonnegative integer. All entries must be distinct.
Format \(\mathbf{1}\) is the standard form used by the Galois field functions in this toolbox, but there are some cases in which format \(\mathbf{2}\) is more convenient. polystandard \(=\) gfrepcov(poly2) converts from the second format to the first, for polynomials of degree at least 2 . poly2 and polystandard are row vectors. The entries of poly2 are distinct integers, and at least one entry must exceed 1 . Each entry of polystandard is either 0 or 1.

Note If poly2 is a binary row vector, gfrepcov assumes that it is already in Format 1 above and returns it unaltered.

\section*{Examples}

The command below converts the representation format of the polynomial \(1+x^{2}+x^{5}\).
```

polystandard = gfrepcov([0 2 5])

```
polystandard =

See Also gfpretty
```

Purpose Find roots of polynomial over prime Galois field
Syntax $\quad r t=\operatorname{gfroots}(f, m, p)$
rt = gfroots(f,prim_poly,p)
[rt,rt_tuple] = gfroots(...)
[rt,rt_tuple,field] = gfroots(...)

```

Note This function performs computations in \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\), where p is prime. To work in \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\), use the roots function with Galois arrays. For details, see "Roots of Polynomials".

For all syntaxes, \(f\) is a row vector that gives the coefficients, in order of ascending powers, of a degree-d polynomial.

Note gfroots lists each root exactly once, ignoring multiplicities of roots.
\(r t=\) gfroots (f,m,p) finds roots in GF(p^m) of the polynomial that \(f\) represents. \(r t\) is a column vector each of whose entries is the exponential format of a root. The exponential format is relative to a root of the default primitive polynomial for \(\mathrm{GF}\left(\mathrm{p}^{\wedge} \mathrm{m}\right)\).
rt = gfroots(f,prim_poly, p) finds roots in \(\mathrm{GF}\left(\mathrm{p}^{m}\right)\) of the polynomial that \(f\) represents. \(r t\) is a column vector each of whose entries is the exponential format of a root. The exponential format is relative to a root of the degree-m primitive polynomial for \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\) that prim_poly represents.
[rt,rt_tuple] = gfroots(...) returns an additional matrix \(r t\) tuple, whose kth row is the polynomial format of the root \(\mathrm{rt}(\mathrm{k})\). The polynomial and exponential formats are both relative to the same primitive element.
[rt,rt_tuple,field] = gfroots(...) returns additional matrices \(r t \_t u p l e\) and field. \(r t\) _tuple is described in the preceding paragraph. field gives the list of elements of the extension field. The list of elements, the polynomial format, and the exponential format are all relative to the same primitive element.

Note For a description of the various formats that gfroots uses, see "Representing Elements of Galois Fields".

\section*{Examples}

See Also gfprimdf

\section*{Purpose Subtract polynomials over Galois field}

Syntax
\(c=\operatorname{gfsub}(a, b, p)\)
\(c=\operatorname{gfsub}(a, b, p, l e n)\)
\(c=g f s u b(a, b, f i e l d)\)

\section*{Description}

Note This function performs computations in GF( \(p^{m}\) ), where p is prime. To work in \(\operatorname{GF}\left(2^{\mathrm{m}}\right)\), apply the - operator to Galois arrays of equal size. For details, see "Example: Addition and Subtraction".
\(c=g f s u b(a, b, p)\) calculates a minus \(b\), where \(a\) and \(b\) represent polynomials over \(G F(p)\) and \(p\) is a prime number. \(a, b\), and \(c\) are row vectors that give the coefficients of the corresponding polynomials in order of ascending powers. Each coefficient is between 0 and p-1. If \(a\) and \(b\) are matrices of the same size, the function treats each row independently.
\(c=\) gfsub(a,b,p,len) subtracts row vectors as in the syntax above, except that it returns a row vector of length len. The output c is a truncated or extended representation of the answer. If the row vector corresponding to the answer has fewer than len entries (including zeros), extra zeros are added at the end; if it has more than len entries, entries from the end are removed.
\(c=\) gfsub(a,b,field) calculates a minus \(b\), where \(a\) and \(b\) are the exponential format of two elements of \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\), relative to some primitive element of \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\). p is a prime number and m is a positive integer. field is the matrix listing all elements of GF( \(\mathrm{p}^{m}\) ), arranged relative to the same primitive element. c is the exponential format of the answer, relative to the same primitive element. See "Representing Elements of Galois Fields" for an explanation of these formats. If a and \(b\) are matrices of the same size, the function treats each element independently.

Examples In the code below, differ is the difference of \(2+3 x+x^{2}\) and \(4+2 x+3 x^{2}\) over GF(5), and linpart is the degree-one part of differ.
```

differ = gfsub([2 3 1],[4 2 3],5)
linpart = gfsub([2 3 1],[4 2 3],5,2)

```

The output is
```

differ =

```
    313
linpart =
    31

The code below shows that \(A^{2}-A^{4}=A^{7}\), where \(A\) is a root of the primitive polynomial \(2+2 x+x^{2}\) for GF(9).
```

p = 3; m = 2;
prim_poly = [2 2 1];
field = gftuple([-1:p^m-2]',prim_poly,p);
d = gfsub(2,4,field)

```

The output is
d =
7
See Also
gfadd | gfconv | gfmul | gfdeconv | gfdiv | gftuple

Purpose
Generate file to accelerate Galois field computations

Syntax
Description
gftable(m,prim_poly);
gftable(m, prim_poly) generates a file that can help accelerate computations in the field \(\mathrm{GF}\left(2^{\wedge} \mathrm{m}\right)\) as described by the nondefault primitive polynomial prim_poly. The integer \(m\) is between 1 and 16 . The integer prim_poly represents a primitive polynomial for GF( \(2^{\wedge}\) m) using the format described in "Specifying the Primitive Polynomial". The function places the file, called userGftable.mat, in your current working folder. If necessary, the function overwrites any writable existing version of the file.

Note If prim_poly is the default primitive polynomial for GF( \(2^{\wedge} \mathrm{m}\) ) listed in the table on the gf reference page, this function has no effect. A MAT-file in your MATLAB installation already includes information that facilitates computations with respect to the default primitive polynomial.

\section*{Examples}

See Also gf

\section*{gftable}

\author{
How To \\ - "Speed and Nondefault Primitive Polynomials"
}

\section*{Purpose Minimize length of polynomial representation}

\section*{Syntax \\ c = gftrunc(a)}

Description \(\quad c=\) gftrunc(a) truncates a row vector, \(a\), that gives the coefficients of a \(\mathrm{GF}(\mathrm{p})\) polynomial in order of ascending powers. If \(\mathrm{a}(\mathrm{k})=0\) whenever \(\mathrm{k}>\mathrm{d}+1\), the polynomial has degree d . The row vector c omits these high-order zeros and thus has length \(\mathrm{d}+1\).

Examples In the code below, zeros are removed from the end, but not from the beginning or middle, of the row-vector representation of \(x^{2}+2 x^{3}+3 x^{4}+4 x^{7}+5 x^{8}\).
\(c=\) gftrunc ([0 00123004500\(])\)
C \(=\)
\begin{tabular}{lllllllll}
0 & 0 & 1 & 2 & 3 & 0 & 0 & 4 & 5
\end{tabular}

See Also
gfadd | gfsub | gfconv | gfdeconv | gftuple
```

Purpose Simplify or convert Galois field element formatting
Syntax $\quad$ tp $=$ gftuple $(a, m)$
tp = gftuple(a, prim_poly)
tp = gftuple(a,m,p)
tp = gftuple(a, prim_poly,p)
tp = gftuple(a,prim_poly,p,prim_ck)
[tp,expform] = gftuple(...)

```

\section*{Description}

Note This function performs computations in \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\), where p is prime. To perform equivalent computations in \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\), apply the . \({ }^{\wedge}\) operator and the log function to Galois arrays. For more information, see "Example: Exponentiation" and "Example: Elementwise Logarithm".

\section*{For All Syntaxes}
gftuple serves to simplify the polynomial or exponential format of Galois field elements, or to convert from one format to another. For an explanation of the formats that gftuple uses, see "Representing Elements of Galois Fields".

In this discussion, the format of an element of \(\mathrm{GF}\left(\mathrm{p}^{\mathrm{m}}\right)\) is called "simplest" if all exponents of the primitive element are
- Between 0 and m-1 for the polynomial format
- Either - Inf, or between 0 and \(\mathrm{p}^{\mathrm{m}-2}\), for the exponential format

For all syntaxes, a is a matrix, each row of which represents an element of a Galois field. The format of a determines how MATLAB interprets it:
- If a is a column of integers, MATLAB interprets each row as an exponential format of an element. Negative integers are equivalent to - Inf in that they all represent the zero element of the field.
- If a has more than one column, MATLAB interprets each row as a polynomial format of an element. (Each entry of a must be an integer between 0 and \(\mathrm{p}-1\).)

The exponential or polynomial formats mentioned above are all relative to a primitive element specified by the second input argument. The second argument is described below.

\section*{For Specific Syntaxes}
tp = gftuple ( \(\mathrm{a}, \mathrm{m}\) ) returns the simplest polynomial format of the elements that a represents, where the kth row of tp corresponds to the kth row of a. The formats are relative to a root of the default primitive polynomial for \(\mathrm{GF}\left(2^{\wedge} \mathrm{m}\right)\), where m is a positive integer.
tp = gftuple(a, prim_poly) is the same as the syntax above, except that prim_poly is a row vector that lists the coefficients of a degree \(m\) primitive polynomial for \(\mathrm{GF}\left(2^{\wedge} \mathrm{m}\right)\) in order of ascending exponents.
\(\mathrm{tp}=\mathrm{gftuple}(\mathrm{a}, \mathrm{m}, \mathrm{p})\) is the same as \(\mathrm{tp}=\mathrm{gftuple}(\mathrm{a}, \mathrm{m})\) except that 2 is replaced by a prime number \(p\).
tp = gftuple(a,prim_poly,p) is the same as tp = gftuple(a,prim_poly) except that 2 is replaced by a prime number \(p\).
tp = gftuple(a,prim_poly, p,prim_ck) is the same as tp = gftuple(a,prim_poly,p) except that gftuple checks whether prim_poly represents a polynomial that is indeed primitive. If not, then gftuple generates an error and \(t p\) is not returned. The input argument prim_ck can be any number or string; only its existence matters.
[tp,expform] = gftuple(...) returns the additional matrix expform. The kth row of expform is the simplest exponential format of the element that the kth row of a represents. All other features are as described in earlier parts of this "Description" section, depending on the input arguments.
- "List of All Elements of a Galois Field" (end of section)
- "Converting to Simplest Polynomial Format"

As another example, the gftuple command below generates a list of elements of \(\mathrm{GF}\left(\mathrm{p}^{\wedge} \mathrm{m}\right)\), arranged relative to a root of the default primitive polynomial. Some functions in this toolbox use such a list as an input argument.

\section*{gftuple}
```

p = 5; % Or any prime number
m = 4; % Or any positive integer
field = gftuple([-1:p^m-2]',m,p);

```

Finally, the two commands below illustrate the influence of the shape of the input matrix. In the first command, a column vector is treated as a sequence of elements expressed in exponential format. In the second command, a row vector is treated as a single element expressed in polynomial format.
```

tp1 = gftuple([0; 1],3,3)
tp2 = gftuple([0, 0, 0, 1],3,3)

```

The output is below.
tp1 =
100
\(0 \quad 1 \quad 0\)
tp2 =

210

The outputs reflect that, according to the default primitive polynomial for \(\mathrm{GF}\left(3^{3}\right)\), the relations below are true.
\[
\begin{aligned}
& \alpha^{0}=1+0 \alpha+0 \alpha^{2} \\
& \alpha^{1}=0+1 \alpha+0 \alpha^{2} \\
& 0+0 \alpha+0 \alpha^{2}+\alpha^{3}=2+\alpha+0 \alpha^{2}
\end{aligned}
\]

\section*{Algorithms}

See Also
gftuple uses recursive callbacks to determine the exponential format.
gfadd | gfmul | gfconv | gfdiv | gfdeconv | gfprimdf

\section*{Purpose Calculate minimum distance of linear block code}
```

Syntax wt = gfweight(genmat)
wt = gfweight(genmat,'gen')
wt = gfweight(parmat,'par')
wt = gfweight(genpoly,n)

```

Description

Examples

The minimum distance, or minimum weight, of a linear block code is defined as the smallest positive number of nonzero entries in any n -tuple that is a codeword.
wt = gfweight(genmat) returns the minimum distance of the linear block code whose generator matrix is genmat.
wt = gfweight(genmat, 'gen') returns the minimum distance of the linear block code whose generator matrix is genmat.
wt = gfweight(parmat, 'par') returns the minimum distance of the linear block code whose parity-check matrix is parmat.
wt = gfweight(genpoly, \(n\) ) returns the minimum distance of the cyclic code whose codeword length is n and whose generator polynomial is represented by genpoly. genpoly is a row vector that gives the coefficients of the generator polynomial in order of ascending powers.

The commands below illustrate three different ways to compute the minimum distance of a \((7,4)\) cyclic code.
```

n = 7;
% Generator polynomial of (7,4) cyclic code
genpoly = cyclpoly(n,4);
[parmat, genmat] = cyclgen(n,genpoly);
wts = [gfweight(genmat,'gen'),gfweight(parmat,'par'),...
gfweight(genpoly,n)]

```

The output is
wts =

\section*{gfweight}
33

\author{
See Also hammgen | cyclpoly | bchgenpoly \\ How To \\ . "Block Codes"
}

\section*{Purpose}

Syntax

Description

Convert Gray-encoded positive integers to corresponding Gray-decoded integers
```

y = gray2bin(x,modulation,M)
[y,map] = gray2bin(x,modulation,M)

```
\(y=\) gray2bin( \(x\), modulation, \(M\) ) generates a Gray-decoded output vector or matrix \(y\) with the same dimensions as its input parameter \(x . x\) can be a scalar, vector, or matrix. modulation is the modulation type and must be a string equal to 'qam', 'pam', 'fsk', 'dpsk', or 'psk'. M is the modulation order that can be an integer power of 2 .
[y,map] = gray2bin(x,modulation, M) generates a Gray-decoded outputy with its respective Gray-encoded constellation map, map.

You can use map output to label a Gray-encoded constellation. The map output gives the Gray encoded labels for the corresponding modulation. See the example below.

Note If you are converting binary coded data to Gray-coded data and modulating the result immediately afterwards, you should use the appropriate modulation object or function with the 'Gray ' option, instead of BIN2GRAY.

\section*{Examples}
```

% To Gray decode a vector x with a 16-QAM Gray encoded
% constellation and return its map, use:
x=randi([0 15],1,100);
[y,map] = gray2bin(x,'qam',16);
% Obtain the symbols for 16-QAM
hMod = modem.qammod('M', 16);
symbols = hMod.Constellation;
% Plot the constellation
scatterplot(symbols);
set(get(gca,'Children'),'Marker','d','MarkerFaceColor','auto');
hold on;

```

\section*{gray2bin}
```

% Label the constellation points according
% to the Gray mapping
for jj=1:16
text(real(symbols(jj))-0.15,imag(symbols(jj))+0.15,...
dec2base(map(jj),2,4));
end
set(gca,'yTick',(-4:2:4),'xTick',(-4:2:4),...
'XLim',[-4 4],'YLim',...
[-4 4],'Box','on','YGrid','on', 'XGrid','on');

```

The example code generates the following plot, which shows the 16 QAM constellation with Gray-encoded labeling.


\section*{See Also}
bin2gray

\section*{Purpose}

Produce parity-check and generator matrices for Hamming code
Syntax
\(\mathrm{h}=\) hammgen (m)
\(\mathrm{h}=\) hammgen(m,pol)
[h,g] = hammgen(...)
[h,g,n,k] = hammgen(...)
Description
For all syntaxes, the codeword length is \(n\). \(n\) has the form \(2^{\mathrm{m}}-1\) for some
positive integer \(m\) greater than or equal to 3 . The message length, \(k\), has the form \(n-m\).
\(\mathrm{h}=\) hammgen(m) produces an m-by-n parity-check matrix for a Hamming code having codeword length \(n=2^{\wedge} m-1\). The input \(m\) is a positive integer greater than or equal to 3 . The message length of the code is \(\mathrm{n}-\mathrm{m}\). The binary primitive polynomial used to produce the Hamming code is the default primitive polynomial for \(\mathrm{GF}\left(2^{\wedge} \mathrm{m}\right)\), represented by gfprimdf(m).
\(\mathrm{h}=\) hammgen (m, pol) produces an m-by-n parity-check matrix for a Hamming code having codeword length \(n=2 \wedge m-1\). The input \(m\) is a positive integer greater than or equal to 3 . The message length of the code is n-m. pol is a row vector that gives the coefficients, in order of ascending powers, of the binary primitive polynomial for \(\mathrm{GF}\left(2^{\wedge} \mathrm{m}\right)\) that is used to produce the Hamming code. hammgen produces an error if pol represents a polynomial that is not, in fact, primitive.
\([\mathrm{h}, \mathrm{g}]=\) hammgen(...) is the same as \(\mathrm{h}=\) hammgen(...) except that it also produces the k -by-n generator matrix g that corresponds to the parity-check matrix \(h\). \(k\), the message length, equals \(n-m\), or 2^m-1-m.
[h,g,n,k] = hammgen(...) is the same as [h,g] = hammgen(...) except that it also returns the codeword length \(n\) and the message length k .

Note If your value of \(m\) is less than 25 and if your primitive polynomial is the default primitive polynomial for \(\mathrm{GF}\left(2^{\wedge} \mathrm{m}\right)\), the syntax hammgen ( m ) is likely to be faster than the syntax hammgen ( \(\mathrm{m}, \mathrm{pol}\) ).

\section*{hammgen}

Examples The command below exhibits the parity-check and generator matrices for a Hamming code with codeword length \(7=2^{3}-1\) and message length \(4=7-3\).


The command below, which uses \(1+x^{2}+x^{3}\) as the primitive polynomial for \(\operatorname{GF}\left(2^{3}\right)\), shows that the parity-check matrix depends on the choice of primitive polynomial. Notice that h1 below is different from hin the example above.
h1 \(=\) hammgen \(\left(3,\left[\begin{array}{llll}1 & 0 & 1 & 1\end{array}\right]\right)\)
\begin{tabular}{lllllll}
1 & 0 & 0 & 1 & 1 & 1 & 0 \\
0 & 1 & 0 & 0 & 1 & 1 & 1 \\
0 & 0 & 1 & 1 & 1 & 0 & 1
\end{tabular}

\section*{Algorithms \\ See Also \\ How To computation result. \\ encode | decode | gen2par \\ - "Block Codes"}

Unlike gftuple, which processes one m-tuple at a time, hammgen generates the entire sequence from 0 to \(2^{\wedge} m-1\). The computation algorithm uses all previously computed values to produce the

\section*{hank2sys}

\section*{Purpose Convert Hankel matrix to linear system model}
```

Syntax
[num,den] = hank2sys(h,ini,tol)
[num,den,sv] = hank2sys(h,ini,tol)
[a,b,c,d] = hank2sys(h,ini,tol)
[a,b,c,d,sv] = hank2sys(h,ini,tol)

```

\section*{Description}

\section*{Examples}
[num, den] = hank2sys(h,ini,tol) converts a Hankel matrix h to a linear system transfer function with numerator num and denominator den. The vectors num and den list the coefficients of their respective polynomials in ascending order of powers of \(\mathrm{z}^{-1}\). The argument ini is the system impulse at time zero. If tol \(>1\), tol is the order of the conversion. If tol \(<1\), tol is the tolerance in selecting the conversion order based on the singular values. If you omit tol, its default value is 0.01 . This conversion uses the singular value decomposition method.
[num,den,sv] = hank2sys(h,ini,tol) returns a vector sv that lists the singular values of \(h\).
[a,b,c,d] = hank2sys(h,ini,tol) converts a Hankel matrix h to a corresponding linear system state-space model. a, b, c, and d are matrices. The input parameters are the same as in the first syntax above.
[a,b,c,d,sv] = hank2sys(h,ini,tol) is the same as the syntax above, except that \(s v\) is a vector that lists the singular values of \(h\).
```

h = hankel([1 0 1]);
[num,den,sv] = hank2sys(h,0,.01)

```

The output is
num =
\begin{tabular}{llll} 
& 0 & 1.0000 & 0.0000 \\
den \(=\) & & & \\
\end{tabular}
```

    1.0000 0.0000 0.0000 0.0000
    sv =
1.6180
1.0000
0.6180

```

\section*{See Also}
rcosflt | hankel

Purpose
Syntax

Description

Restore ordering of symbols permuted using helintrlv
```

[deintrlved,state] = heldeintrlv(data,col,ngrp,stp)
[deintrlved,state] = heldeintrlv(data,col,ngrp,stp,
init_state)
deintrlved = heldeintrlv(data,col,ngrp,stp,init_state)

```
[deintrlved,state] = heldeintrlv(data,col, ngrp,stp) restores the ordering of symbols in data by placing them in an array row by row and then selecting groups in a helical fashion to place in the output, deintrlved. data must have col*ngrp elements. If data is a matrix with multiple rows and columns, it must have col*ngrp rows, and the function processes the columns independently. state is a structure that holds the final state of the array. state. value stores input symbols that remain in the col columns of the array and do not appear in the output.
The function uses the array internally for its computations. The array has unlimited rows indexed by \(1,2,3, \ldots\), and col columns. The function initializes the top of the array with zeros. It then places col*ngrp symbols from the input into the next ngrp rows of the array. The function places symbols from the array in the output, intrlved, placing ngrp symbols at a time; the kth group of ngrp symbols comes from the kth column of the array, starting from row \(1+(\mathrm{k}-1)^{*}\) stp. Some output symbols are default values of 0 rather than input symbols; similarly, some input symbols are left in the array and do not appear in the output.
```

[deintrlved,state] =
heldeintrlv(data,col,ngrp,stp,init_state) initializes the array
with the symbols contained in init_state.value instead of zeros. The
structure init_state is typically the state output from a previous call
to this same function, and is unrelated to the corresponding
interleaver. In this syntax, some output symbols are default
values of 0, some are input symbols from data, and some are
initialization values from init_state.value.
deintrlved = heldeintrlv(data,col,ngrp,stp,init_state) is
the same as the syntax above, except that it does not record the

```
deinterleaver's final state. This syntax is appropriate for the last in a series of calls to this function. However, if you plan to call this function again to continue the deinterleaving process, the syntax above is more appropriate.

\section*{Using an Interleaver-Deinterleaver Pair}

To use this function as an inverse of the helintrlv function, use the same col, ngrp, and stp inputs in both functions. In that case, the two functions are inverses in the sense that applying helintrlv followed by heldeintrlv leaves data unchanged, after you take their combined delay of col*ngrp*ceil(stp*(col-1)/ngrp) into account. To learn more about delays of convolutional interleavers, see "Delays of Convolutional Interleavers".

Note Because the delay is an integer multiple of the number of symbols in data, you must use heldeintrlv at least twice (possibly more times, depending on the actual delay value) before the function returns results that represent more than just the delay.

\section*{Examples}

Recover interleaved data, taking into account the delay of the interleaver-deinterleaver pair.
```

col = 4; ngrp = 3; stp = 2; % Helical interleaver parameters
% Compute the delay of interleaver-deinterleaver pair.
delayval = col * ngrp * ceil(stp * (col-1)/ngrp);
len = col*ngrp; % Process this many symbols at one time.
data = randi([0 9],len,1); % Random symbols
data_padded = [data; zeros(delayval,1)]; % Pad with zeros.
% Interleave zero-padded data.
[i1,istate] = helintrlv(data_padded(1:len),col,ngrp,stp);
[i2,istate] = helintrlv(data_padded(len+1:2*len),col,ngrp, ...
stp,istate);
i3 = helintrlv(data_padded(2*len+1:end),col,ngrp,stp,istate);

```

\section*{heldeintrlv}
```

% Deinterleave.
[d1,dstate] = heldeintrlv(i1,col,ngrp,stp);
[d2,dstate] = heldeintrlv(i2,col,ngrp,stp,dstate);
d3 = heldeintrlv(i3,col,ngrp,stp,dstate);
% Check the results.
dO = [d1; d2; d3]; % All the deinterleaved data
dO_trunc = dO(delayval+1:end); % Remove the delay.
ser = symerr(data,d0_trunc)

```

The output below shows that no symbol errors occurred.
```

ser =

```
    0
See Also ..... helintrlv
How To - "Interleaving"

Permute symbols using helical array
```

intrlved = helintrlv(data,col,ngrp,stp)
[intrlved,state] = helintrlv(data,col,ngrp,stp)
[intrlved,state] = helintrlv(data,col,ngrp,stp,init_state)

```
intrlved \(=\) helintrlv(data, col, ngrp, stp) permutes the symbols in data by placing them in an unlimited-row array in helical fashion and then placing rows of the array in the output, intrlved. data must have col*ngrp elements. If data is a matrix with multiple rows and columns, it must have col*ngrp rows, and the function processes the columns independently.

The function uses the array internally for its computations. The array has unlimited rows indexed by \(1,2,3, \ldots\), and col columns. The function partitions col*ngrp symbols from the input into consecutive groups of ngrp symbols. The function places the kth group in the array along column k , starting from row \(1+(\mathrm{k}-1)^{*}\) stp. Positions in the array that do not contain input symbols have default values of 0 . The function places col*ngrp symbols from the array in the output, intrlved, by reading the first ngrp rows sequentially. Some output symbols are default values of 0 rather than input symbols; similarly, some input symbols are left in the array and do not appear in the output.
[intrlved,state] = helintrlv(data, col, ngrp, stp) returns a structure that holds the final state of the array. state. value stores input symbols that remain in the col columns of the array and do not appear in the output.
[intrlved,state] = helintrlv(data,col,ngrp,stp,init_state) initializes the array with the symbols contained in init_state.value. The structure init_state is typically the state output from a previous call to this same function, and is unrelated to the corresponding deinterleaver. In this syntax, some output symbols are default values of 0 , some are input symbols from data, and some are initialization values from init_state.value.

Examples The example below rearranges the integers from 1 to 24.
```

% Interleave some symbols. Record final state of array.
[i1,state] = helintrlv([1:12]',3,4,1);
% Interleave more symbols, remembering the symbols that
% were left in the array from the earlier command.
i2 = helintrlv([13:24]',3,4,1,state);
disp('Interleaved data:')
disp([i1,i2]')
disp('Values left in array after first interleaving operation:')
state.value{:}

```

During the successive calls to helintrlv, it internally creates the three-column arrays
\begin{tabular}{rrr}
{\([1\)} & 0 & \(0 ;\) \\
2 & 5 & \(0 ;\) \\
3 & 6 & \(9 ;\) \\
4 & 7 & \(10 ;\) \\
0 & 8 & \(11 ;\) \\
0 & 0 & \(12]\)
\end{tabular}
and
\(\begin{array}{lll}{[13} & 8 & 11 ;\end{array}\)
1417 12;
1518 21;
\(161922 ;\)
020 23;
0 0 24]
In the second array shown above, the 8,11 , and 12 are values left in the array from the previous call to the function. Specifying the init_state input in the second call to the function causes it to use those values rather than the default values of 0 .

The output from this example is below. (The actual interleaved data is a tall matrix, but it has been transposed into a wide matrix for display purposes.) The interleaved data comes from the top four rows of the three-column arrays shown above. Notice that some of the symbols in the first half of the interleaved data are default values of 0 , some of the symbols in the second half of the interleaved data were left in the array from the first call to helintrlv, and some of the input symbols (20, 23, and 24) do not appear in the interleaved data at all.
```

Interleaved data:
Columns 1 through 10

```
\begin{tabular}{|c|c|c|c|c|c|c|c|c|c|}
\hline 1 & 0 & 0 & 2 & 5 & 0 & 3 & 6 & 9 & 4 \\
\hline 13 & 8 & 11 & 14 & 17 & 12 & 15 & 18 & 21 & 16 \\
\hline \multicolumn{10}{|l|}{Columns 11 through 12} \\
\hline 7 & 10 & & & & & & & & \\
\hline 19 & 22 & & & & & & & & \\
\hline \multicolumn{10}{|l|}{Values left in array after first interleaving operation:} \\
\hline \multicolumn{10}{|l|}{ans =} \\
\hline \multicolumn{10}{|l|}{[]} \\
\hline \multicolumn{10}{|l|}{ans =} \\
\hline \multicolumn{10}{|l|}{8} \\
\hline \multicolumn{10}{|l|}{ans =} \\
\hline \multicolumn{10}{|c|}{1112} \\
\hline
\end{tabular}

\section*{helintrlv}

The example on the reference page for heldeintrlv also uses this function.

\author{
See Also \\ heldeintrlv \\ How To \\ - "Interleaving"
}

\section*{Purpose}

Syntax
Description

\section*{Examples}

Restore ordering of symbols in helical pattern
```

deintrlvd = helscandeintrlv(data,Nrows,Ncols,hstep)

```
deintrlvd = helscandeintrlv(data, Nrows,Ncols,hstep) rearranges the elements in data by filling a temporary matrix with the elements in a helical fashion and then sending the matrix contents to the output row by row. Nrows and Ncols are the dimensions of the temporary matrix. hstep is the slope of the diagonal, that is, the amount by which the row index increases as the column index increases by one. hstep must be a nonnegative integer less than Nrows.

Helical fashion means that the function places input elements along diagonals of the temporary matrix. The number of elements in each diagonal is exactly Ncols, after the function wraps past the edges of the matrix when necessary. The function traverses diagonals so that the row index and column index both increase. Each diagonal after the first one begins one row below the first element of the previous diagonal.

If data is a vector, it must have Nrows*Ncols elements. If data is a matrix with multiple rows and columns, data must have Nrows*Ncols rows and the function processes the columns independently.

To use this function as an inverse of the helscanintrlv function, use the same Nrows, Ncols, and hstep inputs in both functions. In that case, the two functions are inverses in the sense that applying helscanintrlv followed by helscandeintrlv leaves data unchanged.

The command below rearranges a vector using a 3-by-4 temporary matrix and diagonals of slope 1 .
```

d = helscandeintrlv(1:12,3,4,1)
d =

```
    Columns 1 through 10
\begin{tabular}{llllllllll}
1 & 10 & 7 & 4 & 5 & 2 & 11 & 8 & 9 & 6
\end{tabular}

\section*{helscandeintrlv}
Columns 11 through 1312

Internally, the function creates the 3-by- 4 temporary matrix
\(\left.\begin{array}{rrrr}{\left[\begin{array}{rrr}1 & 10 & 7 \\ 5 & 4 ; \\ 5 & 2 & 11\end{array}\right.} & 8 ; \\ 9 & 6 & 3 & 12\end{array}\right]\)
using length-four diagonals. The function then sends the elements, row by row, to the output d .

\section*{See Also helscanintrlv}

How To . "Interleaving"

\section*{Purpose}

Reorder symbols in helical pattern

\section*{Syntax}

Description

Examples
intrlvd = helscanintrlv(data, Nrows, Ncols,hstep) nonnegative integer less than Nrows.
intrlvd = helscanintrlv(data, Nrows, Ncols, hstep) rearranges the elements in data by filling a temporary matrix with the elements row by row and then sending the matrix contents to the output in a helical fashion. Nrows and Ncols are the dimensions of the temporary matrix. hstep is the slope of the diagonal, that is, the amount by which the row index increases as the column index increases by one. hstep must be a

Helical fashion means that the function selects elements along diagonals of the temporary matrix. The number of elements in each diagonal is exactly Ncols, after the function wraps past the edges of the matrix when necessary. The function traverses diagonals so that the row index and column index both increase. Each diagonal after the first one begins one row below the first element of the previous diagonal.

If data is a vector, it must have Nrows*Ncols elements. If data is a matrix with multiple rows and columns, data must have Nrows*Ncols rows and the function processes the columns independently.

The command below rearranges a vector using diagonals of two different slopes.
```

i1 = helscanintrlv(1:12,3,4,1) % Slope of diagonal is 1.
i2 = helscanintrlv(1:12,3,4,2) % Slope of diagonal is 2.

```

The output is below.
i1 =
Columns 1 through 10
\(\begin{array}{llllllllll}1 & 6 & 11 & 4 & 5 & 10 & 3 & 8 & 9 & 2\end{array}\)
Columns 11 through 12

\section*{helscanintrlv}
```

    7 12
    i2 =
Columns 1 through 10
1 10[10
Columns 11 through 12
3 12

```

In each case, the function internally creates the temporary 3-by-4 matrix
\(\left[\begin{array}{rrr}1 & 2 & 3 \\ 5 & 4 ; \\ 5 & 6 & 8 ; \\ 9 & 10 & 11 \\ 12\end{array}\right]\)

To form i1, the function forms each slope-one diagonal by moving one row down and one column to the right. The first diagonal contains 1 , 6,11 , and 4 , while the second diagonal starts with 5 because that is beneath 1 in the temporary matrix.

To form i2, the function forms each slope-two diagonal by moving two rows down and one column to the right. The first diagonal contains 1 , 10,7 , and 4 , while the second diagonal starts with 5 because that is beneath 1 in the temporary matrix.

\section*{See Also}
helscandeintrlv
How To
- "Interleaving"

\section*{Purpose}

Design Hilbert transform IIR filter
Syntax
```

hilbiir
hilbiir(ts)
hilbiir(ts,dly)
hilbiir(ts,dly,bandwidth)
hilbiir(ts,dly,bandwidth,tol)
[num,den] = hilbiir(...)
[num,den,sv] = hilbiir(...)
[a,b,c,d] = hilbiir(...)
[a,b,c,d,sv] = hilbiir(...)

```

\section*{Description}

The function hilbiir designs a Hilbert transform filter. The output is either
- A plot of the filter's impulse response, or
- A quantitative characterization of the filter, using either a transfer function model or a state-space model

\section*{Background Information}

An ideal Hilbert transform filter has the transfer function \(H(s)=-j \operatorname{sgn}(s)\), where \(\operatorname{sgn}(\).\() is the signum function (sign in\) MATLAB). The impulse response of the Hilbert transform filter is
\[
h(t)=\frac{1}{\pi t}
\]

Because the Hilbert transform filter is a noncausal filter, the hilbiir function introduces a group delay, dly. A Hilbert transform filter with this delay has the impulse response
\[
h(t)=\frac{1}{\pi(t-\mathrm{dly})}
\]

\section*{Choosing a Group Delay Parameter}

The filter design is an approximation. If you provide the filter's group delay as an input argument, these two suggestions can help improve the accuracy of the results:
- Choose the sample time ts and the filter's group delay dly so that dly is at least a few times larger than ts and rem(dly,ts) \(=\mathrm{ts} / 2\). For example, you can set ts to \(2 * d l y / N\), where \(N\) is a positive integer.
- At the point \(t=d l y\), the impulse response of the Hilbert transform filter can be interpreted as 0 , - Inf, or Inf. If hilbiir encounters this point, it sets the impulse response there to zero. To improve accuracy, avoid the point \(t=d l y\).

\section*{Syntaxes for Plots}

Each of these syntaxes produces a plot of the impulse response of the filter that the hilbiir function designs, as well as the impulse response of a corresponding ideal Hilbert transform filter.
hilbiir plots the impulse response of a fourth-order digital Hilbert transform filter with a one-second group delay. The sample time is \(2 / 7\) seconds. In this particular design, the tolerance index is 0.05 . The plot also displays the impulse response of the ideal Hilbert transform filter with a one-second group delay.
hilbiir(ts) plots the impulse response of a fourth-order Hilbert transform filter with a sample time of ts seconds and a group delay of \(\mathrm{ts} * 7 / 2\) seconds. The tolerance index is 0.05 . The plot also displays the impulse response of the ideal Hilbert transform filter having a sample time of ts seconds and a group delay of \(\mathrm{ts} * 7 / 2\) seconds.
hilbiir(ts, dly) is the same as the syntax above, except that the filter's group delay is dly for both the ideal filter and the filter that hilbiir designs. See "Choosing a Group Delay Parameter" on page 1-344 above for guidelines on choosing dly.
hilbiir(ts, dly, bandwidth) is the same as the syntax above, except that bandwidth specifies the assumed bandwidth of the input signal and that the filter design might use a compensator for the input signal.

If bandwidth \(=0\) or bandwidth \(>1 /\left(2^{*}\right.\) ts \()\), hilbiir does not use a compensator.
hilbiir(ts, dly, bandwidth, tol) is the same as the syntax above, except that tol is the tolerance index. If tol \(<1\), the order of the filter is determined by
\[
\frac{\text { truncated-singular-value }}{\text { maximum-singular-value }}<\text { tol }
\]

If tol \(>1\), the order of the filter is tol.

\section*{Syntaxes for Transfer Function and State-Space Quantities}

Each of these syntaxes produces quantitative information about the filter that hilbiir designs, but does not produce a plot. The input arguments for these syntaxes (if you provide any) are the same as those described in "Syntaxes for Plots" on page 1-344.
[num, den] = hilbiir(...) outputs the numerator and denominator of the IIR filter's transfer function.
[num, den,sv] = hilbiir(...) outputs the numerator and denominator of the IIR filter's transfer function, and the singular values of the Hankel matrix that hilbiir uses in the computation.
[a,b, c, d] = hilbiir(...) outputs the discrete-time state-space model of the designed Hilbert transform filter. a, b, c, and d are matrices.
\([a, b, c, d, s v]=\) hilbiir (...) outputs the discrete-time state-space model of the designed Hilbert transform filter, and the singular values of the Hankel matrix that hilbiir uses in the computation.

\section*{Algorithms}

The hilbiir function calculates the impulse response of the ideal Hilbert transform filter response with a group delay. It fits the response curve using a singular-value decomposition method. See the book by Kailath [1].

\section*{hilbiir}
\begin{tabular}{ll} 
Examples & \begin{tabular}{l} 
For an example using the function's default values, type one of the \\
following commands at the MATLAB prompt. \\
hilbiir \\
[num, den] = hilbiir
\end{tabular} \\
References & \begin{tabular}{l} 
[1] Kailath, Thomas, Linear Systems, Englewood Cliffs, NJ, \\
Prentice-Hall, 1980.
\end{tabular} \\
See Also & \begin{tabular}{l} 
grpdelay | rcosiir
\end{tabular} \\
How To & - "Filtering"
\end{tabular}

\section*{Purpose}

Huffman decoder
Syntax
Description

\section*{Examples}
dsig = huffmandeco(comp,dict)
dsig = huffmandeco(comp, dict) decodes the numeric Huffman code vector comp using the code dictionary dict. The argument dict is an N -by- 2 cell array, where N is the number of distinct possible symbols in the original signal that was encoded as comp. The first column of dict represents the distinct symbols and the second column represents the corresponding codewords. Each codeword is represented as a numeric row vector, and no codeword in dict is allowed to be the prefix of any other codeword in dict. You can generate dict using the huffmandict function and comp using the huffmanenco function. If all signal values in dict are numeric, dsig is a vector; if any signal value in dict is alphabetical, dsig is a one-dimensional cell array.

The example below encodes and then decodes a vector of random data that has a prescribed probability distribution.
```

symbols = [1:6]; % Distinct symbols that data source can produce
p = [.5 .125 .125 . 125 .0625 .0625]; % Probability distribution
[dict,avglen] = huffmandict(symbols,p); % Create dictionary.
actualsig = randsrc(1,100,[symbols; p]); % Create data using p.
comp = huffmanenco(actualsig,dict); % Encode the data.
dsig = huffmandeco(comp,dict); % Decode the Huffman code.
isequal(actualsig,dsig) % Check whether the decoding is correct.

```

The output below indicates that the decoder successfully recovered the data in actualsig.
ans \(=\)
1

\section*{References}
[1] Sayood, Khalid, Introduction to Data Compression, San Francisco, Morgan Kaufmann, 2000.

\section*{huffmandeco}

See Also huffmandict \| huffmanenco
How To . "Huffman Coding"

\section*{Purpose}

\section*{Syntax}

\section*{Description}

Generate Huffman code dictionary for source with known probability model
```

[dict,avglen] = huffmandict(symbols,p)
[dict,avglen] = huffmandict(symbols,p,N)
[dict,avglen] = huffmandict(symbols,p,N,variance)

```

\section*{For All Syntaxes}

The huffmandict function generates a Huffman code dictionary corresponding to a source with a known probability model. The required inputs are
- symbols, which lists the distinct signal values that the source produces. It can have the form of a numeric vector, numeric cell array, or alphanumeric cell array. If it is a cell array, it must be either a row or a column.
- \(p\), a probability vector whose kth element is the probability with which the source produces the kth element of symbols. The length of \(p\) must equal the length of symbols.
The outputs of huffmandict are
- dict, a two-column cell array in which the first column lists the distinct signal values from symbols and the second column lists the corresponding Huffman codewords. In the second column, each Huffman codeword is represented as a numeric row vector.
- avglen, the average length among all codewords in the dictionary, weighted according to the probabilities in the vector \(p\).

\section*{For Specific Syntaxes}
[dict,avglen] = huffmandict(symbols, \(p\) ) generates a binary Huffman code dictionary using the maximum variance algorithm.
[dict,avglen] = huffmandict(symbols, p,N) generates an N -ary Huffman code dictionary using the maximum variance algorithm. N is an integer between 2 and 10 that must not exceed the number of source symbols whose probabilities appear in the vector \(p\).

\section*{huffmandict}
[dict,avglen] = huffmandict(symbols, \(p, N\), variance) generates an N -ary Huffman code dictionary with the minimum variance if variance is 'min' and the maximum variance if variance is 'max'. \(N\) is an integer between 2 and 10 that must not exceed the length of the vector \(p\).

\section*{Examples}
avglen =
2.2000
samplecode =

10

\section*{References \\ See Also \\ huffmanenco | huffmandeco}

How To • "Huffman Coding"
\begin{tabular}{ll} 
Purpose & Huffman encoder \\
Syntax & comp = huffmanenco(sig, dict) \\
Description & \begin{tabular}{l} 
comp \(=\) huffmanenco(sig, dict) encodes the signal sig using the \\
Huffman codes described by the code dictionary dict. The argument \\
sig can have the form of a numeric vector, numeric cell array, or \\
alphanumeric cell array. If sig is a cell array, it must be either a row or \\
a column. dict is an N-by-2 cell array, where N is the number of distinct \\
possible symbols to be encoded. The first column of dict represents the \\
distinct symbols and the second column represents the corresponding \\
codewords. Each codeword is represented as a numeric row vector, and
\end{tabular} \\
no codeword in dict can be the prefix of any other codeword in dict. \\
You can generate dict using the huffmandict function.
\end{tabular}
Purpose Inverse discrete Fourier transform
Syntax ..... ifft(x)
Description ifft \((x)\) is the inverse discrete Fourier transform (DFT) of the Galoisvector \(x\). If \(x\) is in the Galois field \(\operatorname{GF}\left(2^{m}\right)\), the length of \(x\) must be \(2^{m}-1\).
Examples For an example using ifft, see the reference page for fft .
Limitations The Galois field over which this function works must have 256 or fewerelements. In other words, \(x\) must be in the Galois field \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\), where\(m\) is an integer between 1 and 8 .
Algorithms If \(x\) is a column vector, ifft applies dftmtx to the multiplicative inverse of the primitive element of the Galois field and multiplies the resulting matrix by \(x\).
See Also ..... fft | dftmtx
How To - "Signal Processing Operations in Galois Fields"

Purpose Integrate and dump
Syntax \(\quad y=\operatorname{intdump}(x\), nsamp \()\)
Description \(y=\) intdump ( \(x\), nsamp) integrates the signal \(x\) for one symbol period, then outputs the averaged one value into Y . nsamp is the number of samples per symbol. For two-dimensional signals, the function treats each column as one channel.

Examples An example in "Combine Pulse Shaping and Filtering with Modulation" uses this function in conjunction with modulation.

Processes two independent channels, each of which contain three symbols of data made up of four samples.
```

s = rng;
rng(68521);
nsamp = 4; % Number of samples per symbol
ch1 = randi([0 1],3*nsamp,1); % Random binary channel
ch2 = rectpulse([1 2 3]',nsamp); % Rectangular pulses
x = [ch1 ch2]; % Two-channel signal
y = intdump(x,nsamp)
rng(s);

```

The output is below. Each column corresponds to one channel, and each row corresponds to one symbol.
```

y =
0.5000 1.0000
0.5000 2.0000
1.0000 3.0000

```
See Also ..... rectpulse

\section*{Purpose}

Reorder sequence of symbols

\section*{Syntax}

Description
```

intrlvd = intrlv(data,elements)

```
intrlvd = intrlv(data, elements) rearranges the elements of data without repeating or omitting any elements. If data is a length-N vector or an N-row matrix, elements is a length-N vector that permutes the integers from 1 to N . The sequence in elements is the sequence in which elements from data or its columns appear in intrlvd. If data is a matrix with multiple rows and columns, the function processes the columns independently.

\section*{Examples}

The command below rearranges the elements of a vector. Your output might differ because the permutation vector is random in this example.
```

p = randperm(10); % Permutation vector
a = intrlv(10:10:100,p)

```

The output is below.
\(\mathrm{a}=\)
\(\begin{array}{llllllllll}10 & 90 & 60 & 30 & 50 & 80 & 100 & 20 & 70 & 40\end{array}\)
The command below rearranges each of two columns of a matrix.
\(\mathrm{b}=\operatorname{intrlv}\left(\left[\begin{array}{llllllllll} & .2 & .3 & .4 & .5 ; & .2 & .4 & .6 & .8 & 1\end{array}\right]^{\prime},\left[\begin{array}{lllll}2 & 4 & 3 & 5 & 1\end{array}\right]\right)\)
\(\mathrm{b}=\)
\begin{tabular}{ll}
0.2000 & 0.4000 \\
0.4000 & 0.8000 \\
0.3000 & 0.6000 \\
0.5000 & 1.0000 \\
0.1000 & 0.2000
\end{tabular}

\section*{See Also}

\section*{intrlv}

> How To . "Interleaving"
Purpose True for trellis corresponding to catastrophic convolutional code
Syntax iscatastrophic(s)
Description iscatastrophic(s) returns true if the trellis s corresponds toa convolutional code that causes catastrophic error propagation.Otherwise, it returns false.
See Also convenc | istrellis | poly2trellis | struct
How To - "Convolutional Codes"

Purpose True for primitive polynomial for Galois field

\section*{Syntax isprimitive(a)}

Description isprimitive(a) returns 1 if the polynomial that a represents is primitive for the Galois field \(\mathrm{GF}\left(2^{\mathrm{m}}\right)\), and 0 otherwise. The input a can represent the polynomial using one of these formats:
- A nonnegative integer less than \(2^{17}\). The binary representation of this integer indicates the coefficients of the polynomial. In this case, \(m\) is floor (log2(a)).
- A Galois row vector in GF(2), listing the coefficients of the polynomial in order of descending powers. In this case, \(m\) is the order of the polynomial represented by a.

\section*{Examples}

The example below finds all primitive polynomials for \(\mathrm{GF}(8)\) and then checks using isprimitive whether specific polynomials are primitive.
```

a = primpoly(3,'all','nodisplay'); % All primitive polys for GF(8)
isp1 = isprimitive(13) % 13 represents a primitive polynomial.
isp2 = isprimitive(14) % 14 represents a nonprimitive polynomial.

```

The output is below. If you examine the vector a, notice that isp1 is true because 13 is an element in a, while isp2 is false because 14 is not an element in a.
```

isp1 =

```

1
isp2 =
0
\begin{tabular}{ll} 
See Also & primpoly \\
How To & - "Galois Field Computations"
\end{tabular}

\section*{istrellis}

Purpose True for valid trellis structure
```

Syntax
[isok,status] = istrellis(s)

```

Description
[isok, status] = istrellis(s) checks if the input s is a valid trellis structure. If the input is a valid trellis structure, isok is 1 and status is an empty string. Otherwise, isok is 0 and status is a string that indicates why s is not a valid trellis structure.

A valid trellis structure is a MATLAB structure whose fields are as in the table below.

Fields of a Valid Trellis Structure for a Rate \(\mathbf{k} / \mathbf{n}\) Code
\begin{tabular}{l|l|l}
\hline \begin{tabular}{l} 
Field in Trellis \\
Structure
\end{tabular} & Dimensions & Meaning \\
\hline numInputSymbols & Scalar & \begin{tabular}{l} 
Number of input \\
symbols to the \\
encoder: \(2^{\mathrm{k}}\)
\end{tabular} \\
\hline numOutputSymbols & Scalar & \begin{tabular}{l} 
Number of output \\
symbols from the \\
encoder: \(2^{\mathrm{n}}\)
\end{tabular} \\
\hline numStates & Scalar & \begin{tabular}{l} 
Number of states in \\
the encoder
\end{tabular} \\
\hline nextStates & \begin{tabular}{l} 
numStates-by- \(2^{\mathrm{k}}\) \\
matrix
\end{tabular} & \begin{tabular}{l} 
Next states for all \\
combinations of \\
current state and \\
current input
\end{tabular} \\
\hline outputs & \begin{tabular}{l} 
numStates-by- \(2^{\mathrm{k}}\) \\
matrix
\end{tabular} & \begin{tabular}{l} 
Outputs (in octal) \\
for all combinations \\
of current state and \\
current input
\end{tabular} \\
\hline
\end{tabular}

In the nextStates matrix, each entry is an integer between 0 and numStates-1. The element in the sth row and uth column denotes the next state when the starting state is s-1 and the input bits have decimal representation \(u-1\). To convert the input bits to a decimal value, use the first input bit as the most significant bit (MSB). For example, the second column of the nextStates matrix stores the next states when the current set of input values is \(\{0, \ldots, 0,1\}\).

To convert the state to a decimal value, use this rule: If k exceeds 1 , the shift register that receives the first input stream in the encoder provides the least significant bits in the state number, and the shift register that receives the last input stream in the encoder provides the most significant bits in the state number.
In the outputs matrix, the element in the sth row and uth column denotes the encoder's output when the starting state is s-1 and the input bits have decimal representation \(u-1\). To convert to decimal value, use the first output bit as the MSB.

These commands assemble the fields into a very simple trellis structure, and then verify the validity of the trellis structure.
```

trellis.numInputSymbols = 2;
trellis.numOutputSymbols = 2;
trellis.numStates = 2;
trellis.nextStates = [0 1;0 1];
trellis.outputs = [0 0;1 1];
[isok,status] = istrellis(trellis)

```

The output is below.
isok =
    1
status =

\section*{istrellis}

Another example of a trellis is in "Trellis Description of a Convolutional Code".

See Also poly2trellis | struct | convenc | vitdec
How To . "Convolutional Codes"

\section*{Purpose}

Toggles random number generation mode for channel objects
Syntax
b = legacychannelsim
legacychannelsim(true)
legacychannelsim(false)
oldmode = legacychannelsim(newmode)
Description
b = legacychannelsim returns FALSE if the code you are running uses the R2009b (or later) version of the random number generator for rayleighchan or ricianchan. (By default, these use the 2009b random number generator.) It returns TRUE if pre-R2009b versions are used. See Version 4.4. (R2009b) Communications System Toolbox Release Notes for more information.
legacychannelsim(true) reverts the random number generation mode for channel objects to pre-2009b version.

Note legacychannelsim(true) will support the reset (chan, randstate) functionality.
legacychannelsim(false) sets the random number generation mode for channel objects to 2009b and later versions.
oldmode = legacychannelsim(newmode) sets the random number generation mode for channel objects to NEWMODE and returns the previous mode, OLDMODE.

\section*{lineareq}

\section*{Purpose Construct linear equalizer object}
```

Syntax eqobj = lineareq(nweights,alg)
eqobj = lineareq(nweights,alg,sigconst)
eqobj = lineareq(nweights,alg,sigconst,nsamp)

```

\section*{Description}

The lineareq function creates an equalizer object that you can use with the equalize function to equalize a signal. To learn more about the process for equalizing a signal, see "Adaptive Algorithms".
eqobj = lineareq(nweights,alg) constructs a symbol-spaced linear equalizer object. The equalizer has nweights complex weights, which are initially all zeros. alg describes the adaptive algorithm that the equalizer uses; you should create alg using any of these functions: lms, signlms, normlms, varlms, rls, or cma. The signal constellation of the desired output is [-1 1] , which corresponds to binary phase shift keying (BPSK).
eqobj = lineareq(nweights,alg,sigconst) specifies the signal constellation vector of the desired output.
eqobj = lineareq(nweights,alg,sigconst,nsamp) constructs a fractionally spaced linear equalizer object. The equalizer has nweights complex weights spaced at \(T / n s a m p\), where \(T\) is the symbol period and nsamp is a positive integer. nsamp \(=1\) corresponds to a symbol-spaced equalizer.

\section*{Properties}

The table below describes the properties of the linear equalizer object. To learn how to view or change the values of a linear equalizer object, see "Accessing Properties of an Equalizer".

Tip To initialize or reset the equalizer object eqobj, enter reset (eqobj).
\begin{tabular}{l|l}
\hline Property & Description \\
\hline EqType & \begin{tabular}{l} 
Fixed value, 'Linear \\
Equalizer'
\end{tabular} \\
\hline AlgType & \begin{tabular}{l} 
Name of the adaptive algorithm \\
represented by alg
\end{tabular} \\
\hline nWeights & Number of weights \\
\hline nSampPerSym & \begin{tabular}{l} 
Number of input samples per \\
symbol (equivalent to nsamp input \\
argument). This value relates \\
to both the equalizer structure \\
(see the use of K in "Fractionally \\
Spaced Equalizers") and an \\
assumption about the signal to be \\
equalized.
\end{tabular} \\
\hline RefTap (except for CMA & \begin{tabular}{l} 
Reference tap index, between 1 \\
and nWeights. Setting this to a \\
value greater than 1 effectively \\
delays the reference signal and \\
the output signal by RefTap-1 \\
with respect to the equalizer's \\
input signal.
\end{tabular} \\
\hline SigConst & \begin{tabular}{l} 
Signal constellation, a vector \\
whose length is typically a power \\
of 2
\end{tabular} \\
\hline Weights & \begin{tabular}{l} 
Vector of complex coefficients. \\
This is the set of wi values in \\
the schematic in "Symbol-Spaced \\
Equalizers".
\end{tabular} \\
\hline WeightInputs & \begin{tabular}{l} 
Vector of tap weight inputs. This \\
is the set of u \(i_{i}\) values in the \\
schematic in "Symbol-Spaced \\
Equalizers".
\end{tabular} \\
\hline
\end{tabular}

\section*{lineareq}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline ResetBeforeFiltering & \begin{tabular}{l} 
If 1, each call to equalize \\
resets the state of eqobj before \\
equalizing. If 0, the equalization \\
process maintains continuity \\
from one call to the next.
\end{tabular} \\
\hline NumSamplesProcessed & \begin{tabular}{l} 
Number of samples the equalizer \\
processed since the last reset. \\
When you create or reset eqobj, \\
this property value is 0.
\end{tabular} \\
\hline \begin{tabular}{l} 
Properties specific to the adaptive \\
algorithm represented by alg
\end{tabular} & \begin{tabular}{l} 
See reference page for the \\
adaptive algorithm function \\
that created alg: lms, signlms, \\
normlms, varlms, rls, or cma.
\end{tabular} \\
\hline
\end{tabular}

\section*{Relationships Among Properties}

If you change nWeights, MATLAB maintains consistency in the equalizer object by adjusting the values of the properties listed below.
\begin{tabular}{l|l}
\hline Property & Adjusted Value \\
\hline Weights & zeros(1, nWeights) \\
\hline WeightInputs & zeros(1, nWeights) \\
\hline \begin{tabular}{l} 
StepSize \\
(Variable-step-size LMS \\
equalizers)
\end{tabular} & InitStep*ones(1, nWeights) \\
\hline \begin{tabular}{l} 
InvCorrMatrix (RLS \\
equalizers)
\end{tabular} & InvCorrInit*eye(nWeights) \\
\hline
\end{tabular}

An example illustrating relationships among properties is in "Linked Properties of an Equalizer Object".
\begin{tabular}{ll} 
Examples & \begin{tabular}{l} 
For examples that use this function, see "Equalize Using a Training \\
Sequence in MATLAB", "Example: Equalizing Multiple Times, Vary \\
the Mode", and "Example: Adaptive Equalization Within a Loop".
\end{tabular} \\
See Also & lms | signlms | normlms | varlms | rls | cma | dfe | equalize \\
How To & - "Equalization"
\end{tabular}

\section*{lloyds}

\section*{Purpose}

Optimize quantization parameters using Lloyd algorithm
```

Syntax
[partition, codebook] = lloyds(training_set,initcodebook)
[partition, codebook] = lloyds(training_set,len)
[partition, codebook] = lloyds(training_set,...,tol)
[partition, codebook,distor] = lloyds(...)
[partition,codebook,distor,reldistor] = lloyds(...)

```

\section*{Description}
[partition,codebook] = lloyds(training_set,initcodebook) optimizes the scalar quantization parameters partition and codebook for the training data in the vector training_set. initcodebook, a vector of length at least 2 , is the initial guess of the codebook values. The output codebook is a vector of the same length as initcodebook. The output partition is a vector whose length is one less than the length of codebook.

See "Represent Partitions", "Represent Codebooks", or the reference page for quantiz in this chapter, for a description of the formats of partition and codebook.

Note lloyds optimizes for the data in training_set. For best results, training_set should be similar to the data that you plan to quantize.
[partition, codebook] = lloyds(training_set,len) is the same as the first syntax, except that the scalar argument len indicates the size of the vector codebook. This syntax does not include an initial codebook guess.
[partition, codebook] = lloyds(training_set,...,tol) is the same as the two syntaxes above, except that tol replaces \(10^{-7}\) in condition 1 of the algorithm description below.
[partition,codebook,distor] = lloyds(...) returns the final mean square distortion in the variable distor.
[partition,codebook,distor,reldistor] = lloyds(...) returns a value reldistor that is related to the algorithm's termination. In
condition 1 of the algorithm below, reldistor is the relative change in distortion between the last two iterations. In condition 2, reldistor is the same as distor.

\section*{Examples}

The code below optimizes the quantization parameters for a sinusoidal transmission via a three-bit channel. Because the typical data is sinusoidal, training_set is a sampled sine wave. Because the channel can transmit three bits at a time, lloyds prepares a codebook of length \(2^{3}\).
```

% Generate a complete period of a sinusoidal signal.
x = sin([0:1000]*pi/500);
[partition,codebook] = lloyds(x,2^3)

```

The output is below.
```

partition =

```
    Columns 1 through 6
        \(\begin{array}{llllll}-0.8540 & -0.5973 & -0.3017 & 0.0031 & 0.3077 & 0.6023\end{array}\)
    Column 7
        0.8572
codebook =
    Columns 1 through 6
        \(\begin{array}{llllll}-0.9504 & -0.7330 & -0.4519 & -0.1481 & 0.1558 & 0.4575\end{array}\)
        Columns 7 through 8
        \(0.7372 \quad 0.9515\)

Algorithms

References
lloyds uses an iterative process to try to minimize the mean square distortion. The optimization processing ends when either
- The relative change in distortion between iterations is less than \(10^{-7}\).
- The distortion is less than eps*max(training_set), where eps is the MATLAB floating-point relative accuracy.
[1] Lloyd, S.P., "Least Squares Quantization in PCM," IEEE Transactions on Information Theory, Vol. IT-28, March, 1982, pp. 129-137.
[2] Max, J., "Quantizing for Minimum Distortion," IRE Transactions on Information Theory, Vol. IT-6, March, 1960, pp. 7-12.
See Also quantiz | dpcmopt
How To . "Source Coding"

\section*{Purpose}

Construct least mean square (LMS) adaptive algorithm object

\section*{Syntax}

Description
```

alg = lms(stepsize)
alg = lms(stepsize,leakagefactor)

```

The lms function creates an adaptive algorithm object that you can use with the lineareq function or dfe function to create an equalizer object. You can then use the equalizer object with the equalize function to equalize a signal. To learn more about the process for equalizing a signal, see "Adaptive Algorithms".
alg = lms(stepsize) constructs an adaptive algorithm object based on the least mean square (LMS) algorithm with a step size of stepsize.
alg = lms(stepsize, leakagefactor) sets the leakage factor of the LMS algorithm. leakagefactor must be between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

\section*{Properties}

The table below describes the properties of the LMS adaptive algorithm object. To learn how to view or change the values of an adaptive algorithm object, see "Access Properties of an Adaptive Algorithm".
\begin{tabular}{l|l}
\hline Property & Description \\
\hline AlgType & Fixed value, 'LMS ' \\
\hline StepSize & \begin{tabular}{l} 
LMS step size parameter, a \\
nonnegative real number
\end{tabular} \\
\hline LeakageFactor & \begin{tabular}{l} 
LMS leakage factor, a real \\
number between 0 and 1
\end{tabular} \\
\hline
\end{tabular}

For examples that use this function, see "Equalize Using a Training Sequence in MATLAB", "Example: Equalizing Multiple Times, Varying the Mode", and "Example: Adaptive Equalization Within a Loop".
Algorithms
ReferencesHow To
Referring to the schematics presented in "Adaptive Algorithms", define w as the vector of all weights \(\mathrm{w}_{\mathrm{i}}\) and define \(u\) as the vector of all inputs \(u_{i}\). Based on the current set of weights, \(w\), this adaptive algorithm creates the new set of weights given by
(LeakageFactor) w + (StepSize) u*e
where the * operator denotes the complex conjugate.
[1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, John Wiley \& Sons, 1998.
[2] Haykin, Simon, Adaptive Filter Theory, Third Ed., Upper Saddle River, NJ, Prentice-Hall, 1996.
[3] Kurzweil, Jack, An Introduction to Digital Communications, New York, John Wiley \& Sons, 2000.
[4] Proakis, John G., Digital Communications, Fourth Ed., New York, McGraw-Hill, 2001.

\section*{See Also \\ See Also}
signlms | normlms | varlms | rls | cma | lineareq | dfe | equalize
- "Equalization"

\section*{Purpose Logarithm in Galois field}

\section*{Syntax \\ \(y=\log (x)\)}

Description \(\quad y=\log (x)\) computes the logarithm of each element in the Galois array \(x\). \(y\) is an integer array that solves the equation \(A .{ }^{\wedge} y=x\), where \(A\) is the primitive element used to represent elements in \(x\). More explicitly, the base \(A\) of the logarithm is \(g f(2, x . m)\) or \(g f\left(2, x . m, x . p r i m \_p o l y\right)\). All elements in \(x\) must be nonzero because the logarithm of zero is undefined.

\section*{Examples}

The code below illustrates how the logarithm operation inverts exponentiation.
```

m = 4; x = gf([8 1 6; 3 5 7; 4 9 2],m);
y = log(x);
primel = gf(2,m); % Primitive element in the field
z = primel .^ y; % This is now the same as x.
ck = isequal(x,z)

```

The output is
ck =

1

The code below shows that the logarithm of 1 is 0 and that the logarithm of the base (primel) is 1 .
\(\mathrm{m}=4 ; \mathrm{primel}=\mathrm{gf}(2, \mathrm{~m})\);
yy \(=\log ([1\), primel])
The output is
yy =
\(0 \quad 1\)

\section*{IteZadoffChuSeq}
\begin{tabular}{ll} 
Purpose & Generate root Zadoff-Chu sequence of complex symbols \\
Syntax & SEQ = lteZadoffChuSeq ( \(\mathrm{R}, \mathrm{N})\) \\
Description & \begin{tabular}{l} 
SEQ \(=1\) leZadoffChuSeq ( \(\mathrm{R}, \mathrm{N})\) generates the Rth root Zadoff-Chu \\
sequence with length \(N\), as defined in the LTE specifications [1]. The \\
output SEQ is an N-length column vector of complex symbols.
\end{tabular} \\
\begin{tabular}{l} 
The function generates the actual sequence using the following \\
algorithm:
\end{tabular}
\end{tabular}
\[
\operatorname{seq}(m+1)=\exp (-j \cdot \pi \cdot R \cdot m \cdot(m+1) / N), \quad \text { for } m=0, \ldots, N-1
\]

This function uses a negative polarity on the argument of the exponent or a clockwise sequence of phases.

R-Root of the Zadoff-Chu sequence
positive integer scalar
Example: 25
Complex Number Support: Yes
N - Length of the Zadoff-Chu sequence.
positive integer scalar
Example: 139
Complex Number Support: Yes

\section*{Output Arguments}

\section*{Examples}

SEQ - Zadoff-Chu output sequence
complex double-type column vector
The output sequence is a complex-valued column vector that contains the Rth root Zadoff-Chu sequence of length N .

\section*{Examine the correlation properties of a Zadoff-Chu sequence}

Generate the 25 th root length-139 Zadoff-Chu sequence.
```

seq = lteZadoffChuSeq(25, 139);
plot(abs(xcorr(seq)./length(seq)))

```

MATLAB displays the following image:


\section*{IteZadoffChuSeq}

\section*{References}
[1] 3rd Generation Partnership Project: Technical Specification Group Radio Access Network. "Evolved Universal Terrestrial Radio Access (E-UTRA)," Physical Channels and Modulation, Release 10, 2010-2012, TS 36.211, Vol. 10.0.0.

\author{
See Also \\ comm.GoldSequence | comm.PNSequence |
}

\section*{Purpose}

Generalized Marcum Q function
Syntax
\(Q=\operatorname{marcumq}(a, b)\)
\(Q=\operatorname{marcumq}(a, b, m)\)
\(Q=\operatorname{marcumq}(a, b)\) computes the Marcum \(Q\) function of \(a\) and \(b\), defined by
\[
Q(a, b)=\int_{b}^{\infty} x \exp \left(-\frac{x^{2}+a^{2}}{2}\right) I_{0}(a x) d x
\]
where a and b are nonnegative real numbers. In this expression, \(I_{0}\) is the modified Bessel function of the first kind of zero order.
\(Q=\operatorname{marcumq}(a, b, m)\) computes the generalized Marcum \(Q\), defined by
\[
Q(a, b)=\frac{1}{a^{m-1}} \int_{b}^{\infty} x^{m} \exp \left(-\frac{x^{2}+a^{2}}{2}\right) I_{m-1}(a x) d x
\]
where \(a\) and \(b\) are nonnegative real numbers, and \(m\) is a positive integer. In this expression, \(I_{\mathrm{m}-1}\) is the modified Bessel function of the first kind of order \(m-1\).

If any of the inputs is a scalar, it is expanded to the size of the other inputs.

\section*{References}
[1] Cantrell, P. E., and A. K. Ojha, "Comparison of Generalized Q-Function Algorithms," IEEE Transactions on Information Theory, Vol. IT-33, July, 1987, pp. 591-596.
[2] Marcum, J. I., "A Statistical Theory of Target Detection by Pulsed Radar: Mathematical Appendix," RAND Corporation, Santa Monica, CA, Research Memorandum RM-753, July 1, 1948. Reprinted in IRE Transactions on Information Theory, Vol. IT-6, April, 1960, pp. 59-267.
[3] Shnidman, D. A., "The Calculation of the Probability of Detection and the Generalized Marcum Q-Function," IEEE Transactions on Information Theory, Vol. IT-35, March, 1989, pp. 389-400.

See Also besseli

\section*{Purpose}

Convert mask vector to shift for shift register configuration

\section*{Syntax}

Description
shift = mask2shift(prpoly,mask)
shift = mask2shift(prpoly,mask) returns the shift that is equivalent to a mask, for a linear feedback shift register whose connections are specified by the primitive polynomial prpoly. The prpoly input can have one of these formats:
- A binary vector that lists the coefficients of the primitive polynomial in order of descending powers
- An integer scalar whose binary representation gives the coefficients of the primitive polynomial, where the least significant bit is the constant term

The mask input is a binary vector whose length is the degree of the primitive polynomial.

Note To save time, mask2shift does not check that prpoly is primitive. If it is not primitive, the output is not meaningful. To find primitive polynomials, use primpoly or see [2].

For more information about how masks and shifts are related to pseudonoise sequence generators, see shift2mask.

\section*{Definition of Equivalent Shift}

If \(A\) is a root of the primitive polynomial and \(m(A)\) is the mask polynomial evaluated at A, the equivalent shift s solves the equation \(A^{s}\) \(=\mathrm{m}(\mathrm{A})\). To interpret the vector mask as a polynomial, treat mask as a list of coefficients in order of descending powers.

\section*{Examples}

The first command below converts a mask of \(x^{3}+1\) into an equivalent shift for the linear feedback shift register whose connections are specified by the primitive polynomial \(x^{4}+x^{3}+1\). The second command shows that a mask of 1 is equivalent to a shift of 0 . In both cases,
notice that the length of the mask vector is one less than the length of the prpoly vector.
```

s = mask2shift([1 1 0 0 1],[1 0 0 1])
s2 = mask2shift([1 1 0 0 1],[0 0 0 1])

```

The output is below.
s =
4
s2 =

0

\author{
References [1] Lee, J. S., and L. E. Miller, CDMA Systems Engineering Handbook, Boston, Artech House, 1998. \\ [2] Simon, Marvin K., Jim K. Omura, et al., Spread Spectrum Communications Handbook, New York, McGraw-Hill, 1994. \\ See Also shift2mask | log | isprimitive | primpoly
}

\section*{Purpose}

\section*{Syntax}

Description

Restore ordering of symbols by filling matrix by columns and emptying it by rows
```

deintrlvd = matdeintrlv(data,Nrows,Ncols)

```
deintrlvd = matdeintrlv(data, Nrows, Ncols) rearranges the elements in data by filling a temporary matrix with the elements column by column and then sending the matrix contents, row by row, to the output. Nrows and Ncols are the dimensions of the temporary matrix. If data is a vector, it must have Nrows*Ncols elements. If data is a matrix with multiple rows and columns, data must have Nrows*Ncols rows and the function processes the columns independently.

To use this function as an inverse of the matintrlv function, use the same Nrows and Ncols inputs in both functions. In that case, the two functions are inverses in the sense that applying matintrlv followed by matdeintrlv leaves data unchanged.

Examples The code below illustrates the inverse relationship between matintrlv and matdeintrlv.
```

Nrows = 2; Ncols = 3;
data = [1 2 3 4 5 6; 2 4 6 8 10 12]';
a = matintrlv(data,Nrows,Ncols); % Interleave.
b = matdeintrlv(a,Nrows,Ncols) % Deinterleave.

```

The output below shows that b is the same as data.
\begin{tabular}{rr}
\(b=\) & \\
& \\
& \\
2 & 2 \\
2 & 4 \\
3 & 6 \\
4 & 8 \\
5 & 10 \\
& 12
\end{tabular}

\section*{matdeintrlv}

See Also
matintrlv
How To . "Interleaving"

\section*{Purpose}

Reorder symbols by filling matrix by rows and emptying it by columns
```

Syntax
intrlvd = matintrlv(data,Nrows,Ncols)

```
intrlvd = matintrlv(data,Nrows,Ncols) rearranges the elements in data by filling a temporary matrix with the elements row by row and then sending the matrix contents, column by column, to the output. Nrows and Ncols are the dimensions of the temporary matrix. If data is a vector, it must have Nrows*Ncols elements. If data is a matrix with multiple rows and columns, data must have Nrows*Ncols rows and the function processes the columns independently.

\section*{Examples}

The command below rearranges each of two columns of a matrix.
b = matintrlv([1 \(23456 ; 246810\) 12]',2,3)
b \(=\)
\(1 \quad 2\)
48
24
510
36
612
To form the first column of the output, the function creates the temporary 2-by-3 matrix [1 2 3; 456 ]. Then the function reads down each column of the temporary matrix to get \(\left[\begin{array}{llllll}1 & 4 & 2 & 5 & 3 & 6\end{array}\right]\).

\section*{See Also matdeintrlv}

How To . "Interleaving"

\section*{minpol}

Purpose Find minimal polynomial of Galois field element

\section*{Syntax \\ pl = minpol(x)}

Description
pl \(=\) minpol \((x)\) finds the minimal polynomial of each element in the Galois column vector, x . The output pl is an array in \(\mathrm{GF}(2)\). The kth row of pl lists the coefficients, in order of descending powers, of the minimal polynomial of the kth element of \(x\).

Note The output is in \(\mathrm{GF}(2)\) even if the input is in a different Galois field.

\section*{Examples}

The code below uses \(m=4\) and finds that the minimal polynomial of \(\mathrm{gf}(2, \mathrm{~m})\) is just the primitive polynomial used for the field \(\mathrm{GF}\left(2^{\wedge} \mathrm{m}\right)\). This is true for any value of \(m\), not just the value used in the example.
```

m = 4;
A = gf(2,m)
pl = minpol(A)

```

The output is below. Notice that the row vector [ \(\left.\begin{array}{lllll}1 & 0 & 0 & 1 & 1\end{array}\right]\) represents the polynomial D^4 + D + 1 .
\(A=G F(2 \wedge 4)\) array. Primitive polynomial \(=D^{\wedge} 4+D+1\) (19 decimal)
Array elements =
    2
\(\mathrm{pl}=\mathrm{GF}(2)\) array.
Array elements =
\begin{tabular}{lllll}
1 & 0 & 0 & 1 & 1
\end{tabular}

\title{
Another example is in "Minimal Polynomials".
}

\section*{See Also \\ cosets}

How To . "Polynomials over Galois Fields"

\section*{mldivide}

Purpose Matrix left division \of Galois arrays

\section*{Syntax \(\quad x=A \backslash B\)}

Description \(\quad x=A \backslash B\) divides the Galois array \(A\) into \(B\) to produce a particular solution of the linear equation \(A * x=B\). In the special case when \(A\) is a nonsingular square matrix, \(x\) is the unique solution, \(\operatorname{inv}(A) * B\), to the equation.

The code below shows that \(\mathrm{A} \backslash\) eye(size(A)) is the inverse of the nonsingular square matrix \(A\).
```

m = 4; A = gf([8 1 6; 3 5 7; 4 9 2],m);
Id = gf(eye(size(A)),m);
X = A \ Id;
ck1 = isequal(X*A, Id)
ck2 = isequal(A*X, Id)

```

The output is below.
ck1 =
1
ck2 =

1
Other examples are in "Solving Linear Equations".

\section*{Limitations The matrix A must be one of these types:}
- A nonsingular square matrix
- A tall matrix such that \(A^{\prime}\) *A is nonsingular
- A wide matrix such that \(\mathrm{A}^{*} \mathrm{~A}^{\prime}\) is nonsingular

\title{
Algorithms If \(A\) is an M-by- \(N\) tall matrix where \(M>N, A \backslash B\) is the same as \(\left(A^{\prime *} A\right) \backslash\left(A^{\prime} * B\right)\). \\ If \(A\) is an \(M\)-by- \(N\) wide matrix where \(M<N, A \backslash B\) is the same as \(A^{\prime} *\left(\left(A^{*} A^{\prime}\right) \backslash B\right)\). This solution is not unique.
}

\author{
How To \\ - "Linear Algebra in Galois Fields"
}

Purpose
Equalize linearly modulated signal using Viterbi algorithm
Syntax
```

y = mlseeq(x,chcffs,const,tblen,opmode)
y = mlseeq(x,chcffs,const,tblen,opmode,nsamp)
y = mlseeq(...,'rst',nsamp,preamble,postamble)
y = mlseeq(...,'cont',nsamp,...
init_metric,init_states,init_inputs)
[y,final_metric,final_states,final_inputs] = ...
mlseeq(...,'cont',...)

```

\section*{Description}
\(y=m l s e e q(x\), chcffs, const,tblen,opmode) equalizes the baseband signal vector \(x\) using the Viterbi algorithm. chcffs is a vector that represents the channel coefficients. const is a complex vector that lists the points in the ideal signal constellation, in the same sequence that the system's modulator uses. tblen is the traceback depth. The equalizer traces back from the state with the best metric. opmode denotes the operation mode of the equalizer; the choices are described in the following table.
\begin{tabular}{l|l}
\hline Value of opmode & Typical Usage \\
\hline 'rst' & \begin{tabular}{l} 
Enables you to specify a preamble and \\
postamble that accompany your data. The \\
function processes x independently of data \\
from any other invocations of this function. \\
This mode incurs no output delay.
\end{tabular} \\
\hline 'cont' & \begin{tabular}{l} 
Enables you to save the equalizer's internal \\
state information for use in a subsequent \\
invocation of this function. Repeated calls \\
to this function are useful if your data is \\
partitioned into a series of smaller vectors that \\
you process within a loop, for example. This \\
mode incurs an output delay of tblen symbols.
\end{tabular} \\
\hline
\end{tabular}

\footnotetext{
\(y=\) mlseeq (x,chcffs, const, tblen,opmode, nsamp) specifies the number of samples per symbol in \(x\), that is, the oversampling factor.
}

The vector length of \(x\) must be a multiple of nsamp. When nsamp \(>1\), the chcffs input represents the oversampled channel coefficients.

\section*{Preamble and Postamble in Reset Operation Mode}
\(y=\) mlseeq(...,'rst', nsamp, preamble, postamble) specifies the preamble and postamble that you expect to precede and follow, respectively, the data in the input signal. The vectors preamble and postamble consist of integers between 0 and \(\mathrm{M}-1\), where M is the order of the modulation, that is, the number of elements in const. To omit a preamble or postamble, specify [].
When the function applies the Viterbi algorithm, it initializes state metrics in a way that depends on whether you specify a preamble and/or postamble:
- If the preamble is nonempty, the function decodes the preamble and assigns a metric of 0 to the decoded state. If the preamble does not decode to a unique state (that is, if the length of the preamble is less than the channel memory), the decoder assigns a metric of 0 to all states that can be represented by the preamble. The traceback path ends at one of the states represented by the preamble.
- If the preamble is unspecified or empty, the decoder initializes the metrics of all states to 0 .
- If the postamble is nonempty, the traceback path begins at the smallest of all possible decoded states that are represented by the postamble.
- If the postamble is unspecified or empty, the traceback path starts at the state with the smallest metric.

\section*{Additional Syntaxes in Continuous Operation Mode}
y = mlseeq(...,'cont', nsamp,...
init_metric, init_states, init_inputs) causes the equalizer to start with \(\bar{i}\) its state metrics, traceback states, and traceback inputs specified by init_metric, init_states, and init_inputs, respectively. These three inputs are typically the extra outputs from a previous call to this function, as in the syntax below. Each real number in init_metric
represents the starting state metric of the corresponding state. init_states and init_inputs jointly specify the initial traceback memory of the equalizer. The table below shows the valid dimensions and values of the last three inputs, where numStates is \(\mathrm{M}^{\mathrm{L}-1}, \mathrm{M}\) is the order of the modulation, and L is the number of symbols in the channel's impulse response (with no oversampling). To use default values for all of the last three arguments, specify them as [], [ ], [ ].
\begin{tabular}{l|l|l|l}
\hline \begin{tabular}{l} 
Input \\
Argument
\end{tabular} & Meaning & Matrix Size & \begin{tabular}{l} 
Range of \\
Values
\end{tabular} \\
\hline init_metric & State metrics & \begin{tabular}{l}
1 row, numStates \\
columns
\end{tabular} & Real numbers \\
\hline init_states & \begin{tabular}{l} 
Traceback \\
states
\end{tabular} & \begin{tabular}{l} 
numStates rows, \\
tblen columns
\end{tabular} & \begin{tabular}{l} 
Integers \\
between 0 and \\
numStates-1
\end{tabular} \\
\hline init_inputs & \begin{tabular}{l} 
Traceback \\
inputs
\end{tabular} & \begin{tabular}{l} 
numStates rows, \\
tblen columns
\end{tabular} & \begin{tabular}{l} 
Integers between \\
0 and M-1
\end{tabular} \\
\hline
\end{tabular}
[y,final_metric,final_states,final_inputs] = ... mlseeq(...,'cont',...) returns the normalized state metrics, traceback states, and traceback inputs, respectively, at the end of the traceback decoding process. final_metric is a vector with numStates elements that correspond to the final state metrics. final_states and final_inputs are both matrices of size numStates-by-tblen.

\section*{Examples}

The example below illustrates how to use reset operation mode on an upsampled signal.
```

% Use 2-PAM.
M = 2; hMod = comm.PAMModulator(M); hDemod = comm.PAMDemodulator(M);
hChan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)', ..
'SNR',5);
const = step(hMod,(0:M-1)'); % PAM constellation
tblen = 10; % Traceback depth for equalizer
nsamp = 2; % Number of samples per symbol

```
```

msgIdx = randi([0 M-1],1000,1); % Random bits
msg = upsample(step(hMod,msgIdx),nsamp); % Modulated message
chcoeffs = [.986; . 845; .237; .12345+.31i]; % Channel coefficients
chanest = chcoeffs; % Channel estimate
hMLSEE = comm.MLSEEqualizer('TracebackDepth',tblen,...
'Channel',chanest, 'Constellation',const, 'SamplesPerSymbol', nsar
filtmsg = filter(chcoeffs,1,msg); % Introduce channel distortion.
msgRx = step(hChan,filtmsg); % Add Gaussian noise.
msgEq = step(hMLSEE,msgRx); % Equalize.
msgEqIdx = step(hDemod,msgEq); % Demodulate.
%Calculate BER
hErrorCalc = comm.ErrorRate;
berVec = step(hErrorCalc, msgIdx, msgEqIdx);
ber = berVec(1)
nerrs = berVec(2)

```

The output is shown below. Your results might vary because this example uses random numbers.
```

nerrs =

```
    1
ber \(=\)
    0.0010

The example in "Example: Continuous Operation Mode" illustrates how to use the final state and initial state arguments when invoking mlseeq repeatedly.
The example in "Use a Preamble in MATLAB" illustrates how to use a preamble.

\section*{References}
[1] Proakis, John G., Digital Communications, Fourth Edition, New York, McGraw-Hill, 2001.
[2] Steele, Raymond, Ed., Mobile Radio Communications, Chichester, England, John Wiley \& Sons, 1996.

\author{
See Also \\ equalize \\ How To \\ - "MLSE Equalizers"
}

\section*{Purpose}

Syntax

Description

Scaling factor for normalizing modulation output
scale \(=\) modnorm(const, 'avpow', avpow) scale \(=\) modnorm(const, 'peakpow', peakpow)
scale \(=\) modnorm(const, 'avpow', avpow) returns a scale factor for normalizing a PAM or QAM modulator output such that its average power is avpow (watts). const is a vector specifying the reference constellation used to generate the scale factor. The function assumes that the signal to be normalized has a minimum distance of 2 .
scale \(=\) modnorm(const, 'peakpow', peakpow) returns a scale factor for normalizing a PAM or QAM modulator output such that its peak power is peakpow (watts).

\section*{Examples}

The code below illustrates how to use modnorm to transmit a quadrature amplitude modulated signal having a peak power of one watt.
```

M = 16; % Alphabet size
% QAM Modulation
hMod = comm.RectangularQAMModulator(M);
hDemod = comm.RectangularQAMDemodulator(M);
% AWGNChannel System object
hChan = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)
'SNR', 10);
const = step(hMod,(0:M-1)'); % Generate the constellation.
x = randi([0 M-1], 100,1);
scale = modnorm(const,'peakpow',1); % Compute scale factor.
y = scale * step(hMod,x); % Modulate and scale.
hChan.SignalPower = (y' * y)/ length(y); % Calculate Signal Power
ynoisy = step(hChan,y); % Transmit along noisy channel.
ynoisy_unscaled = ynoisy/scale; % Unscale at receiver end.
z = step(hDemod,ynoisy_unscaled); % Demodulate.
% See how scaling affects constellation.
h = scatterplot(const,1,0,'ro'); % Unscaled constellation

```
```

hold on; % Next plot will be in same figure window.
scatterplot(const*scale,1,0,'bx',h); % Scaled constellation
hold off;

```

In the plot below, the plotting symbol o marks points on the original QAM signal constellation, and the plotting symbol \(x\) marks points on the signal constellation as scaled by the output of the modnorm function. The channel in this example carries points from the scaled constellation.


Additional examples using modnorm are in "Examples of Signal Constellation Plots".

See Also pammod | pamdemod | qammod | qamdemod
How To
- "Digital Modulation"

\section*{Purpose Minimum shift keying demodulation}

Syntax
\(z=m s k d e m o d(y, n s a m p)\)
\(z=\operatorname{mskdemod}(y, n s a m p, d a t a e n c)\)
z = mskdemod(y,nsamp,dataenc,ini_phase)
z = mskdemod(y,nsamp,dataenc,ini_phase,ini_state)
[z,phaseout] = mskdemod(...)
[z,phaseout,stateout] = mskdemod(...)

\section*{Description}

\section*{Warning}

This function is obsolete and may be removed in the future. We strongly recommend that you use the comm.MSKDemodulator System object instead.
\(z=m s k d e m o d(y, n s a m p)\) demodulates the complex envelope \(y\) of a signal using the differentially encoded minimum shift keying (MSK) method. nsamp denotes the number of samples per symbol and must be a positive integer. The initial phase of the demodulator is 0 . If \(y\) is a matrix with multiple rows and columns, the function treats the columns as independent channels and processes them independently.
z = mskdemod(y,nsamp,dataenc) specifies the method of encoding data for MSK. dataenc can be either 'diff' for differentially encoded MSK or 'nondiff' for nondifferentially encoded MSK.
z = mskdemod(y,nsamp,dataenc,ini_phase) specifies the initial phase of the demodulator. ini_phase is a row vector whose length is the number of channels in \(y\) and whose values are integer multiples of pi/2. To avoid overriding the default value of dataenc, set dataenc to [].
z = mskdemod(y,nsamp,dataenc,ini_phase,ini_state) specifies the initial state of the demodulator. ini_state contains the last half symbol of the previously received signal. ini_state is an nsamp-by-C matrix, where C is the number of channels in y .
[z, phaseout] = mskdemod(...) returns the final phase of \(y\), which is important for demodulating a future signal. The output phaseout has
the same dimensions as the ini_phase input, and assumes the values \(0, \mathrm{pi} / 2\), pi, and \(3 * \mathrm{pi} / 2\).
[z, phaseout, stateout] = mskdemod (...) returns the final nsamp values of \(y\), which is useful for demodulating the first symbol of a future signal. stateout has the same dimensions as the ini_state input.

\section*{Examples}

The example below illustrates how to modulate and demodulate within a loop. To provide continuity from one iteration to the next, the syntaxes for mskmod and mskdemod use initial phases and/or state as both input and output arguments.
```

% Define parameters.
numbits = 99; % Number of bits per iteration
numchans = 2; % Number of channels (columns) in signal
nsamp = 16; % Number of samples per symbol
% Initialize.
numerrs = 0; % Number of bit errors seen so far
demod_ini_phase = zeros(1,numchans); % Modulator phase
mod_ini_phase = zeros(1,numchans); % Demodulator phase
ini_state = complex(zeros(nsamp,numchans)); % Demod. state
% Main loop
for iRuns = 1 : 10
x = randi([0 1],numbits,numchans); % Binary signal
[y,phaseout] = mskmod(x,nsamp,[],mod_ini_phase);
mod_ini_phase = phaseout; % For next mskmod command
[z, phaseout, stateout] = ...
mskdemod(y,nsamp,[],demod_ini_phase,ini_state);
ini_state = stateout; % For next mskdemod command
demod_ini_phase = phaseout; % For next mskdemod command
numerrs = numerrs + biterr(x,z); % Cumulative bit errors
end
disp(['Total number of bit errors = ' num2str(numerrs)])

```

The output is as follows.
Total number of bit errors \(=0\)
References [1] Pasupathy, Subbarayan, "Minimum Shift Keying: A Spectrally Efficient Modulation," IEEE Communications Magazine, July, 1979, pp. 14-22.
See Also mskmod | fskmod | fskdemod
How To - "Digital Modulation"

Purpose Minimum shift keying modulation
```

Syntax $\quad y=\operatorname{mskmod}(x, n s a m p)$
y $=$ mskmod(x, nsamp,dataenc)
y = mskmod(x,nsamp,dataenc,ini_phase)
[y, phaseout] = mskmod(...)

```

\section*{Description}

\section*{Examples}

\section*{Warning}

This function is obsolete and may be removed in the future. We strongly recommend that you use the comm.MSKModulator System object instead.
\(y=\operatorname{mskmod}(x, n s a m p)\) outputs the complex envelope \(y\) of the modulation of the message signal \(x\) using differentially encoded minimum shift keying (MSK) modulation. The elements of \(x\) must be 0 or 1. nsamp denotes the number of samples per symbol in \(y\) and must be a positive integer. The initial phase of the MSK modulator is 0 . If x is a matrix with multiple rows and columns, the function treats the columns as independent channels and processes them independently.
\(\mathrm{y}=\operatorname{mskmod}(\mathrm{x}, \mathrm{nsamp}\), dataenc \()\) specifies the method of encoding data for MSK. dataenc can be either 'diff' for differentially encoded MSK or 'nondiff' for nondifferentially encoded MSK.
\(y=m s k m o d\left(x, n s a m p, d a t a e n c, i n i \_p h a s e\right)\) specifies the initial phase of the MSK modulator. ini_phase is a row vector whose length is the number of channels in y and whose values are integer multiples of pi/2. To avoid overriding the default value of dataenc, set dataenc to [].
[y, phaseout] \(=\operatorname{mskmod}(\ldots)\) returns the final phase of \(y\). This is useful for maintaining phase continuity when you are modulating a future bit stream with differentially encoded MSK. phaseout has the same dimensions as the ini_phase input, and assumes the values 0 , pi/2, pi, and \(3 * p i / 2\).

Create an eye diagram from an MSK signal.
\[
x=\operatorname{randi}([01], 99,1) ; \% \text { Random signal }
\]
```

y = mskmod(x,8,[],pi/2);
y = awgn(y,30,'measured');
eyediagram(y,16);

```


The example on the reference page for mskdemod also uses this function.

\section*{References}

See Also
[1] Pasupathy, Subbarayan, "Minimum Shift Keying: A Spectrally Efficient Modulation," IEEE Communications Magazine, July, 1979, pp. 14-22.

\section*{muxdeintrlv}
\begin{tabular}{ll} 
Purpose & Restore ordering of symbols using specified shift registers \\
Syntax & \\
& deintrlved \(=\) muxdeintrlv(data, delay) \\
& {\([\) deintrlved, state \(=\) muxdeintrlv(data, delay) } \\
& {\([\) deintrlved,state \(]=\) muxdeintrlv(data, delay,init_state) }
\end{tabular}

\section*{Description}

\section*{Examples}

The example below illustrates how to use the state input and output when invoking muxdeintrlv repeatedly. Notice that [deintrlved1; deintrlved2] is the same as deintrlved.
```

delay = [0 4 8 12]; % Delays in shift registers
symbols = 100; % Number of symbols to process
% Interleave random data.
intrlved = muxintrlv(randi([0 1],symbols,1),delay);
% Deinterleave some of the data, recording state for later use.
[deintrlved1,state] = muxdeintrlv(intrlved(1:symbols/2),delay);
% Deinterleave the rest of the data, using state as an input argument.
deintrlved2 = muxdeintrlv(intrlved(symbols/2+1:symbols),delay,state);
% Deinterleave all data in one step.
deintrlved = muxdeintrlv(intrlved,delay);
isequal(deintrlved,[deintrlved1; deintrlved2])

```

The output is below.
ans =
1
Another example using this function is in "Convolutional Interleaving and Deinterleaving Using a Sequence of Consecutive Integers in MATLAB".
\begin{tabular}{ll} 
References & \begin{tabular}{l} 
[1] Heegard, Chris, and Stephen B. Wicker, Turbo Coding, Boston, \\
Kluwer Academic Publishers, 1999.
\end{tabular} \\
See Also & muxintrlv \\
How To & . "Interleaving"
\end{tabular}
\begin{tabular}{ll} 
Purpose & Permute symbols using shift registers with specified delays \\
Syntax & \begin{tabular}{l} 
intrlved \(=\) muxintrlv(data, delay) \\
\\
{\([\) intrlved, state \(]=\) muxintrlv(data, delay) }
\end{tabular} \\
& {\([\) intrlved, state \(]=\) muxintrlv(data, delay, init_state) }
\end{tabular}

\section*{Examples}

References

See Also
intrlved = muxintrlv(data, delay) permutes the elements in data by using internal shift registers, each with its own delay value. delay is a vector whose entries indicate how many symbols each shift register can hold. The length of delay is the number of shift registers. Before the function begins to process data, it initializes all shift registers with zeros. If data is a matrix with multiple rows and columns, the function processes the columns independently.
[intrlved,state] = muxintrlv(data,delay) returns a structure that holds the final state of the shift registers. state. value stores any unshifted symbols. state. index is the index of the next register to be shifted.
[intrlved,state] = muxintrlv(data,delay,init_state) initializes the shift registers with the symbols contained in init_state.value and directs the first input symbol to the shift register referenced by init_state.index. The structure init_state is typically the state output from a previous call to this same function, and is unrelated to the corresponding deinterleaver.

The examples in "Convolutional Interleaving and Deinterleaving Using a Sequence of Consecutive Integers in MATLAB" and on the reference page for the convintrlv function use muxintrlv.

The example on the reference page for muxdeintrlv illustrates how to use the state output and init_state input with that function; the process is analogous for this function.
[1] Heegard, Chris, and Stephen B. Wicker, Turbo Coding, Boston, Kluwer Academic Publishers, 1999.
muxdeintrlv | convintrlv | helintrlv

How To . "Interleaving"

Purpose Equivalent noise bandwidth of filter
Syntax bw = noisebw(num, den, numsamp, Fs)
Description
bw = noisebw(num, den, numsamp, Fs) returns the two-sided equivalent noise bandwidth, in Hz , of a digital lowpass filter given in descending powers of z by numerator vector num and denominator vector den. The bandwidth is calculated over numsamp samples of the impulse response. Fs is the sampling rate of the signal that the filter would process; this is used as a scaling factor to convert a normalized unitless quantity into a bandwidth in Hz .

\section*{Examples}

This example computes the equivalent noise bandwidth of a Butterworth filter over 100 samples of the impulse response.
```

Fs = 16; % Sampling rate
Fnyq = Fs/2; % Nyquist frequency
Fc = 0.5; % Carrier frequency
[num,den] = butter(2,Fc/Fnyq); % Butterworth filter
bw = noisebw(num,den,100,Fs)

```

The output is below.
bw =
1.1049

Algorithms The two-sided equivalent noise bandwidth is
\[
\frac{\mathrm{Fs} \sum_{i=1}^{N}|h(i)|^{2}}{\left|\sum_{i=1}^{N} h(i)\right|^{2}}
\]
where \(h\) is the impulse response of the filter described by num and den, and \(N\) is numsamp.

\section*{References}
[1] Jeruchim, Michel C., Philip Balaban, and K. Sam Shanmugan, Simulation of Communication Systems, New York, Plenum Press, 1992.
```

Purpose Construct normalized least mean square (LMS) adaptive algorithm
object
Syntax
alg = normlms(stepsize)
alg = normlms(stepsize,bias)

```

\section*{Description}

The normlms function creates an adaptive algorithm object that you can use with the lineareq function or dfe function to create an equalizer object. You can then use the equalizer object with the equalize function to equalize a signal. To learn more about the process for equalizing a signal, see "Adaptive Algorithms".
alg = normlms(stepsize) constructs an adaptive algorithm object based on the normalized least mean square (LMS) algorithm with a step size of stepsize and a bias parameter of zero.
alg \(=\) normlms(stepsize,bias) sets the bias parameter of the normalized LMS algorithm. bias must be between 0 and 1 . The algorithm uses the bias parameter to overcome difficulties when the algorithm's input signal is small.

\section*{Properties}

The table below describes the properties of the normalized LMS adaptive algorithm object. To learn how to view or change the values of an adaptive algorithm object, see "Access Properties of an Adaptive Algorithm".
\begin{tabular}{l|l}
\hline Property & Description \\
\hline AlgType & Fixed value, 'Normalized LMS ' \\
\hline StepSize & \begin{tabular}{l} 
LMS step size parameter, a \\
nonnegative real number
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline LeakageFactor & \begin{tabular}{l} 
LMS leakage factor, a real \\
number between 0 and 1. A value \\
of 1 corresponds to a conventional \\
weight update algorithm, while \\
a value of 0 corresponds to a \\
memoryless update algorithm.
\end{tabular} \\
\hline Bias & \begin{tabular}{l} 
Normalized LMS bias parameter, \\
a nonnegative real number
\end{tabular} \\
\hline
\end{tabular}

\section*{Examples \\ Algorithms}

\section*{References}

See Also
How To

For an example that uses this function, see "Delays from Equalization".
Referring to the schematics presented in "Equalizer Structure", define \(w\) as the vector of all weights \(w_{\mathrm{i}}\) and define \(u\) as the vector of all inputs \(u_{\mathrm{i}}\). Based on the current set of weights, \(w\), this adaptive algorithm creates the new set of weights given by
\[
(\text { LeakageFactor }) w+\frac{(\text { StepSize }) u^{*} e}{u^{H} u+\text { Bias }}
\]
where the * operator denotes the complex conjugate and \(H\) denotes the Hermitian transpose.
[1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, John Wiley \& Sons, 1998.
lms | signlms | varlms | rls | cma | lineareq | dfe | equalize
- "Equalization"

Purpose Convert octal to decimal numbers

\section*{Syntax \(\quad d=\operatorname{oct2dec}(c)\)}

Description \(d=\operatorname{oct} 2 d e c(c)\) converts an octal matrix \(c\) to a decimal matrix \(d\), element by element. In both octal and decimal representations, the rightmost digit is the least significant.

Examples The command below converts a 2-by-2 octal matrix.
d = oct2dec([12 144;0 25])
d \(=\)
\(10 \quad 100\)
021
For instance, the octal number 144 is equivalent to the decimal number 100 because \(144(\) octal \()=1 * 8^{2}+4 * 8^{1}+4 * 8^{0}=64+32+4=100\).

See Also bi2de
\begin{tabular}{ll} 
Purpose & Offset quadrature phase shift keying demodulation \\
Syntax & \begin{tabular}{l}
\(z=\operatorname{oqpskdemod}(y)\) \\
\(z=\operatorname{oqpskdemod}(y\), ini_phase \()\)
\end{tabular} \\
Description & \begin{tabular}{l}
\(z=\operatorname{oqpskdemod}(y)\) demodulates the complex envelope \(y\) of an OQPSK \\
modulated signal. The function implicitly downsamples by a factor of 2 \\
because OQPSK does not permit an odd number of samples per symbol. \\
If y is a matrix with multiple rows, the function processes the columns \\
independently. \\
\(z=o\) oqpskdemod \((y\), ini_phase) specifies the phase offset of the \\
modulated signal in radians.
\end{tabular} \\
See Also & \begin{tabular}{l} 
oqpskmod \(\mid\) pskmod \(\mid\) pskdemod \(\mid\) qammod \(\mid\) qamdemod \(\mid\) modnorm
\end{tabular} \\
How To & - "Digital Modulation"
\end{tabular}

\section*{oqpskmod}

Purpose Offset quadrature phase shift keying modulation
```

Syntax
y = oqpskmod(x)
y = oqpskmod(x,ini_phase)

```

Description

See Also
How To
\begin{tabular}{|c|c|}
\hline Purpose & Pulse amplitude demodulation \\
\hline Syntax & \[
\begin{aligned}
& z=\operatorname{pamdemod}(y, M) \\
& z=\operatorname{pamdemod}(y, M, \text { ini_phase }) \\
& z=\operatorname{pamdemod}(y, M, \text { ini_phase, symbol_order })
\end{aligned}
\] \\
\hline Description & \begin{tabular}{l}
\(z=\operatorname{pamdemod}(y, M)\) demodulates the complex envelope \(y\) of a pulse amplitude modulated signal. M is the alphabet size. The ideal modulated signal should have a minimum Euclidean distance of 2. \\
z = pamdemod(y, M,ini_phase) specifies the initial phase of the modulated signal in radians. \\
z = pamdemod(y,M,ini_phase,symbol_order) specifies how the function assigns binary words to corresponding integers. If symbol_order is set to 'bin' (default), the function uses a natural binary-coded ordering. If symbol_order is set to 'gray', it uses a Gray-coded ordering.
\end{tabular} \\
\hline Examples & The example in "Comparing Theoretical and Empirical Error Rates" uses this function. \\
\hline See Also & pammod | qamdemod | qammod \| pskdemod \| pskmod \\
\hline How To & - "Digital Modulation" \\
\hline
\end{tabular}
```

Purpose Pulse amplitude modulation

```
```

Syntax

```
Syntax
y = pammod(x,M)
y = pammod(x,M)
y = pammod(x,M,ini_phase)
y = pammod(x,M,ini_phase)
y = pammod(x,M,ini_phase,symbol_order)
```

y = pammod(x,M,ini_phase,symbol_order)

```

\section*{Description}

\section*{Examples}

How To

See Also pamdemod \| qammod \| qamdemod \| pskmod \| pskdemod
\(y=\operatorname{pammod}(x, M)\) outputs the complex envelope \(y\) of the modulation of the message signal \(x\) using pulse amplitude modulation. \(M\) is the alphabet size. The message signal must consist of integers between 0 and M-1. The modulated signal has a minimum Euclidean distance of 2. If \(x\) is a matrix with multiple rows, the function processes the columns independently.
\(y=\operatorname{pammod}(x, M\), ini_phase \()\) specifies the initial phase of the modulated signal in radians.
\(y=\operatorname{pammod}\left(x, M, i n i \_p h a s e, s y m b o l \_o r d e r\right)\) specifies how the function assigns binary words to corresponding integers. If symbol_order is set to 'bin' (default), the function uses a natural binary-coded ordering. If symbol_order is set to 'gray', it uses a Gray constellation ordering.

The example in "Comparing Theoretical and Empirical Error Rates" uses this function.
- "Digital Modulation"
Purpose Plot channel characteristics with channel visualization tool
Syntax ..... plot(h)
Description plot ( h ), where h is a channel object, launches the channel visualizationtool. This GUI tool allows you to plot channel characteristics in variousways. See "Channel Visualization" for details.
Examples Examples using this plotting tool are found in "Examples of Using the Channel Visualization Tool".
See Also filter | rayleighchan | ricianchan

\section*{pmdemod}

Purpose Phase demodulation
```

Syntax z = pmdemod}(y,Fc,Fs,phasedev
z = pmdemod(y,Fc,Fs,phasedev,ini_phase)

```

Description \(\quad z=\operatorname{pmdemod}(y, F c, F s\), phasedev \()\) demodulates the phase-modulated signal \(y\) at the carrier frequency Fc (hertz). \(z\) and the carrier signal have sampling rate Fs (hertz), where Fs must be at least 2*Fc. The phasedev argument is the phase deviation of the modulated signal, in radians.
z = pmdemod(y,Fc,Fs, phasedev,ini_phase) specifies the initial phase of the modulated signal, in radians.

\section*{Examples \\ The example in "Analog Modulation with Additive White Gaussian Noise (AWGN) Using MATLAB" uses pmdemod.}

See Also pmmod \| fmmod \| fmdemod
How To . "Digital Modulation"
Purpose Phase modulation
Syntax y = pmmod(x,Fc,Fs, phasedev)
y = pmmod(x,Fc,Fs,phasedev,ini_phase)
Description \(y=p m m o d(x, F c, F s, p h a s e d e v)\) modulates the message signal \(x\) usingphase modulation. The carrier signal has frequency Fc (hertz) andsampling rate Fs (hertz), where Fs must be at least 2*Fc. The phasedevargument is the phase deviation of the modulated signal in radians.
y = pmmod(x,Fc,Fs,phasedev,ini_phase) specifies the initial phase of the modulated signal in radians.
Examples The example in "Analog Modulation with Additive White Gaussian Noise (AWGN) Using MATLAB" uses pmmod.
See Also ..... pmdemod | fmmod | fmdemod
How To - "Digital Modulation"

\section*{poly2trellis}

Purpose
Syntax

Description

Convert convolutional code polynomials to trellis description
```

trellis = poly2trellis(ConstraintLength,CodeGenerator)
trellis = poly2trellis(ConstraintLength,CodeGenerator,...
FeedbackConnection)

```

The poly2trellis function accepts a polynomial description of a convolutional encoder and returns the corresponding trellis structure description. The output of poly2trellis is suitable as an input to the convenc and vitdec functions, and as a mask parameter for the Convolutional Encoder, Viterbi Decoder, and APP Decoder blocks in Communications System Toolbox software.
trellis = poly2trellis(ConstraintLength,CodeGenerator) performs the conversion for a rate \(\mathrm{k} / \mathrm{n}\) feedforward encoder. ConstraintLength is a 1-by-k vector that specifies the delay for the encoder's k input bit streams. CodeGenerator is a k-by-n matrix of octal numbers that specifies the n output connections for each of the encoder's k input bit streams.
trellis = poly2trellis(ConstraintLength,CodeGenerator,... FeedbackConnection) is the same as the syntax above, except that it applies to a feedback, not feedforward, encoder. FeedbackConnection is a 1-by-k vector of octal numbers that specifies the feedback connections for the encoder's \(k\) input bit streams.

For both syntaxes, the output is a MATLAB structure whose fields are as in the table below.

Fields of the Output Structure trellis for a Rate k/n Code
\begin{tabular}{l|l|l}
\hline \begin{tabular}{l} 
Field in trellis \\
Structure
\end{tabular} & Dimensions & Meaning \\
\hline numInputSymbols & Scalar & \begin{tabular}{l} 
Number of input \\
symbols to the \\
encoder: \(2^{\mathrm{k}}\)
\end{tabular} \\
\hline numOutputSymbols & Scalar & \begin{tabular}{l} 
Number of output \\
symbols from the \\
encoder: \(2^{\mathrm{n}}\)
\end{tabular} \\
\hline numStates & Scalar & \begin{tabular}{l} 
Number of states in \\
the encoder
\end{tabular} \\
\hline nextStates & \begin{tabular}{l} 
numStates-by-2 \\
matrix
\end{tabular} & \begin{tabular}{l} 
Next states for all \\
combinations of \\
current state and \\
current input
\end{tabular} \\
\hline outputs & \begin{tabular}{l} 
numStates-by-2 \(2^{\mathrm{k}}\) \\
matrix
\end{tabular} & \begin{tabular}{l} 
Outputs (in octal) \\
for all combinations \\
of current state and \\
current input
\end{tabular} \\
\hline
\end{tabular}

For more about this structure, see the reference page for the istrellis function.

\section*{Examples}

An example of a rate \(1 / 2\) encoder is in "Polynomial Description of a Convolutional Code".

As another example, consider the rate \(2 / 3\) feedforward convolutional encoder depicted in the figure below. The reference page for the convenc function includes an example that uses this encoder.


For this encoder, the ConstraintLength vector is [5,4] and the CodeGenerator matrix is [23,35,0; \(0,5,13\) ]. The output below reveals part of the corresponding trellis structure description of this encoder.
```

trellis = poly2trellis([5 4],[23 35 0; 0 5 13])
trellis =
numInputSymbols: 4
numOutputSymbols: 8
numStates: 128
nextStates: [128\times4 double]
outputs: [128x4 double]

```

The scalar field trellis.numInputSymbols has the value 4 because the combination of two input bit streams can produce four different input
symbols. Similarly, trellis.numOutputSymbols is 8 because the three output bit streams can produce eight different output symbols.

The scalar field trellis.numStates is 128 (that is, \(2^{7}\) ) because each of the encoder's seven memory registers can have one of two binary values.

To get details about the matrix fields trellis.nextStates and trellis.outputs, inquire specifically about them. As an example, the command below displays the first five rows of the 128 -by- 4 matrix trellis.nextStates.
trellis.nextStates(1:5,:)
ans =
\begin{tabular}{rrrr}
0 & 64 & 8 & 72 \\
0 & 64 & 8 & 72 \\
1 & 65 & 9 & 73 \\
1 & 65 & 9 & 73 \\
2 & 66 & 10 & 74
\end{tabular}

This first row indicates that if the encoder starts in the zeroth state and receives input bits of \(00,01,10\), or 11 , respectively, the next state will be the 0 th, 64 th, 8 th, or 72 nd state, respectively. The 64 th state means that the bottom-left memory register in the diagram contains the value 1 , while the other six memory registers contain zeros.

\author{
See Also istrellis | convenc | vitdec \\ How To . "Convolutional Codes"
}

\section*{primpoly}

Purpose Find primitive polynomials for Galois field
Syntax
pr = primpoly(m)
pr = primpoly(m,opt)
pr = primpoly(m...,'nodisplay')

\section*{Description}
\(\mathrm{pr}=\mathrm{primpoly}(\mathrm{m})\) returns the primitive polynomial for GF(2^m), where \(m\) is an integer between 2 and 16. The Command Window displays the polynomial using " D " as an indeterminate quantity. The output argument \(p r\) is an integer whose binary representation indicates the coefficients of the polynomial.
\(\mathrm{pr}=\) primpoly (m,opt) returns one or more primitive polynomials for \(\mathrm{GF}\left(2^{\wedge} \mathrm{m}\right)\). The output pol depends on the argument opt as shown in the table below. Each element of the output argument pr is an integer whose binary representation indicates the coefficients of the corresponding polynomial. If no primitive polynomial satisfies the constraints, pr is empty.
\begin{tabular}{l|l}
\hline opt & Meaning of pr \\
\hline 'min' & \begin{tabular}{l} 
One primitive polynomial for \\
GF \(\left(2^{\wedge} m\right)\) having the smallest \\
possible number of nonzero terms
\end{tabular} \\
\hline 'max' & \begin{tabular}{l} 
One primitive polynomial for \\
GF \(\left(2^{\wedge} m\right)\) having the greatest \\
possible number of nonzero terms
\end{tabular} \\
\hline 'all' & \begin{tabular}{l} 
All primitive polynomials for \\
GF \(\left(2^{\wedge} m\right)\)
\end{tabular} \\
\hline Positive integer k & \begin{tabular}{l} 
All primitive polynomials for \\
GF \(\left(2^{\wedge} m\right)\) that have k nonzero \\
terms
\end{tabular} \\
\hline
\end{tabular}
pr = primpoly(m...,'nodisplay') prevents the function from displaying the result as polynomials in "D" in the Command Window. The output argument pr is unaffected by the ' nodisplay' option.

Examples The first example below illustrates the formats that primpoly uses in the Command Window and in the output argument pr. The subsequent examples illustrate the display options and the use of the opt argument.
```

pr = primpoly(4)
pr1 = primpoly(5,'max','nodisplay')
pr2 = primpoly(5,'min')
pr3 = primpoly(5,2)
pr4 = primpoly(5,3);

```
The output is below.
Primitive polynomial(s) =
\(D^{\wedge} 4+D^{\wedge} 1+1\)
pr \(=\)
    19
pr1 =
    61
Primitive polynomial(s) =
\(D^{\wedge} 5+D^{\wedge} 2+1\)
pr2 \(=\)

37

No primitive polynomial satisfies the given constraints. pr3 =
[]

Primitive polynomial(s) =
\(D^{\wedge} 5+D^{\wedge} 2+1\)
\(D^{\wedge} 5+D^{\wedge} 3+1\)
See Also isprimitive
How To - "Galois Field Computations"
\begin{tabular}{ll} 
Purpose & Phase shift keying demodulation \\
Syntax & \(z=\operatorname{pskdemod}(y, M)\) \\
& \(z=\operatorname{pskdemod}(y, M\), ini_phase \()\) \\
& \(z=\operatorname{pskdemod}(y, M\), ini_phase, symbol_order \()\)
\end{tabular}

\section*{Description}

\section*{Examples}

The example below compares PSK and PAM (phase amplitude modulation) to show that PSK is more sensitive to phase noise. This is the expected result because the PSK constellation is circular, and the PAM constellation is linear.
```

len = 10000; % Number of symbols
M = 16; % Size of alphabet
msg = randi([O M-1],len,1); % Original signal
% Modulate using both PSK and PAM,
% to compare the two methods.
txpsk = pskmod(msg,M);
txpam = pammod(msg,M);
% Perturb the phase of the modulated signals.
phasenoise = randn(len,1)*.015;
rxpsk = txpsk.*exp(j*2*pi*phasenoise);

```
```

rxpam = txpam.*exp(j*2*pi*phasenoise);
% Create a scatter plot of the received signals.
scatterplot(rxpsk); title('Noisy PSK Scatter Plot')
scatterplot(rxpam); title('Noisy PAM Scatter Plot')
% Demodulate the received signals.
recovpsk = pskdemod(rxpsk,M);
recovpam = pamdemod(rxpam,M);
% Compute number of symbol errors in each case.
numerrs_psk = symerr(msg,recovpsk)
numerrs_pam = symerr(msg,recovpam)

```

The output and scatter plots are below. Your results might vary because this example uses random numbers.
numerrs_psk = 374
numerrs_pam =

1

\(\begin{array}{ll}\text { See Also } & \text { pskmod | qamdemod | qammod | dpskmod | dpskdemod | modnorm } \\ \text { How To } & \text {. "Digital Modulation" }\end{array}\)
```

Purpose Phase shift keying modulation

```
```

Syntax $\quad y=\operatorname{pskmod}(x, M)$

```
Syntax \(\quad y=\operatorname{pskmod}(x, M)\)
y = pskmod(x,M,ini_phase)
y = pskmod(x,M,ini_phase)
y = pskmod(x,M,ini_phase,symbol_order)
```

y = pskmod(x,M,ini_phase,symbol_order)

```

\section*{Description}

\section*{Examples}

How To

See Also \(\begin{aligned} & \text { dpskmod | dpskdemod | pskdemod | pammod | pamdemod | qammod | } \\ & \text { qamdemod | modnorm }\end{aligned}\)
\(y=p s k m o d(x, M)\) outputs the complex envelope \(y\) of the modulation of the message signal \(x\) using phase shift keying modulation. \(M\) is the alphabet size and must be an integer power of 2 . The message signal must consist of integers between 0 and \(M-1\). The initial phase of the modulation is zero. If \(x\) is a matrix with multiple rows and columns, the function processes the columns independently.
\(y=\operatorname{pskmod}\left(x, M, i n i \_p h a s e\right)\) specifies the initial phase of the modulation in radians.
\(y=p s k m o d\left(x, M, i n i \_p h a s e, s y m b o l \_o r d e r\right)\) specifies how the function assigns binary words to corresponding integers. If symbol_order is set to 'bin' (default), the function uses a natural binary-coded ordering. If symbol_order is set to 'gray', it uses a Gray constellation ordering.

The examples in "Create 16-PSK Constellation Scatter Plot" and on the reference page for pskdemod use this function.
- "Digital Modulation"
\begin{tabular}{ll} 
Purpose & Quadrature amplitude demodulation \\
Syntax & \\
& \(z=\operatorname{qamdemod}(y, M)\) \\
& \(z=\operatorname{qamdemod}(y, M\), ini_phase \()\) \\
& \(z=\operatorname{qamdemod}(y, M\), ini_phase, symbol_order \()\)
\end{tabular}

\section*{Description}

\section*{Examples}
\(z=\) qamdemod \((y, M)\) demodulates the complex envelope \(y\) of a quadrature amplitude modulated signal. \(M\) is the alphabet size and must be an integer power of 2 . The constellation is the same as in qammod. If \(y\) is a matrix with multiple rows, the function processes the columns independently.
z = qamdemod(y,M,ini_phase) specifies the initial phase of the modulated signal in radians.
z = qamdemod(y,M,ini_phase,symbol_order) specifies how the function assigns binary words to corresponding integers. If symbol_order is set to 'bin' (default), the function uses a natural binary-coded ordering. If symbol_order is set to 'gray', it uses a Gray-coded ordering.

The code below suggests which regions in the complex plane are associated with different digits that can form the output of the demodulator. The code demodulates random points, looks for points that were demapped to the digits 0 and 3 , and plots those points in red and blue, respectively. Notice that the regions reflect a rotation of the signal constellation by pi/8.
```

% Construct [in-phase, quadrature] for random points.
y = 4*(rand(1000,1)-1/2)+j*4*(rand(1000,1)-1/2);
% Demodulate using an initial phase of pi/8.
z = qamdemod(y,4,pi/8);
% Find indices of points that mapped to the digits 0 and 3.
red = find(z==0);
blue = find(z==3);
% Plot points corresponding to 0 and 3.
h = scatterplot(y(red,:),1,0,'r.'); hold on
scatterplot(y(blue,:),1,0,'b.',h);

```
legend('Points corresponding to 0 ','Points corresponding to 3 '); hold off


Another example using this function is in "Compute the Symbol Error Rate".
\begin{tabular}{ll} 
See Also & qammod | genqamdemod | genqammod | pamdemod | modnorm \\
How To & - "Digital Modulation"
\end{tabular}
Purpose Quadrature amplitude modulation
```

Syntax
$y=\operatorname{qammod}(x, M)$
y = qammod(x,M,ini_phase)
y = qammod(x,M,ini_phase,symbol_order)

```

\section*{Description}
\(y=q \operatorname{ammod}(x, M)\) outputs the complex envelope \(y\) of the modulation of the message signal \(x\) using quadrature amplitude modulation. \(M\) is the alphabet size and must be an integer power of 2 . The message signal must consist of integers between 0 and \(\mathrm{M}-1\). The signal constellation is rectangular or cross-shaped, and the nearest pair of points in the constellation is separated by 2 . If x is a matrix with multiple rows, the function processes the columns independently.
\(y=\) qammod( \(x, M\), ini_phase) specifies the initial phase of the modulated signal in radians.
y = qammod(x,M,ini_phase,symbol_order) specifies how the function assigns binary words to corresponding integers. If symbol_order is set to 'bin' (default), the function uses a natural binary-coded ordering. If symbol_order is set to 'gray', it uses a Gray constellation ordering.

\section*{Examples}
Examples using this function are in "Compute the Symbol Error Rate" and "Examples of Signal Constellation Plots".

\footnotetext{
See Also qamdemod \| genqammod \| genqamdemod \| pammod \| pamdemod \| modnorm How To - "Digital Modulation"
}

Purpose \(\quad\) Q function

\section*{Syntax \\ \(y=q f u n c(x)\)}

Description
\(y=q f u n c(x)\) is one minus the cumulative distribution function of the standardized normal random variable, evaluated at each element of the real array \(x\). For a scalar x, the formula is
\[
Q(x)=\frac{1}{\sqrt{2 \pi}} \int_{x}^{\infty} \exp \left(-t^{2} / 2\right) d t
\]

The Q function is related to the complementary error function, erfc, according to
\[
Q(x)=\frac{1}{2} \operatorname{erfc}\left(\frac{x}{\sqrt{2}}\right)
\]

\section*{Examples The example below computes the Q function on a matrix, element by element. \\ ```
x = [0 1 2; 3 4 5]; \\ format short e % Switch to floating point format for displays. \\ y = qfunc(x) \\ format % Return to default format for displays.
```}

The output is below.
\(y=\)
\[
\begin{array}{ccc}
5.0000 \mathrm{e}-001 & 1.5866 \mathrm{e}-001 & 2.2750 \mathrm{e}-002 \\
1.3499 \mathrm{e}-003 & 3.1671 \mathrm{e}-005 & 2.8665 \mathrm{e}-007
\end{array}
\]

See Also
qfuncinv | erf | erfc | erfcx | erfinv | erfcinv

\section*{Purpose Inverse Q function}

\section*{Syntax \\ y = qfuncinv(x)}

Description \(\quad y=q f u n c i n v(x)\) returns the argument of the \(Q\) function at which the \(Q\) function's value is \(x\). The input \(x\) must be a real array with elements between 0 and 1 , inclusive.

For a scalar \(x\), the \(Q\) function is one minus the cumulative distribution function of the standardized normal random variable, evaluated at \(x\). The Q function is defined as
\[
Q(x)=\frac{1}{\sqrt{2 \pi}} \int_{x}^{\infty} \exp \left(-t^{2} / 2\right) d t
\]

The \(Q\) function is related to the complementary error function, erfc, according to
\[
Q(x)=\frac{1}{2} \operatorname{erfc}\left(\frac{x}{\sqrt{2}}\right)
\]

\section*{Examples}

The example below illustrates the inverse relationship between qfunc and qfuncinv.
```

x1 = [0 1 2; 3 4 5];
y1 = qfuncinv(qfunc(x1)) % Invert qfunc to recover x1.
x2 = 0:.2:1;
y2 = qfunc(qfuncinv(x2)) % Invert qfuncinv to recover x2.

```

The output is below.
y1 =
\begin{tabular}{lll}
0 & 1 & 2 \\
3 & 4 & 5
\end{tabular}
y2 =
\[
\begin{array}{llllll}
0 & 0.2000 & 0.4000 & 0.6000 & 0.8000 & 1.0000
\end{array}
\]

See Also
qfunc | erf | erfc | erfcx | erfinv | erfcinv

\section*{Purpose}

Produce quantization index and quantized output value

\section*{Syntax}
index = quantiz(sig,partition)
[index,quants] = quantiz(sig,partition,codebook)
[index,quants,distor] = quantiz(sig,partition, codebook)

\section*{Description}

\section*{Examples}
index = quantiz(sig, partition) returns the quantization levels in the real vector signal sig using the parameter partition. partition is a real vector whose entries are in strictly ascending order. If partition has length \(n\), index is a vector whose kth entry is
- 0 if \(\operatorname{sig}(k) \leq\) partition(1)
- m if partition(m) < sig(k) \(\leq\) partition \((m+1)\)
- n if partition(n) < sig(k)
[index, quants] = quantiz(sig, partition, codebook) is the same as the syntax above, except that codebook prescribes a value for each partition in the quantization and quants contains the quantization of sig based on the quantization levels and prescribed values. codebook is a vector whose length exceeds the length of partition by one. quants is a row vector whose length is the same as the length of sig. quants is related to codebook and index by
quants(ii) \(=\) codebook(index(ii)+1);
where ii is an integer between 1 and length(sig).
[index,quants,distor] = quantiz(sig,partition,codebook) is the same as the syntax above, except that distor estimates the mean square distortion of this quantization data set.

The command below rounds several numbers between 1 and 100 up to the nearest multiple of 10 . quants contains the rounded numbers, and index tells which quantization level each number is in.
[index,quants] = quantiz([3 348440 23],10:10:90,10:10:100)

\section*{quantiz}
The output is below.
index =
        \(\begin{array}{lllll}0 & 3 & 8 & 3 & 2\end{array}\)
quants =
        \(10 \quad 40 \quad 90 \quad 40 \quad 30\)
See Also lloyds | dpcmenco | dpcmdeco
How To . "Quantize a Signal"


\section*{Purpose Generate bit error patterns}
```

Syntax out = randerr(m)
out = randerr(m,n)
out = randerr(m,n,errors)
out = randerr(m,n,prob,state)
out = randerr(m,n,prob,s)

```

\section*{Description}

For all syntaxes, randerr treats each row of out independently. out \(=\) randerr(m) generates an m-by-m binary matrix, each row of which has exactly one nonzero entry in a random position. Each allowable configuration has an equal probability.
out \(=\) rander \((m, n)\) generates an \(m\)-by- \(n\) binary matrix, each row of which has exactly one nonzero entry in a random position. Each allowable configuration has an equal probability.
out \(=\) rander (m, n, errors) generates an m-by-n binary matrix, where errors determines how many nonzero entries are in each row:
- If errors is a scalar, it is the number of nonzero entries in each row.
- If errors is a row vector, it lists the possible number of nonzero entries in each row.
- If errors is a matrix having two rows, the first row lists the possible number of nonzero entries in each row and the second row lists the probabilities that correspond to the possible error counts.

Once randerr determines the number of nonzero entries in a given row, each configuration of that number of nonzero entries has equal probability.
out \(=\) randerr(m,n, prob, state) is the same as the syntax above, except that it first resets the state of the uniform random number generator rand to the integer state.

Note This usage is deprecated and may be removed in a future release. Instead of state, use s, as in the following example.

This function uses, by default, the Mersenne Twister algorithm by Nishimura and Matsumoto.

Note Using the state parameter causes this function to switch random generators to use the 'state' algorithm of the rand function.

See rand for details on the generator algorithm.
out = randerr(m,n, prob,s) causes rand to use the random stream s. See RandStream for more details.

\section*{Examples}

The examples below generate an 8 -by- 7 binary matrix, each row of which is equally likely to have either zero or two nonzero entries, and then alter the scenario by making it three times as likely that a row has two nonzero entries. Notice in the latter example that the second row of the error parameter sums to one.
out = randerr(8,7,[0 2])
out2 = randerr(8,7,[0 2; .25 .75])
Sample output is below.
out \(=\)
\begin{tabular}{lllllll}
0 & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 1 & 0 & 0 & 0 & 1 \\
1 & 0 & 1 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 0 & 0 & 0
\end{tabular}
\begin{tabular}{ccccccc}
0 & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & 0 & 0 & 0 & 1 & 1 & 0 \\
1 & 0 & 1 & 0 & 0 & 0 & 0 \\
& & & & & & \\
out2 \(=\) & & & & & & \\
& & & & & & \\
0 & 0 & 0 & 0 & 0 & 0 & 0 \\
1 & 0 & 0 & 0 & 0 & 0 & 1 \\
1 & 0 & 0 & 0 & 0 & 0 & 1 \\
0 & 0 & 0 & 1 & 0 & 1 & 0 \\
0 & 0 & 0 & 0 & 0 & 0 & 0 \\
0 & 1 & 0 & 0 & 0 & 0 & 1 \\
0 & 0 & 0 & 0 & 0 & 0 & 0 \\
1 & 0 & 0 & 0 & 1 & 0 & 0
\end{tabular}
See Also rand | randsrc | randint
How To

- "Sources and Sinks"

\section*{Purpose}

Generate matrix of uniformly distributed random integers

Note MathWorks will remove the randint function from a future version of the Communications System Toolbox software. We strongly recommend that you use randi instead.
```

Synfax out = randint
out = randint(m)
out $=$ randint ( $m, n$ )
out $=$ randint $(m, n, r g)$
out = randint(m,n,rg,state)

```

\section*{Description}
out \(=\) randint generates a random scalar that is either 0 or 1 , with equal probability.
out \(=\) randint ( \(m\) ) generates an \(m\)-by-m binary matrix, each of whose entries independently takes the value 0 with probability \(1 / 2\).
out \(=\) randint \((m, n)\) generates an \(m\)-by-n binary matrix, each of whose entries independently takes the value 0 with probability \(1 / 2\).
out \(=\) randint \((m, n, r g)\) generates an \(m\)-by- \(n\) integer matrix. If rg is zero, out is a zero matrix. Otherwise, the entries are uniformly distributed and independently chosen from the range
- \([0, \mathrm{rg}-1]\) if rg is a positive integer
- \([r g+1,0]\) if \(r g\) is a negative integer
- Between min and max, inclusive, if \(r g=[m i n, m a x]\) or [max, min]
out = randint( \(m, n, r g\), state) is the same as the syntax above, except that it first resets the state of the uniform random number generator rand to the integer state.

This function uses, by default, the Mersenne Twister algorithm by Nishimura and Matsumoto.

Note Using the state parameter causes this function to switch random generators to use the 'state' algorithm of the rand function.

See rand for details on the generator algorithm.

Examples To generate a 10 -by-10 matrix whose elements are uniformly distributed in the range from 0 to 7 , use either of the following commands.
out \(=\) randint( \(10,10,[0,7])\);
out = randint(10,10,8);

\section*{See Also \\ rand | randsrc | randerr}

How To . "Sources and Sinks"

\section*{Purpose Reorder symbols using random permutation}

Syntax intrlvd = randintrlv(data, state)
Description
intrlvd = randintrlv(data, state) rearranges the elements in data using a random permutation. The state parameter initializes the random number generator that the function uses to determine the permutation. state is either a scalar or a \(35 \times 1\) vector, and is described in the rand function, which is used in randintrlv. The function is predictable and invertible for a given state, but different states produce different permutations. If data is a matrix with multiple rows and columns, the function processes the columns independently.
This function uses, by default, the Mersenne Twister algorithm by Nishimura and Matsumoto.

Note Using the state parameter causes this function to switch random generators to use the 'state' algorithm of therand function.

See rand for details on the generator algorithm.

\section*{Examples}

See Also
How To . "Interleaving"

\section*{randseed}

Purpose Generate prime numbers for use as random number seeds

\author{
Syntax \\ \section*{Description}
}
```

out = randseed
out = randseed(state)
out = randseed(state,m)
out = randseed(state,m,n)
out = randseed(state,m,n,rmin)
out = randseed(state,m,n,rmin,rmax)

```

The randseed function produces random prime numbers that work well as seeds for random source blocks or noisy channel blocks in Communications System Toolbox software. It is recommended you use the randseed function when specifying the initial seed parameters of the following blocks: Gaussian, Rayleigh, and Rician Noise Generator.

Note The randseed function uses a local stream of numbers that is independent from the global stream of numbers in the MATLAB software. Use of this function does not affect the state of the global random number stream.
out \(=\) randseed generates a random prime number between 31 and \(2^{17}-1\), using the MATLAB function rand.
out \(=\) randseed(state) generates a random prime number after setting the state of rand to the positive integer state. This syntax produces the same output for a particular value of state.
out \(=\) randseed(state, \(m\) ) generates a column vector of \(m\) random primes.
out \(=\) randseed (state, \(m, n\) ) generates an \(m\)-by- \(n\) matrix of random primes.
out \(=\) randseed(state, \(m, n, r m i n)\) generates an \(m\)-by- \(n\) matrix of random primes between rmin and \(2^{17}-1\).
out \(=\) randseed(state, \(m, n, r m i n, r m a x)\) generates an \(m-b y-n\) matrix of random primes between rmin and rmax.

\section*{Examples}

To generate a two-element sample-based row vector of random bits using the Bernoulli Random Binary Generator block, you can set Probability of a zero to [0.1 0.5] and set Initial seed to randseed (391, 1, 2).

To generate three streams of random data from three different blocks in a single model, you can define out \(=\) randseed \((93,3)\) in the MATLAB workspace and then set the three blocks' Initial seed parameters to out(1), out (2), and out (3), respectively.

Purpose
Generate random matrix using prescribed alphabet

\author{
Syntax \\ \section*{Description}
}
out = randsrc
out \(=\) randsrc(m)
out \(=\) randsrc(m,n)
out \(=\) randsrc(m,n,alphabet)
out \(=\) randsrc(m,n,[alphabet; prob])
out \(=\) randsrc(m,n,...,state);
out \(=\) randsrc(m,n,...,s);
out \(=\) randsrc generates a random scalar that is either -1 or 1 , with equal probability.
out \(=\) randsrc( \(m\) ) generates an \(m\)-by-m matrix, each of whose entries independently takes the value -1 with probability \(1 / 2\), and 1 with probability \(1 / 2\).
out = randsrc(m,n) generates an m-by-n matrix, each of whose entries independently takes the value -1 with probability \(1 / 2\), and 1 with probability \(1 / 2\).
out = randsrc(m,n,alphabet) generates an m-by-n matrix, each of whose entries is independently chosen from the entries in the row vector alphabet. Each entry in alphabet occurs in out with equal probability. Duplicate values in alphabet are ignored.
out \(=\) randsrc( \(m, n,[\) alphabet; prob]) generates an m-by-n matrix, each of whose entries is independently chosen from the entries in the row vector alphabet. Duplicate values in alphabet are ignored. The row vector prob lists corresponding probabilities, so that the symbol alphabet ( \(k\) ) occurs with probability prob( \(k\) ), where \(k\) is any integer between one and the number of columns of alphabet. The elements of prob must add up to 1 .
out \(=\) randsrc( \(m, n, \ldots\), state \()\); is the same as the two preceding syntaxes, except that it first resets the state of the uniform random number generator rand to the integer state.

Note This usage is deprecated and may be removed in a future release. Instead of state, use s, as in the following example.

This function uses, by default, the Mersenne Twister algorithm by Nishimura and Matsumoto.

Note Using the state parameter causes this function to switch random generators to use the 'state' algorithm of the rand function.

See rand for details on the generator algorithm.
out = randsrc(m,n,...,s); causes rand to use the random stream s. See RandStream for more details.

\section*{Examples}

See Also

To generate a 10-by-10 matrix whose elements are uniformly distributed among members of the set \(\{-3,-1,1,3\}\), you can use either of these commands.
```

out = randsrc(10,10,[-3 -1 1 3]);

```
out = randsrc(10,10,[-3 -1 1 3; .25 . 25 . 25 .25]);

To skew the probability distribution so that -1 and 1 each occur with probability .3, while -3 and 3 each occur with probability .2, use this command.
```

out = randsrc(10,10,[-3 -1 1 3; .2 .3 .3 .2]);

```

Purpose Construct Rayleigh fading channel object
Syntax \(\quad \begin{aligned} \text { chan } & =\text { rayleighchan }(t s, f d) \\ \text { chan } & =\text { rayleighchan }(t s, f d, t a u, p d b) \\ \text { chan } & =\text { rayleighchan }\end{aligned}\)

\section*{Description}
chan = rayleighchan(ts,fd) constructs a frequency-flat ("single path") Rayleigh fading channel object. ts is the sample time of the input signal, in seconds. fd is the maximum Doppler shift, in hertz. You can model the effect of the channel on a signal \(x\) by using the syntax \(y=f i l t e r(c h a n, x)\).
chan = rayleighchan(ts,fd,tau,pdb) constructs a frequency-selective ("multiple path") fading channel object that models each discrete path as an independent Rayleigh fading process. tau is a vector of path delays, each specified in seconds. pdb is a vector of average path gains, each specified in \(d B\).
With the above two syntaxes, a smaller fd (a few hertz to a fraction of a hertz) leads to slower variations, and a larger fd (a couple hundred hertz) to faster variations.
chan = rayleighchan constructs a frequency-flat Rayleigh channel object with no Doppler shift. This is a static channel. The sample time of the input signal is irrelevant for frequency-flat static channels.

\section*{Properties}

The tables below describe the properties of the channel object, chan, that you can set and that MATLAB technical computing software sets automatically. To learn how to view or change the values of a channel object, see "Display Object Properties" or "Change Object Properties".

\section*{Writeable Properties}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline InputSamplePeriod & \begin{tabular}{l} 
Sample period of the signal on \\
which the channel acts, measured \\
in seconds.
\end{tabular} \\
\hline DopplerSpectrum & \begin{tabular}{l} 
Doppler spectrum object(s). The \\
default is a Jakes Doppler object.
\end{tabular} \\
\hline MaxDopplerShift & \begin{tabular}{l} 
Maximum Doppler shift of the \\
channel, in hertz (applies to all \\
paths of a channel).
\end{tabular} \\
\hline PathDelays & \begin{tabular}{l} 
Vector listing the delays of the \\
discrete paths, in seconds.
\end{tabular} \\
\hline AvgPathGaindB & \begin{tabular}{l} 
Vector listing the average gain of \\
the discrete paths, in decibels.
\end{tabular} \\
\hline NormalizePathGains & \begin{tabular}{l} 
If 1, the Rayleigh fading process \\
is normalized such that the \\
expected value of the path gains' \\
total power is 1.
\end{tabular} \\
\hline StoreHistory & \begin{tabular}{l} 
If this value is 1, channel \\
state information needed by \\
the channel visualization tool \\
is stored as the channel filter \\
function processes the signal. The \\
default value is 0.
\end{tabular} \\
\hline
\end{tabular}

Writeable Properties (Continued)
\begin{tabular}{l|l}
\hline Property & Description \\
\hline StorePathGains & \begin{tabular}{l} 
If set to 1, the complex path gain \\
vector is stored as the channel \\
filter function processes the \\
signal. The default value is 0.
\end{tabular} \\
\hline ResetBeforeFiltering & \begin{tabular}{l} 
If 1, each call to filter resets the \\
state of chan before filtering. If \\
0, the fading process maintains \\
continuity from one call to the \\
next.
\end{tabular} \\
\hline
\end{tabular}

\section*{Read-Only Properties}
\begin{tabular}{l|l|l}
\hline Property & Description & \begin{tabular}{l} 
When MATLAB \\
Sets or Updates \\
Value
\end{tabular} \\
\hline ChannelType & \begin{tabular}{l} 
Fixed value, \\
'Rayleigh'
\end{tabular} & \begin{tabular}{l} 
When you create \\
object
\end{tabular} \\
\hline PathGains & \begin{tabular}{l} 
Complex vector listing \\
the current gains of the \\
discrete paths. When \\
you create or reset chan, \\
PathGains is a random \\
vector influenced by \\
AvgPathGaindB and \\
NormalizePathGains.
\end{tabular} & \begin{tabular}{l} 
When you create \\
object, reset object, \\
or use it to filter a \\
signal
\end{tabular} \\
\hline
\end{tabular}

\section*{Read-Only Properties (Continued)}
\begin{tabular}{l|l|l}
\hline Property & Description & \begin{tabular}{l} 
When MATLAB \\
Sets or Updates \\
Value
\end{tabular} \\
\hline ChannelFilterDelay & \begin{tabular}{l} 
Delay of the channel \\
filter, measured in \\
samples. \\
The ChannelFilterDelay \\
property returns a delay \\
value that is valid only \\
if the first value of the \\
PathGain is the biggest \\
path gain. In other \\
words, main channel \\
energy is in the first \\
path.
\end{tabular} & \begin{tabular}{l} 
When you \\
create object or \\
change ratio of \\
InputSamplePeriod \\
to PathDelays
\end{tabular} \\
\hline NumSamplesProcessed & \begin{tabular}{l} 
Number of samples the \\
channel processed since \\
the last reset. When you \\
create or reset chan, this \\
property value is 0.
\end{tabular} & \begin{tabular}{l} 
When you create \\
object, reset object, \\
or use it to filter a \\
signal
\end{tabular} \\
\hline
\end{tabular}

\section*{Relationships Among Properties}

The PathDelays and AvgPathGaindB properties of the channel object must always have the same vector length, because this length equals the number of discrete paths of the channel. The DopplerSpectrum property must either be a single Doppler object or a vector of Doppler objects with the same length as PathDelays.

If you change the length of PathDelays, MATLAB truncates or zero-pads the value of AvgPathGaindB if necessary to adjust its vector length (MATLAB may also change the values of read-only properties such as PathGains and ChannelFilterDelay). If DopplerSpectrum is a vector of Doppler objects, and you increase or decrease the length of PathDelays, MATLAB will add Jakes Doppler objects or remove
elements from DopplerSpectrum, respectively, to make it the same length as PathDelays.
If StoreHistory is set to 1 (the default is 0 ), the object stores channel state information as the channel filter function processes the signal. You can then visualize this state information through a GUI using the plot (channel) method.

Note Setting StoreHistory to 1 will result in a slower simulation. If you do not want to visualize channel state information using plot (channel), but want to access the complex path gains, then set StorePathGains to 1 , while keeping StoreHistory as 0 .

\section*{Visualization of Channel}

The characteristics of a channel can be plotted using the channel visualization tool. You can use the channel visualization tool in Normal mode and Accelerator mode. For more information, see "Channel Visualization".

\section*{Examples Several examples using this function are in "Fading Channels".}

The example below illustrates that when you change the value of PathDelays, MATLAB automatically changes the values of other properties to make their vector lengths consistent with that of the new value of PathDelays.
```

c1 = rayleighchan(1e-5,130) % Create object.
c1.PathDelays = [0 1e-6] % Change the number of delays.
% MATLAB automatically changes the size of c1.AvgPathGaindB,
% c1.PathGains, and c1.ChannelFilterDelay.

```

The output below displays all the properties of the channel object before and after the change in the value of the PathDelays property. In the second listing of properties, the AvgPathGaindB, PathGains, and ChannelFilterDelay properties all have different values compared to the first listing of properties.
```

c1 =
InputSamplePeriod: 1.0000e-005
DopplerSpectrum: [1x1 doppler.jakes]
MaxDopplerShift: 130
PathDelays: 0
AvgPathGaindB: 0
NormalizePathGains: 1
StoreHistory: 0
PathGains: 0.2035 + 0.1014i
ChannelFilterDelay: 0
ResetBeforeFiltering: 1
NumSamplesProcessed: 0
c1 =
ChannelType: 'Rayleigh'
InputSamplePeriod: 1.0000e-005
DopplerSpectrum: [1x1 doppler.jakes]
MaxDopplerShift: 130
PathDelays: [0 1.0000e-006]
AvgPathGaindB: [0 0]
NormalizePathGains: 1
StoreHistory: 0
PathGains: [0.6108 - 0.4688i 0.1639 - 0.0027i]
ChannelFilterDelay: 4
ResetBeforeFiltering: 1
NumSamplesProcessed: 0

```

\section*{Algorithms}

The methodology used to simulate fading channels is described in "Methodology for Simulating Multipath Fading Channels:". The properties of the channel object are related to the quantities of the latter section as follows:
- The InputSamplePeriod property contains the value of \(T_{s}\).
- The PathDelays vector property contains the values of \(\left\{\tau_{k}\right\}\), where \(1 \leq k \leq K\).
- The PathGains read-only property contains the values of \(\left\{a_{k}\right\}\), where \(1 \leq k \leq K\).
- The AvgPathGaindB vector property contains the values of \(10 \log _{10}\left\{E\left[\left|a_{k}\right|^{2}\right]\right\}\), where \(1 \leq k \leq K\), and \(E[\cdot]\) denotes statistical
expectation.
- The ChannelFilterDelay read-only property contains the value of \(N_{1}\).

\title{
References \\ [1] Jeruchim, Michel C., Philip Balaban, and K. Sam Shanmugan, Simulation of Communication Systems, Second Edition, New York, Kluwer Academic/Plenum, 2000.
}

See Also ricianchan | filter | plot (channel) | reset
How To . "Fading Channels"

\section*{Purpose}

Design raised cosine finite impulse response (FIR) filter

Note MathWorks will remove the rcosfir function from a future version of the Communications System Toolbox software. While the product still supports this function, you should use fdesign. pulseshaping instead.
```

Syntax

```
```

b = rcosfir(R, n_T,rate,T)

```
b = rcosfir(R, n_T,rate,T)
b = rcosfir(R,n_T,rate,T,filter_type)
b = rcosfir(R,n_T,rate,T,filter_type)
rcosfir(...)
rcosfir(...)
rcosfir(...,colr)
rcosfir(...,colr)
[b,sample_time] = rcosfir(...)
```

[b,sample_time] = rcosfir(...)

```

\section*{Optional} Inputs
\begin{tabular}{l|l}
\hline Input & Default Value \\
\hline \(\mathrm{n}_{-} \mathrm{T}\) & 3 \\
\hline rate & 5 \\
\hline T & 1 \\
\hline
\end{tabular}

\section*{Description}

The rcosfir function designs the same filters that the rcosine function designs when the latter's type_flag argument includes 'fir'. However, rcosine is somewhat easier to use.

The time response of the raised cosine filter has the form
\[
h(t)=\frac{\sin (\pi t / T)}{(\pi t / T)} \cdot \frac{\cos (\pi R t / T)}{\left(1-4 R^{2} t^{2} / T^{2}\right)}
\]
b = rcosfir(R, n_T,rate, \(T\) ) designs a raised cosine filter and returns a vector \(b\) of length \(\left(n_{-} T(2)-n_{-} T(1)\right) *\) rate +1 . The filter's rolloff factor is \(R\), a real number between 0 and 1 , inclusive. \(T\) is the duration
of each bit in seconds. \(n_{-} T\) is a scalar or a vector of length 2 . If \(n \_T\) is specified as a scalar, the filter length is \(2 * n_{-} T+1\) input samples. If \(n_{-} T\) is a vector, it specifies the extent of the filter. In this case, the filter length is \(n_{-} T(2)-n_{-} T(1)+1\) input samples (or ( \(\left.n_{-} T(2)-n_{-} T(1)\right)\) *rate+1 output samples).
rate is the number of points in each input symbol period of length \(T\). rate must be greater than 1 . The input sample rate is T samples per second, while the output sample rate is T*rate samples per second.

The order of the FIR filter is
```

(n_T(2)-n_T(1))*rate

```

The arguments \(n_{-} T\), rate, and \(T\) are optional inputs whose default values are 3,5 , and 1 , respectively.
\(\mathrm{b}=\mathrm{rcosfir}\left(\mathrm{R}, \mathrm{n}_{-} \mathrm{T}\right.\), rate, T, filter_type) designs a square-root raised cosine filter if filter_type is 'sqrt'. If filter_type is 'normal', this syntax is the same as the previous one.

The impulse response of a square root raised cosine filter is
\[
h(t)=4 R \frac{\cos ((1+R) \pi t / T)+\frac{\sin ((1-R) \pi t / T)}{4 R \frac{t}{T}}}{\pi \sqrt{T}\left(1-(4 R t / T)^{2}\right)}
\]
rcosfir(...) produces plots of the time and frequency responses of the raised cosine filter.
rcosfir(..., colr) uses the string colr to determine the plotting color. The choices for colr are the same as those listed for the plot function.
[b, sample_time] = rcosfir(...) returns the FIR filter and its sample time.
Examples The commands below compare different rolloff factors.
rcosfir(0);
subplot(211); hold on;
subplot(212); hold on;
rcosfir(.5,[],[],[],[],'r-');
rcosfir(1,[],[],[],[],'g-');
References [1] Korn, Israel, Digital Communications, New York, Van Nostrand Reinhold, 1985.
See Also rcosiir | rcosflt | rcosine | firrcos
How To - rcosdemo- "Filtering"

Purpose Filter input signal using raised cosine filter
```

Syntax

```
```

y = rcosflt(x,Fd,Fs)

```
y = rcosflt(x,Fd,Fs)
y = rcosflt(x,Fd,Fs,'type_flag',r,delay,tol)
y = rcosflt(x,Fd,Fs,'type_flag',r,delay,tol)
y = rcosflt(x,Fd,Fs,'filter_type/Fs',r,delay,tol)
y = rcosflt(x,Fd,Fs,'filter_type/Fs',r,delay,tol)
y = rcosflt(x,Fd,Fs,'filter_type/filter',num,den)
y = rcosflt(x,Fd,Fs,'filter_type/filter',num,den)
y = rcosflt(x,Fd,Fs,'filter_type/filter',num,den,delay)
y = rcosflt(x,Fd,Fs,'filter_type/filter',num,den,delay)
y = rcosflt(x,Fd,Fs,'filter_type/filter/Fs',num,den...)
y = rcosflt(x,Fd,Fs,'filter_type/filter/Fs',num,den...)
[y,t] = rcosflt(...)
```

[y,t] = rcosflt(...)

```

Note MathWorks will remove the rcosflt function from a future version of the Communications System Toolbox software. We strongly recommend that you use fdesign.pulseshaping instead.

\section*{Optional}

Inputs
\begin{tabular}{l|l}
\hline Input & Default Value \\
\hline filter_type & fir/normal \\
\hline\(r\) & 0.5 \\
\hline delay & 3 \\
\hline tol & 0.01 \\
\hline den & 1 \\
\hline
\end{tabular}

\section*{Description}

The function rcosflt passes an input signal through a raised cosine filter. You can either let rcosflt design a raised cosine filter automatically or you can specify the raised cosine filter yourself using input arguments.

\section*{Designing the Filter Automatically}
\(y=r \operatorname{cosflt}(x, F d, F s)\) designs a raised cosine FIR filter and then filters the input signal \(x\) using it. The sample frequency for the digital input signal x is Fd , and the sample frequency for the output signal y
is Fs. The ratio Fs/Fd must be an integer. In the course of filtering, rcosflt upsamples the data by a factor of \(\mathrm{Fs} / \mathrm{Fd}\), by inserting zeros between samples. The order of the filter is \(1+2 *\) delay*Fs/Fd, where delay is 3 by default. If \(x\) is a vector, then the sizes of \(x\) and \(y\) are related by this equation.
```

length(y) = (length(x) + 2 * delay)*Fs/Fd

```

Otherwise, y is a matrix, each of whose columns is the result of filtering the corresponding column of x .
y = rcosflt(x,Fd,Fs,'type_flag',r,delay,tol) designs a raised cosine FIR or IIR filter and then filters the input signal \(x\) using it. The ratio \(\mathrm{Fs} / \mathrm{Fd}\) must be an integer. r is the rolloff factor for the filter, a real number in the range \([0,1]\). delay is the filter's group delay, measured in input samples. The actual group delay in the filter design is delay/Fd seconds. The input tol is the tolerance in the IIR filter design. FIR filter design does not use tol.

The characteristics of \(x, F d, F s\), and \(y\) are as in the first syntax.
The fourth input argument, 'type_flag', determines the type of filter that rcosflt should design and can have up to three components: filter type, sample frequency, and filter.

Values of filter_type to Determine the Type of Filter
\begin{tabular}{l|l}
\hline Type of Filter & Value of filter_type \\
\hline FIR raised cosine filter & fir or fir/normal \\
\hline IIR raised cosine filter & iir or iir/normal \\
\hline \begin{tabular}{l} 
Square-root FIR raised cosine \\
filter
\end{tabular} & fir/sqrt \\
\hline \begin{tabular}{l} 
Square-root IIR raised cosine \\
filter
\end{tabular} & iir/sqrt \\
\hline
\end{tabular}
\(y=r c o s f l t\left(x, F d, F s, ' f i l t e r \_t y p e / F s ', r, d e l a y, t o l\right)\) is the same as the previous syntax, except that it assumes that \(x\) has sample frequency Fs. This syntax does not upsample \(x\) any further. If \(x\) is a vector, then the relative sizes of \(x\) and \(y\) are related by this equation.
length(y) \(=\) length \((x)+(2\) * delay * Fs/Fd)
As before, if x is a nonvector matrix, y is a matrix, each of whose columns is the result of filtering the corresponding column of \(x\).

\section*{Specifying the Filter Using Input Arguments}
y = rcosflt(x,Fd,Fs,'filter_type/filter', num, den) filters the input signal \(x\) using a filter whose transfer function numerator and denominator are given in num and den, respectively. If type_filter includes fir, then omit den. This syntax uses the same arguments x, Fd, Fs, and type_filter as explained in the first and second syntaxes above.
y = rcosflt(x,Fd,Fs,'filter_type/filter',num, den, delay) uses delay in the same way that the rcosine function uses it. This syntax assumes that the filter described by num, den, and delay was designed using rcosine.

As before, if x is a nonvector matrix, y is a matrix each of whose columns is the result of filtering the corresponding column of \(x\).
\(y=r c o s f l t\left(x, F d, F s, ' f i l t e r \_t y p e / f i l t e r / F s ', n u m, d e n . ..\right)\) is the same as the earlier syntaxes, except that it assumes that \(x\) has sample frequency Fs instead of Fd. This syntax does not upsample \(x\) any further. If \(x\) is a vector, the relative sizes of \(x\) and \(y\) are related by this equation.
```

length(y) = length(x) + (2 * delay * Fs/Fd)

```

\section*{Additional Output}
[y,t] = rcosflt(...) outputs \(t\), a vector that contains the sampling time points of \(y\).

\author{
References Reinhold, 1985. \\ See Also rcosine | rcosfir | rcosiir \\ How To \\ - rcosdemo \\ - "Filtering"
}
[1] Korn, Israel, Digital Communications, New York, Van Nostrand

Purpose Design raised cosine infinite impulse response (IIR) filter
```

Syntax

```
```

[num,den] = rcosiir(R,T_delay,rate,T,tol)

```
[num,den] = rcosiir(R,T_delay,rate,T,tol)
[num,den] = rcosiir(R,T_delay,rate,T,tol,type_filter)
[num,den] = rcosiir(R,T_delay,rate,T,tol,type_filter)
rcosiir(...)
rcosiir(...)
rcosiir(...,colr)
rcosiir(...,colr)
[num,den,sample_time] = rcosiir(...)
```

[num,den,sample_time] = rcosiir(...)

```

Note MathWorks will remove the rcosiir function from a future version of the Communications System Toolbox software.

\section*{Optional Inputs}
\begin{tabular}{l|l}
\hline Input & Default Value \\
\hline T_delay & 3 \\
\hline rate & 5 \\
\hline T & 1 \\
\hline tol & 0.01 \\
\hline
\end{tabular}

Description
The rcosiir function designs the same filters that the rcosine function designs when the latter's type_flag argument includes 'iir'. However, rcosine is somewhat easier to use.

The time response of the raised cosine filter has the form
\[
h(t)=\frac{\sin (\pi t / T)}{(\pi t / T)} \cdot \frac{\cos (\pi R t / T)}{\left(1-4 R^{2} t^{2} / T^{2}\right)}
\]
[num, den] = rcosiir(R,T_delay, rate, T,tol) designs an IIR approximation of an FIR raised cosine filter, and returns the numerator and denominator of the IIR filter. The filter's rolloff factor is R, a real number between 0 and 1 , inclusive. \(T\) is the symbol period in seconds.

The filter's group delay is T_delay symbol periods. rate is the number of sample points in each interval of duration \(T\). rate must be greater than 1 . The input sample rate is \(T\) samples per second, while the output sample rate is \(T^{*}\) rate samples per second. If tol is an integer greater than 1 , it becomes the order of the IIR filter; if tol is less than 1 , it indicates the relative tolerance for rcosiir to use when selecting the order based on the singular values.

The arguments T_delay, rate, T, and tol are optional inputs whose default values are \(3,5,1\), and 0.01 , respectively.
[num,den] = rcosiir(R,T_delay,rate,T,tol,type_filter) designs a square-root raised cosine filter if type_filter is 'sqrt'. If type_filter is 'normal', this syntax is the same as the previous one.
rcosiir(...) plots the time and frequency responses of the raised cosine filter.
rcosiir(..., colr) uses the string colr to determine the plotting color. The choices for colr are the same as those listed for the plot function.
[num,den, sample_time] = rcosiir(...) returns the transfer function and the sample time of the IIR filter.

\section*{Examples}

The script below compares different values of T_delay.
```

rcosiir(0,10);
subplot(211); hold on;
subplot(212); hold on;
col = ['r-';'g-';'b-';'m-';'c-';'w-'];
R = [8,6,4,3,2,1];
for ii = R
rcosiir(0,ii,[],[],[],[],col(find(R==ii),:));
end;

```

This example shows how the filter's frequency response more closely approximates that of the ideal raised cosine filter as T_delay increases.

\section*{rcosiir}
\begin{tabular}{ll} 
References & \begin{tabular}{l} 
[1] Kailath, Thomas, Linear Systems, Englewood Cliffs, N.J., \\
Prentice-Hall, 1980.
\end{tabular} \\
& \begin{tabular}{l} 
[2] Korn, Israel, Digital Communications, New York, Van Nostrand \\
Reinhold, 1985.
\end{tabular} \\
See Also & rcosfir | rcosflt | rcosine \\
How To & - rcosdemo
\end{tabular}

\section*{Purpose Design raised cosine filter}

\section*{Syntax}
```

num = rcosine(Fd,Fs)
[num,den] = rcosine(Fd,Fs,type_flag)
[num,den] = rcosine(Fd,Fs,type_flag,r)
[num,den] = rcosine(Fd,Fs,type_flag,r,delay)
[num,den] = rcosine(Fd,Fs,type_flag,r,delay,tol)

```

Note MathWorks will remove the rcosine function from a future version of the Communications System Toolbox software. We strongly recommend that you use fdesign.pulseshaping instead.

\section*{Description}
num = rcosine(Fd,Fs) designs a finite impulse response (FIR) raised cosine filter and returns its transfer function. The digital input signal has sampling frequency Fd. The sampling frequency for the filter is Fs. The ratio Fs/Fd must be a positive integer greater than 1 . The default rolloff factor is .5 . The filter's group delay, which is the time between the input to the filter and the filter's peak response, is three input samples. Equivalently, the group delay is \(3 /\) Fd seconds.
[num,den] = rcosine(Fd,Fs,type_flag) designs a raised cosine filter using directions in the string variable type_flag. Filter types are listed in the table below, along with the corresponding values of type_flag.

Types of Filter and Corresponding Values of type_flag
\begin{tabular}{l|l}
\hline Type of Filter & Value of type_flag \\
\hline Finite impulse response (FIR) & 'default' or 'fir/normal' \\
\hline Infinite impulse response (IIR) & 'iir' or ''iir/normal'' \\
\hline Square-root raised cosine FIR & 'sqrt' or 'fir/sqrt' \\
\hline Square-root raised cosine IIR & 'iir/sqrt' \\
\hline
\end{tabular}

The default tolerance value in IIR filter design is 0.01 .
[num,den] = rcosine(Fd,Fs,type_flag,r) specifies the rolloff factor, \(r\). The rolloff factor is a real number in the range \([0,1]\).
[num,den] = rcosine(Fd,Fs,type_flag,r,delay) specifies the filter's group delay, measured in input samples. delay is a positive integer. The actual group delay in the filter design is delay/Fd seconds.
[num,den] = rcosine(Fd,Fs,type_flag,r,delay,tol) specifies the tolerance in the IIR filter design. FIR filter design does not use tol.

\author{
References [1] Korn, Israel, Digital Communications, New York, Van Nostrand Reinhold, 1985.
}
```

See Also rcosflt | rcosiir | rcosfir

```
How To • rcosdemo
- "Filtering"
\[
\begin{array}{ll}
\text { Purpose } & \text { Rectangular pulse shaping } \\
\text { Syntax } & y=\text { rectpulse ( } x, n \text { nsamp) } \\
\text { Description } & \begin{array}{l}
y=\text { rectpulse }(x, n s a m p) \text { applies rectangular pulse shaping to } \\
\\
\text { x to produce an output signal having nsamp samples per symbol. } \\
\text { Rectangular pulse shaping means that each symbol from } x \text { is repeated } \\
\text { nsamp times to form the output } y . ~ I f ~ \\
\text { the is a matrix with multiple rows, } \\
\text { independently. }
\end{array}
\end{array}
\]

Note To insert zeros between successive samples of \(x\) instead of repeating the samples of \(x\), use the upsample function instead.

\section*{Examples \\ An example in "Combine Pulse Shaping and Filtering with Modulation" uses this function in conjunction with modulation.}

The code below processes two independent channels, each containing three symbols of data. In the pulse-shaped matrix \(y\), each symbol contains four samples.
```

nsamp = 4; % Number of samples per symbol
nsymb = 3; % Number of symbols
s = RandStream('mt19937ar', 'Seed', 0);
ch1 = randi(s, [0 1], nsymb, 1); % Random binary channel
ch2 = [1:nsymb]';
x = [ch1 ch2] % Two-channel signal
y = rectpulse(x,nsamp)

```

The output is below. In \(y\), each column corresponds to one channel and each row corresponds to one sample. Also, the first four rows of y correspond to the first symbol, the next four rows of \(y\) correspond to the second symbol, and the last four rows of y correspond to the last symbol.

\section*{rectpulse}
\(y=\)\begin{tabular}{lll}
1 & 1 \\
1 & 2 \\
0 & 3 \\
& & \\
& \\
1 & \\
1 & 1 \\
1 & 1 \\
1 & 1 \\
1 & 1 \\
1 & 2 \\
1 & 2 \\
1 & 2 \\
0 & 2 \\
0 & 3 \\
0 & 3 \\
0 & 3
\end{tabular}

See Also intdump | upsample | rcosflt

\section*{Purpose Reset channel object}
```

Syntax reset(chan)
reset(chan,randstate)

```

Description
reset (chan) resets the channel object chan, initializing the PathGains and NumSamplesProcessed properties as well as internal filter states. This syntax is useful when you want the effect of creating a new channel.
reset (chan, randstate) resets the channel object chan and initializes the state of the random number generator that the channel uses. randstate is a two-element column vector. This syntax is useful when you want to repeat previous numerical results that started from a particular state.

Note reset(chan, randstate) will not support randstate in a future release. See the legacychannelsim function for more information.

Examples
The example below shows how to obtain repeatable results. The example chooses a state for the random number generator immediately after defining the channel object and later resets the random number generator to that state.
\% Set up channel.
\% Assume you want to maintain continuity
\% from one filtering operation to the next, except
\% when you explicitly reset the channel.
c = rayleighchan(1e-4,100);
c. ResetBeforeFiltering = 0;
\% Filter all ones.
sig = ones(100,1);
y1 = [filter(c,sig(1:50)); filter(c,sig(51:end))];
```

% Reset the channel and filter all ones.
reset(c); %Generate an independent channel
y2 = [filter(c,sig(1:50)); filter(c,sig(51:end))];
% Plot the magnitude of the channel output
plot(abs([y1; y2]),'*')
grid on

```

This example generates the following figure.


See Also
rayleighchan | ricianchan | filter
How To
- "Fading Channels"
\begin{tabular}{ll} 
Purpose & Reset equalizer object \\
Syntax & reset (eqobj) \\
Description & \begin{tabular}{l} 
reset (eqobj) resets the equalizer object eqobj, initializing the \\
Weights, WeightInputs, and NumSamplesProcessed properties and the \\
adaptive algorithm states. If eqobj is a CMA equalizer, reset does \\
not change the Weights property.
\end{tabular} \\
See Also & \begin{tabular}{l} 
dfe | equalize | lineareq
\end{tabular} \\
How To & . "Equalization"
\end{tabular}

\author{
Purpose \\ Syntax \\ \section*{Description}
}

Construct Rician fading channel object
chan = ricianchan(ts,fd,k)
chan \(=\) ricianchan(ts,fd,k,tau, pdb)
chan = ricianchan(ts,fd,k,tau,pdb,fdLOS)
chan = ricianchan
chan \(=\) ricianchan(ts,fd,k) constructs a frequency-flat (single path) Rician fading-channel object. ts is the sample time of the input signal, in seconds. fd is the maximum Doppler shift, in hertz. k is the Rician K -factor in linear scale. You can model the effect of the channel chan on a signal \(x\) by using the syntax \(y=\) filter (chan, \(x\) ). See filter for more information.
chan \(=\) ricianchan(ts,fd,k,tau,pdb) constructs a frequency-selective (multiple paths) fading-channel object. If \(k\) is a scalar, then the first discrete path is a Rician fading process (it contains a line-of-sight component) with a K-factor of \(k\), while the remaining discrete paths are independent Rayleigh fading processes (no line-of-sight component). If \(k\) is a vector of the same size as tau, then each discrete path is a Rician fading process with a K-factor given by the corresponding element of the vector \(k\). tau is a vector of path delays, each specified in seconds. pdb is a vector of average path gains, each specified in dB.
chan \(=\) ricianchan(ts,fd,k,tau, pdb,fdLOS) specifies fdlos as the Doppler shift(s) of the line-of-sight component(s) of the discrete path(s), in hertz. fdlos must be the same size as \(k\). If \(k\) and fdlos are scalars, the line-of-sight component of the first discrete path has a Doppler shift of fdlos, while the remaining discrete paths are independent Rayleigh fading processes. If fdlos is a vector of the same size as \(k\), the line-of-sight component of each discrete path has a Doppler shift given by the corresponding element of the vector fdlos. By default, fdlos is 0 . The initial phase(s) of the line-of-sight component(s) can be set through the property DirectPathInitPhase.
chan \(=\) ricianchan sets the maximum Doppler shift to 0 , the Rician K-factor to 1, and the Doppler shift and initial phase of the line-of-sight
component to 0 . This syntax models a static frequency-flat channel, and, in this trivial case, the sample time of the signal is unimportant.

\section*{Properties}

The following tables describe the properties of the channel object, chan, that you can set and that MATLAB technical computing software sets automatically. To learn how to view or change the values of a channel object, see "Display Object Properties" or "Change Object Properties".

\section*{Writeable Properties}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline InputSamplePeriod & \begin{tabular}{l} 
Sample period of the signal on \\
which the channel acts, measured \\
in seconds.
\end{tabular} \\
\hline DopplerSpectrum & \begin{tabular}{l} 
Doppler spectrum object(s). The \\
default is a Jakes doppler object.
\end{tabular} \\
\hline MaxDopplerShift & \begin{tabular}{l} 
Maximum Doppler shift of the \\
channel, in hertz (applies to all \\
paths of a channel).
\end{tabular} \\
\hline KFactor & \begin{tabular}{l} 
Rician K-factor (scalar or \\
vector). The default value is 1 \\
(line-of-sight component on the \\
first path only).
\end{tabular} \\
\hline PathDelays & \begin{tabular}{l} 
Vector listing the delays of the \\
discrete paths, in seconds.
\end{tabular} \\
\hline AvgPathGaindB & \begin{tabular}{l} 
Vector listing the average gain of \\
the discrete paths, in decibels.
\end{tabular} \\
\hline DirectPathDopplerShift & \begin{tabular}{l} 
Doppler shift(s) of the line-of-sight \\
component(s) in hertz. The \\
default value is 0.
\end{tabular} \\
\hline
\end{tabular}

\section*{Writeable Properties (Continued)}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline DirectPathInitPhase & \begin{tabular}{l} 
Initial phase(s) of line-of-sight \\
component(s) in radians. The \\
default value is 0.
\end{tabular} \\
\hline NormalizePathGains & \begin{tabular}{l} 
If this value is 1, the Rayleigh \\
fading process is normalized such \\
that the expected value of the \\
path gains' total power is 1.
\end{tabular} \\
\hline StoreHistory & \begin{tabular}{l} 
If this value is 1, channel \\
state information needed by \\
the channel visualization tool \\
is stored as the channel filter \\
function processes the signal. The \\
default value is 0.
\end{tabular} \\
\hline StorePathGains & \begin{tabular}{l} 
If this value is 1, the complex \\
path gain vector is stored as the \\
channel filter function processes \\
the signal. The default value is 0.
\end{tabular} \\
\hline ResetBeforeFiltering & \begin{tabular}{l} 
If this value is 1, each call \\
to filter resets the state of \\
chan before filtering. If it is 0, \\
the fading process maintains \\
continuity from one call to the \\
next.
\end{tabular} \\
\hline
\end{tabular}

\section*{Read-Only Properties}
\begin{tabular}{l|l|l}
\hline Property & Description & \begin{tabular}{l} 
When MATLAB \\
Sets or Updates \\
Value
\end{tabular} \\
\hline ChannelType & Fixed value, 'Rician '. & \begin{tabular}{l} 
When you create \\
object.
\end{tabular} \\
\hline PathGains & \begin{tabular}{l} 
Complex vector listing \\
the current gains of the \\
discrete paths. When \\
you create or reset chan, \\
PathGains is a random \\
vector influenced by \\
AvgPathGaindB and \\
NormalizePathGains.
\end{tabular} & \begin{tabular}{l} 
When you create \\
object, reset object, \\
or use it to filter a \\
signal.
\end{tabular} \\
\hline ChannelFilterDelay & \begin{tabular}{l} 
Delay of the channel \\
filter, measured in \\
samples. \\
The ChannelFilterDelay \\
property returns a delay \\
value that is valid only \\
if the first value of the \\
PathGain is the biggest \\
path gain. In other \\
words, main channel \\
energy is in the first \\
path.
\end{tabular} & \begin{tabular}{l} 
When you \\
create object or \\
change ratio of \\
InputSamplePeriod \\
to PathDelays.
\end{tabular} \\
\hline NumSamplesProcessed & \begin{tabular}{l} 
Number of samples the \\
channel processed since \\
the last reset. When you \\
create or reset chan, this \\
property value is 0.
\end{tabular} & \begin{tabular}{l} 
When you create \\
object, reset object, \\
or use it to filter a \\
signal.
\end{tabular} \\
\hline
\end{tabular}

\section*{Relationships Among Properties}

Changing the length of PathDelays also changes the length of AvgPathGaindB, the length of KFactor if KFactor is a vector (no change if it is a scalar), and the length of DopplerSpectrum if DopplerSpectrum is a vector (no change if it is a single object).
DirectPathDopplerShift and DirectPathInitPhase both follow changes in KFactor.

The PathDelays and AvgPathGaindB properties of the channel object must always have the same vector length, because this length equals the number of discrete paths of the channel. The DopplerSpectrum property must either be a single Doppler object or a vector of Doppler objects with the same length as PathDelays.
If you change the length of PathDelays, MATLAB truncates or zero-pads the value of AvgPathGaindB if necessary to adjust its vector length (MATLAB may also change the values of read-only properties such as PathGains and ChannelFilterDelay). If DopplerSpectrum is a vector of Doppler objects, and you increase or decrease the length of PathDelays, MATLAB will add Jakes Doppler objects or remove elements from DopplerSpectrum, respectively, to make it the same length as PathDelays.

If StoreHistory is set to 1 (the default is 0 ), the object stores channel state information as the channel filter function processes the signal. You can then visualize this state information through a GUI using the plot (channel) method.

Note Setting StoreHistory to 1 will result in a slower simulation. If you do not want to visualize channel state information using plot (channel), but want to access the complex path gains, then set StorePathGains to 1 , while keeping StoreHistory as 0 .

\section*{Reset Method}

If MaxDopplerShift is set to 0 (the default), the channel object, chan, models a static channel.

Use the syntax reset (chan) to generate a new channel realization.

\section*{Algorithm}

The methodology used to simulate fading channels is described in "Methodology for Simulating Multipath Fading Channels:", where the properties specific to the Rician channel object are related to the quantities of this section as follows (see the rayleighchan reference page for properties common to both Rayleigh and Rician channel objects):
- The Kfactor property contains the value of \(K_{r}\) (if it's a scalar) or \(\left\{K_{r, k}\right\}, 1 \leq k \leq K\) (if it's a vector).
- The DirectPathDopplerShift property contains the value of \(f_{d, L O S}\)
(if it's a scalar) or \(\left\{f_{d, L O S, k}\right\}, 1 \leq k \leq K\) (if it's a vector).
- The DirectPathInitPhase property contains the value of \(\theta_{\text {LOS }}\) (if
it's a scalar) or \(\left\{\theta_{L O S, k}\right\}, 1 \leq k \leq K\) (if it's a vector).

\section*{Channel Visualization}

The characteristics of a channel can be plotted using the channel visualization tool. You can use the channel visualization tool in Normal mode and Accelerator mode. For more information, see "Channel Visualization".

\section*{Examples}

References
The example in "Quasi-Static Channel Modeling" uses this function.
[1] Jeruchim, M., Balaban, P., and Shanmugan, K., Simulation of Communication Systems, Second Edition, New York, Kluwer Academic/Plenum, 2000.

See Also rayleighchan | filter | plot (channel) | reset
How To . "Fading Channels"

\section*{Purpose}

Construct recursive least squares (RLS) adaptive algorithm object

\section*{Syntax}

Description
```

alg = rls(forgetfactor)
alg = rls(forgetfactor,invcorr0)

```

The rls function creates an adaptive algorithm object that you can use with the lineareq function or dfe function to create an equalizer object. You can then use the equalizer object with the equalize function to equalize a signal. To learn more about the process for equalizing a signal, see "Adaptive Algorithms".
alg = rls(forgetfactor) constructs an adaptive algorithm object based on the recursive least squares (RLS) algorithm. The forgetting factor is forgetfactor, a real number between 0 and 1 . The inverse correlation matrix is initialized to a scalar value.
alg = rls(forgetfactor, invcorr0) sets the initialization parameter for the inverse correlation matrix. This scalar value is used to initialize or reset the diagonal elements of the inverse correlation matrix.

\section*{Properties}

The table below describes the properties of the RLS adaptive algorithm object. To learn how to view or change the values of an adaptive algorithm object, see "Access Properties of an Adaptive Algorithm".
\begin{tabular}{l|l}
\hline Property & Description \\
\hline AlgType & Fixed value, 'RLS' \\
\hline ForgetFactor & Forgetting factor \\
\hline InvCorrInit & \begin{tabular}{l} 
Scalar value used to initialize or \\
reset the diagonal elements of the \\
inverse correlation matrix
\end{tabular} \\
\hline
\end{tabular}

Also, when you use this adaptive algorithm object to create an equalizer object (via the lineareq function or dfe function), the equalizer object has an InvCorrMatrix property that represents the inverse correlation
matrix for the RLS algorithm. The initial value of InvCorrMatrix is InvCorrInit*eye( \(N\) ), where \(N\) is the total number of equalizer weights.

\section*{Examples}

\section*{Algorithms}

\section*{References}

For examples that use this function, see "Defining an Equalizer Object" and "Example: Adaptive Equalization Within a Loop".

Referring to the schematics presented in "Equalizer Structure", define \(w\) as the vector of all weights \(w_{\mathrm{i}}\) and define u as the vector of all inputs \(u_{i}\). Based on the current set of inputs, u , and the current inverse correlation matrix, \(P\), this adaptive algorithm first computes the Kalman gain vector, \(K\)
\[
K=\frac{P u}{(\text { ForgetFactor })+u^{H} P u}
\]
where \(H\) denotes the Hermitian transpose.
Then the new inverse correlation matrix is given by
\[
(\text { ForgetFactor })^{-1}\left(\mathrm{P}-\mathrm{Ku}^{\mathrm{H}} \mathrm{P}\right)
\]
and the new set of weights is given by
\[
w+K^{\star} \mathrm{e}
\]
where the * operator denotes the complex conjugate.
[1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, John Wiley \& Sons, 1998.
[2] Haykin, S., Adaptive Filter Theory, Third Ed., Upper Saddle River, NJ, Prentice-Hall, 1996.
[3] Kurzweil, J., An Introduction to Digital Communications, New York, John Wiley \& Sons, 2000.
[4] Proakis, John G., Digital Communications, Fourth Ed., New York, McGraw-Hill, 2001.
\begin{tabular}{ll} 
See Also & lms | signlms | normlms | varlms | lineareq | dfe | equalize \\
How To & - "Equalization"
\end{tabular}
```

Purpose
Reed-Solomon decoder
Syntax
decoded $=$ rsdec (code, n,k)
decoded $=$ rsdec(code, n, k,genpoly)
decoded $=$ rsdec(..., paritypos)
[decoded,cnumerr] = rsdec(...)
[decoded,cnumerr, ccode] = rsdec(...)

```

\section*{Description}
decoded \(=\) rsdec (code, \(\mathrm{n}, \mathrm{k}\) ) attempts to decode the received signal in code using an [n,k] Reed-Solomon decoding process with the narrow-sense generator polynomial. code is a Galois array of symbols having \(m\) bits each. Each \(n\)-element row of code represents a corrupted systematic codeword, where the parity symbols are at the end and the leftmost symbol is the most significant symbol. \(n\) is at most \(2^{m}-1\). If \(n\) is not exactly \(2^{\mathrm{m}}-1\), rsdec assumes that code is a corrupted version of a shortened code.

In the Galois array decoded, each row represents the attempt at decoding the corresponding row in code. A decoding failure occurs if rsdec detects more than ( \(\mathrm{n}-\mathrm{k}\) )/2 errors in a row of code. In this case, \(r s d e c\) forms the corresponding row of decoded by merely removing \(n-k\) symbols from the end of the row of code.
decoded \(=r\) sdec (code \(, n, k\), genpoly) is the same as the syntax above, except that a nonempty value of genpoly specifies the generator polynomial for the code. In this case, genpoly is a Galois row vector that lists the coefficients, in order of descending powers, of the generator polynomial. The generator polynomial must have degree n-k. To use the default narrow-sense generator polynomial, set genpoly to [].
decoded \(=\) rsdec (..., paritypos) specifies whether the parity symbols in code were appended or prepended to the message in the coding operation. The string paritypos can be either 'end' or 'beginning'. The default is 'end'. If paritypos is 'beginning', a decoding failure causes rsdec to remove \(\mathrm{n}-\mathrm{k}\) symbols from the beginning rather than the end of the row.
[decoded, cnumerr] = rsdec (...) returns a column vector cnumerr, each element of which is the number of corrected errors in the
corresponding row of code. A value of -1 in cnumerr indicates a decoding failure in that row in code.
[decoded, cnumerr,ccode] = rsdec(...) returns ccode, the corrected version of code. The Galois array ccode has the same format as code. If a decoding failure occurs in a certain row of code, the corresponding row in ccode contains that row unchanged.

\section*{Examples}

The example below encodes three message words using a \((7,3)\) Reed-Solomon encoder. It then corrupts the code by introducing one error in the first codeword, two errors in the second codeword, and three errors in the third codeword. Then rsdec tries to decode the corrupted code.
```

m = 3; % Number of bits per symbol
n = 2^m-1; k = 3; % Word lengths for code
msg = gf([2 7 3; 4 0 6; 5 1 1],m); % Three rows of m-bit symbols
code = rsenc(msg,n,k);
errors = gf([2 0 0 0 0 0 0; 3 4 0 0 0 0 0; 5 6 7 0 0 0 0],m);
noisycode = code + errors;
[dec,cnumerr] = rsdec(noisycode,n,k)

```

The output is below.
```

dec = GF(2^3) array. Primitive polynomial = D^3+D+1 (11 decimal)

```
Array elements =
\begin{tabular}{lll}
2 & 7 & 3 \\
4 & 0 & 6 \\
0 & 7 & 6
\end{tabular}
cnumerr =1
2
-1

The output shows that rsdec successfully corrects the errors in the first two codewords and recovers the first two original message words. However, a (7,3) Reed-Solomon code can correct at most two errors in each word, so rsdec cannot recover the third message word. The elements of the vector cnumerr indicate the number of corrected errors in the first two words and also indicate the decoding failure in the third word.

For additional examples, see "Create and Decode Reed-Solomon Codes".

Limitations
n and k must differ by an even integer. n must be between 3 and 65535 .
rsdec uses the Berlekamp-Massey decoding algorithm. For information about this algorithm, see the works listed in "References" on page 1-482 below.

\author{
References
}

See Also rsenc \| gf \| rsgenpoly
How To . "Block Codes"
Purpose Decode ASCII file encoded using Reed-Solomon code
Syntax

rsdecof(file_in,file_out); rsdecof(file_in,file_out,err_cor);
Description
WarningThis function is obsolete and may be removed in the future. Westrongly recommend that you use the comm. RSDecoder Systemobject instead.
This function is the inverse process of the function rsencof in that it decodes a file that rsencof encoded.
rsdecof(file_in,file_out) decodes the ASCII file file_in that was previously created by the function rsencof using an error-correction capability of 5 . The decoded message is written to file_out. Both file_in and file_out are string variables.
Note If the number of characters in file_in is not an integer multiple of 127 , the function appends char (4) symbols to the data it must decode. If you encode and then decode a file using rsencof and rsdecof, respectively, the decoded file might have char (4) symbols at the end that the original file does not have.
rsdecof(file_in,file_out, err_cor) is the same as the first syntax, except that err_cor specifies the error-correction capability for each block of 127 codeword characters. The message length is 127 - 2 *err_cor. The value in err_cor must match the value used in rsencof when file_in was created.

\section*{Examples An example is on the reference page for rsencof.}

\section*{See Also rsencof}
How To . "Block Codes"

\section*{Purpose Reed-Solomon encoder}
```

Syntax code = rsenc(msg,n,k)
code = rsenc(msg,n,k,genpoly)
code = rsenc(...,paritypos)

```

\section*{Description}

\section*{Examples}
code \(=r \operatorname{senc}(m s g, n, k)\) encodes the message in msg using an \([\mathrm{n}, \mathrm{k}]\) Reed-Solomon code with the narrow-sense generator polynomial. msg is a Galois array of symbols having \(m\) bits each. Each k-element row of msg represents a message word, where the leftmost symbol is the most significant symbol. n is at most \(2^{\mathrm{m}}-1\). If n is not exactly \(2^{\mathrm{m}}-1\), rsenc uses a shortened Reed-Solomon code. Parity symbols are at the end of each word in the output Galois array code.
code \(=r\) senc (msg, \(n, k\), genpoly) is the same as the syntax above, except that a nonempty value of genpoly specifies the generator polynomial for the code. In this case, genpoly is a Galois row vector that lists the coefficients, in order of descending powers, of the generator polynomial. The generator polynomial must have degree \(n-k\). To use the default narrow-sense generator polynomial, set genpoly to [].
code \(=\) rsenc(..., paritypos) specifies whether rsenc appends or prepends the parity symbols to the input message to form code. The string paritypos can be either 'end' or 'beginning'. The default is 'end'.

The example below encodes two message words using a \((7,3)\) Reed-Solomon encoder.
```

m = 3; % Number of bits per symbol
n = 2^m-1; k = 3; % Word lengths for code
msg = gf([2 7 3; 4 0 6],m); % Two rows of m-bit symbols
code = rsenc(msg,n,k)

```

The output is below.
```

code = GF(2^3) array. Primitive polynomial = D^3+D+1 (11 decimal)

```
Array elements =
    \(\begin{array}{lllllll}2 & 7 & 3 & 3 & 6 & 7 & 6\end{array}\)
    \(\begin{array}{lllllll}4 & 0 & 6 & 4 & 2 & 2 & 0\end{array}\)
    For additional examples, see "Represent Words for Reed-Solomon
    Codes" and "Create and Decode Reed-Solomon Codes".
Limitations \(\quad \mathrm{n}\) and k must differ by an even integer. n must be between 3 and 65535 .
See Also rsdec | gf | rsgenpoly
How To . "Block Codes"

\section*{Purpose Encode ASCII file using Reed-Solomon code}
```

Syntax rsencof(file_in,file_out); rsencof(file_in,file_out,err_cor);

```

\section*{Description \\ Warning}

This function is obsolete and may be removed in the future. We strongly recommend that you use the comm. RSEncoder System object instead.
rsencof(file_in,file_out) encodes the ASCII file file_in using (127, 117) Reed-Solomon code. The error-correction capability of this code is 5 for each block of 127 codeword characters. This function writes the encoded text to the file file_out. Both file_in and file_out are string variables.
rsencof(file_in,file_out,err_cor) is the same as the first syntax, except that err_cor specifies the error-correction capability for each block of 127 codeword characters. The message length is 127-2 * err_cor.

Note If the number of characters in file_in is not an integer multiple of 127-2 * err_cor, the function appends char(4) symbols to file_out.

\section*{Examples}

The file matlabroot/toolbox/comm/comm/oct2dec.m contains text help for the oct2dec function in this toolbox. The commands below encode the file using rsencof and then decode it using rsdecof.
```

file_in = [matlabroot '/toolbox/comm/comm/oct2dec.m'];
file_out = 'encodedfile'; % Or use another filename
rsencof(file_in,file_out) % Encode the file.
file_in = file_out;
file_out = 'decodedfile'; % Or use another filename
rsdecof(file_in,file_out) % Decode the file.

```

To see the original file and the decoded file in the MATLAB workspace, use the commands below (or similar ones if you modified the filenames above).
type oct2dec.m
type decodedfile

\section*{See Also rsdecof}

How To . "Block Codes"

\section*{Purpose Generator polynomial of Reed-Solomon code}
```

Syntax genpoly = rsgenpoly ( $\mathrm{n}, \mathrm{k}$ )
genpoly = rsgenpoly(n,k,prim_poly)
genpoly = rsgenpoly(n,k,prim_poly,b)
[genpoly,t] = rsgenpoly(...)

```

\section*{Description}
genpoly = rsgenpoly( \(\mathrm{n}, \mathrm{k}\) ) returns the narrow-sense generator polynomial of a Reed-Solomon code with codeword length n and message length \(k\). The codeword length \(n\) must have the form \(2^{\mathrm{m}}-1\) for some integer m between 3 and 16.
, and \(n-k\) must be an even integer. The output genpoly is a Galois row vector that represents the coefficients of the generator polynomial in order of descending powers. The narrow-sense generator polynomial is \(\left(\mathrm{X}-\mathrm{Alpha}{ }^{1}\right)\left(\mathrm{X}-\mathrm{Alpha}^{2}\right) \ldots\left(\mathrm{X}-\mathrm{Alpha}^{2 \mathrm{t}}\right)\) where:
- Alpha represents a root of the default primitive polynomial for the field GF(n+1),
- and t represents the code's error-correction capability, ( \(\mathrm{n}-\mathrm{k}\) )/2.
genpoly \(=\) rsgenpoly( \(\left.n, k, p r i m \_p o l y\right)\) is the same as the syntax above, except that prim_poly specifies the primitive polynomial for \(\mathrm{GF}(\mathrm{n}+1)\) that has Alpha as a root. prim_poly is an integer whose binary representation indicates the coefficients of the primitive polynomial. To use the default primitive polynomial GF( \(\mathrm{n}+1\) ), set prim_poly to [].
genpoly = rsgenpoly(n,k,prim_poly,b) returns the generator polynomial (X - Alpha \({ }^{\text {b }}\) ) (X - Alpha \({ }^{\text {b+1 }}\) )...(X - Alpha \({ }^{\text {b+2t-1 }}\) ), where:
- b is an integer,
- Alpha is a root of prim_poly,
- and t is the code's error-correction capability, \((\mathrm{n}-\mathrm{k}) / 2\).
[genpoly,t] = rsgenpoly(...) returns \(t\), the error-correction capability of the code.

Examples
The examples below create Galois row vectors that represent generator polynomials for a [7,3] Reed-Solomon code. The vectors \(g\) and \(g 2\) both represent the narrow-sense generator polynomial, but with respect to different primitive elements A. More specifically, g2 is defined such that \(A\) is a root of the primitive polynomial \(D^{3}+D^{2}+1\) for \(\mathrm{GF}(8)\), not of the default primitive polynomial \(\mathrm{D}^{3}+\mathrm{D}+1\). The vector g 3 represents the generator polynomial \(\left(X-A^{3}\right)\left(X-A^{4}\right)\left(X-A^{5}\right)\left(X-A^{6}\right)\), where \(A\) is a root of \(\mathrm{D}^{3}+\mathrm{D}^{2}+1\) in \(\mathrm{GF}(8)\).
```

g = rsgenpoly (7,3)
g2 = rsgenpoly(7,3,13) % Use nondefault primitive polynomial.
g3 = rsgenpoly(7,3,13,3) % Use b = 3.

```

The output is below.
```

g = GF(2^3) array. Primitive polynomial = D^3+D+1 (11 decimal)
Array elements =
1 }
g2 = GF(2^3) array. Primitive polynomial = D^3+D^2+1 (13 decimal)
Array elements =
1 4
g3 = GF(2^3) array. Primitive polynomial = D^3+D^2+1 (13 decimal)
Array elements =
1

```

As another example, the command below shows that the default narrow-sense generator polynomial for a [15,11] Reed-Solomon code is
\(\mathrm{X}^{4}+\left(\mathrm{A}^{3}+\mathrm{A}^{2}+1\right) \mathrm{X}^{3}+\left(\mathrm{A}^{3}+\mathrm{A}^{2}\right) \mathrm{X}^{2}+\mathrm{A}^{3} \mathrm{X}+\left(\mathrm{A}^{2}+\mathrm{A}+1\right)\), where A is a root of the default primitive polynomial for \(\mathrm{GF}(16)\).
gp = rsgenpoly \((15,11)\)
gp = GF(2^4) array. Primitive polynomial = D^4+D+1 (19 decimal)
Array elements =
\(\begin{array}{lllll}1 & 13 & 12 & 8 & 7\end{array}\)
For additional examples, see "Parameters for Reed-Solomon Codes".
Limitations \(\quad n\) and \(k\) must differ by an even integer. The maximum allowable value of \(n\) is 65535 .

\section*{See Also \\ gf | rsenc | rsdec}

How To
- "Block Codes"

\section*{Purpose \\ Generator polynomial coefficients of Reed-Solomon code}
```

Syntax
X = rsgenpolycoeffs(...)
[x,t] = rsgenpolycoeffs(...)

```

Description \(\quad x=r s g e n p o l y c o e f f s(. .\).\() returns the coefficients for the generator\) polynomial of the Reed-Solomon code. The output is identical to genpoly = rsgenpoly(...); \(x=\) genpoly.x.
[x,t] = rsgenpolycoeffs(...) returns \(t\), the error-correction capability of the code.

\author{
See Also \\ rsgenpoly | gf | rsenc | rsdec
}

\section*{Purpose Generate scatter plot}
```

Syntax scatterplot(x)
scatterplot(x,n)
scatterplot(x,n,offset)
scatterplot(x,n,offset,plotstring)
scatterplot(x, n, offset,plotstring,h)
h = scatterplot(...)

```

\section*{Description}
scatterplot(x) produces a scatter plot for the signal \(x\). The interpretation of \(x\) depends on its shape and complexity:
- If \(x\) is a real two-column matrix, scatterplot interprets the first column as in-phase components and the second column as quadrature components.
- If x is a complex vector, scatterplot interprets the real part as in-phase components and the imaginary part as quadrature components.
- If x is a real vector, scatterplot interprets it as a real signal.
scatterplot \((x, n)\) is the same as the first syntax, except that the function plots every nth value of the signal, starting from the first value. That is, the function decimates \(x\) by a factor of \(n\) before plotting.
scatterplot ( \(x, n\), offset) is the same as the first syntax, except that the function plots every nth value of the signal, starting from the (offset+1)st value in \(x\).
scatterplot( \(x, n\), offset, plotstring) is the same as the syntax above, except that plotstring determines the plotting symbol, line type, and color for the plot. plotstring is a string whose format and meaning are the same as in the plot function.
scatterplot ( \(\mathrm{x}, \mathrm{n}\), offset, plotstring, h ) is the same as the syntax above, except that the scatter plot is in the figure whose handle is h , rather than a new figure. h must be a handle to a figure that scatterplot previously generated. To plot multiple signals in the same figure, use hold on.
\(\mathrm{h}=\) scatterplot(...) is the same as the earlier syntaxes, except that \(h\) is the handle to the figure that contains the scatter plot.

\title{
Examples See "View Signals Using Scatter Plots" or the example on the reference page for qamdemod. Both examples illustrate how to plot multiple signals in a single scatter plot.
}

For an online demonstration, type showdemo scattereyedemo.

\author{
See Also \\ How To \\ - scattereyedemo \\ - "Scatter Plots"
}
eyediagram | plot | scatter

Purpose
Syntax

\section*{Alternatives \\ Description}

Calculate bit error rate (BER) using semianalytic technique
```

ber = semianalytic(txsig,rxsig,modtype,M,Nsamp)

```
ber = semianalytic(txsig,rxsig,modtype,M,Nsamp)
ber = semianalytic(txsig,rxsig,modtype,M,Nsamp,num,den)
ber = semianalytic(txsig,rxsig,modtype,M,Nsamp,num,den)
ber = semianalytic(txsig,rxsig,modtype,M,Nsamp,EbNo)
ber = semianalytic(txsig,rxsig,modtype,M,Nsamp,EbNo)
ber =
ber =
semianalytic(txsig,rxsig,modtype,M,Nsamp,num, den, EbNo)
semianalytic(txsig,rxsig,modtype,M,Nsamp,num, den, EbNo)
[ber,avgampl,avgpower] = semianalytic(...)
```

[ber,avgampl,avgpower] = semianalytic(...)

```

As an alternative to the semianalytic function, invoke the BERTool GUI (bertool) and use the Semianalytic tab.
ber \(=\) semianalytic(txsig, rxsig, modtype, \(M\), Nsamp) returns the bit error rate (BER) of a system that transmits the complex baseband vector signal txsig and receives the noiseless complex baseband vector signal rxsig. Each of these signals has Nsamp samples per symbol. Nsamp is also the sampling rate of txsig and rxsig, in Hz. The function assumes that rxsig is the input to the receiver filter, and the function filters rxsig with an ideal integrator. modtype is the modulation type of the signal and \(M\) is the alphabet size. The table below lists the valid values for modtype and \(M\).
\begin{tabular}{l|l|l}
\hline \begin{tabular}{l} 
Modulation \\
Scheme
\end{tabular} & Value of modtype & Valid Values of \(\boldsymbol{M}\) \\
\hline \begin{tabular}{l} 
Differential phase \\
shift keying (DPSK)
\end{tabular} & 'dpsk' & 2,4 \\
\hline \begin{tabular}{l} 
Minimum shift \\
keying (MSK) with \\
differential encoding
\end{tabular} & 'msk/diff' & 2 \\
\hline \begin{tabular}{l} 
Minimum shift \\
keying (MSK) with \\
nondifferential \\
encoding
\end{tabular} & 'msk/nondiff' & 2 \\
\hline
\end{tabular}
\begin{tabular}{l|l|l}
\hline \begin{tabular}{l} 
Modulation \\
Scheme
\end{tabular} & Value of modtype & Valid Values of \(\boldsymbol{M}\) \\
\hline \begin{tabular}{l} 
Phase shift \\
keying (PSK) \\
with differential \\
encoding, where the \\
phase offset of the \\
constellation is 0
\end{tabular} & 'psk/diff' & 2,4 \\
\hline \begin{tabular}{l} 
Phase shift \\
keying (PSK) with \\
nondifferential \\
encoding, where the \\
phase offset of the \\
constellation is 0
\end{tabular} & 'psk/nondiff' & \(2,4,8,16,32\), or 64 \\
\hline \begin{tabular}{l} 
Offset quaternary \\
phase shift keying \\
(OQPSK)
\end{tabular} & 'oqpsk' & 4 \\
\hline \begin{tabular}{l} 
Quadrature \\
amplitude \\
modulation (QAM)
\end{tabular} & 'qam' & \begin{tabular}{l}
\(4,8,16,32,64,128\), \\
\hline
\end{tabular} \\
\hline
\end{tabular}
'msk/diff' is equivalent to conventional MSK (setting the 'Precoding' property of the MSK object to 'off'), while 'msk/nondiff' is equivalent to precoded MSK (setting the 'Precoding ' property of the MSK object to 'on').

Note The output ber is an upper bound on the BER in these cases:
- DQPSK (modtype \(=\) 'dpsk', \(\mathrm{M}=4\) )
- Cross QAM (modtype = 'qam ', M not a perfect square). In this case, note that the upper bound used here is slightly tighter than the upper bound used for cross QAM in the berawgn function.

When the function computes the BER, it assumes that symbols are Gray-coded. The function calculates the BER for values of \(\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}\) in the range of \([0: 20] \mathrm{dB}\) and returns a vector of length 21 whose elements correspond to the different \(\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}\) levels.

Note You must use a sufficiently long vector txsig, or else the calculated BER will be inaccurate. If the system's impulse response is L symbols long, the length of txsig should be at least \(\mathrm{m}^{\mathrm{L}}\). A common approach is to start with an augmented binary pseudonoise (PN) sequence of total length \(\left(\log _{2} M\right) M^{L}\). An augmented \(P N\) sequence is a PN sequence with an extra zero appended, which makes the distribution of ones and zeros equal.
ber = semianalytic(txsig,rxsig,modtype, \(M\), Nsamp, num, den) is the same as the previous syntax, except that the function filters rxsig with a receiver filter instead of an ideal integrator. The transfer function of the receiver filter is given in descending powers of \(z\) by the vectors num and den.
ber = semianalytic(txsig,rxsig, modtype, \(M\), Nsamp, EbNo) is the same as the first syntax, except that EbNo represents \(\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}\), the ratio of bit energy to noise power spectral density, in dB . If EbNo is a vector, then the output ber is a vector of the same size, whose elements correspond to the different \(\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}\) levels.
ber =
semianalytic(txsig, rxsig, modtype, M, Nsamp, num, den, EbNo) combines the functionality of the previous two syntaxes.
[ber, avgampl, avgpower] = semianalytic(...) returns the mean complex signal amplitude and the mean power of rxsig after filtering it by the receiver filter and sampling it at the symbol rate.

\section*{Examples}

A typical procedure for implementing the semianalytic technique is in "Procedure for the Semianalytic Technique". Sample code is in "Example: Using the Semianalytic Technique".
\begin{tabular}{ll} 
Limitations & \begin{tabular}{l} 
The function makes several important assumptions about the \\
communication system. See "When to Use the Semianalytic Technique" \\
to find out whether your communication system is suitable for the \\
semianalytic technique and the semianalytic function.
\end{tabular} \\
References & \begin{tabular}{l} 
[1] Jeruchim, M. C., P. Balaban, and K. S. Shanmugan, Simulation of \\
Communication Systems, New York, Plenum Press, 1992.
\end{tabular} \\
See Also & \begin{tabular}{l} 
[2] Pasupathy, S., "Minimum Shift Keying: A Spectrally Efficient \\
Modulation," IEEE Communications Magazine, July, 1979, pp. 14-22.
\end{tabular} \\
How To & noisebw | qfunc
\end{tabular}

\title{
Purpose Convert shift to mask vector for shift register configuration
}

Syntax mask \(=\operatorname{shift2mask}(\) prpoly, shift \()\)
Description mask = shift2mask(prpoly, shift) returns the mask that is equivalent to the shift (or offset) specified by shift, for a linear feedback shift register whose connections are specified by the primitive polynomial prpoly. The prpoly input can have one of these formats:
- A binary vector that lists the coefficients of the primitive polynomial in order of descending powers
- An integer scalar whose binary representation gives the coefficients of the primitive polynomial, where the least significant bit is the constant term
The shift input is an integer scalar.

Note To save time, shift2mask does not check that prpoly is primitive. If it is not primitive, the output is not meaningful. To find primitive polynomials, use primpoly or see [2].

\section*{Definition of Equivalent Mask}

The equivalent mask for the shift s is the remainder after dividing the polynomial \(x^{s}\) by the primitive polynomial. The vector mask represents the remainder polynomial by listing the coefficients in order of descending powers.

\section*{Shifts, Masks, and Pseudonoise Sequence Generators}

Linear feedback shift registers are part of an implementation of a pseudonoise sequence generator. Below is a schematic diagram of a pseudonoise sequence generator. All adders perform addition modulo 2.


The primitive polynomial determines the state of each switch labeled \(\mathrm{g}_{\mathrm{k}}\), and the mask determines the state of each switch labeled \(\mathrm{m}_{\mathrm{k}}\). The lower half of the diagram shows the implementation of the shift, which delays the starting point of the output sequence. If the shift is zero, the \(\mathrm{m}_{0}\) switch is closed while all other \(\mathrm{m}_{\mathrm{k}}\) switches are open. The table below indicates how the shift affects the shift register's output.
\begin{tabular}{l|l|l|l|l|l|l}
\hline & \(\mathbf{T}=\mathbf{0}\) & \(\mathbf{T}=\mathbf{1}\) & \(\mathbf{T}=\mathbf{2}\) & \(\ldots\) & \(\mathbf{T}=\mathbf{s}\) & \(\mathbf{T}=\mathbf{s + 1}\) \\
\hline \begin{tabular}{l} 
Shift \(=\) \\
\(\mathbf{0}\)
\end{tabular} & \(\mathrm{x}_{0}\) & \(\mathrm{x}_{1}\) & \(\mathrm{x}_{2}\) & \(\ldots\) & \(\mathrm{x}_{\mathrm{s}}\) & \(\mathrm{x}_{\mathrm{s}+1}\) \\
\hline \begin{tabular}{l} 
Shift \(=\) \\
\(\mathbf{s}>\mathbf{0}\)
\end{tabular} & \(\mathrm{x}_{\mathrm{s}}\) & \(\mathrm{x}_{\mathrm{s}+1}\) & \(\mathrm{x}_{\mathrm{s}+2}\) & \(\ldots\) & \(\mathrm{x}_{2 \mathrm{~s}}\) & \(\mathrm{x}_{2 \mathrm{~s}+1}\) \\
\hline
\end{tabular}

If you have Communications System Toolbox software and want to generate a pseudonoise sequence in a Simulink model, see the PN Sequence Generator block reference page.
Examples The command below converts a shift of 5 into the equivalent mask\(x^{3}+x+1\), for the linear feedback shift register whose connections arespecified by the primitive polynomial \(x^{4}+x^{3}+1\).
```

mk = shift2mask([1 1 0 0 1],5)
mk =

```
    \(\begin{array}{llll}1 & 0 & 1 & 1\end{array}\)
References [1] Lee, J. S., and L. E. Miller, CDMA Systems Engineering Handbook, Boston, Artech House, 1998.
[2] Simon, Marvin K., Jim K. Omura, et al., Spread Spectrum Communications Handbook, New York, McGraw-Hill, 1994.
See Also mask2shift | deconv | isprimitive | primpoly

\section*{Purpose}

Construct signed least mean square (LMS) adaptive algorithm object

\section*{Syntax}
```

alg = signlms(stepsize)
alg = signlms(stepsize,algtype)

```

\section*{Description}

The signlms function creates an adaptive algorithm object that you can
use with the lineareq function or dfe function to create an equalizer object. You can then use the equalizer object with the equalize function to equalize a signal. To learn more about the process for equalizing a signal, see "Adaptive Algorithms".
alg = signlms(stepsize) constructs an adaptive algorithm object based on the signed least mean square (LMS) algorithm with a step size of stepsize.
alg = signlms(stepsize,algtype) constructs an adaptive algorithm object of type algtype from the family of signed LMS algorithms. The table below lists the possible values of algtype.
\begin{tabular}{l|l}
\hline Value of algtype & Type of Signed LMS Algorithm \\
\hline 'Sign LMS' & Sign LMS (default) \\
\hline 'Signed Regressor LMS' & Signed regressor LMS \\
\hline 'Sign Sign LMS' & Sign-sign LMS \\
\hline
\end{tabular}

\section*{Properties}

The table below describes the properties of the signed LMS adaptive algorithm object. To learn how to view or change the values of an adaptive algorithm object, see "Access Properties of an Adaptive Algorithm".
\begin{tabular}{l|l}
\hline Property & Description \\
\hline AlgType & \begin{tabular}{l} 
Type of signed LMS algorithm, \\
corresponding to the algtype \\
input argument. You cannot \\
change the value of this property \\
after creating the object.
\end{tabular} \\
\hline StepSize & \begin{tabular}{l} 
LMS step size parameter, a \\
nonnegative real number
\end{tabular} \\
\hline LeakageFactor & \begin{tabular}{l} 
LMS leakage factor, a real \\
number between 0 and 1. A value \\
of 1 corresponds to a conventional \\
weight update algorithm, while \\
a value of 0 corresponds to a \\
memoryless update algorithm.
\end{tabular} \\
\hline
\end{tabular}

\section*{Algorithms}

\section*{References}

Referring to the schematics presented in "Equalizer Structure", define \(w\) as the vector of all weights \(w_{\mathrm{i}}\) and define \(u\) as the vector of all inputs \(u_{i}\). Based on the current set of weights, \(w\), this adaptive algorithm creates the new set of weights given by
- (LeakageFactor) w + (StepSize) \(u^{*} \operatorname{sgn}(\operatorname{Re}(e))\), for sign LMS
- (LeakageFactor) w + (StepSize) \(\operatorname{sgn}(\operatorname{Re}(u)) \operatorname{Re}(e)\), for signed regressor LMS
- (LeakageFactor) \(w+(S t e p S i z e) ~ s g n(R e(u)) ~ s g n(R e(e))\), for sign-sign LMS
where the * operator denotes the complex conjugate and sgn denotes the signum function (sign in MATLAB technical computing software).
[1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, John Wiley \& Sons, 1998.
[2] Kurzweil, J., An Introduction to Digital Communications, New York, John Wiley \& Sons, 2000.
```

See Also lms | normlms | varlms | rls | cma | lineareq | dfe | equalize
How To . "Equalization"

```

Purpose Single sideband amplitude demodulation
```

Syntax
$z=\operatorname{ssbdemod}(y, F c, F s)$
z = ssbdemod(y,Fc,Fs,ini_phase)
z = ssbdemod(y,Fc,Fs,ini_phase,num,den)

```

\section*{Description}

\section*{Examples}

\section*{For All Syntaxes}
\(z=\operatorname{ssbdemod}(y, F c, F s)\) demodulates the single sideband amplitude modulated signal y from the carrier signal having frequency Fc ( Hz ). The carrier signal and \(y\) have sampling rate Fs (Hz). The modulated signal has zero initial phase, and can be an upper- or lower-sideband signal. The demodulation process uses the lowpass filter specified by [num,den] = butter(5,Fc*2/Fs).

Note The Fc and Fs arguments must satisfy Fs \(>2(\) Fc \(+B W\) ), where BW is the bandwidth of the original signal that was modulated.
z = ssbdemod(y,Fc,Fs,ini_phase) specifies the initial phase of the modulated signal in radians.
z = ssbdemod(y,Fc,Fs,ini_phase, num, den) specifies the numerator and denominator of the lowpass filter used in the demodulation.

The code below shows that ssbdemod can demodulate an upper-sideband or lower-sideband signal.
```

Fc = 12000; Fs = 270000;
t = [0:1/Fs:0.01]';
s = sin(2*pi*300*t)+2*sin(2*pi*600*t);
y1 = ssbmod(s,Fc,Fs,0); % Lower-sideband modulated signal
y2 = ssbmod(s,Fc,Fs,0,'upper'); % Upper-sideband modulated signal
s1 = ssbdemod(y1,Fc,Fs); % Demodulate lower sideband
s2 = ssbdemod(y2,Fc,Fs); % Demodulate upper sideband
% Plot results to show that the curves overlap.
figure; plot(t,s1,'r-',t,s2,'k--');

```
legend('Demodulation of upper sideband','Demodulation of lower sideband')


See Also ssbmod \| amdemod
How To
- "Digital Modulation"

Purpose Single sideband amplitude modulation
Syntax
y = ssbmod(x,Fc,Fs)
y = ssbmod(x,Fc,Fs,ini_phase)
y = ssbmod(x,fc,fs,ini_phase,'upper')

\section*{Description}

\section*{Examples An example using ssbmod is on the reference page for ammod.}

\section*{See Also \\ ssbdemod | ammod}

\section*{How To \\ - "Digital Modulation"}
\(y=\operatorname{ssbmod}(x, F c, F s)\) uses the message signal \(x\) to modulate a carrier signal with frequency Fc ( Hz ) using single sideband amplitude modulation in which the lower sideband is the desired sideband. The carrier signal and \(x\) have sample frequency \(\mathrm{Fs}(\mathrm{Hz})\). The modulated signal has zero initial phase.
\(y=\operatorname{ssbmod}\left(x, F c, F s, i n i \_p h a s e\right)\) specifies the initial phase of the modulated signal in radians.
\(y=\operatorname{ssbmod}\left(x, f c, f s, i n i \_p h a s e\right.\), 'upper') uses the upper sideband as the desired sideband.

\section*{Purpose}

Construct channel object from set of standardized channel models
Syntax
chan = stdchan(ts,fd,chantype)
[chan, chanprofile] = stdchan(...)
chan = stdchan(ts,fd, chantype, trms)

\section*{Description}

Channel Models
chan \(=\) stdchan(ts,fd, chantype) constructs a fading channel object chan according to the specified chantype. The input string chantype is chosen from the set of standardized channel profiles listed below. ts is the sample time of the input signal, in seconds. fd is the maximum Doppler shift, in Hertz.
[chan, chanprofile] = stdchan(...) also returns a structure chanprofile containing the parameters of the channel profile specified by chantype.
chan \(=\) stdchan(ts,fd, chantype, trms) is used to create a channel object, chan, when chantype is any one of '802.11a', '802.11b' or '802.11g'. When using '802.11a', '802.11b' or '802.11g' channels, you must specify TRMS, which is the RMS delay spread of the channel model. As per 802.11 specifications, TS should not be larger than TRMS/2.

COST 207 channel models (The Rician K factors for the cases cost207RAx4 and cost207RAx6 are chosen as in 3GPP TS 45.005 V7.9.0 (2007-2)):
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline cost207RAx4 & Rural Area (RAx), 4 taps \\
\hline cost207RAx6 & Rural Area (RAx), 6 taps \\
\hline cost207TUx6 & Typical Urban (TUx), 6 taps \\
\hline cost207TUx6alt & \begin{tabular}{l} 
Typical Urban (TUx), 6 taps, \\
alternative
\end{tabular} \\
\hline cost207TUx12 & Typical Urban (TUx), 12 taps \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline cost207TUx12alt & \begin{tabular}{l} 
Typical Urban (TUx), 12 taps, \\
alternative
\end{tabular} \\
\hline cost207BUx6 & Bad Urban (BUx), 6 taps \\
\hline cost207BUx6alt & \begin{tabular}{l} 
Bad Urban (BUx), 6 taps, \\
alternative
\end{tabular} \\
\hline cost207BUx12 & Bad Urban (BUx), 12 taps \\
\hline cost207BUx12alt & \begin{tabular}{l} 
Bad Urban (BUx), 12 taps, \\
alternative
\end{tabular} \\
\hline cost207HTx6 & Hilly Terrain (HTx), 6 taps \\
\hline cost207HTx6alt & \begin{tabular}{l} 
Hilly Terrain (HTx), 6 taps, \\
alternative
\end{tabular} \\
\hline cost207HTx12 & Hilly Terrain (HTx), 12 taps \\
\hline cost207HTx12alt & \begin{tabular}{l} 
Hilly Terrain (HTx), 12 taps, \\
alternative
\end{tabular} \\
\hline
\end{tabular}

GSM/EDGE channel models (3GPP TS 45.005 V7.9.0 (2007-2), 3GPP TS 05.05 V8.20.0 (2005-11)):
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline gsmRAx6c1 & \begin{tabular}{l} 
Typical case for rural area (RAx), \\
6 taps, case 1
\end{tabular} \\
\hline gsmRAx4c2 & \begin{tabular}{l} 
Typical case for rural area (RAx), \\
4 taps, case 2
\end{tabular} \\
\hline gsmHTx12c1 & \begin{tabular}{l} 
Typical case for hilly terrain \\
(HTx), 12 taps, case 1
\end{tabular} \\
\hline gsmHTx12c2 & \begin{tabular}{l} 
Typical case for hilly terrain \\
(HTx), 12 taps, case 2
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline gsmHTx6c1 & \begin{tabular}{l} 
Typical case for hilly terrain \\
(HTx), 6 taps, case 1
\end{tabular} \\
\hline gsmHTx6c2 & \begin{tabular}{l} 
Typical case for hilly terrain \\
(HTx), 6 taps, case 2
\end{tabular} \\
\hline gsmTUx12c1 & \begin{tabular}{l} 
Typical case for urban area (TUx), \\
12 taps, case 1
\end{tabular} \\
\hline gsmTUx12c1 & \begin{tabular}{l} 
Typical case for urban area (TUx), \\
12 taps, case 2
\end{tabular} \\
\hline gsmTUx6c1 & \begin{tabular}{l} 
Typical case for urban area (TUx), \\
6 taps. case 1
\end{tabular} \\
\hline gsmTUx6c2 & \begin{tabular}{l} 
Typical case for urban area (TUx), \\
6 taps, case 2
\end{tabular} \\
\hline gsmEQx6 & \begin{tabular}{l} 
Profile for equalization test \\
(EQx), 6 taps
\end{tabular} \\
\hline gsmTIx2 & \begin{tabular}{l} 
Typical case for very small cells \\
(TIx), 2 taps
\end{tabular} \\
\hline
\end{tabular}

3GPP channel models for deployment evaluation (3GPP TR 25.943 V6.0.0 (2004-12)):
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline 3gppTUx & \begin{tabular}{l} 
Typical Urban channel model \\
(TUx)
\end{tabular} \\
\hline 3gppRAx & Rural Area channel model (RAx) \\
\hline 3gppHTx & \begin{tabular}{l} 
Hilly Terrain channel model \\
\((\) (HTx \()\)
\end{tabular} \\
\hline
\end{tabular}

ITU-R 3G channel models (ITU-R M. 1225 (1997-2)):

\section*{stdchan}
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline itur3GIAx & Indoor office, channel A \\
\hline itur3GIBx & Indoor office, channel B \\
\hline itur3GPAx & \begin{tabular}{l} 
Outdoor to indoor and pedestrian, \\
channel A
\end{tabular} \\
\hline itur3GPBx & \begin{tabular}{l} 
Outdoor to indoor and pedestrian, \\
channel B
\end{tabular} \\
\hline itur3GVAx & \begin{tabular}{l} 
Vehicular - high antenna, channel \\
A
\end{tabular} \\
\hline itur3GVBx & \begin{tabular}{l} 
Vehicular - high antenna, channel \\
B
\end{tabular} \\
\hline itur3GSAxLOS & Satellite, channel A, LOS \\
\hline itur3GSAxNLOS & Satellite, channel A, NLOS \\
\hline itur3GSBxLOS & Satellite, channel B, LOS \\
\hline itur3GSBxNLOS & Satellite, channel B, NLOS \\
\hline itur3GSCxLOS & Satellite, channel C, LOS \\
\hline itur3GSCxNLOS & Satellite, channel C, NLOS \\
\hline
\end{tabular}

ITU-R HF channel models (ITU-R F. 1487 (2000)) (FD must be 1 to obtain the correct frequency spreads for these models.):
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline iturHFLQ & Low latitudes, Quiet conditions \\
\hline iturHFLM & \begin{tabular}{l} 
Low latitudes, Moderate \\
conditions
\end{tabular} \\
\hline iturHFLD & \begin{tabular}{l} 
Low latitudes, Disturbed \\
conditions
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline iturHFMQ & \begin{tabular}{l} 
Medium latitudes, Quiet \\
conditions
\end{tabular} \\
\hline iturHFMM & \begin{tabular}{l} 
Medium latitudes, Moderate \\
conditions
\end{tabular} \\
\hline iturHFMD & \begin{tabular}{l} 
Medium latitudes, Disturbed \\
conditions
\end{tabular} \\
\hline iturHFMDV & \begin{tabular}{l} 
Medium latitudes, Disturbed \\
conditions near vertical incidence
\end{tabular} \\
\hline iturHFHQ & High latitudes, Quiet conditions \\
\hline iturHFHM & \begin{tabular}{l} 
High latitudes, Moderate \\
conditions
\end{tabular} \\
\hline iturHFHD & \begin{tabular}{l} 
High latitudes, Disturbed \\
conditions
\end{tabular} \\
\hline
\end{tabular}

JTC channel models:
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline jtcInResA & Indoor residential A \\
\hline jtcInResB & Indoor residential B \\
\hline jtcInResC & Indoor residential C \\
\hline jtcInOffA & Indoor office A \\
\hline jtcInOffB & Indoor office B \\
\hline jtcInOffC & Indoor office C \\
\hline jtcInComA & Indoor commercial A \\
\hline jtcInComB & Indoor commercial B \\
\hline jtcInComC & Indoor commercial C \\
\hline
\end{tabular}

\section*{stdchan}
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline jtcOutUrbHRLAA & \begin{tabular}{l} 
Outdoor urban high-rise areas - \\
Low antenna A
\end{tabular} \\
\hline jtcOutUrbHRLAB & \begin{tabular}{l} 
Outdoor urban high-rise areas - \\
Low antenna B
\end{tabular} \\
\hline jtcOutUrbHRLAC & \begin{tabular}{l} 
Outdoor urban high-rise areas - \\
Low antenna C
\end{tabular} \\
\hline jtcOutUrbLRLAA & \begin{tabular}{l} 
Outdoor urban low-rise areas - \\
Low antenna A
\end{tabular} \\
\hline jtcOutUrbLRLAB & \begin{tabular}{l} 
Outdoor urban low-rise areas - \\
Low antenna B
\end{tabular} \\
\hline jtcOutUrbLRLAC & \begin{tabular}{l} 
Outdoor urban low-rise areas - \\
Low antenna C
\end{tabular} \\
\hline jtcOutResLAA & \begin{tabular}{l} 
Outdoor residential areas - Low \\
antenna A
\end{tabular} \\
\hline jtcOutResLAB & \begin{tabular}{l} 
Outdoor residential areas - Low \\
antenna B
\end{tabular} \\
\hline jtcOutResLAC & \begin{tabular}{l} 
Outdoor residential areas - Low \\
antenna C
\end{tabular} \\
\hline jtcOutUrbHRHAA & \begin{tabular}{l} 
Outdoor urban high-rise areas - \\
High antenna A
\end{tabular} \\
\hline jtcOutUrbHRHAB & \begin{tabular}{l} 
Outdoor urban high-rise areas - \\
High antenna B
\end{tabular} \\
\hline jtcOutUrbHRHAC & \begin{tabular}{l} 
Outdoor urban high-rise areas - \\
High antenna C
\end{tabular} \\
\hline\(j t c O u t U r b L R H A A\) & \begin{tabular}{l} 
Outdoor urban low-rise areas - \\
High antenna A
\end{tabular} \\
\hline\(j t c O u t U r b L R H A B\) & \begin{tabular}{l} 
Outdoor urban low-rise areas - \\
High antenna B
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline jtcOutUrbLRHAC & \begin{tabular}{l} 
Outdoor urban low-rise areas - \\
High antenna C
\end{tabular} \\
\hline jtcOutResHAA & \begin{tabular}{l} 
Outdoor residential areas - High \\
antenna A
\end{tabular} \\
\hline jtcOutResHAB & \begin{tabular}{l} 
Outdoor residential areas - High \\
antenna B
\end{tabular} \\
\hline jtcOutResHAC & \begin{tabular}{l} 
Outdoor residential areas - High \\
antenna C
\end{tabular} \\
\hline
\end{tabular}

HIPERLAN/2 channel models:
\begin{tabular}{l|l}
\hline Channel model & Profile \\
\hline hiperlan2A & Model A \\
\hline hiperlan2B & Model B \\
\hline hiperlan2C & Model C \\
\hline hiperlan2D & Model D \\
\hline hiperlan2E & Model E \\
\hline
\end{tabular}
\(802.11 \mathrm{a} / \mathrm{b} / \mathrm{g}\) channel models:
\(802.11 \mathrm{a} / \mathrm{b} / \mathrm{g}\) channel models share a common multipath delay profile

Note TS should not be larger than TRMS/2, as per 802.11 specifications.

\section*{Channel model}
802.11a
802.11b
802.11 g
```

Examples
ts = 0.1e-4; fd = 200;
chan = stdchan(ts, fd, 'cost207TUx6');
chan. NormalizePathGains = 1;
chan. StoreHistory = 1;
$y=$ filter(chan, ones(1,5e4));
plot(chan);

```
See Also doppler | rayleighchan | ricianchan

\section*{Purpose}

Compute number of symbol errors and symbol error rate

\section*{Syntax}
[number, ratio] = symerr(x,y)
[number, ratio] = symerr(x,y,flg)
[number, ratio,loc] = symerr(...)

\section*{Description}

\section*{For All Syntaxes}

The symer function compares binary representations of elements in \(x\) with those in \(y\). The schematics below illustrate how the shapes of \(x\) and y determine which elements symerr compares.


The output number is a scalar or vector that indicates the number of elements that differ. The size of number is determined by the optional input \(f l g\) and by the dimensions of x and y . The output ratio equals number divided by the total number of elements in the smaller input.

\section*{For Specific Syntaxes}
[number, ratio] \(=\operatorname{symerr}(x, y)\) compares the elements in \(x\) and \(y\). The sizes of \(x\) and \(y\) determine which elements are compared:
- If \(x\) and \(y\) are matrices of the same dimensions, then symerr compares x and y element by element. number is a scalar. See schematic (a) in the figure.
- If one is a row (respectively, column) vector and the other is a two-dimensional matrix, then symerr compares the vector element by element with each row (resp., column) of the matrix. The length
of the vector must equal the number of columns (resp., rows) in the matrix. number is a column (resp., row) vector whose mth entry indicates the number of elements that differ when comparing the vector with the mth row (resp., column) of the matrix. See schematics (b) and (c) in the figure.
[number, ratio] \(=\operatorname{symerr}(x, y, f l g)\) is similar to the previous syntax, except that \(f l g\) can override the defaults that govern which elements symerr compares and how symerr computes the outputs. The values of \(f l g\) are 'overall', 'column-wise', and 'row-wise'. The table below describes the differences that result from various combinations of inputs. In all cases, ratio is number divided by the total number of elements in \(y\).

\section*{Comparing a Two-Dimensional Matrix \(x\) with Another Input y}
\begin{tabular}{l|l|l|l}
\hline Shape of \(\mathbf{y}\) & flg & \begin{tabular}{l} 
Type of \\
Comparison
\end{tabular} & number \\
\hline \begin{tabular}{l} 
Two-dim. \\
matrix
\end{tabular} & \begin{tabular}{l} 
' overall' \\
(default)
\end{tabular} & \begin{tabular}{l} 
Element by \\
element
\end{tabular} & \begin{tabular}{l} 
Total number of \\
symbol errors
\end{tabular} \\
\cline { 2 - 4 } & 'column-wise' & \begin{tabular}{l} 
mth column of \(x\) \\
vs. mth column \\
of \(y\)
\end{tabular} & \begin{tabular}{l} 
Row vector \\
whose entries \\
count symbol \\
errors in each \\
column
\end{tabular} \\
\cline { 2 - 4 } & 'row-wise' & \begin{tabular}{l} 
mth row of x vs. \\
mth row of y
\end{tabular} & \begin{tabular}{l} 
Column vector \\
whose entries \\
count symbol \\
errors in each \\
row
\end{tabular} \\
\hline
\end{tabular}

\section*{Comparing a Two-Dimensional Matrix x with Another Input y (Continued)}
\begin{tabular}{l|l|l|l}
\hline Shape of \(\mathbf{y}\) & flg & \begin{tabular}{l} 
Type of \\
Comparison
\end{tabular} & number \\
\hline \multirow{3}{*}{ Column vector } & 'overall' & \begin{tabular}{l} 
y vs. each \\
column of \(x\)
\end{tabular} & \begin{tabular}{l} 
Total number of \\
symbol errors
\end{tabular} \\
\cline { 2 - 4 } & \begin{tabular}{l} 
'column-wise' \\
(default)
\end{tabular} & \begin{tabular}{l}
y vs. each \\
column of x
\end{tabular} & \begin{tabular}{l} 
Row vector \\
whose entries \\
count symbol \\
errors in each \\
column of x
\end{tabular} \\
\hline Row vector & 'overall' & \begin{tabular}{l}
y vs. each row \\
of x
\end{tabular} & \begin{tabular}{l} 
Total number of \\
symbol errors
\end{tabular} \\
\hline & \begin{tabular}{l} 
'row-wise ' \\
(default)
\end{tabular} & \begin{tabular}{l}
y vs. each row \\
of x
\end{tabular} & \begin{tabular}{l} 
Column vector \\
whose entries \\
count symbol \\
errors in each \\
row of x
\end{tabular} \\
\hline
\end{tabular}
[number, ratio,loc] = symerr(...) returns a binary matrix loc that indicates which elements of \(x\) and \(y\) differ. An element of loc is zero if the corresponding comparison yields no discrepancy, and one otherwise.

\section*{Examples}

On the reference page for biterr, the last example uses symerr.
The command below illustrates how symerr works when one argument is a vector and the other is a matrix. It compares the vector [ \(1,2,3\) ] ' to the columns
\[
\left[\begin{array}{l}
1 \\
3 \\
3
\end{array}\right],\left[\begin{array}{l}
1 \\
2 \\
3
\end{array}\right],\left[\begin{array}{l}
3 \\
2 \\
8
\end{array}\right] \text {, and }\left[\begin{array}{l}
1 \\
2 \\
3
\end{array}\right]
\]
of the matrix.
```

num = symerr([1 2 3]',[1 1 3 1;3 2 2 2; 3 3 8 3])
num =
1 0 2 0

```

As another example, the command below illustrates the use of \(f l g\) to override the default row-by-row comparison. Notice that number and ratio are scalars.
```

format rat;
[number,ratio,loc] = symerr([1 2; 3 4],[1 3],'overall')

```

The output is below.
number \(=\)

3
```

ratio =

```
    3/4
loc =
\begin{tabular}{ll}
0 & 1 \\
1 & 1
\end{tabular}
\begin{tabular}{|c|c|}
\hline Purpose & Produce syndrome decoding table \\
\hline Syntax & \(\mathrm{t}=\) syndtable ( h ) \\
\hline \multirow[t]{2}{*}{Description} & \(\mathrm{t}=\) syndtable( h ) returns a decoding table for an error-correcting binary code having codeword length \(n\) and message length \(k\). \(h\) is an ( \(\mathrm{n}-\mathrm{k}\) )-by-n parity-check matrix for the code. t is a \(2^{\mathrm{n}-\mathrm{k}}\)-by- n binary matrix. The rth row of \(t\) is an error pattern for a received binary codeword whose syndrome has decimal integer value r-1. (The syndrome of a received codeword is its product with the transpose of the parity-check matrix.) In other words, the rows of \(t\) represent the coset leaders from the code's standard array. \\
\hline & When converting between binary and decimal values, the leftmost column is interpreted as the most significant digit. This differs from the default convention in the bi2de and de2bi commands. \\
\hline Examples & An example is in "Decoding Table". \\
\hline References & [1] Clark, George C., Jr., and J. Bibb Cain, Error-Correction Coding for Digital Communications, New York, Plenum, 1981. \\
\hline See Also & decode | hammgen | gfcosets \\
\hline How To & - "Block Codes" \\
\hline
\end{tabular}

\section*{testconsole.Results}

Purpose
Description

\section*{Properties}

Gets results from test console simulations

The getResults method of the Error Rate Test Console returns an instance of a testconsole.Results object containing simulation results data. You use methods of the results object to retrieve and plot simulations results data.

A testconsole.Results object has the properties shown on the following table. All properties are writable except for the ones explicitly noted otherwise.
\begin{tabular}{l|l}
\hline Property & Description \\
\hline TestConsoleName & \begin{tabular}{l} 
Error Rate Test Console. This \\
property is not writable.
\end{tabular} \\
\hline System Under Test Name & \begin{tabular}{l} 
Name of the system under test \\
for which the Error Rate Test \\
Console obtained results. This \\
property is not writable.
\end{tabular} \\
\hline IterationMode & \begin{tabular}{l} 
Iteration mode the Error Rate \\
Test Console used for obtaining \\
results. This property is not \\
writable.
\end{tabular} \\
\hline TestPoint & \begin{tabular}{l} 
Specify the name of the registered \\
test point for which the results \\
object parses results. The \\
getData, plot, and semilogy \\
methods of the Results object \\
return data or create a plot for \\
the test point that the TestPoint \\
property specifies.
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline Metric & \begin{tabular}{l} 
Specify the name of the test \\
metric for which the results object \\
parses results. The getData, \\
plot, and semilogy methods of the \\
Results object returns data or \\
creates a plot for the metric that \\
the Metric property specifies.
\end{tabular} \\
\hline TestParameter1 & \begin{tabular}{l} 
Specifies the name of the first \\
independent variable for which \\
the results object parses results.
\end{tabular} \\
\hline TestParameter2 & \begin{tabular}{l} 
Specifies the name of the second \\
independent variable for which \\
the results object parses results.
\end{tabular} \\
\hline
\end{tabular}

\section*{Methods}

A testconsole.Results object has the following methods.

\section*{getData}
d = getData(r) returns results data matrix, \(d\), available in the results object \(r\). The returned results correspond to the test point currently specified in the TestPoint property of \(r\), and to the test metric currently specified in the Metric property of \(r\).
If IterationMode is 'Combinatorial' then \(d\) is a matrix containing results for all the sweep values available in the test parameters specified in the TestParameter1 and TestParameter2 properties. The rows of the matrix correspond to results for all the sweep values available in TestParameter1. The columns of the matrix correspond to results for all sweep values available in TestParameter2. If more than two test parameters are registered to the Error Rate Test Console, \(d\) contains results corresponding to the first value in the sweep vector of all parameters that are not TestParameter1 or TestParameter2.
If IterationMode is 'Indexed', then \(d\) is a vector of results corresponding to each indexed combination of all the test parameter values registered to the Error Rate Test Console.

\section*{plot}
plot ( \(r\) ) creates a plot for the results available in the results object \(r\). The plot corresponds to the test point and test metric, specified by the TestPoint and Metric properties of \(r\)
If IterationMode is 'Combinatorial' then the plot contains a set of curves. The sweep values in TestParameter 1 control the x-axis and the number of sweep values for TestParameter2 specifies how many curves the plot contains. If more than two test parameters are registered to the Error Rate Test Console, the curves correspond to results obtained with the first value in the sweep vector of all parameters that are not TestParameter1, or TestParameter2.
No plots are available when 'IterationMode' is 'Indexed'.

\section*{semilogy}
semilogy (...) is the same as plot(...), except that the Y-Axis uses a logarithmic (base 10) scale.

\section*{surf}
\(\operatorname{surf}(\mathrm{r})\) creates a 3-D, color, surface plot for the results available in the results object, r. The surface plot corresponds to following items:
- The test point you specify using the TestPoint property of the results object
- The test metric currently you specify in the Metric property of the results object

You can specify parameter/value pairs for the results object, which establishes additional properties of the surface plot.

When you select 'Combinatorial' for the IterationMode, the sweep values available in the test parameter you specify for the TestParameter1 property control the x -axis of the surface plot. The sweep values available in the test parameter you specify for the TestParameter2 property control the \(y\)-axis.

If more than two test parameters are registered to the test console, the surface plot corresponds to the results obtained with the parameter
sweep values previously specified with the setParsingValues method of the results object.

You display the current parsing values by calling the getParsingValues method of the results object. The parsing values default to the first value in the sweep vector of each test parameter. By default, the surf method ignores the parsing values for any parameters currently set as TestParameter 1 or TestParameter2.

No surface plots are available if the IterationMode is 'Indexed', when less than two registered test parameters exist, or TestParameter2 is set to 'None'.

\section*{setParsingValues}
setParsingValues(R,'ParameterName1', 'Value1', ... 'ParameterName2', 'Value2', ...) sets the parsing values to the values you specify using the parameter-value pairs. Parameter name inputs must correspond to names of registered test parameters, and value inputs must correspond to a valid test parameter sweep value.
You use this method for specifying single sweep values for test parameters that differ from the values for TestParameter1 and TestParameter2. When you define this method, the results object returns the data values or plots corresponding to the sweep values you set for the setParsingValues method. The parsing values default to the first value in the sweep vector of each test parameter.
You display the current parsing values by calling the getParsingValues method of the results object. You may set parsing values for parameters in TestParameter1 and TestParameter2, but the results object ignores the values when getting data or returning plots.

Parsing values are irrelevant when IterationMode is 'Indexed'.

\section*{getParsingValues}
getParsingValues displays the current parsing values for the Error Rate Test Console.
\(s=\) getParsingValues(r) returns a structure, \(s\), with field names equal to the registered test parameter names and with values corresponding to the current parsing values.

Parsing values are irrelevant when IterationMode is 'Indexed'.
See Also commtest.ErrorRate

\section*{Purpose}

\section*{Syntax}

Description

Construct variable-step-size least mean square (LMS) adaptive algorithm object
alg = varlms(initstep,incstep,minstep,maxstep)
The varlms function creates an adaptive algorithm object that you can use with the lineareq function or dfe function to create an equalizer object. You can then use the equalizer object with the equalize function to equalize a signal. To learn more about the process for equalizing a signal, see "Adaptive Algorithms".
alg = varlms(initstep,incstep, minstep, maxstep) constructs an adaptive algorithm object based on the variable-step-size least mean square (LMS) algorithm. initstep is the initial value of the step size parameter. incstep is the increment by which the step size changes from iteration to iteration. minstep and maxstep are the limits between which the step size can vary.

\section*{Properties}

The table below describes the properties of the variable-step-size LMS adaptive algorithm object. To learn how to view or change the values of an adaptive algorithm object, see "Access Properties of an Adaptive Algorithm".
\begin{tabular}{l|l}
\hline Property & Description \\
\hline AlgType & \begin{tabular}{l} 
Fixed value, 'Variable Step \\
Size LMS'
\end{tabular} \\
\hline LeakageFactor & \begin{tabular}{l} 
LMS leakage factor, a real \\
number between 0 and 1. A value \\
of 1 corresponds to a conventional \\
weight update algorithm, while \\
a value of 0 corresponds to a \\
memoryless update algorithm.
\end{tabular} \\
\hline InitStep & \begin{tabular}{l} 
Initial value of step size when the \\
algorithm starts
\end{tabular} \\
\hline
\end{tabular}
\begin{tabular}{l|l}
\hline Property & Description \\
\hline IncStep & \begin{tabular}{l} 
Increment by which the step \\
size changes from iteration to \\
iteration
\end{tabular} \\
\hline MinStep & Minimum value of step size \\
\hline MaxStep & Maximum value of step size \\
\hline
\end{tabular}

Also, when you use this adaptive algorithm object to create an equalizer object (via the lineareq or dfe function), the equalizer object has a StepSize property. The property value is a vector that lists the current step size for each weight in the equalizer.

\section*{Examples}

Algorithms

For an example that uses this function, see "Linked Properties of an Equalizer Object".

Referring to the schematics presented in "Equalizer Structure", define \(w\) as the vector of all current weights \(w_{\mathrm{i}}\) and define \(u\) as the vector of all inputs \(u_{i}\). Based on the current step size, \(\mu\), this adaptive algorithm first computes the quantity
\[
\mu_{0}=\mu+(\text { IncStep }) \operatorname{Re}\left(\text { gg }_{\text {prev }}\right)
\]
where \(g=u e^{*}, g_{\text {prev }}\) is the analogous expression from the previous iteration, and the \({ }^{*}\) operator denotes the complex conjugate.

Then the new step size is given by
- \(\mu_{0}\), if it is between MinStep and MaxStep
- MinStep, if \(\mu_{0}<\) MinStep
- MaxStep, if \(\mu_{0}>\) MaxStep

The new set of weights is given by
\[
\text { (LeakageFactor) } \mathrm{w}+2 \mu \mathrm{~g}^{*}
\]
References [1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, Wiley, 1998.
See Also lms | signlms | normlms | rls | cma | lineareq | dfe | equalize
How To - "Equalization"

Purpose Convert vector into matrix
```

Syntax mat = vec2mat(vec,matcol)
mat = vec2mat(vec,matcol,padding)
[mat,padded] = vec2mat(...)

```

\section*{Description}

\section*{Examples}
```

vec = [1 2 3 4 5];
[mat,padded] = vec2mat(vec,3)
[mat2,padded2] = vec2mat(vec,4)
mat3 = vec2mat(vec,4,[10 9 8; 7 6 5; 4 3 2])

```

The output is below.
```

mat =

```
    133
```

        4 5 0
    padded =
        1
    mat2 =
    1 2 3 4
    5 0 0
    padded2 =
3
mat3 =
1 2 3 4
5

```
See Also
reshape

Purpose Convolutionally decode binary data using Viterbi algorithm
```

Syntax decoded = vitdec(code,trellis,tblen,opmode,dectype)
decoded = vitdec(code,trellis,tblen,opmode,'soft',nsdec)
decoded = ...
vitdec(code,trellis,tblen,opmode,dectype,puncpat)
decoded = ...
vitdec(code,trellis,tblen,opmode,dectype,puncpat, eraspat)
decoded = ...
vitdec(...,'cont',...,initmetric,initstates,initinputs)
[decoded,finalmetric,finalstates,finalinputs] = ...
vitdec(...,'cont',...)

```

\section*{Description}
decoded \(=\) vitdec (code,trellis,tblen,opmode,dectype) decodes the vector code using the Viterbi algorithm. The MATLAB structure trellis specifies the convolutional encoder that produced code; the format of trellis is described in "Trellis Description of a Convolutional Code" and the reference page for the istrellis function. code contains one or more symbols, each of which consists of log2(trellis.numOutputSymbols) bits. Each symbol in the vector decoded consists of log2(trellis.numInputSymbols) bits. tblen is a positive integer scalar that specifies the traceback depth. If the code rate is \(1 / 2\), a typical value for tblen is about five times the constraint length of the code.

The string opmode indicates the decoder's operation mode and its assumptions about the corresponding encoder's operation. Choices are in the table below.

\section*{Values of opmode Input}
\begin{tabular}{l|l}
\hline Value & Meaning \\
\hline 'cont ' & \begin{tabular}{l} 
The encoder is assumed to have started at the \\
all-zeros state. The decoder traces back from the \\
state with the best metric. A delay equal to tblen \\
symbols elapses before the first decoded symbol \\
appears in the output. This mode is appropriate \\
when you invoke this function repeatedly and want to \\
preserve continuity between successive invocations. \\
See the continuous operation mode syntaxes below.
\end{tabular} \\
\hline 'term' & \begin{tabular}{l} 
The encoder is assumed to have both started and \\
ended at the all-zeros state, which is true for the \\
default syntax of the convenc function. The decoder \\
traces back from the all-zeros state. This mode \\
incurs no delay. This mode is appropriate when the \\
uncoded message (that is, the input to convenc) has \\
enough zeros at the end to fill all memory registers \\
of the encoder. If the encoder has k input streams \\
and constraint length vector constr (using the \\
polynomial description of the encoder), "enough" \\
means k*max (constr-1).
\end{tabular} \\
\hline 'trunc ' & \begin{tabular}{l} 
The encoder is assumed to have started at the \\
all-zeros state. The decoder traces back from the \\
state with the best metric. This mode incurs no \\
delay. This mode is appropriate when you cannot \\
assume the encoder ended at the all-zeros state and \\
when you do not want to preserve continuity between \\
successive invocations of this function.
\end{tabular} \\
\hline
\end{tabular}

For the 'term' and 'trunc' mode, the traceback depth (tblen) must be a positive integer scalar value, not greater than the number of input symbols in code.
The string dectype indicates the type of decision that the decoder makes, and influences the type of data the decoder expects in code. Choices are in the table below.

\section*{Values of dectype Input}
\begin{tabular}{l|l}
\hline Value & Meaning \\
\hline 'unquant' & \begin{tabular}{l} 
code contains real input values, \\
where 1 represents a logical zero \\
and -1 represents a logical one.
\end{tabular} \\
\hline 'hard' & \begin{tabular}{l} 
code contains binary input \\
values.
\end{tabular} \\
\hline 'soft' & \begin{tabular}{l} 
For soft-decision decoding, use the \\
syntax below. nsdec is required \\
for soft-decision decoding.
\end{tabular} \\
\hline
\end{tabular}

\section*{Syntax for Soft Decision Decoding}
decoded = vitdec(code,trellis,tblen,opmode,'soft',nsdec) decodes the vector code using soft-decision decoding. code consists of integers between 0 and \(2^{\wedge}\) nsdec-1, where 0 represents the most confident 0 and \(2^{\wedge}\) nsdec-1 represents the most confident 1 . The existing implementation of the functionality supports up to 13 bits of quantization, meaning nsdec can be set up to 13 . For reference, 3 bits of quantization is about 2 db better than hard decision decoding.

\section*{Syntax for Punctures and Erasures}
decoded = ...
vitdec(code, trellis, tblen, opmode, dectype, puncpat) denotes the input punctured code, where puncpat is the puncture pattern vector, and where 0 s indicate punctured bits in the input code.
```

decoded = ...

```
vitdec(code,trellis,tblen, opmode, dectype, puncpat, eraspat) allows an erasure pattern vector, eraspat, to be specified for the input code, where the 1 s indicate the corresponding erasures. eraspat and code must be of the same length. If puncturing is not used, specify puncpat to be []. In the eraspat vector, 1 s indicate erasures in the input code.

\section*{Additional Syntaxes for Continuous Operation Mode}

Continuous operation mode enables you to save the decoder's internal state information for use in a subsequent invocation of this function. Repeated calls to this function are useful if your data is partitioned into a series of smaller vectors that you process within a loop, for example.
decoded = ...
vitdec(...,'cont',...,initmetric,initstates,initinputs) is the same as the earlier syntaxes, except that the decoder starts with its state metrics, traceback states, and traceback inputs specified by initmetric, initstates, and initinputs, respectively. Each real number in initmetric represents the starting state metric of the corresponding state. initstates and initinputs jointly specify the initial traceback memory of the decoder; both are trellis.numStates-by-tblen matrices. initstates consists of integers between 0 and trellis. numStates-1. If the encoder schematic has more than one input stream, the shift register that receives the first input stream provides the least significant bits in initstates, while the shift register that receives the last input stream provides the most significant bits in initstates. The vector initinputs consists of integers between 0 and trellis.numInputSymbols-1. To use default values for all of the last three arguments, specify them as [ ] , [ ], [ ].
[decoded,finalmetric,finalstates,finalinputs] = ... vitdec (..., 'cont',...) is the same as the earlier syntaxes, except that the final three output arguments return the state metrics, traceback states, and traceback inputs, respectively, at the end of the decoding process. finalmetric is a vector with trellis.numStates elements that correspond to the final state metrics. finalstates and finalinputs are both matrices of size trellis.numStates-by-tblen. The elements of finalstates have the same format as those of initstates.

\section*{Traceback Matrices}

The \(t^{\text {th }}\) column of \(P_{1}\) shows the \(t-1^{\text {th }}\) time step states given the inputs listed in the input matrix. For example, the value in the \(i^{\text {th }}\) row shows the state at time \(t-1\) that transitions to the \(i-1\) state at time \(t\). The input
required for this state transition is given in the \(i^{\text {th }}\) row of the \(t^{\text {th }}\) column of the input matrix.

The \(P_{1}\) output is the states of the traceback matrix. It is a [number of states x traceback length] matrix. The following example uses a \((7,5)\), rate \(1 / 2\) code. This code is easy to follow:
```

t = poly2trellis(3,[7 5]);
k = log2(t.numInputSymbols);
msg = [11001100110011001100110011001100];
code = convenc(msg,t); tblen = 15; [d1 m1 p1
in1]=vitdec(code(1:end/2),t,tblen,'cont','hard')
m1 =
0 3 2 3
p1 =

| 0 | 1 | 1 | 0 | 0 | 1 | 1 | 0 | 0 | 1 | 1 | 0 |
| :--- | :--- | :--- | :--- | :--- | :--- | :--- | :--- | :--- | :--- | :--- | :--- |
| 2 | 3 | 3 | 2 | 2 | 3 | 3 | 2 | 2 | 3 | 3 | 2 |
| 0 | 1 | 1 | 0 | 0 | 1 | 1 | 0 | 0 | 1 | 1 | 0 |
| 2 | 3 | 3 | 2 | 2 | 3 | 3 | 2 | 2 | 3 | 3 | 2 |

in1 =

| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| :--- | :--- | :--- | :--- | :--- | :--- | :--- | :--- | :--- | :--- | :--- |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 |
| 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 |

```

In this example, the message makes the encoder states follow the following sequence:
\(0231 / 0231\) / ...
Since the best state is 0 (column index of smallest metric in \(m_{1}-1\) ), the traceback matrix starts from sate 0 , looking at the first row ( \(0^{\text {th }}\) state)
of the last column of \(P_{1},([1 ; 3 ; 1 ; 3])\), which is 1 . This indicates 1 for the previous state.

Next, the traceback matrix checks in1 ( \([0 ; 0 ; 1 ; 1]\) ), which indicates 0 for the input. The second row (1st state) of the \(14^{\text {th }}\) column of \(P_{1}\) ( \([1 ; 3 ; 1\); \(3]\) ) is 3 . This indicates 3 for the previous state.

The traceback matrix checks in1 \(([0 ; 0 ; 1 ; 1])\), which indicates that the input was 0 . The fourth row (3rd state) of the 13 th column of \(P_{1}\) ( \([0 ; 2\); \(0 ; 2]\) ), is 2 . This indicates 2 for the previous state.

The traceback matrix checks in1 \(([0 ; 0 ; 1 ; 1])\), which indicates the input was 1. The third row (2nd state) of the 12 th column of \(P_{1}([0 ; 2 ; 0 ; 2])\), is 0 . This indicates 0 for the previous state.
The traceback matrix checks in \(1([0 ; 0 ; 1 ; 1])\), which indicates the input was 1. The first row (0th state) of the 11 th column of \(P_{1}([1 ; 3 ; 1 ; 3])\), is 1. This indicates 1 for the previous state. Then, the matrix checks in1 ( \([0 ; 0 ; 1 ; 1]\) ), which indicates 0 for the input.

To determine the best state for a gicen time, use \(m_{1}\). The smallest number in \(m_{1}\) represents the best state.

\section*{Examples}

The example below encodes random data and adds noise. Then it decodes the noisy code three times to illustrate the three decision types that vitdec supports. For unquantized and soft decisions, the output of convenc does not have the same data type that vitdec expects for the input code, so it is necessary to manipulate ncode before invoking vitdec. That the bit error rate computations must account for the delay that the continuous operation mode incurs.
```

s = RandStream.create('mt19937ar', 'seed',131);
prevStream = RandStream.setGlobalStream(s); % seed for repeatability
trel = poly2trellis(3,[6 7]); % Define trellis.
msg = randi([0 1],100,1); % Random data
code = convenc(msg,trel); % Encode.
ncode = rem(code + randerr(200,1,[0 1;.95 .05]),2); % Add noise.
tblen = 3; % Traceback length
decoded1 = vitdec(ncode,trel,tblen,'cont','hard'); %Hard decision

```
```

% Use unquantized decisions.
ucode = 1-2*ncode; % +1 \& -1 represent zero \& one, respectively.
decoded2 = vitdec(ucode,trel,tblen,'cont','unquant');
% To prepare for soft-decision decoding, map to decision values.
[x,qcode] = quantiz(1-2*ncode,[-.75 -.5 -. 25 0 . 25 .5 .75],...
[7 6 5 4 3 2 1 0]); % Values in qcode are between 0 and 2^3-1.
decoded3 = vitdec(qcode',trel,tblen,'cont','soft',3);
% Compute bit error rates, using the fact that the decoder
% output is delayed by tblen symbols.
[n1,r1] = biterr(decoded1(tblen+1:end),msg(1:end-tblen));
[n2,r2] = biterr(decoded2(tblen+1:end),msg(1:end-tblen));
[n3,r3] = biterr(decoded3(tblen+1:end),msg(1:end-tblen));
disp(['The bit error rates are: ',num2str([r1 r2 r3])])
RandStream.setGlobalStream(prevStream); % restore default stream

```

The following example illustrates how to use the final state and initial state arguments when invoking vitdec repeatedly. [decoded4;decoded5] is the same as decoded6.
```

s = RandStream.create('mt19937ar', 'seed',131); % seed for repeatability
prevStream = RandStream.setGlobalStream(s);
trel = poly2trellis(3,[6 7]);
code = convenc(randi([0 1],100,1),trel);
% Decode part of code, recording final state for later use.
[decoded4,f1,f2,f3] = vitdec(code(1:100),trel,3,'cont','hard');
% Decode the rest of code, using state input arguments.
decoded5 = vitdec(code(101:200),trel,3,'cont','hard',f1,f2,f3);
% Decode the entire code in one step.
decoded6 = vitdec(code,trel,3,'cont','hard');
isequal(decoded6,[decoded4; decoded5])
RandStream.setGlobalStream(prevStream); % restore default stream

```

For additional examples, see "Convolutional Codes".
For some commonly used puncture patterns for specific rates and polynomials, see the last three references below.
References
See Also convenc \| poly2trellis \| istrellis
How To - viterbisim
- "Convolutional Codes"

\section*{Purpose Generate white Gaussian noise}

\section*{Syntax}
```

$y=w g n(m, n, p)$
$y=w g n(m, n, p, i m p)$
$y=w g n(m, n, p, i m p, s t a t e)$
y = wgn(..., powertype)
y $=$ wgn(...,outputtype)

```

\section*{Description}
\(y=w g n(m, n, p)\) generates an \(m\)-by-n matrix of white Gaussian noise. \(p\) specifies the power of \(y\) in decibels relative to a watt. The default load impedance is 1 ohm.
\(y=w g n(m, n, p, i m p)\) is the same as the previous syntax, except that imp specifies the load impedance in ohms.
\(\mathrm{y}=\operatorname{wgn}(\mathrm{m}, \mathrm{n}, \mathrm{p}, \mathrm{imp}, \mathrm{s})\) uses \(s\), which is a random stream handle, to generate random noise samples with randn. This syntax is useful to generate repeatable outputs. Type help RandStream for more information.
\(y=w g n(m, n, p, i m p, s t a t e)\) is the same as the previous syntax, except that wgn first resets the state of the normal random number generator randn to the integer state.

Note This usage is deprecated and may be removed in a future release. Instead of state, use s, as in the previous example.
\(\mathrm{y}=\mathrm{wgn}(. . .\), powertype) is the same as the previous syntaxes, except that the string powertype specifies the units of \(p\). Choices for powertype are 'dBW', 'dBm', and 'linear'.
\(\mathrm{y}=\mathrm{wgn}(. .\). ,outputtype \()\) is the same as the previous syntaxes, except that the string outputtype specifies whether the noise is real or complex. Choices for outputtype are 'real' and 'complex'. If outputtype is 'complex', then the real and imaginary parts of \(y\) each have a noise power of \(\mathrm{p} / 2\).

Note The unit of measure for the output of the wgn function is Volts. For power calculations, it is assumed that there is a load of 1 Ohm .

\section*{Examples To generate a column vector of length 100 containing real white Gaussian noise of power 0 dBW , use this command:}
\(\mathrm{y} 1=\operatorname{wgn}(100,1,0) ;\)

To generate a column vector of length 100 containing complex white Gaussian noise, each component of which has a noise power of 0 dBW , use this command:
y2 = wgn(100,1,0,'complex');

\section*{See Also \\ randn | awgn}

How To . "Sources and Sinks"

Blocks - Alphabetical List

\section*{A-Law Compressor}

Purpose Implement A-law compressor for source coding

\section*{Library}

\section*{Source Coding}

Description The A-Law Compressor block implements an A-law compressor for the input signal. The formula for the A-law compressor is
\[
y= \begin{cases}\frac{A|x|}{1+\log A} \operatorname{sgn}(x) & \text { for } 0 \leq|x| \leq \frac{V}{A} \\ \frac{V(1+\log (A|x| / V))}{1+\log A} \operatorname{sgn}(x) & \text { for } \frac{V}{A}<|x| \leq V\end{cases}
\]
where \(A\) is the A-law parameter of the compressor, \(V\) is the peak signal magnitude for \(x\), \(\log\) is the natural logarithm, and sgn is the signum function (sign in MATLAB software).

The most commonly used \(A\) value is 87.6 .
The input can have any shape or frame status. This block processes each vector element independently.

\section*{A-Law Compressor}

\section*{Dialog} Box


A value
The A-law parameter of the compressor.
Peak signal magnitude
The peak value of the input signal. This is also the peak value of the output signal.

\section*{Supported Data Type}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline In & • double \\
\hline Out & • double \\
\hline
\end{tabular}

Pair Block A-Law Expander
See Also
Mu-Law Compressor

\section*{References}
[1] Sklar, Bernard. Digital Communications: Fundamentals and Applications. Englewood Cliffs, N.J., Prentice-Hall, 1988.

\section*{A-Law Expander}

Purpose
Library
Description


Dialog Box

Implement A-law expander for source coding

\section*{Source Coding}

The A-Law Expander block recovers data that the A-Law Compressor block compressed. The formula for the A-law expander, shown below, is the inverse of the compressor function.
\[
x= \begin{cases}\frac{y(1+\log A)}{\mathrm{A}} & \text { for } 0 \leq|y| \leq \frac{\mathrm{V}}{1+\log \mathrm{A}} \\ \exp (|\mathrm{y}|(1+\log A) / V-1) \frac{V}{A} \operatorname{sgn}(y) & \text { for } \frac{\mathrm{V}}{1+\log \mathrm{A}}<|y| \leq V\end{cases}
\]

The input can have any shape or frame status. This block processes each vector element independently.


\footnotetext{
A value
The A-law parameter of the compressor.
}

\section*{A-Law Expander}

\section*{Peak signal magnitude}

The peak value of the input signal. This is also the peak value of the output signal.

Match these parameters to the ones in the corresponding A-Law Compressor block.

\section*{Supported Data Type}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline In & • double \\
\hline Out & • double \\
\hline
\end{tabular}

\author{
Pair Block A-Law Compressor
}

See Also Mu-Law Expander
References [1] Sklar, Bernard. Digital Communications: Fundamentals and Applications. Englewood Cliffs, N.J., Prentice-Hall, 1988.

Purpose

Library
Description

Algebraic Deinterleaver

Restore ordering of input symbols using algebraically derived permutation

Block sublibrary of Interleaving
The Algebraic Deinterleaver block restores the original ordering of a sequence that was interleaved using the Algebraic Interleaver block. In typical usage, the parameters in the two blocks have the same values.

The Number of elements parameter, \(N\), indicates how many numbers are in the input vector. This block accepts a column vector input signal.

The block accepts the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.
The Type parameter indicates the algebraic method that the block uses to generate the appropriate permutation table. Choices are Takeshita-Costello and Welch-Costas. Each of these methods has parameters and restrictions that are specific to it; these are described on the reference page for the Algebraic Interleaver block.

\section*{Algebraic Deinterleaver}

Dialog Box


\section*{Type}

The type of permutation table that the block uses for deinterleaving. Choices are Takeshita-Costello and Welch-Costas.

\section*{Number of elements}

The number of elements, \(N\), in the input vector.

\section*{Algebraic Deinterleaver}

\section*{Multiplicative factor}

The factor the block uses to compute the corresponding interleaver's cycle vector. This field appears only when you set Type to Takeshita-Costello.

\section*{Cyclic shift}

The amount by which the block shifts indices when creating the corresponding interleaver's permutation table. This field appears only when you set Type to Takeshita-Costello.

\section*{Primitive element}

An element of order \(N\) in the finite field GF(N+1). This field appears only if Type is set to Welch-Costas.

\section*{Pair Block}

See Also
General Block Deinterleaver
References
[1] Heegard, Chris and Stephen B. Wicker. Turbo Coding. Boston: Kluwer Academic Publishers, 1999.
[2] Takeshita, O. Y. and D. J. Costello, Jr. "New Classes Of Algebraic Interleavers for Turbo-Codes." Proc. 1998 IEEE International Symposium on Information Theory, Boston, Aug. 16-21, 1998. 419.

\section*{Algebraic Interleaver}

\section*{Purpose}

\section*{Library}

Description

Algebraic Interleaver

Reorder input symbols using algebraically derived permutation table

Block sublibrary of Interleaving
The Algebraic Interleaver block rearranges the elements of its input vector using a permutation that is algebraically derived. The Number of elements parameter, \(N\), indicates how many numbers are in the input vector. This block accepts a column vector input signal.
The block accepts the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.

The Type parameter indicates the algebraic method that the block uses to generate the appropriate permutation table. Choices are Takeshita-Costello and Welch-Costas. Each of these methods has parameters and restrictions that are specific to it:
- If you set Type to Welch-Costas, then \(N+1\) must be prime. The Primitive element parameter is an integer, \(A\), between 1 and \(N\) that represents a primitive element of the finite field GF \((N+1)\). This means that every nonzero element of \(\mathrm{GF}(N+1)\) can be expressed as \(A\) raised to some integer power.

In a Welch-Costas interleaver, the permutation maps the integer \(k\) to \(\bmod \left(A^{k}, N+1\right)-1\).
- If you set Type to Takeshita-Costello, then \(N\) must be \(2^{m}\) for some integer \(m\). The Multiplicative factor parameter, \(k\), must be an odd integer less than \(N\). The Cyclic shift parameter, \(h\), must be a nonnegative integer less than \(N\).

A Takeshita-Costello interleaver uses a length- \(N\) cycle vector whose \(n^{\text {th }}\) element is
\[
c(n)=\bmod \left(k \cdot \frac{n \cdot(n-1)}{2}, N\right)+1, n
\]

\section*{Algebraic Interleaver}
for integers n between 1 and N . The intermediate permutation function is obtained by using the following relationship:
\[
\Pi(c(n))=c(n+1)
\]
where
\[
n=1: N
\]

The interleaver's actual permutation vector is the result of cyclically shifting the elements of the permutation vector, п, by the Cyclic shift parameter, \(h\).

\section*{Algebraic Interleaver}

Function Block Parameters: Algebraic Interleaver
-Algebraic Interleaver (mask) (link)
Interleave the elements of the input vector using an algebraically derived permutation table.

For the Takeshita-Costello type interleaver, the Number of elements N must be a power of 2, the Multiplicative factor must be an odd integer less than \(N\), and the Cyclic shift must be a nonnegative integer less than N .

For the Welch-Costas type interleaver, the Number of elements N must be specified such that \(N+1\) is prime and the Primitive element must be a primitive element from GF(N+1).

In each case, the Number of elements must match the input signal width.
\begin{tabular}{l} 
Parameters \\
Type: Takeshita-Costello \\
Number of elements: \\
256 \\
Multiplicative factor: \\
\hline 13 \\
Cyclic shift: \\
0
\end{tabular}

Dialog Box


\section*{Type}

The type of permutation table that the block uses for interleaving.

\section*{Number of elements}

The number of elements, \(N\), in the input vector.

\section*{Multiplicative factor}

The factor used to compute the interleaver's cycle vector. This field appears only if Type is set to Takeshita-Costello.

\section*{Algebraic Interleaver}

\section*{Cyclic shift}

The amount by which the block shifts indices when creating the permutation table. This field appears only if Type is set to Takeshita-Costello.

\section*{Primitive element}

An element of order \(N\) in the finite field GF(N+1). This field appears only if Type is set to Welch-Costas.

\section*{Pair Block Algebraic Deinterleaver}

See Also General Block Interleaver
References [1] Heegard, Chris and Stephen B. Wicker. Turbo Coding. Boston: Kluwer Academic Publishers, 1999.
[2] Takeshita, O. Y. and D. J. Costello, Jr. "New Classes Of Algebraic Interleavers for Turbo-Codes." Proc. 1998 IEEE International Symposium on Information Theory, Boston, Aug. 16-21, 1998. 419.

\section*{Align Signals}

\section*{Purpose}

\section*{Library}

Description


Align two signals by finding delay between them

Utility Blocks
The Align Signals block aligns two signals by finding the delay between them. This is useful when you want to compare a transmitted and received signal to determine the bit error rate, but do not know the delay in the received signal. This block accepts a column vector or matrix input signal. For a matrix input, the block aligns each channel independently.

The s1 input port receives the original signal, while the s2 input port receives a delayed version. The two input signals must have the same dimensions and sample times. The block calculates the delay between the two signals, and then
- Delays the first signal, s1, by the calculated value, and outputs it through the port labeled s1 del.
- Outputs the second signal s2 without change through the port labeled s2.
- Outputs the delay value through the port labeled delay.

See "Delays" in the Communications System Toolbox online documentation for more information about signal delays.

The block's Correlation window length parameter specifies how many samples of the signals the block uses to calculate the cross-correlation. The delay output is a nonnegative integer less than the Correlation window length.

As the Correlation window length is increased, the reliability of the computed delay also increases. However, the processing time to compute the delay increases as well.

You can make the Align Signals block stop updating the delay after it computes the same delay value for a specified number of samples. To do so, select Disable recurring updates, and enter a positive integer in the Number of constant delay outputs to disable updates
field. For example, if you set Number of constant delay outputs to disable updates to 20 , the block will stop recalculating and updating the delay after it calculates the same value 20 times in succession. Disabling recurring updates causes the simulation to run faster after the target number of constant delays occurs.

\section*{Tips for Using the Block Effectively}
- Set the Correlation window length parameter sufficiently large so that the computed delay eventually stabilizes at a constant value. If the computed delay is not constant, you should increase Correlation window length. If the increased value of Correlation window length exceeds the duration of the simulation, then you should also increase the duration of the simulation accordingly.
- If the cross-correlation between the two signals is broad, then Correlation window length should be much larger than the expected delay, or else the algorithm might stabilize at an incorrect value. For example, a CPM signal has a broad autocorrelation, so it has a broad cross-correlation with a delayed version of itself. In this case, the Correlation window length value should be much larger than the expected delay.
- If the block calculates a delay that is greater than 75 percent of Correlation window length, the signal s1 is probably delayed relative to the signal s2. In this case, you should switch the signal lines leading into the two input ports.
- If you use the Align Signals block with the Error Rate Calculation block, you should set the Receive delay parameter of the Error Rate Calculation block to 0 because the Align Signals block compensates for the delay. Also, you might want to set the Error Rate Calculation block's Computation delay parameter to a nonzero value to account for the possibility that the Align Signals block takes a nonzero amount of time to stabilize on the correct amount by which to delay one of the signals.

\section*{Align Signals}

\section*{Examples}

Dialog Box

See the "Delays" section of Communications System Toolbox User's Guide for an example that uses the Align Signals block in conjunction with the Error Rate Calculation block.

See Setting the Correlation Window Length, on the reference page for the Find Delay block, for an example that illustrates how to set the correlation window length properly.


\section*{Correlation window length}

The number of samples the block uses to calculate the cross-correlations of the two signals.

\section*{Disable recurring updates}

Selecting this option causes the block to stop computing the delay after it computes the same delay value for a specified number of samples.

\section*{Number of constant delay outputs to disable updates}

A positive integer specifying how many times the block must compute the same delay before ceasing to update. This field appears only if Disable recurring updates is selected.

\section*{Align Signals}

Algorithm

See Also Find Delay, Error Rate Calculation

\section*{APP Decoder}

\section*{Purpose}

Decode convolutional code using a posteriori probability (APP) method

\section*{Library}

Description


Convolutional sublibrary of Error Detection and Correction
The APP Decoder block performs a posteriori probability (APP) decoding of a convolutional code.

\section*{Input Signals and Output Signals}

The input \(L(u)\) represents the sequence of log-likelihoods of encoder input bits, while the input \(L(c)\) represents the sequence of log-likelihoods of code bits. The outputs \(L(u)\) and \(L(c)\) are updated versions of these sequences, based on information about the encoder.

If the convolutional code uses an alphabet of \(2^{n}\) possible symbols, this block's \(L(c)\) vectors have length \(Q^{*} n\) for some positive integer \(Q\). Similarly, if the decoded data uses an alphabet of \(2^{k}\) possible output symbols, then this block's \(L(\mathrm{u})\) vectors have length \(Q^{*} k\).

This block accepts a column vector input signal with any positive integer for \(Q\).

If you only need the input \(L(c)\) and output \(L(u)\), you can attach a Simulink Ground block to the input \(L(u)\) and a Simulink Terminator block to the output \(L(c)\).

This block accepts single and double data types. Both inputs, however, must be of the same type. The output data type is the same as the input data type.

\section*{Specifying the Encoder}

To define the convolutional encoder that produced the coded input, use the Trellis structure parameter. This parameter is a MATLAB structure whose format is described in "Trellis Description of a Convolutional Code" in the Communications System Toolbox documentation. You can use this parameter field in two ways:
- If you have a variable in the MATLAB workspace that contains the trellis structure, enter its name as the Trellis structure parameter. This way is preferable because it causes Simulink to spend less time
updating the diagram at the beginning of each simulation, compared to the usage described next.
- If you want to specify the encoder using its constraint length, generator polynomials, and possibly feedback connection polynomials, use a poly2trellis command within the Trellis structure field. For example, to use an encoder with a constraint length of 7 , code generator polynomials of 171 and 133 (in octal numbers), and a feedback connection of 171 (in octal), set the Trellis structure parameter to
```

poly2trellis(7,[171 133],171)

```

To indicate how the encoder treats the trellis at the beginning and end of each frame, set the Termination method parameter to either Truncated or Terminated. The Truncated option indicates that the encoder resets to the all-zeros state at the beginning of each frame. The Terminated option indicates that the encoder forces the trellis to end each frame in the all-zeros state. If you use the Convolutional Encoder block with the Operation mode parameter set to Truncated (reset every frame), use the Truncated option in this block. If you use the Convolutional Encoder block with the Operation mode parameter set to Terminate trellis by appending bits, use the Terminated option in this block.

\section*{Specifying Details of the Algorithm}

You can control part of the decoding algorithm using the Algorithm parameter. The True APP option implements a posteriori probability decoding as per equations 20-23 in section V of [1]. To gain speed, both the Max* and Max options approximate expressions like
\[
\log \sum_{i} \exp \left(a_{i}\right)
\]
by other quantities. The Max option uses \(\max \left(a_{\mathrm{i}}\right)\) as the approximation, while the Max* option uses \(\max \left(a_{\mathrm{i}}\right)\) plus a correction term given by
\[
\ln \left(1+\exp \left(-\left|a_{i-1}-a_{i}\right|\right)\right)[3] .
\]

The Max* option enables the Scaling bits parameter in the dialog box. This parameter is the number of bits by which the block scales the data it processes internally (multiplies the input by ( \(2^{\wedge}\) numScalingBits) and divides the pre-output by the same factor). Use this parameter to avoid losing precision during the computations.

Function Block Parameters: APP Decoder
APP Decoder (mask) (link)
A posteriori probability (APP) decoder. Use the poly2trellis function to create a trellis using the constraint length, code generator (octal), and feedback connection (octal).

Parameters
Trellis structure:
poly2trellis(7, [171 133], 171)|
Termination method: Truncated
Algorithm:
Max*
Number of scaling bits:
3
Disable L(c) output port

\section*{Trellis structure}

MATLAB structure that contains the trellis description of the convolutional encoder.

\section*{Termination method}

Either Truncated or Terminated. This parameter indicates how the convolutional encoder treats the trellis at the beginning and end of frames.

\section*{Algorithm}

Either True APP, Max*, or Max.

\section*{Number of scaling bits}

An integer between 0 and 8 that indicates by how many bits the decoder scales data in order to avoid losing precision. This field is active only when Algorithm is set to Max*.

\section*{Disable L(c) output port}

Select this check box to disable the secondary block output, L(c).

\section*{Examples}

See Also Viterbi Decoder, Convolutional Encoder;poly2trellis
References
For an example using this block, see the Iterative Decoding of a Serially Concatenated Convolutional Code example.
[1] Benedetto, S., G. Montorsi, D. Divsalar, and F. Pollara, "A Soft-Input Soft-Output Maximum A Posterior (MAP) Module to Decode Parallel and Serial Concatenated Codes," JPL TDA Progress Report, Vol. 42-127, November 1996.
[2] Benedetto, Sergio and Guido Montorsi, "Performance of Continuous and Blockwise Decoded Turbo Codes." IEEE Communications Letters, Vol. 1, May 1997, 77-79.
[3] Viterbi, Andrew J., "An Intuitive Justification and a Simplified Implementation of the MAP Decoder for Convolutional Codes," IEEE Journal on Selected Areas in Communications, Vol. 16, February 1998, 260-264.

\section*{Purpose}

\section*{Library}

Description


\section*{Dialog}

Box

Adaptively adjust gain for constant signal-level output
RF Impairments Correction
This automatic gain controller (AGC) block adaptively adjusts its gain to achieve a constant signal level at the output.
\begin{tabular}{|c|c|c|c|}
\hline \multicolumn{3}{|l|}{F Function Block Parameters: AGC} & \(x^{-}\) \\
\hline \multicolumn{4}{|l|}{comm.AGC} \\
\hline \multicolumn{4}{|l|}{Automatic gain controller} \\
\hline \multicolumn{4}{|l|}{View source} \\
\hline Detector method: & Rectifier & & \(\checkmark\) \\
\hline Loop method: & Linear & & \(\checkmark\) \\
\hline Period of gain updates in samples: & & & \\
\hline Step size: & 0.1 & & \\
\hline Maximum gain in dB: & 30 & & \\
\hline OK C & ncel & Help & Apply \\
\hline
\end{tabular}

\section*{Detector method}

Specify the method that the block uses to perform envelope detection. The default is Rectifier.

When you select Rectifier, the AGC detector outputs a voltage value proportional to the envelope amplitude of the output signal. The detector rectifies and then averages the input signal over the period of gain updates in samples. The AGC adjusts the gain to obtain unity voltage at the output of the detector.

When you select Square law, the AGC detector outputs a power value that is proportional to the square of the output voltage. The detector squares and then averages the input signal over the period of gain updates in samples. The AGC adjusts the gain to obtain unity power at the output of the detector.

\section*{Loop method}

Specify the AGC loop implementation that the block uses. The default is Linear.

When you select Linear, the AGC uses the direct value of the detector output to determine the gain value. Typically, a linear loop responds quickly to increases in the input signal level. However, the loop's response to decreases in the input signal level tends to be slow.

When you select Logarithmic, the AGC uses the logarithm of the detector output to determine the gain value. Logarithmic loops respond to decreases in the input signal level much more quickly than linear loops.

\section*{Period of gain updates in samples}

Specify the period of the gain updates as a double- or single-precision, real, integer-valued scalar. The default is 100 .

The number of input samples must be an integer multiple of this parameter value. Setting the period greater than 1 increases the speed of the AGC algorithm.

If you increase the period of the gain updates, you may also need to increase the step size. Similarly, if you decrease the period of the gain updates, you may also need to decrease the step size.

\section*{Step size}

Specify the step size for gain updates as a double- or single-precision, real, positive scalar. The default is 0.1.

If you increase the loop gain, the AGC responds to changes at the input signal level faster. However, gain pumping also increase.

If you increase the period of the gain updates, you may also need to increase the step size. Similarly, if you decrease the period of the gain updates, you may also need to decrease the step size.

\section*{Maximum gain in dB}

Specify the maximum gain of the AGC in decibels as a positive scalar. The default is 30 .

If the input signal to the AGC has a very low signal level, the AGC gain may increase rapidly. Use this parameter to limit the gain that the AGC applies to the input signal.

\section*{Algorithms Linear Loop AGC}

In a linear loop AGC, the detector uses its output directly to generate an error signal. After applying a step size, the AGC passes the error signal to an integrator. The output of the integrator is used as the variable gain. Linear loop AGCs are limited by their decay, or slew, characteristics. In other words, they respond to input signal increases much more quickly than they respond to input signal decreases.

\section*{AGC Block}

\[
\begin{aligned}
& y(n)=g(n) \cdot x(n) \\
& e(n)=A-z(m) \\
& g(n+1)=g(n)+K \cdot e(n) ;
\end{aligned}
\]
where
\(A\) represents the reference value, which is 1
\(K\) represents the step size
\(e\) represents the error signal
\(g\) represents the gain
\(x\) represents the input signal \(y\) represents the output signal \(z\) represents the detector output

\section*{AGC Block}

\section*{Logarithmic Loop AGC}

In a logarithmic loop AGC, the logarithm of the ratio of the detector output and the reference signal represents the error signal. A logarithmic loop uses the exponential of the integrator output as the gain signal. Logarithmic loop AGCs have the same response time to both increases or decreases to the input signal amplitude.


The logarithmic loop has longer attack and decay times. However, the gain pumping of the logarithmic loop is better than that of the linear loop.
\[
\begin{aligned}
& y(n)=e^{g(n)} \cdot x(n) \\
& e(n)=\ln (A)-\ln (z(m)) \\
& g(n+1)=g(n)+K \cdot e(n)
\end{aligned}
\]
where
\(A\) represents the reference value, which is 1
\(K\) represents the step size
\(e\) represents the error signal
\(g\) represents the gain
\(x\) represents the input signal
\(y\) represents the output signal
\(z\) represents the detector output

\section*{AGC Detector}

Two AGC detectors are available:

\section*{Rectifier}
\(\mathrm{z}=|\mathrm{y}|\) when the detector represents a rectifier
\[
z(m)=\frac{1}{N} \sum_{n=m N}^{(m+1) N-1}|y(n)|
\]
where \(N\) represents the period of the gain updates

\section*{Square Law}
\(z=|y|^{2}\) represents the square law detector
\[
z(m)=\frac{1}{N} \sum_{n=m N}^{(m+1) N-1}|y(n)|^{2}
\]
where \(N\) represents the period of the gain updates

\section*{Performance Considerations}

There are three performance criteria for AGCs:
- Attack time: The duration it takes the AGC to respond to an increase in the input amplitude.
- Decay time: The duration it takes the AGC to respond to a decrease in the input amplitude.
- Gain pumping: The variation in the gain value during steady-state operation.

Increasing the step size decreases the attack time and decay times, but it also increases gain pumping.

\section*{Examples}
- To open an example that adaptively adjusts the received signal amplitude to approximately 1 volt, type doc_agc_received_signal_amplitude at the MATLAB command line.
- To open an example that compare the performance of an AGC with a rectifier detector and a square law detector, type doc_agc_compare_rectifier_and_square_law at the MATLAB command line.
- Top open an example that plots the effect of step size on AGC performance, type doc_agc_plot_step_size at the MATLAB command line.
- To open an example that plots the effect of maximum gain on burst signals, type doc_agc_plot_max_gain at the MATLAB command line.

\section*{AWGN Channel}
\begin{tabular}{ll} 
Purpose & Add white Gaussian noise to input signal \\
Library & Channels
\end{tabular}

Description The AWGN Channel block adds white Gaussian noise to a real or complex input signal. When the input signal is real, this block adds AMOGN \(^{\text {rin }}\) real Gaussian noise and produces a real output signal. When the input signal is complex, this block adds complex Gaussian noise and produces a complex output signal. This block inherits its sample time from the input signal.

This block accepts a scalar-valued, vector, or matrix input signal with a data type of type single or double. The output signal inherits port data types from the signals that drive the block.

Note All values of power assume a nominal impedance of 1 ohm .

\section*{Signal Processing and Input Dimensions}

This block can process multichannel signals. When you set the Input Processing parameter to Columns as channels (frame based), the block accepts an \(M\)-by- \(N\) input signal. \(M\) specifies the number of samples per channel and \(N\) specifies the number of channels. Both \(M\) and \(N\) can be equal to 1 . The block adds frames of length- \(M\) Gaussian noise to each of the \(N\) channels, using a distinct random distribution per channel.

\section*{Specifying the Variance Directly or Indirectly}

You can specify the variance of the noise generated by the AWGN Channel block using one of these modes:
- Signal to noise ratio (Eb/No), where the block calculates the variance from these quantities that you specify in the dialog box:
- Eb/No, the ratio of bit energy to noise power spectral density
- Number of bits per symbol
- Input signal power, the actual power of the symbols at the input of the block
- Symbol period
- Signal to noise ratio (Es/No), where the block calculates the variance from these quantities that you specify in the dialog box:
- Es/No, the ratio of signal energy to noise power spectral density
- Input signal power, the actual power of the symbols at the input of the block

\section*{- Symbol period}
- Signal to noise ratio (SNR), where the block calculates the variance from these quantities that you specify in the dialog box:
- SNR, the ratio of signal power to noise power
- Input signal power, the actual power of the samples at the input of the block
- Variance from mask, where you specify the variance in the dialog box. The value must be positive.
- Variance from port, where you provide the variance as an input to the block. The variance input must be positive, and its sampling rate must equal that of the input signal.

Changing the symbol period in the AWGN Channel block affects the variance of the noise added per sample, which also causes a change in the final error rate.
\[
\text { NoiseVariance }=\frac{\text { SignalPower } \times \text { SymbolPeriod }}{\text { SampleTime } \times 10^{\frac{E s / N o}{10}}}
\]

A good rule of thumb for selecting the Symbol period value is to set it to be what you model as the symbol period in the model. The value would depend upon what constitutes a symbol and what the
oversampling applied to it is (e.g., a symbol could have 3 bits and be oversampled by 4).
In both Variance from mask mode and Variance from port mode, these rules describe how the block interprets the variance:
- If the variance is a scalar, then all signal channels are uncorrelated but share the same variance.
- If the variance is a vector whose length is the number of channels in the input signal, then each element represents the variance of the corresponding signal channel.

Note If you apply complex input signals to the AWGN Channel block, then it adds complex zero-mean Gaussian noise with the calculated or specified variance. The variance of each of the quadrature components of the complex noise is half of the calculated or specified value.

\section*{Relationship Among Eb/No, Es/No, and SNR Modes}

For complex input signals, the AWGN Channel block relates \(\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}\), \(\mathrm{E}_{\mathrm{s}} / \mathrm{N}_{0}\), and SNR according to the following equations:
\[
\begin{aligned}
& \mathrm{E}_{\mathrm{s}} / \mathrm{N}_{0}=\left(\mathrm{T}_{\text {sym }} / \mathrm{T}_{\text {samp }}\right) \cdot \mathrm{SNR} \\
& \mathrm{E}_{\mathrm{s}} / \mathrm{N}_{0}=\mathrm{E}_{\mathrm{b}} / \mathrm{N}_{0}+10 \log _{10}(\mathrm{k}) \text { in } \mathrm{dB}
\end{aligned}
\]
where
- \(\mathrm{E}_{\mathrm{s}}=\) Signal energy (Joules)
- \(\mathrm{E}_{\mathrm{b}}=\) Bit energy (Joules)
- \(\mathrm{N}_{0}=\) Noise power spectral density (Watts/Hz)
- \(\mathrm{T}_{\text {sym }}\) is the Symbol period parameter of the block in Es/No mode
- k is the number of information bits per input symbol

\section*{AWGN Channel}
- \(\mathrm{T}_{\text {samp }}\) is the inherited sample time of the block, in seconds

For real signal inputs, the AWGN Channel block relates \(\mathrm{E}_{\mathrm{s}} / \mathrm{N}_{0}\) and SNR according to the following equation:
\[
\mathrm{E}_{\mathrm{s}} / \mathrm{N}_{0}=0.5\left(\mathrm{~T}_{\text {sym }} / \mathrm{T}_{\text {samp }}\right) \cdot \mathrm{SNR}
\]

Note that the equation for the real case differs from the corresponding equation for the complex case by a factor of 2 . This is so because the block uses a noise power spectral density of \(N_{0} / 2 \mathrm{Watts} / \mathrm{Hz}\) for real input signals, versus \(N_{0} \mathrm{Watts} / \mathrm{Hz}\) for complex signals.
For more information about these quantities, see "AWGN Channel Noise Level" in the Communications System Toolbox documentation.

\section*{Tunable Block Parameters}

The following table indicates which parameters are tunable, for different block modes.
\begin{tabular}{l|l}
\hline Mode & Tunable Parameters \\
\hline Eb/No & Eb/No, Input signal power \\
\hline Es/No & Es/No, Input signal power \\
\hline SNR & SNR, Input signal power \\
\hline Variance from mask & Variance \\
\hline
\end{tabular}

You can tune parameters in normal mode, Accelerator mode and the Rapid Accelerator mode.
If you use the Simulink Coder \({ }^{\text {TM }}\) rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameters listed in the previous table without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

\section*{AWGN Channel}


\section*{Input processing}

Specify how the block processes the input signal. You can set this parameter to one of the following options:
- Columns as channels (frame based) - When you select this option, the block treats each column of the input as a separate channel.

\section*{AWGN Channel}

Note The Inherited (this choice will be removed - see release notes) option will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

\section*{Initial seed}

The seed for the Gaussian noise generator.
This block uses the DSP System Toolbox Random Source block to generate noise. Random numbers are generated using the Ziggurat method. The Initial seed parameter in this block initializes the noise generator. Initial seed can be either a scalar or a vector with a length that matches the number of channels in the input signal. Each time you run a simulation, this block outputs the same signal. The first time you run the simulation, the block randomly selects an initial seed. The block reuses the same initial seeds every time you rerun the simulation.

This property is a tunable and allows you to specify different seed values for each DLL build.

\section*{Mode}

The mode by which you specify the noise variance: Signal to noise ratio (Eb/No), Signal to noise ratio (Es/No), Signal to noise ratio (SNR), Variance from mask, or Variance from port.

\section*{Eb/No (dB)}

The ratio of information (i.e., without channel coding) bit energy per symbol to noise power spectral density, in decibels. This field appears only if Mode is set to Eb/No.

\section*{Es/No (dB)}

The ratio of information (i.e., without channel coding) symbol energy per symbol to noise power spectral density, in decibels. This field appears only if Mode is set to Es/No.

\section*{AWGN Channel}

\section*{SNR (dB)}

The ratio of signal power to noise power, in decibels. This field appears only if Mode is set to SNR.

\section*{Number of bits per symbol}

The number of bits in each input symbol. This field appears only if Mode is set to Eb/No.

\section*{Input signal power, referenced to 1 ohm (watts)}

The mean square power of the input symbols (if Mode is Eb/No or Es/No) or input samples (if Mode is SNR), in watts. This field appears only if Mode is set to Eb/No, Es/No, or SNR.

\section*{Symbol period (s)}

The duration of an information channel (i.e., without channel coding) symbol, in seconds. This field appears only if Mode is set to Eb/No or Es/No.

\section*{Variance}

The variance of the white Gaussian noise. This field appears only if Mode is set to Variance from mask.

\section*{Examples Many documentation examples use this block, including:}
- Gray Coded 8-PSK (EbNo mode)
- Phase Noise Effects in 256-QAM (SNR mode)
- Discrete Multitone Signaling (Variance from mask mode)
- "Filter Using Simulink Raised Cosine Filter Blocks"

See Also Random Source (DSP System Toolbox documentation)
Reference [1] Proakis, John G., Digital Communications, 4th Ed., McGraw-Hill, 2001.

\section*{Barker Code Generator}

\section*{Purpose}

Generate Barker Code

\section*{Library}

Description

Sequence Generators sublibrary of Comm Sources
Barker codes, which are subsets of PN sequences, are commonly used for frame synchronization in digital communication systems. Barker codes have length at most 13 and have low correlation sidelobes. A correlation sidelobe is the correlation of a codeword with a time-shifted version of itself. The correlation sidelobe, \(\mathrm{C}_{\mathrm{k}}\), for a k -symbol shift of an N -bit code sequence, \(\left\{\mathrm{X}_{\mathrm{j}}\right\}\), is given by
\[
C_{k}=\sum_{j=1}^{N-k} X_{j} X_{j+k}
\]
where \(\mathrm{X}_{\mathrm{j}}\) is an individual code symbol taking values +1 or -1 for \(\mathrm{j}=1,2,3, \ldots, \mathrm{~N}\), and the adjacent symbols are assumed to be zero.

The Barker Code Generator block provides the codes listed in the following table:
\begin{tabular}{|c|c|}
\hline Code length & Barker Code \\
\hline 1 & [-1] \\
\hline 2 & [-1 1]; \\
\hline 3 & \(\left[\begin{array}{lll}-1 & -1 & 1\end{array}\right]\) \\
\hline 4 & \(\left[\begin{array}{llll}-1 & -1 & 1 & -1\end{array}\right]\) \\
\hline 5 & \(\left[\begin{array}{llllll}-1 & -1 & -1 & 1 & -1\end{array}\right]\) \\
\hline 7 & \(\left[\begin{array}{llllllll}-1 & -1 & -1 & 1 & 1 & -1 & 1\end{array}\right]\) \\
\hline 11 & \(\left[\begin{array}{lllllllllllll}-1 & -1 & -1 & 1 & 1 & 1 & -1 & 1 & 1 & -1 & 1\end{array}\right]\) \\
\hline 13 & \(\left[\begin{array}{lllllllllllllllll}-1 & -1 & -1 & -1 & -1 & 1 & 1 & -1 & -1 & 1 & -1 & 1 & -1\end{array}\right]\) \\
\hline
\end{tabular}

\section*{Barker Code Generator}


\author{
See Also PN Sequence Generator
}

\section*{Baseband PLL}

\section*{Purpose Implement baseband phase-locked loop \\ Library \\ Components sublibrary of Synchronization}

Description

The Baseband PLL (phase-locked loop) block is a feedback control system that automatically adjusts the phase of a locally generated signal to match the phase of an input signal. Unlike the Phase-Locked Loop block, this block uses a baseband method and does not depend on a carrier frequency.

This PLL has these three components:
- An integrator used as a phase detector.
- A filter. You specify the filter's transfer function using the Lowpass filter numerator and Lowpass filter denominator parameters. Each is a vector that gives the respective polynomial's coefficients in order of descending powers of \(s\).

To design a filter, you can use the Signal Processing Toolbox \({ }^{\text {TM }}\) functions cheby1, and cheby2. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.
[num, den] = cheby2(3,40,100,'s')
- A voltage-controlled oscillator (VCO). You specify the sensitivity of the VCO signal to its input using the VCO input sensitivity parameter. This parameter, measured in Hertz per volt, is a scale factor that determines how much the VCO shifts from its quiescent frequency.

This block accepts a sample-based scalar signal. The input signal represents the received signal. The three output ports produce:
- The output of the filter
- The output of the phase detector
- The output of the VCO

\section*{Baseband PLL}

This model is nonlinear; for a linearized version, use the Linearized Baseband PLL block.


\section*{Lowpass filter numerator}

The numerator of the lowpass filter's transfer function, represented as a vector that lists the coefficients in order of descending powers of \(s\).

\section*{Lowpass filter denominator}

The denominator of the lowpass filter's transfer function, represented as a vector that lists the coefficients in order of descending powers of \(s\).

\section*{VCO input sensitivity ( \(\mathrm{Hz} / \mathrm{V}\) )}

This value scales the input to the VCO and, consequently, the shift from the VCO's quiescent frequency.

\author{
See Also Linearized Baseband PLL, Phase-Locked Loop
}

References For more information about phase-locked loops, see the works listed in "Selected Bibliography for Synchronization" in Communications System Toolbox User's Guide .

\section*{BCH Decoder}
\begin{tabular}{ll} 
Purpose & Decode BCH code to recover binary vector data \\
Library & Block sublibrary of Error Detection and Correction
\end{tabular}

Description The BCH Decoder block recovers a binary message vector from a binary BCH codeword vector. For proper decoding, the first two parameter values in this block should match the parameters in the corresponding BCH Encoder block.

This block accepts a column vector input signal with an integer multiple of ( \(N\) - the number of punctures) elements. Each group of \(N\) input elements represents one codeword to be decoded. The values of ( \(N+\) shortening length) and ( \(K+\) shortening length) must produce a valid narrow-sense BCH code.

If the decoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

For a given codeword length \(N\), only specific message lengths \(K\) are valid for a BCH code. For a full length BCH code, \(N\) must be of the form \(2^{M}\) - 1 , where \(3 \leq M \leq 16\). If \(N\) is less than \(2^{M}-1\), the block assumes that the code has been shortened by length \(2^{M-1}-N\). However, if \(N\) is greater than or equal to \(2^{M-1}\), Primitive polynomial must be specified to appropriately set the value of \(M\).

No known analytic formula describes the relationship among the codeword length, message length, and error-correction capability. For a list of some valid values of \(K\) corresponding to values of \(N\) up to 511, see the BCH Encoder reference page in the Communications System Toolbox documentation.

The primitive and generator polynomials may be specified in their respective fields, which appear after selecting their corresponding check boxes.

To have the block output error information, select Output number of corrected errors. Selecting this option causes a second output port to appear. The second output is the number of errors detected during

\section*{BCH Decoder}
decoding of the codeword. A negative integer indicates that the block detected more errors than it could correct using the coding scheme.

In the case of a decoder failure, the message portion of the decoder input is returned unchanged as the decoder output.

The sample times of all input and output signals are equal.
For information about the data types each block port supports, see the "Supported Data Type" on page 2-44 table on this page.

\section*{Punctured Codes}

This block supports puncturing when you select Punctured code. This selection enables the Puncture vector parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 represents that the data symbol passes unaltered, and 0 represents that the data symbol gets punctured, or removed, from the data stream. This convention applies for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

Note 1s and 0s have precisely opposite meanings for the puncture and erasure vectors. For an erasure vector, 1 means that the data symbol is to be replaced with an erasure symbol, and 0 means that the data symbol is passed unaltered. This convention is carried for both the encoder and the decoder.

\section*{BCH Decoder}


\section*{Codeword length, \(\mathbf{N}\)}

The codeword length.

\section*{Message length, \(K\)}

The message length.

\section*{Specify primitive polynomial}

Selecting this check box enables the Primitive polynomial field.

\section*{Primitive polynomial}

A row vector that represents the binary coefficients of the primitive polynomial in order of descending powers.

This field defaults to de2bi(primpoly(4, 'nodisplay'), 'left-msb'), corresponding to a \((15,5)\) code.

This parameter appears only when you select Specify primitive polynomial.

\section*{Specify generator polynomial}

Selecting this check box enables the Generator polynomial field.

\section*{Generator polynomial}

A row vector that represents the binary coefficients of the generator polynomial in order of descending powers.

The length of the Generator polynomial must be \(N-K+1\).
This field defaults to bchgenpoly \((15,5)\).
This parameter appears only when you select Specify generator polynomial.

\section*{Disable generator polynomial checking}

Each time a model initializes, the block performs a polynomial check. This check verifies that \(X^{N}+1\) is divisible by the user-defined generator polynomial, where \(N\) represents the full code word length. Selecting this check box disables the polynomial check. For larger codes, disabling the check speeds up the simulation process. You should always run the check at least once before disabling this feature.

This check box appears only when you select Specify generator polynomial.

\section*{Puncture code}

Selecting this check box enables the field Puncture vector.

\section*{BCH Decoder}

\section*{Puncture vector}

This parameter appears only when you select Puncture code.
A column vector of length \(N-K\). In the Puncture vector, a value of 1 represents that the data symbol passes unaltered, and 0 represents that the data symbol gets punctured, or removed, from the data stream.

The default value is \([\) ones \((8,1)\); zeros \((2,1)]\).

\section*{Enable erasures input port}

Selecting this check box will open the Era port.
Through the Era port, you can input a binary column vector the same size as the codeword input.

Erasure values of 1 correspond to erased bits in the same position in the codeword, and values of 0 correspond to bits that are not erased.

\section*{Output number of corrected errors}

Selecting this check box gives the block an additional output port, Err, which indicates the number of errors the block corrected in the input codeword.

\section*{Supported Data Type}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline In & - Double-precision floating point \\
& - Single-precision floating point \\
& - Boolean \\
& • \(8-, 16\)-, and 32 -bit signed \\
& integers
\end{tabular}

\section*{BCH Decoder}
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline & - 8-, 16-, and 32 -bit unsigned integers \\
\hline Out & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8-, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers
\end{tabular} \\
\hline Era & \begin{tabular}{l}
- Double-precision floating point \\
- Boolean
\end{tabular} \\
\hline Err & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8-, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers
\end{tabular} \\
\hline
\end{tabular}

\section*{Pair Block BCH Encoder}

References [1] Wicker, Stephen B., Error Control Systems for Digital
Communication and Storage, Upper Saddle River, N.J., Prentice Hall, 1995.
[2] Berlekamp, Elwyn R., Algebraic Coding Theory, New York, McGraw-Hill, 1968.

\section*{BCH Decoder}
[3] Clark, George C., Jr., and J. Bibb Cain, Error-Correction Coding for Digital Communications, New York, Plenum Press, 1981.

See Also bchdec (in Communications System Toolbox documentation)

\section*{Purpose \\ Library \\ Description \\ }

Create BCH code from binary vector data
Block sublibrary of Error Detection and Correction
The BCH Encoder block creates a BCH code with message length \(K\) and codeword length ( \(N\) - number of punctures). You specify both \(N\) and \(K\) directly in the dialog box.
This block accepts a column vector input signal with an integer multiple of \(K\) elements. Each group of \(K\) input elements represents one message word to be encoded.

If the encoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

For a given codeword length \(N\), only specific message lengths \(K\) are valid for a BCH code. For a full length BCH code, \(N\) must be of the form \(2^{M}-1\), where \(3 \leq M \leq 16\). If \(N\) is less than \(2^{M}-1\), the block assumes that the code has been shortened by length \(2^{M-1}-N\). However, if \(N\) is greater than or equal to \(2^{M-1}\), Primitive polynomial must be specified to appropriately set the value of \(M\).
No known analytic formula describes the relationship among the codeword length, message length, and error-correction capability. The tables below list valid [ \(n, k\) ] pairs for small values of \(n\), as well as the corresponding values of the error-correction capability, t .
\begin{tabular}{ccc}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\mathbf{t}\) \\
7 & 4 & 1 \\
\hline
\end{tabular}
\begin{tabular}{ccc|}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\mathbf{t}\) \\
15 & 11 & 1 \\
15 & 7 & 2 \\
15 & 5 & 3 \\
\hline
\end{tabular}
\begin{tabular}{ccc}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\mathbf{t}\) \\
31 & 26 & 1 \\
31 & 21 & 2 \\
31 & 16 & 3 \\
31 & 11 & 5 \\
31 & 6 & 7 \\
\hline
\end{tabular}
\begin{tabular}{|lcc|}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\mathbf{t}\) \\
\hline 63 & 57 & 1 \\
63 & 51 & 2 \\
63 & 45 & 3 \\
63 & 39 & 4 \\
63 & 36 & 5 \\
63 & 30 & 6 \\
63 & 24 & 7 \\
63 & 18 & 10 \\
63 & 16 & 11 \\
63 & 10 & 13 \\
\hline
\end{tabular}
\begin{tabular}{|ccc|}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\mathbf{t}\) \\
127 & 120 & 1 \\
127 & 113 & 2 \\
127 & 106 & 3 \\
127 & 99 & 4 \\
127 & 92 & 5 \\
127 & 85 & 6 \\
127 & 78 & 7 \\
127 & 71 & 9 \\
127 & 64 & 10 \\
127 & 57 & 11 \\
127 & 50 & 13 \\
127 & 43 & 14 \\
127 & 36 & 15 \\
127 & 29 & 21 \\
127 & 22 & 23 \\
127 & 15 & 27 \\
127 & 8 & 31 \\
\hline
\end{tabular}
\begin{tabular}{|ccc|}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\boldsymbol{t}\) \\
\hline 255 & 247 & 1 \\
255 & 239 & 2 \\
255 & 231 & 3 \\
255 & 223 & 4 \\
\hline
\end{tabular}

\section*{BCH Encoder}
\begin{tabular}{|c|c|c|}
\hline n & k & † \\
\hline 255 & 215 & 5 \\
\hline 255 & 207 & 6 \\
\hline 255 & 199 & 7 \\
\hline 255 & 191 & 8 \\
\hline 255 & 187 & 9 \\
\hline 255 & 179 & 10 \\
\hline 255 & 171 & 11 \\
\hline 255 & 163 & 12 \\
\hline 255 & 155 & 13 \\
\hline 255 & 147 & 14 \\
\hline 255 & 139 & 15 \\
\hline 255 & 131 & 18 \\
\hline 255 & 123 & 19 \\
\hline 255 & 115 & 21 \\
\hline 255 & 107 & 22 \\
\hline 255 & 99 & 23 \\
\hline 255 & 91 & 25 \\
\hline 255 & 87 & 26 \\
\hline 255 & 79 & 27 \\
\hline 255 & 71 & 29 \\
\hline 255 & 63 & 30 \\
\hline 255 & 55 & 31 \\
\hline 255 & 47 & 42 \\
\hline 255 & 45 & 43 \\
\hline 255 & 37 & 45 \\
\hline
\end{tabular}
\begin{tabular}{ccc|}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\mathbf{t}\) \\
255 & 29 & 47 \\
255 & 21 & 55 \\
255 & 13 & 59 \\
255 & 9 & 63 \\
\hline
\end{tabular}
\begin{tabular}{|ccc|}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\mathbf{t}\) \\
\hline 511 & 502 & 1 \\
511 & 493 & 2 \\
511 & 484 & 3 \\
511 & 475 & 4 \\
511 & 466 & 5 \\
511 & 457 & 6 \\
511 & 448 & 7 \\
511 & 439 & 8 \\
511 & 430 & 9 \\
511 & 421 & 10 \\
511 & 412 & 12 \\
511 & 403 & 13 \\
511 & 394 & 14 \\
511 & 385 & 15 \\
511 & 376 & 16 \\
\hline 511 & 357 & 18 \\
\hline
\end{tabular}
\begin{tabular}{|ccc|}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\mathbf{t}\) \\
\hline 511 & 349 & 19 \\
511 & 340 & 20 \\
511 & 331 & 21 \\
511 & 322 & 22 \\
511 & 313 & 23 \\
511 & 304 & 25 \\
511 & 295 & 26 \\
511 & 286 & 27 \\
511 & 277 & 28 \\
511 & 268 & 29 \\
511 & 259 & 30 \\
511 & 250 & 31 \\
511 & 241 & 36 \\
511 & 238 & 37 \\
511 & 229 & 38 \\
511 & 220 & 39 \\
511 & 211 & 41 \\
511 & 202 & 42 \\
\hline 511 & 193 & 43 \\
511 & 184 & 45 \\
511 & 175 & 46 \\
511 & 166 & 47 \\
511 & 157 & 51 \\
\hline 511 & 148 & 53 \\
\hline 511 & 139 & 54 \\
\hline & & \\
\hline & & \\
\hline
\end{tabular}
\begin{tabular}{|ccc|}
\hline \(\mathbf{n}\) & \(\mathbf{k}\) & \(\mathbf{t}\) \\
\hline 511 & 130 & 55 \\
511 & 121 & 58 \\
511 & 112 & 59 \\
511 & 103 & 61 \\
511 & 94 & 62 \\
511 & 85 & 63 \\
511 & 76 & 85 \\
511 & 67 & 87 \\
511 & 58 & 91 \\
511 & 49 & 93 \\
511 & 40 & 95 \\
511 & 31 & 109 \\
511 & 28 & 111 \\
511 & 19 & 119 \\
511 & 10 & 121 \\
\hline
\end{tabular}

The primitive and generator polynomials may be specified in their respective fields, which appear after selecting their corresponding check boxes.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-57 table on this page.

\section*{Puncture Codes}

This block supports puncturing when you select the Puncture code parameter. This selection enables the Puncture vector parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 represents that the data symbol passes unaltered, and 0 represents that the data symbol gets punctured, or removed, from the data stream. This convention is carried for both the encoder

\section*{BCH Encoder}
and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

Note 1s and 0s have precisely opposite meanings for the puncture and erasure vectors. For an erasure vector, 1 means that the data symbol is to be replaced with an erasure symbol, and 0 means that the data symbol is passed unaltered. This convention is carried for both the encoder and the decoder.

Dialog Box


\section*{Codeword length, \(\mathbf{N}\)}

The codeword length.

\section*{Message length, \(K\)}

The message length.

\section*{Specify primitive polynomial}

Selecting this check box enables the Primitive polynomial field.

\section*{Primitive polynomial}

A row vector that represents the binary coefficients of the primitive polynomial in order of descending powers.

\section*{BCH Encoder}

> This field defaults to de2bi(primpoly(4, 'nodisplay'), 'left-msb'), corresponding to a \((15,5)\) code.

This parameter applies only when you select Specify primitive polynomial.

\section*{Specify generator polynomial}

Selecting this check box enables the Generator polynomial field.

\section*{Generator polynomial}

A row vector that represents the binary coefficients of the generator polynomial in order of descending powers.

The length of the Generator polynomial must be \(N-K+1\).

This field defaults to bchgenpoly \((15,5)\).

This parameter applies only when you select Specify generator polynomial.

\section*{Disable generator polynomial checking}

This check box appears only when you select Specify generator polynomial.

Each time a model initializes, the block performs a polynomial check. This check verifies that \(X^{N}+1\) is divisible by the user-defined generator polynomial, where \(N\) represents the full code word length. Selecting this check box disables the polynomial check. For larger codes, disabling the check speeds up the simulation process. You should always run the check at least once before disabling this feature.

\section*{Puncture code}

Selecting this check box enables the Puncture vector field.

\section*{Puncture vector}

A column vector of length \(N-K\). In the Puncture vector, a value of 1 represents that the data symbol passes unaltered, and 0
represents that the data symbol gets punctured, or removed, from the data stream.

The field defaults to \([\) ones \((8,1) ; \operatorname{zeros}(2,1)]\).
This parameter applies only when you select Puncture code.

\section*{Supported Data Type}
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline In & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8-, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers
\end{tabular} \\
\hline Out & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16 -, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers
\end{tabular} \\
\hline
\end{tabular}

\section*{Pair Block}

BCH Decoder
References
[1] Clark, George C., Jr., and J. Bibb Cain, Error-Correction Coding for Digital Communications, New York, Plenum Press, 1981.

See Also bchenc (in Communications System Toolbox documentation)

\section*{Bernoulli Binary Generator}
\begin{tabular}{ll} 
Purpose & Generate Bernoulli-distributed random binary numbers \\
Library & Random Data Sources sublibrary of Comm Sources \\
Description & \begin{tabular}{l} 
The Bernoulli Binary Generator block generates random binary \\
numbers using a Bernoulli distribution. The Bernoulli distribution with \\
parameter p produces zero with probability p and one with probability \\
1-p. The Bernoulli distribution has mean value 1-p and variance \(p(1-p)\). \\
The Probability of a zero parameter specifies p, and can be any real \\
number between zero and one.
\end{tabular}
\end{tabular}

\section*{Attributes of Output Signal}

The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the Frame-based outputs, Samples per frame, and Interpret vector parameters as 1-D parameters. See "Sources and Sinks" in Communications System Toolbox User's Guide for more details.

The number of elements in the Initial seed and Probability of a zero parameters becomes the number of columns in a frame-based output or the number of elements in a sample-based vector output. Also, the shape (row or column) of the Initial seed and Probability of a zero parameters becomes the shape of a sample-based two-dimensional output signal.

\section*{Bernoulli Binary Generator}


Dialog
Box

\section*{Probability of a zero}
The probability with which a zero output occurs.

\section*{Initial seed}
The initial seed value for the random number generator. The seed can be either a vector of the same length as the Probability of a zero parameter, or a scalar.

\section*{Sample time}
The period of each sample-based vector or each row of a frame-based matrix.

\section*{Frame-based outputs}
Determines whether the output is frame-based or sample-based. This box is active only if Interpret vector parameters as 1-D is unchecked.

\section*{Bernoulli Binary Generator}

\section*{Samples per frame}

The number of samples in each column of a frame-based output signal. This field is active only if Frame-based outputs is checked.

\section*{Interpret vector parameters as 1-D}

If this box is checked, then the output is a one-dimensional signal. Otherwise, the output is a two-dimensional signal. This box is active only if Frame-based outputs is unchecked.

\section*{Output data type}

The output type of the block can be specified as a boolean, int8, uint8, int16, uint16, int32, uint32, single, or double. By default, the block sets this to double. Single outputs may lead to different results when compared with double outputs for the same set of parameters.

See Also
Random Integer Generator, Binary Symmetric Channel; randint (in Communications System Toolboxdocumentation), rand (built-in MATLAB function)

\section*{Binary Cyclic Decoder}

\section*{Purpose}

Decode systematic cyclic code to recover binary vector data

\section*{Library}

Description
Block sublibrary of Error Detection and Correction
The Binary Cyclic Decoder block recovers a message vector from a codeword vector of a binary systematic cyclic code. For proper decoding, the parameter values in this block should match those in the corresponding Binary Cyclic Encoder block.

If the cyclic code has message length \(K\) and codeword length \(N\), then \(N\) must have the form \(2^{\mathrm{M}}-1\) for some integer \(M\) greater than or equal to 3 .

This block accepts a column vector input signal containing \(N\) elements. The output signal is a column vector containing \(K\) elements.

You can determine the systematic cyclic coding scheme in one of two ways:
- To create an \([\mathrm{N}, \mathrm{K}]\) code, enter \(N\) and \(K\) as the first and second dialog parameters, respectively. The block computes an appropriate generator polynomial, namely, cyclpoly ( \(\mathrm{N}, \mathrm{K}, \mathrm{I}^{\prime} \mathrm{min}\) ').
- To create a code with codeword length N and a particular degree-(N-K) binary generator polynomial, enter \(N\) as the first parameter and a binary vector as the second parameter. The vector represents the generator polynomial by listing its coefficients in order of ascending exponents. You can create cyclic generator polynomials using the Communications System Toolbox cyclpoly function.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-63 table on this page.

\section*{Binary Cyclic Decoder}

\section*{Dialog \\ Box}

\section*{Codeword length N}

The codeword length \(N\), which is also the input vector length.

\section*{Message length K, or generator polynomial}

Either the message length, which is also the output vector length; or a binary vector that represents the generator polynomial for the code.

\section*{Binary Cyclic Decoder}

Supported
Data Type
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline In & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers \\
- Fixed-point
\end{tabular} \\
\hline Out & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16-, and 32 -bit unsigned integers \\
- Fixed-point
\end{tabular} \\
\hline
\end{tabular}

Pair Block
Binary Cyclic Encoder
See Also cyclpoly (Communications Toolbox)

\section*{Binary Cyclic Encoder}
\begin{tabular}{|c|c|}
\hline Purpose & Create systematic cyclic code from binary vector data \\
\hline Library & Block sublibrary of Error Detection and Correction \\
\hline \multirow[t]{6}{*}{Description} & The Binary Cyclic Encoder block creates a systematic cyclic code with message length \(K\) and codeword length \(N\). The number \(N\) must have the form \(2^{\mathrm{M}}-1\), where \(M\) is an integer greater than or equal to 3 . \\
\hline & This block accepts a column vector input signal containing \(K\) elements. The out put signal is a column vector containing \(N\) elements. \\
\hline & You can determine the systematic cyclic coding scheme in one of two ways: \\
\hline & - To create an \([\mathrm{N}, \mathrm{K}]\) code, enter N and K as the first and second dialog parameters, respectively. The block computes an appropriate generator polynomial, namely, cyclpoly ( \(\mathrm{N}, \mathrm{K}\), 'min'). \\
\hline & - To create a code with codeword length \(N\) and a particular degree-(N-K) binary generator polynomial, enter \(N\) as the first parameter and a binary vector as the second parameter. The vector represents the generator polynomial by listing its coefficients in order of ascending exponents. You can create cyclic generator polynomials using the Communications System Toolbox cyclpoly function. \\
\hline & For information about the data types each block port supports, see the "Supported Data Type" on page 2-66 table on this page. \\
\hline
\end{tabular}

\section*{Binary Cyclic Encoder}

\section*{Dialog \\ Box}
\begin{tabular}{|c|c|c|c|c|}
\hline \multicolumn{5}{|l|}{Finnction Block Parameters: Binary Cyclic Encoder X} \\
\hline \multicolumn{5}{|l|}{\multirow[t]{2}{*}{\begin{tabular}{l}
Binary Cyclic Encoder (mask) link) \\
Create a systematic cyclic code with message length K and codeword length N . The number \(N\) must have the form \(2^{\wedge} M-1\), where \(M\) is an integer greater than or equal to 3 . \\
The input must contain exactly \(K\) elements. If it is frame-based, then it must be a column vector.
\end{tabular}}} \\
\hline & & & & \\
\hline \multicolumn{4}{|l|}{\multirow[t]{4}{*}{Parameters
Codeword length N :
7
Message length K , or generator polynomia
4}} & \\
\hline & & & & \\
\hline & & & & \\
\hline & & & & \\
\hline \multicolumn{5}{|c|}{OK Cancel Help Apply} \\
\hline
\end{tabular}

\section*{Codeword length N}

The codeword length, which is also the output vector length.

\section*{Message length \(K\), or generator polynomial}

Either the message length, which is also the input vector length; or a binary vector that represents the generator polynomial for the code.

\section*{Binary Cyclic Encoder}

\section*{Supported Data Type}
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline In & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers \\
- Fixed-point
\end{tabular} \\
\hline Out & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers \\
- Fixed-point
\end{tabular} \\
\hline
\end{tabular}

\footnotetext{
Pair Block
Binary Cyclic Decoder
See Also
cyclpoly (in the Communications System Toolbox documentation)
}

\section*{Binary-Input RS Encoder}

\section*{Purpose}

Create Reed-Solomon code from binary vector data

\section*{Library}

Description


Block sublibrary of Error Detection and Correction
The Binary-Input RS Encoder block creates a Reed-Solomon code with message length, K , and codeword length, ( N - number of punctures). You specify both N and K directly in the dialog box. The symbols for the code are binary sequences of length M, corresponding to elements of the Galois field \(\operatorname{GF}\left(2^{\mathrm{M}}\right)\), where the first bit in each sequence is the most significant bit. Restrictions on M and N are given in "Restrictions on the M and the Codeword Length N" on page 2-68 below. The difference \(\mathrm{N}-\mathrm{K}\) must be an even integer.

This block can output shortened codewords when N and K are appropriately specified. To specify output codewords that are shortened by a length \(\mathrm{S}, \mathrm{N}\) and K must be specified in the dialog box as \(\mathrm{N}_{\text {full }}-\mathrm{S}\) and \(\mathrm{K}_{\text {full }}-\mathrm{S}\), where \(\mathrm{N}_{\text {full }}\) and \(\mathrm{K}_{\text {full }}\) are the N and K of an unshortened code. If \(\mathrm{S}<\left(\mathrm{N}_{\text {full }}+1\right) / 2\), the encoder can automatically determine the value of \(\mathrm{N}_{\text {full }}\) and \(\mathrm{K}_{\text {full }}\). However, if \(\mathrm{S} \geq\left(\mathrm{N}_{\text {full }}+1\right) / 2\), Primitive polynomial must be specified in order to properly define the extension field for the code.

The input and output are binary-valued signals that represent messages and codewords, respectively. This block accepts a column vector input signal with a length that is an integer multiple of \(\mathrm{M} * \mathrm{~K}\). This block outputs a column vector with a length that is the same integer multiple of \(\mathrm{M}^{*}(\mathrm{~N}\) - number of punctures). The block inherits the output data type from the input. For information about the data types each block port supports, see the "Supported Data Type" on page 2-73 table on this page.
For more information on representing data for Reed-Solomon codes, see the section "Integer Format (Reed-Solomon Only)" in Communications System Toolbox User's Guide.
If the encoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

\section*{Binary-Input RS Encoder}

The default value of \(M\) is the smallest integer that is greater than or equal to \(\log 2(N+1)\), that is, ceil \((\log 2(N+1))\). You can change the value of M from the default by specifying the primitive polynomial for GF \(\left(2^{\mathrm{M}}\right)\), as described in "Specifying the Primitive Polynomial" on page \(2-68\) below. If N is less than \(2^{\mathrm{M}}-1\), the block assumes that the code has been shortened by length \(2^{\mathrm{M}-1}-\mathrm{N}\).
Each \(\mathrm{M} * \mathrm{~K}\) input bits represent K integers between 0 and \(2^{\mathrm{M}}-1\). Similarly, each M*(N - number of punctures) output bits represent N integers between 0 and \(2^{\mathrm{M}}-1\). These integers in turn represent elements of the Galois field \(\operatorname{GF}\left(2^{\mathrm{M}}\right)\).
An (N,K) Reed-Solomon code can correct up to floor ( (N-K)/2) symbol errors (not bit errors) in each codeword.

\section*{Specifying the Primitive Polynomial}

You can specify the primitive polynomial that defines the finite field \(\mathrm{GF}\left(2^{\mathrm{M}}\right)\), corresponding to the integers that form messages and codewords. To do so, first select Specify primitive polynomial. Then, set Primitive polynomial to a binary row vector that represents a primitive polynomial over \(\mathrm{GF}(2)\) of degree M , in descending order of powers. For example, to specify the polynomial \(x^{3}+x+1\), enter the vector \(\left[\begin{array}{llll}1 & 0 & 1 & 1\end{array}\right]\).
If you do not select Specify primitive polynomial, the block uses the default primitive polynomial of degree \(\mathrm{M}=\) ceil \((\log 2(N+1))\). You can display the default polynomial by entering primpoly (ceil \((\log 2(N+1)))\) at the MATLAB prompt.

\section*{Restrictions on the \(\mathbf{M}\) and the Codeword Length \(\mathbf{N}\)}

The restrictions on the degree \(M\) of the primitive polynomial and the codeword length N are as follows:
- If you do not select Specify primitive polynomial, N must lie in the range \(3<\mathrm{N} \leq 2^{16}-1\).
- If you do select Specify primitive polynomial, N must lie in the range \(3 \leq \mathrm{N} \leq 2^{16}-1\) and M must lie in the range \(3 \leq \mathrm{M} \leq 16\).

\section*{Binary-Input RS Encoder}

\section*{Specifying the Generator Polynomial}

You can specify the generator polynomial for the Reed-Solomon code. To do so, first select Specify generator polynomial. Then, in the Generator polynomial field, enter an integer row vector whose entries are between 0 and \(2^{\mathrm{M}}-1\). The vector represents a polynomial, in descending order of powers, whose coefficients are elements of \(\mathrm{GF}\left(2^{\mathrm{M}}\right)\) represented in integer format. See the section "Integer Format (Reed-Solomon Only)" for more information about integer format. The generator polynomial must be equal to a polynomial with a factored form
\[
g(x)=\left(x+A^{b}\right)\left(x+A^{b+1}\right)\left(x+A^{b+2}\right) \ldots\left(x+A^{b+N-K-1}\right)
\]
where A is the primitive element of the Galois field over which the input message is defined, and \(b\) is a non-negative integer.

If you do not select Specify generator polynomial, the block uses the default generator polynomial, corresponding to \(b=1\), for Reed-Solomon encoding. You can display the default generator polynomial by entering rsgenpoly ( \(\mathrm{N} 1, \mathrm{~K} 1\) ), where \(\mathrm{N} 1=2^{\wedge} \mathrm{M}-1\) and \(\mathrm{K} 1=\mathrm{K}+(\mathrm{N} 1-\mathrm{N})\), at the MATLAB prompt, if you are using the default primitive polynomial. If the Specify primitive polynomial box is selected, and you specify the primitive polynomial specified as poly, the default generator polynomial is rsgenpoly ( \(\mathrm{N} 1, \mathrm{~K} 1\), poly).

\section*{Puncture Codes}

The block supports puncturing when you select the Puncture code parameter. This enables the Puncture vector parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 represents that the data symbol passes unaltered, and 0 represents that the data symbol gets punctured, or removed, from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

\section*{Examples}

Suppose \(\mathrm{M}=3, \mathrm{~N}=2^{3}-1=7\), and \(\mathrm{K}=5\). Then a message is a binary vector of length 15 that represents 5 three-bit integers. A corresponding codeword is a binary vector of length 21 that represents 7 three-bit

\section*{Binary-Input RS Encoder}
integers. The following figure shows the codeword that would result from a particular message word. The integer format equivalents illustrate that the highest order bit is at the left.


\section*{Binary-Input RS Encoder}
Encode the message in the input vector using an ( \(\mathrm{N}, \mathrm{K}\) ) Reed-Solomon encoder with the narrow-sense generator polynomial. This block accepts a column vector input signal with an integer multiple of \(K^{*}\) ceil \((\log 2(\mathrm{~N}+1))\) bits. Each group of \(K^{*}\) ceil \((\log 2(N+1))\) input bits represents one message word to be encoded.
If \(\log 2(N+1)\) does not equal \(M\), where \(3<=M<=16\), then a shortened code is assumed. If the Primitive polynomial is not specified, then the length by which the codeword is shortened is \(2^{\wedge}\) ceil \((\log 2(\mathrm{~N}+1))\) - \((\mathrm{N}+1)\). If it is specified, then the shortening length is \(2^{\wedge}\) (length(Primitive polynomial) -1\()-(N+1)\).
Parameters
Codeword length N :

\section*{7}
Message length K :

\section*{3}
\(\Gamma\) Specify primitive polynomial
\(\Gamma\) Specify generator polynomial
\(\Gamma\) Puncture code

    Binary-Input RS Encoder (mask) (ink)
    Binary-Input RS Encoder (mask) (ink)

Dialog Box

\section*{Codeword length N}

The codeword length. The output has vector length \(\mathrm{NC} * \mathrm{M}^{*}(\mathrm{~N}-\) NP), where NC is the number of codewords being output, and NP is the number of punctures per codeword.

\section*{Message length \(K\)}

The message length. The input has vector length \(N M^{*} M^{*} K\), where NM is the number of messages per frame being input.

\section*{Specify primitive polynomial}

Selecting this check box enables the field Primitive polynomial.

\section*{Binary-Input RS Encoder}

\section*{Primitive polynomial}

This field is available only when Specify primitive polynomial is selected.

Binary row vector representing the primitive polynomial in descending order of powers.

\section*{Specify generator polynomial}

Selecting this check box enables the field Generator polynomial.

\section*{Generator polynomial}

This field is available only when Specify generator polynomial is selected.

Integer row vector, whose entries are in the range from 0 to \(2^{\mathrm{M}}-1\), representing the generator polynomial in descending order of powers.

\section*{Puncture code}

Selecting this check box enables the field Puncture vector.

\section*{Puncture vector}

This field is available only when Puncture code is selected.
A column vector of length N-K. A value of 1 in the Puncture vector corresponds to an M-bit symbol that is not punctured, and a 0 corresponds to an M-bit symbol that is punctured.

The default value is [ones(2,1); zeros(2,1)].

\section*{Output data type}

The output type of the block can be specified as Same as input, boolean, or double. By default, the block sets this to Same as input.

\section*{Supported Data Type}
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline In & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers \\
- 1-bit unsigned integer (ufix(1))
\end{tabular} \\
\hline Out & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16-, and 32 -bit unsigned integers \\
- 1-bit unsigned integer (ufix(1))
\end{tabular} \\
\hline
\end{tabular}

\author{
Pair Block Binary-Output RS Decoder \\ See Also Integer-Input RS Encoder
}

\section*{Binary Linear Decoder}
\[
\begin{array}{ll}
\text { Purpose } & \begin{array}{l}
\text { Decode linear block code to recover binary vector data } \\
\text { Library } \\
\text { Bescription }
\end{array} \begin{array}{l}
\text { The Binary Linear Decoder block recovers a binary message vector from } \\
\text { a binary codeword vector of a linear block code. }
\end{array} \\
\text { The Generator matrix parameter is the generator matrix for the block } \\
\text { code. For proper decoding, this should match the Generator matrix } \\
\text { parameter in the corresponding Binary Linear Encoder block. If } N \text { is } \\
\text { the codeword length of the code, then Generator matrix must have } N \\
\text { columns. If } K \text { is the message length of the code, then the Generator } \\
\text { matrix parameter must have } K \text { rows. }
\end{array}
\]

\section*{Binary Linear Decoder}

Dialog
Box


\section*{Generator matrix}

Generator matrix for the code; same as in Binary Linear Encoder block.

\section*{Decoding table}

Either a \(2^{\text {N-K }}\)-by-N matrix that lists correction vectors for each codeword's syndrome; or the scalar 0 , in which case the block defaults to the table corresponding to the Generator matrix parameter.

\section*{Binary Linear Decoder}

\section*{Supported Data Type}
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline In & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers \\
- Fixed-point
\end{tabular} \\
\hline Out & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers \\
- Fixed-point
\end{tabular} \\
\hline
\end{tabular}

\author{
Pair Block \\ Binary Linear Encoder
}

\section*{Binary Linear Encoder}

\section*{Purpose}

Create linear block code from binary vector data

\section*{Library \\ Description}


Dialog Box

Block sublibrary of Error Detection and Correction
The Binary Linear Encoder block creates a binary linear block code using a generator matrix that you specify. If \(K\) is the message length of the code, then the Generator matrix parameter must have \(K\) rows. If \(N\) is the codeword length of the code, then Generator matrix must have \(N\) columns.

This block accepts a column vector input signal containing \(K\) elements. This block outputs a column vector with a length of \(N\) elements. For information about the data types each block port supports, see "Supported Data Type" on page 2-78.


\section*{Generator matrix}

A \(K\)-by- \(N\) matrix, where \(K\) is the message length and \(N\) is the codeword length.

\section*{Binary Linear Encoder}

\section*{Supported Data Type}
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline In & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16-, and 32 -bit unsigned integers \\
- Fixed-point
\end{tabular} \\
\hline Out & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8-, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers \\
- Fixed-point
\end{tabular} \\
\hline
\end{tabular}

\author{
Pair Block \\ Binary Linear Decoder
}

\section*{Binary-Output RS Decoder}

\section*{Purpose}

Decode Reed-Solomon code to recover binary vector data

\section*{Library}

Description
Block sublibrary of Error Detection and Correction
The Binary-Output RS Decoder block recovers a binary message vector from a binary Reed-Solomon codeword vector. For proper decoding, the parameter values in this block should match those in the corresponding Binary-Input RS Encoder block.

The Reed-Solomon code has message length, \(K\), and codeword length, ( \(N\) - number of punctures). You specify both \(N\) and \(K\) directly in the dialog box. The symbols for the code are binary sequences of length \(M\), corresponding to elements of the Galois field \(\operatorname{GF}\left(2^{\mathrm{M}}\right)\), where the first bit in each sequence is the most significant bit. Restrictions on \(M\) and \(N\) are described in "Restrictions on the M and the Codeword Length N" on page \(2-68\). The difference \(N-K\) must be an even integer.

This block can output shortened codewords when \(N\) and \(K\) are appropriately specified. To specify output codewords that are shortened by a length \(S, N\) and \(K\) must be specified in the dialog box as \(N_{\text {full }}-S\) and \(K_{\text {full }}-S\), where \(N_{\text {full }}\) and \(K_{\text {full }}\) are the \(N\) and \(K\) of an unshortened code. If \(\mathrm{S}<\left(\mathrm{N}_{\text {full }}+1\right) / 2\), the encoder can automatically determine the value of \(N_{\text {full }}\) and \(K_{\text {full }}\). However, if \(\mathrm{S} \geq\left(\mathrm{N}_{\text {full }}+1\right) / 2\), Primitive polynomial must be specified in order to properly define the extension field for the code.

The input and output are binary-valued signals that represent codewords and messages, respectively. This block accepts a column vector input signal with a length that is an integer multiple of \(M^{*}(N-\) number of punctures). This block outputs a column vector with a length that is the same integer multiple of \(M^{*} K\). The output signal inherits its data type from the input signal. For information about the data types each block port supports, see the "Supported Data Type" on page 2-84 table on this page.

For more information on representing data for Reed-Solomon codes, see "Integer Format (Reed-Solomon Only)" in Communications System Toolbox User's Guide.

\section*{Binary-Output RS Decoder}

If the decoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.
The default value of \(M\) is ceil \((\log 2(N+1))\), that is, the smallest integer greater than or equal to \(\log 2(N+1)\). You can change the value of \(M\) from the default by specifying the primitive polynomial for \(\operatorname{GF}\left(2^{\mathrm{M}}\right)\), as described in "Specifying the Primitive Polynomial" on page \(2-68\) below. If N is less than \(2^{\mathrm{M}}-1\), the block uses a shortened Reed-Solomon code.

You can also specify the generator polynomial for the Reed-Solomon code, as described in "Specifying the Generator Polynomial" on page 2-69.

Each \(M^{*} K\) input bits represent \(K\) integers between 0 and \(2^{\mathrm{M}}-1\). Similarly, each \(M^{*}(N\) - number of punctures) output bits represent \(N\) integers between 0 and \(2^{\mathrm{M}}-1\). These integers in turn represent elements of the Galois field \(\operatorname{GF}\left(2^{\mathrm{M}}\right)\).

The second output is a vector of the number of errors detected during decoding of the codeword. A-1 indicates that the block detected more errors than it could correct using the coding scheme. An ( \(N, K\) ) Reed-Solomon code can correct up to floor ( (N-K)/2) symbol errors (not bit errors) in each codeword.

You can disable the second output by deselecting Output port for number of corrected errors. This removes the block's second output port.

\section*{Punctured Codes}

This block supports puncturing when you select the Punctured code parameter. This selection enables the Puncture vector parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 represents that the data symbol passes unaltered, and 0 represents that the data symbol gets punctured, or removed, from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

\section*{Binary-Output RS Decoder}

> Note 1 s and 0 s have precisely opposite meanings for the puncture and erasure vectors. For an erasure vector, 1 means that the data symbol is to be replaced with an erasure symbol, and 0 means that the data symbol is passed unaltered. This convention is carried for both the encoder and the decoder.

\section*{Binary-Output RS Decoder}


Dialog Box

\section*{Codeword length \(\mathbf{N}\)}

The codeword length. The input has vector length \(N C^{*} M^{*}(N\) \(N P\) ), where \(N C\) is the number of codewords being output, and \(N P\) is the number of punctures per codeword.

\section*{Message length \(K\)}

The message length. The first output has vector length \(N M^{*} M^{*} K\), where \(N M\) is the number of messages per frame being output.

\section*{Binary-Output RS Decoder}

\section*{Specify primitive polynomial}

Selecting this check box enables the Primitive polynomial field.

\section*{Primitive polynomial}

Binary row vector representing the primitive polynomial in descending order of powers. When you provide a Primitive polynomial, the number of input bits must be an integer multiple of \(K\) times the order of the Primitive polynomial instead.

This parameter applies only when you select Specify primitive polynomial.

\section*{Specify generator polynomial}

Selecting this check box enables the Generator polynomial field.

\section*{Generator polynomial}

Integer row vector, whose entries are in the range from 0 to \(2^{\mathrm{M}}-1\), representing the generator polynomial in descending order of powers. Each coefficient is an element of the Galois field defined by the primitive polynomial.

This parameter applies only when you select Specify generator polynomial.

\section*{Puncture code}

Selecting this check box enables the Puncture vector field.

\section*{Puncture vector}

A column vector of length \(N-K\). For a puncture vector, 1 represents an \(M\)-bit symbol that passes unaltered, and 0 represents an \(M\)-bit symbol that gets punctured, or removed, from the data stream.

The default value is \([\) ones \((2,1) ; \operatorname{zeros}(2,1)]\).
This parameter applies only when you select Punctured code.

\section*{Enable erasures input port}

Select this check to open the erasures port, Era.

\section*{Binary-Output RS Decoder}

Through the port, you can input a binary column vector that is \(1 / \mathrm{M}\) times as long as the codeword input.

Erasure values of 1 correspond to erased symbols in the same position in the bit-packed codeword, and values of 0 correspond to nonerased symbols.

\section*{Output number of corrected errors}

When you select this box, the block outputs the number of corrected errors in each word through a second output port. A decoding failure occurs when a certain received word in the input contains more than ( \(N-K\) )/2 symbol errors. The value -1 indicates the corresponding position in the second output vector.

\section*{Output data type}

The output type of the block can be specified as Same as input, boolean, or double. By default, the block sets this to Same as input.

\section*{Supported Data Type}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline In & - Double-precision floating point \\
& - Single-precision floating point \\
& - Boolean \\
& - \(8-, 16\)-, and 32 -bit signed \\
& integers \\
& - \(8-, 16\)-, and 32-bit unsigned \\
& integers \\
& - 1-bit unsigned integer (ufix(1)) \\
\hline Out & - Double-precision floating point \\
& - Single-precision floating point \\
& - Boolean \\
&
\end{tabular}

\section*{Binary-Output RS Decoder}
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline & \begin{tabular}{l}
- 8 -, 16 -, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers \\
- 1-bit unsigned integer (ufix(1))
\end{tabular} \\
\hline Era & \begin{tabular}{l}
- Double-precision floating point \\
- Boolean
\end{tabular} \\
\hline Err & - Double-precision floating point \\
\hline
\end{tabular}

\footnotetext{
\section*{Algorithm}

Pair Block
References

This block uses the Berlekamp-Massey decoding algorithm. For information about this algorithm, see the references listed below.

Binary-Input RS Encoder
[1] Wicker, Stephen B., Error Control Systems for Digital Communication and Storage, Upper Saddle River, N.J., Prentice Hall, 1995.
[2] Berlekamp, Elwyn R., Algebraic Coding Theory, New York, McGraw-Hill, 1968.
[3] Clark, George C., Jr., and J. Bibb Cain, Error-Correction Coding for Digital Communications, New York, Plenum Press, 1981.
}

\section*{See Also}

Integer-Output RS Decoder

\section*{Binary Symmetric Channel}

\section*{Purpose Introduce binary errors}

Library
Channels
Description The Binary Symmetric Channel block introduces binary errors to the signal transmitted through this channel.


The input port represents the transmitted binary signal. This block accepts a scalar or vector input signal. The block processes each vector element independently, and introduces an error in a given spot with probability Error probability.

This block uses the DSP System Toolbox Random Source block to generate the noise. The block generates random numbers using the Ziggurat method, which is the same method used by the MATLAB randn function. The Initial seed parameter in this block initializes the noise generator. Initial seed can be either a scalar or a vector, with a length that matches the number of channels in the input signal. For details on Initial seed, see the Random Source block reference page in the DSP System Toolbox documentation set.
The first output port is the binary signal the channel processes. The second output port is the vector of errors the block introduces. To suppress the second output port, clear Output error vector.

\section*{Binary Symmetric Channel}

\section*{Dialog Box}
```

O. Function Block Parameters: Binary Symmetric Channel83
Binary Symmetric Channel (mask) (link)
Add binary errors to the input signal.
Parameters
Error probability:
0.05
Initial seed:
7 1
Output error vector
Output data type: double
OK Cancel Help Apply

```

\section*{Error probability}

The probability that a binary error occurs. Set the value of this parameter between 0 and 1.

\section*{Initial seed}

The initial seed value for the random number generator.

\section*{Output error vector}

When you select this box the block outputs the vector of errors.

\section*{Output data type}

Select the output data type as double or boolean.

\section*{Binary Symmetric Channel}

Supported
Data
Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & • Double-precision floating point \\
& - Single-precision floating point \\
& - Fixed point (signed and unsigned) \\
& - Boolean \\
& - 8-, 16-, and 32-bit signed integers \\
& - 8-, 16-, and 32-bit unsigned integers \\
\hline Output & • Double-precision floating point \\
& - Boolean \\
\hline
\end{tabular}

See Also Bernoulli Binary Generator

\section*{Bipolar to Unipolar Converter}

\section*{Purpose}

Map bipolar signal into unipolar signal in range [0, M-1]

\section*{Library}

Description

Bipolar to Unipolar Converter

Utility Blocks
The Bipolar to Unipolar Converter block maps the bipolar input signal to a unipolar output signal. If the input consists of integers in the set \(\{-M+1,-M+3,-M+5, \ldots, M-1\}\), where \(M\) is the \(\mathbf{M}\)-ary number parameter, then the output consists of integers between 0 and \(\mathrm{M}-1\). This block is
only designed to work when the input value is within the set \(\{-M+1\), \(-M+3,-M+5, \ldots, M-1\}\), where M is the \(\mathbf{M}\)-ary number parameter. If the input value is outside of this set of integers the output may not be valid.

The table below shows how the block's mapping depends on the Polarity parameter.
\begin{tabular}{l|l}
\hline Polarity Parameter Value & \begin{tabular}{l} 
Output Corresponding to \\
Input Value of \(\mathbf{k}\)
\end{tabular} \\
\hline Positive & \((\mathrm{M}-1+\mathrm{k}) / 2\) \\
\hline Negative & \((\mathrm{M}-1-\mathrm{k}) / 2\) \\
\hline
\end{tabular}

\section*{Bipolar to Unipolar Converter}


Dialog Box

\section*{M-ary number}

The number of symbols in the bipolar or unipolar alphabet.
Polarity
A value of Positive causes the block to maintain the relative ordering of symbols in the alphabets. A value of Negative causes the block to reverse the relative ordering of symbols in the alphabets.

\section*{Output Data Type}

The type of bipolar signal produced at the block's output.
The block supports the following output data types:
- Inherit via internal rule
- Same as input
- double
- int8
- uint8
- int16

\section*{Bipolar to Unipolar Converter}
- uint16
- int32
- uint32
- boolean

When the parameter is set to its default setting, Inherit via internal rule, the block determines the output data type based on the input data type.
- If the input signal is floating-point (either single or double), the output data type is the same as the input data type.
- If the input data type is not floating-point:
- Based on the M-ary number parameter, the output data type is the ideal unsigned integer output word length required to contain the range [ \(0 \mathrm{M}-1\) ] and is computed as follows:
ideal word length \(=\operatorname{ceil}(\log 2(M))\)
- The block sets the output data type to be an unsigned integer, based on the smallest word length (in bits) that can fit best the computed ideal word length.

Note The selections in the Hardware Implementation pane pertaining to word length constraints do not affect how this block determines output data types.

\section*{Examples}

If the input is \([-3 ;-1 ; 1 ; 3]\), the \(\mathbf{M}\)-ary number parameter is 4 , and the Polarity parameter is Positive, then the output is \([0 ; 1 ; 2 ; 3]\). Changing the Polarity parameter to Negative changes the output to [3; 2; 1; 0].

If the value for the M-ary number is \(2^{8}\) the block gives an output of uint8.

\section*{Bipolar to Unipolar Converter}

If the value for the M-ary number is \(2^{8}+1\) the block gives an output of uint16.

\author{
Pair Block Unipolar to Bipolar Converter
}

\section*{Bit to Integer Converter}

\section*{Purpose}

Map vector of bits to corresponding vector of integers

\section*{Library}

Description

Bit to Integer
Converter
Utility Blocks
The Bit to Integer Converter block maps groups of bits in the input vector to integers in the output vector. \(M\) defines how many bits are mapped for each output integer.

For unsigned integers, if \(M\) is the Number of bits per integer, then the block maps each group of \(M\) bits to an integer between 0 and \(2^{\mathrm{M}}-1\). As a result, the output vector length is \(1 / M\) times the input vector length. For signed integers, if \(M\) is the Number of bits per integer , then the block maps each group of \(M\) bits to an integer between \(-2^{\mathrm{M}-1}\) and \(2^{\mathrm{M}-1}-1\).

This block accepts a column vector input signal with an integer multiple equal to the value you specify for Number of bits per integer parameter. The block accepts double, single, boolean, int8, uint8, int16, uint16, int32, uint32 and ufix1 input data types.

\section*{Bit to Integer Converter}

Dialog Box


\section*{Number of bits per integer}

The number of input bits that the block maps to each integer of the output. This parameter must be an integer between 1 and 32 .

\section*{Input bit order}

Defines whether the first bit of the input signal is the most significant bit (MSB) or the least significant bit (LSB). The default selection is MSB.

\section*{After bit packing, treat resulting integer value as}

Indicates if the integer value input ranges should be treated as signed or unsigned. The default setting is Unsigned.

> Note This parameter setting determines which Output data type selections are available.

\section*{Bit to Integer Converter}

\section*{Output data type}

If the input values are unsigned integers, you can choose from the following Output data type options:
- Inherit via internal rule
- Smallest integer
- Same as input
- double
- single
- int8
- uint8
- int16
- uint16
- int32
- uint32

If the input values are signed integers, you can choose from the following Output data type options:
- Inherit via internal rule
- Smallest integer
- double
- single
- int8
- int16
- int32

The default selection for this parameter is Inherit via internal rule.

\section*{Bit to Integer Converter}

When you set the parameter to Inherit via internal rule, the block determines the output data type based on the input data type.
- If the input signal is floating-point (either double or single), the output data type is the same as the input data type.
- If the input data type is not floating-point, the output data type is determined as if the parameter is set to Smallest integer.

When you set the parameter to Smallest integer, the software selects the output data type based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box.
- If ASIC/FPGA is selected, the output data type is the smallest ideal integer or fixed-point data type, based on the setting for the Number of bits per integer parameter.
- For all other selections, the output data type is the smallest available (signed or unsigned) integer word length that is large enough to fit the ideal minimum bit size.

\section*{Examples Refer to the example on the Integer to Bit Converter reference page: Fixed-Point Integer To Bit and Bit To Integer Conversion (Audio Scrambling and Descrambling Example)}

\section*{See Also}
bi2de, bin2dec

\author{
Pair Block Integer to Bit Converter
}

\section*{BPSK Demodulator Baseband}

\section*{Purpose Demodulate BPSK-modulated data}

\author{
Library \\ PM, in Digital Baseband sublibrary of Modulation
}

Description

The BPSK Demodulator Baseband block demodulates a signal that was modulated using the binary phase shift keying method. The input is a baseband representation of the modulated signal. This block accepts a scalar or column vector input signal. The input signal must be be a discrete-time complex signal. The block maps the points \(\exp (\mathrm{j} \theta)\) and \(-\exp (\mathrm{j} \theta)\) to 0 and 1 , respectively, where \(\theta\) is the Phase offset parameter.

For information about the data types each block port supports, see "Supported Data Types" on page 2-105.

\section*{BPSK Demodulator Baseband}


Hard-Decision BPSK Demodulator Signal Diagram for Trivial Phase
Offset (multiple of \(\frac{\pi}{2}\) )

\section*{BPSK Demodulator Baseband}


\section*{Hard-Decision BPSK Demodulator Floating-Point Signal Diagram for Nontrivial Phase Offset}

\section*{BPSK Demodulator Baseband}


\section*{Hard-Decision BPSK Demodulator Fixed-Point Signal Diagram for Nontrivial Phase Offset}

The exact LLR and approximate LLR cases (soft-decision) are described in "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the Communications System Toolbox User's Guide.

\section*{BPSK Demodulator Baseband}

\section*{Function Block Parameters: BPSK Demodulator Baseband}

BPSK Demodulator Baseband
Demodulate the input signal using the binary phase shift keying method.
For sample-based input, the input must be a scalar. For frame-based input, the input must be a column ve
The Decision type parameter allows a choice between Hard decision demodulation, Log-likelihood ratio a Approximate log-likelihood ratio. The output values for Log-likelihood ratio and Approximate log-likelihood Decision types are of the same data type as the input values.
Main \(\mid\) Data Types \(\mid\)
Phase offset(rad):

Decision type:

Dialog Box

\section*{Phase offset (rad)}

The phase of the zeroth point of the signal constellation.

\section*{Decision type}

Specifies the use of hard decision, LLR, or approximate LLR during demodulation. The output values for Log-likelihood ratio and Approximate log-likelihood ratio are of the same data type as the input values. See "Exact LLR Algorithm" and "Approximate

\section*{BPSK Demodulator Baseband}

LLR Algorithm" in the Communications System Toolbox User's Guide for algorithm details.

\section*{Noise variance source}

This field appears when Approximate log-likelihood ratio or Log-likelihood ratio is selected for Decision type.

When set to Dialog, the noise variance can be specified in the Noise variance field. When set to Port, a port appears on the block through which the noise variance can be input.

\section*{Noise variance}

This parameter appears when the Noise variance source is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:
- Inf to - Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.

\section*{BPSK Demodulator Baseband}


Data Types Pane for Hard-Decision

\section*{Output}

When Decision type is set to Hard decision, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, uint32, or boolean.

When this parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is a floating-point type (single or double). If the input data type is fixed-point, the output data type will work as if this parameter is set to 'Smallest unsigned integer'.

When this parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration

\section*{BPSK Demodulator Baseband}

Parameters dialog box of the model. If ASIC/FPGA is selected in the Hardware Implementation pane, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

\section*{Derotate factor}

This parameter only applies when the input is fixed-point and
Phase offset is not a multiple of \(\frac{\pi}{2}\).
This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input.


\section*{Data Types Pane for Soft-Decision}

\section*{BPSK Demodulator Baseband}

Supported Data Types
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline Input & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Signed fixed point (only for Hard decision mode)
\end{tabular} \\
\hline Var & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point
\end{tabular} \\
\hline Output & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16 -, and 32 -bit signed integers \\
- 8-, 16 -, and 32 -bit unsigned integers \\
- ufix(1) in ASIC/FPGA and when Decision type is Hard decision modes
\end{tabular} \\
\hline
\end{tabular}

BPSK Modulator Baseband
See Also \(\begin{aligned} & \text { M-PSK Demodulator Baseband, QPSK Demodulator Baseband, DBPSK } \\ & \text { Demodulator Baseband }\end{aligned}\)

\section*{BPSK Modulator Baseband}
\begin{tabular}{ll} 
Purpose & Modulate using binary phase shift keying method \\
Library & PM, in Digital Baseband sublibrary of Modulation \\
Description & \begin{tabular}{l} 
The BPSK Modulator Baseband block modulates using the binary \\
phase shift keying method. The output is a baseband representation of \\
the modulated signal.
\end{tabular} \\
\begin{tabular}{l} 
This block accepts a column vector input signal. The input must be a \\
discrete-time binary-valued signal. If the input bit is 0 or 1, respectively, \\
then the modulated symbol is exp(je) or -exp(je), respectively, where \(\theta\) \\
represents the Phase offset parameter.
\end{tabular} \\
& \begin{tabular}{l} 
For information about the data types each block port supports, see the \\
"Supported Data Types" on page 2-109 table on this page. \\
Constellation Visualization
\end{tabular} \\
& \begin{tabular}{l} 
The BPSK Modulator Baseband block provides the capability to \\
visualize a signal constellation from the block mask. This Constellation \\
Visualization feature allows you to visualize a signal constellation for \\
specific block parameters. For more information, see the "Constellation \\
Visaalization" section of the Communications System Toolbox User's \\
Guide.
\end{tabular}
\end{tabular}

Dialog Box


\section*{Phase offset (rad)}

The phase of the zeroth point of the signal constellation.


\section*{Output data type}

The output data type can be set to double, single, Fixed-point, User-defined, or Inherit via back propagation.

\section*{BPSK Modulator Baseband}

Setting this parameter to Fixed-point or User-defined enables fields in which you can further specify details. Setting this parameter to Inherit via back propagation, sets the output data type and scaling to match the following block.

\section*{Output word length}

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed-point for the Output data type parameter.

\section*{Set output fraction length to}

Specify the scaling of the fixed-point output by either of the following two methods:
- Choose Best precision to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose User-defined to specify the output scaling in the Output fraction length parameter.

This parameter is only visible when you select Fixed-point for the Output data type parameter or when you select User-defined and the specified output data type is a fixed-point data type.

\section*{User-defined data type}

Specify any signed built-in or signed fixed-point data type. You can specify fixed-point data types using the sfix, sint, sfrac, and fixdt functions from Fixed-Point Designer \({ }^{\text {TM }}\). This parameter is only visible when you select User-defined for the Output data type parameter.

\section*{Output fraction length}

For fixed-point output data types, specify the number of fractional bits, or bits to the right of the binary point. This parameter is only visible when you select Fixed-point or User-defined for the Output data type parameter and User-defined for the Set output fraction length to parameter.

\section*{BPSK Modulator Baseband}

Supported Data
Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
& \(\bullet\) Boolean \\
& \(\bullet 8-, 16\)-, and 32-bit signed integers \\
& \(\bullet\) 8-, 16-, and 32-bit unsigned integers \\
& \(\bullet\) ufix(1) \\
\hline Output & • Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
& \(\bullet\) Fixed point (signed only) \\
\hline
\end{tabular}

Pair Block
See Also

BPSK Demodulator Baseband

M-PSK Modulator Baseband, QPSK Modulator Baseband, DBPSK Modulator Baseband

Purpose
Library
Description

Charge Filt
Pump
PD
PLL vCO.

Implement charge pump phase-locked loop using digital phase detector

Components sublibrary of Synchronization
The Charge Pump PLL (phase-locked loop) block automatically adjusts the phase of a locally generated signal to match the phase of an input signal. It is suitable for use with digital signals.
This PLL has these three components:
- A sequential logic phase detector, also called a digital phase detector or a phase/frequency detector.
- A filter. You specify the filter transfer function using the Lowpass filter numerator and Lowpass filter denominator parameters. Each is a vector that gives the respective polynomial's coefficients in order of descending powers of \(s\).
To design a filter, use functions such as butter, cheby1, and cheby2 in Signal Processing Toolbox software. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.
[num, den] = cheby2(3,40,100,'s')
- A voltage-controlled oscillator (VCO). You specify characteristics of the VCO using the VCO input sensitivity, VCO quiescent frequency, VCO initial phase, and VCO output amplitude parameters.

This block accepts a sample-based scalar input signal. The input signal represents the received signal. The three output ports produce:
- The output of the filter
- The output of the phase detector
- The output of the VCO

A sequential logic phase detector operates on the zero crossings of the signal waveform. The equilibrium point of the phase difference between the input signal and the VCO signal equals п. The sequential logic detector can compensate for any frequency difference that might exist between a VCO and an incoming signal frequency. Hence, the sequential logic phase detector acts as a frequency detector.


Dialog Box

\section*{Lowpass filter numerator}

The numerator of the lowpass filter transfer function, represented as a vector that lists the coefficients in order of descending powers of \(s\).

\section*{Lowpass filter denominator}

The denominator of the lowpass filter transfer function, represented as a vector that lists the coefficients in order of descending powers of \(s\).

\section*{VCO input sensitivity ( \(\mathrm{Hz} / \mathrm{V}\) )}

This value scales the input to the VCO and, consequently, the shift from the VCO quiescent frequency value. The units of VCO input sensitivity are Hertz per volt.

VCO quiescent frequency \((\mathrm{Hz})\)
The frequency of the VCO signal when the voltage applied to it is zero. This should match the frequency of the input signal.

\section*{VCO initial phase (rad)}

The initial phase of the VCO signal.

\section*{VCO output amplitude}

The amplitude of the VCO signal.

\section*{See Also \\ Phase-Locked Loop}

References For more information about digital phase-locked loops, see the works listed in"Selected Bibliography for Synchronization" in Communications System Toolbox User's Guide.

\section*{CMA Equalizer}
\begin{tabular}{ll} 
Purpose & Equalize using constant modulus algorithm \\
Library & Equalizers
\end{tabular}

Description


The CMA Equalizer block uses a linear equalizer and the constant modulus algorithm (CMA) to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the CMA to update the weights, once per symbol. If the Number of samples per symbol parameter is 1 , then the block implements a symbol-spaced equalizer; otherwise, the block implements a fractionally spaced equalizer.

When using this block, you should initialize the equalizer weights with a nonzero vector. Typically, CMA is used with differential modulation; otherwise, the initial weights are very important. A typical vector of initial weights has a 1 corresponding to the center tap and zeros elsewhere.

\section*{Input and Output Signals}

The Input port accepts a scalar-valued or column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal.

You can configure the block to have one or more of the extra ports listed in the table below.
\begin{tabular}{l|l|l}
\hline Port & Meaning & How to Enable \\
\hline Err output & \begin{tabular}{l}
\(y\left(R-|y|^{2}\right)\), where \(y\) is \\
the equalized signal \\
and \(R\) is a constant \\
related to the signal \\
constellation
\end{tabular} & Select Output error. \\
\hline Wts output & \begin{tabular}{l} 
A vector listing the \\
weights after the \\
block has processed \\
either the current
\end{tabular} & \begin{tabular}{l} 
Select Output \\
weights.
\end{tabular} \\
\hline
\end{tabular}

\section*{CMA Equalizer}
\begin{tabular}{l|l|l}
\hline Port & Meaning & How to Enable \\
& \begin{tabular}{l} 
input frame or \\
sample.
\end{tabular} & \\
\hline
\end{tabular}

\section*{Algorithms}

Referring to the schematics in "Equalizer Structure", define \(w\) as the vector of all weights \(w_{\mathrm{i}}\) and define \(u\) as the vector of all inputs \(u_{\mathrm{i}}\). Based on the current set of weights, \(w\), this adaptive algorithm creates the new set of weights given by
\[
\text { (LeakageFactor) w + (StepSize) } u^{*} e
\]
where the * operator denotes the complex conjugate.

\section*{Equalizer Delay}

The delay between the transmitter's modulator output and the CMA equalizer output is typically unknown (unlike the delay for other adaptive equalizers in this product). If you need to determine the delay, you can use the Find Delay block.

\section*{CMA Equalizer}

Dialog Box


\section*{Parameters}

Number of taps:
4
Number of samples per symbol:
1
Signal constellation:
\([1+\mathrm{j}-1+\mathrm{j}-1-\mathrm{j} 1-\mathrm{j}]\)
Step size:
0.01

Leakage factor:
1
Initial weights:
[1000]
\(\sqrt{\checkmark}\) Output error
\(\Gamma\) Output weights

\section*{Number of taps}

The number of taps in the filter of the equalizer.

\section*{Number of samples per symbol}

The number of input samples for each symbol.
When you set this parameter to 1 , the filter weights are updated once for each symbol, for a symbol spaced (i.e. T-spaced) equalizer. When you set this parameter to a value greater than one, the
weights are updated once every \(\mathrm{N}^{\text {th }}\) sample, for a fractionally spaced (i.e. T/N-spaced) equalizer.

\section*{Signal constellation}

A vector of complex numbers that specifies the constellation for the modulation.

\section*{Step size}

The step size of the CMA.

\section*{Leakage factor}

The leakage factor of the CMA, a number between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

\section*{Initial weights}

A vector that lists the initial weights for the taps.

\section*{Output error}

If you check this box, the block outputs the error signal described in the table above.

\section*{Output weights}

If you check this box, the block outputs the current weights.

\author{
References \\ [1] Haykin, Simon, Adaptive Filter Theory, Third Ed., Upper Saddle River, N.J., Prentice-Hall, 1996. \\ [2] Johnson, Richard C. Jr., Philip Schniter, Thomas. J. Endres, et al., "Blind Equalization Using the Constant Modulus Criterion: A Review," Proceedings of the IEEE, vol. 86, pp. 1927-1950, October 1998.
}

See Also
LMS Linear Equalizer, LMS Decision Feedback Equalizer, RLS Linear Equalizer, RLS Decision Feedback Equalizer

\section*{Complex Phase Difference}

Purpose Output phase difference between two complex input signals

Library
Description

Complex Phase Difference

Dialog Box

See Also

Utility Blocks
The Complex Phase Difference block accepts two complex input signals that have the same size and frame status. The output is the phase difference from the second to the first, measured in radians. The elements of the output are between -п and п.

The input signals can have any size or frame status. This block processes each pair of elements independently.


Complex Phase Shift

\section*{Complex Phase Shift}

\section*{Purpose}

\section*{Library}

Description

Dialog Box

\section*{See Also}

Shift phase of complex input signal by second input value

Utility Blocks
The Complex Phase Shift block accepts a complex signal at the port labeled In. The output is the result of shifting this signal's phase by an amount specified by the real signal at the input port labeled Ph. The Ph input is measured in radians, and must have the same size and frame status as the In input.

The input signals can have any size or frame status. This block processes each pair of corresponding elements independently.


Complex Phase Difference

\section*{Continuous-Time VCO}

Purpose Implement voltage-controlled oscillator

Library
Description

Continuous-Time VCO

Continuous-Time VCO

Components sublibrary of Synchronization
The Continuous-Time VCO (voltage-controlled oscillator) block generates a signal with a frequency shift from the Quiescent frequency parameter that is proportional to the input signal. The input signal is interpreted as a voltage. If the input signal is \(u(t)\), then the output signal is
\[
y(t)=A_{c} \cos \left(2 \pi f_{c} t+2 \pi k_{c} \int_{0}^{t} u(\tau) d \tau+\varphi\right)
\]
where \(A_{c}\) is the Output amplitude parameter, \(f_{c}\) is the Quiescent frequency parameter, \(k_{\mathrm{c}}\) is the Input sensitivity parameter, and \(\varphi\) is the Initial phase parameter.

This block uses a continuous-time integrator to interpret the equation above.

The input and output are both sample-based scalar signals.

\section*{Continuous-Time VCO}


\section*{Dialog} Box

\section*{Output amplitude}

The amplitude of the output.

\section*{Quiescent frequency}

The frequency of the oscillator output when the input signal is zero.

\section*{Input sensitivity}

This value scales the input voltage and, consequently, the shift from the Quiescent frequency value. The units of Input sensitivity are Hertz per volt.

\section*{Initial phase}

The initial phase of the oscillator in radians.

\section*{Continuous-Time VCO}

\author{
See Also \\ Discrete-Time VCO
}

\section*{Convolutional Deinterleaver}

\section*{Purpose}

Restore ordering of symbols that were permuted using shift registers

\section*{Library}

Description

Convolutional Deinterleaver

Convolutional sublibrary of Interleaving
The Convolutional Deinterleaver block recovers a signal that was interleaved using the Convolutional Interleaver block. Internally, this block uses a set of shift registers. The delay value of the \(k^{\text {th }}\) shift register is ( \(N-k\) ) times the Register length step parameter.

The number of shift registers, \(N\), is the value of the Rows of shift registers parameter. The parameters in the two blocks must have the same values.

This block accepts a scalar or column vector input signal, which can be real or complex. The output signal has the same sample time as the input signal.

This block accepts the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point.

\section*{Convolutional Deinterleaver}


\section*{Rows of shift registers}

The number of shift registers that the block uses internally.

\section*{Register length step}

The difference in symbol capacity of each successive shift register, where the last register holds zero symbols.

\section*{Initial conditions}

Indicates the values that fill each shift register at the beginning of the simulation (except for the last shift register, which has zero delay).
- When you select a scalar value for Initial conditions, the value fills all shift registers (except for the last one)

\section*{Convolutional Deinterleaver}

\section*{Examples}

For an example that uses this block, see "Adaptive Algorithms".

\section*{Pair Block Convolutional Interleaver}

See Also General Multiplexed Deinterleaver, Helical Deinterleaver
References [1] Clark, George C. Jr. and J. Bibb Cain. Error-Correction Coding for Digital Communications. New York: Plenum Press, 1981.
[2] Forney, G., D., Jr. "Burst-Correcting Codes for the Classic Bursty Channel." IEEE Transactions on Communications, vol. COM-19, October 1971. 772-781.
[3] Ramsey, J. L. "Realization of Optimum Interleavers." IEEE Transactions on Information Theory, IT-16 (3), May 1970. 338-345.

\section*{Convolutional Encoder}
\begin{tabular}{ll} 
Purpose & Create convolutional code from binary data \\
Library & Convolutional sublibrary of Error Detection and Correction
\end{tabular}

Description
The Convolutional Encoder block encodes a sequence of binary input vectors to produce a sequence of binary output vectors. This block can process multiple symbols at a time.
This block can accept inputs that vary in length during simulation. For more information about variable-size signals, see "Variable-Size Signal Basics" in the Simulink documentation.

\section*{Input and Output Sizes}

If the encoder takes \(k\) input bit streams (that is, it can receive \(2^{k}\) possible input symbols), the block input vector length is \(\mathrm{L}^{*} k\) for some positive integer L. Similarly, if the encoder produces \(n\) output bit streams (that is, it can produce \(2^{n}\) possible output symbols), the block output vector length is \(\mathrm{L}^{*} n\).

This block accepts a column vector input signal with any positive integer for \(L\). For variable-size inputs, the \(L\) can vary during simulation. The operation of the block is governed by the Operation mode parameter.

For both its inputs and outputs for the data ports, the block supports double, single, boolean, int8, uint8, int16, uint16, int32, uint32, and ufix1. The port data types are inherited from the signals that drive the block. The input reset port supports double and boolean typed signals.

\section*{Specifying the Encoder}

To define the convolutional encoder, use the Trellis structure parameter. This parameter is a MATLAB structure whose format is described in the "Trellis Description of a Convolutional Code" section of the Communications System Toolbox documentation. You can use this parameter field in two ways:
- If you have a variable in the MATLAB workspace that contains the trellis structure, enter its name in the Trellis structure parameter.

\section*{Convolutional Encoder}

This way is preferable because it causes Simulink to spend less time updating the diagram at the beginning of each simulation, compared to the usage described next.
- If you want to specify the encoder using its constraint length, generator polynomials, and possibly feedback connection polynomials, use a poly2trellis command in the Trellis structure parameter. For example, to use an encoder with a constraint length of 7, code generator polynomials of 171 and 133 (in octal numbers), and a feedback connection of 171 (in octal), set the Trellis structure parameter to
```

poly2trellis(7,[171 133],171)

```

The encoder registers begin in the all-zeros state. Set the Operation mode parameter to Reset on nonzero input via port to reset all encoder registers to the all-zeros state during the simulation. This selection opens a second input port, labeled Rst, which accepts a scalar-valued input signal. When the input signal is nonzero, the block resets before processing the data at the first input port. To reset the block after it processes the data at the first input port, select Delay reset action to next time step.

\section*{Convolutional Encoder}

Dialog
Box

Function Block Parameters: Convolutional Encoder
Convolutional Encoder (mask) (link)
Convolutionally encode binary data. Use the poly2trellis function to create a trellis using the constraint length, code generator (octal) and feedback connection (octal).

Select the "Terminate trellis by appending bits" operation mode to terminate the trellis at the all-zero state by appending tail bits at the end of each input frame. Check the Puncture code checkbox to puncture the encoded data for all other operation modes.

Use the istrellis function in MATLAB to check if a structure is a valid trellis structure.

Parameters
Trellis structure:

\section*{poly2trellis(7, [171 133])}

Operation mode: ContinuousOutput final statePuncture code

\section*{Trellis structure}

MATLAB structure that contains the trellis description of the convolutional encoder.

\section*{Operation mode}

In Continuous mode, the block retains the encoder states at the end of each input, for use with the next frame.

In Truncated (reset every frame) mode, the block treats each input independently. The encoder states are reset to all-zeros state at the start of each input.

\section*{Convolutional Encoder}

Note When this block outputs sequences that vary in length during simulation and you set the Operation mode to Truncated (reset every frame) or Terminate trellis by appending bits, the block's state resets at every input time step.

In Terminate trellis by appending bits mode, the block treats each input independently. For each input frame, extra bits are used to set the encoder states to all-zeros state at the end of the frame. The output length is given by \(y=n \cdot(x+s) / k\), where \(x\) is the number of input bits, and \(s=\) constraint length -1 (or, in the case of multiple constraint lengths, \(s=\operatorname{sum}(\) ConstraintLength(i)-1)).

Note This block works for cases \(k \geq 1\), where it has the same values for constraint lengths in each input stream (e.g., constraint lengths of [2 2] or [7 7] will work, but [5 4] will not).

In Reset on nonzero input via port mode, the block has an additional input port, labeled Rst. When the Rst input is nonzero, the encoder resets to the all-zeros state.

\section*{Delay reset action to next time step}

When you select Delay reset action to next time step, the Convolutional Encoder block resets after computing the encoded data. This check box only appears when you set the Operation mode parameter to Reset on nonzero input via port.

The delay in the reset action allows the block to support HDL code generation. In order to generate HDL code, you must have an HDL Coder \({ }^{\mathrm{TM}}\) license.

\section*{Convolutional Encoder}

\section*{Output final state}

When you select Output final state, the second output port signal specifies the output state for the block. The output signal is a scalar, integer value. You can select Output final state for all operation modes except Terminate trellis by appending bits.

\section*{Specify initial state via input port}

When you select Specify initial state via input port the second input port signal specifies the starting state for every frame in the block. The input signal must be a scalar, integer value. Specify initial state via input port appears only in Truncated operation mode.

\section*{Puncture code}

Selecting this option opens the field Puncture vector.

\section*{Puncture vector}

Vector used to puncture the encoded data. The puncture vector is a pattern of 1 s and 0 s where the 0 s indicate the punctured bits. This field appears when you select Punctured code.

\section*{Puncture Pattern Examples}

See Also Viterbi Decoder, APP Decoder
References [1] Clark, George C. Jr. and J. Bibb Cain, Error-Correction Coding for Digital Communications, New York, Plenum Press, 1981.
[2] Gitlin, Richard D., Jeremiah F. Hayes, and Stephen B. Weinstein, Data Communications Principles, New York, Plenum, 1992.
[3] Yasuda, Y., et. al., "High rate punctured convolutional codes for soft decision Viterbi decoding," IEEE Transactions on Communications, Vol. COM-32, No. 3, pp 315-319, March 1984.

\section*{Convolutional Encoder}
[4] Haccoun, D., and Begin, G., "High-rate punctured convolutional codes for Viterbi and Sequential decoding," IEEE Transactions on Communications, Vol. 37, No. 11, pp 1113-1125, Nov. 1989.
[5] Begin, G., et.al., "Further results on high-rate punctured convolutional codes for Viterbi and sequential decoding," IEEE Transactions on Communications, Vol. 38, No. 11, pp 1922-1928, Nov. 1990.

\section*{Convolutional Interleaver}

\section*{Purpose Permute input symbols using set of shift registers \\ Library \\ Convolutional sublibrary of Interleaving}

Description

Convolutional Interleaver

The Convolutional Interleaver block permutes the symbols in the input signal. Internally, it uses a set of shift registers. The delay value of the kth shift register is (k-1) times the Register length step parameter. The number of shift registers is the value of the Rows of shift registers parameter.

The Initial conditions parameter indicates the values that fill each shift register at the beginning of the simulation (except for the first shift register, which has zero delay). If Initial conditions is a scalar, then its value fills all shift registers except the first; if Initial conditions is a column vector whose length is the Rows of shift registers parameter, then each entry fills the corresponding shift register. The value of the first element of the Initial conditions parameter is unimportant, since the first shift register has zero delay.

This block accepts a scalar or column vector input signal, which can be real or complex. The output signal has the same sample time as the input signal.

The block can accept the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The data type of this output will be the same as that of the input signal.

\section*{Convolutional Interleaver}

Dialog Box


\section*{Rows of shift registers}
The number of shift registers that the block uses internally.

\section*{Register length step}
The number of additional symbols that fit in each successive shift register, where the first register holds zero symbols.

\section*{Initial conditions}
The values that fill each shift register when the simulation begins.

\section*{Examples}
For an example that uses this block, see "Convolutional Interleaving".
Pair Block Convolutional Deinterleaver
See Also General Multiplexed Interleaver, Helical Interleaver

\section*{Convolutional Interleaver}

References [1] Clark, George C. Jr. and J. Bibb Cain. Error-Correction Coding for Digital Communications. New York: Plenum Press, 1981.
[2] Forney, G., D., Jr. "Burst-Correcting Codes for the Classic Bursty Channel." IEEE Transactions on Communications, vol. COM-19, October 1971. 772-781.
[3] Ramsey, J. L. "Realization of Optimum Interleavers." IEEE Transactions on Information Theory, IT-16 (3), May 1970. 338-345.

\section*{CPFSK Demodulator Baseband}

\section*{Purpose \\ Library \\ Description}

Demodulate CPFSK-modulated data

CPM, in Digital Baseband sublibrary of Modulation
The CPFSK Demodulator Baseband block demodulates a signal that was modulated using the continuous phase frequency shift keying method. The input to this block is a baseband representation of the modulated signal. The M-ary number parameter, \(M\), is the size of the input alphabet. \(M\) must have the form \(2^{K}\) for some positive integer \(K\).

This block supports multi-h Modulation index. See CPM Modulator Baseband for details.

\section*{Integer-Valued Signals and Binary-Valued Signals}

When you set the Output type parameter to Integer, then the block produces odd integers between -(M-1) and M-1.

When you set the Output type parameter to Bit, then the block produces groupings of \(K\) bits. Each grouping is called a binary word.

In binary output mode, the block first maps each input symbol to an intermediate value as in the integer output mode. The block then maps the odd integer \(k\) to the nonnegative integer \((k+M-1) / 2\). Finally, the block maps each nonnegative integer to a binary word, using a mapping that depends on whether the Symbol set ordering parameter is set to Binary or Gray.

This block accepts a scalar-valued or column vector input signal with a data type of single or double.

\section*{Single-Rate Processing}

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. The input width must be an integer multiple of the Samples per symbol parameter value, and the input can be a column vector.
- When you set Output type to Bit, the output width is \(K\) times the number of input symbols.

\section*{CPFSK Demodulator Baseband}
- When you set Output type to Integer, the output width is the number of input symbols.

\section*{Multirate Processing}

In multirate processing mode, the input and output signals have different port sample times. The input must be a scalar. The output symbol time is the product of the input sample time and the Samples per symbol parameter value.
- When you set Output type to Bit, the output width equals the number of bits per symbol.
- When you set Output type to Integer, the output is a scalar.

\section*{Traceback Depth and Output Delays}

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The Traceback depth parameter, \(D\), in this block is the number of trellis branches that the algorithm uses to construct each traceback path. \(D\) influences the output delay, which is the number of zero symbols that precede the first meaningful demodulated value in the output.
- When you set the Rate options parameter to Allow multirate processing, and the model uses a variable-step solver or a fixed-step solver with the Tasking Mode parameter set to SingleTasking, then the delay consists of \(\mathrm{D}+1\) zero symbols.
- When you set the Rate options parameter to Enforce single-rate processing, then the delay consists of D zero symbols.

The optimal Traceback depth parameter value is dependent on minimum squared Euclidean distance calculations. Alternatively, a typical value, dependent on the number of states, can be chosen using the "five-times-the-constraint-length" rule, which corresponds to \(5 \cdot \log 2(\) numStates \()\).
For the definition of the number of states, see CPM Demodulator Baseband Help page.

\section*{CPFSK Demodulator Baseband}


Dialog Box

\section*{M-ary number}

The size of the alphabet.

\section*{Output type}

Determines whether the output consists of integers or groups of bits.

\section*{Symbol set ordering}

Determines how the block maps each integer to a group of output bits. This field is active only when Output type is set to Bit.

\section*{CPFSK Demodulator Baseband}

\section*{Modulation index}

Specify the modulation index \(\left\{h_{i}\right\}\). The default is 0.5 . The value of this property must be a real, nonnegative scalar or column vector.

This block supports multi-h Modulation index. See CPM Modulator Baseband for details.

\section*{Phase offset (rad)}

The initial phase of the modulated waveform.

\section*{Samples per symbol}

The number of input samples that represent each modulated symbol, which must be a positive integer. For more information, see "Upsample Signals and Rate Changes" in Communications System Toolbox User's Guide.

\section*{Rate options}

Select the rate processing method for the block.
- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width is the number of symbols (which is given by dividing the input length by the Samples per symbol parameter value when the Output type parameter is set to Integer).
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output period is the same as the symbol period and equals the product of the input period and the Samples per symbol parameter value.

\section*{CPFSK Demodulator Baseband}

> Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

For more information, see Single-Rate Processing and Multirate Processing in the Description section of this page.

\section*{Traceback depth}

The number of trellis branches that the CPFSK Demodulator Baseband block uses to construct each traceback path.

\section*{Output datatype}

The output data type can be boolean, int8, int16, int32, or double.

\section*{Supported Data Types}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & • Double-precision floating point \\
& • Single-precision floating point
\end{tabular}

Pair Block CPFSK Modulator Baseband
See Also CPM Demodulator Baseband, Viterbi Decoder, M-FSK Demodulator Baseband

\author{
References \\ [1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. Digital Phase Modulation. New York: Plenum Press, 1986.
}

\section*{CPFSK Modulator Baseband}
\begin{tabular}{ll} 
Purpose & Modulate using continuous phase frequency shift keying method \\
Library & CPM, in Digital Baseband sublibrary of Modulation
\end{tabular}

Description
いTNM CPFSK

CPM, in Digital Baseband sublibrary of Modulation
The CPFSK Modulator Baseband block modulates a signal using the continuous phase frequency shift keying method. The output is a baseband representation of the modulated signal. The M-ary number parameter, \(M\), represents the size of the input alphabet. \(M\) must have the form \(2^{K}\) for some positive integer \(K\).

This block supports multi-h Modulation index. See CPM Modulator Baseband for details.

\section*{Integer-Valued Signals and Binary-Valued Signals}

When you set the Input type parameter to Integer, the block accepts odd integers between -(M-1) and \(\mathrm{M}-1\).

When you set the Input type parameter to Bit, the block accepts groupings of \(K\) bits. Each grouping is called a binary word. The input vector length must be an integer multiple of \(K\).

In binary input mode, the block maps each binary word to an integer between 0 and M-1, using a mapping scheme that depends on whether you set the Symbol set ordering parameter to Binary or Gray. The block then maps the integer \(k\) to the intermediate value 2 k -(M-1) and proceeds as if it operates in the integer input mode. For more information, see "Integer-Valued Signals and Binary-Valued Signals" in Communications System Toolbox User's Guide.

This block accepts a scalar-valued or column vector input signal. If you set Input type to Bit, then the input signal can also be a vector of length \(K\).

\section*{Single-Rate Processing}

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. In this mode, the input to the block can be multiple symbols.

\section*{CPFSK Modulator Baseband}
- When you set Input type to Integer, the input can be a column vector, the length of which is the number of input symbols.
- When you set Input type to Bit, the input must be a column vector with a width that is an integer multiple of \(K\), the number of bits per symbol.

The output width equals the product of the number of input symbols and the Samples per symbol parameter value.

\section*{Multirate Processing}

In multirate processing mode, the input and output signals have different port sample times. In this mode, the input to the block must be one symbol.
- When you set Input type to Integer, the input must be a scalar.
- When you set Input type to Bit, the input width must equal the number of bits per symbol.

The output sample time equals the symbol period divided by the Samples per symbol parameter value.

\section*{CPFSK Modulator Baseband}
 Box

\section*{M-ary number}

The size of the alphabet.

\section*{Input type}

Indicates whether the input consists of integers or groups of bits.

\section*{Symbol set ordering}

Determines how the block maps each group of input bits to a corresponding integer. This field is active only when Input type is set to Bit.

\section*{CPFSK Modulator Baseband}

\section*{Modulation index}

Specify the modulation index \(\left\{h_{i}\right\}\). The default is 0.5 . The value of this property must be a real, nonnegative scalar or column vector.

This block supports multi-h Modulation index. See CPM Modulator Baseband for details.

\section*{Phase offset (rad)}

The initial phase of the output waveform, measured in radians.

\section*{Samples per symbol}

The number of output samples that the block produces for each integer or binary word in the input, which must be a positive integer. For all non-binary schemes, as defined by the pulse shapes, this value must be greater than 1 .

For more information, see "Upsample Signals and Rate Changes" in Communications System ToolboxUser's Guide.

\section*{Rate options}

Select the rate processing option for the block.
- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width equals the product of the number of symbols and the Samples per symbol parameter value.
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output sample time equals the symbol period divided by the Samples per symbol parameter value.

\section*{CPFSK Modulator Baseband}

> Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

\section*{Output data type}

Select the data type of the output signal. The output data type can be single or double.

\section*{Supported Data Types}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & • Double-precision floating point \\
& • Boolean (When Input type set to Bit) \\
& \begin{tabular}{l} 
• 8-, 16-, and 32-bit signed integers (When Input type \\
set to Integer)
\end{tabular} \\
\hline Output & \begin{tabular}{l} 
• Double-precision floating point \\
\\
\\
• Single-precision floating point
\end{tabular} \\
\hline
\end{tabular}

\section*{Pair Block}

CPFSK Demodulator Baseband
See Also CPM Modulator Baseband, M-FSK Modulator Baseband
References
[1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. Digital Phase Modulation. New York: Plenum Press, 1986.

\section*{CPM Demodulator Baseband}

\section*{Purpose}

Demodulate CPM-modulated data

\section*{Library}

Description
 CPM

CPM, in Digital Baseband sublibrary of Modulation

The Continuous Phase Modulation (CPM) Demodulator Baseband block demodulates a signal that was modulated using continuous phase modulation. The input is a baseband representation of the modulated signal:
\[
\begin{aligned}
& s(t)=\exp \left[j 2 \pi \sum_{i=0}^{n} \alpha_{i} h_{i} q(t-i T)\right], \\
& n T<t<(n+1) T
\end{aligned}
\]

See the CPM Modulator Baseband block reference page for the definition of \(\left\{\alpha_{i}\right\},\left\{h_{i}\right\}\), and \(q(t)\).

This block accepts a scalar-valued or a column vector input signal with a data type of single or double. CPM is a modulation method with memory. The optimum receiver consists of a correlator followed by a maximum-likelihood sequence detector (MLSD) that searches the paths through the state trellis for the minimum Euclidean distance path. When the Modulation index \(h\) is rational, i.e., \(h=m / p\), there are a finite number of phase states and the block uses the Viterbi algorithm to perform MLSD.
\(\left\{h_{i}\right\}\) represents a sequence of modulation indices that moves cyclically through a set of indices \(\left\{\mathrm{h}_{0}, \mathrm{~h}_{1}, \mathrm{~h}_{2}, \ldots, \mathrm{~h}_{\mathrm{H}-1}\right\}\).
- \(h_{i}=m_{i} / p_{i}\) represents the modulation index in proper rational form
- \(\mathrm{m}_{\mathrm{i}}\) represents the numerator of modulation index
- \(p_{i}\) represents the denominator of modulation index
- \(\mathrm{m}_{\mathrm{i}}\) and \(\mathrm{p}_{\mathrm{i}}\) are relatively prime positive numbers
- The Least Common Multiple (LCM) of \(\left\{\mathrm{p}_{0}, \mathrm{p}_{1}, \mathrm{p}_{2}, \ldots, \mathrm{p}_{\mathrm{H}-1}\right\}\) is denoted as \(p\)

\section*{CPM Demodulator Baseband}
- \(\mathrm{h}_{\mathrm{i}}=\mathrm{m}_{\mathrm{i}}^{\prime} / \mathrm{p}\)
\(\left\{h_{i}\right\}\) determines the number of phase states:
\[
\text { numPhaseStates }=\left\{\begin{array}{l}
p, \text { for all even } m_{i}^{\prime} \\
2 p, \text { for any odd } m_{i}^{\prime}
\end{array}\right\}
\]
and affects the number of trellis states:
numStates \(=\) numPhaseStates \({ }^{*} M^{(L-1)}\)
where
- \(L\) represents the Pulse length
- \(M\) represents the M-ary number

\section*{Integer-Valued Signals and Binary-Valued Signals}

When you set the Output type parameter to Integer, then the block produces odd integers between -(M-1) and M-1. When you set the Output type to Integer, you cannot set Output datatype to boolean.

When you set the Output type parameter to Bit, then the block produces groupings of \(K\) bits. Each grouping is called a binary word. When you set the Output type to Bit, the Output datatype can only be double or boolean.

In binary output mode, the block first maps each input symbol to an intermediate value as in the integer output mode. Then, the block maps the odd integer \(k\) to the nonnegative integer \((k+M-1) / 2\). Finally, the block maps each nonnegative integer to a binary word, using a mapping that depends on whether you set the Symbol set ordering parameter to Binary or Gray.

\section*{Single-Rate Processing}

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the

\section*{CPM Demodulator Baseband}
input. The input width must be an integer multiple of the Samples per symbol parameter value, and the input can be a column vector.
- When you set Output type to Bit, the output width is \(K\) times the number of input symbols.
- When you set Output type to Integer, the output width is the number of input symbols.

\section*{Multirate Processing}

In multirate processing mode, the input and output signals have different port sample times. The input must be a scalar. The output symbol time is the product of the input sample time and the Samples per symbol parameter value.
- When you set Output type to Bit, the output width equals the number of bits per symbol.
- When you set Output type to Integer, the output is a scalar.

\section*{Traceback Depth and Output Delays}

The Traceback depth parameter, D, in this block is the number of trellis branches used to construct each traceback path. D influences the output delay, which is the number of zero symbols that precede the first meaningful demodulated value in the output.
- When you set the Rate options parameter to Allow multirate processing, and the model uses a variable-step solver or a fixed-step solver with the Tasking Mode parameter set to SingleTasking, then the delay consists of \(\mathrm{D}+1\) zero symbols.
- When you set the Rate options parameter to Enforce single-rate processing, the delay consists of D zero symbols.

The optimal Traceback depth parameter value is dependent on minimum squared Euclidean distance calculations. Alternatively, a typical value, dependent on the number of states, can be chosen

\section*{CPM Demodulator Baseband}
using the "five-times-the-constraint-length" rule, which corresponds to \(5 \cdot \log 2\) (numStates).

For a binary raised cosine pulse shape with a pulse length of \(3, \mathrm{~h}=2 / 3\),
this rule \(\left(5^{*} \log 2\left(3^{*} 2^{2}\right)=18\right)\) gives a result close to the optimum value of 20 .

\section*{CPM Demodulator Baseband}

Function Block Parameters: CPM Demodulator Basebant CPM Demodulator Baseband (mask) (link)

Demodulate the CPM modulated input signal using the Viterbi algorithm.
For the multirate processing option, this block accepts a scalar input signal. For the single-rate processing option, this block accepts a column vector input signal whose width is an integer multiple of the Samples per symbol parameter.

The output signal can be either bits or integers. When you set the 'Output type' parameter to 'Bit', the output width is an integer multiple of the number of bits per symbol.

Parameters
M-ary number:
4
Output type: Integer
Symbol set ordering: Binary
Modulation index:
0.5

Frequency pulse shape: Rectangular
Pulse length (symbol intervals):
1
Symbol prehistory:
1
Phase offset (rad):
0
Samples per symbol:
8
Rate options: Enforce single-rate processing
Traceback depth:
16
Output datatype: double

Dialog Box

\section*{M-ary number}

The size of the alphabet.

\section*{CPM Demodulator Baseband}

\section*{Output type}

Determines whether the output consists of integers or groups of bits.

\section*{Symbol set ordering}

Determines how the block maps each integer to a group of output bits. This field applies only when you set Output type to Bit.

\section*{Modulation index}

Specify the modulation index \(\left\{h_{i}\right\}\). The default is 0.5 . The value of this property must be a real, nonnegative scalar or column vector.

\section*{Frequency pulse shape}

Specify the type of pulse shaping that the corresponding modulator uses to smooth the phase transitions of the modulated signal. You can select from the following pulse shapes:
- Rectangular
- Raised Cosine
- Spectral Raised Cosine

This option requires an additional parameter, Rolloff. The Rolloff parameter, which affects the spectrum of the pulse, is a scalar between zero and one.
- Gaussian

This option requires an additional parameter, BT product. The BT product parameter, which represents bandwidth multiplied by time, is a nonnegative scalar. It is used to reduce the bandwidth at the expense of increased intersymbol interference.
- Tamed FM (tamed frequency modulation)

\section*{Main lobe pulse duration (symbol intervals)}

Number of symbol intervals of the largest lobe of the spectral raised cosine pulse. This field is active only when Frequency pulse shape is set to Spectral Raised Cosine.

\section*{CPM Demodulator Baseband}

\section*{Rolloff}

The rolloff factor of the specified raised cosine filter. This field appears only when you set Frequency pulse shape to Spectral Raised Cosine.

\section*{BT product}

The product of bandwidth and time. This field appears only when Frequency pulse shape is set to Gaussian.

\section*{Pulse length (symbol intervals)}

The length of the frequency pulse shape.

\section*{Symbol prehistory}

The data symbols the modulator uses before the start of the simulation.

\section*{Phase offset (rad)}

The initial phase of the modulated waveform.

\section*{Samples per symbol}

The number of input samples that represent each modulated symbol. For more information, see "Upsample Signals and Rate Changes" in Communications System ToolboxUser's Guide.

\section*{Rate options}

Select the rate processing method for the block.
- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width is the number of symbols (which is given by dividing the input length by the Samples per symbol parameter value when the Output type parameter is set to Integer).
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output period is the same as the symbol period and equals the product of the input period and the Samples per symbol parameter value.

\section*{CPM Demodulator Baseband}

> Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

For more information, see Single-Rate Processing and Multirate Processing in the Description section of this page.

\section*{Traceback depth}

The number of trellis branches that the CPM Demodulator block uses to construct each traceback path.

\section*{Output datatype}

The output data type can be boolean, int8, int16, int32, or double.

\section*{Supported Data Types}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & • Double-precision floating point \\
& • Single-precision floating point
\end{tabular}

Pair Block CPM Modulator Baseband
See Also CPFSK Demodulator Baseband, GMSK Demodulator Baseband, MSK Demodulator Baseband, Viterbi Decoder

References [1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. Digital Phase Modulation. New York: Plenum Press, 1986.

\section*{CPM Modulator Baseband}

\section*{Purpose}

Modulate using continuous phase modulation

\section*{Library}

Description
万Tmm
CPM
CPM, in Digital Baseband sublibrary of Modulation
The Continuous Phase Modulation (CPM) Modulator Baseband block modulates an input signal using continuous phase modulation. The output is a baseband representation of the modulated signal:
\[
\begin{aligned}
& s(t)=\exp \left[j 2 \pi \sum_{i=0}^{n} \alpha_{i} h_{i} q(t-i T)\right], \\
& n T<t<(n+1) T
\end{aligned}
\]
where
- \(\left\{a_{i}\right\}\) represents a sequence of \(M\)-ary data symbols selected from the alphabet \(\pm 1, \pm 3, \pm(M-1)\).
- \(M\) must have the form \(2^{K}\) for some positive integer \(K\). You specify the value of \(M\) using the \(\mathbf{M}\)-ary number parameter.
- \(\left\{h_{i}\right\}\) represents a sequence of modulation indices and \(h_{i}\) moves cyclically through a set of indices \(\left\{\mathrm{h}_{0}, \mathrm{~h}_{1}, \mathrm{~h}_{2}, \ldots, \mathrm{~h}_{\mathrm{H}-1}\right\}\). When \(H=1\), there is only one modulation index, \(\mathrm{h}_{0}\), which is denoted as \(h\).

When \(h_{i}\) varies from interval to interval, the block operates in multi-h. To ensure a finite number of phase states, \(h_{\mathrm{i}}\) must be a rational number. You specify the value(s) of \(h_{i}\) using the Modulation index parameter.

Continuous phase modulation uses pulse shaping to smooth the phase transitions of the modulated signal. The function \(q(t)\) is the phase response obtained from the frequency pulse, \(g(t)\), through the relation:
\[
q(t)=\int_{-\infty}^{t} g(t) d t
\]

\section*{CPM Modulator Baseband}

Using the Frequency pulse shape parameter, you can select the following pulse shapes:
- Rectangular
- Raised Cosine
- Spectral Raised Cosine
- Gaussian
- Tamed FM (tamed frequency modulation)

For the exact definitions of these pulse shapes, see the work by Anderson, Aulin, and Sundberg among the references at the end of this page. Each pulse shape has a corresponding pulse duration. The Pulse length (symbol intervals) parameter measures this quantity in symbol intervals.

\section*{Integer-Valued Signals and Binary-Valued Signals}

When you set the Input type parameter to Integer, then the block accepts odd integers between -(M-1) and M-1. \(M\) represents the M-ary number block parameter.

When you set the Input type parameter to Bit, the block accepts binary-valued inputs that represent integers. The block collects binary-valued signals into groups of \(K=\log _{2}(M)\) bits
where
\(K\) represents the number of bits per symbol.
The input vector length must be an integer multiple of \(K\). In this configuration, the block accepts a group of \(K\) bits and maps that group onto a symbol at the block output. The block outputs one modulated symbol, oversampled by the Samples per symbol parameter value, for each group of \(K\) bits.
This block accepts a scalar-valued or column vector input signal. For a column vector input signal, the width of the output frame equals the product of the number of symbols and the value for the Samples per

\section*{CPM Modulator Baseband}
symbol parameter. For a sample-based input signal, the output sample time equals the symbol period divided by the value for the Samples per symbol parameter. For information about the data types each block port supports, see the "Supported Data Types" on page 2-160 table on this page.

\section*{Symbol Sets}

In binary input mode, the block maps each binary word to an integer between 0 and M-1, using a mapping that depends on whether the Symbol set ordering parameter is set to Binary or Gray. The block then maps the integer k to the intermediate value 2 k -( \(\mathrm{M}-1\) ) and proceeds as in the integer input mode. For more information, see Integer-Valued Signals and Binary-Valued Signals on the M-PSK Modulator ref page.

\section*{Single-Rate Processing}

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. In this mode, the input to the block can be multiple symbols.
- When you set Input type to Integer, the input can be a column vector, the length of which is the number of input symbols.
- When you set Input type to Bit, the input width must be an integer multiple of \(K\), the number of bits per symbol.

The output width equals the product of the number of input symbols and the Samples per symbol parameter value.

\section*{Multirate Processing}

In multirate processing mode, the input and output signals have different port sample times. In this mode, the input to the block must be one symbol.
- When you set Input type to Integer, the input must be a scalar.
- When you set Input type to Bit, the input width must equal the number of bits per symbol.

\section*{CPM Modulator Baseband}

The output sample time equals the symbol period divided by the Samples per symbol parameter value.

\section*{CPM Modulator Baseband}

目 Function Block Parameters: CPM Modulator Baseband X CPM Modulator Baseband (mask) (link)

Output the complex envelope representation of the selected continuous phase modulation.

The input signal can be either bits or integers. For the single-rate processing option with bit inputs, the input width must be an integer multiple of the number of bits per symbol. For the multirate processing option with bit inputs, the input width must equal the number of bits per symbol.

For the single-rate processing option with integer inputs, this block accepts a scalar or column vector input signal. For the multirate processing option with integer inputs, this block accepts a scalar input signal.

Parameters
M-ary number:
4


Symbol set ordering: Binary
Modulation index:
0.5

Frequency pulse shape: Rectangular
Pulse length (symbol intervals):
1
Symbol prehistory:
1

Phase offset (rad):
0
Samples per symbol:

8

Rate options: Enforce single-rate processing
Output data type: double

\section*{M-ary number}

The size of the alphabet.

\section*{CPM Modulator Baseband}

\section*{Input type}

Indicates whether the input consists of integers or groups of bits.

\section*{Symbol set ordering}

Determines how the block maps each group of input bits to a corresponding integer. This field is active only when Input type is set to Bit.

\section*{Modulation index}

Specify the modulation index \(\left\{h_{i}\right\}\). The default is 0.5 . The value of this property must be a real, nonnegative scalar or column vector.

\section*{Frequency pulse shape}

Specify the type of pulse shaping that the block uses to smooth the phase transitions of the modulated signal. You can select from the following pulse shapes:
- Rectangular
- Raised Cosine
- Spectral Raised Cosine

This option requires an additional parameter, Rolloff. The Rolloff parameter, which affects the spectrum of the pulse, is a scalar between zero and one.
- Gaussian

This option requires an additional parameter, BT product. The BT product parameter, which represents bandwidth multiplied by time, is a nonnegative scalar. It is used to reduce the bandwidth at the expense of increased intersymbol interference.
- Tamed FM (tamed frequency modulation)

\section*{Main lobe pulse duration (symbol intervals)}

Number of symbol intervals of the largest lobe of the spectral raised cosine pulse. This field is active only when Frequency pulse shape is set to Spectral Raised Cosine.

\section*{CPM Modulator Baseband}

\section*{Rolloff}

The rolloff factor of the specified spectral raised cosine pulse shape. This field appears only when you set Frequency pulse shape to Spectral Raised Cosine.

\section*{BT product}

The product of bandwidth and time. This field appears only when Frequency pulse shape is set to Gaussian.

\section*{Pulse length (symbol intervals)}

The length of the frequency pulse shape.

\section*{Symbol prehistory}

A scalar-valued or vector signal that specifies the data symbols used before the start of the simulation, in reverse chronological order. If Symbol prehistory is a vector, then its length must be one less than the Pulse length parameter.

\section*{Phase offset (rad)}

The initial phase of the output waveform, measured in radians.

\section*{Samples per symbol}

The number of output samples that the block produces for each integer or binary word in the input, which must be a positive integer. For all non-binary schemes, as defined by the pulse shapes, this value must be greater than 1 .

For more information, see "Upsample Signals and Rate Changes" in Communications System ToolboxUser's Guide.

\section*{Rate options}

Select the rate processing option for the block.
- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width equals the product of the number of symbols and the Samples per symbol parameter value.

\section*{CPM Modulator Baseband}
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output sample time equals the symbol period divided by the Samples per symbol parameter value.

Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

\section*{Output data type}

Specify the block output data type as double and single.
Supported
Data
Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & - Double-precision floating point \\
& - Boolean (when Input type set to Bit) \\
& - 8-, 16-, and 32-bit signed integers (when Input type set \\
& to Integer) \\
\hline Output & - Double-precision floating point \\
& - Single-precision floating point \\
\hline
\end{tabular}

\author{
Pair Block \\ See Also \\ References \\ CPM Demodulator Baseband \\ CPFSK Modulator Baseband, GMSK Modulator Baseband, MSK Modulator Baseband \\ [1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. Digital Phase Modulation. New York: Plenum Press, 1986.
}

\section*{Purpose}

Recover carrier phase using 2P-Power method

\section*{Library}

Description


Carrier Phase Recovery sublibrary of Synchronization
The CPM Phase Recovery block recovers the carrier phase of the input signal using the 2P-Power method. This feedforward, non-data-aided, clock-aided method is suitable for systems that use these types of baseband modulation: continuous phase modulation (CPM), minimum shift keying (MSK), continuous phase frequency shift keying (CPFSK), and Gaussian minimum shift keying (GMSK). This block is suitable for use with blocks in the Baseband Continuous Phase Modulation library.

If you express the modulation index for CPM as a proper fraction, \(\mathrm{h}=\mathrm{K} / \mathrm{P}\), then P is the number to which the name "2P-Power" refers. The observation interval parameter must be an integer multiple of the input signal vector length.

The 2P-Power method assumes that the carrier phase is constant over a series of consecutive symbols, and returns an estimate of the carrier phase for the series. The Observation interval parameter is the number of symbols for which the carrier phase is assumed constant. This number must be an integer multiple of the input signal's vector length.

\section*{Input and Output Signals}

This block accepts a scalar or column vector input signal of type double or single. The input signal represents a baseband signal at the symbol rate, so it must be complex-valued and must contain one sample per symbol.
The outputs are as follows:
- The output port labeled Sig gives the result of rotating the input signal counterclockwise, where the amount of rotation equals the carrier phase estimate. The Sig output is thus a corrected version of the input signal, and has the same sample time and vector size as the input signal.
- The output port labeled Ph outputs the carrier phase estimate, in degrees, for all symbols in the observation interval. The Ph output is a scalar signal.

Note Because the block internally computes the argument of a complex number, the carrier phase estimate has an inherent ambiguity. The carrier phase estimate is between -90/P and 90/P degrees and might differ from the actual carrier phase by an integer multiple of 180/P degrees.

\section*{Delays and Latency}

The block's algorithm requires it to collect symbols during a period of length Observation interval before computing a single estimate of the carrier phase. Therefore, each estimate is delayed by Observation interval symbols and the corrected signal has a latency of Observation interval symbols, relative to the input signal.

Dialog Box


P
The denominator of the modulation index for CPM ( \(\mathrm{h}=\mathrm{K} / \mathrm{P}\) ) when expressed as a proper fraction.

\section*{Observation interval}

The number of symbols for which the carrier phase is assumed constant. The observation interval parameter must be an integer multiple of the input signal vector length.

When this parameter is exactly equal to the vector length of the input signal, then the block always works. When the integer multiple is not equal to 1 , select Simulation \(>\) Configuration Parameters > Solver
and set Tasking mode for periodic sample times to SingleTasking.

\footnotetext{
Algorithm If the symbols occurring during the observation interval are \(x(1), x(2)\), \(\mathrm{x}(3), \ldots, \mathrm{x}(\mathrm{L})\), then the resulting carrier phase estimate is
\[
\frac{1}{2 P} \arg \left\{\sum_{k=1}^{L}(x(k))^{2 P}\right\}
\]
where the arg function returns values between - 180 degrees and 180 degrees.

\author{
References [1] Mengali, Umberto, and Aldo N. D'Andrea, Synchronization Techniques for Digital Receivers, New York, Plenum Press, 1997.
}

\author{
See Also M-PSK Phase Recovery, CPM Modulator Baseband
}
}

\section*{CRC-N Generator}

\section*{Purpose}

\section*{Library}

Description Generator

Generate CRC bits according to CRC method and append to input data frames

CRC sublibrary of Error Detection and Correction
The CRC-N Generator block generates cyclic redundancy code (CRC) bits for each input data frame and appends them to the frame. The input must be a binary column vector. The CRC-N Generator block is a simplified version of the General CRC Generator block. With the CRC-N Generator block, you can select the generator polynomial for the CRC algorithm from a list of commonly used polynomials, given in the CRC-N method field in the block's dialog. N is degree of the generator polynomial. The table below lists the options for the generator polynomial.
\begin{tabular}{l|l|l}
\hline CRC Method & Generator Polynomial & Number of Bits \\
\hline CRC-32 & \begin{tabular}{l}
\(\mathrm{x}^{32}+\mathrm{x}^{26}+\mathrm{x}^{23}+\mathrm{x}^{22}+\mathrm{x}^{16}+\mathrm{x}^{12}+\mathrm{x}^{11}\) \\
\(+\mathrm{x}^{10}+\mathrm{x}^{8}+\mathrm{x}^{7}+\mathrm{x}^{5}+\mathrm{x}^{4}+\mathrm{x}^{2}+\mathrm{x}+1\)
\end{tabular} & 32 \\
\hline CRC-24 & \(\mathrm{x}^{24}+\mathrm{x}^{23}+\mathrm{x}^{14}+\mathrm{x}^{12}+\mathrm{x}^{8}+1\) & 24 \\
\hline CRC-16 & \(\mathrm{x}^{16}+\mathrm{x}^{15}+\mathrm{x}^{2}+1\) & 16 \\
\hline \begin{tabular}{l} 
Reversed \\
CRC-16
\end{tabular} & \(\mathrm{x}^{16+\mathrm{x}^{14}+\mathrm{x}+1}\) & 16 \\
\hline CRC-8 & \(\mathrm{x}^{8}+\mathrm{x}^{7}+\mathrm{x}^{6}+\mathrm{x}^{4}+\mathrm{x}^{2}+1\) & 8 \\
\hline CRC-4 & \(\mathrm{x}^{4}+\mathrm{x}^{3}+\mathrm{x}^{2}+\mathrm{x}+1\) & 4 \\
\hline
\end{tabular}

You specify the initial state of the internal shift register using the Initial states parameter. You specify the number of checksums that the block calculates for each input frame using the Checksums per frame parameter. For more detailed information, see the reference page for the General CRC Generator block.

This block supports double and boolean data types. The output data type is inherited from the input.

\section*{CRC-N Generator}

\section*{Signal Attributes}

The General CRC Generator block has one input port and one output port. Both ports accept binary column vector input signals.


Dialog Box

\section*{CRC-N method}

The generator polynomial for the CRC algorithm.

\section*{Initial states}

A binary scalar or a binary row vector of length equal to the degree of the generator polynomial, specifying the initial state of the internal shift register.

\section*{Checksums per frame}

A positive integer specifying the number of checksums the block calculates for each input frame.

\section*{Algorithm}

Schematic of the CRC Implementation

For a description of the CRC algorithm as implemented by this block, see "CRC Non-Direct Algorithm" in Communications System Toolbox User's Guide.


The above circuit divides the polynomial
\(a(x)=a_{k-1} x^{k-1}+a_{k-2} x^{k-2}+\cdots+a_{1} x+a_{0} \quad\) by
\(g(x)=g_{r-1} x^{r-1}+g_{r-2} x^{r-2}+\cdots+g_{1} x+g_{0}\), and returns the remainder
\(d(x)=d_{r-1} x^{r-1}+d_{r-2} x^{r-2}+\cdots+d_{1} x+d_{0}\).
The input symbols \(\left\{a_{k-1}, a_{k-2}, \ldots, a_{2}, a_{1}, a_{0}\right\}\) are fed into the shift register one at a time in order of decreasing index. When the last symbol ( \({ }^{a_{0}}\) ) works its way out of the register (achieved by augmenting the message with r zeros), the register contains the coefficients of the remainder polynomial \(d(x)\).

This remainder polynomial is the checksum that is appended to the original message, which is then transmitted.

\section*{References}

Pair Block
[1] Sklar, Bernard, Digital Communications: Fundamentals and Applications. Englewood Cliffs, N.J., Prentice-Hall, 1988.
[2] Wicker, Stephen B., Error Control Systems for Digital Communication and Storage, Upper Saddle River, N.J., Prentice Hall, 1995.

CRC-N Syndrome Detector

CRC-N Generator

\author{
See Also \\ General CRC Generator, General CRC Syndrome Detector
}

\section*{CRC-N Syndrome Detector}

\section*{Purpose}

Detect errors in input data frames according to selected CRC method

\section*{Library}

Description

CRC-N Syndrome Detector Err

CRC sublibrary of Error Detection and Correction
The CRC-N Syndrome Detector block computes checksums for its entire input frame. This block has two output ports. The first output port contains the set of message words with the CRC bits removed. The second output port contains the checksum result, which is a vector of a size equal to the number of checksums. A value of 0 indicates no checksum errors. A value of 1 indicates a checksum error occurred.

The CRC-N Syndrome Detector block is a simplified version of the General CRC Syndrome Detector block. You can select the generator polynomial for the CRC algorithm from a list of commonly used polynomials, given in the CRC-N method field in the block's dialog. N is the degree of the generator polynomial. The reference page for the CRC-N Generator block contains a list of the options for the generator polynomial.

The parameter settings for the CRC-N Syndrome Detector block should match those of the CRC-N Generator block.

You specify the initial state of the internal shift register by the Initial states parameter. You specify the number of checksums that the block calculates for each input frame by the Checksums per frame parameter. For more detailed information, see the reference page for the General CRC Syndrome Detector block.

This block supports double and boolean data types. The output data type is inherited from the input.

\section*{Signal Attributes}

The CRC-N Syndrome Detector block has one input port and two output ports. All three ports accept binary column vector signals.

\section*{CRC-N Syndrome Detector}

Dialog
Box


\section*{CRC-N method}

The generator polynomial for the CRC algorithm.

\section*{Initial states}

A binary scalar or a binary row vector of length equal to the degree of the generator polynomial, specifying the initial state of the internal shift register.

\section*{Checksums per frame}

A positive integer specifying the number of checksums the block calculates for each input frame.

Algorithm
For a description of the CRC algorithm as implemented by this block, see "Cyclic Redundancy Check Codes" in Communications System Toolbox User's Guide.
References [1] Sklar, Bernard. Digital Communications: Fundamentals and Applications. Englewood Cliffs, N.J., Prentice-Hall, 1988.[2] Wicker, Stephen B., Error Control Systems for DigitalCommunication and Storage, Upper Saddle River, N.J., Prentice Hall,1995.
Pair Block CRC-N Generator
See Also General CRC Generator, General CRC Syndrome Detector

\section*{Data Mapper}
\begin{tabular}{ll} 
Purpose & Map integer symbols from one coding scheme to another \\
Library & Utility Blocks
\end{tabular}

Description The Data Mapper block accepts integer inputs and produces integer outputs. You can select one of four mapping modes: Binary to Gray, Gray to Binary, User Defined, or Straight Through.

This block accepts a scalar, column vector, or full matrix input signal. It can accept multichannel inputs and allows for input and output data types of double, single, int32, int16, int8, uint32, uint16, and uint8. The input signal must be a non-negative value. The block truncates non-integer input signals as integer values.

Gray coding is an ordering of binary numbers such that all adjacent numbers differ by only one bit. However, the inputs and outputs of this block are integers, not binary vectors. As a result, the first two mapping modes perform code conversions as follows:
- In the Binary to Gray mode, the output from this block is the integer equivalent of the Gray code bit representation for the input integer.
- In the Gray to Binary mode, the output from this block is the integer position of the binary equivalent of the input integer in a Gray code ordering.

As an example, the table below shows both the Binary to Gray and Gray to Binary mappings for integers in the range 0 to 7 . In the Binary to Gray Mode Output column, notice that binary representations in successive rows differ by exactly one bit. In the Gray to Binary Mode columns, notice that sorting the rows by Output value creates a Gray code ordering of Input binary representations.
\begin{tabular}{l|l|l|l}
\hline \multicolumn{2}{|l|}{ Binary to Gray Mode } & \multicolumn{2}{l}{ Gray to Binary Mode } \\
\hline Input & Output & Input & Output \\
\hline 0 & \(0(000)\) & \(0(000)\) & 0 \\
\hline 1 & \(1(001)\) & \(1(001)\) & 1 \\
\hline 2 & \(3(011)\) & \(2(010)\) & 3 \\
\hline 3 & \(2(010)\) & \(3(011)\) & 2 \\
\hline 4 & \(6(110)\) & \(4(100)\) & 7 \\
\hline 5 & \(7(111)\) & \(5(101)\) & 6 \\
\hline 6 & \(5(101)\) & \(6(110)\) & 4 \\
\hline 7 & \(4(100)\) & \(7(111)\) & 5 \\
\hline
\end{tabular}

When you select the User Defined mode, you can use any arbitrary mapping by providing a vector to specify the output ordering. For example, the vector [ \(1,5,0,4,2,3\) ] defines the following mapping:
\[
\begin{aligned}
& 0 \rightarrow 1 \\
& 1 \rightarrow 5 \\
& 2 \rightarrow 0 \\
& 3 \rightarrow 4 \\
& 4 \rightarrow 2 \\
& 5 \rightarrow 3
\end{aligned}
\]

When you select the Straight Through mode, the output equals the input.

\section*{Data Mapper}


Dialog Box

\section*{Mapping mode}

The type of data mapping that the block performs.

\section*{Symbol set size}

Symbol set size of M restricts this block's inputs and outputs to integers in the range 0 to \(\mathrm{M}-1\).

\section*{Mapping vector}

A vector of length M that contains the integers from 0 to \(\mathrm{M}-1\). The order of the elements of this vector specifies the mapping of inputs to outputs. This parameter appears only when you set Mapping mode to User Defined.

\section*{DBPSK Demodulator Baseband}

\section*{Purpose Demodulate DBPSK-modulated data}

\section*{Library \\ PM, in Digital Baseband sublibrary of Modulation}

Description
The DBPSK Demodulator Baseband block demodulates a signal that was modulated using the differential binary phase shift keying method. The input is a baseband representation of the modulated signal.

The input must be a discrete-time complex signal. The block compares the current symbol to the previous symbol. It maps phase differences of \(\theta\) and \(п+\theta\), respectively, to outputs of 0 and 1 , respectively, where \(\theta\) is the Phase rotation parameter. The first element of the block's output is the initial condition of zero because there is no previous symbol with which to compare the first symbol.

This block accepts a scalar or column vector input signal. The input signal can be of data types single and double. For information about the data types each block port supports, see "Supported Data Types" on page 2-177.

\section*{DBPSK Demodulator Baseband}


\section*{Phase rotation (rad)}

This phase difference between the current and previous modulated symbols results in an output of zero.

\section*{Output data type}

When the parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is of type single or double.

For additional information, see "Supported Data Types" on page 2-177.

\section*{DBPSK Demodulator Baseband}

Supported Data
Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
\hline Output & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
& \(\bullet\) Boolean \\
& \(\bullet 8-, 16\)-, and 32-bit signed integers \\
& \(\bullet 8-, 16\)-, and 32 -bit unsigned integers \\
\hline
\end{tabular}

\section*{Pair Block}

DBPSK Modulator Baseband

\author{
See Also
}

M-DPSK Demodulator Baseband, DQPSK Demodulator Baseband, BPSK Demodulator Baseband

\section*{DBPSK Modulator Baseband}

\section*{Purpose Modulate using differential binary phase shift keying method \\ Library \\ Description \\ ■ NOM DBPSK \\ PM, in Digital Baseband sublibrary of Modulation \\ The DBPSK Modulator Baseband block modulates using the differential binary phase shift keying method. The output is a baseband representation of the modulated signal. \\ This block accepts a scalar or column vector input signal. The input must be a discrete-time binary-valued signal. For information about the data types each block port supports, see "Supported Data Types" on page 2-179. \\ The following rules govern this modulation method when the Phase rotation parameter is \(\theta\) : \\ - If the first input bit is 0 or 1 , respectively, then the first modulated symbol is \(\exp (\mathrm{j} \theta)\) or \(-\exp (\mathrm{j} \theta)\), respectively. \\ - If a successive input bit is 0 or 1 , respectively, then the modulated symbol is the previous modulated symbol multiplied by \(\exp (\mathrm{j} \theta)\) or \(-\exp (\mathrm{j} \theta)\), respectively.}

\section*{DBPSK Modulator Baseband}

Dialog Box


\section*{Phase rotation (rad)}

The phase difference between the previous and current modulated symbols when the input is zero.

\section*{Output Data type}

The output data type can be either single or double. By default, the block sets this to double.

\section*{Supported Data \\ Types}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
& \(\bullet\) Boolean \\
& \(\bullet 8-, 16-\), and 32 -bit signed integers \\
& \(\bullet 8-, 16-\), and 32-bit unsigned integers \\
\hline Output & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
\hline
\end{tabular}

\section*{DBPSK Modulator Baseband}

\author{
Pair Block DBPSK Demodulator Baseband \\ See Also DQPSK Modulator Baseband, BPSK Modulator Baseband
}

\section*{Deinterlacer}

\section*{Purpose}

\section*{Library}

Description


\section*{Dialog Box}

\section*{Examples}

Distribute elements of input vector alternately between two output vectors

Sequence Operations
The Deinterlacer block accepts an even length column vector input signal. The block alternately places the elements in each of two output vectors. As a result, each output vector size is half the input vector size. The output vectors have the same complexity and sample time of the input.

This block accepts a column vector input signal with an even integer length. The block supports the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.
The Deinterlacer block can be useful for separating in-phase and quadrature information from a single vector into separate vectors.

\footnotetext{
Function Block Parameters: Deinterlacer
X
Deinterlacer (mask) (link)
Separate the elements of the input signal to generate the output signals. The oddnumbered elements of the input signal become the first output signal, while the even-numbered elements of the input signal become the second output signal.

This block accepts an even length column vector input signal.
}


If the input vector has the values \([1 ; 5 ; 2 ; 6 ; 3 ; 7 ; 4 ; 8\) ], then the two output vectors are \([1 ; 2 ; 3 ; 4]\) and \([5 ; 6 ; 7 ; 8]\). Notice that this example is the inverse of the example on the reference page for the Interlacer block.

\section*{Deinterlacer}

If the input vector has the values \([1 ; 2 ; 3 ; 4 ; 5 ; 6]\), then the two output vectors are \([1 ; 3 ; 5]\) and \([2 ; 4 ; 6]\).

\section*{Pair Block Interlacer}

See Also Demux (Simulink documentation)

\section*{Purpose \\ Library \\ Description \\ Derepeat \\ 15}

Reduce sampling rate by averaging consecutive samples

Sequence Operations
The Derepeat block resamples the discrete input at a rate \(1 / \mathrm{N}\) times the input sample rate by averaging N consecutive samples. This is one possible inverse of the DSP System Toolbox Repeat block. The positive integer N is the Derepeat factor parameter in the Derepeat dialog.

The Initial condition parameter prescribes elements of the output when it is still too early for the input data to show up in the output. If the dimensions of the Initial condition parameter match the output dimensions, then the parameter represents the initial output value. If Initial condition is a scalar, then it represents the initial value of each element in the output. The block does not support empty matrices for initial conditions.

The input can have any shape or frame status. The block accepts the data types single and double. The output signal inherits its data type from the input signal.

This block works within a triggered subsystem, as long as you use it in the single-rate mode.

\section*{Single-Rate Processing}

The block derepeats each frame, treating distinct channels independently. Each element of the output is the average of N consecutive elements along a column of the input matrix. The Derepeat factor must be less than the frame size.

When you set the Rate options parameter to Enforce single-rate processing, the input and output of the block have the same sample rate. The block reduces the sampling rate by using a proportionally smaller frame size than the input. Derepeat factor should be an integer factor of the number of rows in the input vector or matrix. For derepetition by a factor of N , the output frame size is \(1 / \mathrm{N}\) times the input frame size, but the input and output frame rates are equal. When you use this option, the Initial condition parameter does not apply

\section*{Derepeat}
and the block incurs no delay, because the input data immediately shows up in the output.

For example, if a single-channel input with 64 elements is derepeated by a factor of 4 , then the output contains 16 elements. The input and output frame periods are equal.

\section*{Multirate Processing}

When you set the Rate options parameter to Allow multirate processing, the input and output of the Derepeat block are the same size, but the sample rate of the output is \(N\) times slower than that of the input. When the block is in multirate processing mode, you must also specify a value for the Input processing parameter:
- When you set the Input processing parameter to Elements as channels (sample based), then the block assumes that the input is a vector or matrix whose elements represent samples from independent channels. The block averages samples from each channel independently over time. The output period is N times the input period, and the input and output sizes are identical. The output is delayed by one output period, and the first output value is the Initial condition value. If you set Rate options to Enforce single-rate processing, the block will generate an error message.
- When you set the Input processing parameter to Columns as channels (frame based), The block reduces the sampling rate by using a proportionally longer frame period at the output port than at the input port. For derepetition by a factor of N, the output frame period is N times the input frame period, but the input and output frame sizes are equal. The output is delayed by one output frame, and the first output frame is determined only by the Initial condition value. The block derepeats each frame, treating distinct channels independently. Each element of the output is the average of N consecutive elements along a column of the input matrix. The Derepeat factor must be less than the frame size.

\section*{Derepeat}

For example, if a single-channel input with a frame period of 1 second is derepeated by a factor of 4 , then the output has a frame period of 4 seconds. The input and output frame sizes are equal.
\begin{tabular}{l} 
19. Function Block Parameters: Derepeat \\
Derepeat (mask) (link) \\
Derepeat by an integer factor. The value of each output sample is the mean value \\
of N consecutive input samples. \\
Parameters \\
Derepeat factor, N : \\
\begin{tabular}{l} 
Input processing: Columns as channels (frame based) \\
Rate options: Allow multirate processing \\
Initial conditions: \\
\hline 0
\end{tabular} \\
\hline
\end{tabular}

\section*{Derepeat factor, \(\mathbf{N}\)}

The number of consecutive input samples to average in order to produce each output sample.

\section*{Input processing}

Specify how the block processes the input signal. You can set this parameter to one of the following options:

\section*{Derepeat}
- Columns as channels (frame based) - When you select this option, the block treats each column of the input as a separate channel.
- Elements as channels (sample based) - When you select this option, the block treats each element of the input as a separate channel.

Note The Inherited (this choice will be removed - see release notes) option will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

\section*{Rate options}

Select the rate processing option for the block.
- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width equals the product of the number of symbols and the Samples per symbol parameter value.
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output sample time equals the symbol period divided by the Samples per symbol parameter value.

> Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

\section*{Initial condition}

The value with which to initialize the block.
See Also Repeat (DSP System Toolbox documentation), Downsample (DSP System Toolbox documentation)

\section*{Descrambler}

\section*{Purpose Descramble input signal}

Library
Sequence Operations
Description The Descrambler block descrambles a scalar or column vector input signal. The Descrambler block is the inverse of the Scrambler block. If you use the Scrambler block in a transmitter, then you use the Descrambler block in the related receiver.
In the following descrambler schematic, the adders and subtracter operate modulo \(N\), where \(N\) is the Calculation base parameter. You must specify integer input values between 0 and \(\mathrm{N}-1\).


At each time step, the input causes the contents of the registers to shift sequentially. Using the Scramble polynomial parameter, you specify if each switch in the descrambler is on or off. To make the Descrambler block reverse the operation of the Scrambler block, use the same Scramble polynomial parameters in both blocks. If there is no signal delay between the scrambler and the descrambler, then the Initial states in the two blocks must be the same. See the reference page for the Scrambler block for more information about these parameters.

\section*{Descrambler}


\section*{Dialog Box}

\section*{Calculation base}

The calculation base \(N\). The input and output of this block are integers in the range [ \(0, N-1\) ].

\section*{Scramble polynomial}

A polynomial that defines the connections in the scrambler.

\section*{Initial states}

The states of the scrambler's registers when the simulation starts.
Pair Block Scrambler

\section*{Differential Decoder}

Purpose Decode binary signal using differential coding

\section*{Library}

Description
Source Coding
The Differential Decoder block decodes the binary input signal. The output is the logical difference between the consecutive input element within a channel. More specifically, the block's input and output are related by
\[
\begin{aligned}
& m\left(i_{0}\right)=d\left(i_{0}\right) \text { XOR Initial condition parameter value } \\
& m\left(i_{k}\right)=d\left(i_{k}\right) \operatorname{XOR} d\left(i_{k-1}\right)
\end{aligned}
\]
where
- \(d\) is the differentially encoded input.
- \(m\) is the output message.
- \(\mathrm{i}_{\mathrm{k}}\) is the kth element.
- XOR is the logical exclusive-or operator.

This block accepts a scalar, column vector, or matrix input signal and treats columns as channels.

\section*{Differential Decoder}


\section*{Initial conditions}

The logical exclusive-or of this value with the initial input value forms the initial output value.

Supported
Data Type Data Type
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline In & \(\bullet\) double \\
& \(\bullet\) single \\
& • boolean \\
& \(\bullet\) integer \\
& \(\bullet\) fixed-point \\
\hline Out & \(\bullet\) double \\
& \(\bullet\) single \\
& \(\bullet\) boolean
\end{tabular}

\section*{Differential Decoder}
\begin{tabular}{|c|c|c|}
\hline & Port & Supported Data Types \\
\hline References & [1] Couch, Leon W., II, Digital and Sixth edition, Upper Saddle River, & \begin{tabular}{l}
Andigteger \({ }^{2}\) munication Systems, \\
N. JJ fixederqderiftHall, 2001.
\end{tabular} \\
\hline Pair Block & Differential Encoder & \\
\hline
\end{tabular}

\section*{Purpose Encode binary signal using differential coding}

\section*{Library}

Source Coding
Description
The Differential Encoder block encodes the binary input signal within a channel. The output is the logical difference between the current input element and the previous output element. More specifically, the Encoder input and output are related by
\[
\begin{aligned}
& d\left(i_{0}\right)=m\left(i_{0}\right) \text { XOR Initial condition parameter value } \\
& d\left(i_{k}\right)=d\left(i_{k-1}\right) \operatorname{XOR} m\left(i_{k}\right)
\end{aligned}
\]
where
- \(m\) is the input message.
- d is the differentially encoded output.
- \(\mathrm{i}_{\mathrm{k}}\) is the kth element.
- XOR is the logical exclusive-or operator.

This block accepts a scalar or column vector input signal and treats columns as channels.

\section*{Differential Encoder}
\begin{tabular}{|c|c|c|c|c|c|}
\hline \multirow[t]{3}{*}{} & \multicolumn{5}{|l|}{國 Function Block Parameters: Differential Encoder X} \\
\hline & \multicolumn{5}{|l|}{\begin{tabular}{l}
Differential Encoder (mask) (link) \\
Differentially encode the input data. This block treats columns as channels. \\
The output of this block is the logical difference between the current input element and the previous output element.
\end{tabular}} \\
\hline & \multicolumn{5}{|l|}{\begin{tabular}{l}
Parameters \\
Initial conditions:
\[
0
\]
\end{tabular}} \\
\hline Dialog & OK & Cancel & Help & Apply & \\
\hline & \multicolumn{5}{|l|}{\begin{tabular}{l}
Initial conditions \\
The logical exclusive-or of this value with the initial input value forms the initial output value.
\end{tabular}} \\
\hline \multirow[t]{3}{*}{Supported Data Type} & Port & \multicolumn{4}{|c|}{Supported Data Types} \\
\hline & In & \multicolumn{4}{|r|}{\begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- Integer \\
- Fixed-point
\end{tabular}} \\
\hline & Out & \multicolumn{4}{|l|}{\begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean
\end{tabular}} \\
\hline
\end{tabular}
\begin{tabular}{|c|c|}
\hline & Port \({ }^{\text {P }}\) ( Supported Data Types \\
\hline References & [1] Couch, Leon W., II, Digital and Analytegermmunication Systems, Sixth edition, Upper Saddle River, N.oJ.FiReentigenall, 2001. \\
\hline Pair Block & Differential Decoder \\
\hline
\end{tabular}

\section*{Discrete-Time Eye Diagram Scope}

\section*{Purpose \\ Library}

Display multiple traces of modulated signal

Comm Sinks
Description The Discrete-Time Eye Diagram Scope block displays multiple traces of a modulated signal to produce an eye diagram. You can use the block to reveal the modulation characteristics of the signal, such as pulse shaping or channel distortions.

The Discrete-Time Eye Diagram Scope block has one input port. This block accepts a scalar-valued or column vector input signal. The block accepts a signal with the following data types: double, single, boolean, base integer, and fixed-point data types for input, but casts as double prior to displaying the results.

\section*{Marker and Line Styles}

The Marker, Line style, and Line color parameters, on the Rendering Properties panel, control the appearance of the signal trajectory. The Marker parameter specifies the marker style for points in the eye diagram. The following table lists some of the available line markers.
\begin{tabular}{l|l|ll}
\hline \begin{tabular}{l} 
Marker \\
Style
\end{tabular} & \begin{tabular}{l} 
Parameter \\
Symbol
\end{tabular} & Appearance
\end{tabular}

The Line style parameter specifies the style for lines in the eye diagram. The following lists some of the available line styles.

\section*{Discrete-Time Eye Diagram Scope}
\begin{tabular}{|c|c|}
\hline Line Style & Appearance \\
\hline Solid & —— \\
\hline Dashed & ------- \\
\hline Dotted & \\
\hline Dash-dot & -.-------.-.-.- \\
\hline
\end{tabular}

The Line color parameter specifies the color of the eye diagram. These settings plot the signal channels in the following colors (8-bit RGB equivalents are shown in the center column).
\begin{tabular}{l|l|l}
\hline Color & \begin{tabular}{l} 
RGB \\
Equivalent
\end{tabular} & Appearance \\
\hline Black & \((0,0,0)\) & - \\
\hline Blue & \((0,0,255)\) & - \\
\hline Red & \((255,0,0)\) & - \\
\hline Green & \((0,255,0)\) & - \\
\hline \begin{tabular}{l} 
Dark \\
purple
\end{tabular} & \((192,0,192)\) & - \\
\hline
\end{tabular}

See the line function in the MATLAB documentation for more information about the available markers, colors, and line styles.

\section*{Recommended Settings}

The following table summarizes the recommended parameter settings for the Discrete-Time Eye Diagram Scope.

\section*{Discrete-Time Eye Diagram Scope}
\begin{tabular}{l|l}
\hline Parameter & Recommended Setting \\
\hline Samples per symbol & \begin{tabular}{l} 
Same as the Samples per \\
symbol setting in the modulator \\
block, or the Interpolation \\
factor setting in the interpolation \\
block
\end{tabular} \\
\hline Offset (samples) & \begin{tabular}{l}
0 to view the open part of the eye \\
(Samples per symbol)/2 to view \\
the closed part of the eye
\end{tabular} \\
\hline Symbols per trace & An integer between 1 and 4 \\
\hline Traces displayed & \begin{tabular}{l}
10 times the alphabet size of the \\
modulator, M
\end{tabular} \\
\hline New traces per display & \begin{tabular}{l} 
Same as Traces displayed for \\
greater speed A small positive \\
integer for best animation
\end{tabular} \\
\hline Marker & \begin{tabular}{l} 
None or a point (.) to see where \\
the samples are plotted
\end{tabular} \\
\hline Line style & Solid dash (-) \\
\hline Line color & Blue (b) \\
\hline Duplicate points at trace & \begin{tabular}{l} 
Check Duplicate points at \\
trace boundary for modulations \\
such as PSK and QAM. \\
boundary
\end{tabular} \\
\hline \begin{tabular}{l} 
Clear to display the phase trees \\
for MSK, CPFSK, GFSK, GMSK, \\
and other continuous phase \\
modulations.
\end{tabular} \\
\hline Color fading & \begin{tabular}{l} 
Check Color fading for \\
animation that resembles an \\
oscilloscope. \\
Clear for greater speed and \\
animation that resembles a plot.
\end{tabular} \\
\hline
\end{tabular}

\section*{Discrete-Time Eye Diagram Scope}
\begin{tabular}{l|l}
\hline Parameter & Recommended Setting \\
\hline High quality rendering & \begin{tabular}{l} 
Check High quality rendering \\
for better animation. \\
Clear for greater speed.
\end{tabular} \\
\hline Eye diagram to display & \begin{tabular}{l} 
Select In-phase and \\
Quadrature to view real and \\
imaginary components. \\
Select In-phase Only to view \\
real component only and for \\
greater speed. \\
When the input is real and \\
you choose In-phase and \\
Quadrature, the quadrature \\
component of the eye diagram is \\
zero.
\end{tabular} \\
\hline Open at start of simulation & \begin{tabular}{l} 
Check Open at start of \\
simulation to view the signal at \\
the start of simulation.
\end{tabular} \\
\hline Y-axis minimum & \begin{tabular}{l} 
Clear to view the signal after \\
convergence to steady state and \\
for greater initial speed.
\end{tabular} \\
\hline Y-axis maximum & \begin{tabular}{l} 
Approximately 10\% less than the \\
expected minimum value of the \\
signal
\end{tabular} \\
\hline & \begin{tabular}{l} 
Approximately 10\% greater than \\
the expected maximum value of \\
the signal
\end{tabular} \\
\hline
\end{tabular}

For Rapid Accelerator or External mode, set the scope up for single rate mode. To guarantee the satisfactory behavior of single rate mode, the subsystem below the block mask for this block must operate as a single-rate entity, which means the following conditions are true:

\section*{Discrete-Time Eye Diagram Scope}
\[
\mathrm{sps}^{*}((\mathrm{td} *(\mathrm{spt}-1))+\mathrm{ntpd})=\mathrm{Sf}
\]
where:
- \(\mathrm{sps}=\) Samples per symbol
- td \(=\) Traces displayed
- \(\mathrm{spt}=\) Symbols per trace
- \(n t p d=\) New traces per display
- \(\mathrm{Sf}=\) Input frame size, in samples

This equation guarantees that the subsystem below the mask for this block operates as a single rate entity.

\section*{Warning}

If you want to use Rapid Accelerator or External mode, set this block up to run as a single rate entity because the block does not support multi-rate in these modes.

Note Before running a model that contains a Discrete-Time Eye Diagram Scope block in Accelerator, Rapid Accelerator, or External mode, you must select Open scope at start of simulation. If you do not select this check box before running your model for the first time, the scope will not display your simulation data

\section*{Scope Options}

The scope title (in the window title bar) is the same as the block title. You can set the axis scaling by setting the \(y\)-axis minimum and \(y\)-axis maximum parameters on the Axes Properties panel.
In addition to the standard MATLAB figure window menus (File, Edit, Window, Help), the Vector Scope window has an Axes and a Channels menu.

\section*{Discrete-Time Eye Diagram Scope}

The properties listed in the Axes menu apply to all channels. Many of the parameters in this menu are also accessible through the block parameter dialog box. These are Autoscale, Show grid, Frame \#, and Save Position. Below are descriptions of the other parameters listed in the Axes menu:
- Autoscale resizes the \(y\)-axis to best fit the vertical range of the data. The numerical limits selected by the autoscale feature are displayed in the Minimum Y-limit and Maximum Y-limit parameters in the parameter dialog box. You can change them by editing those values.
- Show grid - When selected, the scope displays a grid according to tick marks on the \(x\) - and \(y\)-axes.
- Frame \# - When selected, the scope displays the current frame number at the bottom of the scope window.
- Save Position automatically updates the Scope position parameter in the Figure properties panel to reflect the scope window's current position and size. To make the scope window open at a particular location on the screen when the simulation runs, simply drag the window to the desired location, resize it as needed, and select Save Position.

The properties listed in the Channels menu apply to a particular channel. The parameters listed in this menu are Style, Marker, and Color. They correspond to the parameters Line style, Marker, and Line color, respectively.

You can also access many of these options by right-clicking with the mouse anywhere on the scope display. The menu that pops up contains a combination of the options available in both the Axes and Channels menus.

\section*{Behavior in Enabled Subsystems}

You can use the Discrete-Time Eye Diagram Scope block inside an enabled subsystem. However, you cannot use the scope block inside an enabled subsystem when the model is in a multirate multitasking environment.

\section*{Discrete-Time Eye Diagram Scope}

When you use the scope in a multirate singletasking environment, it may generate unexpected results inside enabled subsystems. To workaround this issue, configure the scope for single-rate mode. See "Recommended Settings" on page 2-197 for the parameter settings that enable single-rate mode.


\section*{Samples per symbol}

Number of samples per symbol. Use with Symbols per trace to determine the number of samples per trace.

\section*{Discrete-Time Eye Diagram Scope}

\section*{Offset (samples)}

Nonnegative integer less than the product of Samples per symbol and Symbols per trace, specifying the number of samples to omit before plotting the first point. Tunable.

\section*{Symbols per trace}

Positive integer specifying the number of symbols plotted per trace.

\section*{Traces displayed}

Number of traces plotted.
New traces per display
Positive integer less than Traces displayed, specifying the number of new traces that appear in each display.
\begin{tabular}{|c|c|c|c|}
\hline Plotting Properties & Rendering Properties & Axes Properties & Figure Properties \\
\hline \multicolumn{4}{|l|}{Markers:} \\
\hline \multicolumn{4}{|l|}{Line style:} \\
\hline \multicolumn{4}{|l|}{-} \\
\hline \multicolumn{4}{|l|}{Line color:} \\
\hline \(b\) & & & \\
\hline \begin{tabular}{l}
Duplicate points \\
Color fading \\
High quality rend \\
Show grid
\end{tabular} & \begin{tabular}{l}
trace boundary \\
ing
\end{tabular} & & \\
\hline
\end{tabular}

\section*{Markers}

The marker for points in the eye diagram. Tunable.

\section*{Line style}

The line style in the eye diagram. Tunable.
Line color
The line color in the eye diagram. Tunable.

\section*{Discrete-Time Eye Diagram Scope}

\section*{Duplicate points at trace boundary}

Check to enable duplicate points at the trace boundary. Clear to disable.

\section*{Color fading}

When selected, the points in the eye diagram fade as the interval of time after they are first plotted increases. Tunable.

\section*{High quality rendering}

When selected, the block renders a slow, higher-quality picture with overwrite raster operations. When cleared, the block renders a fast, lower-quality picture with XOR raster operations. Tunable.

\section*{Show grid}

Toggles the scope grid on and off. Tunable.
\begin{tabular}{|l|l|l|}
\hline Ploting Properties & Rendering Properties & Axes Properties: \\
\hline Y axis minimum: & Figure Properties \\
\hline-1.5 & \\
\hline Y axis maximum: & \\
\hline 1.5 & \\
\hline In-phase \(Y\)-axis label: & \\
\hline In-phase Amplitude & \\
Quadrature \(Y\)-axis label: \\
\hline Quadrature Amplitude \\
\hline
\end{tabular}

\section*{Y-axis minimum}

Minimum signal value the scope displays. Tunable.

\section*{Y-axis maximum}

Maximum signal value the scope displays. Tunable.

\section*{In-phase Y-axis label}

Label for \(y\)-axis of the in-phase diagram. Tunable.

\section*{Discrete-Time Eye Diagram Scope}

\section*{Quadrature Y-axis label}

Label for \(y\)-axis of the quadrature diagram. Tunable.
\begin{tabular}{|c|c|c|c|}
\hline Plotting Properties & Rendering Properties & Axes Properties & Figure Properties \\
\hline \multicolumn{4}{|l|}{\(\sqrt{\checkmark}\) Open scope at start of simulation} \\
\hline \multicolumn{3}{|l|}{Eye diagram to display: \(1 n\)-phase and Quadrature} & \(\checkmark\) \\
\hline \multicolumn{4}{|l|}{\(\square\) Trace number} \\
\hline \multicolumn{4}{|l|}{Scope position:} \\
\hline \multicolumn{4}{|l|}{get(0,'defaultigureposition');} \\
\hline \multicolumn{4}{|l|}{Title:} \\
\hline Eye Diagram & & & \\
\hline
\end{tabular}

\section*{Open scope at start of simulation}

When selected, the scope opens at the start of simulation.
When cleared, you must double-click the block after the start of simulation to open the scope. Tunable.

Note Before running a model that contains a Discrete-Time Eye Diagram Scope block in Accelerator, Rapid Accelerator, or External mode, you must select Open scope at start of simulation. If you do not select this check box before running your model for the first time, the scope will not display your simulation data

\section*{Eye diagram to display}

Type of eye diagram to display. Choose In-phase and
Quadrature to display real and complex components, or In-phase Only to display only the real component. Tunable.

\section*{Discrete-Time Eye Diagram Scope}

\section*{Trace number}

Displays the number of the current trace in the input sequenced. Tunable.

\section*{Scope position}

A four-element vector of the form [left bottom width height] specifying the position of the scope window. \((0,0)\) is the lower left corner of the display. Tunable.

\section*{Title}

Title of eye diagram figure window. Tunable.

\section*{Examples \\ For documentation examples that use this block, see "View a Sinusoid" and "View a Modulated Signal". \\ Also, the following Communications System Toolbox demos illustrate how to use the Discrete-Time Eye Diagram Scope block:}
- CPM Phase Tree Example
- Filtered Offset QPSK vs. Filtered QPSK
- Rayleigh Fading Channel
- QPSK vs. MSK

See Also Discrete-Time Scatter Plot Scope, Discrete-Time Signal Trajectory Scope

\section*{Discrete-Time Scatter Plot Scope}

\section*{Purpose}

\section*{Library}

Description

Display in-phase and quadrature components of modulated signal constellation

Comm Sinks
The Discrete-Time Scatter Plot Scope block displays scatter plots of a modulated signal, to reveal the modulation characteristics, such as pulse shaping or channel distortions of the signal.

The Discrete-Time Scatter Plot Scope block has one input port. This block accepts a complex scalar-valued or column vector input signal. The block accepts a signal with the following data types: double, single, base integer, and fixed-point for input, but will cast it as double.

See the reference page for the Discrete-Time Signal Trajectory Scope block to compare the preceding scatter plot with the trajectory of the same signal. The Discrete-Time Signal Trajectory Scope block connects the points displayed by the Discrete-Time Scatter Plot Scope block to display the signal trajectory.

Setting Samples per symbol to 8, increasing Points displayed to 100 , and running the model for 100 seconds produces the following scatter plot.

\section*{Discrete-Time Scatter Plot Scope}


\section*{Markers and Color}

The Markers and Color parameters, on the Rendering Properties panel, specify the style and color of markers in the scatter plot. For details on the options for these parameters, see the reference page for the Discrete-Time Eye Diagram Scope block.

\section*{Recommended Settings}

The following table summarizes the recommended parameter settings for the Discrete-Time Scatter Plot Scope.

\section*{Discrete-Time Scatter Plot Scope}
\begin{tabular}{l|l}
\hline Parameter & Recommended Setting \\
\hline Samples per symbol & \begin{tabular}{l} 
Same as the Samples per \\
symbol setting in the modulator \\
block, or the Interpolation \\
factor setting in the interpolation \\
block
\end{tabular} \\
\hline Points displayed & \begin{tabular}{l}
10 times the alphabet size of the \\
modulator
\end{tabular} \\
\hline New points per display & \begin{tabular}{l} 
Same as Points displayed for \\
greater speed \\
A small positive integer for best \\
animation
\end{tabular} \\
\hline Line style & Solid dash (-) \\
\hline Line color & Blue (b) \\
\hline Color fading & \begin{tabular}{l} 
Check Color fading for \\
animation that resembles an \\
oscilloscope. \\
Clear for greater speed and \\
animation that resembles a plot.
\end{tabular} \\
\hline High quality rendering & \begin{tabular}{l} 
Check High quality rendering \\
for higher quality rendering. \\
Clear for greater speed.
\end{tabular} \\
\hline Open at start of simulation & \begin{tabular}{l} 
Check Open at start of \\
simulation to view the signal at \\
the start of simulation. \\
Clear to view the signal after \\
convergence to steady state and \\
for greater initial speed.
\end{tabular} \\
\hline
\end{tabular}

\section*{Discrete-Time Scatter Plot Scope}
\begin{tabular}{l|l}
\hline Parameter & Recommended Setting \\
\hline X-axis minimum & \begin{tabular}{l} 
Approximately \(10 \%\) less than the \\
expected minimum value of the \\
signal
\end{tabular} \\
\hline X-axis maximum & \begin{tabular}{l} 
Approximately \(10 \%\) greater than \\
the expected maximum value of \\
the signal
\end{tabular} \\
\hline
\end{tabular}

For Rapid Accelerator or External mode, set the scope up for single rate mode. To guarantee the satisfactory behavior of single rate mode, the subsystem below the block mask for this block must operate as a single-rate entity, which means the following conditions are true:
\[
\mathrm{sps} * \mathrm{nppd}=\mathrm{Sf}
\]
where:
- sps \(=\) Samples per symbol
- nppd = New points per display
- \(\mathrm{Sf}=\) Input frame size, in samples

This equation guarantees that the subsystem below the mask for this block operates as a single rate entity.

\section*{Warning}

If you want to use Rapid Accelerator or External mode, set this block up to run as a single rate entity because multi-rate does not support these modes.

\section*{Discrete-Time Scatter Plot Scope}

> Note Before running a model that contains a Discrete-Time Scatter Plot Scope block in Accelerator, Rapid Accelerator, or External mode, you must select Open scope at start of simulation. If you do not select this check box before running your model for the first time, the scope will not display your simulation data

\section*{Behavior in Enabled Subsystems}

You can use the Discrete-Time Scatter Plot Scope block inside an enabled subsystem. However, you cannot use the scope block inside an enabled subsystem when the model is in a multirate multitasking environment.

When you use the scope in a multirate singletasking environment, it may generate unexpected results inside enabled subsystems. To workaround this issue, configure the scope for single-rate mode. See "Recommended Settings" on page 2-208 for the parameter settings that enable single-rate mode.

\section*{Discrete-Time Scatter Plot Scope}


\section*{Samples per symbol}

Number of samples per symbol.

\section*{Offset (samples)}

Nonnegative integer less than the number of samples per symbol, specifying the number of samples to skip before plotting points.

\section*{Discrete-Time Scatter Plot Scope}

\section*{Points displayed}

Total number of points plotted.

\section*{New points per display}

Number of new points that appear in each display.
\begin{tabular}{|c|c|c|c|}
\hline Plotting Properties & Rendering Properties & Axes Properties & Figure Properties \\
\hline \multicolumn{4}{|l|}{Markers:} \\
\hline 1. & & & \\
\hline \multicolumn{4}{|l|}{Line color:} \\
\hline \multicolumn{4}{|l|}{b} \\
\hline \begin{tabular}{l}
Color fading \\
High quality ren \\
Show grid
\end{tabular} & & & \\
\hline
\end{tabular}

\section*{Markers}

Line markers used in the scatter plot. Tunable.

\section*{Line color}

The line color used in the scatter plot. Tunable.

\section*{Color fading}

When selected, the points in the scatter plot fade as the interval of time after they are first plotted increases. Tunable.

\section*{High quality rendering}

When selected, the block renders a slow, higher-quality picture with overwrite raster operations. When cleared, the block renders a fast, lower-quality picture with XOR raster operations. Tunable.

\section*{Show grid}

Toggles the scope grid on and off. Tunable.

\section*{Discrete-Time Scatter Plot Scope}
\begin{tabular}{|l|l|l|}
\hline Ploting Properties & Rendering Properties & Axes Properties \\
\hline X-axis minimum: & Figure Properties \\
\hline-1.5 & \\
\hline\(X\)-axis maximum: & \\
\hline 1.5 & \\
\hline Y-axis minimum: & \\
\hline-1.5 & \\
\(Y\) Yaxis maximum: & \\
\hline 1.5 & \\
In-phase \(X\)-axis label: \\
\hline In-phase Amplitude \\
Quadrature \(Y\)-axis label: \\
\hline Quadrature Amplitude \\
\hline
\end{tabular}

\section*{X-axis minimum}

Minimum value the scope displays on the \(x\)-axis. Tunable.

\section*{X -axis maximum}

Maximum value the scope displays on the \(x\)-axis. Tunable.

\section*{Y-axis minimum}

Minimum signal value the scope displays on the \(y\)-axis. Tunable.

\section*{Y -axis maximum}

Maximum signal value the scope displays on the \(y\)-axis. Tunable.

\section*{In-phase X-axis label}

Label for \(x\)-axis. Tunable.

\section*{Quadrature Y-axis label}

Label for \(y\)-axis. Tunable.
\begin{tabular}{|l|l|l|}
\hline Plotting Properties & Rendering Properties & Axes Properties \\
\hline Figure Properties: \\
\hline I Open scope at start of simulation \\
\hline Point number \\
Scope position: \\
\hline get(0,'defaultigureposition'); \\
Title: \\
\hline Scatter Plot & \\
\hline
\end{tabular}

\section*{Open at start of simulation}

When selected, the scope opens at the start of simulation. When cleared, you must double-click the block after the start of simulation to open the scope.

Note Before running a model that contains a Discrete-Time Scatter Plot Scope block in Accelerator, Rapid Accelerator, or External mode, you must select Open scope at start of simulation. If you do not select this check box before running your model for the first time, the scope will not display your simulation data

\section*{Point number}

Displays the number of the current point in the input sequence. Tunable.

\section*{Scope position}

A four-element vector of the form [left bottom width height] specifying the position of the scope window. \((0,0)\) is the lower left corner of the display. Tunable.

\section*{Discrete-Time Scatter Plot Scope}

\section*{Title}

Title of scatter plot. Tunable.

\author{
Examples For documentation examples that use this block, see "View a Sinusoid" and "View a Modulated Signal". \\ The following demos in Communications System Toolbox software illustrate how to use the Discrete-Time Scatter Plot Scope block: \\ - Digital Video Broadcasting Model - Terrestrial \\ - HiperLAN/2 \\ - Phase Noise Effects in 256 QAM \\ - Multipath Rayleigh Fading Channel \\ See Also Discrete-Time Eye Diagram Scope, Discrete-Time Signal Trajectory Scope, Real-Imag to Complex
}

\section*{Discrete-Time Signal Trajectory Scope}

\section*{Purpose}

\section*{Library}

Description

Plot modulated signal's in-phase component versus its quadrature component

Comm Sinks
The Discrete-Time Signal Trajectory Scope displays the trajectory of a modulated signal in its signal space by plotting its in-phase component versus its quadrature component.

The Discrete-Time Signal Trajectory Scope block has one input port. This block accepts a complex scalar-valued or column vector input signal. The block accepts a signal with the following data types: double, single, base integer, and fixed-point for input, but will cast it as double.

\section*{Line Style and Color}

The Line style and Line color parameters on the Rendering Properties panel control the appearance of the signal trajectory. The Line style parameter specifies the style for lines in the signal trajectory. For details on the options for these parameters, see the reference page for the Discrete-Time Eye Diagram Scope block.

\section*{Recommended Settings}

The following table summarizes the recommended parameter settings for the Discrete-Time Signal Trajectory Scope.
\begin{tabular}{l|l}
\hline Parameter & Recommended Setting \\
\hline Samples per symbol & \begin{tabular}{l} 
Same as the Samples per \\
symbol setting in the modulator \\
block, or the Interpolation \\
factor used in the interpolation \\
block
\end{tabular} \\
\hline Symbols displayed & \begin{tabular}{l}
10 times the alphabet size of the \\
modulator, M
\end{tabular} \\
\hline
\end{tabular}

\section*{Discrete-Time Signal Trajectory Scope}
\(\left.\begin{array}{l|l}\hline \text { Parameter } & \text { Recommended Setting } \\
\hline \text { New symbols per display } & \begin{array}{l}\text { Same as Symbols displayed for } \\
\text { greater speed } \\
\text { A small positive integer for best } \\
\text { animation }\end{array} \\
\hline \text { Line style } & \text { Solid dash (-) } \\
\hline \text { Line color } & \text { Blue (b) } \\
\hline \text { Color fading } & \begin{array}{l}\text { Check Color fading for } \\
\text { animation that resembles an } \\
\text { oscilloscope. } \\
\text { Clear for greater speed and } \\
\text { animation that resembles a plot. }\end{array} \\
\hline \text { High quality rendering } & \begin{array}{l}\text { Check High quality rendering } \\
\text { for higher quality rendering. } \\
\text { Clear for greater speed. }\end{array} \\
\hline \text { Open at start of simulation } & \begin{array}{l}\text { Check Open at start of } \\
\text { simulation to view the signal at } \\
\text { the start of simulation. } \\
\text { Clear to view the signal after }\end{array} \\
\text { convergence to steady state and } \\
\text { for greater initial speed. }\end{array}\right\}\)\begin{tabular}{l} 
Approximately 10\% less than the \\
expected minimum value of the \\
signal
\end{tabular}

For Rapid Accelerator or External mode, set the scope up for single rate mode. To guarantee the satisfactory behavior of single rate mode,

\section*{Discrete-Time Signal Trajectory Scope}
the subsystem below the block mask for this block must operate as a single-rate entity, which means the following conditions are true:
\[
\mathrm{sps} * \mathrm{nspd}=\mathrm{Sf}
\]
where:
- \(\mathrm{sps}=\) Samples per symbol
- \(n s p d=\) New symbols per display
- \(\mathrm{Sf}=\) Input frame size, in samples

This equation guarantees that the subsystem below the mask for this block operates as a single-rate entity.

\section*{Warning}

If you want to use Rapid Accelerator or External mode, set this block up to run as a single rate entity because the block does not support multi-rate in these modes.

Note Before running a model that contains a Discrete-Time Signal Trajectory Scope block in Accelerator, Rapid Accelerator, or External mode, you must select Open scope at start of simulation. If you do not select this check box before running your model for the first time, the scope will not display your simulation data

\section*{Behavior in Enabled Subsystems}

You can use the Discrete-Time Signal Trajectory Scope block inside an enabled subsystem. However, you cannot use the scope block inside an enabled subsystem when the model is in a multirate multitasking environment.

When you use the scope in a multirate singletasking environment, it may generate unexpected results inside enabled subsystems. To workaround this issue, configure the scope for single-rate mode. See

\section*{Discrete-Time Signal Trajectory Scope}
"Recommended Settings" on page 2-217 for the parameter settings that enable single-rate mode.
 Box

\section*{Samples per symbol}

Number of samples per symbol.

\section*{Symbols displayed}

Total number of symbols plotted.

\section*{Discrete-Time Signal Trajectory Scope}

\section*{New symbols per display}

Number of new symbols that appear in each display.
\begin{tabular}{|c|c|c|c|}
\hline Plotting Properties & Rendering Properties & Axes Properties & Figure Properties \\
\hline \multicolumn{4}{|l|}{Line style:} \\
\hline \multicolumn{4}{|l|}{-} \\
\hline \multicolumn{4}{|l|}{Line color:} \\
\hline \multicolumn{4}{|l|}{\(b\)} \\
\hline \begin{tabular}{l}
Color fading \\
High quality ren \\
Show grid
\end{tabular} & & & \\
\hline
\end{tabular}

\section*{Line markers}

The line markers used in the signal trajectory. Tunable.

\section*{Line color}

The line color used in the signal trajectory. Tunable.

\section*{Color fading}

When selected, the points in the signal trajectory fade as the interval of time after they are first plotted increases. Tunable.

\section*{High quality rendering}

When selected, the block renders a slow, higher-quality picture with overwrite raster operations. When cleared, the block renders a fast, lower-quality picture with XOR raster operations. Tunable.

\section*{Show grid}

Toggles the scope grid on and off. Tunable.

\section*{Discrete-Time Signal Trajectory Scope}
\begin{tabular}{|l|l|l|l|}
\hline Plotting Properties & Rendering Properties & Axes Properties & Figure Properties \\
\hline X-axis minimum: & & \\
\hline-1.5 & & \\
\hline X-axis maximum: & \\
\hline 1.5 & \\
\hline\(Y\)-axis minimum: & \\
\hline-1.5 & \\
\hline Y axis maximum: & \\
\hline 1.5 \\
In-phase X-axis label: & \\
\hline In-phase Amplitude & \\
\hline Quadrature \(Y\)-axis label: & \\
\hline Quadrature Amplitude & \\
\hline
\end{tabular}

\section*{X -axis minimum}

Minimum value the scope displays on the x-axis. Tunable.

\section*{X -axis maximum}

Maximum value the scope displays on the \(x\)-axis. Tunable.

\section*{Y-axis minimum}

Minimum signal value the scope displays on the \(y\)-axis. Tunable.

\section*{Y-axis maximum}

Maximum signal value the scope display on the \(y\)-axis. Tunable.

\section*{In-phase X-axis label}

Label for \(x\)-axis. Tunable.

\section*{Quadrature Y-axis label}

Label for \(y\)-axis. Tunable.

\section*{Discrete-Time Signal Trajectory Scope}
\begin{tabular}{|l|l|l|}
\hline Plotting Properties & Rendering Properties & Axes Properties \\
\hline Figure Properties \\
\hline I Open scope at start of simulation & \\
\hline Symbol number & \\
\hline Scope position: \\
\hline get(0,'defaultigureposition'): \\
\hline Title: \\
\hline Signal Trajectory & \\
\hline \\
\\
\hline
\end{tabular}

\section*{Open at start of simulation}

When selected, the scope opens at the start of simulation. When cleared, you must double-click the block after the start of simulation to open the scope. Tunable

Note Before running a model that contains a Discrete-Time Signal Trajectory Scope block in Accelerator, Rapid Accelerator, or External mode, you must select Open scope at start of simulation. If you do not select this check box before running your model for the first time, the scope will not display your simulation data

\section*{Symbol number}

Displays the number of the current symbol in the input sequence. Tunable.

\section*{Discrete-Time Signal Trajectory Scope}

\section*{Scope position}

A four-element vector of the form [left bottom width height] specifying the position of the scope window. \((0,0)\) is the lower left corner of the display. Tunable.

\section*{Title}

Title of signal trajectory plot. Tunable.

\section*{Examples}

For documentation examples that use this block, see "View a Sinusoid" and "View a Modulated Signal".

Also, the following demos in Communications System Toolbox software illustrate how to use the Discrete-Time Signal Trajectory Scope:
- Filtered Offset QPSK vs. Filtered QPSK
- GMSK vs. MSK

See Also Discrete-Time Eye Diagram Scope, Discrete-Time Scatter Plot Scope

\section*{Discrete-Time VCO}

\section*{Purpose}

Implement voltage-controlled oscillator in discrete time

\section*{Library}

Description

Discrete-Time
vco
Components sublibrary of Synchronization
The Discrete-Time VCO (voltage-controlled oscillator) block generates a signal whose frequency shift from the Quiescent frequency parameter is proportional to the input signal. The input signal is interpreted as a
voltage. If the input signal is \(u(t)\), then the output signal is
\[
y(t)=A_{c} \cos \left(2 \pi f_{c} t+2 \pi k_{c} \int_{0}^{t} u(\tau) d \tau+\varphi\right)
\]
where \(A_{\mathrm{c}}\) is the Output amplitude, \(f_{\mathrm{c}}\) is the Quiescent frequency, \(k_{\mathrm{c}}\) is the Input sensitivity, and \(\varphi\) is the Initial phase

This block uses a discrete-time integrator to interpret the equation above.

This block accepts a scalar-valued input signal with a data type of single or double. The output signal inherits its data type from the input signal. The block supports double precision only for code generation.

\section*{Discrete-Time VCO}


Dialog
Box

\section*{Output amplitude}

The amplitude of the output.

\section*{Quiescent frequency ( Hz )}

The frequency of the oscillator output when the input signal is zero.

\section*{Input sensitivity}

This value scales the input voltage and, consequently, the shift from the Quiescent frequency value. The units of Input sensitivity are Hertz per volt.

\section*{Initial phase (rad)}

The initial phase of the oscillator in radians.

\section*{Sample time}

The calculation sample time.
See Also Continuous-Time VCO

\section*{DQPSK Demodulator Baseband}

\author{
Purpose Demodulate DQPSK-modulated data \\ Library \\ PM, in Digital Baseband sublibrary of Modulation
}

Description The DQPSK Demodulator Baseband block demodulates a signal that was modulated using the differential quaternary phase shift keying method. The input is a baseband representation of the modulated signal.

The input must be a discrete-time complex signal. The output depends on the phase difference between the current symbol and the previous symbol. The first integer (or binary pair, if you set the Output type parameter to Bit) at the block output is the initial condition of zero because there is no previous symbol.

This block accepts either a scalar or column vector input signal. For information about the data types each block port supports, see "Supported Data Types" on page 2-230.

\section*{Outputs and Constellation Types}

When you set Output type parameter to Integer, the block maps a phase difference of
\[
\theta+\pi m / 2
\]
to \(m\), where \(\theta\) represents the Phase rotation parameter and \(m\) is 0,1 , 2 , or 3 .

When you set the Output type parameter to Bit, then the output contains pairs of binary values. The reference page for the DQPSK Modulator Baseband block shows which phase differences map to each binary pair, for the cases when the Constellation ordering parameter is either Binary or Gray.

\section*{DQPSK Demodulator Baseband}

\section*{Dialog Box}


\section*{Output type}

Determines whether the output consists of integers or pairs of bits.

\section*{Constellation ordering}

Determines how the block maps each integer to a pair of output bits.

\section*{Phase rotation (rad)}

This phase difference between the current and previous modulated symbols results in an output of zero.

\section*{Output data type}

When the parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is of type single or double.

For integer outputs, this block can output the data types int8, uint8, int16, uint16, int32, uint32, single, and double. For bit outputs, output can be int8, uint8, int16, uint16, int32, uint32, boolean, single, or double.

\section*{DQPSK Demodulator Baseband}

Supported
Data
Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & \(\bullet\) Double-precision floating point \\
& • Single-precision floating point \\
\hline Output & \(\bullet\) Double-precision floating point \\
& • Single-precision floating point \\
& • Boolean when Output type isBit \\
& \(\bullet 8-, 16\)-, and 32-bit signed integers \\
& \(\bullet 8-, 16\)-, and 32 -bit unsigned integers \\
\hline
\end{tabular}

\author{
Pair Block DQPSK Modulator Baseband \\ See Also \\ M-DPSK Demodulator Baseband, DBPSK Demodulator Baseband, QPSK Demodulator Baseband
}

\section*{DQPSK Modulator Baseband}

\section*{Purpose}

Modulate using differential quaternary phase shift keying method

\section*{Library}

PM, in Digital Baseband sublibrary of Modulation
Description
च—MMA
DQPSK
The DQPSK Modulator Baseband block modulates using the differential quaternary phase shift keying method. The output is a baseband representation of the modulated signal.

The input must be a discrete-time signal. For information about the data types each block port supports, see "Supported Data Types" on page 2-235.

\section*{Integer-Valued Signals and Binary-Valued Signals}

When you set the Input type parameter to Integer, the valid input values are \(0,1,2\), and 3 . In this case, the block accepts a scalar or column vector input signal. If the first input is \(m\), then the modulated symbol is
\[
\exp (j \theta+j \Pi m / 2)
\]
where \(\theta\) represents the Phase rotation parameter. If a successive input is \(m\), then the modulated symbol is the previous modulated symbol multiplied by \(\exp (\mathrm{j} \theta+\mathrm{j} \Pi \mathrm{m} / 2)\).

When you set the Input type parameter to Bit, the input contains pairs of binary values. In this case, the block accepts a column vector whose length is an even integer. The following figure shows the complex numbers by which the block multiples the previous symbol to compute the current symbol, depending on whether you set the Constellation ordering parameter to Binary or Gray. The following figure assumes
that you set the Phase rotation parameter to \(\frac{\Pi}{4}\); in other cases, the
two schematics would be rotated accordingly.

\section*{DQPSK Modulator Baseband}


The following figure shows the signal constellation for the DQPSK
modulation method when you set the Phase rotation parameter to \(\frac{\Pi}{4}\).
The arrows indicate the four possible transitions from each symbol to the next symbol. The Binary and Gray options determine which transition is associated with each pair of input values.


More generally, if the Phase rotation parameter has the form \(\frac{\Pi}{k}\) for
some integer \(k\), then the signal constellation has \(2 k\) points.

\section*{DQPSK Modulator Baseband}


Dialog Box

\section*{Input type}

Indicates whether the input consists of integers or pairs of bits.

\section*{Constellation ordering}

Determines how the block maps each pair of input bits to a corresponding integer, using either a Binary or Gray mapping scheme.

\section*{Phase rotation (rad)}

The phase difference between the previous and current modulated symbols when the input is zero.

\section*{Output Data type}

The output data type can be either single or double. By default, the block sets this to double.

Supported Data
Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
& \(\bullet\) Boolean when Input type is Bit \\
& \(\bullet 8-, 16\)-, and 32 -bit signed integers \\
& \(\bullet 8-, 16\)-, and 32 -bit unsigned integers \\
\hline Output & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
\hline
\end{tabular}

\section*{Pair Block}

DQPSK Demodulator Baseband

\author{
See Also
}

M-DPSK Modulator Baseband, DBPSK Modulator Baseband, QPSK Modulator Baseband

\section*{DSB AM Demodulator Passband}
Purpose Demodulate DSB-AM-modulated data
LibraryAnalog Passband Modulation, in Modulation
Description The DSB AM Demodulator Passband block demodulates a signal thatwas modulated using double-sideband amplitude modulation. The block uses the envelope detection method. The input is a passband representation of the modulated signal. Both the input and output signals are real scalar signals.
In the course of demodulating, this block uses a filter whose order, coefficients, passband ripple and stopband ripple are described by their respective lowpass filter parameters.
Typically, an appropriate Carrier frequency value is much higher than the highest frequency of the input signal. By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the Carrier frequency parameter.
This block works only with real inputs of type double. This block does not work inside a triggered subsystem.

\section*{DSB AM Demodulator Passband}


Dialog
Box

\section*{Input signal offset}

The same as the Input signal offset parameter in the corresponding DSB AM Modulator Passband block.

\section*{Carrier frequency ( Hz )}

The frequency of the carrier in the corresponding DSB AM Modulator Passband block.

\section*{Initial phase (rad)}

The initial phase of the carrier in radians.

\section*{DSB AM Demodulator Passband}

\section*{Lowpass filter design method}

The method used to generate the filter. Available methods are Butterworth, Chebyshev type I, Chebyshev type II, and Elliptic.

Filter order
The order of the lowpass digital filter specified in the Lowpass filter design method field .

Cutoff frequency ( Hz )
The cutoff frequency of the lowpass digital filter specified in the Lowpass filter design method field in Hertz.

\section*{Passband ripple (dB)}

Applies to Chebyshev type I and Elliptic filters only. This is peak-to-peak ripple in the passband in dB .

\section*{Stopband ripple ( dB )}

Applies to Chebyshev type II and Elliptic filters only. This is the peak-to-peak ripple in the stopband in dB .

\author{
Pair Block DSB AM Modulator Passband
}

\section*{DSB AM Modulator Passband}
Purpose Modulate using double-sideband amplitude modulation
Library Analog Passband Modulation, in Modulation
Description The DSB AM Modulator Passband block modulates usingdouble-sideband amplitude modulation. The output is a passband representation of the modulated signal. Both the input and output signals are real scalar signals.
If the input is \(u(t)\) as a function of time \(t\), then the output is
\[
(u(t)+k) \cos \left(2 \pi f_{c} t+\theta\right)
\]
where:
- \(k\) is the Input signal offset parameter.
- \(f_{\mathrm{c}}\) is the Carrier frequency parameter.
- \(\theta\) is the Initial phase parameter.
It is common to set the value of \(k\) to the maximum absolute value of the negative part of the input signal \(u(t)\).
Typically, an appropriate Carrier frequency value is much higher than the highest frequency of the input signal. By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the Carrier frequency parameter.
This block works only with real inputs of type double. This block does not work inside a triggered subsystem.

\section*{DSB AM Modulator Passband}
 Box

\section*{Input signal offset}

The offset factor \(k\). This value should be greater than or equal to the absolute value of the minimum of the input signal.

\section*{Carrier frequency (Hz)}

The frequency of the carrier.

\section*{Initial phase (rad)}

The initial phase of the carrier.

\section*{Pair Block \\ DSB AM Demodulator Passband}

\author{
See Also DSBSC AM Modulator Passband, SSB AM Modulator Passband
}

\title{
DSBSC AM Demodulator Passband
}

\section*{Purpose \\ Demodulate DSBSC-AM-modulated data}

\section*{Library}

Analog Passband Modulation, in Modulation

Description
WMWN
DSBSC AM

The DSBSC AM Demodulator Passband block demodulates a signal that was modulated using double-sideband suppressed-carrier amplitude modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real scalar signals.

In the course of demodulating, this block uses a filter whose order, coefficients, passband ripple and stopband ripple are described by the their respective lowpass filter parameters.

Typically, an appropriate Carrier frequency value is much higher than the highest frequency of the input signal. By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the Carrier frequency parameter.

This block works only with real inputs of type double. This block does not work inside a triggered subsystem.

\section*{DSBSC AM Demodulator Passband}


\section*{DSBSC AM Demodulator Passband}

\section*{Cutoff frequency (Hz)}

The cutoff frequency of the lowpass digital filter specified in the Lowpass filter design method field in Hertz.

\section*{Passband Ripple (dB)}

Applies to Chebyshev type I and Elliptic filters only. This is peak-to-peak ripple in the passband in dB .

\section*{Stopband Ripple (dB)}

Applies to Chebyshev type II and Elliptic filters only. This is the peak-to-peak ripple in the stopband in dB .

\author{
Pair Block DSBSC AM Modulator Passband
}

See Also DSB AM Demodulator Passband, SSB AM Demodulator Passband

\section*{DSBSC AM Modulator Passband}
\begin{tabular}{ll} 
Purpose & \begin{tabular}{l} 
Modulate using double-sideband suppressed-carrier amplitude \\
modulation
\end{tabular} \\
Library & \begin{tabular}{l} 
Analog Passband Modulation, in Modulation
\end{tabular} \\
Description & \begin{tabular}{l} 
The DSBSC AM Modulator Passband block modulates using \\
double-sideband suppressed-carrier amplitude modulation. The output \\
is a passband representation of the modulated signal. Both the input \\
and output signals are real scalar signals.
\end{tabular} \\
If the input is \(u(t)\) as a function of time \(t\), then the output is \\
\(u(t) \cos \left(2 \pi f_{c} t+\theta\right)\)
\end{tabular}\(\quad\)\begin{tabular}{l} 
where \(f_{\mathrm{c}}\) is the Carrier frequency parameter and \(\theta\) is the Initial \\
phase parameter. \\
Typically, an appropriate Carrier frequency value is much higher \\
than the highest frequency of the input signal. By the Nyquist sampling \\
theorem, the reciprocal of the model's sample time (defined by the \\
model's signal source) must exceed twice the Carrier frequency \\
parameter.
\end{tabular}

\section*{DSBSC AM Modulator Passband}


\section*{Dialog Box}

\section*{Carrier frequency ( Hz )}

The frequency of the carrier.

\section*{Initial phase (rad)}

The initial phase of the carrier in radians.

\section*{Pair Block DSBSC AM Demodulator Passband}

See Also DSB AM Modulator Passband, SSB AM Modulator Passband

\section*{Early-Late Gate Timing Recovery}
\begin{tabular}{ll} 
Purpose & Recover symbol timing phase using early-late gate method \\
Library & Timing Phase Recovery sublibrary of Synchronization \\
Description & \begin{tabular}{l} 
The Early-Late Gate Timing Recovery block recovers the symbol timing \\
phase of the input signal using the early-late gate method. This block \\
implements a non-data-aided feedback method.
\end{tabular} \\
\begin{tabular}{|ll}
\begin{tabular}{l} 
Ealy-Late Gate sym, \\
Timing Recoveny
\end{tabular} \\
\hline
\end{tabular} & \begin{tabular}{l} 
Inputs
\end{tabular}
\end{tabular}

By default, the block has one input port. Typically, the input signal is the output of a receive filter that is matched to the transmitting pulse shape.

This block accepts a scalar-valued or column vector input signal. The input uses \(N\) samples to represent each symbol, where \(N>1\) is the Samples per symbol parameter.
- For a column vector input signal, the block operates in single-rate processing mode. In this mode, the output signal inherits its sample rate from the input signal. The input length must be a multiple of \(N\).
- For a scalar input signal, the block operates in multirate processing mode. In this mode, the input and output signals have different sample rates. The output sample rate equals \(N\) multiplied by the input sample rate.
- This block accepts input signals of type Double or Single

If you set the Reset parameter to On nonzero input via port, then the block has a second input port, labeled Rst. The Rst input determines when the timing estimation process restarts, and must be a scalar.
- If the input signal is a scalar value, the sample time of the Rst input equals the symbol period
- If the input signal is a column vector, the sample time of the Rst input equals the input port sample time
- This block accepts reset signals of type Double or Boolean

\section*{Outputs}

The block has two output ports, labeled Sym and Ph:
- The Sym output is the result of applying the estimated phase correction to the input signal. This output is the signal value for each symbol, which can be used for decision purposes. The values in the Sym output occur at the symbol rate:
- For a column vector input signal of length \(N^{*} R\), the Sym output is a column vector of length \(R\) having the same sample rate as the input signal.
- For a scalar input signal, the sample rate of the Sym output equals \(N\) multiplied by the input sample rate.
- The Ph output gives the phase estimate for each symbol in the input.

The Ph output contains nonnegative real numbers less than \(N\). Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time of the Ph output is the same as that of the Sym output.

Note If the Ph output is very close to either zero or Samples per symbol, or if the actual timing phase offset in your input signal is very close to zero, then the block's accuracy might be compromised by small amounts of noise or jitter. The block works well when the timing phase offset is significant rather than very close to zero.
- The output signal inherits its data type from the input signal.

\section*{Delays}

When the input signal is a vector, this block incurs a delay of two symbols. When the input signal is a scalar, this block incurs a delay of three symbols.

\section*{Early-Late Gate Timing Recovery}

Dialog Box


\section*{Samples per symbol}

The number of samples, N , that represent each symbol in the input signal. This must be greater than 1.

\section*{Error update gain}

A positive real number representing the step size that the block uses for updating successive phase estimates. Typically, this number is less than \(1 / \mathrm{N}\), which corresponds to a slowly varying phase.

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink User's Guide.

\section*{Reset}

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are None, Every frame, and On nonzero input via port. The last option causes the block to have a second input port, labeled Rst.

\section*{Algorithm}

\section*{References}
[1] Mengali, Umberto and Aldo N. D'Andrea, Synchronization Techniques for Digital Receivers, New York, Plenum Press, 1997.
[2] Sklar, Bernard. Digital Communications: Fundamentals and Applications. Englewood Cliffs, N.J., Prentice-Hall, 1988.

See Also
Gardner Timing Recovery, Squaring Timing Recovery, Mueller-Muller Timing Recovery

\section*{Error Rate Calculation}

\section*{Purpose}

Compute bit error rate or symbol error rate of input data

\section*{Library}

Description

T× Error Rate \(R x^{\text {Calculation }}\)

Comm Sinks

The Error Rate Calculation block compares input data from a transmitter with input data from a receiver. It calculates the error rate as a running statistic, by dividing the total number of unequal pairs of data elements by the total number of input data elements from one source.

Use this block to compute either symbol or bit error rate, because it does not consider the magnitude of the difference between input data elements. If the inputs are bits, then the block computes the bit error rate. If the inputs are symbols, then it computes the symbol error rate.

Note When you set the Output data parameter to Workspace, the block generates no code. If you need error rate information from generated code, set Output data to Port.

\section*{Input Data}

This block has between two and four input ports, depending on how you set the dialog parameters. The inports marked Tx and Rx accept transmitted and received signals, respectively. The Tx and Rx signals must share the same sampling rate.

The Tx and Rx input ports accept scalar or column vector signals. For information about the data types each block port supports, see the "Supported Data Types" on page 2-260 table on this page. If Tx is a scalar and Rx is a vector, or vice-versa, then the block compares the scalar with each element of the vector. (Overall, the block behaves as if you had preprocessed the scalar signal with the Communications System Toolbox Repeat block with the Rate options parameter set to Enforce single rate.)

If you select Reset port, then an additional input port appears, labeled Rst. The Rst input accepts only a scalar signal (of type double or

\section*{Error Rate Calculation}
boolean) and must have the same port sample time as the Tx and Rx ports. When the Rst input is nonzero, the block clears and then recomputes the error statistics.
If you set the Computation mode parameter to Select samples from port, then an additional input port appears, labeled Sel. The Sel input indicates which elements of a frame are relevant for the computation. The Sel input can be a column vector of type double.
The guidelines below indicate how you should configure the inputs and the dialog parameters depending on how you want this block to interpret your Tx and Rx data.
- If both data signals are scalar, then this block compares the Tx scalar signal with the Rx scalar signal. For this configuration, use the Computation mode parameter default value, Entire frame.
- If both data signals are vectors, then this block compares some or all of the Tx and Rx data:
- If you set the Computation mode parameter to Entire frame, then the block compares all of the Tx frame with all of the Rx frame.
- If you set the Computation mode parameter to Select samples from mask, then the Selected samples from frame field appears in the dialog. This parameter field accepts a vector that lists the indices of those elements of the Rx frame that you want the block to consider. For example, to consider only the first and last elements of a length-six receiver frame, set the Selected samples from frame parameter to [1 6]. If the Selected samples from frame vector includes zeros, then the block ignores them.
- If you set the Computation mode parameter to Select samples from port, then an additional input port, labeled Sel, appears on the block icon. The data at this input port must have the same format as that of the Selected samples from frame parameter described above.
- If one data signal is a scalar and the other is a vector, then this block compares the scalar with each entry of the vector. The three

\section*{Error Rate Calculation}
subbullets above are still valid for this mode, except that if Rx is a scalar, then the phrase "Rx frame" above refers to the vector expansion of Rx.

Note This block does not support variable-size signals. If you choose the Select samples from port option and want the number of elements in the subframe to vary during the simulation, then you should pad the Sel signal with zeros. The Error Rate Calculation block ignores zeros in the Sel signal.

\section*{Output Data}

This block produces a vector of length three, whose entries correspond to:
- The error rate
- The total number of errors, that is, the number of instances that an Rx element does not match the corresponding Tx element
- The total number of comparisons that the block made

The block sends this output data to the base MATLAB workspace or to an output port, depending on how you set the Output data parameter:
- If you set the Output data parameter to Workspace and fill in the Variable name parameter, then that variable in the base MATLAB workspace contains the current value when the simulation ends. Pausing the simulation does not cause the block to write interim data to the variable.

If you plan to use this block along with the Simulink Coder software, then you should not use the Workspace option. Instead, use the Port option and connect the output port to a Simulink To Workspace block.
- If you set the Output data parameter to Port, then an output port appears. This output port contains the running error statistics.

\section*{Error Rate Calculation}

\section*{Delays}

The Receive delay and Computation delay parameters implement two different types of delays for this block. One delay is useful if you want this block to compensate for the delay in the received signal. The other is useful if you want to ignore the initial transient behavior of both input signals.
- The Receive delay parameter represents the number of samples by which the received data lags behind the transmitted data. The transmit signal is implicitly delayed by that same amount before the block compares it to the received data. This value is helpful when you delay the transmit signal so that it aligns with the received signal. The receive delay persists throughout the simulation.
- The Computation delay parameter represents the number of samples the block ignores at the beginning of the comparison.

If you do not know the receive delay in your model, you can use the Align Signals block, which automatically compensates for the delay. If you use the Align Signals block, set the Receive delay in the Error Rate Calculation block to 0 and the Computation delay to the value coming out of the Delay port of the Align Signals block.
Alternatively, you can use the Find Delay block to find the value of the delay, and then set the Receive delay parameter in the Error Rate Calculation block to the delay value.
If you use the Select samples from mask or Select samples from port option, then each delay parameter refers to the number of samples that the block receives, whether the block ultimately ignores some of them or not.

\section*{Stopping the Simulation Based on Error Statistics}

You can configure this block so that its error statistics control the duration of simulation. This is useful for computing reliable steady-state error statistics without knowing in advance how long transient effects might last. To use this mode, check Stop simulation. The block attempts to run the simulation until it detects the number of

\section*{Error Rate Calculation}
errors the Target number of errors parameter specifies. However, the simulation stops before detecting enough errors if the time reaches the model's Stop time setting (in the Configuration Parameters dialog box), if the Error Rate Calculation block makes Maximum number of symbols comparisons, or if another block in the model directs the simulation to stop.

To ignore either of the two stopping criteria in this block, set the corresponding parameter (Target number of errors or Maximum number of symbols) to Inf. For example, to reach a target number of errors without stopping the simulation early, set Maximum number of symbols to Inf and set the model's Stop time to Inf.

\section*{Examples \\ The figure below shows how the block compares pairs of elements and counts the number of error events. The Tx and Rx inputs are column vectors.}


This example assumes that the sample time of each input signal is 1 second and that the block's parameters are as follows:
- Receive delay \(=2\)
- Computation delay \(=0\)
- Computation mode = Entire frame

Both input signals are column vectors of length three. However, the schematic arranges each column vector horizontally and aligns pairs

\section*{Error Rate Calculation}
of vectors so as to reflect a receive delay of two samples. At each time step, the block compares elements of the Rx signal with those of the Tx signal that appear directly above them in the schematic. For instance, at time 1 , the block compares 2,4 , and 1 from the Rx signal with 2,3 , and 1 from the Tx signal.

The values of the first two elements of Rx appear as asterisks because they do not influence the output. Similarly, the 6 and 5 in the Tx signal do not influence the output up to time 3, though they would influence the output at time 4.

In the error rates on the right side of the figure, each numerator at time \(t\) reflects the number of errors when considering the elements of Rx up through time \(t\).

If the block's Reset port box had been checked and a reset had occurred at time \(=3\) seconds, then the last error rate would have been \(2 / 3\) instead of \(4 / 10\). This value \(2 / 3\) would reflect the comparison of 3,2 , and 1 from the Rx signal with 7,7 , and 1 from the Tx signal. The figure below illustrates this scenario. The Tx and Rx inputs are column vectors.


\section*{Tuning Parameters in an RSim Executable (Simulink Coder Software)}

If you use the Simulink Coder rapid simulation (RSim) target to build an RSim executable, then you can tune the Target number of errors and Maximum number of symbols parameters without recompiling the model. This is useful for Monte Carlo simulations in which you

\section*{Error Rate Calculation}
run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

\section*{Error Rate Calculation}


Dialog Box

\section*{Receive delay}

Number of samples by which the received data lags behind the transmitted data. (If Tx or Rx is a vector, then each entry represents a sample.)

\section*{Error Rate Calculation}

\section*{Computation delay}

Number of samples that the block should ignore at the beginning of the comparison.

\section*{Computation mode}

Either Entire frame, Select samples from mask, or Select samples from port, depending on whether the block should consider all or only part of the input frames.

\section*{Selected samples from frame}

A vector that lists the indices of the elements of the Rx frame vector that the block should consider when making comparisons. This field appears only if Computation mode is set to Select samples from mask.

\section*{Output data}

Either Workspace or Port, depending on where you want to send the output data.

\section*{Variable name}

Name of variable for the output data vector in the base MATLAB workspace. This field appears only if Output data is set to Workspace.

\section*{Reset port}

If you check this box, then an additional input port appears, labeled Rst.

\section*{Stop simulation}

If you check this box, then the simulation runs only until this block detects a specified number of errors or performs a specified number of comparisons, whichever comes first.

\section*{Target number of errors}

The simulation stops after detecting this number of errors. This field is active only if Stop simulation is checked.

\section*{Maximum number of symbols}

The simulation stops after making this number of comparisons. This field is active only if Stop simulation is checked.

\section*{Error Rate Calculation}

\section*{Supported Data Types}
\begin{tabular}{|c|c|}
\hline Port & Supported Data Types \\
\hline Tx & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers
\end{tabular} \\
\hline Rx & \begin{tabular}{l}
- Double-precision floating point \\
- Single-precision floating point \\
- Boolean \\
- 8 -, 16-, and 32 -bit signed integers \\
- 8 -, 16 -, and 32 -bit unsigned integers
\end{tabular} \\
\hline Sel & - Double-precision floating point \\
\hline Reset & \begin{tabular}{l}
- Double-precision floating point \\
- Boolean
\end{tabular} \\
\hline
\end{tabular}

\section*{See Also}

Align Signals, Find Delay

\section*{Purpose}

\section*{Library}

Description


Calculate vector magnitude difference between ideal reference signal and measured signal

\section*{Utility Blocks}

Error Vector Magnitude (EVM) is a measurement of modulator or demodulator performance in an impaired signal.
While certain mask selections can change EVM block behavior, the block always has two input signals: a reference signal (at the reference port, Ref) and a corrupted signal (at the input port, In). You must select which normalization method the block uses when performing EVM calculations and which calculations you want as outputs.

The block either normalizes to the average reference signal power, average constellation power, or peak constellation power. For RMS EVM, Max EVM, and X-percentile EVM, the output computations reflect the normalization method.

The default EVM output is RMS EVM in percent, with an option of Output maximum EVM or Output X-percentile EVM values. The maximum EVM represents the worst-case EVM value per burst. For the X-percentile option, you can select to output the number of symbols processed in the percentile computations.
The following table shows the output type, the activation (what selects the output computation), computation units, and the corresponding computation duration.
\begin{tabular}{l|l|l|l}
\hline Output & Activation & Units & \begin{tabular}{l} 
Computation \\
Duration
\end{tabular} \\
\hline RMS EVM & Default & Percentage & Per burst \\
\hline Max EVM & \begin{tabular}{l} 
Parameter \\
setting
\end{tabular} & Percentage & Per burst \\
\hline
\end{tabular}
\begin{tabular}{l|l|l|l}
\hline Output & Activation & Units & \begin{tabular}{l} 
Computation \\
Duration
\end{tabular} \\
\hline Percentile EVM & \begin{tabular}{l} 
Parameter \\
setting
\end{tabular} & Percentage & Continuous \\
\hline \begin{tabular}{l} 
Number of \\
symbols
\end{tabular} & \begin{tabular}{l} 
Parameter \\
setting if you \\
select Output \\
X-percentile \\
EVM
\end{tabular} & None & Continuous \\
\hline
\end{tabular}

The computation duration in per burst mode spans the symbols in the present burst. The computation duration in continuous mode spans all the symbols across multiple bursts.

\section*{Dimension}

The block computes measurements for bursts of data. The data must be of length \(N\), where \(N\) is the size of the burst. When computing RMS EVM or Max EVM, the block computes a unique output for each incoming burst; therefore, the computation duration is per burst.

The block computes the X-percentile for all incoming symbols across multiple bursts. This computation duration is the continuous mode (in contrast to the per burst duration for RMS EVM or Max EVM).

\section*{Input Signals}

This block accepts scalar-valued or column vector input signals. The input and reference signals must have identical dimensions.

\section*{Output Signals}

The output is always a scalar value.

\section*{Data Type}

The block accepts double, single, and fixed-point data types. The output of the block is always double type.

\section*{Algorithms}

The EVM block provides three different normalization methods. You can normalize measurements according to the average power of the reference signal, average constellation power, or peak constellation power. Different industry standards follow one of these normalization methods.

The following table lists how the block calculates the RMS EVM value for different normalization methods.
\begin{tabular}{l|l}
\hline \begin{tabular}{l} 
EVM \\
Normalization \\
Method
\end{tabular} & Algorithm \\
\hline Reference Signal & \(E V M_{R M S}=\sqrt{\frac{\frac{1}{N} \sum_{k=1}^{N}\left(e_{k}\right)}{\frac{1}{N} \sum_{k=1}^{N}\left(I_{k}^{2}+Q_{k}^{2}\right)}} * 100\) \\
\hline Average Power & \(E V M_{R M S}=\sqrt{\frac{\frac{1}{N} \sum_{k=1}^{N}\left(e_{k}\right)}{P_{a v g}}} * 100\) \\
\hline Peak Power & \(E V M_{R M S}=\sqrt{\frac{1}{N} \sum_{k=1}^{P_{\text {max }}\left(e_{k}\right)}} * 100\) \\
\hline
\end{tabular}
where,
\(e_{k}=\left(I_{k}-\tilde{I_{k}}\right)^{2}+\left(Q_{k}-\tilde{Q}_{k}\right)^{2}\)
\(I_{k}=\) In-phase measurement of the \(k t h\) symbol in the burst
\(Q_{k}=\) Quadrature phase measurement of the \(k t h\) symbol in the burst
\(N=\) Input vector length
\(P_{\text {avg }}=\) The value for Average constellation power
\(P_{\max }=\) The value for Peak constellation power
\(I_{k}\) and \(Q_{k}\) represent ideal (reference) values. \(\tilde{I}_{k}\) and \(\tilde{Q}_{k}\) represent measured (received) symbols.

The max EVM is the maximum EVM value in a frame or
\[
E V M_{\max }=\max _{k \in[1, \ldots, N]}\left\{E V M_{k}\right\}
\]
where \(k\) is the \(k\) th symbol in a burst of length \(N\).
The definition for \(\mathrm{EVM}_{\mathrm{k}}\) varies depending upon which normalization method you select for computing measurements. The block supports the algorithms in the following table.
\begin{tabular}{l|l}
\hline \begin{tabular}{l} 
EVM \\
Normalization
\end{tabular} & Algorithm \\
\hline Reference Signal & \(E V M_{k}=\sqrt{\frac{e_{k}}{\frac{1}{N} \sum_{k=1}^{N}\left(I_{k}^{2}+Q_{k}^{2}\right)}} * 100\) \\
\hline Average Power & \(E V M_{k}=\sqrt{\frac{e_{k}}{P_{a v g}}} * 100\) \\
\hline Peak Power & \(E V M_{k}=\sqrt{\frac{e_{k}}{P_{\text {max }}}} * 100\) \\
\hline
\end{tabular}

The block computes X-percentile EVM by creating a histogram of all the incoming \(E V M_{k}\) values. The output provides the EVM value below which X\% of the EVM values lay.


Dialog Box

\section*{Normalize RMS error vector by:}

Selects the method by which the block normalizes measurements:
- Average reference signal power
- Average constellation power
- Peak constellation power

This parameter defaults to Average reference signal power.

\section*{Average constellation power:}

Normalizes EVM measurement by the average constellation power. This parameter only appears if you set Normalize RMS error vector to Average constellation power.

\section*{Peak constellation power:}

Normalizes EVM measurement by the peak constellation power. This parameter only appears if you set Normalize RMS error vector to Peak constellation power.

\section*{Output maximum EVM}

Outputs the maximum EVM of an input vector or frame.

\section*{Output X-percentile EVM}

Enables an output X-percentile EVM measurement. When you select this option, specify X-percentile value (\%).

X -percentile value (\%)
This parameter only appears when you select Output X-percentile EVM. The Xth percentile is the EVM value below which X\% of all the computed EVM values lie. The parameter defaults to the 95 th percentile. Therefore, \(95 \%\) of all EVM values are below this output.

\section*{Output the number of symbols processed}

Outputs the number of symbols that the block uses to compute the Output X-percentile EVM. This parameter only appears when you select Output X-percentile EVM.

\author{
Examples To see an example using the EVM block, refer to Measuring Modulator Accuracy in the Communications System Toolbox User's Guide.
}

\section*{References}

\section*{References}
[1] IEEE Standard 802.16-2004: "Part 16: Air interface for fixed broadband wireless access systems," October 2004. http://ieee802.org/16/published.html
[2] 3 GPP TS 45.005 V8.1.0 (2008-05): "Radio Access Network: Radio transmission and reception"
[3] IEEE Standard 802.11a-1999: "Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications: High-speed Physical Layer in the 5 GHz Band," 1999.

See Also
MER Measurement

\section*{Purpose}

Find delay between two signals

\section*{Library}

Description


Utility Blocks

The Find Delay block finds the delay between a signal and a delayed, and possibly distorted, version of itself. This is useful when you want to compare a transmitted and received signal to find the bit error rate, but do not know the delay in the received signal. This block accepts a column vector or matrix input signal. For a matrix input, the block outputs a row vector, and finds the delay in each channel of the matrix independently. See "Delays" for more information about signal delays.

The sRef input port receives the original signal, while the sDel input port receives the delayed version of the signal. The two input signals must have the same dimensions and sample times.

The output port labeled delay outputs the delay in units of samples. If you select Include "change signal" output port, then an output port labeled chg appears. The chg output port outputs 1 when there is a change from the delay computed at the previous sample, and 0 when there is no change. The delay output port outputs signals of type double, and the chg output port outputs signals of type boolean.

The block's Correlation window length parameter specifies how many samples of the signals the block uses to calculate the cross-correlation. The delay output is a nonnegative integer less than the Correlation window length.
As the Correlation window length is increased, the reliability of the computed delay also increases. However, the processing time to compute the delay increases as well.

You can make the Find Delay block stop updating the delay after it computes the same delay value for a specified number of samples. To do so, select Disable recurring updates, and enter a positive integer in the Number of constant delay outputs to disable updates field. For example, if you set Number of constant delay outputs to disable updates to 20 , the block will stop recalculating and updating the delay after it calculates the same value 20 times in succession.

Disabling recurring updates causes the simulation to run faster after the target number of constant delays occurs.

\section*{Tips for Using the Block Effectively}
- Set Correlation window length sufficiently large so that the computed delay eventually stabilizes at a constant value. When this occurs, the signal from the optional chg output port stabilizes at the constant value of zero. If the computed delay is not constant, you should increase Correlation window length. If the increased value of Correlation window length exceeds the duration of the simulation, then you should also increase the duration of the simulation accordingly.
- If the cross-correlation between the two signals is broad, then the Correlation window length value should be much larger than the expected delay, or else the algorithm might stabilize at an incorrect value. For example, a CPM signal has a broad autocorrelation, so it has a broad cross-correlation with a delayed version of itself. In this case, the Correlation window length value should be much larger than the expected delay.
- If the block calculates a delay that is greater than 75 percent of the Correlation window length, the signal sRef is probably delayed relative to the signal sDel. In this case, you should switch the signal lines leading into the two input ports.

\section*{Examples}

\section*{Finding the Delay Before Calculating an Error Rate}

A typical use of this block is to determine the correct Receive delay parameter in the Error Rate Calculation block. This is illustrated in "Use the Find Delay and Align Signals Blocks". In that example, the modulation/demodulation operation introduces a computational delay into the received signal and the Find Delay block determines that the delay is 6 samples. This value of 6 becomes a parameter in the Error Rate Calculation block, which computes the bit error rate of the system. Another example of this usage is in "Delays".

\section*{Finding the Delay to Help Align Words}

Another typical use of this block is to determine how to align the boundaries of frames with the boundaries of codewords or other types of data blocks. "Delays" describes when such alignment is necessary and also illustrates, in the "Aligning Words of a Block Code" discussion, how to use the Find Delay block to solve the problem.

\section*{Setting the Correlation Window Length}

The next example illustrates how to tell when the Correlation window length is not sufficiently large.


The model uses a Delay block to delay a signal by 10 samples. The Find Delay block compares the original signal with the delayed version. In this model, the Input processing parameter of the Delay block is set to Columns as channels. The model then displays the output of the Find Delay block in a scope. If the Correlation window length is 15, the scope shows that the calculated delay is not constant over time, as you can see in the following image.


This result tells you to increase the Correlation window length. If you increase it to 50 , the calculated delay stabilizes at 10 , as shown below.



\section*{Correlation window length}

The number of samples the block uses to calculate the cross-correlations of the two signals.

\section*{Include "change signal" output port}

If you select this option, then the block has an extra output port that emits an impulse when the current computed delay differs from the previous computed delay.

\section*{Disable recurring updates}

Selecting this option causes the block to stop computing the delay after it computes the same delay value for a specified number of samples.

\section*{Number of constant delay outputs to disable updates}

A positive integer specifying how many times the block must compute the same delay before ceasing to update. This field appears only if Disable recurring updates is selected.

\section*{Algorithm}

See Also

The Find Delay block finds the delay by calculating the cross-correlations of the first signal with time-shifted versions of the second signal, and then finding the index at which the cross-correlation is maximized.

\author{
Align Signals, Error Rate Calculation
}

\section*{Purpose Demodulate FM-modulated data}

Library
Analog Passband Modulation, in Modulation
Description The FM Demodulator Passband block demodulates a signal that was modulated using frequency modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real scalar signals.
For best results, use a carrier frequency which is estimated to be larger than \(10 \%\) of the reciprocal of your input signal's sample rate. This is due to the implementation of the Hilbert transform by means of a filter.

In the following example, we sample a 10 Hz input signal at 8000 samples per second. We then designate a Hilbert Transform filter of order 100. Below is the response of the Hilbert Transform filter as returned by fvtool.


Note the bandwidth of the filter's magnitude response. By choosing a carrier frequency larger than \(10 \%\) (but less than \(90 \%\) ) of the reciprocal of your input signal's sample time ( 8000 samples per second, in this example) or equivalently, a carrier frequency larger than 400 Hz , we ensure that the Hilbert Transform Filter will be operating in the flat section of the filter's magnitude response (shown in blue), and that our modulated signal will have the desired magnitude and form.
Typically, an appropriate Carrier frequency value is much higher than the highest frequency of the input signal. By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the Carrier frequency parameter.

\section*{FM Demodulator Passband}

This block works only with real inputs of type double. This block does not work inside a triggered subsystem.

Dialog Box
\begin{tabular}{|c|c|}
\hline (7) Function Block Parameters: FM Demodulator Passband & x \\
\hline \multicolumn{2}{|l|}{FM Demodulator Passband (mask) (link)} \\
\hline \multicolumn{2}{|l|}{Demodulate a frequency modulated signal with a discriminator.} \\
\hline \multicolumn{2}{|l|}{The input signal must be a scalar.} \\
\hline \multicolumn{2}{|l|}{\multirow[t]{2}{*}{Parameters Carrier frequency \((\mathrm{Hz})\) :}} \\
\hline & \\
\hline \multicolumn{2}{|l|}{300} \\
\hline \multicolumn{2}{|l|}{Initial phase (rad):} \\
\hline \multicolumn{2}{|l|}{0} \\
\hline \multicolumn{2}{|l|}{Frequency deviation \((\mathrm{Hz})\) :} \\
\hline \multicolumn{2}{|l|}{50} \\
\hline \multicolumn{2}{|l|}{Hilbert transform filter order (must be even):} \\
\hline \multicolumn{2}{|l|}{100} \\
\hline OK Cancel Help & Apply \\
\hline
\end{tabular}

\section*{Carrier frequency (Hz)}

The frequency of the carrier.

\section*{Initial phase (rad)}

The initial phase of the carrier in radians.

\section*{Frequency deviation ( Hz )}

The frequency deviation of the carrier frequency in Hertz.
Sometimes it is referred to as the "variation" in the frequency.

\section*{Hilbert transform filter order}

The length of the FIR filter used to compute the Hilbert transform.

\author{
Pair Block FM Modulator Passband
}

\section*{Purpose Modulate using frequency modulation}

\section*{Library \\ Analog Passband Modulation, in Modulation}

Description
The FM Modulator Passband block modulates using frequency modulation. The output is a passband representation of the modulated signal. The output signal's frequency varies with the input signal's amplitude. Both the input and output signals are real scalar signals.

If the input is \(u(t)\) as a function of time \(t\), then the output is
\[
\cos \left(2 \pi f_{c} t+2 \pi K_{c} \int_{0}^{t} u(\tau) d \tau+\theta\right)
\]
where:
- \(f_{\mathrm{c}}\) represents the Carrier frequency parameter.
- \(\theta\) represents the Initial phase parameter.
- \(K_{\mathrm{c}}\) represents the Frequency deviation parameter.

Typically, an appropriate Carrier frequency value is much higher than the highest frequency of the input signal.

By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the Carrier frequency parameter.

This block works only with real inputs of type double. This block does not work inside a triggered subsystem.

\section*{FM Modulator Passband}


\section*{Purpose}

Reduce amplitude of input signal by amount specified

\section*{Library}

Description
Free Space Path Loss 10 dB

RF Impairments
The Free Space Path Loss block simulates the loss of signal power due to the distance between transmitter and receiver. The block reduces the amplitude of the input signal by an amount that is determined in either of two ways:
- By the Distance (km) and Carrier frequency (MHz) parameters, if you specify Distance and Frequency in the Mode field
- By the Loss (dB) parameter, if you specify Decibels in the Mode field

This block accepts a column vector input signal. The input signal to this block must be a complex signal.

\section*{Free Space Path Loss}

Dialog Box


\section*{Mode}

Method of specifying the amount by which the signal power is reduced. The choices are Decibels and Distance and Frequency.

\section*{Loss}

The signal loss in decibels. This parameter appears when you set Mode to Decibels. The decibel amount shown on the mask is rounded for display purposes only.

\section*{Distance}

Distance between transmitter and receiver in kilometers. This parameter appears when you set Mode to Distance and Frequency.

\section*{Carrier frequency \((\mathbf{M H z})\)}

The carrier frequency in megahertz. This parameter appears when you set Mode to Distance and Frequency.

Examples The model below illustrates the effect of the Free Space Path Loss block with the following parameter settings:
- Mode is set to Distance and Frequency.
- Distance (km) is set to 0.5
- Carrier frequency \((\mathbf{M H z})\) is set to 180


See Also Memoryless Nonlinearity

\section*{Gardner Timing Recovery}

\section*{Purpose \\ Library \\ Description}
\(\begin{gathered}\text { Gardner Sym } \\ \text { Timing Recovery }\end{gathered}\) Ph

Recover symbol timing phase using Gardner's method

Timing Phase Recovery sublibrary of Synchronization
The Gardner Timing Recovery block recovers the symbol timing phase of the input signal using Gardner's method. This block implements a non-data-aided feedback method that is independent of carrier phase recovery. The timing error detector that forms part of this block's algorithm requires at least two samples per symbol, one of which is the point at which the decision can be made.

The recovery method estimates the symbol timing phase offset for each incoming symbol and outputs the signal value corresponding to the estimated symbol sampling instant.

The second output returns the estimated timing phase recovery offset for each symbol, which is a nonnegative real number less than N , where N is the number of samples per symbol.

The error update gain parameter is the step size used for updating the successive phase estimates.

\section*{Inputs}

By default, this block has one input port. Typically, the input signal is the output of a receive filter that is matched to the transmitting pulse shape. For best results, the input signal power should be less than 1.

This block accepts a scalar-valued or column vector input signal. The input uses \(N\) samples to represent each symbol, where \(N>1\) is the Samples per symbol parameter.
- For a column vector input signal, the block operates in single-rate processing mode. In this mode, the output signal inherits its sample rate from the input signal. The input length must be a multiple of \(N\).
- For a scalar input signal, the block operates in multirate processing mode. In this mode, the input and output signals have different sample rates. The output sample rate equals \(N\) multiplied by the input sample rate.
- This block accepts input signals of type Double or Single

If you set the Reset parameter to On nonzero input via port, then the block has a second input port, labeled Rst. The Rst input determines when the timing estimation process restarts, and must be a scalar.
- If the input signal is a scalar value, the sample time of the Rst input equals the symbol period
- If the input signal is a column vector, the sample time of the Rst input equals the input port sample time
- This block accepts reset signals of type Double or Boolean

\section*{Outputs}

The block has two output ports, labeled Sym and Ph:
- The Sym output is the result of applying the estimated phase correction to the input signal. This output is the signal value for each symbol, which can be used for decision purposes. The values in the Sym output occur at the symbol rate:
- For a column vector input signal of length \(N^{*} R\), the Sym output is a column vector of length \(R\) having the same sample rate as the input signal.
- For a scalar input signal, the sample rate of the Sym output equals \(N\) multiplied by the input sample rate.
- The Ph output gives the phase estimate for each symbol in the input.

The Ph output contains nonnegative real numbers less than \(N\). Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time of the Ph output is the same as that of the Sym output.

> Note If the Ph output is very close to either zero or Samples per symbol, or if the actual timing phase offset in your input signal is very close to zero, then the block's accuracy might be compromised by small amounts of noise or jitter. The block works well when the timing phase offset is significant rather than very close to zero.
- The output signal inherits its data type from the input signal.

\section*{Delays}

When the input signal is a vector, this block incurs a delay of two symbols. When the input signal is a scalar, this block incurs a delay of three symbols.

Dialog
Box


\section*{Samples per symbol}

The number of samples, N , that represent each symbol in the input signal. This must be greater than 1.

\section*{Error update gain}

A positive real number representing the step size that the block uses for updating successive phase estimates. Typically, this number is less than \(1 / \mathrm{N}\), which corresponds to a slowly varying phase.

\section*{Gardner Timing Recovery}

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink User's Guide.

\section*{Reset}

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are None, Every frame, and On nonzero input via port. The last option causes the block to have a second input port, labeled Rst.

\section*{Algorithm}

This block uses a timing error detector whose result for the kth symbol is \(\mathrm{e}(\mathrm{k})\), given by
\[
\begin{aligned}
e(k) & =a_{I}(k)+a_{Q}(k) \\
a_{I}(k) & =\left\{y_{I}\left((k-1) T+d_{k-1}\right)-y_{I}\left(k T+d_{k}\right)\right\} y_{I}\left(k T-T / 2+d_{k-1}\right) \\
a_{Q}(k) & =\left\{y_{Q}\left((k-1) T+d_{k-1}\right)-y_{Q}\left(k T+d_{k}\right)\right\} y_{Q}\left(k T-T / 2+d_{k-1}\right)
\end{aligned}
\]
where
- \(y_{I}\) and \(y_{Q}\) are the in-phase and quadrature components, respectively, of the block's input signal
- T is the symbol period
- \(d_{k}\) is the phase estimate for the kth symbol

Notice from the expressions in curly braces above that the timing error detector approximates the derivative of y using finite differences.

For more information about the role that e(k) plays in this block's algorithm, see "Feedback Methods for Timing Phase Recovery" in Communications System Toolbox User's Guide.

Supported Data Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & • Double-precision floating point \\
& • Single-precision floating point \\
\hline Reset & • Double-precision floating point \\
& • Boolean \\
\hline Output & • Double-precision floating point \\
& • Single-precision floating point \\
\hline
\end{tabular}

Examples
References

The commgardnerphrecov demonstration model uses this block.
[1] Gardner, F. M., "A BPSK/QPSK Timing-Error Detector for Sampled Receivers", IEEE Transactions on Communications, Vol. COM-34, No. 5, May 1986, pp. 423-429.
[2] Mengali, Umberto and Aldo N. D'Andrea, Synchronization Techniques for Digital Receivers, New York, Plenum Press, 1997.
[3] Meyr, Heinrich, Marc Moeneclaey, and Stefan A. Fechtel, Digital Communication Receivers, Vol 2, New York, Wiley, 1998.
[4] Oerder, M., "Derivation of Gardner's Timing-Error Detector from the ML principle", IEEE Transactions on Communications, Vol. COM-35, No. 6, June 1987, pp. 684-685.

See Also Early-Late Gate Timing Recovery, Squaring Timing Recovery, Mueller-Muller Timing Recovery

\section*{Gaussian Filter}

Purpose
Library
Description


Filter input signal, possibly downsampling, using Gaussian FIR filter

Comm Filters

The Gaussian Filter block filters the input signal using a Gaussian FIR filter. The block expects the input signal to be upsampled as its input, so that the Input samples per symbol parameter, N , is at least 2. The block's icon shows the filter's impulse response."

\section*{Characteristics of the Filter}

The impulse response of the Gaussian filter is
\[
h(t)=\frac{\exp \left(\frac{-t^{2}}{2 \delta^{2}}\right)}{\sqrt{2 \pi} \cdot \delta}
\]
where
\[
\delta=\frac{\sqrt{\ln (2)}}{2 \pi B T}
\]
and B is the filter's \(3-\mathrm{dB}\) bandwidth. The BT product parameter is B times the input signal's symbol period. For a given BT product, the Signal Processing Toolbox gaussfir function generates a filter that is half the bandwidth of the filter generated by the Communications System Toolbox Gaussian Filter block.

The Group delay parameter is the number of symbol periods between the start of the filter's response and the peak of the filter's response. The group delay and N determine the length of the filter's impulse response, which is 2 * N * Group delay +1 .

The Filter coefficient normalization parameter indicates how the block scales the set of filter coefficients:
- Sum of coefficients means that the sum of the coefficients equals 1.

\section*{Gaussian Filter}
- Filter energy means that the sum of the squares of the coefficients equals 1.
- Peak amplitude means that the maximum coefficient equals 1.

After the block normalizes the set of filter coefficients as above, it multiplies all coefficients by the Linear amplitude filter gain parameter. As a result, the output is scaled by \(\sqrt{N}\). If the output of this block feeds the input to the AWGN Channel block, specify the AWGN signal power parameter to be \(1 / \mathrm{N}\).

\section*{Input and Output Signals}

This block accepts scalar, column vector, and \(M\)-by- \(N\) matrix input signals. The block filters an \(M\)-by- \(N\) input matrix as follows:
- When you set the Input processing parameter to Columns as channels (frame based), the block treats each column as a separate channel. In this mode, the block creates \(N\) instances of the same filter, each with its own independent state buffer. Each of the \(N\) filters process \(M\) input samples at every Simulink time step.
- When you set the Input processing parameter to Elements as channels (sample based), the block treats each element as a separate channel. In this mode, the block creates \(M^{*} N\) instances of the same filter, each with its own independent state buffer. Each filter processes one input sample at every Simulink time step.

The output dimensions always equal those of the input signal. For information about the data types each block port supports, see the table on this page.

\section*{Exporting Filter Coefficients to the MATLAB Workspace}

To examine or manipulate the coefficients of the filter that this block designs, select Export filter coefficients to workspace. Then set the Coefficient variable name parameter to the name of a variable that you want the block to create in the MATLAB workspace. Running

\section*{Gaussian Filter}
the simulation causes the block to create the variable, overwriting any previous contents in case the variable already exists.

Dialog Box


\section*{Input samples per symbol}

A positive integer representing the number of samples per symbol in the input signal.

\section*{BT product}

The product of the filter's \(3-\mathrm{dB}\) bandwidth and the input signal's symbol period

\section*{Group delay}

A positive integer that represents the number of symbol periods between the start of the filter response and its peak.

\section*{Filter coefficient normalization}

The block scales the set of filter coefficients so that this quantity equals 1. Choices are Sum of coefficients, Filter energy, and Peak amplitude.

\section*{Input processing}

Specify how the block processes the input signal. You can set this parameter to one of the following options:
- Columns as channels (frame based) - When you select this option, the block treats each column of the input as a separate channel.
- Elements as channels (sample based) - When you select this option, the block treats each element of the input as a separate channel.

Note The Inherited (this choice will be removed - see release notes) option will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

This parameter is available only when you set the Rate options parameter to Allow multirate processing.

\section*{Linear amplitude filter gain}

A positive scalar used to scale the filter coefficients after the block uses the normalization specified in the Filter coefficient normalization parameter.

\section*{Export filter coefficients to workspace}

If you check this box, then the block creates a variable in the MATLAB workspace that contains the filter coefficients.

\section*{Coefficient variable name}

The name of the variable to create in the MATLAB workspace. This field appears only if Export filter coefficients to workspace is selected.

\section*{Visualize filter with FVTool}

If you click this button, then MATLAB launches the Filter Visualization Tool, fvtool, to analyze the Gaussian filter whenever you apply any changes to the block's parameters. If you launch fvtool for the filter, and subsequently change parameters in the mask, fvtool will not update. You will need to launch a new fvtool in order to see the new filter characteristics. Also note that if you have launched fvtool, then it will remain open even after the model is closed.

\section*{Gaussian Filter}


\section*{Rounding mode}

Select the rounding mode for fixed-point operations. The filter coefficients do not obey this parameter; they always round to Nearest. The block uses the Rounding selection if a number cannot be represented exactly by the specified data type and scaling, it is rounded to a representable number. For more information, see Rounding Modes in the DSP System Toolbox documentation or "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

\section*{Gaussian Filter}

\section*{Overflow mode}

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

\section*{Coefficients}

The block implementation uses a Direct-Form FIR filter. The Coefficients parameter controls which data type represents the coefficients when the input data is a fixed-point signal.

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator). See "Filter Structure Diagrams" in the DSP System Toolbox Reference Guide for illustrations depicting the use of the coefficient data types in this block:
- When you select Same word length as input, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Specify word length, you are able to enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the coefficients, in bits. If applicable, you are able to enter separate fraction lengths for the numerator and denominator coefficients.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the coefficients. If applicable, you are able to enter separate slopes for the numerator and denominator coefficients. This block requires power-of-two slope and a bias of zero.

\section*{Gaussian Filter}
- The filter coefficients do not obey the Rounding mode and the Overflow mode parameters; they are always saturated and rounded to Nearest.

\section*{Product output}

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Multiplication Data Types" and "Multiplication Data Types" in the DSP System Toolbox Reference Guide for illustrations depicting the use of the product output data type in this block:
- When you select Same as input, these characteristics match those of the input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the product output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

\section*{Accumulator}

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:
- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as product output, these characteristics match those of the product output.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.

\section*{Gaussian Filter}
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

\section*{Output}

Choose how you specify the output word length and fraction length:
- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as accumulator, these characteristics match those of the accumulator.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

\section*{Lock scaling against changes by the autoscaling tool}

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.

\section*{Supported Data Type}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline In & - Double-precision floating point \\
& - Single-precision floating point \\
& - Signed fixed-point \\
\hline Out & - Double-precision floating point \\
& - Single-precision floating point \\
& - Signed fixed-point \\
\hline
\end{tabular}

\section*{See Also}

Raised Cosine Receive Filter, gaussfir

\section*{Gaussian Filter}

\footnotetext{
References
[1] 3GPP TS 05.04 V8.4.0 - 3rd Generation Partnership Project; Technical Specification Group GSM/EDGE Radio Access Network; Digital cellular telecommunications system (Phase 2+); Modulation (Release 1999)
}

\section*{Gaussian Noise Generator}

\section*{Purpose Generate Gaussian distributed noise with given mean and variance values \\ Library Noise Generators sublibrary of Comm Sources \\ Description \\ trand \\ Gaussian \\ The Gaussian Noise Generator block generates discrete-time white Gaussian noise. You must specify the Initial seed vector in the simulation.}

The Mean Value and the Variance can be either scalars or vectors. If either of these is a scalar, then the block applies the same value to each element of a sample-based output or each column of a frame-based output. Individual elements or columns, respectively, are uncorrelated with each other.

When the Variance is a vector, its length must be the same as that of the Initial seed vector. In this case, the covariance matrix is a diagonal matrix whose diagonal elements come from the Variance vector. Since the off-diagonal elements are zero, the output Gaussian random variables are uncorrelated.

When the Variance is a square matrix, it represents the covariance matrix. Its off-diagonal elements are the correlations between pairs of output Gaussian random variables. In this case, the Variance matrix must be positive definite, and it must be N -by- N , where N is the length of the Initial seed.

The probability density function of \(n\)-dimensional Gaussian noise is
\[
f(x)=\left((2 \pi)^{n} \operatorname{det} K\right)^{-1 / 2} \exp \left(-(x-\mu)^{T} K^{-1}(x-\mu) / 2\right)
\]
where \(x\) is a length \(-n\) vector, \(K\) is the \(n\)-by- \(n\) covariance matrix, \(\mu\) is the mean value vector, and the superscript \(T\) indicates matrix transpose.

\section*{Initial Seed}

The Initial seed parameter initializes the random number generator that the Gaussian Noise Generator block uses to add noise to the input signal. For best results, the Initial seed should be a prime number

\section*{Gaussian Noise Generator}
greater than 30. Also, if there are other blocks in a model that have an Initial seed parameter, you should choose different initial seeds for all such blocks.

You can choose seeds for the Gaussian Noise Generator block using the Communications System Toolbox randseed function. At the MATLAB prompt, enter
randseed

This returns a random prime number greater than 30. Entering randseed again produces a different prime number. If you supply an integer argument, randseed always returns the same prime for that integer. For example, randseed (5) always returns the same answer.

\section*{Attributes of Output Signal}

The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the Frame-based outputs, Samples per frame, and Interpret vector parameters as 1-D parameters. See "Sources and Sinks" in the Communications System Toolbox User's Guide for more details.

If the Initial seed parameter is a vector, then its length becomes the number of columns in a frame-based output or the number of elements in a sample-based vector output. In this case, the shape (row or column) of the Initial seed parameter becomes the shape of a sample-based two-dimensional output signal. If the Initial seed parameter is a scalar but either the Mean value or Variance parameter is a vector, then the vector length determines the output attributes mentioned above.

\section*{Gaussian Noise Generator}
 Box

\section*{Mean value}

The mean value of the random variable output.

\section*{Variance}

The covariance among the output random variables.

\section*{Initial seed}

The initial seed value for the random number generator.

\section*{Sample time}

The period of each sample-based vector or each row of a frame-based matrix.

\section*{Gaussian Noise Generator}

\section*{Frame-based outputs}

Determines whether the output is frame-based or sample-based. This box is active only if Interpret vector parameters as 1-D is unchecked.

\section*{Samples per frame}

The number of samples in each column of a frame-based output signal. This field is active only if Frame-based outputs is checked.

\section*{Interpret vector parameters as 1-D}

If this box is checked, then the output is a one-dimensional signal. Otherwise, the output is a two-dimensional signal. This box is active only if Frame-based outputs is unchecked.

\section*{Output data type}

The output can be set to double or single data types.
See Also Random Source (DSP System Toolbox documentation), AWGN Channel, rand (built-in MATLAB function), randseed

\section*{General Block Deinterleaver}

Description

General Block Deinterleaver

\section*{Purpose Restore ordering of symbols in input vector \\ Library \\ Block sublibrary of Interleaving}

The General Block Deinterleaver block rearranges the elements of its input vector without repeating or omitting any elements. If the input contains \(N\) elements, then the Elements parameter is a column vector of length \(N\). The column vector indicates the indices, in order, of the output elements that came from the input vector. That is, for each integer \(k\) between 1 and \(N\),

Output(Elements( \(k\) )) \(=\operatorname{Input}(k)\)
The Elements parameter must contain unique integers between 1 and \(N\).

Both the input and the Elements parameter must be column vector signals.

This block accept the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.

To use this block as an inverse of the General Block Interleaver block, use the same Elements parameter in both blocks. In that case, the two blocks are inverses in the sense that applying the General Block Interleaver block followed by the General Block Deinterleaver block leaves data unchanged.


\section*{Dialog} Box

Examples

See Also perms (MATLAB function) block is \([40 ; 32 ; 59 ; 1]\).

\section*{Pair Block \\ General Block Interleaver}

\section*{Elements} that came from the input vector.
This example reverses the operation in the example on the General Block Interleaver block reference page. If you set Elements to \([4,1,3,2]\) ' and you set the General Block Deinterleaver block input to \([1 ; 40 ; 59 ; 32\) ], then the output of the General Block Deinterleaver
A vector of length N that lists the indices of the output elements

\section*{General Block Interleaver}

Purpose Reorder symbols in input vector

Library
Description
General Block Interleaver

Block sublibrary of Interleaving
The General Block Interleaver block rearranges the elements of its input vector without repeating or omitting any elements. If the input contains \(N\) elements, then the Elements parameter is a column vector of length \(N\). The column vector indicates the indices, in order, of the input elements that form the length- \(N\) output vector; that is,
\[
\text { Output }(k)=\operatorname{Input}(\text { Elements(k)) }
\]
for each integer \(k\) between 1 and \(N\). The contents of Elements must be integers between 1 and N , and must have no repetitions.

Both the input and the Elements parameter must be column vector signals.

This block accept the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.

\section*{General Block Interleaver}


\section*{Dialog} Box

Examples

Pair Block
See Also

If Elements is \([4 ; 1 ; 3 ; 2]\) and the input vector is \([40 ; 32 ; 59 ; 1]\), then the output vector is \([1 ; 40 ; 59 ; 32]\). Notice that all of these vectors have the same length and that the vector Elements is a permutation of the vector [1:4]'.

General Block Deinterleaver
ments is \([4 ; 1 ; 3 ; 2]\)
put vector is \([1 ; 40 ; 5\)
me length and that th
\([1: 4]^{\prime}\).
perms (MATLAB function)

\section*{General CRC Generator}

\section*{Purpose}

Library
Description
General CRC Generator

Generate CRC bits according to generator polynomial and append to input data frames

CRC sublibrary of Error Correction and Detection
The General CRC Generator block generates cyclic redundancy code (CRC) bits for each input data frame and appends them to the frame. This block accepts a binary column vector input signal.

You specify the generator polynomial for the CRC algorithm using the Generator polynomial parameter. This block is general in the sense that the degree of the polynomial does not need to be a power of two. You represent the polynomial in one of these ways:
- As a binary row vector containing the coefficients in descending order of powers. For example, [ \(\left.\begin{array}{llll}1 & 1 & 0 & 1\end{array}\right]\) represents the polynomial \(x^{3}+x^{2}+1\).
- As an integer row vector containing the powers of nonzero terms in the polynomial, in descending order. For example, [llll 32200\(]\) represents the polynomial \(x^{3}+x^{2}+1\).

You specify the initial state of the internal shift register by the Initial states parameter. The Initial states parameter is either a scalar or a binary row vector of length equal to the degree of the generator polynomial. A scalar value is expanded to a row vector of length equal to the degree of the generator polynomial. For example, the default initial state of [ 0 ] is expanded to a row vector of all zeros.

You specify the number of checksums that the block calculates for each input frame by the Checksums per frame parameter. The Checksums per frame value must evenly divide the size of the input frame. If the value of Checksums per frame is \(k\), the block does the following:

1 Divides each input frame into \(k\) subframes of equal size
2 Prefixes the Initial states vector to each of the \(k\) subframes

\section*{General CRC Generator}

3 Applies the CRC algorithm to each augmented subframe
4 Appends the resulting checksums at the end of each subframe
5 Outputs concatenated subframes
If the size of the input frame is \(m\) and the degree of the generator polynomial is \(r\), the output frame has size \(m+k * r\).

This block supports double and boolean data types. The block inherits the output data type from the input signal.

\section*{Example}

Suppose the size of the input frame is 10 , the degree of the generator polynomial is 3 , Initial states is [0], and Checksums per frame is 2. The block divides each input frame into two subframes of size 5 and appends a checksum of size 3 to each subframe, as shown below. The initial states are not shown in this example, because an initial state of [0] does not affect the output of the CRC algorithm. The output frame then has size \(5+3+5+3=16\).

\section*{General CRC Generator}


\section*{Example of Cyclic Redundancy Check Encoding}

This example clarifies the operation of the General CRC Generator block by comparing the generated CRC bits from the library block with those generated from primitive Simulink blocks. To open the model, enter doc_crcgen at the MATLAB command line.

\section*{General CRC Generator}

\section*{Cyclic Redundancy Check Encoding}


For a known input message with a length of 6 bits, the model generates
CRC bits for a generator polynomial, \(g(x)=x^{3}+x+1\), and a specific initial state of the register.

You can experiment with different initial states by changing the value of Initial states prior to running the simulation. For all values, the comparison (generated CRC bits from the library block with those generated from primitive Simulink blocks) holds true.

Using the General CRC Generator block allows you to easily specify the generator polynomial (especially for higher order polynomials).

\section*{Signal Attributes}

The General CRC Generator block has one input port and one output port. Both ports support binary column vector signals.

\section*{General CRC Generator}


\section*{Box}

\section*{Generator polynomial}

A binary or integer row vector specifying the generator polynomial, in descending order of powers.

\section*{General CRC Generator}

\section*{Initial conditions}

Binary scalar or a binary row vector of length equal to the degree of the generator polynomial, specifying the initial state of the internal shift register.

\section*{Direct method}

When you select this check box, the object uses the direct algorithm for CRC checksum calculations. When you clear this check box, the object uses the non-direct algorithm for CRC checksum calculations.

\section*{Reflect input bytes}

When you select this check box, the block flips the input data on a bytewise basis prior to entering the data into the shift register. For this application, the input frame length (and any current input frame length for variable-size signals) divided by the value for the Checksums per frame parameter must be a multiple of eight. When you clear this check box, the block does not flip the input data.

\section*{Reflect checksums before final XOR}

When you select this check box, the block flips the CRC checksums around their centers after the input data are completely through the shift register. When you clear this check box, the block does not flip the CRC checksums.

\section*{Final XOR}

Specify the value with which the CRC checksum is to be XORed as a binary scalar or vector. The block applies the XOR operation just prior to appending the input data. The vector length is the degree of the generator polynomial that you specify in the Generator polynomial parameter. When you specify the final XOR value as a scalar, the block expands the value to a row vector with a length equal to the degree of the generator polynomial. The default value of this parameter is 0 , which is equivalent to no XOR operation.

\section*{Checksums per frame}

Specify the number of checksums the block calculates for each input frame. This value must be a positive integer. The input

\section*{General CRC Generator}
frame length (and any current input frame length for variable-size signals) must be a multiple of this parameter value.

\author{
Algorithm For a description of the CRC algorithm as implemented by this block, see "Cyclic Redundancy Check Codes" in Communications System Toolbox User's Guide. \\ References [1] Sklar, Bernard, Digital Communications: Fundamentals and Applications. Englewood Cliffs, N.J., Prentice-Hall, 1988. \\ [2] Wicker, Stephen B., Error Control Systems for Digital Communication and Storage, Upper Saddle River, N.J., Prentice Hall, 1995. \\ \section*{Pair Block General CRC Syndrome Detector} \\ See Also CRC-N Generator, CRC-N Syndrome Detector
}

\section*{General CRC Generator HDL Optimized}


This hardware-friendly CRC generator block, like the General CRC Generator block, generates the CRC bits and appends them to the input message bits. The output consists of CRC checksum plus the message. With the General CRC Generator HDL Optimized block, the processing is optimized for HDL code generation. Instead of processing an entire frame at once, the block processes samples of data. Control signals are added at both input and output for easy data synchronization.

\section*{Signal Attributes}

The General CRC Generator HDL Optimized block has four input ports and four output ports.


\section*{General CRC Generator HDL Optimized}
\begin{tabular}{l|l|l|l}
\hline Port & Direction & Description & \begin{tabular}{l} 
Data \\
Type
\end{tabular} \\
\hline dataIn & Input & \begin{tabular}{l} 
Message data. Data width is \\
less than or equal to the CRC \\
length, and the CRC length \\
should be divisible by the \\
data width. For example, for \\
CRC-CCITT/CRC-16, the valid \\
data widths are 16, 8, 4, 2 and 1.
\end{tabular} & \begin{tabular}{l} 
Column \\
vector of \\
double, \\
boolean, \\
or ufix1
\end{tabular} \\
\hline startIn & Input & \begin{tabular}{l} 
Indicates the start of a frame of \\
data.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline endIn & Input & \begin{tabular}{l} 
Indicates the end of a frame of \\
data.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline validIn & Input & \begin{tabular}{l} 
Indicates that input data is \\
valid.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline dataOut & Output & \begin{tabular}{l} 
Message data with the checksum \\
appended. The data width is the \\
same as the input data port.
\end{tabular} & \begin{tabular}{l} 
Column \\
vector of \\
double, \\
boolean, \\
or ufiv1
\end{tabular} \\
\hline startOut & Output & \begin{tabular}{l} 
Indicates the start of a frame of \\
data.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline endOut & Output & \begin{tabular}{l} 
Indicates the end of a frame of \\
data, including checksum.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline validOut & Output & \begin{tabular}{l} 
Indicates that output data is \\
valid.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline
\end{tabular}

\section*{General CRC Generator HDL Optimized}

Dialog Box

Function Block Parameters: General CRC Generator HDL Optimized
\(x\)
General CRC Generator HDL Optimized
Generate CRC code bits and append to input data
Polynomial: \(\quad[1,0,0,0,1,0,0,0,0,0,0,1,0,0,0,0,1]\)
Initial state: \(\quad 0\)
\(\square \quad\) Reflect input
\(\square \quad\) Reflect CRC checksum
Final XOR value: \(\quad 0\) Cancel Help Apply

\section*{Polynomial}

A double, boolean, or ufix1 row or column vector specifying the polynomial, in descending order of powers. CRC length is length(polynomial)-1. The default value is \([1000100000\) 0100001 ].

\section*{Initial state}

A double, boolean, or ufix1 scalar or vector of length equal to the CRC length, specifying the initial state of the internal shift register. The default value is 0 .

\section*{Reflect input}
- The input data width must be a multiple of 8 .
- When checked, each input byte is flipped before entering the shift register.
- When unchecked, the message data is passed to the shift register unchanged.
The default value is unchecked.

\section*{General CRC Generator HDL Optimized}

\section*{Reflect CRC checksum}
- The CRC length must be a multiple of 8 .
- When checked, each checksum byte is flipped before it is passed to the final XOR stage.
- When unchecked, the checksum byte is passed to the final XOR stage unchanged.
The default value is unchecked.
Final XOR value
The value with which the CRC checksum is to be XORed just prior to being appended to the input data. A double, boolean, or ufix1 scalar or vector of length equal to the CRC length, specifying the FinalXOR value. The default value is 0 .

\section*{Algorithm \\ Timing Diagram}

Timing diagram of CRC generator

\section*{General CRC Generator HDL Optimized}


\section*{Initial Delay}

The General CRC Generator HDL Optimized block introduces a latency on the output. This latency can be computed with the following equation:
```

initialdelay = CRC length/input data width + 2

```

Example See Using HDL Optimized CRC Library Blocks.

\section*{General CRC Generator HDL Optimized}

\author{
Pair Block General CRC Syndrome Detector HDL Optimized \\ See Also General CRC Generator
}

\section*{General CRC Syndrome Detector}

\section*{Purpose}

Detect errors in input data frames according to generator polynomial

\section*{Library}

Description

General CRC Syndrome Detector Err

CRC sublibrary of Error Correction and Detection
The General CRC Syndrome Detector block computes checksums for its entire input frame. This block accepts a binary column vector input signal.
The block's second output is a vector whose size is the number of checksums, and whose entries are 0 if the checksum computation yields a zero value, and 1 otherwise. The block's first output is the set of message words with the checksums removed.

The first output is the data frame with the CRC bits removed and the second output indicates if an error was detected in the data frame.

The block's parameter settings should agree with those in the General CRC Generator block.

You specify the number of checksums the block calculates for each frame by the Checksums per frame parameter. If the Checksums per frame value is \(k\), the size of the input frame is \(n\), and the degree of the generator polynomial is \(r\), then \(k\) must divide \(n-k^{*} r\), which is the size of the message word.

This block supports double and boolean data types. The block inherits the output data type from the input signal.

\section*{Example}

Suppose the received codeword has size 16, the generator polynomial has degree 3, Initial states is [0], and Checksums per frame is 2. The block computes the two checksums of size 3, one from the first half of the received codeword, and the other from the second half of the received codeword, as shown in the following figure. The initial states are not shown in this example, because an initial state of [0] does not affect the output of the CRC algorithm. The block concatenates the two halves of the message word as a single vector of size 10 and outputs this vector through the first output port. The block outputs a 2 -by- 1 binary frame vector whose entries depend on whether the computed checksums

\section*{General CRC Syndrome Detector}
are zero. The following figure shows an example in which the first checksum is nonzero and the second checksum is zero. This indicates that an error occurred in transmitting the first half of the codeword.


\section*{Signal Attributes}

The General CRC Syndrome Detector block has one input port and two output ports. These ports accept binary column vector signals.

\section*{General CRC Syndrome Detector}

Function Block Parameters: General CRC Syndrome Detector
General CRC Syndrome Detector (mask) (link)
Detect errors in the input data frames according to the generator polynomial parameter. Specify the generator polynomial as a binary vector or a descending ordered polynomial to indicate the connection points.

The first output is the data frame with the CRC bits removed and the second output indicates if an error was detected in the data frame.

This block accepts a binary column vector input signal.
Parameters
Generator polynomial:
[10001000000100001]
Initial states:
0Direct methodReflect input bytesReflect checksums before final XOR
Final XOR:
0
Checksums per frame:
1

Dialog
Box

\section*{Generator polynomial}

A binary or integer row vector specifying the generator polynomial, in descending order of powers.

\section*{General CRC Syndrome Detector}

\section*{Initial conditions}

A binary scalar or a binary row vector of length equal to the degree of the generator polynomial, specifying the initial state of the internal shift register.

\section*{Direct method}

When you select this check box, the object uses the direct algorithm for CRC checksum calculations. When you clear this check box, the object uses the non-direct algorithm for CRC checksum calculations.

\section*{Reflect input bytes}

When you select this check box, the block flips the input data on a bytewise basis prior to entering the data into the shift register. For this application, the input frame length (and any current input frame length for variable-size signals) divided by the value for the Checksums per frame parameter minus the degree of the generator polynomial, which you specify in the Generator polynomial parameter, must be a multiple of eight. When you clear this check box, the block does not flip the input data.

\section*{Reflect checksums before final XOR}

When you select this check box, the block flips the CRC checksums around their centers after the input data are completely through the shift register. When you clear this check box, the block does not flip the CRC checksums.

\section*{Final XOR}

Specify the value with which the CRC checksum is to be XORed as a binary scalar or vector. The block applies the XOR operation just prior to appending the input data. The vector length is the degree of the generator polynomial that you specify in the Generator polynomial parameter. When you specify the final XOR value as a scalar, the block expands the value to a row vector with a length equal to the degree of the generator polynomial. The default value of this parameter is 0 , which is equivalent to no XOR operation.

\section*{General CRC Syndrome Detector}

\section*{Checksums per frame}

Specify the number of checksums the block calculates for each input frame. This value must be a positive integer. The input frame length (and any current input frame length for variable-size signals) must be a multiple of this parameter value.

\author{
Algorithm For a description of the CRC algorithm as implemented by this block, see "Cyclic Redundancy Check Codes" in Communications System Toolbox User's Guide. \\ References [1] Sklar, Bernard. Digital Communications: Fundamentals and Applications. Englewood Cliffs, N.J., Prentice-Hall, 1988. \\ [2] Wicker, Stephen B., Error Control Systems for Digital Communication and Storage, Upper Saddle River, N.J., Prentice Hall, 1995. \\ \section*{Pair Block General CRC Generator} \\ See Also CRC-N Generator, CRC-N Syndrome Detector
}

\section*{General CRC Syndrome Detector HDL Optimized}

Purpose Detect errors in input data using CRC
Library
CRC sublibrary of Error Correction and Detection

\section*{Description}


GeneralCRC Syndrome Detector HDL Optimized
This hardware-friendly CRC detector block performs a CRC on data and compares the resulting checksum with the appended checksum. An error is detected if the two checksums do not match. Instead of frame processing, the block processes data at the streaming mode. Control signals are added at both input and output of the block for easy data synchronization.

\section*{Signal Attributes}

The General CRC Syndrome Detector HDL Optimized block has four input ports and five output ports.


\section*{General CRC Syndrome Detector HDL Optimized}
\begin{tabular}{l|l|l|l}
\hline Port & Direction & Description & \begin{tabular}{l} 
Data \\
Type
\end{tabular} \\
\hline dataIn & Input & \begin{tabular}{l} 
Message data. Data width is \\
less than or equal to the CRC \\
length, and the CRC length \\
should be divisible by the \\
data width. For example, for \\
CRC-CCITT/CRC-16, the valid \\
data widths are 16, 8, 4, 2 and 1.
\end{tabular} & \begin{tabular}{l} 
Column \\
vector of \\
double, \\
Boolean, \\
or ufix1
\end{tabular} \\
\hline startIn & Input & \begin{tabular}{l} 
Indicates the start of a frame of \\
data.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline endIn & Input & \begin{tabular}{l} 
Indicates the end of a frame of \\
data.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline validIn & Input & \begin{tabular}{l} 
Indicates that input data is \\
valid.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline dataOut & Output & \begin{tabular}{l} 
Message data with the checksum \\
appended. The data width is the \\
same as the input data port.
\end{tabular} & \begin{tabular}{l} 
Column \\
vector of \\
double, \\
Boolean, \\
or ufiv1
\end{tabular} \\
\hline startOut & Output & \begin{tabular}{l} 
Indicates the start of a frame of \\
data.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline endOut & Output & \begin{tabular}{l} 
Indicates the end of a frame of \\
data, including checksum.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline validOut & Output & \begin{tabular}{l} 
Indicates that output data is \\
valid.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline err & Output & \begin{tabular}{l} 
Indicates the corruption of the \\
received data when error is high.
\end{tabular} & \begin{tabular}{l} 
Boolean \\
or ufix1
\end{tabular} \\
\hline
\end{tabular}

\section*{General CRC Syndrome Detector HDL Optimized}

\section*{Dialog Box}

Function Block Parameters: General CRC Syndrome Detector HDL Optimized

\section*{General CRC Syndrome Detector HDL Optimized}

Detect errors in input data using CRC
Polynomial: \(\quad[1,0,0,0,1,0,0,0,0,0,0,1,0,0,0,0,1]\)

Initial state:
0
Reflect input
Reflect CRC checksum
Final XOR value:

Cancel
Help
Apply

\section*{Polynomial}

A double, boolean, or ufix1 row or column vector specifying the polynomial, in descending order of powers. CRC length is length(polynomial)-1. The default value is \(\left[\begin{array}{lll}100 & 0 & 100000\end{array}\right.\) 0100001 ].

\section*{Initial state}

A double, boolean, or ufix1 scalar or vector of length equal to the CRC length, specifying the initial state of the internal shift register. The default value is 0 .

\section*{Reflect input}
- The input data width must be a multiple of 8 .
- When checked, each input byte is flipped before entering the shift register.
- When unchecked, the message data is passed to the shift register unchanged.
The default value is unchecked.

\section*{General CRC Syndrome Detector HDL Optimized}

\section*{Reflect CRC checksum}
- The CRC length must be a multiple of 8 .
- When checked, each checksum byte is flipped before it is passed to the final XOR stage.
- When unchecked, the checksum byte is passed to the final XOR stage unchanged.
The default value is unchecked.

\section*{Final XOR value}

The value with which the CRC checksum is to be XORed just prior to being appended to the input data. A double, boolean, or ufix1 scalar or vector of length equal to the CRC length, specifying the FinalXOR value. The default value is 0 .

\section*{Algorithm}

Timing Diagram
Timing diagram of CRC detector

\section*{General CRC Syndrome Detector HDL Optimized}


\section*{Initial Delay}

The General CRC Syndrome Detector HDL Optimized block introduces a latency on the output. This latency can be computed with the following equation:
initialdelay \(=(3\) * CRC length/input data width \()+2\)
Example See Using HDL Optimized CRC Library Blocks.
Pair Block General CRC Generator HDL Optimized
See Also General CRC Syndrome Detector

\section*{General Multiplexed Deinterleaver}

\section*{Purpose}

Restore ordering of symbols using specified-delay shift registers

\section*{Library}

Description

General Multiplexed Deinterleaver

Convolutional sublibrary of Interleaving
The General Multiplexed Deinterleaver block restores the original ordering of a sequence that was interleaved using the General Multiplexed Interleaver block.

In typical usage, the parameters in the two blocks have the same
values. As a result, the Interleaver delay parameter, \(V\), specifies the delays for each shift register in the corresponding interleaver, so that the delays of the deinterleaver's shift registers are actually \(\max (V)-V\).

This block accepts a scalar or column vector input signal, which can be real or complex. The output signal has the same sample time as the input signal.

The block can accept the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The block inherist data type of this output will be the same as that of the input signal.

\section*{General Multiplexed Deinterleaver}


Dialog Box

\section*{Interleaver delay (samples)}

A vector that lists the number of symbols that fit in each shift register of the corresponding interleaver. The length of this vector is the number of shift registers.

\section*{Initial conditions}

The values that fill each shift register when the simulation begins.

\section*{Pair Block \\ General Multiplexed Interleaver}

See Also Convolutional Deinterleaver, Helical Deinterleaver

\section*{References}
[1] Heegard, Chris and Stephen B. Wicker. Turbo Coding. Boston: Kluwer Academic Publishers, 1999.

\section*{General Multiplexed Interleaver}

Purpose
Permute input symbols using set of shift registers with specified delays

\section*{Library}

Description

General Multiplexed Interleaver

Convolutional sublibrary of Interleaving
The General Multiplexed Interleaver block permutes the symbols in the input signal. Internally, it uses a set of shift registers, each with its own delay value.

This block accepts a scalar or column vector input signal, which can be
real or complex. The input and output signals have the same sample time.

The block can accept the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal has the same data type as the input signal.

\section*{General Multiplexed Interleaver}


Dialog Box

\section*{Interleaver delay (samples)}

A column vector listing the number of symbols that fit into each shift register. The length of this vector is the number of shift registers. (In sample-based mode, it can also be a row vector.)

\section*{Initial conditions}

The values that fill each shift register at the beginning of the simulation.

If Initial conditions is a scalar, then its value fills all shift registers. If Initial conditions is a column vector, then each entry fills the corresponding shift register. (In sample-based mode, Initial conditions can also be a row vector.) If a given shift register has zero delay, then the value of the corresponding entry in the Initial conditions vector is unimportant.

\author{
Pair Block \\ General Multiplexed Deinterleaver
}

\section*{General Multiplexed Interleaver}

See Also Convolutional Interleaver, Helical Interleaver
References [1] Heegard, Chris and Stephen B. Wicker. Turbo Coding. Boston: Kluwer Academic Publishers, 1999.

\section*{General QAM Demodulator Baseband}

\author{
Purpose Demodulate QAM-modulated data
}

Library
Description

AM, in Digital Baseband sublibrary of Modulation
The General QAM Demodulator Baseband block demodulates a signal that was modulated using quadrature amplitude modulation. The input is a baseband representation of the modulated signal.
The input must be a discrete-time complex signal. The Signal constellation parameter defines the constellation by listing its points in a length-M vector of complex numbers. The block maps the \(m\) th point in the Signal constellation vector to the integer \(m-1\).

This block accepts a scalar or column vector input signal. For information about the data types each block port supports, see the "Supported Data Types" on page 2-348 table on this page.

\section*{General QAM Demodulator Baseband}


A real or complex vector that lists the constellation points.

\section*{General QAM Demodulator Baseband}

\section*{Output type}

Determines whether the block produces integers or binary representations of integers.

If you set this parameter to Integer, the block produces integers.
If you set this parameter to Bit, the block produces a group of K bits, called a binary word, for each symbol, when Decision type is set to Hard decision. If Decision type is set to Log-likelihood ratio or Approximate log-likelihood ratio, the block outputs bitwise LLR and approximate LLR, respectively.

\section*{Decision type}

This field appears when Bit is selected in the pull-down list Output type.

Specifies the use of hard decision, LLR, or approximate LLR during demodulation. See "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the Communications System Toolbox User's Guide for algorithm details.

\section*{Noise variance source}

This field appears when you set Approximate log-likelihood ratio or Log-likelihood ratio for Decision type.

When you set this parameter to Dialog, you can then specify the noise variance in the Noise variance field. When you set this option to Port, a port appears on the block through which the noise variance can be input.

\section*{Noise variance}

This parameter appears when the Noise variance source is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter

\section*{General QAM Demodulator Baseband}

Fixed-Point Signal Flow Diagrams
without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:
- Inf to - Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.


\section*{Fixed-Point Signal Flow Diagram for Hard Decision Mode}

Note In the figure above, \(M\) represents the size of the Signal constellation .

The general QAM Demodulator Baseband block supports fixed-point operations for computing Hard Decision (Output type set to Bit

\section*{General QAM Demodulator Baseband}
and Decision type is set to Hard decision) and Approximate LLR (Output type is set to Bit and Decision type is set to Approximate Log-Likelihood ratio) output values. The input values must have fixed-point data type for fixed-point operations.

Note Fixed-Point operations are NOT yet supported for Exact LLR output values.


\section*{Fixed-Point Signal Flow Diagram for Approximate LLR Mode}

\section*{General QAM Demodulator Baseband}

Note In the figure above, \(M\) represents the size of the Signal constellation.


Fixed-Point Signal Flow Diagram for Approximate LLR Mode: Noise Variance Operation Modes

Note If Noise variance is set to Dialog, the block performs the operations shown inside the dotted line once during initialization. The block also performs these operations if the Noise variance value changes during simulation.

\section*{General QAM Demodulator Baseband}

\section*{Data \\ Types Attributes}


Fixed-Point Attributes for Hard Decision Mode

\section*{General QAM Demodulator Baseband}

\section*{Output}

The block supports the following Output options:
When you set the parameter to Inherit via internal rule (default setting), the block inherits the output data type from the input port. The output data type is the same as the input data type if the input is of type single or double.

For integer outputs, you can set this block's output to Inherit via internal rule (default setting), Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, single, and double.

For bit outputs, when you set Decision type to Hard decision, you can set the output to Inherit via internal rule, Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, boolean, single, or double.

When you set Decision type to Hard decision or Approximate log-likelihood ratio and the input is a floating point data type, then the output inherits its data type from the input. For example, if the input is of data type double, the output is also of data type double. When you set Decision type to Hard decision or Approximate log-likelihood ratio, and the input is a fixed-point signal, the Output parameter, located in the Fixed-Point algorithm parameters region of the Data-Type tab, specifies the output data type.

When you set the parameter to Smallest unsigned integer, the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box. If you select ASIC/FPGA in the Hardware Implementation pane, the output data type is the ideal
minimum size, i.e., ufix (1) for bit outputs, and ufix \(\left(\left\lceil\log _{2} M\right\rceil\right)\) for integer outputs. For all other choices, the Output data type is an unsigned integer with the smallest available word length large

\section*{General QAM Demodulator Baseband}
enough to fit the ideal minimum size, usually corresponding to the size of a char (e.g., uint8).

\section*{Rounding Mode Parameter}

Use this parameter to specify the rounding method to be used when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result.

For more information, see "Rounding Modes" in the DSP System Toolbox documentation "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

\section*{Overflow Mode Parameter}

Use this parameter to specify the method to be used if the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result:
- Saturate represents positive overflows as the largest positive number in the range being used, and negative overflows as the largest negative number in the range being used.
- Wrap uses modulo arithmetic to cast an overflow back into the representable range of the data type. See Modulo Arithmetic for more information.

For more information, see the Rounding Mode Parameter subsection of "Specify Fixed-Point Attributes for Blocks".

\section*{Signal constellation}

Use this parameter to define the data type of the Signal constellation parameter.
- When you select Same word length as input the word length of the Signal constellation parameter matches that of the input to the block. The fraction length is computed to provide the best precision for given signal constellation values.
- When you select Specify word length, the Word Length field appears, and you may enter a value for the word length.

\section*{General QAM Demodulator Baseband}

The fraction length is computed to provide the best precision for given signal constellation values.

\section*{Accumulator 1}

Use this parameter to specify the data type for Accumulator 1:
- When you select Inherit via internal rule, the block automatically calculates the output word and fraction lengths. For more information, see the "Inherit via Internal Rule" subsection of the DSP System Toolbox User's Guide.
- When you select Binary point scaling, you can enter the word length and the fraction length of Accumulator 1, in bits.

\section*{Product Input}

Use this parameter to specify the data type for Product input.
- When you select Same as accumulator 1, the Product Input characteristics match those of Accumulator 1.
- When you select Binary point scaling you can enter the word length and the fraction length of Product input, in bits.

\section*{Product Output}

Use this parameter to select the data type for Product output.
- When you select Inherit via internal rule, the block automatically calculates the output signal type. For more information, see the Inherit via Internal Rule subsection of the DSP System Toolbox User's Guide.
- When you select Binary point scaling enter the word length and the fraction length for Product output, in bits.

\section*{Accumulator 2}

Use this parameter to specify the data type for Accumulator 2:
- When you select Inherit via internal rule, the block automatically calculates the accumulator data type. The internal rule calculates the ideal, full-precision word length and fraction length as follows:
\[
W L_{\text {ideal accumulator } 2}=W L_{\text {input to accumulator } 2}
\]

\section*{General QAM Demodulator Baseband}
\(F L_{\text {ideal accumulator 2 }}=F L_{\text {input to accumulator 2 }}\)
After the full-precision result is calculated, your particular hardware may still affect the final word and fraction lengths set by the internal rule. For more information, see The Effect of the Hardware Implementation Pane on the Internal Rule subsection of the DSP System Toolbox User's Guide.

The internal rule always sets the sign of data-type to Unsigned .
- When you select Binary point scaling, you are able to enter the word length and the fraction length of Accumulator 2, in bits.

\section*{General QAM Demodulator Baseband}

\section*{Function Block Parameters: General QAM Demodulator Baseband}

General QAM Demodulator Baseband
Demodulate the input signal using the quadrature amplitude modulation method.
The input must be 1-D or a 2-D column vector.
The output can be either bits or integers. For bit output, the size of signal constellation must be an integer power of two and the output width is an integer multiple of the number of bits per symbol. In this case, Decision type parameter allows to choose between Hard decision demodulation, Log-likelihood ratio and Approximate log-likelihood ratio. The output values for Log-likelihood ratio and Approximate log-likelihood ratio Decision types are of the same data type as the input values. For integer output, the block always performs Hard decision demodulation.

Main Data Types

Output: Inherit via internal rule

Floating-point inheritance takes precedence over the settings in the 'Data Type' column below. When the block input is floating point, all block data types match the input.

Fixed-point operational parameters
Rounding mode: Floor Overflow mode: Wrap

Fixed-point algorithm parameters
\begin{tabular}{|c|c|c|c|}
\hline & \multicolumn{2}{|l|}{Data Type} & Signed \\
\hline Signal constellation: & Same word length as input & \(\checkmark\) & Yes \\
\hline Accumulator 1: & Inherit via internal rule & \(\square\) & Yes \\
\hline Product input: & Same as accumulator 1 & \(\square\) & Yes \\
\hline Product output: & Inherit via internal rule & \(\nabla\) & Yes \\
\hline Accumulator 2: & Inherit via internal rule & \(\square\) & No \\
\hline
\end{tabular}
\(\square\)
Fixed-Point Attributes for Approximate LLR Mode

\section*{General QAM Demodulator Baseband}

The settings for the following fixed-point parameters only apply when you set Decision type to Approximate log-likelihood ratio.

\section*{Accumulator 3}

When you select Inherit via internal rule, the block automatically calculates the accumulator data type. The internal rule first calculates ideal, full-precision word length and fraction length as follows:
\[
\begin{aligned}
& W L_{\text {ideal accumulator 3 }}=W L_{\text {input to accumulator 3 }}+1 \\
& F L_{\text {ideal accumulator 3 }}=F L_{\text {input to accumulator 3 }} .
\end{aligned}
\]

After the full-precision result is calculated, your particular hardware may still affect the final word and fraction lengths set by the internal rule. For more information, see The Effect of the Hardware Implementation Pane on the Internal Rule subsection of the DSP System Toolbox User's Guide.

The internal rule always sets the sign of data-type to Signed.

\section*{Noise scaling input}
- When you select Same as accumulator 3, the Noise scaling input characteristics match those of Accumulator 3.
- When you select Binary point scaling you are able to enter the word length and the fraction length of Noise scaling input, in bits.

\section*{Inverse noise variance}

This field appears when Noise variance source is set to Dialog.
- When you select Same word length as input the word length of the Inverse noise variance parameter matches that of the input to the block. The fraction length is computed to provide the best precision for a given inverse noise variance value.
- When you select Specify word length, the Word Length field appears, and you may enter a value for the word length.

\section*{General QAM Demodulator Baseband}

The fraction length is computed to provide the best precision for a given inverse noise variance value.

\section*{Output}

When you select Inherit via internal rule, the Output data type is automatically set for you.

If you set the Noise variance source parameter to Dialog, the output is a result of product operation as shown in the Noise Variance Operation Modes Signal Flow Diagram Fixed-Point Signal Flow Diagram for Approximate LLR Mode: Noise Variance Operation Modes on page 2-339. In this case, it follows the internal rule for Product data types specified in the Inherit via Internal Rule subsection of the DSP System Toolbox User's Guide.

If the Noise variance source parameter is set to Port, the output is a result of division operation as shown in the signal flow diagram. In this case, the internal rule calculates the ideal, full-precision word length and fraction length as follows:
\(W L_{\text {output }}=\max \left(W L_{\text {Noise scaling input }}, W L_{\text {Noise variance }}\right)\)
\(F L_{\text {output }}=F L_{\text {Noise scaling input (dividend) }}-F L_{\text {Noise variance (divisor) }}\).
After the full-precision result is calculated, your particular hardware may still affect the final word and fraction lengths set by the internal rule. For more information, see "The Effect of the Hardware Implementation Pane on the Internal Rule" subsection of the DSP System Toolbox User's Guide.

The internal rule for Output always sets the sign of data-type to Signed.

For additional information about the parameters pertaining to fixed-point applications, see "Specify Fixed-Point Attributes for Blocks".

\section*{General QAM Demodulator Baseband}

Supported
Data
Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & • Double-precision floating point \\
& - Single-precision floating point \\
& - Signed fixed-point when Output type is Integer or Output type is Bit \\
and Decision type is either Hard-decision or Approximate LLR
\end{tabular}

\author{
Pair Block General QAM Modulator Baseband \\ See Also Rectangular QAM Demodulator Baseband
}

\section*{General QAM Modulator Baseband}
\begin{tabular}{|c|c|}
\hline Purpose & Modulate using quadrature amplitude modulation \\
\hline Library & AM, in Digital Baseband sublibrary of Modulation \\
\hline \multirow[t]{5}{*}{\[
\begin{gathered}
\text { Gemmmal } \\
\substack{\text { General } \\
\text { QAM }}
\end{gathered}
\]} & The General QAM Modulator Baseband block modulates using quadrature amplitude modulation. The output is a baseband representation of the modulated signal. \\
\hline & The Signal constellation parameter defines the constellation by listing its points in a length-M vector of complex numbers. The input signal values must be integers between 0 and \(\mathrm{M}-1\). The block maps an input integer \(m\) to the \((m+1)\) st value in the Signal constellation vector. \\
\hline & This block accepts a scalar or column vector input signal. For information about the data types each block port supports, see the "Supported Data Types" on page 2-351 table on this page. \\
\hline & Constellation Visualization \\
\hline & The General QAM Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the Communications System Toolbox User's Guide. \\
\hline
\end{tabular}

\section*{General QAM Modulator Baseband}


Dialog Box

\section*{Signal constellation}

A real or complex vector that lists the constellation points.

\section*{Output data type}

The output data type can be set to double, single, Fixed-point, User-defined, or Inherit via back propagation.

Setting this to Fixed-point or User-defined will enable fields in which you can further specify details. Setting this to Inherit via back propagation, sets the output data type and scaling to match the following block..

\section*{Output word length}

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed-point for the Output data type parameter.

\section*{User-defined data type}

Specify any signed built-in or signed fixed-point data type. You can specify fixed-point data types using the sfix, sint, sfrac, and fixdt functions from Fixed-Point Designer software. This

\section*{General QAM Modulator Baseband}
parameter is only visible when you select User-defined for the Output data type parameter.

\section*{Set output fraction length to}

Specify the scaling of the fixed-point output by either of the following two methods:
- Choose Best precision to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose User-defined to specify the output scaling in the Output fraction length parameter.

This parameter is only visible when you select Fixed-point for the Output data type parameter, or when you select User-defined and the specified output data type is a fixed-point data type.

\section*{Output fraction length}

For fixed-point output data types, specify the number of fractional bits, or bits to the right of the binary point. This parameter is only visible when you select Fixed-point or User-defined for the Output data type parameter and User-defined for the Set output fraction length to parameter.

\section*{Supported \\ Data \\ Types}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
& \(\bullet 8-, 16-, 32\)-bit signed integers \\
& \(\bullet 8-, 16-, 32\)-bit unsigned integers \\
&
\end{tabular}

\section*{General QAM Modulator Baseband}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline & - ufix \(\left(\left\lceil\log _{2} M\right\rceil\right)\) \\
\hline Output & • Double-precision floating point \\
& - Single-precision floating point \\
& - Signed fixed-point \\
\hline
\end{tabular}

\author{
Pair Block General QAM Demodulator Baseband
}

\author{
See Also Rectangular QAM Modulator Baseband
}

\section*{General TCM Decoder}

\section*{Purpose}

\section*{Library}

Description

Decode trellis-coded modulation data, mapped using arbitrary constellation

TCM, in Digital Baseband sublibrary of Modulation
The General TCM Decoder block uses the Viterbi algorithm to decode a trellis-coded modulation (TCM) signal that was previously modulated using an arbitrary signal constellation.

The Trellis structure and Signal constellation parameters in this block should match those in the General TCM Encoder block, to ensure proper decoding. In particular, the Signal constellation parameter must be in set-partitioned order.

\section*{Input and Output Signals}

This block accepts a column vector input signal containing complex numbers. The input signal must be double or single. The reset port signal must be double or Boolean. For information about the data types each block port supports, see "Supported Data Types" on page 2-356.

If the convolutional encoder described by the trellis structure represents a rate \(k / n\) code, then the General TCM Decoder block's output is a binary column vector whose length is \(k\) times the vector length of the input signal.

\section*{Operation Modes}

The block has three possible methods for transitioning between successive frames. The Operation mode parameter controls which method the block uses. This parameter also affects the range of possible values for the Traceback depth parameter, D.
- In Continuous mode, the block initializes all state metrics to zero at the beginning of the simulation, waits until it accumulates D symbols, and then uses a sequence of \(D\) symbols to compute each of the traceback paths. D can be any positive integer. At the end of each frame, the block saves its internal state metric for use with the next frame.

\section*{General TCM Decoder}

If you select Enable the reset input port, the block displays another input port, labeled Rst. This port receives an integer scalar signal. Whenever the value at the Rst port is nonzero, the block resets all state metrics to zero and sets the traceback memory to zero.
- In Truncated mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In Terminated mode, the block treats each frame independently. The traceback path always starts at the all-zeros state. D must be less than or equal to the vector length of the input. If you know that each frame of data typically ends at the all-zeros state, then this mode is an appropriate choice.

\section*{Decoding Delay}

If you set Operation mode to Continuous, then this block introduces a decoding delay equal to Traceback depth*k bits for a rate \(\mathrm{k} / \mathrm{n}\) convolutional code. The decoding delay is the number of zeros that precede the first decoded bit in the output.

The block incurs no delay for other values of Operation mode.

\section*{General TCM Decoder}

Function Block Parameters: General TCM Decoder X
General TCM Decoder (mask) (link)
Use the Viterbi algorithm to decode trellis-coded modulation data, mapped using the Signal constellation parameter that expects complex constellation points in the set-partitioned order.

The Trellis structure parameter must be a valid MATLAB trellis structure. To check if a structure is a valid trellis structure, use the istrellis function in MATLAB.

Parameters
Trellis structure:
poly2trellis([1 3],[10 0; 05 2])
Signal constellation:

Traceback depth:
21
Operation mode: Continuous \(\nabla\)

Enable the reset input port
Output data type: double

Dialog Box


\section*{Trellis structure}

MATLAB structure that contains the trellis description of the convolutional encoder.

\section*{Signal constellation}

A complex vector that lists the points in the signal constellation in set-partitioned order.

\section*{General TCM Decoder}

\section*{Traceback depth}

The number of trellis branches (equivalently, the number of symbols) the block uses in the Viterbi algorithm to construct each traceback path.

\section*{Operation mode}

The operation mode of the Viterbi decoder. The choices are Continuous, Truncated, and Terminated.

\section*{Enable the reset input port}

When you select this check box, the block has a second input port labeled Rst. Providing a nonzero value to this port causes the block to set its internal memory to the initial state before processing the input data. This field appears only if you set Operation mode to Continuous.

\section*{Output data type}

Select the data type for the block output signal as boolean or single. By default, the block sets this to double.

\section*{Supported Data Types}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & • Double-precision floating point \\
& - Single-precision floating point \\
\hline Reset & \begin{tabular}{l} 
- Double-precision floating point \\
\\
- Boolean
\end{tabular} \\
\hline Output & - Double-precision floating point \\
& - Boolean \\
\hline
\end{tabular}

\section*{Pair Block}

General TCM Encoder
See Also
M-PSK TCM Decoder, Rectangular QAM TCM Decoder, poly2trellis

\section*{General TCM Decoder}

\author{
References \\ [1] Biglieri, E., D. Divsalar, P. J. McLane, and M. K. Simon, Introduction to Trellis-Coded Modulation with Applications, New York, Macmillan, 1991.
}
[2] Proakis, John G., Digital Communications, Fourth edition, New York, McGraw-Hill, 2001.

\section*{General TCM Encoder}

\section*{Purpose}

Library
Description

General
TCM

Convolutionally encode binary data and map using arbitrary constellation

TCM, in Digital Baseband sublibrary of Modulation
The General TCM Encoder block implements trellis-coded modulation (TCM) by convolutionally encoding the binary input signal and mapping the result to an arbitrary signal constellation. The Signal constellation parameter lists the signal constellation points in set-partitioned order. This parameter is a complex vector with a length, \(M\), equal to the number of possible output symbols from the convolutional encoder. (That is, \(\log _{2} M\) is equal to n for a rate \(k / n\) convolutional code.)

\section*{Input Signals and Output Signals}

If the convolutional encoder represents a rate \(k / n\) code, then the General TCM Encoder block's input must be a binary column vector with a length of \(L^{*} k\) for some positive integer \(L\).

This block accepts a binary-valued input signal. The output signal is a complex column vector of length \(L\). For information about the data types each block port supports, see "Supported Data Types" on page 2-362.

\section*{Specifying the Encoder}

To define the convolutional encoder, use the Trellis structure parameter. This parameter is a MATLAB structure whose format is described in the section "Trellis Description of a Convolutional Code" in the Communications System Toolbox documentation. You can use this parameter field in two ways:
- If you want to specify the encoder using its constraint length, generator polynomials, and possibly feedback connection polynomials, then use a poly2trellis command within the Trellis structure field. For example, to use an encoder with a constraint length of 7 , code generator polynomials of 171 and 133 (in octal numbers), and a feedback connection of 171 (in octal), set the Trellis structure parameter to

\section*{General TCM Encoder}
```

poly2trellis(7,[171 133],171)

```
- If you have a variable in the MATLAB workspace that contains the trellis structure, then enter its name as the Trellis structure parameter. This way is faster because it causes Simulink software to spend less time updating the diagram at the beginning of each simulation, compared to the usage in the previous bulleted item.

The encoder registers begin in the all-zeros state. You can configure the encoder so that it resets its registers to the all-zeros state during the course of the simulation. To do this, set the Operation mode to Reset on nonzero input via port. The block then opens a second input port, labeled Rst. The signal at the Rst port is a scalar signal. When it is nonzero, the encoder resets before processing the data at the first input port.

\section*{Signal Constellations}

The trellis-coded modulation technique partitions the constellation into subsets called cosets so as to maximize the minimum distance between pairs of points in each coset.

Note When you set the Signal constellation parameter, you must ensure that the constellation vector is already in set-partitioned order. Otherwise, the block might produce unexpected or suboptimal results.

As an example, the diagram below shows one way to devise a set-partitioned order for the points for an 8-PSK signal constellation. The figure at the top of the tree is the entire 8-PSK signal constellation, while the eight figures at the bottom of the tree contain one constellation point each. Each level of the tree corresponds to a different bit in a binary sequence ( \(\mathrm{b}_{3}, \mathrm{~b}_{2}, \mathrm{~b}_{1}\) ), while each branch in a given level of the tree corresponds to a particular value for that bit. Listing the constellation points using the sequence at the bottom of the tree leads to the vector
```

exp(2*pi*j*[0 4 2 6 1 5 3 7]/8)

```

\section*{General TCM Encoder}
which is a valid value for the Signal constellation parameter in this block.


For other examples of signal constellations in set-partitioned order, see [1] or the reference pages for the M-PSK TCM Encoder and Rectangular QAM TCM Encoder blocks.

\section*{Coding Gains}

Coding gains of 3 to 6 decibels, relative to the uncoded case can be achieved in the presence of AWGN with multiphase trellis codes [3].

\section*{General TCM Encoder}

\section*{Dialog Box}


\section*{Trellis structure}

MATLAB structure that contains the trellis description of the convolutional encoder.

\section*{Operation mode}

In Continuous mode (default setting), the block retains the encoder states at the end of each frame, for use with the next frame.

In Truncated (reset every frame) mode, the block treats each frame independently. I.e., the encoder states are reset to all-zeros state at the start of each frame.

In Terminate trellis by appending bits mode, the block treats each frame independently. For each input frame,

\section*{General TCM Encoder} state at the end of the frame. The output length is given by \(y=n \cdot(x+s) / k\), where \(x\) is the number of input bits, and \(s=\) constraint length -1 (or, in the case of multiple constraint lengths, \(s=\) sum(ConstraintLength(i) -1 )). The block supports this mode for column vector input signals.

In Reset on nonzero input via port mode, the block has an additional input port, labeled Rst. When the Rst input is nonzero, the encoder resets to the all-zeros state.

\section*{Signal constellation}

A complex vector that lists the points in the signal constellation in set-partitioned order.

\section*{Output data type}

The output type of the block can be specified as a single or double. By default, the block sets this to double.

\section*{Supported}

Data Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
& \(\bullet\) Boolean \\
& \(\bullet 8-, 16\)-, and 32 -bit signed integers \\
& \(\bullet 8-, 16-\), and 32 -bit unsigned integers \\
& \(\bullet\) ufix(1) \\
\hline Output & \(\bullet\) Double-precision floating point \\
& \(\bullet\) Single-precision floating point \\
\hline
\end{tabular}

Pair Block
General TCM Decoder

See Also
References [1] Biglieri, E., D. Divsalar, P. J. McLane, and M. K. Simon, Introduction to Trellis-Coded Modulation with Applications, New York, Macmillan, 1991.
[2] Proakis, John G., Digital Communications, Fourth edition, New York, McGraw-Hill, 2001.
[3] Ungerboeck, G., "Channel Coding with Multilevel/Phase Signals", IEEE Trans. on Information Theory, Vol IT28, Jan. 1982, pp. 55-67.

\section*{GMSK Demodulator Baseband}

\section*{Purpose Demodulate GMSK-modulated data \\ Library \\ Description \\ MMTGMSK \\ CPM, in Digital Baseband sublibrary of Modulation \\ The GMSK Demodulator Baseband block uses a Viterbi algorithm to demodulate a signal that was modulated using the Gaussian minimum shift keying method. The input to this block is a baseband representation of the modulated signal. \\ Integer-Valued Signals and Binary-Valued Signals \\ This block accepts a scalar-valued or column vector input signal with a data type of single or double. If you set the Output type parameter to Integer, then the block produces values of 1 and -1 . If you set the Output type parameter to Bit, then the block produces values of 0 and 1.}

\section*{Single-Rate Processing}

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. The input width must be an integer multiple of the Samples per symbol parameter value, and the input can be a column vector.
- When you set Output type to Bit, the output width is two times the number of input symbols.
- When you set Output type to Integer, the output width is the number of input symbols.

For a column vector input signal, the width of the input equals the product of the number of symbols and the value for the Samples per symbol parameter.

\section*{Multirate Processing}

In multirate processing mode, the input and output signals have different port sample times. The input must be a scalar. The output

\section*{GMSK Demodulator Baseband}
symbol time is the product of the input sample time and the Samples per symbol parameter value.
- When you set Output type to Bit, the output width equals the number of bits per symbol.
- When you set Output type to Integer, the output is a scalar.

\section*{Traceback Depth and Output Delays}

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The Traceback depth parameter, D , in this block is the number of trellis branches used to construct each traceback path. D influences the output delay, which is the number of zero symbols that precede the first meaningful demodulated value in the output.
- When you set the Rate options parameter to Allow multirate processing, and the model uses a variable-step solver or a fixed-step solver with the Tasking Mode parameter set to SingleTasking, then the delay consists of \(\mathrm{D}+1\) zero symbols.
- When you set the Rate options parameter to Enforce single-rate processing, then the delay consists of D zero symbols.

The optimal Traceback depth parameter value is dependent on minimum squared Euclidean distance calculations. Alternatively, a typical value, dependent on the number of states, can be chosen using the "five-times-the-constraint-length" rule, which corresponds
to \(5 \cdot \log 2\) (numStates). The number of states is determined by the following equation:
numStates \(=\left\{\begin{array}{l}p \cdot 2^{(L-1)}, \text { for even } m \\ 2 p \cdot 2^{(L-1)}, \text { for odd } m\end{array}\right\}\)
where:
- \(h=m / p\) is the modulation index in proper rational form

\section*{GMSK Demodulator Baseband}
- \(m=\) numerator of modulation index
- \(p=\) denominator of modulation index
- \(L\) is the Pulse length

\begin{tabular}{l}
\hline Function Block Parameters: GMSK Demodulator Baseband \\
\begin{tabular}{|l|l|}
\hline GMSK Demodulator Baseband (mask) (link) - \\
Demodulate the GMSK modulated input signal using the Viterbi algorithm. \\
Parameters - \\
Output type: Integer \\
BT product: \\
\hline .3 \\
Pulse length (symbol intervals): \\
\hline 4 \\
Symbol prehistory: \\
\hline 1 & Cancel \\
Phase offset (rad): \\
\hline 0 & Help \\
\hline Samples per symbol: \\
\hline 8 \\
Rate options: Enforce single-rate processing \\
Traceback depth: \\
\hline 16 \\
\hline
\end{tabular} \\
\hline Output datatype double \\
\hline
\end{tabular}

\section*{Output type}

Determines whether the output consists of bipolar or binary values.

\section*{GMSK Demodulator Baseband}

\section*{BT product}

The product of bandwidth and time.

\section*{Pulse length (symbol intervals)}

The length of the frequency pulse shape.

\section*{Symbol prehistory}

The data symbols the modulator uses before the start of the simulation.

\section*{Phase offset (rad)}

The initial phase of the modulated waveform.

\section*{Samples per symbol}

The number of input samples that represent each modulated symbol, which must be a positive integer. For more information, see "Upsample Signals and Rate Changes" in Communications System Toolbox User’s Guide.

\section*{Rate options}

Select the rate processing method for the block.
- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width is the number of symbols (which is given by dividing the input length by the Samples per symbol parameter value when the Output type parameter is set to Integer).
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output period is the same as the symbol period and equals the product of the input period and the Samples per symbol parameter value.

\section*{GMSK Demodulator Baseband}

> Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

For more information, see Single-Rate Processing and Multirate Processing in the Description section of this page.

\section*{Traceback depth}

The number of trellis branches that the GMSK Demodulator Baseband block uses to construct each traceback path.

\section*{Output data type}

The output data type can be boolean, int8, int16, int32, or double.

Supported Data Types
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & • Double-precision floating point \\
& - Single-precision floating point \\
\hline Output & • Double-precision floating point \\
& - Boolean (When Output type set to Bit) \\
& • 8-, 16-, and 32-bit signed integers (When Output type \\
& set to Integer)
\end{tabular}

\author{
Pair Block GMSK Modulator Baseband \\ See Also CPM Demodulator Baseband, Viterbi Decoder \\ References [1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. Digital Phase Modulation. New York: Plenum Press, 1986.
}

\title{
GMSK Modulator Baseband
}

\section*{Purpose}

Modulate using Gaussian minimum shift keying method

\section*{Library}

Description


CPM, in Digital Baseband sublibrary of Modulation
The GMSK Modulator Baseband block modulates using the Gaussian minimum shift keying method. The output is a baseband representation of the modulated signal.

The BT product parameter represents bandwidth multiplied by
time. This parameter is a nonnegative scalar. It is used to reduce the bandwidth at the expense of increased intersymbol interference. The Pulse length parameter measures the length of the Gaussian pulse shape, in symbol intervals. For an explanation of the pulse shape, see the work by Anderson, Aulin, and Sundberg among the references listed below. The frequency pulse shape is defined by the following equations.
\[
\begin{aligned}
& g(t)=\frac{1}{2 T}\left\{Q\left[2 \pi B_{b} \frac{t-\frac{T}{2}}{\sqrt{\ln (2)}}\right]-Q\left[2 \pi B_{b} \frac{t+\frac{T}{2}}{\sqrt{\ln (2)}}\right]\right\} \\
& Q(t)=\int_{t}^{\infty} \frac{1}{\sqrt{2 \pi}} e^{-\tau^{2} / 2} d \tau
\end{aligned}
\]

For this block, an input symbol of 1 causes a phase shift of \(п / 2\) radians.
The group delay is the number of samples between the start of a filter's response and its peak. The group delay that the block introduces is Pulse length/2 * Samples per symbol (using a reference of output sample periods). For GMSK, Pulse length denotes the truncated frequency pulse length in symbols. The net delay effect at the receiver (demodulator) is due to the Traceback depth parameter, which in most cases would be larger than the group delay.

\section*{Integer-Valued Signals and Binary-Valued Signals}

When you set the Input type parameter to Integer, then the block accepts values of 1 and -1 .

\section*{GMSK Modulator Baseband}

When you set the Input type parameter to Bit, then the block accepts values of 0 and 1 .

This block accepts a scalar-valued or column vector input signal. For a column vector input signal, the width of the output equals the product of the number of symbols and the value for the Samples per symbol parameter.

\section*{Single-Rate Processing}

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. In this mode, the input to the block can be multiple symbols.
- When you set Input type to Integer, the input can be a column vector, the length of which is the number of input symbols.
- When you set Input type to Bit, the input width must be an integer multiple of 2 .

The output width equals the product of the number of input symbols and the Samples per symbol parameter value.

\section*{Multirate Processing}

In multirate processing mode, the input and output signals have different port sample times. In this mode, the input to the block must be one symbol.
- When you set Input type to Integer, the input must be a scalar.
- When you set Input type to Bit, the input width must equal the number of bits per symbol.

The output sample time equals the symbol period divided by the Samples per symbol parameter value.

\section*{GMSK Modulator Baseband}


Dialog Box

\section*{Input type}

Indicates whether the input consists of bipolar or binary values.

\section*{BT product}

The product of bandwidth and time.
The block uses this parameter to reduce bandwidth at the expense of increased intersymbol interference. Enter a nonnegative scalar value for this parameter.

\section*{Pulse length (symbol intervals)}

The length of the frequency pulse shape.

\section*{GMSK Modulator Baseband}

\section*{Symbol prehistory}

A scalar or vector value that specifies the data symbols the block uses before the start of the simulation, in reverse chronological order. If it is a vector, then its length must be one less than the Pulse length parameter.

Phase offset (rad)
The initial phase of the output waveform, measured in radians.

\section*{Samples per symbol}

The number of output samples that the block produces for each integer or bit in the input, which must be a positive integer. For all non-binary schemes, as defined by the pulse shapes, this value must be greater than 1 .

For more information, see "Upsample Signals and Rate Changes" in Communications System ToolboxUser's Guide.

\section*{Rate options}

Select the rate processing option for the block.
- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width equals the product of the number of symbols and the Samples per symbol parameter value.
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output sample time equals the symbol period divided by the Samples per symbol parameter value.

\section*{GMSK Modulator Baseband}

Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

\section*{Output data type}

The output type of the block can be specified as a single or double. By default, the block sets this to double.

\section*{Supported Data Types}
\begin{tabular}{l|l}
\hline Port & Supported Data Types \\
\hline Input & • Double-precision floating point \\
& • Boolean (When Input type set to Bit) \\
& \begin{tabular}{l} 
• 8-, 16-, and 32-bit signed integers (When Input type \\
set to Integer)
\end{tabular} \\
\hline Output & \begin{tabular}{l} 
• Double-precision floating point \\
\\
\\
\end{tabular} \begin{tabular}{l} 
• Single-precision floating point
\end{tabular} \\
\hline
\end{tabular}

\section*{Pair Block}

GMSK Demodulator Baseband

\section*{See Also CPM Modulator Baseband}

References [1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. Digital Phase Modulation. New York: Plenum Press, 1986.

\section*{Gold Sequence Generator}
\[
\begin{array}{ll}
\text { Purpose } & \text { Generate Gold sequence from set of sequences } \\
\text { Library } & \text { Sequence Generators sublibrary of Comm Sources } \\
\text { Description } & \begin{array}{l}
\text { The Gold Sequence Generator block generates a Gold sequence. Gold } \\
\text { sequences form a large class of sequences that have good periodic } \\
\text { cross-correlation properties. }
\end{array} \\
\begin{array}{c}
\text { Gold Sequence } \\
\text { Generator }
\end{array} & \begin{array}{l}
\text { This block can output sequences that vary in length during simulation. } \\
\text { For more information about variable-size signals, see "Variable-Size } \\
\text { Signal Basics" in the Simulink documentation. }
\end{array}
\end{array}
\]

\section*{Gold Sequences}

The Gold sequences are defined using a specified pair of sequences \(u\) and \(v\), of period \(N=2^{\mathrm{n}}-1\), called a preferred pair, as defined in "Preferred Pairs of Sequences" on page 2-377 below. The set \(G(u, v)\) of Gold sequences is defined by
\[
G(u, v)=\left\{u, v, u \oplus v, u \oplus T v, u \oplus T^{2} v, \ldots, u \oplus T^{N-1} v\right\}
\]
where \(T\) represents the operator that shifts vectors cyclically to the left by one place, and \(\oplus\) represents addition modulo 2 . Note that G(u,v) contains \(N+2\) sequences of period \(N\). The Gold Sequence Generator block outputs one of these sequences according to the block's parameters.

Gold sequences have the property that the cross-correlation between any two, or between shifted versions of them, takes on one of three values: \(-t(n),-1\), or \(t(n)-2\), where
\[
t(n)= \begin{cases}1+2^{(n+1) / 2} & n \text { even } \\ 1+2^{(n+2) / 2} & n \text { odd }\end{cases}
\]

The Gold Sequence Generator block uses two PN Sequence Generator blocks to generate the preferred pair of sequences, and then XORs these

\section*{Gold Sequence Generator}
sequences to produce the output sequence, as shown in the following diagram.


You can specify the preferred pair by the Preferred polynomial [1] and Preferred polynomial [2] parameters in the dialog for the Gold Sequence Generator block. These polynomials, both of which must have degree \(n\), describe the shift registers that the PN Sequence Generator blocks use to generate their output. For more details on how these sequences are generated, see the reference page for the PN Sequence Generator block. You can specify the preferred polynomials using either of the following formats:
- A vector that lists the coefficients of the polynomial in descending order of powers. The first and last entries must be 1. Note that the length of this vector is one more than the degree of the generator polynomial.
- A vector containing the exponents of \(z\) for the nonzero terms of the polynomial in descending order of powers. The last entry must be 0 .

For example, the vectors [5 2 0] and [ \(\left.\begin{array}{lllll}1 & 0 & 0 & 1 & 0\end{array} 1\right]\) both represent the polynomial \(z^{5}+z^{2}+1\).

The following table provides a short list of preferred pairs.
\begin{tabular}{l|l|l|l}
\hline \(\mathbf{n}\) & \(\mathbf{N}\) & \begin{tabular}{l} 
Preferred \\
Polynomial[1]
\end{tabular} & \begin{tabular}{l} 
Preferred \\
Polynomial[2]
\end{tabular} \\
\hline 5 & 31 & {\(\left[\begin{array}{lll}5 & 2 & 0\end{array}\right]\)} & {\(\left[\begin{array}{lllll}5 & 4 & 3 & 2 & 0\end{array}\right]\)} \\
\hline 6 & 63 & {\(\left[\begin{array}{lll}6 & 1 & 0\end{array}\right]\)} & {\(\left[\begin{array}{lllll}6 & 5 & 2 & 1 & 0\end{array}\right]\)} \\
\hline
\end{tabular}

\section*{Gold Sequence Generator}
\begin{tabular}{l|l|l|l}
\hline \(\mathbf{n}\) & \(\mathbf{N}\) & \begin{tabular}{l} 
Preferred \\
Polynomial[1]
\end{tabular} & \begin{tabular}{l} 
Preferred \\
Polynomial[2]
\end{tabular} \\
\hline 7 & 127 & {\(\left[\begin{array}{lll}7 & 3 & 0\end{array}\right]\)} & {\(\left[\begin{array}{lllll}7 & 3 & 2 & 1 & 0\end{array}\right]\)} \\
\hline 9 & 511 & {\(\left[\begin{array}{lll}9 & 4 & 0\end{array}\right]\)} & {\(\left[\begin{array}{llll}9 & 6 & 4 & 3\end{array}\right]\)} \\
\hline 10 & 1023 & {\(\left[\begin{array}{lll}10 & 3 & 0\end{array}\right]\)} & {\(\left[\begin{array}{lllll}10 & 8 & 3 & 2 & 0\end{array}\right]\)} \\
\hline 11 & 2047 & {\(\left[\begin{array}{lll}11 & 2 & 0\end{array}\right]\)} & {\(\left[\begin{array}{lllll}11 & 8 & 5 & 2 & 0\end{array}\right]\)} \\
\hline
\end{tabular}

The Initial states[1] and Initial states[2] parameters are vectors specifying the initial values of the registers corresponding to Preferred polynomial [1] and Preferred polynomial [2], respectively. These parameters must satisfy these criteria:
- All elements of the Initial states[1] and Initial states[2] vectors must be binary numbers.
- The length of the Initial states[1] vector must equal the degree of the Preferred polynomial[1], and the length of the Initial states[2] vector must equal the degree of the Preferred polynomial[2].

Note At least one element of the Initial states vectors must be nonzero in order for the block to generate a nonzero sequence. That is, the initial state of at least one of the registers must be nonzero.

The Sequence index parameter specifies which sequence in the set \(G(u, v)\) of Gold sequences the block outputs. The range of Sequence index is \(\left[-2,-1,0,1,2, \ldots, 2^{\mathrm{n}}-2\right]\). The correspondence between Sequence index and the output sequence is given in the following table.
\begin{tabular}{l|l}
\hline Sequence Index & Output Sequence \\
\hline-2 & u \\
\hline-1 & v \\
\hline
\end{tabular}

\section*{Gold Sequence Generator}
\begin{tabular}{l|l}
\hline Sequence Index & Output Sequence \\
\hline 0 & \(u \oplus v\) \\
\hline 1 & \(u \oplus T v\) \\
\hline 2 & \(u \oplus T^{2} v\) \\
\hline\(\ldots\) & \(\ldots\) \\
\hline \(2^{\mathrm{n}-2}\) & \(u \oplus T^{2^{n}-2} v\) \\
\hline
\end{tabular}

You can shift the starting point of the Gold sequence with the Shift parameter, which is an integer representing the length of the shift.

You can use an external signal to reset the values of the internal shift register to the initial state by selecting Reset on nonzero input. This creates an input port for the external signal in the Gold Sequence Generator block. The way the block resets the internal shift register depends on whether its output signal and the reset signal are sample-based or frame-based. The following example demonstrates the possible alternatives. See "Example: Resetting a Signal" on page 2-723 for an example.

\section*{Preferred Pairs of Sequences}

The requirements for a pair of sequences \(u, v\) of period \(\mathrm{N}=2^{\mathrm{n}}-1\) to be a preferred pair are as follows:
- n is not divisible by 4
- \(v=u[q]\), where
- \(q\) is odd
- \(\mathrm{q}=2^{\mathrm{k}}+1\) or \(\mathrm{q}=2^{2 \mathrm{k}}-2^{\mathrm{k}}+1\)
- \(v\) is obtained by sampling every \(q\) th symbol of \(u\)

\section*{Gold Sequence Generator}
- \(\operatorname{gcd}(n, k)= \begin{cases}1 & n \equiv 1 \bmod 2 \\ 2 & n \equiv 2 \bmod 4\end{cases}\)

\section*{Gold Sequence Generator}
```

Source Block Parameters: Gold Sequence Generator
Gold Sequence Generator
Generate a Gold sequence from a set of sequences by specifying a preferred pair of polynomials.
The polynomial parameter values represent the shift register connections. Enter these values as either a binary vector or a descending ordered polynomial to indicate the connection points.
The sequence index parameter denotes the single sequence outputted from the set of Gold sequences. Specify it as a scalar integer in the range [ -2 , $\left.2^{\wedge} n-2\right]$ where $n$ is the degree of the generator polynomial. The index values -2 and -1 correspond to the first and second PN sequences as generated by the preferred polynomials (1) and (2), respectively.
For variable-size output signals, the current output size is either specified from the 'oSiz' input or inherited from the 'Ref' input.
Parameters
Preferred polynomial (1): [10 000000111$]$
Initial states (1): [ $\left.0 \begin{array}{lllll}0 & 0 & 0 & 0 & 0\end{array}\right]$
Preferred polynomial (2): [11 10001111$]$
Initial states (2): [ $00000 c c c c c c]$
Sequence index:
Shift: 0
$\square$ Output variable-size signals
Sample time: 1

```
```Frame-based outputs
```

```Reset on nonzero input
Output data type: double
```


## Preferred polynomial[1]

Vector specifying the polynomial for the first sequence of the preferred pair.

## Gold Sequence Generator

## Initial states[1]

Vector of initial states of the shift register for the first sequence of the preferred pair.

## Preferred polynomial[2]

Vector specifying the polynomial for the second sequence of the preferred pair.

## Initial states[2]

Vector of initial states of the shift register for the second sequence of the preferred pair.

## Sequence index

Integer specifying the index of the output sequence from the set of sequences.

## Shift

Integer scalar that determines the offset of the Gold sequence from the initial time.

## Output variable-size signals

Select this check box if you want the output sequences to vary in length during simulation. The default selection outputs fixed-length signals.

## Maximum output size source

Specify how the block defines maximum output size for a signal.

- When you select Dialog parameter, the value you enter in the Maximum output size parameter specifies the maximum size of the output. When you make this selection, the oSiz input port specifies the current size of the output signal and the block output inherits sample time from the input signal. The input value must be less than or equal to the Maximum output size parameter.
- When you select Inherit from reference port, the block output inherits sample time, maximum size, and current size from the variable-sized signal at the Ref input port.


## Gold Sequence Generator

This parameter only appears when you select Output variable-size signals. The default selection is Dialog parameter.

## Maximum output size

Specify a two-element row vector denoting the maximum output size for the block. The second element of the vector must be 1 For example, [10 1] gives a 10-by-1 maximum sized output signal. This parameter only appears when you select Output variable-size signals.

## Sample time

Period of each element of the output signal.

## Frame-based outputs

Determines whether the output is frame-based or sample-based.

## Samples per frame

The number of samples in a frame-based output signal. This field is active only if you select Frame-based outputs.

## Reset on nonzero input

When selected, you can specify an input signal that resets the internal shift registers to the original values of the Initial states parameter

## Output data type

The output type of the block can be specified as boolean, double or Smallest unsigned integer. By default, the block sets this to double.

When the parameter is set to Smallest unsigned integer, the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in the Hardware Implementation pane, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

See Also Kasami Sequence Generator, PN Sequence Generator

## References <br> [1] Proakis, John G., Digital Communications, Third edition, New York,

 McGraw Hill, 1995.[2] Gold, R., "Maximal Recursive Sequences with 3 -valued Recursive Cross-Correlation Functions," IEEE Trans. Infor. Theory, Jan., 1968, pp. 154-156.
[3] Gold, R., "Optimal Binary Sequences for Spread Spectrum Multiplexing, IEEE Trans. Infor. Theory, Oct., 1967, pp. 619-621.
[4] Sarwate, D.V., and M.B. Pursley, "Crosscorrelation Properties of Pseudorandom and Related Sequences," Proc. IEEE, Vol. 68, No. 5, May, 1980, pp. 583-619.
[5] Dixon, Robert, Spread Spectrum Systems with Commercial Applications, Third Edition, Wiley-Interscience, 1994.

## Purpose

Generate Hadamard code from orthogonal set of codes

## Library

Description

Hadamard Code Generator

Sequence Generators sublibrary of Comm Sources
The Hadamard Code Generator block generates a Hadamard code from a Hadamard matrix, whose rows form an orthogonal set of codes. Orthogonal codes can be used for spreading in communication systems
in which the receiver is perfectly synchronized with the transmitter. In these systems, the despreading operation is ideal, as the codes are decorrelated completely.

The Hadamard codes are the individual rows of a Hadamard matrix. Hadamard matrices are square matrices whose entries are +1 or -1 , and whose rows and columns are mutually orthogonal. If N is a nonnegative power of 2, the N-by-N Hadamard matrix, denoted $\mathrm{H}_{\mathrm{N}}$, is defined recursively as follows.

$$
\begin{aligned}
H_{1} & =[1] \\
H_{2 N} & =\left[\begin{array}{cc}
H_{N} & H_{N} \\
H_{N} & -H_{N}
\end{array}\right]
\end{aligned}
$$

The N-by-N Hadamard matrix has the property that

$$
\mathrm{H}_{\mathrm{N}} \mathrm{H}_{\mathrm{N}}{ }^{\mathrm{T}}=\mathrm{NI}_{\mathrm{N}}
$$

where $\mathrm{I}_{\mathrm{N}}$ is the N -by- N identity matrix.
The Hadamard Code Generator block outputs a row of $\mathrm{H}_{\mathrm{N}}$. The output is bipolar. You specify the length of the code, N , by the Code length parameter. The Code length must be a power of 2 . You specify the index of the row of the Hadamard matrix, which is an integer in the range $[0,1, \ldots, \mathrm{~N}-1]$, by the Code index parameter.

## Hadamard Code Generator

## Dialog

Box


## Code length

A positive integer that is a power of two specifying the length of the Hadamard code.

## Code index

An integer between 0 and $\mathrm{N}-1$, where N is the Code length, specifying a row of the Hadamard matrix.

## Sample time

A positive real scalar specifying the sample time of the output signal.

## Frame-based outputs

Determines whether the output is frame-based or sample-based.

## Samples per frame

The number of samples in a frame-based output signal. This field is active only if you select Frame-based outputs.

## Output data type

The output type of the block can be specified as an int8 or double. By default, the block sets this to double.

See Also<br>OVSF Code Generator, Walsh Code Generator

## Hamming Decoder

| Purpose | Decode Hamming code to recover binary vector data |
| :--- | :--- |
| Library | Block sublibrary of Error Detection and Correction |

Description The Hamming Decoder block recovers a binary message vector from a binary Hamming codeword vector. For proper decoding, the parameter values in this block should match those in the corresponding Hamming Encoder block.

If the Hamming code has message length $K$ and codeword length $N$, then $N$ must have the form $2^{\mathrm{M}}-1$ for some integer $M$ greater than or equal to 3 . Also, $K$ must equal $N-M$.

This block accepts a column vector input signal of length $N$. The output signal is a column vector of length $K$.
The coding scheme uses elements of the finite field GF $\left(2^{\mathrm{M}}\right)$. You can either specify the primitive polynomial that the algorithm should use, or you can rely on the default setting:

- To use the default primitive polynomial, simply enter $N$ and $K$ as the first and second dialog parameters, respectively. The algorithm uses gfprimdf( $M$ ) as the primitive polynomial for $\mathrm{GF}\left(2^{\mathrm{M}}\right)$.
- To specify the primitive polynomial, enter $N$ as the first parameter and a binary vector as the second parameter. The vector represents the primitive polynomial by listing its coefficients in order of ascending exponents. You can create primitive polynomials using the Communications System Toolbox gfprimfd function.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-388 table on this page.

## Hamming Decoder

## Dialog <br> Box



## Codeword length $\mathbf{N}$

The codeword length $N$, which is also the input vector length.

## Message length K, or M-degree primitive polynomial

Either the message length, which is also the output vector length; or a binary vector that represents a primitive polynomial for $\mathrm{GF}\left(2^{\mathrm{M}}\right)$.

## Hamming Decoder

## Supported Data Type

| Port | Supported Data Types |
| :---: | :---: |
| In | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean <br> - 8 -, 16-, and 32 -bit signed integers <br> - 8 -, 16 -, and 32 -bit unsigned integers <br> - Fixed-point |
| Out | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean <br> - 8 -, 16-, and 32 -bit signed integers <br> - 8 -, 16 -, and 32 -bit unsigned integers <br> - Fixed-point |

Pair Block Hamming Encoder<br>See Also hammgen (Communications System Toolbox)

## Purpose

Create Hamming code from binary vector data

## Library

Description
Block sublibrary of Error Detection and Correction
The Hamming Encoder block creates a Hamming code with message length $K$ and codeword length $N$. The number $N$ must have the form $2^{\mathrm{M}}-1$, where $M$ is an integer greater than or equal to 3 . Then $K$ equals $N-M$.

This block accepts a column vector input signal of length $K$. The output signal is a column vector of length $N$.

The coding scheme uses elements of the finite field GF $\left(2^{\mathrm{M}}\right)$. You can either specify the primitive polynomial that the algorithm should use, or you can rely on the default setting:

- To use the default primitive polynomial, simply enter $N$ and $K$ as the first and second dialog parameters, respectively. The algorithm uses gfprimdf(M) as the primitive polynomial for $\mathrm{GF}\left(2^{\mathrm{M}}\right)$.
- To specify the primitive polynomial, enter $N$ as the first parameter and a binary vector as the second parameter. The vector represents the primitive polynomial by listing its coefficients in order of ascending exponents. You can create primitive polynomials using the Communications System Toolbox gfprimfd function.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-390 table on this page.

## Hamming Encoder

## Dialog <br> Box



## Codeword length $\mathbf{N}$

The codeword length, which is also the output vector length.

## Message length K, or M-degree primitive polynomial

Either the message length, which is also the input vector length; or a binary vector that represents a primitive polynomial for $\mathrm{GF}\left(2^{\mathrm{M}}\right)$.

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | $\bullet$ Double-precision floating point |
|  | • Single-precision floating point |
|  | • Boolean |
|  | • 8-, 16-, and 32 -bit signed |
|  | integers |

## Hamming Encoder

| Port | Supported Data Types |
| :---: | :---: |
|  | - 8 -, 16 -, and 32 -bit unsigned integers <br> - Fixed-point |
| Out | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean <br> - 8-, 16-, and 32 -bit signed integers <br> - 8-, 16-, and 32 -bit unsigned integers <br> - Fixed-point |

## Pair Block

Hamming Decoder
See Also
hammgen (Communications System Toolbox)

## Helical Deinterleaver

Purpose
Library
Description

Restore ordering of symbols permuted by helical interleaver
Convolutional sublibrary of Interleaving
The Helical Deinterleaver block permutes the symbols in the input signal by placing them in an array row by row and then selecting groups in a helical fashion to send to the output port.

The block uses the array internally for its computations. If $C$ is the Number of columns in helical array parameter, then the array has $C$ columns and unlimited rows. If $N$ is the Group size parameter, then the block accepts an input of length $C \cdot N$ at each time step and inserts them into the next $N$ rows of the array. The block also places the Initial condition parameter into certain positions in the top few rows of the array (not only to accommodate the helical pattern but also to preserve the vector indices of symbols that pass through the Helical Interleaver and Helical Deinterleaver blocks in turn).

The output consists of consecutive groups of $N$ symbols. Counting from the beginning of the simulation, the block selects the kth output group in the array from column $k$ mod $C$. The selection is helical because of the reduction modulo C and because the first symbol in the $k^{\text {th }}$ group is in row $1+(k-1)^{*}$ s, where $s$ is the Helical array step size parameter.

This block accepts a column vector input signal containing $C \cdot N$ elements.

The block can accept the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The data type of this output will be the same as that of the input signal.

## Delay of Interleaver-Deinterleaver Pair

After processing a message with the Helical Interleaver block and the Helical Deinterleaver block, the deinterleaved data lags the original message by

$$
C N\left\lceil\frac{s(C-1)}{N}\right\rceil
$$

## Helical Deinterleaver

samples. Before this delay elapses, the deinterleaver output is either the Initial condition parameter in the Helical Deinterleaver block or the Initial condition parameter in the Helical Interleaver block.

If your model incurs an additional delay between the interleaver output and the deinterleaver input, then the restored sequence lags the original sequence by the sum of the additional delay and the amount in the formula above. For proper synchronization, the delay between the interleaver and deinterleaver must be $\mathrm{m} \cdot C \cdot N$ for some nonnegative integer $m$. You can use the DSP System Toolbox Delay block to adjust delays manually, if necessary.

## Helical Deinterleaver

Dialog Box


## Number of columns in helical array

The number of columns, $C$, in the helical array.

## Group size

The size, $N$, of each group of symbols. The input width is $C$ times $N$.

## Helical array step size

The number of rows of separation between consecutive output groups as the block selects them from their respective columns of the helical array.

## Initial conditions

A scalar that fills the array before the first input is placed.

Pair Block Helical Interleaver<br>See Also General Multiplexed Deinterleaver<br>References [1] Berlekamp, E. R. and P. Tong. "Improved Interleavers for Algebraic Block Codes." U. S. Patent 4559625, Dec. 17, 1985.

## Helical Interleaver

Purpose
Library
Description

Helical Interleaver

Permute input symbols using helical array
Convolutional sublibrary of Interleaving
The Helical Interleaver block permutes the symbols in the input signal by placing them in an array in a helical fashion and then sending rows of the array to the output port.

The block uses the array internally for its computations. If $C$ is the Number of columns in helical array parameter, then the array has $C$ columns and unlimited rows. If $N$ is the Group size parameter, then the block accepts an input of length $C \cdot N$ at each time step and partitions the input into consecutive groups of $N$ symbols. Counting from the beginning of the simulation, the block places the $k^{\text {th }}$ group in the array along column $k \bmod C$. The placement is helical because of the reduction modulo $C$ and because the first symbol in the $k^{\text {th }}$ group is in row $1+(k-1) \cdot s$, where $s$ is the Helical array step size parameter. Positions in the array that do not contain input symbols have default contents specified by the Initial condition parameter.

The block sends $C \cdot N$ symbols from the array to the output port by reading the next $N$ rows sequentially. At a given time step, the output symbols might be the Initial condition parameter value, symbols from that time step's input vector, or symbols left in the array from a previous time step.
This block accepts a column vector input signal containing $C \cdot N$ elements.

The block can accept the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The data type of this output will be the same as that of the input signal.

## Helical Interleaver

Function Block Parameters: Helical Interleaver X Helical Interleaver (mask) (link)

Permute input vector using a helical array with C columns. The input to the helical interleaver must have width $\mathrm{C}^{*} \mathrm{~N}$. The block processes the input in groups of size N and assigns an index to each group, beginning with $\mathrm{k}=1$ at the start of the simulation. The $k$ th group of N symbols is entered sequentially down column k mod $C$ of the helical array and beginning in row $1+(k-1)^{*} s$, where $s$ is the helical array step size. The helical interleaver output is then read row-by-row from the helical array.

The helical array step size must be a nonnegative integer and the initial condition must be a scalar.

Parameters
Number of columns in helical array:
5

Group size:
4
Helical array step size:
1
Initial conditions:
0

Dialog Box


## Number of columns in helical array

The number of columns, C, in the helical array.

## Group size

The size, N, of each group of input symbols. The input width is C times N .

## Helical Interleaver

## Helical array step size

The number of rows of separation between consecutive input groups in their respective columns of the helical array.

## Initial conditions

A scalar that fills the array before the first input is placed.
Examples Suppose that $\mathrm{C}=3, \mathrm{~N}=2$, the Helical array step size parameter is 1 , and the Initial condition parameter is -1. After receiving inputs of [1:6]', [7:12]', and [13:18]', the block's internal array looks like the schematic below. The coloring of the inputs and the array indicate how the input symbols are placed within the array. The outputs at the first three time steps are $[1 ;-1 ;-1 ; 2 ; 3 ;-1],[7 ; 4 ; 5 ; 8 ; 9 ; 6]$, and $[13 ; 10 ; 11 ; 14 ; 15 ; 12]$. (The outputs are not color-coded in the schematic.)


Pair Block Helical Deinterleaver<br>See Also General Multiplexed Interleaver

## Helical Interleaver

## References

[1] Berlekamp, E. R. and P. Tong. "Improved Interleavers for Algebraic Block Codes." U. S. Patent 4559625, Dec. 17, 1985.

| Purpose | Shape input signal using ideal rectangular pulses |
| :--- | :--- |
| Library | Comm Filters |

Description Pulse Filter

The Ideal Rectangular Pulse Filter block upsamples and shapes the input signal using rectangular pulses. The block replicates each input sample N times, where N is the Pulse length parameter. After replicating input samples, the block can also normalize the output signal and/or apply a linear amplitude gain.

If the Pulse delay parameter is nonzero, then the block outputs that number of zeros at the beginning of the simulation, before starting to replicate any of the input values.

This block accepts a scalar, column vector, or matrix input signal. For information about the data types each block port supports, see the "Supported Data Type" on page 2-408 table on this page.

The vector size, the pulse length, and the pulse delay are mutually independent. They do not need to satisfy any conditions with respect to each other.

## Single-Rate Processing

When you set the Rate options parameter to Enforce single-rate processing, the input and output of the block have the same sample rate. To generate the output while maintaining the input sample rate, the block resamples the data in each column of the input such that the frame size of the output $\left(M_{o}\right)$ is $L$ times larger than that of the input ( $M_{o}=M_{i}^{*} L$ ), where $L$ is the Pulse length (number of samples) parameter value.

## Multirate Processing

When you set the Rate options parameter to Allow multirate processing, the input and output of the block are the same size. However, the sample rate of the output is $L$ times faster than that of the input (i.e. the output sample time is $1 / \mathrm{N}$ times the input sample time). When the block is in multirate processing mode, you must also specify a value for the Input processing parameter:

## Ideal Rectangular Pulse Filter

- When you set the Input processing parameter to Elements as channels (sample based), the block treats an $M$-by- $N$ matrix input as $M^{*} N$ independent channels, and processes each channel over time. The output sample period ( $T_{s o}$ ) is $L$ times shorter than the input sample period ( $T_{s o}=T_{s i} / L$ ), while the input and output sizes remain identical.
- When you set the Input processing parameter to Columns as channels (frame based), the block treats an $M_{i}$-by- $N$ matrix input as $N$ independent channels. The block processes each column of the input over time by keeping the frame size constant $\left(M_{i}=M_{o}\right)$, while making the output frame period $\left(T_{f o}\right) L$ times shorter than the input frame period ( $T_{f o}=T_{f i} / L$ ).


## Normalization Methods

You determine the block's normalization behavior using the Normalize output signal and Linear amplitude gain parameters.

- If you clear Normalize output signal, then the block multiplies the set of replicated values by the Linear amplitude gain parameter. This parameter must be a scalar.
- If you select Normalize output signal, then the Normalization method parameter appears. The block scales the set of replicated values so that one of these conditions is true:
- The sum of the samples in each pulse equals the original input value that the block replicated.
- The energy in each pulse equals the energy of the original input value that the block replicated. That is, the sum of the squared samples in each pulse equals the square of the input value.

After the block applies the scaling specified in the Normalization method parameter, it multiplies the scaled signal by the constant scalar value specified in the Linear amplitude gain parameter.

## Ideal Rectangular Pulse Filter

The output is scaled by $\sqrt{N}$. If the output of this block feeds the input to the AWGN Channel block, specify the AWGN signal power parameter to be $1 / \mathrm{N}$.

Dialog
Box


## Pulse length (number of samples)

The number of samples in each output pulse; that is, the number of times the block replicates each input value when creating the output signal.

## Pulse delay (number of samples)

The number of zeros that appear in the output at the beginning of the simulation, before the block replicates any input values.

## Input processing

Specify how the block processes the input signal. You can set this parameter to one of the following options:

- Columns as channels (frame based) - When you select this option, the block treats each column of the input as a separate channel.
- Elements as channels (sample based) - When you select this option, the block treats each element of the input as a separate channel.

Note The Inherited (this choice will be removed - see release notes) option will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

## Rate options

Specify the method by which the block should upsample and shape the input signal. You can select one of the following options:

- Enforce single-rate processing - When you select this option, the block maintains the input sample rate, and processes the signal by increasing the output frame size by a factor of $L$. To select this option, you must set the Input processing parameter to Columns as channels (frame based).


## Ideal Rectangular Pulse Filter

- Allow multirate processing - When you select this option, the block processes the signal such that the output sample rate is $L$ times faster than the input sample rate.


## Normalize output signal

If you select this, then the block scales the set of replicated values before applying the linear amplitude gain.

## Normalization method

The quantity that the block considers when scaling the set of replicated values. Choices are Sum of samples and Energy per pulse. This field appears only if you select Normalize method.

## Linear amplitude gain

A positive scalar used to scale the output signal.


## Rounding mode

Use this parameter to specify the rounding method to be used when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. The filter coefficients do not obey this parameter; they always round to Nearest.

For more information, see Rounding Modes in the DSP System Toolbox documentation or "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

## Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

## Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator). See "Filter Structure Diagrams" in DSP System Toolbox Reference Guide for illustrations depicting the use of the coefficient data types in this block:

- When you select Same word length as input, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Specify word length, you are able to enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the coefficients, in bits. If applicable, you are able to enter separate fraction lengths for the numerator and denominator coefficients.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the coefficients. If applicable, you are able to enter separate slopes for the numerator and denominator coefficients. This block requires power-of-two slope and a bias of zero.
- The filter coefficients do not obey the Rounding mode and the Overflow mode parameters; they are always saturated and rounded to Nearest.


## Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" in DSP System Toolbox Reference Guide for illustrations depicting the use of the product output data type in this block:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the product output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.


## Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as product output, these characteristics match those of the product output.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.


## Ideal Rectangular Pulse Filter

## Output

Choose how you specify the output word length and fraction length:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as accumulator, these characteristics match those of the accumulator.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool
Select this check box to prevent any fixed-point scaling you specify in the block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.


## Examples

| Port | Supported Data Types |
| :--- | :--- |
| In | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Fixed-point |
| Out | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed-point |

If Pulse length is 4 and Pulse delay is the scalar 3, then the table below shows how the block treats the beginning of a ramp (1, 2, 3,...) in several situations. (The values shown in the table do not reflect vector sizes but merely indicate numerical values.)

| Normalization Method, If Any | Linear Amplitude Gain | First Several Output Values |
| :---: | :---: | :---: |
| None (Normalize output signal cleared) | 1 | $\begin{aligned} & 0,0,0,1,1,1,1,2,2, \\ & 2,2,3,3,3,3, \ldots \end{aligned}$ |
| None (Normalize output signal cleared) | 10 | $\begin{aligned} & 0,0,0,10,10,10,10, \\ & 20,20,20,20,30,30, \\ & 30,30, \ldots \end{aligned}$ |
| Sum of samples | 1 | $\begin{aligned} & 0,0,0,0.25,0.25, \\ & 0.25,0.25,0.5,0.5, \\ & 0.5,0.5,0.75,0.75, \\ & 0.75,0.75, \ldots, \text { where } \\ & 0.25 * 4=1 \end{aligned}$ |
| Sum of samples | 10 | $\begin{aligned} & \begin{array}{l} 0,0,0,2.5,2.5,2.5, \\ 2.5,5,5,5,5,7.5, \\ 7.5,7.5,7.5, \ldots \end{array} \end{aligned}$ |
| Energy per pulse | 1 | $\begin{aligned} & 0,0,0,0.5,0.5, \\ & 0.5,0.5,1.0,1.0 \\ & 1.0,1.0,1.5,1.5, \\ & 1.5,1.5, \ldots, \text { where } \\ & (0.5)^{\wedge} 2 * 4=1 \wedge 2 \end{aligned}$ |
| Energy per pulse | 10 | $\begin{aligned} & 0,0,0,5,5,5,5,10 \\ & 10,10,10,15,15,15, \\ & 15, \ldots \end{aligned}$ |

See Also Upsample, Integrate and Dump

## Purpose Distribute input elements in output vector

Library
Description

## Sequence Operations

The Insert Zero block constructs an output vector by inserting zeros among the elements of the input vector. The input signal can be real or complex. Both the input signal and the Insert zero vector parameter are column vector signals. The number of 1 s in the Insert zero vector parameter must be evenly divisible by the input data length. If the input vector length is greater than the number of 1 s in the Insert zero vector parameter, then the block repeats the insertion pattern until it has placed all input elements in the output vector.

The block determines where to place the zeros by using the Insert zero vector parameter.

- For each 1 the block places the next element of the input vector in the output vector
- For each 0 the block places a 0 in the output vector

The block accepts the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.

To implement punctured coding using the Puncture and Insert Zero blocks, use the same vector for the Insert zero vector parameter in this block and for the Puncture vector parameter in the Puncture block.

## Dialog Box

## Examples



## Insert zero vector

A binary vector with a pattern of 0 s and 1 s that indicate where the block places either 0s or input vector elements in the output vector.

If the Insert zero vector parameter is the six-element vector [ $1 ; 0 ; 1 ; 1 ; 1 ; 0]$, then the block inserts zeros after the first and last elements of each consecutive grouping of four input elements. It considers groups of four elements because the Insert zero vector parameter has four 1 s .

The diagram below depicts the block's operation using this Insert zero vector parameter. Notice that the insertion pattern applies twice.


Compare this example with that on the reference page for the Puncture block.

See Also
Puncture

## Integer-Input RS Encoder

## Purpose <br> Library <br> Description <br> $\underset{\text { RS encoder }}{\mathrm{ER}}=\mathrm{l}$

Create Reed-Solomon code from integer vector data
Block sublibrary of Error Detection and Correction
The Integer-Input RS Encoder block creates a Reed-Solomon code with message length, K , and codeword length, ( N - number of punctures). You specify both N and K directly in the block dialog. The symbols for the code are integers between 0 and $2^{\mathrm{M}}-1$, which represent elements of the finite field $\mathrm{GF}\left(2^{\mathrm{M}}\right)$. Restrictions on M and N are described in "Restrictions on M and the Codeword Length N" on page 2-415 below. The difference N - K must be an even integer.

This block can output shortened codewords when N and K are appropriately specified. To specify output codewords that are shortened by a length $\mathrm{S}, \mathrm{N}$ and K must be specified in the dialog box as $\mathrm{N}_{\text {full }}-\mathrm{S}$ and $\mathrm{K}_{\text {full }}-\mathrm{S}$, where $\mathrm{N}_{\text {full }}$ and $\mathrm{K}_{\text {full }}$ are the N and K of an unshortened code. If $\mathrm{S}<\left(\mathrm{N}_{\text {full }}+1\right) / 2$, the encoder can automatically determine the value of $\mathrm{N}_{\text {full }}$ and $\mathrm{K}_{\text {full }}$. However, if $\mathrm{S} \geq\left(\mathrm{N}_{\text {full }}+1\right) / 2$, Primitive polynomial must be specified in order to properly define the extension field for the code.

The input and output are integer-valued signals that represent messages and codewords, respectively. This block accepts a column vector input signal with a length that is an integer multiple of $K$. The column vector output, with a length that is the same integer multiple of N , inherits its data type from the input signal. For information about the data types each block port supports, see the "Supported Data Type" on page 2-419 table on this page.

For more information on representing data for Reed-Solomon codes, see the section "Integer Format (Reed-Solomon Only)" in Communications System Toolbox User's Guide.

If the encoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

The default value of M is the smallest integer that is greater than or equal to $\log 2(N+1)$, that is, ceil $(\log 2(N+1))$. You can change the

## Integer-Input RS Encoder

value of M from the default by specifying the primitive polynomial for GF $\left(2^{\mathrm{M}}\right)$, as described in "Specifying the Primitive Polynomial" on page 2-415 below. If N is less than $2^{\mathrm{M}}-1$, the block uses a shortened Reed-Solomon code.

An (N, K) Reed-Solomon code can correct up to floor ( (N-K)/2) symbol errors (not bit errors) in each codeword.

## Specifying the Primitive Polynomial

You can specify the primitive polynomial that defines the finite field $\mathrm{GF}\left(2^{\mathrm{M}}\right)$, corresponding to the integers that form messages and codewords. To do so, first select Specify primitive polynomial. Then, in the Primitive polynomial field, enter a binary row vector that represents a primitive polynomial over $\mathrm{GF}(2)$ of degree M , in descending order of powers. For example, to specify the polynomial $x^{3}+x+1$, enter the vector $\left[\begin{array}{llll}1 & 0 & 1 & 1\end{array}\right]$.

If you do not select Specify primitive polynomial, the block uses the default primitive polynomial of degree $\mathrm{M}=$ ceil $(\log 2(N+1))$. You can display the default polynomial by entering primpoly (ceil (log2 $(\mathrm{N}+1))$ ) at the MATLAB prompt.

## Restrictions on $\mathbf{M}$ and the Codeword Length $\mathbf{N}$

The restrictions on the degree M of the primitive polynomial and the codeword length N are as follows:

- If you do not select Specify primitive polynomial, N must lie in the range $3<\mathrm{N} \leq 2^{16}-1$.
- If you do select Specify primitive polynomial, N must lie in the range $3 \leq \mathrm{N} \leq 2^{\mathrm{M}}-1$ and M must lie in the range $3 \leq \mathrm{M} \leq 16$.


## Specifying the Generator Polynomial

You can specify the generator polynomial for the Reed-Solomon code. To do so, first select Specify generator polynomial. Then, in the Generator polynomial field, enter an integer row vector whose entries are between 0 and $2^{\mathrm{M}}-1$. The vector represents a polynomial, in descending order of powers, whose coefficients are elements of

## Integer-Input RS Encoder

$\mathrm{GF}\left(2^{\mathrm{M}}\right)$ represented in integer format. See the section "Integer Format (Reed-Solomon Only)" for more information about integer format. The generator polynomial must be equal to a polynomial with a factored form

$$
\mathrm{g}(\mathrm{x})=\left(\mathrm{x}+\mathrm{A}^{\mathrm{b}}\right)\left(\mathrm{x}+\mathrm{A}^{\mathrm{b}+1}\right)\left(\mathrm{x}+\mathrm{A}^{\mathrm{b}+2}\right) \ldots\left(\mathrm{x}+\mathrm{A}^{\mathrm{b}+\mathrm{N}-\mathrm{K}-1}\right)
$$

where A is the primitive element of the Galois field over which the input message is defined, and $b$ is an integer.

If you do not select Specify generator polynomial, the block uses the default generator polynomial, corresponding to $b=1$, for Reed-Solomon encoding. You can display the default generator polynomial by entering rsgenpoly ( $\mathrm{N} 1, \mathrm{~K} 1$ ), where $\mathrm{N} 1=2^{\wedge} \mathrm{M}-1$ and $\mathrm{K} 1=\mathrm{K}+(\mathrm{N} 1-\mathrm{N})$, at the MATLAB prompt, if you are using the default primitive polynomial. If the Specify primitive polynomial box is selected, and you specify the primitive polynomial specified as poly, the default generator polynomial is rsgenpoly ( $\mathrm{N} 1, \mathrm{~K} 1$, poly).

## Puncture Codes

The block supports puncturing when you select the Puncture code parameter. This enables the Puncture vector parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 represents that the data symbol passes unaltered, and 0 represents that the data symbol gets punctured, or removed, from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

Examples Suppose $\mathrm{M}=3, \mathrm{~N}=2^{3}-1=7$, and $\mathrm{K}=5$. Then a message is a vector of length 5 whose entries are integers between 0 and 7 . A corresponding codeword is a vector of length 7 whose entries are integers between 0 and 7. The following figure illustrates possible input and output signals to this block when Codeword length $\mathbf{N}$ is set to 7, Message length $\mathbf{K}$ is set to 5 , and the default primitive and generator polynomials are used.


Dialog Box


## Integer-Input RS Encoder

## Codeword length $\mathbf{N}$

The codeword length.

## Message length $K$

The message length.

## Specify primitive polynomial

Selecting this check box enables the Primitive polynomial parameter.

## Primitive polynomial

Binary row vector representing the primitive polynomial in descending order of powers. When you provide a Primitive polynomial, the number of input bits must be an integer multiple of $K$ times the order of the Primitive polynomial instead.

This parameter applies when only when you select Specify primitive polynomial.

## Specify generator polynomial

Selecting this check box enables the Generator polynomial parameter.

## Generator polynomial

This field is available only when Specify generator polynomial is selected.

Integer row vector, whose entries are in the range from 0 to $2^{\mathrm{M}}-1$, representing the generator polynomial in descending order of powers. Each coefficient is an element of the Galois field defined by the primitive polynomial.

## Puncture code

Selecting this check box enables the Puncture vector parameter.

## Puncture vector

This field is available only when Puncture code is selected.

## Integer-Input RS Encoder

A column vector of length $N-K$. In a puncture vector, 1 represents that the data symbol passes unaltered, and 0 represents that the data symbol gets punctured, or removed, from the data stream.

The default value is $[$ ones $(2,1) ; \operatorname{zeros}(2,1)]$.
This parameter applies only when you select Puncture code.

## Supported Data Type

| Port | Supported Data Types |
| :---: | :---: |
| In | - Double-precision floating point <br> - Single-precision floating point <br> - 8 -, 16 -, and 32 -bit signed integers <br> - 8-, 16-, and 32 -bit unsigned integers |
| Out | - Double-precision floating point <br> - Single-precision floating point <br> - 8 -, 16 -, and 32 -bit signed integers <br> - 8-, 16-, and 32 -bit unsigned integers |

Pair Block<br>See Also<br>Binary-Input RS Encoder

## Integer-Input RS Encoder HDL Optimized

Purpose Encode data using a Reed-Solomon encoder
Library Block sublibrary of Error Correction and Detection

## Description

nteger-Input
RS Encoder
HDL Optimized
Integer-Input RS Encoder HDL Optimized

Reed-Solomon encoding follows the same standards as any other cyclic redundancy code. The Integer-Input RS Encoder HDL Optimized block can be used to model many communication system Forward Error Correcting (FEC) codes.

For more about the Reed-Solomon encoder, see the Integer-Input RS Encoder block reference. For more information on representing data for Reed-Solomon codes, see "Integer Format (Reed-Solomon Only)".

## Signal Attributes

The Integer-Input RS Encoder HDL Optimized block has four input ports and four output ports.

## Integer-Input RS Encoder HDL Optimized



| Port | Direction | Description | Data <br> Type |
| :--- | :--- | :--- | :--- |
| datain | Input | Message data. Data width is <br> less than or equal to the CRC <br> length, and the CRC length <br> should be divisible by the <br> data width. For example, for <br> CRC-CCITT/CRC-16, the valid <br> data widths are 16, 8, 4, 2, and 1. | Column <br> vector of <br> double, <br> Boolean, <br> or ufix1 |
| start | Input | Indicates the start of a frame of <br> data. | Boolean <br> or ufix1 |
| end | Input | Indicates the end of a frame of <br> data. | Boolean <br> or ufix1 |
| valid | Input | Indicates that input data is <br> valid. | Boolean <br> or ufix1 |

## Integer-Input RS Encoder HDL Optimized

| Port | Direction | Description | Data <br> Type |
| :--- | :--- | :--- | :--- |
| dataout | Output | Message data with the checksum <br> appended. The data width is the <br> same as the input data port. | Column <br> vector of <br> double, <br> Boolean, <br> or ufix1 |
| startout | Output | Indicates the start of a frame of <br> data. | Boolean <br> or ufix1 |
| endout | Output | Indicates the end of a frame of <br> data, including checksum. | Boolean <br> or ufix1 |
| validout | Output | Indicates that output data is <br> valid. | Boolean <br> or ufix1 |

## Limitations

- The length of the code word $N$ must be less than 2^16-1. The number of parity symbols $\mathrm{N}-\mathrm{K}$ must be a positive even integer. A shortened code is inferred anytime the number of input data samples is less than $2^{\wedge} \mathrm{M}-1$ for M between 3 and 16 .
- The generator polynomial is not specified explicitly. However, it is defined by the code word length, the message length, and the B value for the starting exponent of the roots.
- For HDL code generation, the block does not handle double-precision input data. You can simulate using double-precision values, but if you attempt HDL code generation, you receive a error message.
- The Control Signals (start, end, valid) must be the Boolean datatype.


## Integer-Input RS Encoder HDL Optimized

## Block Dialog

## Integer-Input RS Encoder HDL Optimized Block Mask, Default View

| 嵒 Function Block Parameters: Integer-Input RS Encoder HDL Optimized |  | $x^{x}$ |
| :---: | :---: | :---: |
| Integer-Input RS Encoder HDL Optimized Encode data using a Reed-Solomon encoder |  |  |
|  |  |  |
| Codeword length: | 7 |  |
| Message length: | 3 |  |
| Source of primitive polynomial: | Auto | $\checkmark$ |
| Source of puncture pattern: | None | $\checkmark$ |
| Source of B, the starting power for roots of the primitive polynomial: | Auto | $\checkmark$ |
| OK Cancel | Help | Apply |

Integer-Input RS Encoder HDL Optimized Block Mask, Expanded View

| Function Block Parameters: Integer-Input RS Encoder HDL Optimized |  |  |
| :---: | :---: | :---: |
| Integer-Input RS Encoder HDL Optimized <br> Encode data using a Reed-Solomon encoder |  |  |
|  |  |  |
| Codeword length: 7 |  |  |
| Message length: |  |  |
| Source of primitive polynomial: Property |  |  |
| Primitive polynomial: [1, 0, 1, 1] |  |  |
| Source of puncture pattern: Property |  |  |
| Puncture pattern vector: <br> Source of B, the starting power for roots of the primitive polynomial: <br> $B$ value: | ( 2,1$)$; zeros( 2,1$)$ ] |  |
|  | Property | $\checkmark$ |
|  | 1 |  |
| OK Cancel | Help | Apply |

## Integer-Input RS Encoder HDL Optimized

## Codeword length

The codeword length.

## Message length

The message length.

## Source of primitive polynomial

Select Property to enable the Primitive polynomial parameter.

## Primitive polynomial

Binary row vector representing the primitive polynomial in descending order of powers. When you provide a primitive polynomial, the number of input bits must be an integer multiple of $K$ times the order of the primitive polynomial instead.

This parameter applies when only when Property is selected for Primitive polynomial.

## Source of puncture pattern

Select Property to enable the Puncture pattern vector parameter.

## Puncture pattern vector

A column vector of length $N-K$. In a puncture vector, 1 represents that the data symbol passes unaltered. The value 0 represents that the data symbol is punctured, or removed from the data stream.

The default value is [ones(2,1); zeros(2,1)].
This field is available only when Property is selected for Source of puncture pattern.

## Source of B, the starting power for roots of the primitive polynomial <br> Select Property to enable the $\mathbf{B}$ value parameter.

$B$ value
The starting exponent of the roots.

## Integer-Input RS Encoder HDL Optimized

This field is available only when you select Property for
Source of B, the starting power for roots of the primitive polynomial.

## Algorithm Timing Diagram

Serial Data Packet


Pair Block Integer-Output RS Decoder HDL Optimized
See Also Integer-Input RS Encoder

## Integer-Output RS Decoder

Purpose | Decode Reed-Solomon code to recover integer vector data |
| :--- |
| Library |
| Block sublibrary of Error Detection and Correction |

## Integer-Output RS Decoder

If the decoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

The default value of $M$ is ceil $(\log 2(N+1))$, that is, the smallest integer greater than or equal to $\log 2(\mathrm{~N}+1)$. You can change the value of M from the default by specifying the primitive polynomial for $\mathrm{GF}\left(2^{\mathrm{M}}\right)$, as described in "Specifying the Primitive Polynomial" on page 2-415 below. If N is less than $2^{\mathrm{M}}-1$, the block uses a shortened Reed-Solomon code.

You can also specify the generator polynomial for the Reed-Solomon code, as described in "Specifying the Generator Polynomial" on page 2-415.

An (N, K) Reed-Solomon code can correct up to floor ((N-K)/2) symbol errors (not bit errors) in each codeword.
The second output is the number of errors detected during decoding of the codeword. A-1 indicates that the block detected more errors than it could correct using the coding scheme. An (N,K) Reed-Solomon code can correct up to floor ( ( $\mathrm{N}-\mathrm{K}$ )/2) symbol errors (not bit errors) in each codeword. The data type of this output is also inherited from the input signal.

You can disable the second output by deselecting Output number of corrected errors. This removes the block's second output port.

In the case of a decoder failure, the message portion of the decoder input is returned unchanged as the decoder output.

The sample times of the input and output signals are equal.

## Punctured Codes

This block supports puncturing when you select the Punctured code parameter. This selection enables the Puncture vector parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 represents that the data symbol passes unaltered, and 0 represents that the data symbol gets punctured, or removed, from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

## Integer-Output RS Decoder

Note 1s and 0s have precisely opposite meanings for the puncture and erasure vectors. For an erasure vector, 1 means that the data symbol is to be replaced with an erasure symbol, and 0 means that the data symbol is passed unaltered. This convention is carried for both the encoder and the decoder.

## Dialog <br> Box



## Integer-Output RS Decoder

## Codeword length $\mathbf{N}$

The codeword length.

## Message length $K$

The message length.

## Specify primitive polynomial

Selecting this check box enables the field Primitive polynomial.

## Primitive polynomial

This parameter applies only when you select Specify primitive polynomial.

Binary row vector representing the primitive polynomial in descending order of powers.

## Specify generator polynomial

Selecting this check box enables the field Generator polynomial.

## Generator polynomial

Integer row vector, whose entries are in the range from 0 to $2^{\mathrm{M}}-1$, representing the generator polynomial in descending order of powers. Each coefficient is an element of the Galois field defined by the primitive polynomial.

This parameter applies only when you select Specify generator polynomial.

## Puncture code

Selecting this check box enables the field Puncture vector.

## Puncture vector

A column vector of length $N-K$. In the Puncture vector, a value of 1 represents that the data symbol passes unaltered, and 0 represents that the data symbol gets punctured, or removed, from the data stream.

The default value is [ones(2,1); zeros(2,1)].
This parameter applies only when you select Puncture code.

## Integer-Output RS Decoder

## Enable erasures input port

Selecting this check box will open the port, Era. This port accepts a binary column vector input signal with the same size as the codeword.

Erasure values of 1 represents symbols in the same position in the codeword that get erased, and values of 0 represent symbols that do not get erased.

## Output number of corrected errors

When you select this check box, the block outputs the number of corrected errors in each word through a second output port. A decoding failure occurs when a certain word in the input contains more than ( $\mathrm{N}-\mathrm{K}$ )/2 errors. A value of -1 indicates a decoding failure in the corresponding position in the second output vector.

## Algorithm

## Supported Data Type

This block uses the Berlekamp-Massey decoding algorithm. For information about this algorithm, see the references listed below.

| Port | Supported Data Types |
| :--- | :--- |
| In | $\bullet$ Double-precision floating point |
|  | $\bullet$ Single-precision floating point |
|  | $\bullet 8-, 16$-, and 32 -bit signed integers |
|  | $\bullet 8-, 16-$, and 32 -bit unsigned integers |
| Out | $\bullet$ Double-precision floating point |
|  | $\bullet$ Single-precision floating point |
|  | $\bullet 8-, 16-$, and 32 -bit signed integers |
|  | $\bullet 8-, 16-$, and 32 -bit unsigned integers |

## Integer-Output RS Decoder

| Port | Supported Data Types |
| :--- | :--- |
| Era | • Double-precision floating point |
|  | - Boolean |
| Err | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - 8-, 16-, and 32-bit signed integers |
|  | - If the input is uint8, uint16, or uint32, then the |
|  | number of errors output datatype is int8, int16, <br> or int32, respectively. |

## Pair Block Integer-Input RS Encoder <br> References [1] Wicker, Stephen B., Error Control Systems for Digital Communication and Storage, Upper Saddle River, N.J., Prentice Hall, 1995. <br> [2] Berlekamp, Elwyn R., Algebraic Coding Theory, New York, McGraw-Hill, 1968. <br> [3] Clark, George C., Jr., and J. Bibb Cain, Error-Correction Coding for Digital Communications, New York, Plenum Press, 1981.

See Also Binary-Output RS Decoder

## Integer-Output RS Decoder HDL Optimized

| Purpose | Decode data using a Reed-Solomon decoder |
| :--- | :--- |
| Library | Block sublibrary of Error Correction and Detection |

## Description



Reed-Solomon encoding follows the same standards as any other cyclic redundancy code. The Integer-Output RS Decoder HDL Optimized block can be used to model many communication system Forward Error Correcting (FEC) codes.

For more about the Reed-Solomon decoder, see the Integer-Output RS Decoder block reference. For more information on representing data for Reed-Solomon codes, see "Integer Format (Reed-Solomon Only)".

## Signal Attributes

The Integer-Output RS Decoder HDL Optimized block has four input ports and six output ports (5 required, 1 optional).

## Integer-Output RS Decoder HDL Optimized



| Port | Direction | Description | Data Type |
| :--- | :--- | :--- | :--- |
| datain | Input | Message data. Data width is <br> less than or equal to the CRC <br> length, and the CRC length <br> should be divisible by the <br> data width. For example, <br> for CRC-CCITT/CRC-16, the <br> valid data widths are 16, 8, <br> 4, 2, and 1. | Must be <br> an integer <br> (uint8, <br> uint16, <br> uint32) or fi(). <br> Doubles are <br> allowed for <br> simulation <br> but not for <br> HDL code <br> generation. |
| start | Input | Indicates the start of a <br> frame of data. | Boolean |

## Integer-Output RS Decoder HDL Optimized

| Port | Direction | Description | Data Type |
| :--- | :--- | :--- | :--- |
| end | Input | Indicates the end of a frame <br> of data. | Boolean |
| valid | Input | Indicates that input data is <br> valid. | Boolean |
| dataout | Output | Message data with the <br> checksum appended. The <br> data width is the same as <br> the input data port. | Must be <br> an integer <br> (uint8, <br> uint16, <br> uint32) or fi(). <br> Doubles are <br> ollowad far |
| startout | Output | Indicates the start of a <br> frame of data. | Boolean |
| endout | Output | Indicates the end of a frame <br> of data, including checksum. | Boolean |
| validout | Output | Indicates that output data <br> is valid. | Boolean |
| err | Output | Indicates the corruption <br> of the received data when <br> error is high. | Boolean in <br> and out |
| numerrs | Output | Optional. | Always a <br> uint8 output |

## Limitations

- The length of the code word $N$ must be less than 2^16-1. The number $^{\wedge}$ of parity symbols $\mathrm{N}-\mathrm{K}$ must be a positive even integer. A shortened code is inferred when the number of valid data samples between start and is less than the codeword length.


## Integer-Output RS Decoder HDL Optimized

- The generator polynomial is not specified explicitly. However, it is defined by the code word length, the message length, and the B value for the starting exponent of the roots.
- For HDL code generation, the block does not handle double-precision floating point datatype numbers. You can simulate using double-precision values, but if you attempt HDL code generation, you receive a error message.


## Dialog Box

## Integer-Ouput RS Decoder HDL Optimized Block Mask, Default View

| Function Block Parameters: Integer-Input RS Decoder HDL Optimized |  | $x^{x}$ |
| :---: | :---: | :---: |
| Integer-Input RS Decoder HDL Optimized <br> Decode data using a Reed-Solomon decoder |  |  |
|  |  |  |
| Codeword length: 7 |  |  |
| Message length: | 3 |  |
| Source of primitive polynomial: | Auto |  |
| Source of B, the starting power for roots of the primitive polynomial: | Auto | $\checkmark$ |
| $\checkmark$ Enable number of errors output |  |  |
| OK Cancel | Help | Apply |

[^0]
## Integer-Output RS Decoder HDL Optimized

| Function Block Parameters: Integer-Input RS Decoder HDL Optimized |  | $\pm$ |
| :---: | :---: | :---: |
| Integer-Input RS Decoder HDL Optimized |  |  |
| Decode data using a Reed-Solomon decoder |  |  |
| Codeword length: | 7 |  |
| Message length: | 3 |  |
| Source of primitive polynomial: | Property | - |
| Primitive polynomial: | [ 1, 0, 1, |  |
| Source of B, the starting power for roots of the primitive polynomial: | Property | $\checkmark$ |
| $B$ value: | 1 |  |
| V Enable number of errors output |  |  |
| OK Cancel | Help | Apply |

## Codeword length

The codeword length.

## Message length

The message length.

## Source of primitive polynomial

Select Property to enable the Primitive polynomial parameter.

## Primitive polynomial

Binary row vector representing the primitive polynomial in descending order of powers. When you provide a primitive polynomial, the number of input bits must be an integer multiple of $K$ times the order of the primitive polynomial instead.

This parameter applies when only when Property is selected for Primitive polynomial.

Source of B, the starting power for roots of the primitive polynomial

Select Property to enable the B value parameter.

## Integer-Output RS Decoder HDL Optimized

$B$ value
The starting exponent of the roots.
This field is available only when you select Property for Source of B, the starting power for roots of the primitive polynomial.

Enable number of errors output
Check this box to enable the number of errors output port.

## Algorithm Timing Diagram



Pair Block Integer-Input RS Encoder HDL Optimized
See Also Integer-Output RS Decoder

## Integer to Bit Converter

| Purpose | Map vector of integers to vector of bits |
| :--- | :--- |
| Library | Utility Blocks |

Description The Integer to Bit Converter block maps each integer (or fixed-point value) in the input vector to a group of bits in the output vector.
Integer to Bit Converter

The block maps each integer value (or stored integer when you use a fixed point input) to a group of $M$ bits, using the selection for the Output bit order to determine the most significant bit. The resulting output vector length is $M$ times the input vector length.

When you set the Number of bits per integer parameter to $M$ and Treat input values as to Unsigned, then the input values must be between 0 and $2^{\mathrm{M}}-1$. When you set Number of bits per integer to $M$ and Treat input values as to Signed, then the input values must be between $-2^{\mathrm{M}-1}$ and $2^{\mathrm{M}-1}-1$. During simulation, the block performs a run-time check and issues an error if any input value is outside of the appropriate range. When the block generates code, it does not perform this run-time check.

This block is single-rate and single-channel. It accepts a length $N$ column vector or a scalar-valued $(N=1)$ input signal and outputs a length $N \cdot M$ column vector.

The block can accept the data types int8, uint8, int16, uint16, int32, uint32, single, double, and fixed point.

## Integer to Bit Converter

## Dialog Box



## Number of bits per integer

The number of bits the block uses to represent each integer of the input. This parameter must be an integer between 1 and 32 .

## Treat input values as

Indicate if the integer value input ranges should be treated as signed or unsigned. The default setting is Unsigned.

## Output bit order

Define whether the first bit of the output signal is the most significant bit (MSB) or the least significant bit (LSB). The default selection is MSB first.

## Output data type

You can choose the following Output data type options:

- Inherit via internal rule
- Smallest integer


## Integer to Bit Converter

- Same as input
- double
- single
- uint8
- uint16
- uint32

The default selection for this parameter is Inherit via internal rule.

When the parameter is set to Inherit via internal rule, the block determines the output data type based on the input data type.

- If the input signal is floating-point (either single or double), the output data type is the same as the input data type.
- If the input data type is not floating-point, the output data type is determined as if the parameter is set to Smallest integer.

When the parameter is set to Smallest integer, the block selects the output data type based on settings used in the Hardware Implementation pane of the Configuration Parameters dialog box.

- If you select ASIC/FPGA, the output data type is the ideal one-bit size; ufix1.
- For all other selections, the output data type is an unsigned integer with the smallest available word length, as defined in the Hardware Implementation settings (e.g. uint8)


## Examples Fixed-Point Integer To Bit and Bit To Integer Conversion (Audio Scrambling and Descrambling Example) Overview

This example illustrates how to use the Bit to Integer and Integer to Bit Converter blocks with fixed-point signals.

This example uses a simplified audio scrambler configuration and a 16 -bit, fixed-point digital audio source, which is recorded speech. The left-side of the model represents the audio scrambler subsystem and the right-side represents the descrambler subsystem.

## Opening the Model

You can open the model by typing doc_audioscrambler at the MATLAB command line.

## Structure

In the audio scrambler subsystem, the Integer to Bit Converter block unpacks each 16 -bit audio sample into a binary, 1-bit signal. The binary signal passes to a linear feedback shift register (LFSR) scrambler, which randomizes the bits in a controllable way, thereby scrambling the signal. The Communications System Toolbox Scrambler block is used in the LFSR implementation. From the LFSR, the scrambled audio bits pass to the Bit to Integer Converter block. This block packs the scrambled 1 -bit samples into 16 -bit audio samples. The audio samples pass to the Data Type Conversion block, which converts the integer-based audio samples back into fixed-point samples.
The fixed-point samples pass from the scrambler subsystem to a channel. The channel sends the samples to the descrambler subsystem. For illustrative purposes, this example uses a noiseless channel. In an actual system, a channel may introduce noise. Removing such noise requires a more sophisticated design.

In the audio descrambler subsystem, the Integer to Bit Converter block unpacks each 16 -bit audio sample into a binary, 1-bit signal. The binary signal passes to a linear feedback shift register (LFSR) descrambler, which randomizes the bits in a controllable way, reversing the scrambling process. This LFSR descrambler implementation uses the Communications System Toolbox Descrambler block. From the

## Integer to Bit Converter

LFSR, the descrambled audio bits pass to the Bit to Integer Converter block. This block packs the descrambled 1-bit samples into 16 -bit audio samples. The audio samples pass to the Data Type Conversion block, which converts the integer-based audio samples back into fixed-point samples.
In Simulink, the sfix16_En15 data type represents a signed (s) fixed-point (fix) signal with word length 16 and fraction length 15. Therefore, this model represents audio signals using the sfix16_En15 data type, except when converting to and from 1-bit binary signals. All 1-bit signals are represented by ufix1, as seen at the output of the Integer to Bit Converter block. The audio source has a frame size (or number of samples per frame) of 1024. For more information on fixed-point signals, please refer to Fixed-Point Numbers in the Simulink documentation.

## Integer to Bit Converter

Fixed-Point Integer to Bit and Bit to Integer Conversion (Audio Scrambler and Descrambler Example)


## Running the Model

You must run the example before you can listen to any of the audio signals.

You can run the example by clicking Simulation $>$ Run.
You can hear the audio signals by clicking the model's yellow, audio icons.

## Converter Block Settings

In the audio scrambler and descrambler subsystems, the Integer to Bit Converter block settings are:

## Integer to Bit Converter

- Number of bits per integer: 16
- Treat input values as: Signed
- Output bit order: MSB first
- Output data type: Inherit via internal rule

In the audio scrambler and descrambler subsystems, the Bit to Integer Converter block settings are:

- Number of bits per integer: 16
- Input bit order: MSB first
- After bit packing, treat resulting integer values as: Signed
- Output data type: Inherit via internal rule

Pair Block Bit to Integer Converter
See Also de2bi and dec2bin

## Purpose

Integrate discrete-time signal, resetting to zero periodically

## Library

Description

## Integrate

 and DumpComm Filters
The Integrate and Dump block creates a cumulative sum of the discrete-time input signal, while resetting the sum to zero according to a fixed schedule. When the simulation begins, the block discards the number of samples specified in the Offset parameter. After this initial period, the block sums the input signal along columns and resets the sum to zero every $N$ input samples, where $N$ is the Integration period parameter value. The reset occurs after the block produces its output at that time step.
Receiver models often use the integrate-and-dump operation when the system's transmitter uses a simple square-pulse model. Fiber optics and in spread-spectrum communication systems, such as CDMA (code division multiple access) applications, also use the operation.
This block accepts a scalar, column vector, or matrix input signal. When the input signal is not a scalar value, it must contain $k \cdot N$ rows for some positive integer $k$. For these input signals, the block processes each column independently.

Selecting Output intermediate values affects the contents, dimensions, and sample time as follows:

- If you clear the check box, then the block outputs the cumulative sum at each reset time.
- If the input is a scalar value, then the output sample time is $N$ times the input sample time and the block experiences a delay whose duration is one output sample period. In this case, the output dimensions match the input dimensions.
- If the input is a ( $k \cdot N$ )-by-n matrix, then the output is $k$-by- $n$. In this case, the block experiences no delay and the output period matches the input period.


## Integrate and Dump

- If you select the check box, then the block outputs the cumulative sum at each time step. The output has the same sample time and the same matrix dimensions as the input.


## Transients and Delays

A nonzero value in the Offset parameter causes the block to output one or more zeros during the initial period while it discards input samples. If the input is a matrix with $n$ columns and the Offset parameter is a length $-n$ vector, then the $m^{\text {th }}$ element of the Offset vector is the offset for the $m^{\text {th }}$ column of data. If Offset is a scalar, then the block applies the same offset to each column of data. The output of initial zeros due to a nonzero Offset value is a transient effect, not a persistent delay.

When you clear Output intermediate values, the block's output is delayed, relative to its input, throughout the simulation:

- If the input is a scalar value, then the output is delayed by one sample after any transient effect is over. That is, after removing transients from the input and output, you can see the result of the $m^{\text {th }}$ integration period in the output sample indexed by $\mathrm{m}+1$.
- If the input is a column vector or matrix and the Offset parameter is nonzero, then after the transient effect is over, the result of each integration period appears in the output frame corresponding to the last input sample of that integration period. This is one frame later than the output frame corresponding to the first input sample of that integration period, in cases where an integration period spans two input frames. For an example of this situation, see "Example of Transient and Delay" on page 2-452.


## Integrate and Dump

## Dialog

 Box

## Integration period

The number of input samples between resets.

## Offset

A nonnegative integer vector or scalar specifying the number of input samples to discard from each column of input data at the beginning of the simulation.

## Output intermediate values

Determines whether the block outputs the intermediate cumulative sums between successive resets.

## Integrate and Dump

## Fixed-Point Signal Flow Diagram



## Integrate and Dump

## Fixed-Point Attributes



The settings for the following parameters only apply when block inputs are fixed-point signals.

## Rounding mode

Use this parameter to specify the rounding method to be used when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result.

## Integrate and Dump

For more information, see "Rounding Modes" in the DSP System Toolbox documentation or "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

## Overflow mode

Use this parameter to specify the method to be used if the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result:

- Saturate represents positive overflows as the largest positive number in the range being used, and negative overflows as the largest negative number in the range being used.
- Wrap uses modulo arithmetic to cast an overflow back into the representable range of the data type. See "Modulo Arithmetic" for more information.


## Accumulator-Mode

Use the Accumulator-Mode parameter to specify how you would like to designate the accumulator word and fraction lengths:

- When you select Inherit via internal rule, the block automatically calculates the accumulator output word and fraction lengths.
- When you select Same as input, these characteristics match those of the input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator.


## Output

Use the Output parameter to choose how you specify the word length and fraction length of the output of the block:

- When you select Same as accumulator, these characteristics match those of the accumulator.


## Integrate and Dump

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Binary point scaling, enter the word length and the fraction length of the output, in bits.
- When you select Slope and bias scaling, enter the word length, in bits, and the slope of the output.

For additional information about the parameters pertaining to fixed-point applications, see "Specify Fixed-Point Attributes for Blocks".

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Fixed-point |
| Out | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Fixed-point |

## Examples

If Integration period is 4 and Offset is the scalar 3, then the table below shows how the block treats the beginning of a ramp ( $1,2,3,4, \ldots$ ) in several situations. (The values shown in the table do not reflect vector sizes but merely indicate numerical values.)

| Output intermediate values Check Box | Input Signal Properties | First Several Output Values |
| :---: | :---: | :---: |
| Cleared | Scalar | $0,0,4+5+6+7$, and $8+9+10+11$, where one 0 is an initial transient value and the other 0 is a delay value that results from the cleared check box and scalar value input. |
| Cleared | Column vector of length 4 | $0,4+5+6+7$, and $8+9+10+11$, where 0 is an initial delay value that results from the nonzero offset. The output is a scalar value. |
| Selected | Scalar | $\begin{aligned} & 0,0,0,4,4+5,4+5+6, \\ & 4+5+6+7,8,8+9,8+9+10, \\ & 8+9+10+11, \text { and } 12 \text {, where } \\ & \text { the three } 0 \text { s are initial } \\ & \text { transient values. } \end{aligned}$ |
| Selected | Column vector of length 4 | $\begin{aligned} & 0,0,0,4,4+5,4+5+6, \\ & 4+5+6+7,8,8+9,8+9+10, \\ & 8+9+10+11, \text { and } 12, \text { where } \end{aligned}$ <br> the three 0s are initial transient values. The output is a column vector of length 4. |

In all cases, the block discards the first three input samples (1, 2, and 3).

## Example of Transient and Delay

The figure below illustrates a situation in which the block exhibits both a transient effect for three output samples, as well as a one-sample delay in alternate subsequent output samples for the rest of the
simulation. The figure also indicates how the input and output values are organized as column vectors. In each vector in the figure, the last sample of each integration period is underlined, discarded input samples are white, and transient zeros in the output are white.


The transient effect lasts for ceil(13/5) output samples because the block discards 13 input samples and the integration period is 5 . The first output sample after the transient effect is over, 80, corresponds to the sum $14+15+16+17+18$ and appears at the time of the input sample 18. The next output sample, 105, corresponds to the sum $19+20+21+22+23$ and appears at the time of the input sample 23. Notice that the input sample 23 is one frame later than the input sample 19; that is, this five-sample integration period spans two input frames. As a result, the output of 105 is delayed compared to the first input (19) that contributes to that sum.

See Also
Windowed Integrator, Discrete-Time Integrator (Simulink documentation), Ideal Rectangular Pulse Filter

## Interlacer

## Purpose

Library
Description


Dialog Box

## Examples

Alternately select elements from two input vectors to generate output vector

Sequence Operations
The Interlacer block accepts two inputs that have the same vector size, complexity, and sample time. It produces one output vector by alternating elements from the first input (labeled 0 for odd) and from the second input (labeled E for even). As a result, the output vector size is twice that of either input. The output vector has the same complexity and sample time of the inputs.

Both input ports accept scalars or column vectors with the same number of elements. The block accepts the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.
This block can be useful for combining in-phase and quadrature information from separate vectors into a single vector.


If the two input vectors have the values $[1 ; 2 ; 3 ; 4]$ and [5; 6; 7; 8], then the output vector is $[1 ; 5 ; 2 ; 6 ; 3 ; 7 ; 4 ; 8]$.
Pair Block Deinterlacer
See Also General Block Interleaver; Mux (Simulink documentation)

## I/Q Imbalance

## Purpose Create complex baseband model of signal impairments caused by

 imbalances between in-phase and quadrature receiver components
## Library

RF Impairments
Description
The I/Q Imbalance block creates a complex baseband model of the signal impairments caused by imbalances between in-phase and quadrature receiver components. Typically, these are caused by differences in the physical channels for the two components of the signal.

The I/Q Imbalance block applies amplitude and phase imbalances to the in-phase and quadrature components of the input signal, and then combines the results into a complex signal. The block

1 Separates the signal into its in-phase and quadrature components.
2 Applies amplitude and phase imbalances, specified by the I/Q amplitude imbalance (dB) and I/Q phase imbalance (deg) parameters, respectively, to both components.

3 Combines the in-phase and quadrature components into a complex signal.

4 Applies an in-phase dc offset, specified by the I dc offset parameter, and a quadrature offset, specified by the $\mathbf{Q}$ dc offset parameter, to the signal.

The block performs these operations in the subsystem shown in the following diagram, which you can view by right-clicking the block and selecting Mask > Look under mask:


合
Let
$I_{a}=\mathrm{I} / \mathrm{Q}$ amplitude imbalance
$I_{p}=\mathrm{I} / \mathrm{Q}$ phase imbalance
$I_{D C}=$ in-phase DC offset
$Q_{D C}=$ quadrature DC offset
Also let $x=x_{r}+j * x_{i}$ be the complex input to the block, with $x_{r}$ and $x_{i}$ being the real and imaginary parts, respectively, of $x$. Let $y$ be the complex output of the block.

Then, for an I/Q amplitude imbalance, $I_{a}$

$$
\begin{aligned}
& y_{\text {AmplitudeImbalance }}=\left[10^{\left(0.5^{*} \frac{I_{a}}{20}\right)} * x_{r}\right]+j\left[10^{\left(-0.5^{*} \frac{I_{a}}{20}\right)} * x_{i}\right] \\
& y_{r \text { Anpplitudelmbalanee }}+j^{*} y_{i_{\text {Anpplitudelmbolalanee }}}
\end{aligned}
$$

For an I/Q phase imbalance $I_{p}$

## I/Q Imbalance

$y_{\text {PhaseImbalance }}=$

$$
\begin{aligned}
& {\left[\exp \left(-0.5 * j * \pi * \frac{I_{p}}{180}\right) * y_{r_{\text {Amplitude } \operatorname{Im} \text { balance }}}\right]+\left\{\exp \left[j\left(\frac{\pi}{2}+0.5 * \pi * \frac{I_{p}}{180}\right)\right] * y_{i_{\text {Amplitude } \operatorname{Im} \text { balance }}}\right\}} \\
& \left.\square y_{\text {rPhaselmbalance }}+j * y_{\text {iPhasesImbalance }}\right\}
\end{aligned}
$$

For DC offsets $I_{D C}$ and $Q_{D C}$
$\mathrm{y}=\left(\mathrm{y}_{\mathrm{r} \text { Phasesmbalanee }}+I_{D C}\right)+\mathrm{j} *\left(y_{\text {iPhnsesmbdannee }}+Q_{D C}\right)$

The value of the $\mathbf{I} / \mathbf{Q}$ amplitude imbalance (dB) parameter is divided between the in-phase and quadrature components:

- If you enter a positive value $X$ for the $\mathbf{I} / \mathbf{Q}$ amplitude imbalance (dB), the block applies a gain of $+X / 2 \mathrm{~dB}$ to the in-phase component and a gain of $-X / 2 \mathrm{~dB}$ to the quadrature component.
- If you enter a negative value $X$ for the $I / Q$ amplitude imbalance (dB), the block applies a gain of $-X / 2 \mathrm{~dB}$ to the in-phase component and a gain of $+X / 2 \mathrm{~dB}$ to the quadrature component.

The effects of changing the block's parameters are illustrated by the following scatter plots of a signal modulated by 16 -ary quadrature amplitude modulation (QAM) with an average power of 0.01 watts. The usual 16-ary QAM constellation without distortion is shown in the first scatter plot:

## I/Q Imbalance



The following figure shows a scatter plot of an output signal, modulated by 16 -ary QAM, from the I/Q block with I/Q amplitude imbalance (dB) set to 8 and all other parameters set to 0 :

## I/Q Imbalance



Observe that the scatter plot is stretched horizontally and compressed vertically compared to the undistorted constellation.

If you set IQ phase imbalance (deg) to 30 and all other parameters to 0 , the scatter plot is skewed clockwise by 30 degrees, as shown below:

## I/Q Imbalance



Setting the I dc offset to 0.02 and the $\mathbf{Q}$ dc offset to 0.04 shifts the constellation 0.02 to the right and 0.04 up, as shown below:

## I/Q Imbalance



See "Illustrate RF Impairments That Distort a Signal" for a description of the model that generates this plot.

## I/Q Imbalance



Dialog Box

I/Q amplitude imbalance ( dB )
Scalar specifying the I/Q amplitude imbalance in decibels.

## I/Q phase imbalance (deg)

Scalar specifying the I/Q phase imbalance in degrees.

## I dc offset

Scalar specifying the in-phase dc offset.

## Q dc offset

Scalar specifying the amplitude dc offset.
See Also Memoryless Nonlinearity

| Purpose | Generate Kasami sequence from set of Kasami sequences |
| :--- | :--- |
| Library | Sequence Generators sublibrary of Comm Sources |
| Description | The Kasami Sequence Generator block generates a sequence from the <br> set of Kasami sequences. The Kasami sequences are a set of sequences <br> that have good cross-correlation properties. |
| Kasami <br> Sequence <br> Generator This block can output sequences that vary in length during simulation. <br>  For more information about variable-size signals, see "Variable-Size <br> Signal Basics" in the Simulink documentation.. |  |

## Kasami Sequences

There are two sets of Kasami sequences: the small set and the large set. The large set contains all the sequences in the small set. Only the small set is optimal in the sense of matching Welch's lower bound for correlation functions.

Kasami sequences have period $N=2^{\mathrm{n}}-1$, where $n$ is a nonnegative, even integer. Let $u$ be a binary sequence of length $N$, and let $w$ be the sequence obtained by decimating u by $2^{n / 2}+1$. The small set of Kasami sequences is defined by the following formulas, in which $T$ denotes the left shift operator, $m$ is the shift parameter for $w$, and $\oplus$ denotes addition modulo 2.

$$
K_{s}(u, n, m)= \begin{cases}u & m=-1 \\ u \oplus T^{m} w & m=0, \ldots, 2^{n / 2}-2\end{cases}
$$

## Small Set of Kasami Sequences for $\mathbf{n}$ Even

Note that the small set contains $2^{\mathrm{n} / 2}$ sequences.
For $\bmod (n, 4)=2$, the large set of Kasami sequences is defined as follows. Let $v$ be the sequence formed by decimating the sequence $u$ by $2^{\mathrm{n} / 2+1}+1$. The large set is defined by the following table, in which $k$ and $m$ are the shift parameters for the sequences $v$ and $w$, respectively.

$$
K_{L}(u, n, k, m)= \begin{cases}u & k=-2 ; m=-1 \\ v & k=-1 ; m=-1 \\ u \oplus T^{k} v & k=0, \ldots, 2^{n}-2 ; m=-1 \\ u \oplus T^{m} w & k=-2 ; m=0, \ldots, 2^{n / 2}-2 \\ v \oplus T^{m} w & k=-1 ; m=0, \ldots, 2^{n / 2}-2 \\ u \oplus T^{k} v \oplus T^{m} w & k=0, \ldots, 2^{n}-2 ; m=0, \ldots, 2^{n / 2}-2\end{cases}
$$

## Large Set of Kasami Sequences for $\bmod (\mathbf{n}, 4)=2$

The sequences described in the first three rows of the preceding figure correspond to the Gold sequences for $\bmod (\mathrm{n}, 4)=2$. See the reference page for the Gold Sequence Generator block for a description of Gold sequences. However, the Kasami sequences form a larger set than the Gold sequences.

The correlation functions for the sequences takes on the values

$$
\{-t(n),-s(n),-1, s(n)-2, t(n)-2\}
$$

where

$$
\begin{aligned}
& t(n)=1+2^{(n+2) / 2}, n \text { even } \\
& s(n)=\frac{1}{2}(t(n)+1)
\end{aligned}
$$

## Block Parameters

The Generator polynomial parameter specifies the generator polynomial, which determines the connections in the shift register that generates the sequence $u$. You can specify the Generator polynomial parameter using either of these formats:

- A vector that lists the coefficients of the polynomial in descending order of powers. The first and last entries must be 1 . Note that the length of this vector is one more than the degree of the generator polynomial.


## Kasami Sequence Generator

- A vector containing the exponents of $z$ for the nonzero terms of the polynomial in descending order of powers. The last entry must be 0 .

For example, $\left[\begin{array}{lllllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0 & 1\end{array}\right]$ and [ 8220$]$ represent the same polynomial, $p(z)=z^{8}+z^{2}+1$.

The Initial states parameter specifies the initial states of the shift register that generates the sequence u. Initial States is a binary scalar or row vector of length equal to the degree of the Generator polynomial. If you choose a binary scalar, the block expands the scalar to a row vector of length equal to the degree of the Generator polynomial, all of whose entries equal the scalar.

The Sequence index parameter specifies the shifts of the sequences $v$ and $w$ used to generate the output sequence. You can specify the parameter in either of two ways:

- To generate sequences from the small set, for $n$ is even, you can specify the Sequence index as an integer $m$. The range of $m$ is $\left[-1, \ldots, 2^{\mathrm{n} / 2}-2\right]$. The following table describes the output sequences corresponding to Sequence index m:

| Sequence <br> Index | Range of Indices | Output Sequence |
| :--- | :--- | :--- |
| -1 | $m=-1$ | u |
| m | $m=0, \ldots, 2^{\mathrm{n} / 2}-2$ | $u \oplus T^{m} w$ |

- To generate sequences from the large set, for $\bmod (n, 4)=2$, where $n$ is the degree of the Generator polynomial, you can specify Sequence index as an integer vector $[k \mathrm{~m}]$. In this case, the output sequence is from the large set. The range for $k$ is $\left[-2, \ldots, 2^{\mathrm{n}}-2\right]$, and the range for $m$ is $\left[-1, \ldots, 2^{\text {n/2 }}-2\right]$. The following table describes the output sequences corresponding to Sequence index [k m]:

| Sequence Index <br> [k m] | Range of Indices | Output Sequence |
| :--- | :--- | :--- |
| $\left[\begin{array}{ll}-2 & -1\end{array}\right]$ | $k=-2, m=-1$ | u |
| $\left[\begin{array}{ll}-1 & -1\end{array}\right]$ | $k=-1, m=-1$ | v |
| $\left[\begin{array}{ll}\mathrm{k}-1\end{array}\right]$ | $k=0,1, \ldots, 2^{\mathrm{n}}-2$ <br> $m=-1$ | $u \oplus T^{k} v$ |
| $[-2 \mathrm{~m}]$ | $k=-2$ <br> $m=0,1, \ldots, 2^{\mathrm{n} / 2}-2$ | $u \oplus T^{m} w$ |
| $\left[\begin{array}{ll}-1 \mathrm{~m}] & m=-1 \\ m=0, \ldots, 2^{\mathrm{n} / 2}-2\end{array}\right.$ | $v \oplus T^{m} w$ |  |
| $[\mathrm{k} \mathrm{m}]$ | $k=0, \ldots, 2^{\mathrm{n}}-2$ <br> $m=0, \ldots, 2^{\mathrm{n} / 2}-2$ | $u \oplus T^{k} v \oplus T^{m} w$ |

You can shift the starting point of the Kasami sequence with the Shift parameter, which is an integer representing the length of the shift.

You can use an external signal to reset the values of the internal shift register to the initial state by selecting Reset on nonzero input. This creates an input port for the external signal in the Kasami Sequence Generator block. The way the block resets the internal shift register depends on whether its output signal and the reset signal are sample-based or frame-based. See "Example: Resetting a Signal" on page 2-723 for an example.

## Polynomials for Generating Kasami Sequences

The following table lists some of the polynomials that you can use to generate the Kasami set of sequences.

| $\mathbf{n}$ | $\mathbf{N}$ | Polynomial | Set |
| :--- | :--- | :--- | :--- |
| 4 | 15 | $\left[\begin{array}{lll}4 & 1 & 0\end{array}\right]$ | Small |
| 6 | 63 | $\left[\begin{array}{lll}6 & 1 & 0\end{array}\right]$ | Large |

## Kasami Sequence Generator

| $\mathbf{n}$ | $\mathbf{N}$ | Polynomial | Set |
| :--- | :--- | :--- | :--- |
| 8 | 255 | $\left[\begin{array}{llll}8 & 4 & 3 & 2\end{array}\right]$ | Small |
| 10 | 1023 | $\left[\begin{array}{llll}1 & 3 & 0\end{array}\right]$ | Large |
| 12 | 4095 | $\left[\begin{array}{llll}12 & 6 & 4 & 1\end{array}\right]$ |  |

## Kasami Sequence Generator

## Source Block Parameters: Kasami Sequence Generator

Kasami Sequence Generator
Generate a Kasami sequence from the set of Kasami sequences by specifying the generator polynomial.

The generator polynomial parameter value represents the shift register connections. Enter these values as either a binary vector or a descending ordered polynomial to indicate the connection points.

The sequence index parameter denotes the single sequence outputted from the set of Kasami sequences. Specify it as a 2-element integer vector for the Large set of sequences or as a scalar integer for the Small set of sequences.

For variable-size output signals, the current output size is either specified from the 'oSiz' input or inherited from the 'Ref' input.

Parameters

Initial states: [ $\left.\begin{array}{lllllll}0 & 0 & 0 & 0 & 0 & 1\end{array}\right]$
Sequence index(es):
Shift: 0
$\square$ Output variable-size signals
Sample time: 1
$\square$ Frame-based outputsReset on nonzero input
Output data type: double

Dialog


## Generator polynomial

Binary vector specifying the generator polynomial for the sequence $u$.

## Kasami Sequence Generator

## Initial states

Binary scalar or row vector of length equal to the degree of the Generator polynomial, which specifies the initial states of the shift register that generates the sequence $u$.

## Sequence index

Integer or vector specifying the shifts of the sequences $v$ and $w$ used to generate the output sequence.

## Shift

Integer scalar that determines the offset of the Kasami sequence from the initial time.

## Output variable-size signals

Select this if you want the output sequences to vary in length during simulation. The default selection outputs fixed-length signals.

## Maximum output size source

Specify how the block defines maximum output size for a signal.

- When you select Dialog parameter, the value you enter in the Maximum output size parameter specifies the maximum size of the output. When you make this selection, the oSiz input port specifies the current size of the output signal and the block output inherits sample time from the input signal. The input value must be less than or equal to the Maximum output size parameter.
- When you select Inherit from reference port, the block output inherits sample time, maximum size, and current size from the variable-sized signal at the Ref input port.

This parameter only appears when you select Output variable-size signals. The default selection is Dialog parameter.

## Maximum output size

Specify a two-element row vector denoting the maximum output size for the block. The second element of the vector must be

1. For example, [10 1] gives a 10-by-1 maximum sized output signal. This parameter only appears when you select Output variable-size signals.

## Sample time

Period of each element of the output signal.

## Frame-based outputs

Determines whether the output is frame-based or sample-based.

## Samples per frame

The number of samples in a frame-based output signal. This field is active only if you select Frame-based outputs.

## Reset on nonzero input

When selected, you can specify an input signal that resets the internal shift registers to the original values of the Initial states.

## Output data type

The output type of the block can be specified as a boolean or double. By default, the block sets this to double.

## See Also

Gold Sequence Generator, PN Sequence Generator

## Reference

[1] Peterson and Weldon, Error Correcting Codes, 2nd Ed., MIT Press, Cambridge, MA, 1972.
[2] Proakis, John G., Digital Communications, Third edition, New York, McGraw Hill, 1995.
[3] Sarwate, D. V. and Pursley, M.B., "Crosscorrelation Properties of Pseudorandom and Related Sequences," Proc. IEEE, Vol. 68, No. 5, May 1980, pp. 583-619.

## LDPC Decoder

## Purpose

Library
Description


Decode binary low-density parity-check code specified by parity-check matrix

Block sublibrary of Error Detection and Correction
This block implements the message-passing algorithm for decoding low-density parity-check (LDPC) codes, which are linear error control codes with sparse parity-check matrices and long block lengths that can attain performance near the Shannon limit.

The LDPC Decoder block is designed to:

- Decode generic binary LDPC codes where no patterns in the parity-check matrix are assumed.
- Execute a number of iterations you specify or run until all parity-checks are satisfied.
- Output hard decisions or soft decisions (log-likelihood ratios) for decoded bits.
( $n-k$ ) and $n$ are the number of rows and columns, respectively, in the parity-check matrix.

This block accepts a real-valued, $n \times 1$ column vector input signal of type double. Each element is the log-likelihood ratio for a received bit (more likely to be 0 if the log-likelihood ratio is positive). The first $k$ elements correspond to the information part of a codeword.

Both the input and the output are discrete-time signals. The ratio of the output sample time to the input sample time is $n / k$ if only the information part is decoded, and 1 if the entire codeword is decoded.

## Decoding Algorithm



The input to the LDPC decoder is the log-likelihood ratio (LLR), $L\left(c_{i}\right)$, which is defined by the following equation

$$
L\left(c_{i}\right)=\log \left(\frac{\operatorname{Pr}\left(c_{i}=0 \mid \text { channel output for } c_{i}\right)}{\operatorname{Pr}\left(c_{i}=1 \mid \text { channel output for } c_{i}\right)}\right)
$$

where $c_{i}$ is the $i$ th bit of the transmitted codeword, $c$. There are three key variables in the algorithm: $L\left(r_{j i}\right), L\left(q_{i j}\right)$, and $L\left(Q_{i}\right) . L\left(q_{i j}\right)$ is initialized as $L\left(q_{i j}\right)=L\left(c_{i}\right)$. For each iteration, update $L\left(r_{j i}\right), L\left(q_{i j}\right)$, and $L\left(Q_{i}\right)$ using the following equations

$$
\begin{aligned}
& L\left(r_{j i}\right)=2 \operatorname{atanh}\left(\prod_{i^{\prime} \in V_{j} \backslash i} \tanh \left(\frac{1}{2} L\left(q_{i^{\prime} j}\right)\right)\right) \\
& L\left(q_{i j}\right)=L\left(c_{i}\right)+\sum_{j^{\prime} \in C_{i} \backslash j} L\left(r_{j^{\prime} i^{\prime}}\right) \\
& L\left(Q_{i}\right)=L\left(c_{i}\right)+\sum_{j^{\prime} \in C_{i}} L\left(r_{j^{\prime} i}\right)
\end{aligned}
$$

where the index sets, $C_{i} \backslash j$ and $V_{j} \backslash i$, are chosen as shown in the following example.

Suppose you have the following parity-check matrix $\mathbf{H}$ :

$$
\mathbf{H}=\left(\begin{array}{llllllllll}
1 & 1 & 1 & 1 & 0 & 0 & 0 & 0 & 0 & 0 \\
1 & 0 & 0 & 0 & 1 & 1 & 1 & 0 & 0 & 0 \\
0 & 1 & 0 & 0 & 1 & 0 & 0 & 1 & 1 & 0 \\
0 & 0 & 1 & 0 & 0 & 1 & 0 & 1 & 0 & 1 \\
0 & 0 & 0 & 1 & 0 & 0 & 1 & 0 & 1 & 1
\end{array}\right)
$$

For $i=5$ and $j=3$, the index sets would be


At the end of each iteration, $L\left(Q_{i}\right)$ provides an updated estimate of the a posteriori log-likelihood ratio for the transmitted bit $c_{i}$.

The soft-decision output for $c_{i}$ is $L\left(Q_{i}\right)$. The hard-decision output for $c_{i}$ is 1 if $L\left(Q_{i}\right)<0$, and 0 otherwise.
If the property DoParityCheck is set to 'no', the algorithm iterates as many times as specified by the Number of iterations parameter.

If the property DoParityCheck is set to 'yes', then at the end of each iteration the algorithm verifies the parity check equation ( $\mathbf{H c}^{\mathrm{T}}=0$ ) and stops if it is satisfied.

In this algorithm, $\operatorname{atanh}(1)$ and $\operatorname{atanh}(-1)$ are set to be 19.07 and -19.07 respectively to avoid infinite numbers from being used in the algorithm's equations. These numbers were chosen because MATLAB returns 1 for $\tanh (19.07)$ and -1 for $\tanh (-19.07)$, due to finite precision.

## LDPC Decoder



## Dialog Box

## Parity-check matrix

This parameter accepts a sparse matrix with dimension $n-k$ by $n$ (where $n>k>0$ ) of real numbers. All nonzero elements must be equal to 1 . The upper bound limit for the value of $n$ is $2^{31}-1$

## Output format

The output is a real-valued column vector signal. The options are Information part and Whole codeword.

## LDPC Decoder

- When you this parameter to Information part, the output contains $k$ elements.
- When you set this parameter to whole codeword, the output contains $n$ elements


## Decision type

The options are Hard decision and Soft decision.

- When you set this parameter to Hard decision, the output is decoded bits (of type double or boolean).
- When you set this parameter to Soft decision, the output is log-likelihood ratios (of type double).


## Output data type

This parameter appears only when Decision type is set to Hard decision.

The options are boolean and double.

## Number of iterations

This can be any positive integer.
Stop iterating when all parity checks are satisfied
If checked, the block will determine whether the parity checks are satisfied after each iteration and stop if all are satisfied.

## Output number of iterations executed

Creates an output port on the block when selected.

## Output final parity checks

Creates an output port on the block when selected.

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | - Double-precision floating point |
| Out | - Double-precision floating point |
|  | - Boolean when Decision type <br> is Hard decision |

## LDPC Decoder

Examples<br>Enter commdvbs2 at the command line to see an example that uses this block.<br>References [1] Gallager, Robert G., Low-Density Parity-Check Codes, Cambridge, MA, MIT Press, 1963 .

See Also LDPC Encoder \| comm.LDPCDecoder \| dvbs2ldpc

## LDPC Encoder

Purpose Encode binary low-density parity-check code specified by parity-check matrix

Library
Description


Block sublibrary of Error Detection and Correction
This block supports encoding of low-density parity-check (LDPC) codes, which are linear error control codes with sparse parity-check matrices and long block lengths that can attain performance near the Shannon limit.

Both the input and the output are discrete-time signals. The ratio of the output sample time to the input sample time is $k / n$. The input must be a real $k \times 1$ column vector signal.

The output signal inherits the data type from the input signal, and the input must be binary-valued (0 or 1). For information about the data types each block port supports, see the "Supported Data Type" on page 2-480 table on this page.

Note Model initialization or update may take a long time, because a large matrix may need to be inverted (when the last ( $n-k$ ) columns of the parity-check matrix is not triangular).

Dialog
Box


## Parity-check matrix

This block can accept a sparse matrix with dimension $n-k$ by $n$ (where $n>k>0$ ) of real numbers. All nonzero elements must be equal to 1 . The upper bound limit for the value of $n$ is $2^{31}-1$

The default value is the parity-check matrix of the half-rate LDPC code from the DVB-S. 2 standard.

## LDPC Encoder

## Supported Data Type

| Port | Supported Data Types |
| :---: | :---: |
| In | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean <br> - 8-, 16-, and 32 -bit signed integers <br> - 8 -, 16 -, and 32 -bit unsigned integers |
| Out | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean <br> - 8 -, 16-, and 32 -bit signed integers <br> - 8 -, 16 -, and 32 -bit unsigned integers |

## Examples

See Also
Enter commdvbs2 at the command line to see an example that uses this block.

LDPC Decoder | comm.LDPCEncoder | dvbs2ldpc

## Linearized Baseband PLL

## Purpose

## Library

Description
Linearized Filt $=$
Baseband PD PLL VCO.

Implement linearized version of baseband phase-locked loop

Components sublibrary of Synchronization
The Linearized Baseband PLL block is a feedback control system that automatically adjusts the phase of a locally generated signal to match the phase of an input signal. Unlike the Phase-Locked Loop block, this block uses a baseband model method. Unlike the Baseband PLL block, which uses a nonlinear model, this block simplifies the computations by using $x$ to approximate $\sin (x)$. The baseband PLL model depends on the amplitude of the incoming signal but does not depend on a carrier frequency.

This PLL has these three components:

- An integrator used as a phase detector.
- A filter. You specify the filter's transfer function using the Lowpass filter numerator and Lowpass filter denominator parameters. Each is a vector that gives the respective polynomial's coefficients in order of descending powers of $s$.

To design a filter, you can use functions such as butter, cheby1, and cheby2 in Signal Processing Toolbox software. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.

```
[num, den] = cheby2(3,40,100,'s')
```

- A voltage-controlled oscillator (VCO). You specify the sensitivity of the VCO signal to its input using the VCO input sensitivity parameter. This parameter, measured in Hertz per volt, is a scale factor that determines how much the VCO shifts from its quiescent frequency.

This block accepts a sample-based scalar input signal. The input signal represents the received signal. The three output ports produce:

- The output of the filter


## Linearized Baseband PLL

- The output of the phase detector
- The output of the VCO

Dialog Box


## Lowpass filter numerator

The numerator of the lowpass filter transfer function, represented as a vector that lists the coefficients in order of descending powers of $s$.

## Lowpass filter denominator

The denominator of the lowpass filter transfer function, represented as a vector that lists the coefficients in order of descending powers of $s$.

## Linearized Baseband PLL

## VCO input sensitivity ( $\mathrm{Hz} / \mathrm{V}$ )

This value scales the input to the VCO and, consequently, the shift from the VCO's quiescent frequency.

See Also Baseband PLL, Phase-Locked Loop
References For more information about phase-locked loops, see the works listed in "Selected Bibliography for Synchronization" in Communications System Toolbox User's Guide.

## LMS Decision Feedback Equalizer

| Purpose | Equalize using decision feedback equalizer that updates weights with LMS algorithm |
| :---: | :---: |
| Library | Equalizers |
| Description | The LMS Decision Feedback Equalizer block uses a decision feedback equalizer and the LMS algorithm to equalize a linearly modulated |
| Epit Emalized | baseband signal through a dispersive channel. During the simulation, the block uses the LMS algorithm to update the weights, once per |
|  | bol. If the Number of samples per symbol parameter is 1 , then |

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.

Set the Reference tap parameter so it is greater than zero and less than the value for the Number of forward taps parameter.

The port labeled Equalized outputs the result of the equalization process.

You can configure the block to have one or more of these extra ports:

- Mode input, as described in "Reference Signal and Operation Modes" in Communications System Toolbox User's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.
- Weights output, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.


## LMS Decision Feedback Equalizer

## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Using Adaptive Equalizers" in Communications System Toolbox User's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to
$1+($ Reference tap-1)/(Number of samples per symbol)
Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

## LMS Decision Feedback Equalizer



## Number of forward taps

The number of taps in the forward filter of the decision feedback equalizer.

## LMS Decision Feedback Equalizer

## Number of feedback taps

The number of taps in the feedback filter of the decision feedback equalizer.

## Number of samples per symbol

The number of input samples for each symbol.

## Signal constellation

A vector of complex numbers that specifies the constellation for the modulation.

## Reference tap

A positive integer less than or equal to the number of forward taps in the equalizer.

## Step size

The step size of the LMS algorithm.

## Leakage factor

The leakage factor of the LMS algorithm, a number between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

## Initial weights

A vector that concatenates the initial weights for the forward and feedback taps.

## Mode input port

If you select this check box, the block has an input port that enables you to toggle between training and decision-directed mode. For training, the mode input must be 1, and for decision directed, the mode must be 0 . For every frame in which the mode input is 1 or not present, the equalizer trains at the beginning of the frame for the length of the desired signal.

## Output error

If you select this check box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## LMS Decision Feedback Equalizer

## Output weights

If you select this check box, the block outputs the current forward and feedback weights, concatenated into one vector.

## References

See Also
[1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, Wiley, 1998.
[2] Haykin, Simon, Adaptive Filter Theory, Third Ed., Upper Saddle River, N.J., Prentice-Hall, 1996.
[3] Kurzweil, Jack, An Introduction to Digital Communications, New York, Wiley, 2000.
[4] Proakis, John G., Digital Communications, Fourth Ed., New York, McGraw-Hill, 2001.

LMS Linear Equalizer, Normalized LMS Decision Feedback Equalizer, Sign LMS Decision Feedback Equalizer, Variable Step LMS Decision Feedback Equalizer, RLS Decision Feedback Equalizer, CMA Equalizer

## LMS Linear Equalizer

## Purpose

## Library

Description


Equalize using linear equalizer that meditorsupdates weights with LMS algorithm

## Equalizers

The LMS Linear Equalizer block uses a linear equalizer and the LMS algorithm to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the LMS algorithm to update the weights, once per symbol. When you set the Number of samples per symbol parameter to 1, then the block implements a symbol-spaced (i.e. T-spaced) equalizer. When you set the Number of samples per symbol parameter to a value greater than one, the block updates the weights once every $N^{\text {th }}$ sample for a $\mathrm{T} / \mathrm{N}$-spaced equalizer.

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.

Set the Reference tap parameter so it is greater than zero and less than the value for the Number of taps parameter.

The Equalized port outputs the result of the equalization process.
You can configure the block to have one or more of these extra ports:

- Mode input, as described in "Reference Signal and Operation Modes" in Communications System Toolbox User's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.
- Weights output, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.


## LMS Linear Equalizer

## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Using Adaptive Equalizers" in Communications System Toolbox User's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to

## $1+($ Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## LMS Linear Equalizer

Dialog
Box


## Number of taps

The number of taps in the filter of the linear equalizer.

## Number of samples per symbol

The number of input samples for each symbol.

## LMS Linear Equalizer

## Signal constellation

A vector of complex numbers that specifies the constellation for the modulated signal, as determined by the modulator in your model

## Reference tap

A positive integer less than or equal to the number of taps in the equalizer.

## Step size

The step size of the LMS algorithm.

## Leakage factor

The leakage factor of the LMS algorithm, a number between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

## Initial weights

A vector that lists the initial weights for the taps.

## Mode input port

If you select this check box, the block has an input port that allows you to toggle between training and decision-directed mode. For training, the mode input must be 1 , and for decision directed, the mode must be 0 . For every frame in which the mode input is 1 or not present, the equalizer trains at the beginning of the frame for the length of the desired signal.

## Output error

If you select this check box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## Output weights

If you select this check box, the block outputs the current weights.
Examples See "Implement LMS Linear Equalizer Using Simulink" for an example that uses this block.

## LMS Linear Equalizer

References<br>See Also<br>LMS Decision Feedback Equalizer, Normalized LMS Linear Equalizer, Sign LMS Linear Equalizer, Variable Step LMS Linear Equalizer, RLS Linear Equalizer, CMA Equalizer

## Matrix Deinterleaver

$$
\begin{array}{ll}
\text { Purpose } & \begin{array}{l}
\text { Permute input symbols by filling matrix by columns and emptying it } \\
\text { by rows }
\end{array} \\
\text { Library } & \begin{array}{l}
\text { Block sublibrary of Interleaving }
\end{array} \\
\text { Description } \quad \begin{array}{l}
\text { The Matrix Deinterleaver block performs block deinterleaving by filling } \\
\text { a matrix with the input symbols column by column and then sending } \\
\text { the matrix contents to the output port row by row. The Number of } \\
\text { rows and Number of columns parameters are the dimensions of the } \\
\text { matrix that the block uses internally for its computations. }
\end{array} \\
\begin{array}{l}
\text { Deinterleaver }
\end{array} \\
\begin{array}{l}
\text { This block accepts a column vector input signal. The length of the input } \\
\text { vector must be Number of rows times Number of columns. } \\
\text { The block accepts the following data types: int8, uint8, int16, uint16, } \\
\text { int32, uint32, boolean, single, double, and fixed-point. The output } \\
\text { signal inherits its data type from the input signal. }
\end{array}
\end{array}
$$

## Matrix Deinterleaver



Dialog Box

## Number of rows

The number of rows in the matrix that the block uses for its computations.

## Number of columns

The number of columns in the matrix that the block uses for its computations.

## Examples

## Pair Block Matrix Interleaver

See Also
If the Number of rows and Number of columns parameters are 2 and 3 , respectively, then the deinterleaver uses a 2 -by- 3 matrix for its internal computations. Given an input signal of $[1 ; 2 ; 3 ; 4 ; 5 ; 6]$, the block produces an output of $[1 ; 3 ; 5 ; 2 ; 4 ; 6]$.

General Block Deinterleaver

## Matrix Helical Scan Deinterleaver


#### Abstract

Purpose Library Description

Matrix Helical Scan Deinterleaver

Restore ordering of input symbols by filling matrix along diagonals Block sublibrary of Interleaving The Matrix Helical Scan Deinterleaver block performs block deinterleaving by filling a matrix with the input symbols in a helical fashion and then sending the matrix contents to the output port row by row. The Number of rows and Number of columns parameters are the dimensions of the matrix that the block uses internally for its computations.

Helical fashion means that the block places input symbols along diagonals of the matrix. The number of elements in each diagonal matches the Number of columns parameter, after the block wraps past the edges of the matrix when necessary. The block traverses diagonals so that the row index and column index both increase. Each diagonal after the first one begins one row below the first element of the previous diagonal.

The Array step size parameter is the slope of each diagonal, that is, the amount by which the row index increases as the column index increases by one. This parameter must be an integer between zero and the Number of rows parameter. If the Array step size parameter is zero, then the block does not deinterleave and the output is the same as the input. This block accepts a column vector input signal. The number of elements of the input vector must be the product of Number of rows and Number of columns.

The block accepts the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.


## Matrix Helical Scan Deinterleaver



Dialog Box

## Number of rows

The number of rows in the matrix that the block uses for its computations.

## Number of columns

The number of columns in the matrix that the block uses for its computations.

## Array step size

The slope of the diagonals that the block writes.

## Matrix Helical Scan Deinterleaver

Pair Block Matrix Helical Scan Interleaver<br>See Also General Block Deinterleaver

## Purpose

Permute input symbols by selecting matrix elements along diagonals

## Library

Description

Block sublibrary of Interleaving
The Matrix Helical Scan Interleaver block performs block interleaving by filling a matrix with the input symbols row by row and then sending the matrix contents to the output port in a helical fashion. The Number of rows and Number of columns parameters are the dimensions of
the matrix that the block uses internally for its computations.

Helical fashion means that the block selects output symbols by selecting elements along diagonals of the matrix. The number of elements in each diagonal matches the Number of columns parameter, after the block wraps past the edges of the matrix when necessary. The block traverses diagonals so that the row index and column index both increase. Each diagonal after the first one begins one row below the first element of the previous diagonal.
The Array step size parameter is the slope of each diagonal, that is, the amount by which the row index increases as the column index increases by one. This parameter must be an integer between zero and the Number of rows parameter. If the Array step size parameter is zero, then the block does not interleave and the output is the same as the input.

This block accepts a column vector input signal. The number of elements of the input vector must be the product of Number of rows and Number of columns.

The block accepts the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.

## Matrix Helical Scan Interleaver

Dialog Box

> Function Block Parameters: Matrix Helical Scan Interleaver $x$ -Matrix Helical Scan Interleaver (mask) (link)

> Interleave input vector by writing elements row-by-row into an array with a specified number of rows and columns and then reading them out by scanning along diagonals of this array. The pitch of the diagonal scans is determined by the array step size.

> The array step size must be a nonnegative integer less than the specified number of rows. An array step size of zero indicates no interleaving.

> The product of Number of rows and Number of columns must match the input signal width.
Parameters
Number of rows:
54
Number of columns:
64
Array step size:
1

## Number of rows

The number of rows in the matrix that the block uses for its computations.

## Number of columns

The number of columns in the matrix that the block uses for its computations.

## Array step size

The slope of the diagonals that the block reads.

## Matrix Helical Scan Interleaver



## Matrix Interleaver

Purpose Permute input symbols by filling matrix by rows and emptying it by columns

## Library

Block sublibrary of Interleaving
Description Interleaver

The Matrix Interleaver block performs block interleaving by filling a matrix with the input symbols row by row and then sending the matrix contents to the output port column by column.

The Number of rows and Number of columns parameters are the dimensions of the matrix that the block uses internally for its computations.

This block accepts a column vector input signal. The number of elements of the input vector must be the product of Number of rows and Number of columns.

The block accepts the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.

## Matrix Interleaver



Dialog
Box

## Number of rows

The number of rows in the matrix that the block uses for its computations.

## Number of columns

The number of columns in the matrix that the block uses for its computations.

## Examples

## See Also

If the Number of rows and Number of columns parameters are 2 and 3 , respectively, then the interleaver uses a 2 -by- 3 matrix for its internal computations. Given an input signal of [1; 2; 3; 4; 5; 6], the block produces an output of $[1 ; 4 ; 2 ; 5 ; 3 ; 6]$.

## Pair Block <br> Matrix Deinterleaver

General Block Interleaver

## M-DPSK Demodulator Baseband

## Purpose <br> Library <br> Description <br> Wunt- <br> M-DPSK

Demodulate DPSK-modulated data

PM, in Digital Baseband sublibrary of Modulation
The M-DPSK Demodulator Baseband block demodulates a signal that was modulated using the M-ary differential phase shift keying method. The input is a baseband representation of the modulated signal. The input and output for this block are discrete-time signals. This block accepts a scalar-valued or column vector input signal. For information about the data types each block port supports, see the "Supported Data Types" on page 2-506 table on this page.

The M-ary number parameter, M, is the number of possible output symbols that can immediately follow a given output symbol. The block compares the current symbol to the previous symbol. The block's first output is the initial condition of zero (or a group of zeros, if the Output type parameter is set to Bit) because there is no previous symbol.

## Integer-Valued Signals and Binary-Valued Signals

If you set the Output type parameter to Integer, then the block demodulates a phase difference of

$$
\theta+2 \pi k / M
$$

to $k$, where $\theta$ represents the Phase rotation parameter and $k$ represents an integer between 0 and $M-1$.

When you set the Output type parameter to Bit, the block outputs binary-valued signals that represent integers. The block represents each integer using a group of $K=\log _{2}(M)$ bits, where $K$ represents the number of bits per symbol. The output vector length must be an integer multiple of $K$.
In binary output mode, the symbols can be either binary-demapped or Gray-demapped. The Constellation ordering parameter indicates how the block maps an integer to a corresponding group of $K$ output bits. See the reference pages for the M-DPSK Modulator Baseband and M-PSK Modulator Baseband blocks for details.

## M-DPSK Demodulator Baseband

Dialog Box


## M-ary number

The number of possible modulated symbols that can immediately follow a given symbol.

## Output type

Determines whether the output consists of integers or groups of bits.

## Constellation ordering

Determines how the block maps each integer to a group of output bits.

## M-DPSK Demodulator Baseband

## Phase rotation (rad)

This phase difference between the current and previous modulated symbols that results in an output of zero.

## Output data type

When the parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is of type single or double.

For integer outputs, this block can output the data types int8, uint8, int16, uint16, int32, uint32, single, and double. For bit outputs, output can be int8, uint8, int16, uint16, int32, uint32, boolean, single, or double.

Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | - Single-precision floating point |
| Output | • Double-precision floating point |
|  | - Single-precision floating point |
|  | - Boolean when Output type set to Bit |
|  | - 8-, 16-, and 32-bit signed integers |
|  | • 8-, 16-, and 32-bit unsigned integers |

[^1]
# M-DPSK Modulator Baseband 

## Purpose

Modulate using M-ary differential phase shift keying method

## Library

PM, in Digital Baseband sublibrary of Modulation
Description
ㄴNuM M-DPSK

The M-DPSK Modulator Baseband block modulates using the M-ary differential phase shift keying method. The output is a baseband representation of the modulated signal. The M-ary number parameter,

M , is the number of possible output symbols that can immediately follow a given output symbol.

The input must be a discrete-time signal. For integer inputs, the block can accept the data types int8, uint8, int16, uint16, int32, uint32, single, and double. For bit inputs, the block can accept int8, uint8, int16, uint16, int32, uint32, boolean, single, and double.

The input can be either bits or integers, which are binary-mapped or Gray-mapped into symbols.

This block accepts column vector input signals. For a bit input, the input width must be an integer multiple of the number of bits per symbol.

## Integer-Valued Signals and Binary-Valued Signals

If you set the Input type parameter to Integer, then valid input values are integers between 0 and $\mathrm{M}-1$. In this case, the input can be either a scalar or a frame-based column vector. If the first input is $\mathrm{k}_{1}$, then the modulated symbol is

$$
\exp \left(j \theta+j 2 \pi \frac{k_{1}}{m}\right)
$$

where $\theta$ represents the Phase rotation parameter. If a successive input is $k$, then the modulated symbol is

$$
\exp \left(j \theta+j 2 \pi \frac{k}{m}\right) \cdot(\text { previous modulated symbol })
$$

## M-DPSK Modulator Baseband

When you set the Input type parameter to Bit, the block accepts binary-valued inputs that represent integers. The block collects binary-valued signals into groups of $K=\log _{2}(M)$ bits
where
$K$ represents the number of bits per symbol.
The input vector length must be an integer multiple of $K$. In this configuration, the block accepts a group of $K$ bits and maps that group onto a symbol at the block output. The block outputs one modulated symbol for each group of $K$ bits.
The input can be a column vector with a length that is an integer multiple of $K$.
In binary input mode, the Constellation ordering parameter indicates how the block maps a group of $K$ input bits to a corresponding phase difference. The Binary option uses a natural binary-to-integer mapping, while the Gray option uses a Gray-coded assignment of phase differences. For example, the following table indicates the assignment of phase difference to three-bit inputs, for both the Binary and Gray options. $\theta$ is the Phase rotation parameter. The phase difference is between the previous symbol and the current symbol.

| Current Input | Binary-Coded <br> Phase Difference | Gray-Coded Phase <br> Difference |
| :--- | :--- | :--- |
| $\left[\begin{array}{lll}0 & 0 & 0\end{array}\right]$ | $\mathrm{j} \theta$ | $\mathrm{j} \theta$ |
| $\left[\begin{array}{lll}0 & 0 & 1\end{array}\right]$ | $\mathrm{j} \theta+\mathrm{j} \Pi / 4$ | $\mathrm{j} \theta+\mathrm{j} \Pi / 4$ |
| $\left[\begin{array}{lll}0 & 1 & 0\end{array}\right]$ | $\mathrm{j} \theta+\mathrm{j} \Pi 2 / 4$ | $\mathrm{j} \theta+\mathrm{j} \Pi 3 / 4$ |
| $\left[\begin{array}{lll}0 & 1 & 1\end{array}\right]$ | $\mathrm{j} \theta+\mathrm{j} \Pi 3 / 4$ | $\mathrm{j} \theta+\mathrm{j} \Pi 2 / 4$ |
| $\left[\begin{array}{lll}1 & 0 & 0\end{array}\right]$ | $\mathrm{j} \theta+\mathrm{j} \Pi 4 / 4$ | $\mathrm{j} \theta+\mathrm{j} \Pi 7 / 4$ |
| $\left[\begin{array}{lll}1 & 0 & 1\end{array}\right]$ | $\mathrm{j} \theta+\mathrm{j} \Pi 5 / 4$ | $\mathrm{j} \theta+\mathrm{j} \Pi 6 / 4$ |
| $\left[\begin{array}{lll}1 & 1 & 0\end{array}\right]$ | $\mathrm{j} \theta+\mathrm{j} \Pi 6 / 4$ | $\mathrm{j} \theta+\mathrm{j} \Pi 4 / 4$ |
| $\left[\begin{array}{lll}1 & 1 & 1\end{array}\right]$ | $\mathrm{j} \theta+\mathrm{j} \Pi 7 / 4$ | $\mathrm{j} \theta+\mathrm{j} \Pi 5 / 4$ |

## M-DPSK Modulator Baseband

For more details about the Binary and Gray options, see the reference page for the M-PSK Modulator Baseband block. The signal constellation for that block corresponds to the arrangement of phase differences for this block.

## Dialog Box



## M-ary number

The number of possible output symbols that can immediately follow a given output symbol.

## M-DPSK Modulator Baseband

## Input type

Indicates whether the input consists of integers or groups of bits. If this parameter is set to Bit, then the M-ary number parameter must be $2^{\mathrm{K}}$ for some positive integer K .

## Constellation ordering

Determines how the block maps each group of input bits to a corresponding integer.

## Phase rotation (rad)

The phase difference between the previous and current modulated symbols when the input is zero.

## Output data type

The output data type can be either single or double. By default, the block sets this to double.

Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | - Single-precision floating point |
|  | - Boolean (binary input mode only) |
|  | - 8-, 16-, and 32-bit signed integers |
|  | - 8-, 16-, and 32 -bit unsigned integers |
| Output | - Double-precision floating point |
|  | - Single-precision floating point |

## Pair Block

M-DPSK Demodulator Baseband

See Also

DBPSK Modulator Baseband, DQPSK Modulator Baseband, M-PSK Modulator Baseband

## M-DPSK Modulator Baseband

References [1] Pawula, R. F., "On M-ary DPSK Transmission Over Terrestrial and Satellite Channels," IEEE Transactions on Communications, Vol. COM-32, July 1984, 752-761.

## Memoryless Nonlinearity

| Purpose | Apply memoryless nonlinearity to complex baseband signal |
| :--- | :--- |
| Library | RF Impairments |
| Description | The Memoryless Nonlinearity block applies a memoryless nonlinearity <br> to a complex, baseband signal. You can use the block to model radio <br> frequency (RF) impairments to a signal at the receiver. |
| cubio <br> Polynomial | This block accepts a column vector input signal. |

Note All values of power assume a nominal impedance of 1 ohm .

The Memoryless Nonlinearity block provides five different methods for modeling the nonlinearity, which you specify by the Method parameter. The options for the Method parameter are

- Cubic polynomial
- Hyperbolic tangent
- Saleh model
- Ghorbani model
- Rapp model

The bock implements these five methods using subsystems underneath the block mask. For each of the first four methods, the nonlinearity subsystem has the same basic structure, as shown in the following figure.


## Nonlinearity Subsystem

For the first four methods, each subsystem applies a nonlinearity to the input signal as follows:

1 Multiply the signal by a gain factor.
2 Split the complex signal into its magnitude and angle components.
3 Apply an AM/AM conversion to the magnitude of the signal, according to the selected Method, to produce the magnitude of the output signal.

4 Apply an AM/PM conversion to the phase of the signal, according to the selected Method, and adds the result to the angle of the signal to produce the angle of the output signal.

5 Combine the new magnitude and angle components into a complex signal and multiply the result by a gain factor, which is controlled by the Linear gain parameter.

Each subsystem implements the AM/AM and AM/PM conversions differently, according to the Method you specify. The Rapp model does not apply a phase change to the input signal. The nonlinearity subsystem for Rapp model has following structure:

## Memoryless Nonlinearity



## Nonlinearity Subsystem for Rapp Model

The Rapp Subsystem applies nonlinearity as follows:
1 Multiply the signal by a gain factor.
2 Split the complex signal into its magnitude and angle components.
3 Apply an AM/AM conversion to the magnitude of the signal, according to the selected Method, to produce the magnitude of the output signal.

4 Combine the new magnitude and angle components into a complex signal and multiply the result by a gain factor, which is controlled by the Linear gain parameter.

If you want to see exactly how the Memoryless Nonlinearity block implements the conversions for a specific method, you can view the $\mathrm{AM} / \mathrm{AM}$ and $\mathrm{AM} / \mathrm{PM}$ subsystems that implement these conversions as follows:

1 Right-click on the Memoryless Nonlinearity block and select Mask > Look under mask. This displays the block's configuration underneath the mask. The block contains five subsystems corresponding to the five nonlinearity methods.

2 Double-click the subsystem for the method you are interested in. This displays the subsystem shown in the preceding figure, Nonlinearity Subsystem on page 2-513.

3 Double-click on one of the subsystems labeled AM/AM or AM/PM to view how the block implements the conversions.

## AM/PM Characteristics of the Cubic Polynomial and Hyperbolic Tangent Methods

The following illustration shows the AM/PM behavior for the Cubic polynomial and Hyperbolic tangent methods:


The AM/PM conversion scales linearly with input power value between the lower and upper limits of the input power level (specified by Lower input power limit for AM/PM conversion ( dBm ) and Upper input power limit for $A M / P M$ conversion ( dBm )). Beyond these values, $\mathrm{AM} / \mathrm{PM}$ conversion is constant at the values corresponding to the lower and upper input power limits, which are zero and

## Memoryless Nonlinearity

(AM/PM conversion)•(upper input power limit - lower input power limit) respectively.

## AM/AM and AM/PM Characteristics of the Saleh Method

The following figure shows, for the Saleh method, plots of

- Output voltage against input voltage for the $\mathrm{AM} / \mathrm{AM}$ conversion
- Output phase against input voltage for the AM/PM conversion



## Example with 16-ary QAM

You can see the effect of the Memoryless Nonlinearity block on a signal modulated by 16 -ary quadrature amplitude modulation (QAM) in a scatter plot. The constellation for 16 -ary QAM without the effect of the Memoryless Nonlinearity block is shown in the following figure:


You can generate a scatter plot of the same signal after it passes through the Memoryless Nonlinearity block, with the Method parameter set to Saleh Model, as shown in the following figure.

## Memoryless Nonlinearity



This plot is generated by the model described in "Illustrate RF Impairments That Distort a Signal" with the following parameter settings for the Rectangular QAM Modulator Baseband block:

- Normalization method set to Average Power
- Average power (watts) set to $1 \mathrm{e}-2$

The following sections discuss parameters specific to the Saleh, Ghorbani, and Rapp models.

## Parameters for the Saleh Model

The Input scaling ( $\mathbf{d B}$ ) parameter scales the input signal before the nonlinearity is applied. The block multiplies the input signal by the parameter value, converted from decibels to linear units. If you set the parameter to be the inverse of the input signal amplitude, the scaled signal has amplitude normalized to 1 .

## Memoryless Nonlinearity

The AM/AM parameters, alpha and beta, are used to compute the amplitude gain for an input signal using the following function:

$$
F_{A M / A M}(u)=\frac{\text { alpha } * u}{1+\operatorname{beta}^{*} u^{2}}
$$

where $u$ is the magnitude of the scaled signal.
The AM/PM parameters, alpha and beta, are used to compute the phase change for an input signal using the following function:

$$
F_{A M / P M}(u)=\frac{\text { alpha } * u^{2}}{1+\text { beta } * u^{2}}
$$

where $u$ is the magnitude of the scaled signal. Note that the AM/AM and AM/PM parameters, although similarly named alpha and beta, are distinct.

The Output scaling (dB) parameter scales the output signal similarly.

## Parameters for the Ghorbani Model

The Input scaling (dB) parameter scales the input signal before the nonlinearity is applied. The block multiplies the input signal by the parameter value, converted from decibels to linear units. If you set the parameter to be the inverse of the input signal amplitude, the scaled signal has amplitude normalized to 1 .

The AM/AM parameters, $\left[\mathrm{x}_{1} \mathrm{x}_{2} \mathrm{x}_{3} \mathrm{x}_{4}\right.$ ], are used to compute the amplitude gain for an input signal using the following function:

$$
F_{A M / A M}(u)=\frac{x_{1} u^{x_{2}}}{1+x_{3} u^{x_{2}}}+x_{4} u
$$

where $u$ is the magnitude of the scaled signal.
The AM/PM parameters, $\left[y_{1} y_{2} y_{3} y_{4}\right]$, are used to compute the phase change for an input signal using the following function:

## Memoryless Nonlinearity

$$
F_{A M / P M}(u)=\frac{y_{1} u^{y_{2}}}{1+y_{3} u^{y_{2}}}+y_{4} u
$$

where $u$ is the magnitude of the scaled signal.
The Output scaling (dB) parameter scales the output signal similarly.

## Parameters for the Rapp Model

The Linear gain (dB) parameter scales the input signal before the nonlinearity is applied. The block multiplies the input signal by the parameter value, converted from decibels to linear units. If you set the parameter to be the inverse of the input signal amplitude, the scaled signal has amplitude normalized to 1 .

The Smoothness factor and Output saturation level parameters are used to compute the amplitude gain for the input signal:

$$
F_{A M / A M}(u)=\frac{u}{\left(1+\left(\frac{u}{O_{s a t}}\right)^{2 S}\right)^{1 / 2 S}}
$$

where $u$ is the magnitude of the scaled signal, $S$ is the Smoothness factor, and $O_{\text {sat }}$ is the Output saturation level.

The Rapp model does not apply a phase change to the input signal.
The Output saturation level parameter limits the output signal level.

## Memoryless Nonlinearity

$$
\begin{aligned}
& \text { Function Block Parameters: Memoryless Nonlinearity } \\
& \begin{array}{|l|}
\text { Memoryless Nonlinearity (mask) (link) - } \\
\text { Add memoryless nonlinearity to complex baseband signal. } \\
\text { Two of the five methods (Cubic Polynomial and Hyperbolic Tangent) fit AM/AM } \\
\text { curves to measured data provided by the gain and third order intercept point } \\
\text { (IIP3) parameters. They generate a linear AM/PM characteristic within the user- } \\
\text { specified input power limits. Outside those limits, the AM/PM is constant. } \\
\text { The other three methods use models originated by Saleh, Ghorbani, and Rapp. The } \\
\text { Saleh and Ghorbani models are based on normalized nonlinear transfer functions. } \\
\text { Use the Input scaling and Output scaling parameters to adjust signal levels up or } \\
\text { down from their normalized values. } \\
\text { All values of power assume a nominal impedance of } 1 \text { ohm. This block accepts a } \\
\text { scalar or column vector input signal. }
\end{array}
\end{aligned}
$$



## Method

The nonlinearity method.
The following describes specific parameters for each method.

## Memoryless Nonlinearity



## Linear gain (db)

Scalar specifying the linear gain for the output function.

## IIP3 (dBm)

Scalar specifying the third order intercept.

## AM/PM conversion (degrees per dB)

Scaler specifying the AM/PM conversion in degrees per decibel.

## Lower input power limit (dBm)

Scalar specifying the minimum input power for which AM/PM conversion scales linearly with input power value. Below this value, the phase shift resulting from AM/PM conversion is zero.

## Upper input power limit ( dBm )

Scalar specifying the maximum input power for which AM/PM conversion scales linearly with input power value. Above this value, the phase shift resulting from AM/PM conversion is constant. The value of this maximum shift is given by:
(AM/PM conversion)•(upper input power limit - lower input power limit)


## Linear gain (db)

Scalar specifying the linear gain for the output function.

## IIP3 (dBm)

Scalar specifying the third order intercept.

## AM/PM conversion (degrees per dB)

Scalar specifying the AM/PM conversion in degrees per decibel.

## Lower input power limit (dBm)

Scalar specifying the minimum input power for which AM/PM conversion scales linearly with input power value. Below this value, the phase shift resulting from AM/PM conversion is zero.

## Upper input power limit ( dBm )

Scalar specifying the maximum input power for which AM/PM conversion scales linearly with input power value. Above this value, the phase shift resulting from AM/PM conversion is constant. The value of this maximum shift is given by:
(AM/PM conversion)•(upper input power limit - lower input power limit)

## Memoryless Nonlinearity

```
Parameters
Method: Saleh model 
Input scaling (dB):
0
AM/AM parameters [alpha beta]:
[2.15871.1517]
AM/PM parameters [alpha beta]:
[4.0033 9.1040]
Output scaling (dB):
0
```


## Input scaling (dB)

Number that scales the input signal level.

## AM/AM parameters [alpha beta]

Vector specifying the AM/AM parameters.

## AM/PM parameters [alpha beta]

Vector specifying the AM/PM parameters.
Output scaling (dB)
Number that scales the output signal level.

| Parameters |
| :--- |
| Method: Ghorbani model |
| Input scaling (dB): |
| 0 |
| AM/AM parameters $[\times 1 \times 2 \times 3 \times 4]$ : |
| [8.1081 $1.54136 .5202-0.0718]$ |
| AM/PM parameters $[y 1 \mathrm{y} 2 \mathrm{y} 3 \mathrm{y} 4]$ : |
| [4.6645 2.0965 10.88-0.003] |
| Output scaling (dB): |
| 0 |

## Input scaling (dB)

Number that scales the input signal level.
AM/AM parameters [ $\mathrm{x} 1 \times 2 \times 3 \times 4$ ]
Vector specifying the AM/AM parameters.

## AM/PM parameters [y1 y2 y3 y4]

Vector specifying the AM/PM parameters.

## Output scaling (dB)

Number that scales the output signal level.

| Parameters |
| :--- |
| Method: $\mid$ Rapp model |
| Linear gain (dB): |
| 0 |
| Smoothness factor: |
| 0.5 |
| Output saturation level: |
| 1 |

## Linear gain (db)

Scalar specifying the linear gain for the output function.

## Smoothness factor

Scalar specifying the smoothness factor

## Output saturation level

Scalar specifying the the output saturation level.

## See Also <br> I/Q Imbalance

## Reference

[1] Saleh, A.A.M., "Frequency-independent and frequency-dependent nonlinear models of TWT amplifiers," IEEE Trans. Communications, vol. COM-29, pp.1715-1720, November 1981.
[2] A. Ghorbani, and M. Sheikhan, "The effect of Solid State Power Amplifiers (SSPAs) Nonlinearities on MPSK and M-QAM Signal Transmission", Sixth Int'l Conference on Digital Processing of Signals in Comm., 1991, pp. 193-197.
[3] C. Rapp, "Effects of HPA-Nonlinearity on a 4-DPSK/OFDM-Signal for a Digitial Sound Broadcasting System", in Proceedings of the Second

## Memoryless Nonlinearity

European Conference on Satellite Communications, Liege, Belgium, Oct. 22-24, 1991, pp. 179-184.

# M-FSK Demodulator Baseband 

## Purpose

Demodulate FSK-modulated data

## Library

Description
MMW-5
M-FSK
FM, in Digital Baseband sublibrary of Modulation

The M-FSK Demodulator Baseband block demodulates a signal that was modulated using the M-ary frequency shift keying method. The input is a baseband representation of the modulated signal. The input and output for this block are discrete-time signals. This block accepts a scalar value or column vector input signal of type single or double. For information about the data types each block port supports, see "Supported Data Types" on page 2-533.

The M-ary number parameter, $M$, is the number of frequencies in the modulated signal. The Frequency separation parameter is the distance, in Hz , between successive frequencies of the modulated signal.

The M-FSK Demodulator Baseband block implements a non-coherent energy detector. To obtain the same BER performance as that of coherent FSK demodulation, use the CPFSK Demodulator Baseband block.

## Integer-Valued Signals and Binary-Valued Signals

When you set the Output type parameter to Integer, the block outputs integer values between 0 and $M-1 . M$ represents the $\mathbf{M}$-ary number block parameter.

When you set the Output type parameter to Bit, the block outputs binary-valued signals that represent integers. The block represents each integer using a group of $K=\log _{2}(M)$ bits, where $K$ represents the number of bits per symbol. The output vector length must be an integer multiple of $K$.

The Symbol set ordering parameter indicates how the block maps a symbol to a group of $K$ output bits. When you set the parameter to Binary, the block maps the integer, I, to $[u(1) u(2) \ldots u(\mathrm{~K})]$ bits, where the individual $u(i)$ are given by

## M-FSK Demodulator Baseband

$$
I=\sum_{i=1}^{K} u(i) 2^{K-i}
$$

$u(1)$ is the most significant bit.
For example, if $M=8$, you set Symbol set ordering to Binary, and the demodulated integer symbol value is 6 , then the binary output word is $\left[\begin{array}{lll}1 & 1 & 0\end{array}\right]$.
When you set Symbol set ordering to Gray, the block assigns binary outputs from points of a predefined Gray-coded signal constellation. The predefined M-ary Gray-coded signal constellation assigns the binary representation

```
M = 8; P = [0:M-1]';
de2bi(bitxor(P,floor(P/2)), log2(M),'left-msb')
```

to the $P^{\text {th }}$ integer.
The typical Binary to Gray mapping for $M=8$ is shown in the following tables.

## Binary to Gray Mapping for Bits

| Binary Code | Gray Code |
| :--- | :--- |
| 000 | 000 |
| 001 | 001 |
| 010 | 011 |
| 011 | 010 |
| 100 | 110 |
| 101 | 111 |
| 110 | 101 |
| 111 | 100 |

## M-FSK Demodulator Baseband

## Binary to Gray Mapping for Integers

| Binary Code | Gray Code |
| :--- | :--- |
| 0 | 0 |
| 1 | 1 |
| 2 | 3 |
| 3 | 2 |
| 4 | 6 |
| 5 | 7 |
| 6 | 5 |
| 7 | 4 |

Whether the output is an integer or a binary representation of an integer, the block maps the highest frequency to the integer 0 and maps the lowest frequency to the integer M-1. In baseband simulation, the lowest frequency is the negative frequency with the largest absolute value.

## Single-Rate Processing

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. The input width must be an integer multiple of the Samples per symbol parameter value, and the input can be a column vector.

- When you set Output type to Bit, the output width is $K$ times the number of input symbols.
- When you set Output type to Integer, the output width is the number of input symbols.


## Multirate Processing

In multirate processing mode, the input and output signals have different port sample times. The input must be a scalar. The output

## M-FSK Demodulator Baseband

symbol time is the product of the input sample time and the Samples per symbol parameter value.

- When you set Output type to Bit, the output width equals the number of bits per symbol.
- When you set Output type to Integer, the output is a scalar.

To run the M-FSK Demodulator block in multirate mode, set Tasking mode for periodic sample times (in Simulation > Configuration Parameters > Solver) to SingleTasking.

## M-FSK Demodulator Baseband



## Dialog Box

## M-ary number

The number of frequencies in the modulated signal.

## Output type

Determines whether the output consists of integers or groups of bits. If this parameter is set to Bit, then the M-ary number parameter must be $2^{\mathrm{K}}$ for some positive integer K .

## M-FSK Demodulator Baseband

## Symbol set ordering

Determines how the block maps each integer to a group of output bits.

## Frequency separation (Hz)

The distance between successive frequencies in the modulated signal.

## Samples per symbol

The number of input samples that represent each modulated symbol.

## Rate options

Select the rate processing method for the block.

- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample times. The block implements the rate change by making a size change at the output when compared to the input. The output width is the number of symbols (which is given by dividing the input length by the Samples per symbol parameter value when the Output type parameter is set to Integer).
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output period is the same as the symbol period and equals the product of the input period and the Samples per symbol parameter value.

> Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

For more information, see Single-Rate Processing and Multirate Processing in the Description section of this page.

## M-FSK Demodulator Baseband

## Output data type

The output type of the block can be specified here as boolean, int8, uint8, int16, uint16, int32, uint32, or double. By default, the block sets this to double.

## Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | $\bullet$ Double-precision floating point |
|  | $\bullet$ Single-precision floating point |
| Output | $\bullet$ Double-precision floating point |
|  | $\bullet$ Boolean |
|  | $\bullet 8-, 16$-, and 32 -bit signed integers |
|  | $\bullet 8-, 16$-, and 32 -bit unsigned integers |

## Pair Block M-FSK Modulator Baseband

See Also CPFSK Demodulator Baseband

## M-FSK Modulator Baseband

| Purpose | Modulate using M-ary frequency shift keying method |
| :---: | :---: |
| Library | FM, in Digital Baseband sublibrary of Modulation |
| Descripti | The M-FSK Modulator Baseband block modulates using the M-ary frequency shift keying method. The output is a baseband representation of the modulated signal. For information about the data types each block port supports, see "Supported Data Types" on page 2-540. |
|  | To prevent aliasing from occurring in the output signal, set the sampling frequency greater than the product of $M$ and the Frequency separation parameter. Sampling frequency is Samples per symbol divided by the input symbol period (in seconds). |
|  | Integer-Valued Signals and Binary-Valued Signals |
|  | The input and output signals for this block are discrete-time signals. |
|  | When you set the Input type parameter to Integer, the block accepts integer values between 0 and $M-1$. $M$ represents the M-ary number block parameter. |
|  | When you set the Input type parameter to Bit, the block accepts binary-valued inputs that represent integers. The block collects binary-valued signals into groups of $K=\log _{2}(M)$ bits |
|  | where |
|  | $K$ represents the number of bits per symbol. |
|  | The input vector length must be an integer multiple of $K$. In this configuration, the block accepts a group of $K$ bits and maps that group onto a symbol at the block output. The block outputs one modulated symbol, oversampled by the Samples per symbol parameter value, for each group of $K$ bits. |
|  | The Symbol set ordering parameter indicates how the block maps a group of $K$ input bits to a corresponding symbol. When you set the parameter to Binary, the block maps $[u(1) u(2) \ldots u(\mathrm{~K})]$ to the integer |

## M-FSK Modulator Baseband

$$
\sum_{i=1}^{K} u(i) 2^{K-i}
$$

and assumes that this integer is the input value. $u(1)$ is the most significant bit.

If you set $M=8$, Symbol set ordering to Binary, and the binary input word is [ $\left.\begin{array}{lll}1 & 1 & 0\end{array}\right]$, the block converts $\left[\begin{array}{lll}1 & 1 & 0\end{array}\right]$ to the integer 6 . The block produces the same output when the input is 6 and the Input type parameter is Integer.

When you set Symbol set ordering to Gray, the block uses a Gray-coded arrangement and assigns binary inputs to points of a predefined Gray-coded signal constellation. The predefined M-ary Gray-coded signal constellation assigns the binary representation

```
M = 8; P = [0:M-1]';
de2bi(bitxor(P,floor(P/2)), log2(M),'left-msb')
to the \(P^{\text {th }}\) integer.
```

The following tables show the typical Binary to Gray mapping for $M=8$.

## Binary to Gray Mapping for Bits

| Binary Code | Gray Code |
| :--- | :--- |
| 000 | 000 |
| 001 | 001 |
| 010 | 011 |
| 011 | 010 |
| 100 | 110 |
| 101 | 111 |
| 110 | 101 |
| 111 | 100 |

## M-FSK Modulator Baseband

Binary to Gray Mapping for Integers

| Binary Code | Gray Code |
| :--- | :--- |
| 0 | 0 |
| 1 | 1 |
| 2 | 3 |
| 3 | 2 |
| 4 | 6 |
| 5 | 7 |
| 6 | 5 |
| 7 | 4 |

## Single-Rate Processing

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. In this mode, the input to the block can be multiple symbols.

- When you set Input type to Integer, the input can be a column vector, the length of which is the number of input symbols.
- When you set Input type to Bit, the input width must be an integer multiple of $K$, the number of bits per symbol.

The output width equals the product of the number of input symbols and the Samples per symbol parameter value.

## Multirate Processing

In multirate processing mode, the input and output signals have different port sample times. In this mode, the input to the block must be one symbol.

- When you set Input type to Integer, the input must be a scalar.
- When you set Input type to Bit, the input width must equal the number of bits per symbol.

The output sample time equals the symbol period divided by the Samples per symbol parameter value.

To run the M-FSK Modulator block in multirate mode, set Tasking mode for periodic sample times (in Simulation > Configuration Parameters > Solver) to SingleTasking.

## M-FSK Modulator Baseband

| Function Block Parameters: M-FSK Modulator Baseband |
| :--- |
| M-FSK Modulator Baseband (mask) (ink) -  <br> Modulate the input signal using the frequency shift keying method.  <br> The input signal can be either bits or integers. For the single-rate processing option  <br> with bit inputs, the input width must be an integer multiple of the number of bits per  <br> symbol. For the multirate processing option with bit inputs, the input width must equal  <br> the number of bits per symbol.  <br> For the single-rate processing option with integer inputs, this block accepts a scalar or  <br> column vector input signal. For the multirate processing option with integer inputs, this  <br> block accepts a scalar input signal.  <br> Parameters -  <br> M-ary number:  <br> 8 Input type: Integer <br> Symbol set ordering: Gray  <br> Frequency separation (Hz):  <br> 6 Cancel <br> Phase continuity: Continuous  <br> Samples per symbol:  <br> 17 Help <br> Rate options: Enforce single-rate processing  <br> Output data type: double  |

## M-ary number

The number of frequencies in the modulated signal.

## Input type

Indicates whether the input consists of integers or groups of bits. If you set this parameter to Bit, then the M-ary number parameter must be $2^{\mathrm{K}}$ for some positive integer K .

## Symbol set ordering

Determines how the block maps each group of input bits to a corresponding integer.
Frequency separation (Hz)
The distance between successive frequencies in the modulated signal.

## Phase continuity

Determines whether the modulated signal changes phases in a continuous or discontinuous way.

If you set the Phase continuity parameter to Continuous, then the modulated signal maintains its phase even when it changes its frequency. If you set the Phase continuity parameter to Discontinuous, then the modulated signal comprises portions of $M$ sinusoids of different frequencies. Thus, a change in the input value sometimes causes a change in the phase of the modulated signal.

## Samples per symbol

The number of output samples that the block produces for each integer or binary word in the input.

## Rate options

Select the rate processing option for the block.

- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width equals the product of the number of symbols and the Samples per symbol parameter value.
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output sample time equals the symbol period divided by the Samples per symbol parameter value.


## M-FSK Modulator Baseband

> Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

## Output data type

You can specify the output type of the block as either a double or a single. By default, the block sets this value to double.

## Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | - Double-precision floating point |
|  | - Boolean (bit input mode only) |
|  | - 8-, 16-, and 32-bit signed integers (integer input mode |
|  | only) |
|  | 8-, 16-, and 32 -bit unsigned integers (integer input mode <br> only) |
| Output | - Double-precision floating point |
|  | - Single-precision floating point |

Pair Block<br>M-FSK Demodulator Baseband<br>See Also CPFSK Modulator Baseband

## Purpose

Equalize using Viterbi algorithm

## Library

Description modeled as a finite input response (FIR) filter.

The MLSE Equalizer block uses the Viterbi algorithm to equalize a linearly modulated signal through a dispersive channel. The block processes input frames and outputs the maximum likelihood sequence estimate (MLSE) of the signal, using an estimate of the channel

This block supports single and double data types.

## Channel Estimates

The channel estimate takes the form of a column vector containing the coefficients of an FIR filter in descending order of powers. The length of this vector is the channel memory, which must be a multiple of the block's Samples per input symbol parameter.

To specify the channel estimate vector, use one of these methods:

- Set Specify channel via to Dialog and enter the vector in the Channel coefficients field.
- Set Specify channel via to Input port and the block displays an additional input port, labeled Ch, which accepts a column vector input signal.


## Signal Constellation

The Signal constellation parameter specifies the constellation for the modulated signal, as determined by the modulator in your model. Signal constellation is a vector of complex numbers, where the kth complex number in the vector is the constellation point to which the modulator maps the integer k-1.

Note The sequence of constellation points must be consistent between the modulator in your model and the Signal constellation parameter in this block.

For example, to specify the constellation given by the mapping

$$
\begin{aligned}
& 0 \rightarrow+1+i \\
& 1 \rightarrow-1+i \\
& 2 \rightarrow-1-i \\
& 3 \rightarrow+1-i
\end{aligned}
$$

set Constellation points to [1+i, -1+i, -1-i, 1-i]. Note that the sequence of numbers in the vector indicates how the modulator maps integers to the set of constellation points. The labeled constellation is shown below.


## MLSE Equalizer

## Preamble and Postamble

If your data is accompanied by a preamble (prefix) or postamble (suffix), then configure the block accordingly:

- If you select Input contains preamble, then the Expected preamble parameter specifies the preamble that you expect to precede the data in the input signal.
- If you check the Input contains postamble, then the Expected postamble parameter specifies the postamble that you expect to follow the data in the input signal.

The Expected preamble or Expected postamble parameter must be a vector of integers between 0 and $\mathrm{M}-1$, where M is the number of constellation points. An integer value of k-1 in the vector corresponds to the kth entry in the Constellation points vector and, consequently, to a modulator input of $\mathrm{k}-1$.

The preamble or postamble must already be included at the beginning or end, respectively, of the input signal to this block. If necessary, you can concatenate vectors in Simulink software using the Matrix Concatenation block.

To learn how the block uses the preamble and postamble, see "Reset Every Frame" Operation Mode" on page 2-543 below.

## "Reset Every Frame" Operation Mode

One way that the Viterbi algorithm can transition between successive frames is called Reset every frame mode. You can choose this mode using the Operation mode parameter.

In Reset every frame mode, the block decodes each frame of data independently, resetting the state metric at the end of each frame. The traceback decoding always starts at the state with the minimum state metric.

The initialization of state metrics depends on whether you specify a preamble and/or postamble:

- If you do not specify a preamble, the decoder initializes the metrics of all states to 0 at the beginning of each frame of data.
- If you specify a preamble, the block uses it to initialize the state metrics at the beginning of each frame of data. More specifically, the block decodes the preamble and assigns a metric of 0 to the decoded state. If the preamble does not decode to a unique state -- that is, if the length of the preamble is less than the channel memory -- the decoder assigns a metric of 0 to all states that can be represented by the preamble. Whenever you specify a preamble, the traceback path ends at one of the states represented by the preamble.
- If you do not specify a postamble, the traceback path starts at the state with the smallest metric.
- If you specify a postamble, the traceback path begins at the state represented by the postamble. If the postamble does not decode to a unique state, the decoder identifies the smallest of all possible decoded states that are represented by the postamble and begins traceback decoding at that state.

Note In Reset every frame mode, the input to the MLSE Equalizer block must contain at least T symbols, not including an optional preamble, where T is the Traceback depth parameter.

## Continuous Operation Mode

An alternative way that the Viterbi algorithm can transition between successive frames is called Continuous with reset option mode. You can choose this mode using the Operation mode parameter.

In Continuous with reset option mode, the block initializes the metrics of all states to 0 at the beginning of the simulation. At the end of each frame, the block saves the internal state metric for use in computing the traceback paths in the next frame.

If you select Enable the reset input port, the block displays another input port, labeled Rst. In this case, the block resets the state metrics whenever the scalar value at the Rst port is nonzero.

## Decoding Delay

The MLSE Equalizer block introduces an output delay equal to the Traceback depth in the Continuous with reset option mode, and no delay in the Reset every frame mode.

## MLSE Equalizer

Dialog Box


## Specify channel via

The method for specifying the channel estimate. If you select Input port, the block displays a second input port that receives the channel estimate. If you select Dialog, you can specify the channel estimate as a vector of coefficients for an FIR filter in the Channel coefficients field.

## Channel coefficients

Vector containing the coefficients of the FIR filter that the block uses for the channel estimate. This field is visible only if you set Specify channel via to Dialog.

## Signal constellation

Vector of complex numbers that specifies the constellation for the modulation.

## Traceback depth

The number of trellis branches (equivalently, the number of symbols) the block uses in the Viterbi algorithm to construct each traceback path.

## Operation mode

The operation mode of the Viterbi decoder. Choices are Continuous with reset option and Reset every frame.

## Input contains preamble

When checked, you can set the preamble in the Expected preamble field. This option appears only if you set Operation mode to Reset every frame.

## Expected preamble

Vector of integers between 0 and $\mathrm{M}-1$ representing the preamble, where M is the size of the constellation. This field is visible and active only if you set Operation mode to Reset every frame and then select Input contains preamble.

## Input contains postamble

When checked, you can set the postamble in the Expected postamble field. This option appears only if you set Operation mode to Reset every frame.

## Expected postamble

Vector of integers between 0 and M-1 representing the postamble, where M is the size of the constellation. This field is visible and active only if you set Operation mode to Reset every frame and then select Input contains postamble.

## Samples per input symbol

The number of input samples for each constellation point.

## Enable the reset input port

When you check this box, the block has a second input port labeled Rst. Providing a nonzero input value to this port causes the block to set its internal memory to the initial state before processing the input data. This option appears only if you set Operation mode to Continuous with reset option.

## Example <br> MLSE Equalization with dynamically changing channel

This example shows how to equalize the effects of a Multipath Rayleigh Fading Channel block. Maximum Likelihood Sequence Estimation (MLSE) estimates the data the model transmits through a time varying dispersive channel with the least possible number of errors. This model inputs the dynamically evolving channel coefficients of a two-path channel to the MLSE Equalizer block. The model shows the MLSE block being used in a typical multipath wireless Rayleigh channel. It applies the same channel estimate to 50 samples in the frame that is processed by the MLSE Equalizer. This is similar to a practical system, where the training sequence is transmitted in regular intervals and a channel estimate is used until the next training symbol is transmitted.
To open the example, type doc_mlse_dynamic_coeffs at the MATLAB command line.

## Block Parameters in the model

- The sample time of the Bernoulli Binary Generator block is set to $5 \mathrm{e}-6$, which corresponds to a bit rate of 200 kbps , and a QPSK symbol rate of $100 \mathrm{ksym} / \mathrm{sec}$.
- The Multipath Rayleigh Fading Channel block has a Maximum Doppler shift of 30 Hz , which is a realistic physical value. The Delay vector of the MRFC block is [0 1e-5], which corresponds to two consecutive sample times of the input QPSK symbol data. This reflects the simplest delay vector for a two-path channel. The Average path gain vector is set arbitrarily to [0-10]. The gain


## MLSE Equalizer

vector is normalized to 0 dB , so that the average power input to the AWGN block is 1 W .

- The MLSE Equalizer block has the Traceback depth set to 10 and may be varied to study its effect on Bit Error rate (BER).
- The QPSK Demodulator accepts an N-by-1 input frame and generates a 2 N -by- 1 output frame. This, along with the traceback depth of 10 results in a delay of 20 bits. The model performs frame-based processing with 100 samples per frame. Thus, there is a delay of 100 bits inherent in the model. The combined receive delay of 120 is set in the Receive delay parameter of the Error Rate Calculation block, aligning the samples.


## Block Parameters in the model

The sample time of the Bernoulli Binary Generator block is set to 5e-6, which corresponds to a bit rate of 200 kbps , and a QPSK symbol rate of $100 \mathrm{ksym} / \mathrm{sec}$. Multipath Rayleigh Fading Channel (MRFC) block: The MRFC block has a max Doppler shift of 30 Hz , which is a realistic physical value. The Delay vector of the MRFC block is [ $01 \mathrm{e}-5$ ], which corresponds to two consecutive sample times of the input QPSK symbol data. This reflects the simplest delay vector for a two-path channel. The Gain vector of the MRFC block is set arbitrarily to [0-10]. The gain vector is normalized to 0 dB , so that the average power input to the AWGN block is 1 W .

See Also LMS Linear Equalizer, LMS Decision Feedback Equalizer, RLS Linear Equalizer, RLS Decision Feedback Equalizer, CMA Equalizer

References
[1] Proakis, John G., Digital Communications, Fourth edition, New York, McGraw-Hill, 2001.
[2] Steele, Raymond, Ed., Mobile Radio Communications, Chichester, England, Wiley, 1996.

> Purpose
> Library
> Measure signal-to-noise ratio (SNR) in digital modulation applications
> Utility Blocks

Description


The Modulation Error Ratio (MER) is a measure of the signal-to-noise ratio (SNR) in digital modulation applications. You can use these types of measurements to determine system performance in communications applications. For example, determining if an EDGE system conforms to 3GPP radio transmission standards requires accurate MER, Minimum MER, and 95 th percentile for the MER measurements. The block measures all outputs in decibels (dB).

The MER block receives an ideal input signal (at reference port, Ref) and an AWGN corrupted signal (at input port, In). The MER block then outputs a measure of the modulation accuracy by comparing these inputs. The Modulation Error Ratio is the ratio of the average reference signal power to the mean square error. This ratio corresponds to the SNR of the AWGN channel.

The block output defaults to MER in decibels (dB), with an option of Output minimum MER or Output X-percentile MER values. The minimum MER represents the best-case MER value per burst. For the X-percentile option, you can select to output the number of symbols processed in the percentile computations.

The following table shows the output type, the activation (what selects the output computation), computation units, and the corresponding computation duration.

| Output | Activation | Units | Computation <br> Duration |
| :--- | :--- | :--- | :--- |
| MER | Default | Decibels | Per burst |
| Min MER | Parameter <br> setting | Decibels | Per burst |

# MER Measurement 

| Output | Activation | Units | Computation <br> Duration |
| :--- | :--- | :--- | :--- |
| Percentile MER | Parameter <br> setting | Decibels | Continuous |
| Number of <br> symbols | Parameter <br> setting if you <br> select Output <br> X-percentile <br> MER | None | Continuous |

## Dimension

The block computes measurements for bursts of data. The data must be of length $N$ symbols, where $N$ is the size of the burst. The block computes a unique output for each incoming burst; therefore, the computation duration is per burst.

## Input Signals

The input signals must be 1-D or 2-D sample-based column vectors or 2 -D frame-based column vectors. The input and reference signals must have identical dimensions.

## Output Signals

The output is always a scalar value.

## Data Type

The block accepts double, single, and fixed-point data types. The output of the block is always double type.

## Algorithms

MER is a measure of the SNR in a modulated signal calculated in dB . MER over $N$ symbols is

$$
M E R=10 * \log _{10}\left(\frac{\sum_{n=1}^{N}\left(I_{k}^{2}+Q_{k}^{2}\right)}{\sum_{n=1}^{N}\left(e_{k}\right)}\right) d B
$$

The MER for the $k_{t h}$ symbol is

$$
M E R_{k}=10 * \log _{10}\left(\frac{\frac{1}{N} \sum_{n=1}^{N}\left(I_{k}^{2}+Q_{k}^{2}\right)}{e_{k}}\right) d b
$$

The minimum MER represents the minimum MER value in a burst or
$M E R_{\min }=\min _{k \in[1, \ldots, N]}\left\{M E R_{k}\right\}$
where
$e_{k}=\left(I_{k}-\tilde{I_{k}}\right)^{2}+\left(Q_{k}-\tilde{Q}_{k}\right)^{2}$
$I_{k}=$ In-phase measurement of the $k t h$ symbol in the burst
$Q_{k}=$ Quadrature phase measurement of the $k t h$ symbol in the burst
$I_{k}$ and $Q_{k}$ represent ideal (reference) values. $\tilde{I}_{k}$ and $\tilde{Q}_{k}$ represent measured (received) symbols.

The block computes X-percentile MER by creating a histogram of all the incoming $M E R_{k}$ values. The output provides the MER value above which X\% of the MER values lay.


Dialog Box

## Output Minimum MER

Outputs the minimum MER of an input vector or frame.

## Output X-percentile MER

Enables an output X-percentile MER measurement. When you select this option, specify X-percentile value (\%).

## X -Percentile value (\%)

This parameter only appears when you select Output X-percentile MER. The Xth percentile is the MER value above which X\% of all the computed MER values lie. The parameter defaults to the 95 th percentile. Therefore, $95 \%$ of all MER values are above this output.

## Output the number of symbols processed

Outputs the number of symbols that the block uses to compute the Output X-percentile MER. This parameter only appears when you select Output X-percentile MER.

[^2]
# M-PAM Demodulator Baseband 

## Purpose

Demodulate PAM-modulated data

## Library

AM, in Digital Baseband sublibrary of Modulation
Description
MMW-
M.PAM

The M-PAM Demodulator Baseband block demodulates a signal that was modulated using M-ary pulse amplitude modulation. The input is a baseband representation of the modulated signal.

The signal constellation has M points, where M is the M -ary number parameter. M must be an even integer. The block scales the signal constellation based on how you set the Normalization method parameter. For details on the constellation and its scaling, see the reference page for the M-PAM Modulator Baseband block.

This block accepts a scalar or column vector input signal. For information about the data types each block port supports, see "Supported Data Types" on page 2-564.

Note All values of power assume a nominal impedance of 1 ohm .

## Integer-Valued Signals and Binary-Valued Signals

When you set the Output type parameter to Integer, the block outputs integer values between 0 and $M-1$. $M$ represents the M-ary number block parameter.

When you set the Output type parameter to Bit, the block outputs binary-valued signals that represent integers. The block represents each integer using a group of $K=\log _{2}(M)$ bits, where $K$ represents the number of bits per symbol. The output vector length must be an integer multiple of $K$.

The Constellation ordering parameter indicates how the block assigns binary words to points of the signal constellation. More details are on the reference page for the M-PAM Modulator Baseband block.

## M-PAM Demodulator Baseband

Algorithm
The demodulator algorithm maps received input signal constellation values to M-ary integer symbol indices between 0 and $\mathrm{M}-1$ and then maps these demodulated symbol indices to formatted output values.

The integer symbol index computation is performed by first scaling the real part of the input signal constellation (possibly with noise) by a denormalization factor derived from the Normalization method and related parameters. This denormalized value is added to M-1 to translate it into an approximate range between 0 and $2 \times(\mathrm{M}-1)$ plus noise. The resulting value is then rescaled via a divide-by-two (or, equivalently, a right-shift by one bit for fixed-point operation) to obtain a range approximately between 0 and $\mathrm{M}-1$ (plus noise). The noisy index value is rounded to the nearest integer and clipped, via saturation, to the exact range of [0 M-1]. Finally, based on other block parameters, the integer index is mapped to a symbol value that is formatted and cast to the selected Output data type.

The following figures contains signal flow diagrams for floating-point and fixed-point algorithm operation. The floating-point diagrams apply when the input signal data type is double or single. The fixed-point diagrams apply when the input signal is a signed fixed-point data type. Note that the diagram is simplified when using normalized constellations (i.e., denormalization factor is 1 ).

## M-PAM Demodulator Baseband



## Fixed Point



Signal-Flow Diagrams with Denormalization Factor Equal to 1

## M-PAM Demodulator Baseband



Signal-Flow Diagrams with Nonunity Denormalization Factor

## M-PAM Demodulator Baseband

## Dialog

Box

Function Block Parameters: M-PAM Demodulator Baseband
M-PAM Demodulator Baseband
Demodulate the input signal using the pulse amplitude modulation method.
This block accepts a scalar or column vector input signal.
The output signal can be either bits or integers. When you set the 'Output type' parameter to 'Bit', the output width is an integer multiple of the number of bits per symbol.


## M-ary number

The number of points in the signal constellation. It must be an even integer.

## M-PAM Demodulator Baseband

## Output type

Determines whether the output consists of integers or groups of bits. If this parameter is set to Bit, then the M-ary number parameter must be $2^{\mathrm{K}}$ for some positive integer K .

## Constellation ordering

Determines how the block maps each integer to a group of output bits.

## Normalization method

Determines how the block scales the signal constellation. Choices are Min. distance between symbols, Average Power, and Peak Power.

## Minimum distance

The distance between two nearest constellation points. This field appears only when Normalization method is set to Min. distance between symbols.

Average power, referenced to 1 ohm (watts)
The average power of the symbols in the constellation, referenced to 1 ohm . This field appears only when Normalization method is set to Average Power.

## Peak power, referenced to 1 ohm (watts)

The maximum power of the symbols in the constellation, referenced to 1 ohm . This field appears only when Normalization method is set to Peak Power.

## M-PAM Demodulator Baseband

Function Block Parameters: M-PAM Demodulator Baseband
M-PAM Demodulator Baseband
Demodulate the input signal using the pulse amplitude modulation method.
This block accepts a scalar or column vector input signal.
The output signal can be either bits or integers. When you set the 'Output type' parameter to 'Bit', the output width is an integer multiple of the number of bits per symbol.

## Main Data Types

Output: Inherit via internal rule

Fixed-point algorithm parameters
Floating-point inheritance takes precedence over the settings in the 'Data Type' column below. When the block input is floating point, all block data types match the input.

| Denormalization factor: | Data Type | Signed | Word Length | Fraction Length | Rounding | Overflow <br> Saturate |
| :---: | :---: | :---: | :---: | :---: | :---: | :---: |
|  | Same word length as input - | Yes | Same as input | Best precision | Nearest |  |
|  | Inherit via internal rule $\quad$ - | Yes | Inherited | Inherited | Floor | Wrap - |
| Sum: | Inherit via internal rule $\quad$ - | Yes | Inherited | Inherited | Nearest | Saturate |
|  |  |  |  | OK Can | Hel Help | Apply |

## Output

When the parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is of type single or double. Otherwise, the output data type will be as if this parameter is set to 'Smallest unsigned integer'.

When the parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in

## M-PAM Demodulator Baseband

the Hardware Implementation pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in the Hardware Implementation pane, the output data type is the ideal minimum size, i.e., ufix(1) for bit outputs, and ufix (ceil(log2(M))) for integer outputs. For all other selections, it is an unsigned integer with the smallest available word length large enough to fit the ideal minimum size, usually corresponding to the size of a char (e.g., uint8).

For integer outputs, this parameter can be set to Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, single, and double. For bit outputs, the options are Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, boolean, single, or double.

## Denormalization factor

This parameter applies when a fixed-point input is not normalized. It can be set to Same word length as input or Specify word length, in which case a field is enabled for user input. A best-precision fraction length is always used.

## Product output

This parameter only applies when the input is a fixed-point signal and there is a nonunity (not equal to 1 ) denormalized factor. It can be set to Inherit via internal rule or Specify word length, which enables a field for user input.

Setting to Inherit via internal rule computes the full-precision product word length and fraction length. Internal Rule for Product Data Types in DSP System Toolbox User's Guide describes the full-precision Product output internal rule.

Setting to Specify word length allows you to define the word length. The block computes a best-precision fraction length based on the word length specified and the pre-computed worst-case ( $\min / \max$ ) real world value Product output result. The worst-case Product output result is precomputed by multiplying

## M-PAM Demodulator Baseband

the denormalized factor with the worst-case ( $\mathrm{min} / \mathrm{max}$ ) input signal range, purely based on the input signal data type.

The block uses the Rounding method when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. For more information, see "Rounding Modes" in the DSP System Toolbox documentation or "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

## Sum

This parameter only applies when the input is a fixed-point signal. It can be set to Inherit via internal rule, Same as product output, or Specify word length, in which case a field is enabled for user input

Setting Inherit via internal rule computes the full-precision sum word length and fraction length, based on the two inputs to the Sum in the fixed-point Hard Decision Algorithm signal flow diagram. The rule is the same as the fixed-point inherit rule of the internal Accumulator data type parameter in the Simulink Sum block.

Setting Specify word length allows you to define the word length. A best precision fraction length is computed based on the word length specified in the pre-computed maximum range necessary for the demodulated algorithm to produce accurate results. The signed fixed-point data type that has the best precision fully contains the values in the range 2 * (M-1) for the specified word length.

Setting to Same as product output allows the Sum data type to be the same as the Product output data type (when Product output is used). If the Product output is not used, then this setting will be ignored and the Inherit via internal rule Sum setting will be used.

## M-PAM Demodulator Baseband

Supported
Data
Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed-point |
| Output | • Double-precision floating point |
|  | - Single-precision floating point |
|  | - Boolean when Output type is Bit |
|  | - 8-, 16-, and 32-bit signed integers |
|  | - 8-, 16-, and 32 -bit unsigned integers |
|  | - ufix $(1)$ in ASIC/FPGA when Output type is Bit |
|  | • ufix $\left(\left\lceil\log _{2} M\right\rceil\right)$ in ASIC/FPGA when Output type is Integer |

Pair Block M-PAM Modulator Baseband<br>See Also General QAM Demodulator Baseband

## M-PAM Modulator Baseband

Purpose Modulate using M-ary pulse amplitude modulation<br>Library AM, in Digital Baseband sublibrary of Modulation<br>Description The M-PAM Modulator Baseband block modulates using M-ary pulse amplitude modulation. The output is a baseband representation of the<br>■ WhM<br>M-PAM modulated signal. The M-ary number parameter, M, is the number of points in the signal constellation. It must be an even integer.

Note All values of power assume a nominal impedance of 1 ohm .

## Constellation Size and Scaling

Baseband M-ary pulse amplitude modulation using the block's default signal constellation maps an integer $m$ between 0 and M-1 to the complex value

$$
2 \mathrm{~m}-\mathrm{M}+1
$$

Note This value is actually a real number. The block's output signal is a complex data-type signal whose imaginary part is zero.

The block scales the default signal constellation based on how you set the Normalization method parameter. The following table lists the possible scaling conditions.

## M-PAM Modulator Baseband

| Value of Normalization <br> Method Parameter | Scaling Condition |
| :--- | :--- |
| Min. distance between <br> symbols | The nearest pair of points in the <br> constellation is separated by the <br> value of the Minimum distance <br> parameter |
| Average Power | The average power of the symbols <br> in the constellation is the <br> Average power parameter |
| Peak Power | The maximum power of the <br> symbols in the constellation is the <br> Peak power parameter |

## Integer-Valued Signals and Binary-Valued Signals

This block accepts a scalar or column vector input signal.
When you set the Input type parameter to Integer, the block accepts integer values between 0 and $M-1 . M$ represents the $\mathbf{M}$-ary number block parameter.

When you set the Input type parameter to Bit, the block accepts binary-valued inputs that represent integers. The block collects binary-valued signals into groups of $K=\log _{2}(M)$ bits
where
$K$ represents the number of bits per symbol.
The input vector length must be an integer multiple of $K$. In this configuration, the block accepts a group of $K$ bits and maps that group onto a symbol at the block output. The block outputs one modulated symbol for each group of $K$ bits.
The Constellation ordering parameter indicates how the block assigns binary words to points of the signal constellation.

- If Constellation ordering is set to Binary, then the block uses a natural binary-coded constellation.


## M-PAM Modulator Baseband

- If Constellation ordering is set to Gray, then the block uses a Gray-coded constellation.

For details about the Gray coding, see the reference page for the M-PSK Modulator Baseband block.

## Constellation Visualization

The M-PAM Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the Communications System Toolbox User's Guide.

## M-PAM Modulator Baseband

Dialog Box


## M-ary number

The number of points in the signal constellation. It must be an even integer.

## Input type

Indicates whether the input consists of integers or groups of bits. If this parameter is set to Bit, then the M-ary number parameter must be $2^{\mathrm{K}}$ for some positive integer K .

## Constellation ordering

Determines how the block maps each group of input bits to a corresponding integer.

# M-PAM Modulator Baseband 

## Normalization method

Determines how the block scales the signal constellation. Choices are Min. distance between symbols, Average Power, and Peak Power.

## Minimum distance

The distance between two nearest constellation points. This field appears only when Normalization method is set to Min. distance between symbols.

## Average power, referenced to 1 ohm (watts)

The average power of the symbols in the constellation, referenced to 1 ohm. This field appears only when Normalization method is set to Average Power.

## Peak power, referenced to 1 ohm (watts)

The maximum power of the symbols in the constellation, referenced to 1 ohm . This field appears only when Normalization method is set to Peak Power.

## Output data type

The output data type can be set to double, single, Fixed-point, User-defined, or Inherit via back propagation.

Setting this parameter to Fixed-point or User-defined enables fields in which you can further specify details. Setting this parameter to Inherit via back propagation, sets the output data type and scaling to match the following block.

## Output word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed-point for the Output data type parameter.

## User-defined data type

Specify any signed built-in or signed fixed-point data type. You can specify fixed-point data types using the sfix, sint, sfrac, and fixdt functions from Fixed-Point Designer software. This parameter is only visible when you select User-defined for the Output data type parameter.

## M-PAM Modulator Baseband

## Set output fraction length to

Specify the scaling of the fixed-point output by either of the following methods:

- Choose Best precision to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose User-defined to specify the output scaling in the Output fraction length parameter.

This parameter is only visible when you select Fixed-point for the Output data type parameter or when you select User-defined and the specified output data type is a fixed-point data type.

## Output fraction length

For fixed-point output data types, specify the number of fractional bits, or bits to the right of the binary point. This parameter is only visible when you select Fixed-point or User-defined for the Output data type parameter and User-defined for the Set output fraction length to parameter.

## Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | $\bullet$ Double-precision floating point |
|  | - Single-precision floating point |
|  | • Boolean when Input type is Bit |
|  | $\bullet 8-, 16-, 32$-bit signed integers |
|  | $\bullet 8-, 16-, 32$-bit unsigned integers |

## M-PAM Modulator Baseband

| Port | Supported Data Types |
| :--- | :--- |
|  | • ufix $\left(\left\lceil\log _{2} M\right\rceil\right)$ when Input type is Integer |
| Output | • Double-precision floating point |
|  | • Single-precision floating point |
|  | - Signed fixed-point |

Pair Block M-PAM Demodulator Baseband
See Also General QAM Modulator Baseband

## M-PSK Demodulator Baseband

Purpose Demodulate PSK-modulated data
LibraryPM, in Digital Baseband sublibrary of Modulation
Description The M-PSK Demodulator Baseband block demodulates a signal thatMOHE—
M-PSKwas modulated using the M-ary phase shift keying method. The inputis a baseband representation of the modulated signal. The input andoutput for this block are discrete-time signals. This block accepts ascalar-valued or column vector input signal. For information aboutthe data types each block port supports, see "Supported Data Types"on page 2-586.
The M-ary number parameter, M, is the number of points in the signal constellation.

## Integer-Valued Signals and Binary-Valued Signals

When you set the Output type parameter to Integer, the block outputs integer values between 0 and $M-1 . M$ represents the M-ary number block parameter.
When you set the Output type parameter to Bit, the block outputs binary-valued signals that represent integers. The block represents each integer using a group of $K=\log _{2}(M)$ bits, where $K$ represents the number of bits per symbol. The output vector length must be an integer multiple of $K$.
Depending on the demodulation scheme, the Constellation ordering or Symbol set ordering parameter indicates how the block maps a symbol to a group of $K$ output bits. When you set the parameter to Binary, the block maps the integer, I, to $[u(1) u(2) \ldots u(\mathrm{~K})]$ bits, where the individual $u(1)$ are given by

$$
\sum_{i=1}^{K} u(i) 2^{K-i}
$$

$u(1)$ is the most significant bit.

## M-PSK Demodulator Baseband

For example, if $M=8$, Constellation ordering (or Symbol set ordering) is set to Binary, and the integer symbol value is 6 , then the binary input word is $\left[\begin{array}{lll}1 & 1 & 0\end{array}\right]$.

When you set Constellation ordering (or Symbol set ordering) to Gray, the block assigns binary outputs from points of a predefined Gray-coded signal constellation. The predefined M-ary Gray-coded signal constellation assigns the binary representation

```
de2bi(bitxor(M,floor(M/2)), log2(M),'left-msb')
```

to the $M^{\text {th }}$ phase. The zeroth phase in the constellation is the Phase offset parameter, and successive phases are counted in a counterclockwise direction.

Note This transformation might seem counterintuitive because it constitutes a Gray-to-binary mapping. However, the block must use it to impose a Gray ordering on the signal constellation, which has a natural binary ordering.

In other words, if the block input is the natural binary representation, $u$, of the integer $U$, the block output has phase

$$
j \theta+j 2 п m / M
$$

where $\theta$ is the Phase offset parameter and $m$ is an integer between 0 and $\mathrm{M}-1$ that satisfies

$$
m \operatorname{XOR}\lfloor m / 2\rfloor=U
$$

For example, if $M=8$, the binary representations that correspond to the zeroth through seventh phases are as follows.

```
M = 8; m = [0:M-1]';
de2bi(bitxor(m,floor(m/2)), log2(M),'left-msb')
```


## M-PSK Demodulator Baseband

ans $=$

| 0 | 0 | 0 |
| :--- | :--- | :--- |
| 0 | 0 | 1 |
| 0 | 1 | 1 |
| 0 | 1 | 0 |
| 1 | 1 | 0 |
| 1 | 1 | 1 |
| 1 | 0 | 1 |
| 1 | 0 | 0 |

The following diagram shows the 8-ary Gray-coded constellation that the block uses if the Phase offset parameter is $\frac{\Pi}{8}$.


[^3]For M=2, refer to the BPSK Demodulator Baseband block reference page.

## M-PSK Demodulator Baseband

For M=4, refer to the QPSK Demodulator Baseband block reference page.

For $\mathrm{M}=8$ and greater, see the following signal diagrams.

## M-PSK Demodulator Baseband



Hard-Decision 8-PSK Demodulator Floating-Point Signal Diagram

## M-PSK Demodulator Baseband



Hard-Decision 8-PSK Demodulator Fixed-Point Signal Diagram

## M-PSK Demodulator Baseband



## Hard-Decision M-PSK Demodulator ( $\mathbf{M} \boldsymbol{>}$ 8) Floating-Point Signal Diagram for Nontrivial Phase Offset

## M-PSK Demodulator Baseband

For $M>8$, in order to improve speed and implementation costs, no derotation arithmetic is performed when Phase offset is $0, \pi / 2, \pi$, or $3 \pi / 2$ (i.e., when it is trivial).

Also, for $M>8$, this block will only support inputs of type double and single.

The exact LLR and approximate LLR cases (soft-decision) are described in "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the Communications System Toolbox User's Guide.

## M-PSK Demodulator Baseband



Dialog Box

## M-ary number

The number of points in the signal constellation.

## Phase offset

The phase of the zeroth point of the signal constellation.

## Constellation ordering

Determines how the block maps a symbol to the corresponding K output bits or integer. See the reference page for the M-PSK Modulator Baseband block for details. Selecting User-defined displays the field Constellation mapping, allowing for user-specified mapping.

## M-PSK Demodulator Baseband

## Constellation mapping

This field appears when User-defined is selected in the drop-down list Constellation ordering.

This parameter is a row or column vector of size M and must have unique integer values in the range [0, M-1]. The values must be of data type double.

The first element of this vector corresponds to the constellation point at $0+$ Phase offset angle, with subsequent elements running counterclockwise. The last element corresponds to the $-2 \pi / M+$ Phase offset constellation point.

## Output type

Determines whether the output consists of integers or groups of bits. If this parameter is set to Bit, the M-ary number parameter must be $2^{\mathrm{K}}$ for some positive integer K .

## Decision type

Specifies the output to be bitwise hard decision, LLR, or approximate LLR. This parameter appears when you select Bit from the Output type drop-down list. The output values for Log-likelihood ratio and Approximate log-likelihood ratio decision types are of the same data type as the input values

See "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the Communications System Toolbox User's Guide for algorithm details.

## Noise variance source

This field appears when Approximate log-likelinood ratio or Log-likelihood ratio is selected for Decision type.

When set to Dialog, the noise variance can be specified in the Noise variance field. When set to Port, a port appears on the block through which the noise variance can be input.

## M-PSK Demodulator Baseband

## Noise variance

This parameter appears when the Noise variance source is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

If you use theSimulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:

- Inf to - Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.

## M-PSK Demodulator Baseband

Function Block Parameters: M-PSK Demodulator Baseband
M-PSK Demodulator Baseband
Demodulate the input signal using the phase shift keying method.
For sample-based input, the input must be a scalar. For frame-based input, the input must be a column vector.
The output can be either bits or integers. For bit output, the output width is an integer multiple of the number of bits per symbol. In this case, Decision type parameter allows a choice between Hard decision demodulation, Loglikelihood ratio and Approximate log-likelihood ratio. The output values for Log-likelihood ratio and Approximate log-likelihood ratio Decision types are of the same data type as the input values. For integer output, the block always performs Hard decision demodulation.

## Main Data Types

Output: Inherit via internal rule

Fixed-point algorithm parameters
Floating-point inheritance takes precedence over the settings in the 'Data Type' column below. When the block input is floating point, all block data types match the input.

|  | Data Type | Signed Word Length Fraction Length Rounding Overflow |
| :--- | :--- | :--- |
| Derotate factor: | Same word length as input Yes Same as input Best precision Nearest Saturate |  |

$\square$

$\square$ Apply

## Data Types Pane for Hard-Decision

## Output

For bit outputs, when Decision type is set to Hard decision, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, uint32, or boolean.

For integer outputs, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, or uint32.

## M-PSK Demodulator Baseband

When this parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is a floating-point type (single or double). If the input data type is fixed-point (supported only when M-ary number is 2 , 4 , or 8 ), the output data type will work as if this parameter is set to 'Smallest unsigned integer'.

When this parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box of the model.

If ASIC/FPGA is selected in the Hardware Implementation pane, and Output type is Bit, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

If ASIC/FPGA is selected in the Hardware Implementation pane, and Output type is Integer, the output data type is the ideal minimum integer size, i.e., ufix (ceil ( $\log 2(M))$ ). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit the ideal minimum size, usually corresponding to the size of a char (e.g., uint8).

## Derotate factor

This parameter only applies when $\mathbf{M}$-ary number is 2 , 4 , or 8 , the input is fixed-point, and Phase offset is nontrivial. The phase offset is trivial when:

- You set M-ary number to 2 and Phase offset to a multiple

$$
\text { of } \frac{\pi}{2}
$$

## M-PSK Demodulator Baseband

- You set M-ary number to 4 and Phase offset to an even multiple of $\frac{\pi}{4}$

When you set M-ary number to 8 there are no trivial phase offsets.


## Data Types Pane for Soft-Decision

For bit outputs, when Decision type is set to Log-likelinood ratio or Approximate log-likelihood ratio, the output data type is inherited from the input (e.g., if the input is of data type double, the output is also of data type double).

## M-PSK Demodulator Baseband

Supported Data Types

| Port | Supported Data Types |
| :---: | :---: |
| Input | - Double-precision floating point <br> - Single-precision floating point <br> - Signed fixed point only when $M \leq 8$ and: <br> - Output type is Integer <br> - Output type is Bit and Decision type is Hard-decision |
| Var | - Double-precision floating point <br> - Single-precision floating point |
| Output | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean when Output type is Bit and Decision type is Hard-decision <br> - 8 -, 16-, and 32 -bit signed integers <br> - 8 -, 16 -, and 32 -bit unsigned integers <br> - ufix(1) when Output type is Bit <br> - ufix $\left(\log _{2} M\right)$ when Output type is Integer |

Pair Block M-PSK Modulator Baseband<br>See Also<br>BPSK Demodulator Baseband, QPSK Demodulator Baseband, M-DPSK Demodulator Baseband

# M-PSK Modulator Baseband 

## Purpose

Modulate using M-ary phase shift keying method

## Library

PM, in Digital Baseband sublibrary of Modulation
Description


The M-PSK Modulator Baseband block modulates using the M-ary phase shift keying method. The output is a baseband representation of the modulated signal. The M-ary number parameter, $M$, is the number of points in the signal constellation.

The block accepts scalar or column vector input signals. For information about the data types each block port supports, see "Supported Data Types" on page 2-594.

Alternative configurations of the block determine how the block interprets its input and arranges its output, as explained in the following sections.

## Integer-Valued Signals and Binary-Valued Signals

When you set the Input type parameter to Integer, the block accepts integer values between 0 and $M$-1. $M$ represents the $\mathbf{M}$-ary number block parameter.

When you set the Input type parameter to Bit, the block accepts binary-valued inputs that represent integers. The block collects binary-valued signals into groups of $K=\log _{2}(M)$ bits
where
$K$ represents the number of bits per symbol.
The input vector length must be an integer multiple of $K$. In this configuration, the block accepts a group of $K$ bits and maps that group onto a symbol at the block output. The block outputs one modulated symbol for each group of $K$ bits.

For example, the following schematics illustrate how the block processes two 8 -ary integers or binary words in one time step. The block processes all input signals as frames. In both cases, the Phase offset parameter is 0 .

## M-PSK Modulator Baseband



The Constellation ordering parameter indicates how the block maps a group of $K$ input bits to a corresponding symbol. When you set the parameter to Binary, the block maps $[u(1) u(2) \ldots u(\mathrm{~K})]$ to the integer

$$
\sum_{i=1}^{K} u(i) 2^{K-i}
$$

and behaves as if this integer were the input value. $u(1)$ is the most significant bit.
For example, if you set $M=8$, Constellation ordering to Binary, and the binary input word is $\left[\begin{array}{lll}1 & 1 & 0\end{array}\right]$, the block converts $\left[\begin{array}{lll}1 & 1 & 0\end{array}\right]$ to the integer 6 . The block produces the same output when the input is 6 and the Input type parameter is Integer.

When you set Constellation ordering to Gray, the block uses a Gray-coded arrangement and assigns binary inputs to points of a predefined Gray-coded signal constellation. The predefined M-ary Gray-coded signal constellation assigns the binary representation
de2bi(bitxor(M,floor(M/2)), log2(M),'left-msb')
to the $M^{\text {th }}$ phase. The zeroth phase in the constellation is the Phase offset parameter. Successive phases are in the counterclockwise direction.

Note This transformation seems counterintuitive because it constitutes a Gray-to-binary mapping. However, the block must use it to impose a Gray ordering on the signal constellation, which has a natural binary ordering.

In other words, if the block input is the natural binary representation, $u$, of the integer $U$, the block output has phase

$$
\mathrm{j} \theta+\mathrm{j} 2 \pi m / \mathrm{M}
$$

where $\theta$ is the Phase offset parameter and $m$ is an integer between 0 and M-1 that satisfies

$$
m \mathrm{XOR}\lfloor m / 2\rfloor=U
$$

For example, if $M=8$, the binary representations that correspond to the zeroth through seventh phases are as follows.

```
M = 8; m = [O:M-1]';
de2bi(bitxor(m,floor(m/2)), log2(M),'left-msb')
ans =
\begin{tabular}{lll}
0 & 0 & 0 \\
0 & 0 & 1 \\
0 & 1 & 1 \\
0 & 1 & 0 \\
1 & 1 & 0 \\
1 & 1 & 1
\end{tabular}
```


## M-PSK Modulator Baseband

| 1 | 0 | 1 |
| :--- | :--- | :--- |
| 1 | 0 | 0 |

The following diagram shows the 8-ary Gray-coded constellation that the block uses if the Phase offset parameter is $\frac{\Pi}{8}$.


## Constellation Visualization

The M-PSK Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the Communications System Toolbox User's Guide.

Dialog Box


## M-ary number

The number of points in the signal constellation.

## Phase offset

The phase of the zeroth point of the signal constellation.

## Constellation ordering

Determines how the block maps an integer or group of K input bits to the corresponding symbol.

If set to Binary, baseband M-ary phase shift keying modulation with a phase offset of $\theta$ maps an integer $m$ between 0 and M-1 to the complex value

## M-PSK Modulator Baseband

$$
\exp (j \theta+j 2 \pi m / M)
$$

If set to Gray, the block uses a Gray-coded signal constellation. As a result, binary representations that differ in more than one bit cannot map to consecutive integers modulo M .

Selecting User-defined displays the Constellation mapping parameter, which allows you to specify the mapping technique for the block.

## Constellation mapping

This field appears when you select User-defined from the Constellation ordering drop-down list.

This parameter is a row or column vector of size M and must have unique integer values in the range [0, M-1]. The values must be of data type double.

The first element of this vector corresponds to the constellation point at $0+$ Phase offset angle, with subsequent elements running counterclockwise. The last element corresponds to the $-2 \pi / M+$ Phase offset constellation point.

## Input type

Indicates whether the input consists of integers or groups of bits.
To use integer values between 0 and $\mathrm{M}-1$ as inputs, set this parameter to Integer.

If this parameter is set to Bit, the M-ary number parameter must be $2^{\mathrm{K}}$ for some positive integer $K$. $K$ consecutive elements in the input represent a symbol, where $K=\log 2(\mathrm{M})$.

## Output data type

This block supports the following output data types: double, single, Fixed-point, User-defined, or Inherit via back propagation.

Set this property to Fixed-point or User-defined to enable parameters in which you specify additional details. Set this property to Inherit via back propagation to match the output data type and scaling to the following block in the model.

## Output word length

Specify the word length, in bits, of the fixed-point output data type. This parameter appears when you select Fixed-point for the Output data type parameter.

## User-defined data type

Specify any signed built-in or signed fixed-point data type. You can specify fixed-point data types using the sfix, sint, sfrac, and fixdt functions from Fixed-Point Designer software. This parameter appears when you select User-defined for the Output data type parameter.

## Output fraction length

Specify the scaling of the fixed-point output by either of the following methods:

- Select Best precision to automatically scale the output signal so that it has the best possible precision.
- Select User-defined to specify the output scaling in the Output fraction length parameter.

This parameter appears when you select Fixed-point for the Output data type parameter or when you select User-defined and the specified output data type is a fixed-point data type.

## Output fraction length

For fixed-point output data types, specify the number of fractional bits, or bits to the right of the binary point. This parameter appears when you select Fixed-point or User-defined for the Output data type parameter and User-defined for the Set output fraction length to parameter.

## M-PSK Modulator Baseband

Supported Data Types

| Port | Supported Data Types |
| :---: | :---: |
| Input | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean when Input type is Bit <br> - 8-, 16-, and 32 -bit signed integers <br> - 8-, 16-, and 32 -bit unsigned integers <br> - ufix(1) when Input type is Bit <br> - ufix $\left(\log _{2} M\right)$ when Input type is Integer |
| Output | - Double-precision floating point <br> - Single-precision floating point <br> - Signed Fixed point |

## Pair Block

See Also
M-PSK Demodulator Baseband
BPSK Modulator Baseband, QPSK Modulator Baseband, M-DPSK Modulator Baseband

## Purpose

Recover carrier phase using M-Power method

## Library

Description


Carrier Phase Recovery sublibrary of Synchronization
The M-PSK Phase Recovery block recovers the carrier phase of the input signal using the M-Power method. This feedforward, non-data-aided, clock-aided method is suitable for systems that use baseband phase shift keying (PSK) modulation. It is also suitable for systems that use baseband quadrature amplitude modulation (QAM), although the results are less accurate than those for comparable PSK systems. The alphabet size for the modulation must be an even integer.

For PSK signals, the M-ary number parameter represents the alphabet size. For QAM signals, the M-ary number should be 4 regardless of the alphabet size because the 4 -power method is the most appropriate for QAM signals.

The M-Power method assumes that the carrier phase is constant over a series of consecutive symbols, and returns an estimate of the carrier phase for the series. The Observation interval parameter is the number of symbols for which the carrier phase is assumed constant. This number must be an integer multiple of the input signal's vector length.

## Input and Output Signals

This block accepts a scalar or column vector input signal of type double or single. The input signal represents a baseband signal at the symbol rate, so it must be complex-valued and must contain one sample per symbol.
The outputs are as follows:

- The output port labeled Sig gives the result of rotating the input signal counterclockwise, where the amount of rotation equals the carrier phase estimate. The Sig output is thus a corrected version of the input signal, and has the same sample time and vector size as the input signal.
- The output port labeled Ph outputs the carrier phase estimate, in degrees, for all symbols in the observation interval. The Ph output is a scalar signal.

Note Because the block internally computes the argument of a complex number, the carrier phase estimate has an inherent ambiguity. The carrier phase estimate is between -180/M and 180/M degrees and might differ from the actual carrier phase by an integer multiple of $360 / \mathrm{M}$ degrees.

## Delays and Latency

The block's algorithm requires it to collect symbols during a period of length Observation interval before computing a single estimate of the carrier phase. Therefore, each estimate is delayed by Observation interval symbols and the corrected signal has a latency of Observation interval symbols, relative to the input signal.

Dialog Box


## M-ary number

The number of points in the signal constellation of the transmitted PSK signal. This value as an even integer.

## Observation interval

The number of symbols for which the carrier phase is assumed constant. The observation interval parameter must be an integer multiple of the input signal vector length.

When this parameter is exactly equal to the vector length of the input signal, then the block always works. When the integer multiple is not equal to 1 , select Simulation $>$ Configuration Parameters > Solver
and set Tasking mode for periodic sample times to SingleTasking.

## Examples See "Carrier Phase Recovery Example" in Communications System Toolbox User's Guide.

Algorithm

## References

[1] Mengali, Umberto, and Aldo N. D'Andrea, Synchronization Techniques for Digital Receivers, New York, Plenum Press, 1997.
[2] Moeneclaey, Marc, and Geert de Jonghe, "ML-Oriented NDA Carrier Synchronization for General Rotationally Symmetric Signal Constellations," IEEE Transactions on Communications, Vol. 42, No. 8, Aug. 1994, pp. 2531-2533.

See Also CPM Phase Recovery, M-PSK Modulator Baseband

## Purpose

Decode trellis-coded modulation data, modulated using PSK method

## Library

Description
TCM, in Digital Baseband sublibrary of Modulation
The M-PSK TCM Decoder block uses the Viterbi algorithm to decode a trellis-coded modulation (TCM) signal that was previously modulated using a PSK signal constellation.

The M-ary number parameter represents the number of points in the signal constellation, which also equals the number of possible output symbols from the convolutional encoder. (That is, $\log _{2}(\mathbf{M}$-ary number) is the number of output bit streams from the convolutional encoder.)

The Trellis structure and M-ary number parameters in this block should match those in the M-PSK TCM Encoder block, to ensure proper decoding.

## Input and Output Signals

This block accepts a column vector input signal containing complex numbers. The input signal must be double or single. The reset port signal must be double or Boolean. For information about the data types each block port supports, see "Supported Data Types" on page 2-602.
If the convolutional encoder described by the trellis structure represents a rate $k / n$ code, then the M-PSK TCM Decoder block's output is a binary column vector whose length is $k$ times the vector length of the input signal.

## Operation Modes

The block has three possible methods for transitioning between successive frames. The Operation mode parameter controls which method the block uses. This parameter also affects the range of possible values for the Traceback depth parameter, D.

- In Continuous mode, the block initializes all state metrics to zero at the beginning of the simulation, waits until it accumulates D symbols, and then uses a sequence of $D$ symbols to compute each of the traceback paths. D can be any positive integer. At the end of


## M-PSK TCM Decoder

each frame, the block saves its internal state metric for use with the next frame.

If you select Enable the reset input, the block displays another input port, labeled Rst. This port receives an integer scalar signal. Whenever the value at the Rst port is nonzero, the block resets all state metrics to zero and sets the traceback memory to zero.

- In Truncated mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In Terminated mode, the block treats each frame independently. The traceback path always starts at the all-zeros state. D must be less than or equal to the vector length of the input. If you know that each frame of data typically ends at the all-zeros state, then this mode is an appropriate choice.


## Decoding Delay

If you set Operation mode to Continuous, then this block introduces a decoding delay equal to Traceback depth*k bits, for a rate k/n convolutional code. The decoding delay is the number of zeros that precede the first decoded bit in the output.

The block incurs no delay for other values of Operation mode.

## M-PSK TCM Decoder

Dialog Box


## Trellis structure

MATLAB structure that contains the trellis description of the convolutional encoder.

## M-ary number

The number of points in the signal constellation.

## Traceback depth

The number of trellis branches (equivalently, the number of symbols) the block uses in the Viterbi algorithm to construct each traceback path.

## M-PSK TCM Decoder

Operation mode
The operation mode of the Viterbi decoder. Choices are Continuous, Truncated, and Terminated.

## Enable the reset input port

When you check this box, the block has a second input port labeled Rst. Providing a nonzero input value to this port causes the block to set its internal memory to the initial state before processing the input data. This option appears only if you set Operation mode to Continuous.

## Output data type

The output type of the block can be specified as a boolean or double. By default, the block sets this to double.

## Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | • Single-precision floating point |

## Pair Block

M-PSK TCM Encoder

## See Also

General TCM Decoder, poly2trellis

## References

[1] Biglieri, E., D. Divsalar, P. J. McLane and M. K. Simon, Introduction to Trellis-Coded Modulation with Applications, New York, Macmillan, 1991.
[2] Proakis, John G., Digital Communications, Fourth edition, New York, McGraw-Hill, 2001.

## Purpose

Convolutionally encode binary data and modulate using PSK method

## Library

Description
TCM, in Digital Baseband sublibrary of Modulation
The M-PSK TCM Encoder block implements trellis-coded modulation (TCM) by convolutionally encoding the binary input signal and mapping the result to a PSK signal constellation.

The M-ary number parameter is the number of points in the signal constellation, which also equals the number of possible output symbols from the convolutional encoder. (That is, $\log _{2}(\mathbf{M}$-ary number) is equal to n for a rate $\mathrm{k} / \mathrm{n}$ convolutional code.)

## Input Signals and Output Signals

If the convolutional encoder described by the trellis structure represents a rate $k / n$ code, then the block input signal must be a binary column vector with a length of $L^{*} k$ for some positive integer $L$.

This block accepts a binary-valued input signal. The output signal is a complex column vector of length $L$.

## Specifying the Encoder

To define the convolutional encoder, use the Trellis structure parameter. This parameter is a MATLAB structure whose format is described in "Trellis Description of a Convolutional Code" in the Communications System Toolbox documentation. You can use this parameter field in two ways:

- If you want to specify the encoder using its constraint length, generator polynomials, and possibly feedback connection polynomials, then use a poly2trellis command within the Trellis structure field. For example, to use an encoder with a constraint length of 7 , code generator polynomials of 171 and 133 (in octal numbers), and a feedback connection of 171 (in octal), set the Trellis structure parameter to


## M-PSK TCM Encoder

- If you have a variable in the MATLAB workspace that contains the trellis structure, then enter its name as the Trellis structure parameter. This way is faster because it causes Simulink software to spend less time updating the diagram at the beginning of each simulation, compared to the usage in the previous bulleted item.

The encoder registers begin in the all-zeros state. You can configure the encoder so that it resets its registers to the all-zeros state during the course of the simulation. To do this, set the Operation mode to Reset on nonzero input via port. The block then opens a second input port, labeled Rst. The signal at the Rst port is a scalar signal. When it is nonzero, the encoder resets before processing the data at the first input port.

## Signal Constellations

The trellis-coded modulation technique partitions the constellation into subsets called cosets, so as to maximize the minimum distance between pairs of points in each coset. This block internally forms a valid partition based on the value you choose for the $\mathbf{M}$-ary number parameter.

The figure below shows the labeled set-partitioned signal constellation that the block uses when M-ary number is 8 . For constellations of other sizes, see [1].


## Coding Gains

Coding gains of 3 to 6 decibels, relative to the uncoded case can be achieved in the presence of AWGN with multiphase trellis codes [3].

## M-PSK TCM Encoder

Dialog

## Box



## Trellis structure

MATLAB structure that contains the trellis description of the convolutional encoder.

## Operation mode

In Continuous mode (default setting), the block retains the encoder states at the end of each frame, for use with the next frame.

In Truncated (reset every frame) mode, the block treats each frame independently. I.e., the encoder states are reset to all-zeros state at the start of each frame.

In Terminate trellis by appending bits mode, the block treats each frame independently. For each input frame, extra bits are used to set the encoder states to all-zeros state at the end of the frame. The output length is given by

## M-PSK TCM Encoder

$y=n \cdot(x+s) / k$, where $x$ is the number of input bits, and $s=$ constraint length -1 (or, in the case of multiple constraint lengths, $s=$ sum(ConstraintLength(i)-1)). The block supports this mode for column vector input signals.

In Reset on nonzero input via port mode, the block has an additional input port, labeled Rst. When the Rst input is nonzero, the encoder resets to the all-zeros state.

## M-ary number

The number of points in the signal constellation.

## Output data type

The output type of the block can be specified as a single or double. By default, the block sets this to double.

## Pair Block M-PSK TCM Decoder

See Also General TCM Encoder, poly2trellis
References [1] Biglieri, E., D. Divsalar, P. J. McLane and M. K. Simon, Introduction to Trellis-Coded Modulation with Applications, New York, Macmillan, 1991.
[2] Proakis, John G., Digital Communications, Fourth edition, New York, McGraw-Hill, 2001
[3] Ungerboeck, G., "Channel Coding with Multilevel/Phase Signals", IEEE Trans. on Information Theory, Vol IT28, Jan. 1982, pp. 55-67.

## MSK Demodulator Baseband

Purpose Demodulate MSK-modulated data<br>Library<br>CPM, in Digital Baseband sublibrary of Modulation

## Description The MSK Demodulator Baseband block demodulates a signal that was

 modulated using the minimum shift keying method. The input signal is a baseband representation of the modulated signal. The Phase offset parameter represents the initial phase of the modulated waveform.

## Integer-Valued Signals and Binary-Valued Signals

This block accepts a scalar-valued or column vector input signal with a data type of single or double. If you set the Output type parameter to Integer, then the block produces values of 1 and -1 . If you set the Output type parameter to Bit, then the block produces values of 0 and 1.

## Single-Rate Processing

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. The input width must be an integer multiple of the Samples per symbol parameter value, and the input can be a column vector.

- When you set Output type to Bit, the output width is $K$ times the number of input symbols.
- When you set Output type to Integer, the output width is the number of input symbols.


## Multirate Processing

In multirate processing mode, the input and output signals have different port sample times. The input must be a scalar. The output symbol time is the product of the input sample time and the Samples per symbol parameter value.

- When you set Output type to Bit, the output width equals the number of bits per symbol.


## MSK Demodulator Baseband

- When you set Output type to Integer, the output is a scalar.


## Traceback Depth and Output Delays

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The Traceback depth parameter, D , in this block is the number of trellis branches used to construct each traceback path. D influences the output delay, which is the number of zero symbols that precede the first meaningful demodulated value in the output.

- When you set the Rate options parameter to Allow multirate processing, and the model uses a variable-step solver or a fixed-step solver with the Tasking Mode parameter set to SingleTasking, then the delay consists of $\mathrm{D}+1$ zero symbols.
- When you set the Rate options parameter to Enforce single-rate processing, then the delay consists of D zero symbols.

The optimal Traceback depth parameter value is dependent on minimum squared Euclidean distance calculations. Alternatively, a typical value, dependent on the number of states, can be chosen using the "five-times-the-constraint-length" rule, which corresponds
to $5 \cdot \log 2$ (numStates). The number of states is determined by the following equation:
numStates $=\left\{\begin{array}{l}p \cdot 2^{(L-1)}, \text { for even } m \\ 2 p \cdot 2^{(L-1)}, \text { for oddm }\end{array}\right\}$
where:

- $h=m / p$ is the modulation index proper rational form
- $m=$ numerator of modulation index
- $p=$ denominator of modulation index
- $L$ is the Pulse length


## MSK Demodulator Baseband

## Dialog <br> Box



## Output type

Determines whether the output consists of bipolar or binary values.

## Phase offset (rad)

The initial phase of the modulated waveform.

## Samples per symbol

The number of input samples that represent each modulated symbol, which must be a positive integer. For more information, see "Upsample Signals and Rate Changes" in Communications System Toolbox User's Guide.

## Rate options

Select the rate processing method for the block.

- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample


## MSK Demodulator Baseband

time. The block implements the rate change by making a size change at the output when compared to the input. The output width is the number of symbols (which is given by dividing the input length by the Samples per symbol parameter value when the Output type parameter is set to Integer).

- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output period is the same as the symbol period and equals the product of the input period and the Samples per symbol parameter value.

Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

For more information, see Single-Rate Processing and Multirate Processing in the Description section of this page.

## Traceback depth

The number of trellis branches that the MSK Demodulator Baseband block uses to construct each traceback path.

## Output data type

The output data type can be boolean, int8, int16, int32, or double.

## MSK Demodulator Baseband

Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | $\bullet$ Single-precision floating point |
| Output | • Double-precision floating point |
|  | • Boolean (When Output type set to Bit) |
|  | • 8-, 16-, and 32-bit signed integers (When Output type |
|  | set to Integer) |

## Pair Block <br> MSK Modulator Baseband

## See Also <br> CPM Demodulator Baseband, Viterbi Decoder

References [1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg, Digital Phase Modulation, New York, Plenum Press, 1986.

## Purpose

Modulate using minimum shift keying method

## Library

Description


MSK
CPM, in Digital Baseband sublibrary of Modulation
The MSK Modulator Baseband block modulates using the minimum shift keying method. The output is a baseband representation of the modulated signal.

This block accepts a scalar-valued or column vector input signal. For a column vector input signal, the width of the output equals the product of the number of symbols and the value for the Samples per symbol parameter.

## Integer-Valued Signals and Binary-Valued Signals

When you set the Input type parameter to Integer, then the block accepts values of 1 and -1 .

When you set the Input type parameter to Bit, then the block accepts values of 0 and 1 .

For information about the data types each block port supports, see the "Supported Data Types" on page 2-616 table on this page.

## Single-Rate Processing

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. In this mode, the input to the block can be multiple symbols.

- When you set Input type to Integer, the input can be a column vector, the length of which is the number of input symbols.
- When you set Input type to Bit, the input width must be an integer multiple of $K$, the number of bits per symbol.

The output width equals the product of the number of input symbols and the Samples per symbol parameter value.

## MSK Modulator Baseband

## Multirate Processing

In multirate processing mode, the input and output signals have different port sample times. In this mode, the input to the block must be one symbol.

- When you set Input type to Integer, the input must be a scalar.
- When you set Input type to Bit, the input width must equal the number of bits per symbol.

The output sample time equals the symbol period divided by the Samples per symbol parameter value.

Dialog Box


## Input type

Indicates whether the input consists of bipolar or binary values.

## Phase offset (rad)

The initial phase of the output waveform, measured in radians.

## MSK Modulator Baseband

## Samples per symbol

The number of output samples that the block produces for each integer or binary word in the input, which must be a positive integer. For all non-binary schemes, as defined by the pulse shapes, this value must be greater than 1 .

For more information, see "Upsample Signals and Rate Changes" in Communications System ToolboxUser's Guide.

## Rate options

Select the rate processing option for the block.

- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width equals the product of the number of symbols and the Samples per symbol parameter value.
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output sample time equals the symbol period divided by the Samples per symbol parameter value.

Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

## Output data type

Specify the block output data type as double and single. By default, the block sets this to double.

## MSK Modulator Baseband

Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | $\bullet$ Double-precision floating point |
|  | $\bullet$ Boolean (when Input type set to Bit) |
|  | $\bullet$ 8-, 16-, and 32-bit signed integers (when Input type set |
|  | to Integer) |
| Output | • Double-precision floating point |
|  | • Single-precision floating point |

## Pair Block <br> See Also <br> CPM Modulator Baseband

References [1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg, Digital Phase Modulation, New York, Plenum Press, 1986.

## Purpose

Recover symbol timing phase using fourth-order nonlinearity method

## Library

Description
Timing Phase Recovery sublibrary of Synchronization
The MSK-Type Signal Timing Recovery block recovers the symbol timing phase of the input signal using a fourth-order nonlinearity method. This block implements a general non-data-aided feedback method that is independent of carrier phase recovery but requires prior
compensation for the carrier frequency offset. This block is suitable for systems that use baseband minimum shift keying (MSK) modulation or Gaussian minimum shift keying (GMSK) modulation.

## Inputs

By default, the block has one input port. The input signal could be (but is not required to be) the output of a receive filter that is matched to the transmitting pulse shape, or the output of a lowpass filter that limits the amount of noise entering this block.

This block accepts a scalar-valued or column vector input signal. The input uses $N$ samples to represent each symbol, where $N>1$ is the Samples per symbol parameter.

- For a column vector input signal, the block operates in single-rate processing mode. In this mode, the output signal inherits its sample rate from the input signal. The input length must be a multiple of $N$.
- For a scalar input signal, the block operates in multirate processing mode. In this mode, the input and output signals have different sample rates. The output sample rate equals $N$ multiplied by the input sample rate.
- This block accepts input signals of type Double or Single

If you set the Reset parameter to On nonzero input via port, then the block has a second input port, labeled Rst. The Rst input determines when the timing estimation process restarts, and must be a scalar.

## MSK-Type Signal Timing Recovery

- If the input signal is a scalar value, the sample time of the Rst input equals the symbol period
- If the input signal is a column vector, the sample time of the Rst input equals the input port sample time
- This block accepts reset signals of type Double or Boolean


## Outputs

The block has two output ports, labeled Sym and Ph:

- The Sym output is the result of applying the estimated phase correction to the input signal. This output is the signal value for each symbol, which can be used for decision purposes. The values in the Sym output occur at the symbol rate:
- For a column vector input signal of length $N^{*} R$, the Sym output is a column vector of length $R$ having the same sample rate as the input signal.
- For a scalar input signal, the sample rate of the Sym output equals $N$ multiplied by the input sample rate.
- The Ph output gives the phase estimate for each symbol in the input.

The Ph output contains nonnegative real numbers less than $N$. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time of the Ph output is the same as that of the Sym output.

Note If the Ph output is very close to either zero or Samples per symbol, or if the actual timing phase offset in your input signal is very close to zero, then the block's accuracy might be compromised by small amounts of noise or jitter. The block works well when the timing phase offset is significant rather than very close to zero.

- The output signal inherits its data type from the input signal.


## MSK-Type Signal Timing Recovery

## Delays

When the input signal is a vector, this block incurs a delay of two symbols. When the input signal is a scalar, this block incurs a delay of three symbols.

Dialog Box


## Modulation type

The type of modulation in the system. Choices are MSK and GMSK.

## Samples per symbol

The number of samples, N , that represent each symbol in the input signal. This must be greater than 1.

## Error update gain

A positive real number representing the step size that the block uses for updating successive phase estimates. Typically, this number is less than $1 / \mathrm{N}$, which corresponds to a slowly varying phase.

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink documentation.

## Reset

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are None, Every frame, and On nonzero input via port. The last option causes the block to have a second input port, labeled Rst.

Algorithm
This block's algorithm extracts timing information by passing the sampled baseband signal through a fourth-order nonlinearity followed by a digital differentiator whose output is smoothed to yield an error signal. The algorithm then uses the error signal to make the sampling adjustments.

More specifically, this block uses a timing error detector whose result for the kth symbol is $\mathrm{e}(\mathrm{k})$, given in [2] by

$$
\begin{array}{r}
e(k)=(-1)^{D+1} \operatorname{Re}\left\{r^{2}\left(k T-T_{s}+d_{k-1}\right) r^{* 2}\left((k-1) T-T_{s}+d_{k-2}\right)\right\} \\
\quad-(-1)^{D+1} \operatorname{Re}\left\{r^{2}\left(k T+T_{s}+d_{k-1}\right) r^{* 2}\left((k-1) T+T_{s}+d_{k-1}\right)\right\} \\
e(k)=(-1)^{D+1} \operatorname{Re}\left\{r^{2}\left(k T-T_{s}+d_{k-1}\right) r^{* 2}\left((k-1) T-T_{s}+d_{k-2}\right)\right\} \\
-(-1)^{D+1} \operatorname{Re}\left\{r^{2}\left(k T+T_{s}+d_{k-1}\right) r^{* 2}\left((k-1) T+T_{s}+d_{k-1}\right)\right\}
\end{array}
$$

where

## MSK-Type Signal Timing Recovery

- $r$ is the block's input signal
- T is the symbol period
- $\mathrm{T}_{\mathrm{s}}$ is the sampling period
-     * means complex conjugate
- $d_{k}$ is the phase estimate for the kth symbol
- D is 1 for MSK and 2 for Gaussian MSK modulation

For more information about the role that e(k) plays in this block's algorithm, see "Feedback Methods for Timing Phase Recovery" in Communications System Toolbox User's Guide.

References [1] D'Andrea, A. N., U. Mengali, and R. Reggiannini, "A Digital Approach to Clock Recovery in Generalized Minimum Shift Keying," IEEE Transactions on Vehicular Technology, Vol. 39, No. 3, August 1990, pp. 227-234.

[2] Mengali, Umberto and Aldo N. D'Andrea, Synchronization Techniques for Digital Receivers, New York, Plenum Press, 1997.

See Also Early-Late Gate Timing Recovery, Squaring Timing Recovery

## Purpose

Library
Description
Mueller-Muller Symp
Mueller-Muller
Timing Recovery
Ph

Recover symbol timing phase using Mueller-Muller method
Timing Phase Recovery sublibrary of Synchronization
The Mueller-Muller Timing Recovery block recovers the symbol timing phase of the input signal using the Mueller-Muller method. This block implements a decision-directed, data-aided feedback method that requires prior recovery of the carrier phase.

## Inputs

By default, the block has one input port. Typically, the input signal is the output of a receive filter that is matched to the transmitting pulse shape.

This block accepts a scalar-valued or column vector input signal. The input uses $N$ samples to represent each symbol, where $N>1$ is the Samples per symbol parameter.

- For a column vector input signal, the block operates in single-rate processing mode. In this mode, the output signal inherits its sample rate from the input signal. The input length must be a multiple of $N$.
- For a scalar input signal, the block operates in multirate processing mode. In this mode, the input and output signals have different sample rates. The output sample rate equals $N$ multiplied by the input sample rate.
- This block accepts input signals of type Double or Single

If you set the Reset parameter to On nonzero input via port, then the block has a second input port, labeled Rst. The Rst input determines when the timing estimation process restarts, and must be a scalar.

- If the input signal is a scalar value, the sample time of the Rst input equals the symbol period
- If the input signal is a column vector, the sample time of the Rst input equals the input port sample time
- This block accepts reset signals of type Double or Boolean


## Outputs

The block has two output ports, labeled Sym and Ph:

- The Sym output is the result of applying the estimated phase correction to the input signal. This output is the signal value for each symbol, which can be used for decision purposes. The values in the Sym output occur at the symbol rate:
- For a column vector input signal of length $N^{*} R$, the Sym output is a column vector of length $R$ having the same sample rate as the input signal.
- For a scalar input signal, the sample rate of the Sym output equals $N$ multiplied by the input sample rate.
- The Ph output gives the phase estimate for each symbol in the input.

The Ph output contains nonnegative real numbers less than $N$. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time of the Ph output is the same as that of the Sym output.

Note If the Ph output is very close to either zero or Samples per symbol, or if the actual timing phase offset in your input signal is very close to zero, then the block's accuracy might be compromised by small amounts of noise or jitter. The block works well when the timing phase offset is significant rather than very close to zero.

- The output signal inherits its data type from the input signal.


## Delays

When the input signal is a vector, this block incurs a delay of two symbols. When the input signal is a scalar, this block incurs a delay of three symbols.

## Mueller-Muller Timing Recovery



Dialog Box

## Samples per symbol

The number of samples, N , that represent each symbol in the input signal. This must be greater than 1.

## Error update gain

A positive real number representing the step size that the block uses for updating successive phase estimates. Typically, this number is less than $1 / \mathrm{N}$, which corresponds to a slowly varying phase.

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink User's Guide.

## Reset

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are None, Every , and On nonzero input via port. The last option causes the block to have a second input port, labeled Rst.

## Algorithm

## References

This block uses a timing error detector whose result for the kth symbol is $\mathrm{e}(\mathrm{k})$, given by

$$
e(k)=\operatorname{Re}\left\{c_{k-1}^{*} y\left(k T+d_{k}\right)-c_{k}^{*} y\left((k-1) T+d_{k-1}\right)\right\}
$$

where

- y is the block's input signal
- $c_{k}$ is the decision based on the sample value $y\left(k T+d_{k}\right)$
- T is the symbol period
- $d_{k}$ is the phase estimate for the kth symbol

For more information about the role that e(k) plays in this block's algorithm, see "Feedback Methods for Timing Phase Recovery" in Communications System ToolboxUser's Guide.
[1] Mengali, Umberto and Aldo N. D'Andrea, Synchronization Techniques for Digital Receivers, New York, Plenum Press, 1997.
[2] Meyr, Heinrich, Marc Moeneclaey, and Stefan A. Fechtel, Digital Communication Receivers, Vol 2, New York, Wiley, 1998.

## Mueller-Muller Timing Recovery

[3] Mueller, K. H., and M. S. Muller, "Timing Recovery in Digital Synchronous Data Receivers," IEEE Transactions on Communications, Vol. COM-24, May 1976, pp. 516-531.

See Also Early-Late Gate Timing Recovery, Squaring Timing Recovery

## Mu-Law Compressor

## Purpose

Implement $\mu$-law compressor for source coding

## Library

Source Coding

Description

Mu-Law Compressor

## Dialog Box

The Mu-Law Compressor block implements a $\mu$-law compressor for the input signal. The formula for the $\mu$-law compressor is

$$
y=\frac{V \log (1+\mu|x| / V)}{\log (1+\mu)} \operatorname{sgn}(x)
$$

where $\mu$ is the $\mu$-law parameter of the compressor, $V$ is the peak magnitude of $x, \log$ is the natural logarithm, and sgn is the signum function (sign in MATLAB).

The input can have any shape or frame status. This block processes each vector element independently.


## mu value

The $\mu$-law parameter of the compressor.

## Mu-Law Compressor

Peak signal magnitude
The peak value of the input signal. This is also the peak value of the output.

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | $\bullet$ double |
| Out | $\bullet$ double |

Pair Block Mu-Law Expander<br>See Also<br>References<br>[1] Sklar, Bernard. Digital Communications: Fundamentals and Applications. Englewood Cliffs, N.J.: Prentice-Hall, 1988.

## Mu-Law Expander

## Purpose

## Library

Description

Mu-Law Expander

Implement $\mu$-law expander for source coding
Source Coding
The Mu-Law Expander block recovers data that the Mu-Law Compressor block compressed. The formula for the $\mu$-law expander, shown below, is the inverse of the compressor function.

$$
x=\frac{V}{\mu}\left(e^{|y| \log (1+\mu) / V}-1\right) \operatorname{sgn}(y)
$$

The input can have any shape or frame status. This block processes each vector element independently.


## mu value

The $\mu$-law parameter of the compressor.

## Mu-Law Expander

## Peak signal magnitude

The peak value of the input signal. This is also the peak value of the output.

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | $\bullet$ double |
| Out | $\bullet$ double |

## Pair Block Mu-Law Compressor

See Also
A-Law Expander
References [1] Sklar, Bernard. Digital Communications: Fundamentals and Applications. Englewood Cliffs, N.J.: Prentice-Hall, 1988.

# Multipath Rayleigh Fading Channel 

## Purpose

Simulate multipath Rayleigh fading propagation channel

## Library

Description

Multipath Rayleigh
Fading Channel
Channels
The Multipath Rayleigh Fading Channel block implements a baseband simulation of a multipath Rayleigh fading propagation channel. You can use this block to model mobile wireless communication systems. For details about fading channels, see the references listed below.

This block accepts a scalar value or column vector input signal. The block inherits sample time from the input signal. The input signal must
have a discrete sample time greater than 0 .

Relative motion between the transmitter and receiver causes Doppler shifts in the signal frequency. You can specify the Doppler spectrum of the Rayleigh process using the Doppler spectrum type parameter. For channels with multiple paths, you can assign each path a different Doppler spectrum, by entering a vector of doppler objects in the Doppler spectrum field.

Because a multipath channel reflects signals at multiple places, a transmitted signal travels to the receiver along several paths, each of which may have differing lengths and associated time delays. In the block's parameter dialog box, the Discrete path delay vector specifies the time delay for each path. If you do not check Normalize gain vector to 0 dB overall gain, then the Average path gain vector specifies the gain for each path. When you check the box, the block uses a multiple of Average path gain vector instead of the Average path gain vector itself, choosing the scaling factor so that the channel's effective gain, considering all paths, is 0 dB .

The number of paths indicates the length of Discrete path delay vector or Average path gain vector, whichever is larger. If both of these parameters are vectors, then they must have the same length; if exactly one of these parameters contains a scalar value, then the block expands it into a vector whose size matches that of the other vector parameter.

## Multipath Rayleigh Fading Channel

The block multiplies the input signal by samples of a Rayleigh-distributed complex random process. The scalar Initial seed parameter seeds the random number generator and the block generates random numbers using the Ziggurat method.
Double-clicking this block during simulation with Inline parameters off or selecting Open channel visualization at start of simulation plots the channel characteristics using the channel visualization tool. See "Channel Visualization" in Communications System Toolbox User's Guide for details.

## Multipath Rayleigh Fading Channel



Dialog Box

## Maximum Doppler shift (Hz)

A positive scalar value that indicates the maximum Doppler shift.

## Doppler spectrum type

Specifies the Doppler spectrum of the Rayleigh process.

## Multipath Rayleigh Fading Channel

This parameter defaults to Jakes Doppler spectrum. Alternatively, you can also choose any of the following types:

- Flat
- Gaussian
- Rounded
- Restricted Jakes
- Asymmetrical Jakes
- Bi-Gaussian
- Bell

For all Doppler spectrum types except Jakes and Flat, you can choose one or more parameters to control the shape of the spectrum.

You can also select Specify as dialog parameter for the Doppler spectrum type. Specify the Doppler spectrum by entering an object in the Doppler spectrum field. See the doppler function reference in Communications System Toolbox User's Guide for details on how to construct Doppler objects, and also for the meaning of the parameters associated with the various Doppler spectrum types.

## Discrete path delay vector (s)

A vector that specifies the propagation delay for each path.

## Average path gain vector (dB)

A vector that specifies the gain for each path.

## Normalize gain vector to 0 dB overall gain

Checking this box causes the block to scale the Gain vector parameter so that the channel's effective gain (considering all paths) is 0 dB .

## Initial seed

The scalar seed for the Gaussian noise generator.

## Multipath Rayleigh Fading Channel

## Open channel visualization at start of simulation

Select this check box to open the channel visualization tool when a simulation begins.

## Complex path gains port

Select this check box to create a port that outputs the values of the complex path gains for each path. In this $N$-by- $M$ multichannel output, $N$ represents the number of samples the input signal contains and $M$ represents the number of discrete paths (number of delays).

## Channel filter delay port

Select this check box to create a port that outputs the value of the delay (in samples) that results from the filtering operation of this block. This delay is zero if only one path is simulated, but can be greater than zero if more than one path is present. See "Methodology for Simulating Multipath Fading Channels:" in Communications System Toolbox User's Guide for a definition of this delay, where it is denoted as $N_{1}$.

## Algorithm

This implementation is based on the direct-form simulator described in Reference [1]. A detailed explanation of the implementation, including a review of the different Doppler spectra, can be found in [4].

Some wireless applications, such as standard GSM (Global System for Mobile Communication) systems, prefer to specify Doppler shifts in terms of the speed of the mobile. If the mobile moves at speed $v$ making an angle of $\theta$ with the direction of wave motion, then the Doppler shift is

$$
f_{\mathrm{d}}=(v f / c) \cos \theta
$$

where $f$ is the transmission carrier frequency and $c$ is the speed of light. The Doppler frequency represents the maximum Doppler shift arising from motion of the mobile.

## Multipath Rayleigh Fading Channel

## Example Generating Ideal Theoretical BER Results for a Rayleigh Fading Channel

This example illustrates how to generate ideal theoretical BER results for a flat Rayleigh fading channel. The model uses reproduces known theoretical results and shows the correct BER performance for a flat Rayleigh fading channel. In this example, you will run the model and compare the simulation results to the BERTool theoretical results for verification purposes. Note that the EbNo value for the model's AWGN block is 5 dB . You can change the noise power by double-clicking the AWGN block and entering another numeric value in the EbNo parameter.

## Opening the Model

You can open the model by clicking here in the MATLAB Help browser. Alternatively, you can type doc_qpsk_rayleigh_derotated at the MATLAB command line.

## Running the Model and Comparing Results

1 You can run the example by clicking Simulation > Run.
2 After the model collects more than 5000 errors, click the stop button.
3 Close the three scopes.
4 In the Simulink model window, double-click the Transmitter Output block. In the mask window, click the Figure Properties tab, uncheck Open scope at start of Simulation, then click OK.

## Multipath Rayleigh Fading Channel



5 In the Simulink model window, double-click the Rayleigh Channel Output block. In the mask window, click the Figure Properties tab, uncheck Open scope at start of Simulation, then click OK.

6 In the Simulink model window, double-click the Noisy Rayleigh Channel Output block. In the mask window, click the Figure Properties tab, uncheck Open scope at start of Simulation, then clock OK.

7 In the Simulink model window, double-click the Error Rate Calculation block, check Stop simulation, enter 5000 for Target number of error, then click OK.

8 Click the play button to rerun the example.

## Multipath Rayleigh Fading Channel

9 Open BERTool by typing bertool at the MATLAB command line.
10 In BERTool, click the Theoretical tab and make the following selections:


- For Eb/No range enter 0:10
- For Channel type, select Rayleigh
- For Diversity Order enter 1
- For Modulation Type, select PSK
- For Modulation order, select 4


## Multipath Rayleigh Fading Channel

## 11 Click Plot.

12 Since the Simulink model uses an EbNo value of 5 dB , verify the probability of error on the BERTool curve at 5 dB . The two values should be approximately equal.


Click the Data Cursor button (second from right) and click on the BERTool curve at 5dB.

## See Also

References

Rayleigh Noise Generator, Multipath Rician Fading Channel, doppler
[1] Jeruchim, Michel C., Balaban, Philip, and Shanmugan, K. Sam, Simulation of Communication Systems, Second edition, New York, Kluwer Academic/Plenum, 2000.
[2] Jakes, William C., ed. Microwave Mobile Communications, New York, IEEE Press, 1974.

## Multipath Rayleigh Fading Channel

[3] Lee, William C. Y., Mobile Communications Design Fundamentals, 2nd Ed. New York, Wiley, 1993.
[4] Iskander, Cyril-Daniel, A MATLAB-based Object-Oriented Approach to Multipath Fading Channel Simulation, a MATLAB Central submission available from www.mathworks.com.

# Multipath Rician Fading Channel 

## Purpose

Simulate multipath Rician fading propagation channel

## Library

Description
포 Rician 포
Fading
Channels
The Multipath Rician Fading Channel block implements a baseband simulation of a multipath Rician fading propagation channel. You can use this block to model mobile wireless communication systems when
the transmitted signal can travel to the receiver along a dominant line-of-sight or direct path. For more details, see "Fading Channels".
This block accepts a scalar value or column vector input signal. The block inherits sample time from the input signal. The input signal must have a discrete sample time greater than 0 .

Relative motion between the transmitter and receiver causes Doppler shifts in the signal frequency. You can specify the Doppler spectrum of the Rician process using the Doppler spectrum type pop-up menu. For channels with multiple paths, you can assign each path a different Doppler spectrum, by entering a vector of doppler objects in the Doppler spectrum field.

Because a multipath channel reflects signals at multiple places, a transmitted signal travels to the receiver along several paths, each of which may have differing lengths and associated time delays. In the block's parameter dialog box, the Discrete path delay vector specifies the time delay for each path. If you do not check the Normalize gain vector to 0 dB overall gain box, then the Average path gain vector specifies the gain for each path. When you check the box, the block uses a multiple of Average path gain vector instead of the Average path gain vector itself, choosing the scaling factor so that the channel's effective gain considering all paths is 0 dB .

The number of paths indicates the length of Discrete path delay vector or Average path gain vector, whichever is larger. If both of these parameters are vectors, they must have the same length; if exactly one of these parameters contains a scalar value, the block expands it into a vector whose size matches that of the other vector parameter.

## Multipath Rician Fading Channel

Fading causes the signal to become diffuse. The K-factor parameter, which is part of the statistical description of the Rician distribution, represents the ratio between the power in the line-of-sight component and the power in the diffuse component. The ratio is expressed linearly, not in decibels. While the Average path gain vector parameter controls the overall gain through the channel, the K-factor parameter controls the gain's partition into line-of-sight and diffuse components.

You can specify the K-factor parameter as a scalar or a vector. If the K-factor parameter is a scalar, then the first discrete path of the channel is a Rician fading process (it contains a line-of-sight component) with the specified K-factor, while the remaining discrete paths indicate independent Rayleigh fading processes (with no line-of-sight component). If the $\mathbf{K}$-factor parameter is a vector of the same size as Discrete path delay vector, then each discrete path is a Rician fading process with a K-factor given by the corresponding element of the vector. You can attribute the line-of-sight component a Doppler shift, through the Doppler shift(s) of line-of-sight component(s) parameter, and an initial phase, through the Initial phase(s) of line-of-sight component(s). The Doppler shift(s) of line-of-sight component(s) and Initial phase(s) of line-of-sight component(s) parameters must be of the same size as the K-factor parameter.
The block multiplies the input signal by samples of a Rician-distributed complex random process. The scalar Initial seed parameter seeds the random number generator and the block generates random numbers using the Ziggurat method.
Double-clicking this block during simulation with Inline parameters off or selecting the block dialog's check box labeled Open channel visualization at start of simulation plots the channel characteristics using the channel visualization tool. See "Channel Visualization" in Communications System Toolbox User's Guide for details.

## Multipath Rician Fading Channel



## K-factor

The ratio of power in the line-of-sight component to the power in the diffuse component. The ratio is expressed linearly, not in decibels. If $\mathbf{K}$-factor is a scalar value, then the first discrete path

## Multipath Rician Fading Channel

is a Rician fading process (it contains a line-of-sight component) with the specified K-factor, while the remaining discrete paths are independent Rayleigh fading processes (with no line-of-sight component). If K-factor is a vector of the same size as Discrete path delay vector, then each discrete path is a Rician fading process with a K-factor given by the corresponding element of the vector.

## Doppler shift(s) of line-of-sight components(s) (Hz)

The Doppler shift of the line-of-sight component. It must be a scalar (if K-factor is a scalar) or a vector of the same size as K-factor. If this parameter contains a scalar value, then the line-of-sight component of the first discrete path has the specified Doppler shift, while the remaining discrete paths become independent Rayleigh fading processes. If the parameter contains a vector, then the line-of-sight component of each discrete path has a Doppler shift given by the corresponding element of the vector.
Initial phase(s) of line-of-sight component(s) (rad)
The initial phase of the line-of-sight component. It must be either a scalar (if $\mathbf{K}$-factor is a scalar value) or a vector of the same size as K-factor.

Maximum diffuse Doppler shift (Hz)
A positive scalar value that indicates the maximum diffuse Doppler shift.

## Doppler spectrum type

Specifies the Doppler spectrum of the Rician process.
This parameter defaults to Jakes Doppler spectrum. Alternately, you can choose any of the following types:

- Flat
- Gaussian
- Rounded
- Restricted Jakes


# Multipath Rician Fading Channel 

- Asymmetrical Jakes
- Bi-Gaussian
- Bell

For all Doppler spectrum types except Jakes and Flat, You can use one or more parameters to control the shape of the spectrum.

You can also select Specify as dialog parameter for the Doppler spectrum type. Specify the Doppler spectrum by entering an object in the Doppler spectrum field. See the doppler function reference in Communications System Toolbox User's Guide for details on how to construct doppler objects, and for the meaning of the parameters associated with the various Doppler spectrum types.

## Discrete delay vector(s)

A vector that specifies the propagation delay for each path.

## Average path gain vector (dB)

A vector that specifies the gain for each path.

## Initial seed

The scalar seed for the Gaussian noise generator.

## Open channel visualization at start of simulation

Select this check box to open the channel visualization tool when a simulation begins. This block supports channel visualization for a column vector input signal.

## Complex path gains port

Select this check box to create a port that outputs the values of the complex path gains for each path. In this $N$-by- $M$ multichannel output, $N$ represents the number of samples the input contains and $M$ represents the number of discrete paths (number of delays).

## Channel filter delay port

Select this check box to create a port that outputs the value of the delay (in samples) that results from the filtering operation of this block. This delay is zero if only one path is simulated,

## Multipath Rician Fading Channel

but can be greater than zero if more than one path is present. See "Methodology for Simulating Multipath Fading Channels:" in Communications System Toolbox User's Guide for a definition of this delay, where it is denoted as $N_{1}$.

## Algorithm

See Also Rician Noise Generator, Multipath Rayleigh Fading Channel, doppler
References
This implementation is based on the direct form simulator described in Reference [1]. A detailed explanation of the implementation, including a review of the different Doppler spectra, can be found in [4].

Some wireless applications, such as standard GSM (Global System for Mobile Communication) systems, prefer to specify Doppler shifts in terms of the speed of the mobile. If the mobile moves at speed $v$ making an angle of $\theta$ with the direction of wave motion, the Doppler shift is

$$
f_{\mathrm{d}}=(v f / c) \cos \theta
$$

where $f$ is the transmission carrier frequency and $c$ is the speed of light. The Doppler frequency is the maximum Doppler shift arising from the motion of the mobile.
[1] Jeruchim, Michel C., Balaban, P., and Shanmugan, K. Sam, Simulation of Communication Systems, Second edition, New York, Kluwer Academic/Plenum, 2000.
[2] Jakes, William C., ed., Microwave Mobile Communications, New York, IEEE Press, 1974.
[3] Lee, William C. Y., Mobile Communications Design Fundamentals, 2nd ed., New York, John Wiley \& Sons, Inc., 1993.
[4] Iskander, Cyril-Daniel, A MATLAB-based Object-Oriented Approach to Multipath Fading Channel Simulation, a MATLAB Central submission available from www.mathworks.com.

## Normalized LMS Decision Feedback Equalizer

## Purpose

## Library

Description


Equalize using decision feedback equalizer that updates weights with normalized LMS algorithm

Equalizer Block
The Normalized LMS Decision Feedback Equalizer block uses a decision feedback equalizer and the normalized LMS algorithm to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the normalized LMS algorithm to update the weights, once per symbol. When you set the Number of samples per symbol parameter to 1 , then the block implements a symbol-spaced (i.e. T-spaced) equalizer. When you set the Number of samples per symbol parameter to a value greater than 1 , , the weights are updated once every $N^{\text {th }}$ sample, for a $T / N$-spaced equalizer.

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.

Set the Reference tap parameter so it is greater than zero and less than the value for the Number of forward taps parameter.

The port labeled Equalized outputs the result of the equalization process.

You can configure the block to have one or more of these extra ports:

- Mode input, as described in "Reference Signal and Operation Modes" in Communications System Toolbox User's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.
- Weights output, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.


## Normalized LMS Decision Feedback Equalizer

## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Using Adaptive Equalizers" in Communications System Toolbox User's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to

## $1+($ Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

## Normalized LMS Decision Feedback Equalizer

Dialog Box


## Number of forward taps

The number of taps in the forward filter of the decision feedback equalizer.

## Normalized LMS Decision Feedback Equalizer

## Number of feedback taps

The number of taps in the feedback filter of the decision feedback equalizer.

## Number of samples per symbol

The number of input samples for each symbol.

## Signal constellation

A vector of complex numbers that specifies the constellation for the modulation.

## Reference tap

A positive integer less than or equal to the number of forward taps in the equalizer.

## Step size

The step size of the normalized LMS algorithm.

## Leakage factor

The leakage factor of the normalized LMS algorithm, a number between 0 and 1. A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

## Bias

The bias parameter of the normalized LMS algorithm, a nonnegative real number. This parameter is used to overcome difficulties when the algorithm's input signal is small.

## Initial weights

A vector that concatenates the initial weights for the forward and feedback taps.

## Mode input port

If you select this check box, the block has an input port that enables you to toggle between training and decision-directed mode. For training, the mode input must be 1 , for decision directed, the mode should be 0 . The equalizer will train for the length of the Desired signal. If the mode input is not present, the equalizer will train at the beginning of every frame for the length of the Desired signal.

## Output error

If you select this check box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## Output weights

If you select this check box, the block outputs the current forward and feedback weights, concatenated into one vector.

References [1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, Wiley, 1998.

See Also Normalized LMS Linear Equalizer, LMS Decision Feedback Equalizer

## Normalized LMS Linear Equalizer

| Purpose | Equalize using linear equalizer that updates weights with normalized <br> LMS algorithm |
| :--- | :--- |
| Library | Equalizers |
| Description | The Normalized LMS Linear Equalizer block uses a linear equalizer <br> and the normalized LMS algorithm to equalize a linearly modulated <br> baseband signal through a dispersive channel. During the simulation, |
| the block uses the normalized LMS algorithm to update the weights, |  |
| once per symbol. When you set the Number of samples per symbol |  |
| parameter to 1, the block implements a symbol-spaced (i.e. T-spaced) |  |
| equalizer and updates the filter weights once for each symbol. When |  |
| you set the Number of samples per symbol parameter to a value |  |
| greater than 1, the weights are updated once every $N^{\text {th }}$ sample, for a |  |
| $T / N$-spaced equalizer. |  |

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.

Set the Reference tap parameter so it is greater than zero and less than the value for the Number of taps parameter.

The port labeled Equalized outputs the result of the equalization process.
You can configure the block to have one or more of these extra ports:

- Mode input, as described in "Reference Signal and Operation Modes" in Communications System Toolbox User's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.


## Normalized LMS Linear Equalizer

- Weights output, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.


## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Using Adaptive Equalizers" in Communications System Toolbox User's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to
$1+($ Reference tap-1)/(Number of samples per symbol)
Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## Normalized LMS Linear Equalizer



## Number of taps

The number of taps in the filter of the linear equalizer.

## Number of samples per symbol

The number of input samples for each symbol.

## Signal constellation

A vector of complex numbers that specifies the constellation for the modulation.

## Reference tap

A positive integer less than or equal to the number of taps in the equalizer.

## Step size

The step size of the normalized LMS algorithm.

## Leakage factor

The leakage factor of the normalized LMS algorithm, a number between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

## Bias

The bias parameter of the normalized LMS algorithm, a nonnegative real number. This parameter is used to overcome difficulties when the algorithm's input signal is small.

## Initial weights

A vector that lists the initial weights for the taps.

## Mode input port

When you select this check box, the block has an input port that allows you to toggle between training and decision-directed mode. For training, the mode input must be 1 , for decision directed, the mode should be 0 . For every frame in which the mode input is 1 or not present, the equalizer trains at the beginning of the frame for the length of the desired signal.

## Output error

If you check this box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## Normalized LMS Linear Equalizer

## Output weights

If you check this box, the block outputs the current weights.

## Examples

References $\quad \begin{aligned} & \text { [1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, } \\ & \text { Chichester, England, Wiley, 1998. }\end{aligned}$
See Also Normalized LMS Decision Feedback Equalizer, LMS Linear Equalizer

## OQPSK Demodulator Baseband

## Purpose

Demodulate OQPSK-modulated data

## Library

Description


OQPSK
PM, in Digital Baseband sublibrary of Modulation

The OQPSK Demodulator Baseband block demodulates a signal that was modulated using the offset quadrature phase shift keying method. The input is a baseband representation of the modulated signal.

The input must be a discrete-time complex signal. This block accepts a scalar-valued or column vector input signal. For information about the data types each block port supports, see "Supported Data Types" on page 2-666.

When you set the Output type parameter to Integer, the block outputs integer symbol values between 0 and 3 . When you set the Output type parameter to Bit, the block outputs a 2 -bit binary representation of integers, in a binary-valued vector with a length that is an even number.
The block produces one output symbol for each pair of input samples. The input sample period is half the period of each output integer or bit pair. The constellation used to map bit pairs to symbols is on the reference page for the OQPSK Modulator Baseband block.

## Single-Rate Processing

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. The input can be an even length column vector.

- When you set Output type to Bit, the output width is 2 times the number of input symbols.
- When you set Output type to Integer, the output width is the number of input symbols.


## OQPSK Demodulator Baseband



To open this model, type doc_moqpskdemod_fb at the MATLAB command line.

## Multirate Processing

In multirate processing mode, the input and output signals have different port sample times. The input must be a scalar. The output symbol time is two times the input sample time.

- When you set Output type to Bit, the output width equals 2.
- When you set Output type to Integer, the output is a scalar.


To open this model, type doc_moqpskdemod_sb at the MATLAB command line.

## Delays

The modulator-demodulator pair incurs a delay, as described in "Example: Delays from Demodulation".

# OQPSK Demodulator Baseband 

Signal Flow
Diagram


OQPSK Fixed-Point Signal Flow Diagram

Note Every two input samples produce one output symbol. In the preceding figure, the dotted line represents the region comprised of input sample processing.


## OQPSK Floating Point Signal Flow Diagram

Note Every two input samples produce one output symbol. In the preceding figure, the dotted line represents the region comprised of input sample processing.

## OQPSK Demodulator Baseband

Function Block Parameters: OQPSK Demodulator Baseband
OQPSK Demodulator Baseband
Demodulate the input signal using the offset quadrature phase shift keying method.
For the multirate processing option, this block accepts a scalar input signal. For the single-rate processing option, this block accepts a column vector input signal whose width must be even.

The output signal can be either bits or integers. When you set the 'Output type' parameter to 'Bit', the output width is even.
Main $\mid$ Data Types
Parameters

Phase offset(rad): 0

Output type:

> Integer
$\nabla$

Rate options:
Enforce single-rate processing $\nabla$

## Dialog Box

## Phase offset (rad)

The amount by which the phase of the zeroth point of the signal constellation is shifted from $\Pi / 4$.

## Output type

Determines whether the output consists of integers or pairs of bits.

## Rate options

Select the rate processing option for the block.

- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size


## OQPSK Demodulator Baseband

change at the output when compared to the input. The output width equals half the input width for integer outputs.

- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output symbol time is two times the input sample time.

Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

## OQPSK Demodulator Baseband



## Data Type Pane

## Output

For bit outputs, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, uint32, or boolean.

For integer outputs, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, or uint32.

When this parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input

## OQPSK Demodulator Baseband

data type if the input is a floating-point type (single or double). If the input data type is fixed-point, the output data type will work as if this parameter is set to 'Smallest unsigned integer'.

When this parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box of the model.

If ASIC/FPGA is selected in the Hardware Implementation pane, and Output type is Bit, the output data type is the ideal minimum one-bit size, i.e., ufix1. For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

If ASIC/FPGA is selected in the Hardware Implementation pane, and Output type is Integer, the output data type is the ideal minimum two-bit size, i.e., ufix2. For all other selections, it is an unsigned integer with the smallest available word length large enough to fit two bits, usually corresponding to the size of a char (e.g., uint8).

## Derotate factor

This parameter only applies when the input is fixed-point and
Phase offset is not a multiple of $\frac{\Pi}{2}$.
This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input.

## Accumulator

Specify the data type for the Accumulator. You can set this parameter to Inherit via internal rule, Same as input or Binary point scaling.

## OQPSK Demodulator Baseband

The Accumulator parameter only applies for fixed-point inputs. The selections you make for the Rounding and Overflow parameters affect the Accumulator.

Fixed-point Communications System Toolbox blocks that must hold summation results for further calculation usually allow you to specify the data type and scaling of the accumulator. Most such blocks cast to the accumulator data type prior to summation:


Use the Accumulator-Mode parameter to specify how you would like to designate the accumulator word and fraction lengths:

- When you select Inherit via internal rule, the accumulator output word and fraction lengths are automatically calculated for you. Refer to "Inherit via Internal Rule" for more information.
- When you select Same as product output, these characteristics match those of the product output.
- When you select Same as input, these characteristics match those of the first input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.


## OQPSK Demodulator Baseband

- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator. The bias of all signals in DSP System Toolbox software is zero.


## Rounding

The block uses the Rounding method when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. For more information, see "Rounding Modes" in DSP System Toolbox User's Guide or "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

## Overflow

Specify the method of storing the result when the magnitude of a fixed-point calculation result that does not does not fit within the range of the data type selected. You can select either Wrap or Saturate for this parameter.

For more information refer to Overflow in the Precision and Range subsection of DSP System Toolbox.

## Mapping input

This can be set to Same as accumulator or Binary point scaling. This parameter only applies for fixed-point inputs.

## Supported <br> Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed point |
| Output | • Double-precision floating point |
|  | - Single-precision floating point |

## OQPSK Demodulator Baseband

## Port $\quad$ Supported Data Types

- Boolean when Output type is Bit
- 8-, 16-, and 32 -bit signed integers
- 8-, 16-, and 32 -bit unsigned integers
- ufix(1) in ASIC/FPGA when Output type is Bit
- ufix(2) in ASIC/FPGA when Output type is Integer

Pair Block OQPSK Modulator Baseband
See Also QPSK Demodulator Baseband

## OQPSK Modulator Baseband

## Purpose Modulate using offset quadrature phase shift keying method <br> Library <br> PM, in Digital Baseband sublibrary of Modulation <br> Description <br> च— $\quad \mathrm{mm}$ OQPSK <br> The OQPSK Modulator Baseband block modulates using the offset quadrature phase shift keying method. The block outputs a baseband representation of the modulated signal.

Note The OQPSK modulator block upsamples by a factor of 2 .

When you set the Input type parameter to Integer, valid input values are $0,1,2$, and 3 . In this case, the block accepts a scalar or a column vector input signal.

When you set the Input type parameter to Bit, a binary-valued vector is a valid input value. In this case, the block accepts a column vector input signal with a length that is an even integer.

For information about the data types each block port supports, see "Supported Data Types" on page 2-674.

The constellation the block uses to map bit pairs to symbols is shown in the following figure. If you set the Phase offset parameter to a nonzero value, then the constellation rotates by that value.


## Single-Rate Processing

In single-rate processing mode, the input and output signals have the same port sample time. The block implicitly implements the rate change by making a size change at the output when compared to the input. In this mode, the input to the block can be multiple symbols.

- When you set Input type to Integer, the input can be a scalar value or column vector, the length of which is the number of input symbols.
- When you set Input type to Bit, the input width must be an integer multiple of two.

The output sample period is half the period of each integer or bit pair in the input.

## OQPSK Modulator Baseband



To open this model, type doc_moqpskmod_fb at the MATLAB command line.

## Multirate Processing

In multirate processing mode, the input and output signals have different port sample times. In this mode, the input to the block must be one symbol.

- When you set Input type to Integer, the input must be a scalar value.
- When you set Input type to Bit, the input width must equal 2.

The output sample time equals one-half the symbol period. The first output symbol is an initial condition of zero that is unrelated to the input values.

## OQPSK Modulator Baseband



To open this model, type doc_moqpskmod_sb at the MATLAB command line.

## Delays

The modulator-demodulator pair incurs a delay, as described in "Delays in Digital Modulation".

## OQPSK Modulator Baseband



Dialog
Box

## Phase offset (rad)

The amount by which the block shifts the phase of the zeroth point of the signal constellation from $\pi / 4$.

## Input type

Indicates whether the input consists of integers or pairs of bits.

## Rate options

Select the rate processing option for the block.

- Enforce single-rate processing - When you select this option, the input and output signals have the same port sample time. The block implements the rate change by making a size change at the output when compared to the input. The output width equals two times the number of symbols for integer inputs.
- Allow multirate processing - When you select this option, the input and output signals have different port sample times. The output sample time equals one-half the symbol period.

Note The option Inherit from input (this choice will be removed - see release notes) will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

## Output data type

Select the output data type as double, single, Fixed-point, User-defined, or Inherit via back propagation.

Setting this parameter to Fixed-point or User-defined enables fields in which you can further specify details. Setting this parameter to Inherit via back propagation, sets the output data type and scaling to match the following block.

## Output word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed-point for the Output data type parameter.

## User-defined data type

Specify any signed built-in or signed fixed-point data type. You can specify fixed-point data types using the sfix, sint, sfrac, and fixdt functions from Fixed-Point Designer. This parameter is only visible when you select User-defined for the Output data type parameter.

## Set output fraction length to

Specify the scaling of the fixed-point output by either of the following methods:

- When you select Best precision the block sets the output scaling so the output signal has the best possible precision.


## OQPSK Modulator Baseband

- When you select User-defined you specify the output scaling using the Output fraction length parameter.

This parameter applies when you select Fixed-point for the Output data type parameter or you select User-defined and the specified output data type is a fixed-point data type.

## Output fraction length

For fixed-point output data types, specify the number of fractional bits, or bits to the right of the binary point. This parameter applies when you select Fixed-point or User-defined for the Output data type parameter and User-defined for the Set output fraction length to parameter.

Supported Data Types

| Port | Supported Data Types |
| :---: | :---: |
| Input | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean when Input type is Bit <br> - 8 -, 16 -, and 32 -bit signed integers <br> - 8-, 16-, and 32 -bit unsigned integers <br> - ufix(1) when Input type is Bit <br> - ufix(2) when Input type is Integer |
| Output | - Double-precision floating point <br> - Single-precision floating point <br> - Signed fixed point |

Pair Block OQPSK Demodulator Baseband<br>See Also QPSK Modulator Baseband

## OSTBC Combiner

## Purpose

## Library

Description


OSTBC Combiner

Combine inputs for received signals and channel estimate according to orthogonal space-time block code (OSTBC)

MIMO
The OSTBC Combiner block combines the input signal (from all of the receive antennas) and the channel estimate signal to extract the soft information of the symbols that were encoded using an OSTBC. The input channel estimate may not be constant during each codeword block transmission and the combining algorithm uses only the estimate for the first symbol period per codeword block. A symbol demodulator or decoder would follow the Combiner block in a MIMO communications system.

The block conducts the combining operation for each symbol independently. The combining algorithm depends on the structure of the OSTBC. For more information, see the OSTBC Combining Algorithms section of this help page.

## Dimension

Along with the time and spatial domains for OSTBC transmission, the block supports an optional dimension, over which the combining calculation is independent. This dimension can be thought of as the frequency domain for OFDM-based applications. The following illustration indicates the supported dimensions for inputs and output of the OSTBC Combiner block.


The following table describes each variable for the block.

## OSTBC Combiner

| Variable | Description |
| :--- | :--- |
| F | The additional dimension; typically the frequency <br> dimension. The combining calculation is <br> independent of this dimension. |
| N | Number of transmit antennas. |
| M | Number of receive antennas. |
| T | Output symbol sequence length in time domain. |
| R | Symbol rate of the code. |

Note On the two inputs, T/R is the symbol sequence length in the time domain.
$F$ can be any positive integers. $M$ can be 1 through 8 , indicated by the Number of receive antennas parameter. $N$ can be 2,3 or 4 , indicated by the Number of transmit antennas parameter. The time domain length $T / R$ must be a multiple of the codeword block length ( 2 for Alamouti; 4 for all other OSTBC). For $N=2, T / R$ must be a multiple of 2 . When $N>2$, T/R must be a multiple of 4 . $R$ defaults to 1 for 2 antennas. $R$ can be either $\frac{3}{4}$ or $\frac{1}{2}$ for more than 2 antennas. The supported dimensions for the block depend upon the values of $F$ and $M$. For one receive antenna $(M=1)$, the received signal input must be a column vector or a full 2-D matrix, depending on the value for $F$. The corresponding channel estimate input must be a full 2-D or 3-D matrix.

For more than one receive antenna ( $M>1$ ), the received signal input must be a full 2-D or 3-D matrix, depending on the value for $F$. Correspondingly, the channel estimate input must be a $3-\mathrm{D}$ or 4 -D matrix, depending on the value for $F$.

To understand the block's dimension propagation, refer to the following table.

## OSTBC Combiner

|  | Input 1 <br> (Received <br> Signal) | Input 2 <br> (Channel <br> Estimate) | Output |
| :--- | :--- | :--- | :--- |
| $\mathrm{F}=1$ and $\mathrm{M}=1$ | Column vector | 2-D | Column vector |
| $\mathrm{F}=1$ and $\mathrm{M}>1$ | 2-D | 3-D | Column vector |
| $\mathrm{F}>1$ and $\mathrm{M}=1$ | 2-D | 3-D | 2-D |
| $\mathrm{F}>1$ and $\mathrm{M}>1$ | 3-D | 4-D | 2-D |

## Data Type

For information about the data types each block port supports, see the "Supported Data Type" on page 2-686 table on this page. The output signal inherits the data type from the inputs. The block supports different fixed-point properties for the two inputs. For fixed-point signals, the output word length and fractional length depend on the block's mask parameter settings. See Fixed-Point Signals for more information about fixed-point data propagation of this block.

## Frames

The output inherits the frameness of the received signal input. For either column vector or full 2-D matrix input signal, the input can be either frame-based or sample-based. A 3-D or 4-D matrix input signal must have sample-based input.

## OSTBC Combining Algorithms

The OSTBC Combiner block supports five different OSTBC combining computation algorithms. Depending on the selection for Rate and Number of transmit antennas, you can select one of the algorithms shown in the following table.

## OSTBC Combiner

| Transmit Antenna | Rate | Computational Algorithm per Codeword Block Length |
| :---: | :---: | :---: |
| 2 | 1 | $\binom{\hat{s}_{1}}{\hat{s}_{2}}=\frac{1}{\\|H\\|^{2}} \sum_{j=1}^{M}\binom{h_{1, j}^{*} r_{1, j}+h_{2},{ }_{j} r_{2, j}^{*}}{h_{2, j}^{*} r_{1, j}-h_{1},{ }_{j} r_{2, j}^{*}}$. |
| 3 | 1/2 | $\binom{\hat{s}_{1}}{\hat{s}_{2}}=\frac{1}{\\|H\\|^{2}} \sum_{j=1}^{M}\binom{h_{1, j}^{*} r_{1, j}+h_{2, j} r_{2, j}^{*}+h_{3, j}^{*} r_{3, j}^{*}}{h_{2, j}^{*} r_{1, j}-h_{1, j} r_{2, j}^{*}-h_{3, j} r_{4, j}^{*}}$. |
| 3 | 3/4 | $\left(\begin{array}{l}\hat{s}_{1} \\ \hat{s}_{2} \\ \hat{s}_{3}\end{array}\right)=\frac{1}{\\|H\\|^{2}} \sum_{j=1}^{M}\left(\begin{array}{l}h_{1, j}^{*} r_{1, j}+h_{2, j} r_{2, j}^{*}-h_{3, j} r_{3, j}^{*} \\ h_{2, j}^{*} r_{1, j}-h_{1, j} r_{2, j}^{*}-h_{3, j} r_{4, j}^{*} \\ h_{3, j}^{*} r_{1, j}+h_{1, j} r_{3, j}^{*}+h_{2, j} r_{4, j}^{*}\end{array}\right)$. |
| 4 | 1/2 | $\binom{\hat{s}_{1}}{\hat{s}_{2}}=\frac{1}{\\|H\\|^{2}} \sum_{j=1}^{M}\binom{h_{1, j}^{*} r_{1, j}+h_{2, j} r_{2, j}^{*}+h_{3, j 3, j}^{*} r_{3, j}+h_{4, j} r_{4, j}^{*}}{h_{2, j}^{*} r_{1, j} h_{1, j} r_{2, j}^{*}+h_{4, j}^{*} r_{3, j}-h_{3, j} r_{4, j}^{*}}$. |
| 4 | 3/4 | $\left(\begin{array}{l}\hat{s}_{1} \\ \hat{s}_{2} \\ \hat{s}_{3}\end{array}\right)=\frac{1}{\\|H\\|^{2}} \sum_{j=1}^{M}\left(\begin{array}{c}h_{1, j}^{*} r_{1, j}+h_{2, j} r_{2, j}^{*} h_{3, j} r_{3, j}^{*}-h_{4, j}^{*} r_{4 j} \\ h_{2, j}^{*} r_{1 j}-h_{1, j} r_{2, j}^{*}+h_{4, j}^{*} r_{3, j}-h_{3, j}^{*} r_{4, j}^{*} \\ h_{3, j}^{*} r_{1 j}+h_{4, j}^{*} r_{2, j}+h_{1, j} r_{3, j}^{*}+h_{2, j} r_{4, j}^{* *}\end{array}\right)$. |

$\hat{s}_{k}$ represents the estimated $k$ th symbol in the OSTBC codeword matrix. $h_{i j}$ represents the estimate for the channel from the $i$ th transmit antenna and the $j$ th receive antenna. The values of $i$ and $j$ can range from 1 to $N$ (the number of transmit antennas) and to $M$ (the number of receive antennas) respectively. $r_{l j}$ represents the $l$ th symbol at the $j$ th receive antenna per codeword block. The value of $l$ can range from 1 to

## OSTBC Combiner

the codeword block length. $\|H\|^{2}$ represents the summation of channel
power per link, i.e., $\|H\|^{2}=\sum_{i=1}^{N} \sum_{j=1}^{M}\left\|h_{i j}\right\|^{2}$

## Fixed-Point Signals

Use the following formula for $\hat{s}_{1}$ for Alamouti code with 1 receive antenna to highlight the data types used for fixed-point signals.
$\hat{s}_{1}=\frac{h_{1,1,}^{*} r_{1,1}+h_{2,1}, r_{2,1}^{*}}{\|H\|^{2}}=\frac{h_{1,1,}^{*} r_{1,1}+h_{2,1} r_{2,1}^{*}}{h_{1,1} h_{1,1}^{*}+h_{2,1} h_{2,1}^{*}}$
In this equation, the data types for Product output and Accumulator correspond to the product and summation in the numerator. Similarly, the types for Energy product output and Energy accumulator correspond to the product and summation in the denominator.


Signal Flow Diagram for $\mathrm{s}_{1}$ Combining Calculation of Alamouti Code
with One Receive Antenna

## OSTBC Combiner

The following formula shows the data types used within the OSTBC Combiner block for fixed-point signals for more than one receive antenna for Alamouti code, where $M$ represents the number of receive antennas.
$\hat{s}_{1}=\frac{h_{1,1,}^{*} r_{1,1}+h_{2,1,} r_{2,1}^{*}+h_{1,2}^{*} r_{1,2}+h_{2,2} r_{2,2}^{*}+\ldots+h_{1, M,}^{*} r_{1, M}+h_{2, M} r_{2, M}^{*}}{h_{1.1} h_{1.1}^{*}+h_{2.1} h_{2.1}^{*}+h_{1.2} h_{1.2}^{*}+h_{2.2} h_{2.2}^{*}+\ldots+h_{1 . M} h_{1 . M}^{*}+h_{2 . M} h_{2 . M}^{*}}$


## Signal Flow Diagram for Complex Multiply of a+iband c+id

For Binary point scaling, you can not specify $W L_{p}$ and $F L_{p}$. Instead, the blocks determine these values implicitly from $W L_{a}$ and $F L_{a}$
The Internal Rule for Product output and Energy product output are:

- When you select Inherit via internal rule, the internal rule determines $W L_{p}$ and $F L_{p}$. Therefore, $W L_{a}=W L_{p}+1$ and $F L_{a}=F L_{p}$
- For Binary point scaling, you specify $W L_{a}$ and $F L_{a}$. Therefore, $W L_{p}=W L_{a}-1$ and $F L_{a}=F L_{p}$.


## OSTBC Combiner

For information on how the Internal Rule applies to the Accumulator and Energy Accumulator, see Inherit via Internal Rule in the DSP System Toolbox User's Guide.


## Number of transmit antennas

Sets the number of transmit antennas. The block supports 2,3 , or 4 transmit antennas. This value defaults to 2 .

## OSTBC Combiner

## Rate

Sets the symbol rate of the code. You can specify either $3 / 4$ or $1 / 2$. This field only appears when you use more than 2 transmit
antennas. This field defaults to $\frac{3}{4}$ for more than 2 transmit antennas. For 2 transmit antennas, there is no rate option and the implicit (default) rate defaults to 1 .

## Number of receive antennas

The number of antennas the block uses to receive signal streams. The block supports from 1 to 8 receive antennas. This value defaults to 1 .

## OSTBC Combiner



## Rounding mode

Sets the rounding mode for fixed-point calculations. The block uses the rounding mode if a value cannot be represented exactly by the specified data type and scaling. When this occurs, the value is rounded to a representable number. For more information refer to Rounding in Fixed-Point Designer.

## OSTBC Combiner

## Overflow mode

Sets the overflow mode for fixed-point calculations. Use this parameter to specify the method to be used if the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result. For more information refer to Precision and Range in the Precision and Range section of the DSP System Toolbox User's Guide.

## Product Output

Complex product in the numerator for the diversity combining. For more information refer to the Fixed-Point Signals section of this help page.

## Accumulator

Summation in the numerator for the diversity combining.
Fixed-point Communications System Toolbox blocks that must hold summation results for further calculation usually allow you to specify the data type and scaling of the accumulator. Most such blocks cast to the accumulator data type prior to summation:


Use the Accumulator-Mode parameter to specify how you would like to designate the accumulator word and fraction lengths:

## OSTBC Combiner

- When you select Inherit via internal rule, the accumulator output word and fraction lengths are automatically calculated for you. Refer to Inherit via Internal Rule for more information.
- When you select Same as product output, these characteristics match those of the product output.
- When you select Same as input, these characteristics match those of the first input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator. The bias of all signals in DSP System Toolbox software is zero.


## Energy product output

Complex product in the denominator for calculating total energy in the MIMO channel.

## Energy accumulator

Summation in the denominator for calculating total energy in the MIMO channel.

## Division output

Normalized diversity combining by total energy in the MIMO channel.

## OSTBC Combiner

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| Rx | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed Fixed-point |
| cEst | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed Fixed-point |
| Out | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed Fixed-point |

## Examples

See Also OSTBC Encoder

## Purpose

Encode input message using orthogonal space-time block code (OSTBC)

## Library

MIMO
Description


The OSTBC Encoder block encodes an input symbol sequence using orthogonal space-time block code (OSTBC). The block maps the input symbols block-wise and concatenates the output codeword matrices in the time domain. For more information, see the OSTBC Encoding Algorithms section of this help page.

## Dimension

The block supports time and spatial domains for OSTBC transmission. It also supports an optional dimension, over which the encoding calculation is independent. This dimension can be thought of as the frequency domain. The following illustration indicates the supported dimensions for the inputs and output of the OSTBC Encoder block.


The following table describes the variables.

| Variable | Description |
| :--- | :--- |
| F | The additional dimension; <br> typically the frequency domain. <br> The encoding does not depend on <br> this dimension. |
| T | Input symbol sequence length for <br> the time domain. |


| Variable | Description |
| :--- | :--- |
| R | Symbol rate of the code. |
| N | Number of transmit antennas. |

Note On the output, T/R is the symbol sequence length in time domain.
$F$ can be any positive integer. $N$ can be 2,3 or 4 , indicated by Number of transmit antennas. For $N=2, R$ must be 1 . For $N=3$ or $4, R$ can be $3 / 4$ or $1 / 2$, indicated by Rate. The time domain length $T$ must be a multiple of the number of symbols in each codeword matrix. Specifically, for $N=2$ or $R=1 / 2, T$ must be a multiple of 2 and when $R$ $=3 / 4, T$ must be a multiple of 3 .

To understand the block's dimension propagation, refer to the following table.

| Dimension | Input | Output |
| :--- | :--- | :--- |
| $\mathrm{F}=1$ | Column vector | 2-D |
| $\mathrm{F}>1$ | 2-D | 3-D |

## Data Type

For information about the data types each block port supports, see the "Supported Data Type" on page 2-692 table on this page. The output signal inherits the data type from the input signal. For fixed-point signals, the complex conjugation may cause overflows which the fixed-point parameter Overflow mode must handle.

## Frames

The output signal inherits frame type from the input signal. A column vector input requires either frame-based or sample-based input; otherwise, the input must be sample-based.

## OSTBC Encoding Algorithms

The OSTBC Encoder block supports five different OSTBC encoding algorithms. Depending on the selection for Rate and Number of transmit antennas, the block implements one of the algorithms in the following table:

| Transmit <br> Antenna | Rate | OSTBC Codeword Matrix |
| :--- | :--- | :--- |
| 2 | 1 | $\left(\begin{array}{cc}s_{1} & s_{2} \\ -s_{2}^{*} & s_{1}^{*}\end{array}\right)$ |
| 3 | $1 / 2$ | $\left(\begin{array}{ccc}s_{1} & s_{2} & 0 \\ -s_{2}^{*} & s_{1}^{*} & 0 \\ 0 & 0 & s_{1} \\ 0 & 0 & -s_{2}^{*}\end{array}\right)$ |
| 3 | $3 / 4$ | $\left(\begin{array}{ccc}s_{1} & s_{2} & s_{3} \\ -s_{2}^{*} & s_{1}^{*} & 0 \\ s_{3}^{*} & 0 & -s_{1}^{*} \\ 0 & s_{3}^{*} & -s_{2}^{*}\end{array}\right)$ |

## OSTBC Encoder

| Transmit <br> Antenna | Rate | OSTBC Codeword Matrix |
| :--- | :--- | :--- |
| 4 | $1 / 2$ | $\left(\begin{array}{cccc}s_{1} & s_{2} & 0 & 0 \\ -s_{2}^{*} & s_{1}^{*} & 0 & 0 \\ 0 & 0 & s_{1} & s_{2} \\ 0 & 0 & -s_{2}^{*} & s_{1}^{*}\end{array}\right)$ |
| 4 | $3 / 4$ | $\left(\begin{array}{cccc}s_{1} & s_{2} & s_{3} & 0 \\ -s_{2}^{*} & s_{1}^{*} & 0 & s_{3} \\ s_{3}^{*} & 0 & -s_{1}^{*} & s_{2} \\ 0 & s_{3}^{*} & -s_{2}^{*} & -s_{1}\end{array}\right)$ |

In each matrix, its $(l, i)$ entry indicates the symbol transmitted from the $i$ th antenna in the $l$ th time slot of the block. The value of $i$ can range from 1 to $N$ (the number of transmit antennas). The value of $l$ can range from 1 to the codeword block length.


Dialog Box

## Block Parameters

## Number of transmit antennas

Sets the number of antennas at the transmitter side. The block supports 2,3 , or 4 transmit antennas. The value defaults to 2 .

## Rate

Sets the symbol rate of the code. You can specify either $3 / 4$ or $1 / 2$. This field only appears when using more than 2 transmit
antennas. This field defaults to $\frac{3}{4}$ for more than 2 transmit antennas. For 2 transmit antennas, there is no rate option and the rate defaults to 1 .

## OSTBC Encoder



## Overflow mode

Sets the overflow mode for fixed-point calculations. Use this parameter to specify the method to be used if the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result. For more information refer to "Precision and Range" in DSP System Toolbox.

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | • Double-precision floating point |
|  | • Single-precision floating point |
|  | • Signed Fixed-point |
| Out | • Double-precision floating point |
|  | • Single-precision floating point |
|  | • Signed Fixed-point |

# Examples For an example of this block in use, see OSTBC Over 3x2 Rayleigh Fading Channel in the Communications System Toolbox documentation. The model shows the use of a rate $3 / 4$ OSTBC for 3 transmit and 2 receive antennas with BPSK modulation using independent fading links and AWGN <br> You can also see the block in the Concatenated OSTBC with TCM example by typing commtcmostbc. View the IEEE 802.16-2004 OFDM PHY Link, Including Space-Time Block Coding example by typing commwman80216dstbc at the MATLAB command line. 

See Also<br>OSTBC Combiner

## Purpose

## Library

Description

Generate orthogonal variable spreading factor (OVSF) code from set of orthogonal codes

## Spreading Codes

The OVSF Code Generator block generates an OVSF code from a set of orthogonal codes. OVSF codes were first introduced for 3G communication systems. OVSF codes are primarily used to preserve orthogonality between different channels in a communication system.

OVSF codes are defined as the rows of an N -by- N matrix, $\mathrm{C}_{\mathrm{N}}$, which is defined recursively as follows. First, define $\mathrm{C}_{1}=[1]$. Next, assume that $\mathrm{C}_{\mathrm{N}}$ is defined and let $\mathrm{C}_{\mathrm{N}}(\mathrm{k})$ denote the kth row of $\mathrm{C}_{\mathrm{N}}$. Define $\mathrm{C}_{2 \mathrm{~N}}$ by

$$
C_{2 N}=\left[\begin{array}{cc}
C_{N}(0) & C_{N}(0) \\
C_{N}(0) & -C_{N}(0) \\
C_{N}(1) & C_{N}(1) \\
C_{N}(1) & -C_{N}(1) \\
\cdots & \cdots \\
C_{N}(N-1) & C_{N}(N-1) \\
C_{N}(N-1) & -C_{N}(N-1)
\end{array}\right]
$$

Note that $C_{\mathrm{N}}$ is only defined for $N$ a power of 2 . It follows by induction that the rows of $C_{\mathrm{N}}$ are orthogonal.

The OVSF codes can also be defined recursively by a tree structure, as shown in the following figure.


If [C] is a code length $2^{\mathrm{r}}$ at depth r in the tree, where the root has depth 0 , the two branches leading out of C are labeled by the sequences [C C] and [C - C], which have length $2^{r+1}$. The codes at depth $r$ in the tree are the rows of the matrix $\mathrm{C}_{\mathrm{N}}$, where $\mathrm{N}=2^{\mathrm{r}}$.

Note that two OVSF codes are orthogonal if and only if neither code lies on the path from the other code to the root. Since codes assigned to different users in the same cell must be orthogonal, this restricts the number of available codes for a given cell. For example, if the code $\mathrm{C}_{41}$

## OVSF Code Generator

in the tree is assigned to a user, the codes $\mathrm{C}_{10}, \mathrm{C}_{20}, \mathrm{C}_{82}, \mathrm{C}_{83}$, and so on, cannot be assigned to any other user in the same cell.

## Block Parameters

You specify the code the OVSF Code Generator block outputs by two parameters in the block's dialog: the Spreading factor, which is the length of the code, and the Code index, which must be an integer in the range $[0,1, \ldots, \mathrm{~N}-1]$, where N is the spreading factor. If the code appears at depth $r$ in the preceding tree, the Spreading factor is $2^{r}$. The Code index specifies how far down the column of the tree at depth $r$ the code appears, counting from 0 to $N-1$. For $\mathrm{C}_{\mathrm{N}, \mathrm{k}}$ in the preceding diagram, $N$ is the Spreading factor and $k$ is the Code index.

You can recover the code from the Spreading factor and the Code index as follows. Convert the Code index to the corresponding binary number, and then add 0 s to the left, if necessary, so that the resulting binary sequence $x_{1} x_{2} \ldots x_{\mathrm{r}}$ has length $r$, where $r$ is the logarithm base 2 of the Spreading factor. This sequence describes the path from the root to the code. The path takes the upper branch from the code at depth $i$ if $\mathrm{x}_{\mathrm{i}}=0$, and the lower branch if $\mathrm{x}_{\mathrm{i}}=1$.

To reconstruct the code, recursively define a sequence of codes $\mathrm{C}_{\mathrm{i}}$ for as follows. Let $\mathrm{C}_{0}$ be the root [1]. Assuming that $\mathrm{C}_{\mathrm{i}}$ has been defined, for $i<r$, define $\mathrm{C}_{\mathrm{i}+1}$ by

$$
C_{i+1}= \begin{cases}C_{i} C_{i} & \text { if } x_{i}=0 \\ C_{i}\left(-C_{i}\right) & \text { if } x_{i}=1\end{cases}
$$

The code $\mathrm{C}_{\mathrm{N}}$ has the specified Spreading factor and Code index.
For example, to find the code with Spreading factor 16 and Code index 6 , do the following:

1 Convert 6 to the binary number 110.
2 Add one 0 to the left to obtain 0110 , which has length $4=\log _{2} 16$.
3 Construct the sequences $\mathrm{C}_{\mathrm{i}}$ according to the following table.

| i | $\mathrm{x}_{\mathbf{i}}$ | C |
| :---: | :---: | :---: |
| 0 |  | $\mathrm{C}_{0}=[1]$ |
| 1 | 0 | $\mathrm{C}_{1}=\mathrm{C}_{0} \mathrm{C}_{0}=[1][1]$ |
| 2 | 1 | $\mathrm{C}_{2}=\mathrm{C}_{1}-\mathrm{C}_{1}=\left[\begin{array}{lll}1 & 1\end{array}\right][-1-1]$ |
| 3 | 1 | $\mathrm{C}_{3}=\mathrm{C}_{2}-\mathrm{C}_{2}=\left[\begin{array}{llll}1 & 1 & -1 & -1\end{array}\right]\left[\begin{array}{lllll}-1 & -1 & 1 & 1\end{array}\right]$ |
| 4 | 0 |  |

The code $\mathrm{C}_{4}$ has Spreading factor 16 and Code index 6.

## OVSF Code Generator



Dialog Box

## Spreading factor

Positive integer that is a power of 2 , specifying the length of the code.

## Code index

Integer in the range $[0,1, \ldots, N-1]$ specifying the code, where N is the Spreading factor.

## Sample time

A positive real scalar specifying the sample time of the output signal.

## Frame-based outputs

Determines whether the output is frame-based or sample-based.

## Samples per frame

The number of samples in a frame-based output signal. This field is active only if you select Frame-based outputs.

## Output data type

The output type of the block can be specified as an int8 or double. By default, the block sets this to double.

See Also
Hadamard Code Generator, Walsh Code Generator

## Phase/Frequency Offset

## Purpose

## Library

Description
Phase/ Frequency Offset

Apply phase and frequency offsets to complex baseband signal
RF Impairments
The Phase/Frequency Offset block applies phase and frequency offsets to an incoming signal.

The block inherits its output data type from the input signal. If the input signal is $u(t)$, then the output signal is:

$$
y(t)=u(t) \cdot\left(\cos \left(2 \pi \int_{0}^{t} f(\tau) d \tau+\varphi(t)\right)+j \sin \left(2 \pi \int_{0}^{t} f(\tau) d \tau+\varphi(t)\right)\right)
$$

where
$f(t)=$ Frequency offset
$\varphi(t)=$ Phase offset
The discrete-time output is:

$$
\begin{aligned}
& y(0)=u(0)(\cos (\varphi(0))+j \sin (\varphi(0))) \\
& y(i)=u(i)(\cos (2 \pi f(i-1) \Delta t+\varphi(i))+j \sin (2 \pi f(i-1) \Delta t+\varphi(i))) \quad i>0
\end{aligned}
$$

where
$\Delta t=$ Sample time
This block accepts real and complex inputs of data type double or single.

## Phase Offset

The block applies a phase offset to the input signal, specified by the Phase offset parameter.

## Frequency Offset

The block applies a frequency offset to the input signal, specified by the Frequency offset parameter. Alternatively, when you select

## Phase/Frequency Offset

Frequency offset from port, the Frq input port provides the offset to the block. The frequency offset must be a scalar value, vector with the same number of rows or columns as the data input, or a matrix with the same size as the data input. For more information, see "Interdependent Parameter-Port Dimensions" on page 2-703.

The effects of changing the block's parameters are illustrated by the following scatter plots of a signal modulated by 16 -ary quadrature amplitude modulation (QAM). The usual 16-ary QAM constellation without the effect of the Phase/Frequency Offset block is shown in the first scatter plot:


The following figure shows a scatter plot of an output signal, modulated by 16 -ary QAM, from the Phase/Frequency Offset block with Phase offset set to 20 and Frequency offset set to 0:

## Phase/Frequency Offset



Observe that each point in the constellation is rotated by a 20 degree angle counterclockwise.
If you set Frequency offset to 2 and Phase offset to 0, the angles of points in the constellation change linearly over time. This causes points in the scatter plot to shift radially, as shown in the following figure:

## Phase/Frequency Offset



Note that every point in the scatter plot has magnitude equal to a point in the original constellation.

See "Illustrate RF Impairments That Distort a Signal" for a description of the model that generates this plot.

Interdependent Parameter-Port Dimensions

| Number of <br> Dimensions | Data I/O <br> Dimension | Frame <br> Size | Number <br> of <br> Channels | Frequency/Phase <br> Offset Parameter <br> Dimension | Frequency <br> Offset Input <br> Port Dimension |
| :--- | :--- | :--- | :--- | :--- | :--- |
| Any | Scalar | 1 | 1 | Scalar | Scalar |
| 2 | $M$-by-1 | $M$ | 1 | $M$-by-1, 1-by- $M$, <br> 1 -by-1 | $M, M$-by-1, 1, <br> 1 -by-1 |

## Phase/Frequency Offset

| Number of <br> Dimensions | Data I/O <br> Dimension | Frame <br> Size | Number <br> of <br> Channels | Frequency/Phase <br> Offset Parameter <br> Dimension | Frequency <br> Offset Input <br> Port Dimension |
| :--- | :--- | :--- | :--- | :--- | :--- |
| 2 | 1 -by- $N$ | 1 | $N$ | $N$-by-1, 1-by- $N$, <br> 1 -by-1 | $N, 1$-by- $N, 1$, <br> 1 -by-1 |
| 2 | $M$-by- $N$ | $M$ | $N$ | $M$-by- $N, N$-by-1, <br> 1 -by- $N, M$-by-1, <br> 1 -by- $M, 1$-by-1 | $M$-by- $N, N$, <br> 1 -by- $N, 1,1$-by-1, <br> $M, M$-by-1 |

- When you specify a scalar offset parameter the block applies the same offset to all elements of the input signal
- When you specify a 2 -by- 1 offset parameter for a 2 -by- 3 input signal (one offset value per sample), the block applies the same sample offset across the three channels.
- When you specify a 1 -by- 3 offset parameter for a 2 -by- 3 input signal (one offset value per channel), the same channel offset is applied across the two samples of a channel.
- When you specify a 2 -by- 3 offset parameter for a 2 -by- 3 input signal (one offset value per sample for each channel), the offsets are applied element-wise to the input signal.


## Phase/Frequency Offset

Dialog Box


## Frequency offset from port

Selecting this option opens a port on the block through which you can input the frequency offset information.

## Frequency offset

Specifies the frequency offset in hertz.
This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink User's Guide.

## Phase offset

Specifies the phase offset in degrees.
This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more

## Phase/Frequency Offset

information, see Tunable Parameters in the Simulink User's Guide.

If Frequency offset and Phase offset are both vectors or both matrices, their dimensions (vector lengths, or number of rows and columns) must be the same.

See Also Phase Noise

## Phase-Locked Loop

## Purpose

Implement phase-locked loop to recover phase of input signal

## Library

Description


Components sublibrary of Synchronization
The Phase-Locked Loop (PLL) block is a feedback control system that automatically adjusts the phase of a locally generated signal to match the phase of an input signal. This block is most appropriate when the input is a narrowband signal.

This PLL has these three components:

- A multiplier used as a phase detector.
- A filter. You specify the filter transfer function using the Lowpass filter numerator and Lowpass filter denominator parameters. Each is a vector that gives the respective polynomial's coefficients in order of descending powers of $s$.
To design a filter, you can use functions such as butter, cheby1, and cheby2 in Signal Processing Toolbox software. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.
[num, den] = cheby2(3,40,100,'s')
- A voltage-controlled oscillator (VCO). You specify characteristics of the VCO using the VCO quiescent frequency, VCO initial phase, and VCO output amplitude parameters.

This block accepts a sample-based scalar input signal. The input signal represents the received signal. The three output ports produce:

- The output of the filter
- The output of the phase detector
- The output of the VCO


## Phase-Locked Loop



Dialog Box

## Lowpass filter numerator

The numerator of the lowpass filter transfer function, represented as a vector that lists the coefficients in order of descending powers of $s$.

## Lowpass filter denominator

The denominator of the lowpass filter transfer function, represented as a vector that lists the coefficients in order of descending powers of $s$.

## VCO input sensitivity ( $\mathrm{Hz} / \mathrm{V}$ )

This value scales the input to the VCO and, consequently, the shift from the VCO quiescent frequency value. The units of VCO input sensitivity are Hertz per volt.

VCO quiescent frequency $(\mathrm{Hz})$
The frequency of the VCO signal when the voltage applied to it is zero. This should match the carrier frequency of the input signal.

## VCO initial phase (rad)

The initial phase of the VCO signal.

## VCO output amplitude

The amplitude of the VCO signal.

See Also
Baseband PLL, Linearized Baseband PLL, Charge Pump PLL
References For more information about phase-locked loops, see the works listed in "Selected Bibliography for Synchronization" in Communications System Toolbox User's Guide.

## Phase Noise

Description

## Purpose Apply receiver phase noise to complex baseband signal <br> Library <br> RF Impairments

The Phase Noise block applies receiver phase noise to a complex, baseband signal. The block applies the phase noise as follows:

1 Generates additive white Gaussian noise (AWGN) and filters it with a digital filter.

2 Adds the resulting noise to the angle component of the input signal.
Using this block, you can specify phase noise as a scalar frequency offset or a vector of frequency offsets.

- For a scalar frequency offset, the block generates phase noise over the entire spectral observation window, from 0 Hz (or as close as possible to 0 Hz ) to $\pm \frac{F_{S}}{2}$, where $F_{s}$ represents the sampling frequency. The noise is scaled ${ }^{2}$ so that it is at the block-specific phase noise level at the specified frequency offset. The block generates a phase noise with
$\frac{1}{f}$ characteristic over the entire frequency range.
- For a vector of frequency offsets, the block interpolates the spectrum mask across $\log 10$ (frequency), and is flat from the highest frequency offset to half the sample rate.

You can view the block's implementation of phase noise by right-clicking on the block and selecting Mask > Look under mask. This displays the following figure:


You can view the construction of the Noise Source subsystem by double-clicking it.
The effects of changing the block's parameters are illustrated by the following scatter plots of a signal modulated by 16 -ary quadrature amplitude modulation (QAM). The usual 16-ary QAM constellation without distortion is shown in the first scatter plot:


## Phase Noise

The following figure shows a scatter plot of an output signal, modulated by 16-ary QAM, from the Phase Noise block with Phase noise level $(\mathbf{d B c} / \mathrm{Hz})$ set to -70 and Frequency offset $(\mathrm{Hz})$ set to 100:


This plot is generated by the model described in "Illustrate RF Impairments That Distort a Signal" with the following parameter settings for the Rectangular QAM Modulator Baseband block:

- Normalization method set to Average Power
- Average power (watts) set to $1 \mathrm{e}-12$

Function Block Parameters: Phase Noise
Phase Noise (link)
Add receiver phase noise to complex baseband signal.
The phase noise is filtered according to the specified spectral mask. For a vector frequency offset specification, the spectrum mask is interpolated across log10(frequency), and is flat from the highest frequency offset to half the sample rate. For a scalar frequency offset specification, the spectrum has a $1 / \mathrm{f}$ characteristic that passes through the specified point.

For a scalar frequency offset specification, the Sample rate parameter is not used.

Use the plotPhaseNoiseFilter function to visualize the response of the phase noise filter.

Parameters
Phase noise level ( $\mathrm{dBc} / \mathrm{Hz}$ ):

```
-60-80]
```

Frequency offset (Hz):
[20 200]
Sample rate (Hz):
1024
Initial seed:
2137

## Phase noise level ( $\mathrm{dBc} / \mathrm{Hz}$ )

Scalar or vector that specifies the phase noise level. Specify the phase noise level in decibels relative to carrier per Hertz ( $\mathrm{dBc} / \mathrm{Hz}$ ). The lengths of the phase noise level and frequency offset vectors must be equal.

## Phase Noise

## Frequency offset (Hz)

Specifies the frequency offset in Hertz. If the frequency offset is a vector, then the vector must be monotonically increasing. The lengths of the phase noise level and frequency offset vectors must be equal.

## Sample rate (Hz)

Must be greater than twice the largest value of the Frequency offset vector to avoid aliasing. Specify in Hertz. When you specify a vector of frequency offsets, the block uses this parameter. The block does not use this parameter when you specify a scalar frequency offset.

The sample rate must match the sample rate of the input signal. This quantity is the actual sample rate, not the frame rate of a frame-based signal.

## Initial seed

Nonnegative integer specifying the initial seed for the random number generator the block uses to generate noise.

Examples<br>See Also Phase/Frequency Offset<br>See Also Phase/Frequency Offset<br>References<br>For an example model that uses this block, see "View Phase Noise Effects on Signal Spectral".<br>[1] Kasdin, N.J., "Discrete Simulation of Colored Noise and Stochastic Processes and 1/(f^alpha); Power Law Noise Generation," The Proceedings of the IEEE, May, 1995, Vol. 83, No. 5

## PM Demodulator Passband

## Purpose Demodulate PM-modulated data

## Library Analog Passband Modulation, in Modulation

Description The PM Demodulator Passband block demodulates a signal that was modulated using phase modulation. The input is a passband
 representation of the modulated signal. Both the input and output signals are real scalar signals.

For best results, use a carrier frequency which is estimated to be larger than $10 \%$ of your input signal's sample rate. This is due to the implementation of the Hilbert transform by means of a filter.

In the following example, we sample a 10 Hz input signal at 8000 samples per second. We then designate a Hilbert Transform filter of order 100. Below is the response of the Hilbert Transform filter as returned by fvtool.

## PM Demodulator Passband



Note the bandwidth of the filter's magnitude response. By choosing a carrier frequency larger than $10 \%$ (but less than $90 \%$ ) of the input signal's sample rate ( 8000 samples per second, in this example) or equivalently, a carrier frequency larger than 400 Hz , we ensure that the Hilbert Transform Filter will be operating in the flat section of the filter's magnitude response (shown in blue), and that our modulated signal will have the desired magnitude and form.
Typically, an appropriate Carrier frequency value is much higher than the highest frequency of the input signal. By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the Carrier frequency parameter.
This block works only with real inputs of type double. This block does not work inside a triggered subsystem.

## PM Demodulator Passband



Dialog Box

## Carrier frequency ( Hz )

The frequency of the carrier.

## Initial phase (rad)

The initial phase of the carrier in radians.

## Phase deviation (rad)

The phase deviation of the carrier frequency in radians.
Sometimes it is referred to as the "variation" in the phase.

## Hilbert transform filter order

The length of the FIR filter used to compute the Hilbert transform.
Pair Block PM Modulator Passband

## PM Modulator Passband

Purpose Modulate using phase modulation
Library
Analog Passband Modulation, in Modulation
Description The PM Modulator Passband block modulates using phase modulation. The output is a passband representation of the modulated signal. The
 output signal's frequency varies with the input signal's amplitude. Both the input and output signals are real scalar signals.

If the input is $u(t)$ as a function of time $t$, then the output is

$$
\cos \left(2 \pi f_{c} t+K_{c} u(t)+\theta\right)
$$

where

- $f_{\mathrm{c}}$ represents the Carrier frequency parameter
- $\theta$ represents the Initial phase parameter
- $K_{\mathrm{c}}$ represents the Phase deviation parameter

An appropriate Carrier frequency value is generally much higher than the highest frequency of the input signal. By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the Carrier frequency parameter.

This block works only with real inputs of type double. This block does not work inside a triggered subsystem.

## PM Modulator Passband

| Function Block Parameters: PM Modulator Passband |
| :--- |
| PM Modulator Passband (mask) (link) - <br> Modulate the input signal using the phase modulation method. <br> The input signal must be a scalar. <br> Parameters -   <br> Carrier frequency (Hz):   <br> 300 Cancel Help <br> Initial phase (rad): 0 Apply  <br> Phase deviation (rad):   |

Dialog Box

## Carrier frequency (Hz)

The frequency of the carrier.

## Initial phase (rad)

The initial phase of the carrier in radians.
Phase deviation (rad)
The phase deviation of the carrier frequency in radians. This is sometimes referred to as the "variation" in the phase.

Pair Block PM Demodulator Passband

## PN Sequence Generator

## Purpose Generate pseudonoise sequence

Library
Description

PN Sequence Generator

Sequence Generators sublibrary of Comm Sources
The PN Sequence Generator block generates a sequence of pseudorandom binary numbers using a linear-feedback shift register (LFSR). This block implements LFSR using a simple shift register generator (SSRG, or Fibonacci) configuration. A pseudonoise sequence can be used in a pseudorandom scrambler and descrambler. It can also be used in a direct-sequence spread-spectrum system.

This block can output sequences that vary in length during simulation. For more information about variable-size signals, see "Variable-Size Signal Basics" in the Simulink documentation.

The PN Sequence Generator block uses a shift register to generate sequences, as shown below.

XOR addition


## PN Sequence Generator

All $r$ registers in the generator update their values at each time step, according to the value of the incoming arrow to the shift register. The adders perform addition modulo 2 . The shift register is described by the Generator Polynomial parameter, which is a primitive binary polynomial in $z, \mathrm{~g}_{\mathrm{r}} \mathrm{z}^{\mathrm{r}}+\mathrm{g}_{\mathrm{r}-1} \mathrm{r}^{\mathrm{r}-1}+\mathrm{g}_{\mathrm{r}-2^{\mathrm{r}}} \mathrm{r}^{-2}+\ldots+\mathrm{g}_{0}$. The coefficient $\mathrm{g}_{\mathrm{k}}$ is 1 if there is a connection from the kth register, as labeled in the preceding diagram, to the adder. The leading term $\mathrm{g}_{\mathrm{r}}$ and the constant term $\mathrm{g}_{0}$ of the Generator Polynomial parameter must be 1 because the polynomial must be primitive.

You can specify the Generator polynomial parameter using either of these formats:

- A vector that lists the coefficients of the polynomial in descending order of powers. The first and last entries must be 1 . Note that the length of this vector is one more than the degree of the generator polynomial.
- A vector containing the exponents of $z$ for the nonzero terms of the polynomial in descending order of powers. The last entry must be 0 .

For example, $\left[\begin{array}{lllllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0 & 1\end{array}\right]$ and $\left[\begin{array}{lll}8 & 2 & 0\end{array}\right]$ represent the same polynomial, $p(z)=z^{8}+z^{2}+1$.

The Initial states parameter is a vector specifying the initial values of the registers. The Initial states parameter must satisfy these criteria:

- All elements of the Initial states vector must be binary numbers.
- The length of the Initial states vector must equal the degree of the generator polynomial.

Note At least one element of the Initial states vector must be nonzero in order for the block to generate a nonzero sequence. That is, the initial state of at least one of the registers must be nonzero.

## PN Sequence Generator

For example, the following table indicates two sets of parameter values that correspond to a generator polynomial of $p(z)=z^{8}+z^{2}+1$.

| Quantity | Example 1 |  | Example 2 |
| :--- | :--- | :--- | :--- |
| Generator <br> polynomial | $\left.\begin{array}{l}g 1=\left[\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0 \\ 1\end{array}\right]\end{array}\right] g 2=\left[\begin{array}{llll}8 & 2 & 0\end{array}\right]$ |  |  |
| Degree of <br> generator <br> polynomial | 8, which is length $(g 1)-1$ | 8 |  |
| Initial <br> states | $\left[\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0\end{array}\right]$ | $\left[\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0\end{array}\right]$ |  |

Output mask vector (or scalar shift value) shifts the starting point of the output sequence. With the default setting for this parameter, the only connection is along the arrow labeled $m_{0}$, which corresponds to a shift of 0 . The parameter is described in greater detail below.
You can shift the starting point of the PN sequence with Output mask vector (or scalar shift value). You can specify the parameter in either of two ways:

- An integer representing the length of the shift
- A binary vector, called the mask vector, whose length is equal to the degree of the generator polynomial

The difference between the block's output when you set Output mask vector (or scalar shift value) to 0 , versus a positive integer $d$, is shown in the following table.

|  | $\mathbf{T}=\mathbf{0}$ | $\mathbf{T}=\mathbf{1}$ | $\mathbf{T}=\mathbf{2}$ | $\ldots$ | $\mathbf{T}=\mathbf{d}$ | $\mathbf{T}=\mathbf{1}$ <br> $\mathbf{d}+\mathbf{1}$ |
| :--- | :--- | :--- | :--- | :--- | :--- | :--- |
| Shift $=0$ | $x_{0}$ | $x_{1}$ | $x_{2}$ | $\ldots$ | $x_{\mathrm{d}}$ | $x_{\mathrm{d}+1}$ |
| Shift $=\mathrm{d}$ | $x_{\mathrm{d}}$ | $x_{\mathrm{d}+1}$ | $x_{\mathrm{d}+2}$ | $\ldots$ | $x_{2 \mathrm{~d}}$ | $x_{2 \mathrm{~d}+1}$ |

## PN Sequence Generator

Alternatively, you can set Output mask vector (or scalar shift value) to a binary vector, corresponding to a polynomial in $z, \mathrm{~m}_{\mathrm{r}-1} \mathrm{z}^{\mathrm{r}-1}$ $+\mathrm{m}_{\mathrm{r}-2} \mathrm{z}^{\mathrm{r}-2}+\ldots+\mathrm{m}_{1} \mathrm{z}+\mathrm{m}_{0}$, of degree at most $\mathrm{r}-1$. The mask vector corresponding to a shift of $d$ is the vector that represents $m(z)=z^{d}$ modulo $\mathrm{g}(z)$, where $\mathrm{g}(z)$ is the generator polynomial. For example, if the degree of the generator polynomial is 4 , then the mask vector corresponding to $\mathrm{d}=2$ is $\left[\begin{array}{lll}0 & 1 & 0\end{array} 0\right]$, which represents the polynomial $m(z)=z^{2}$. The preceding schematic diagram shows how Output mask vector (or scalar shift value) is implemented when you specify it as a mask vector. The default setting for Output mask vector (or scalar shift value) is 0 . You can calculate the mask vector using the Communications System Toolbox function shift2mask.

You can use an external signal to reset the values of the internal shift register to the initial state by selecting Reset on nonzero input. This creates an input port for the external signal in the PN Sequence Generator block. The way the block resets the internal shift register depends on whether its output signal and the reset signal are sample-based or frame-based. The following example demonstrates the possible alternatives.

## Example: Resetting a Signal

Suppose that the PN Sequence Generator block outputs $\left[\begin{array}{llllll}1 & 0 & 0 & 1 & 1 & 0\end{array}\right.$ 1 1] when there is no reset. You then select Reset on nonzero input and input a reset signal $\left[\begin{array}{lll}0 & 0 & 0\end{array}\right]$. The following table shows three possibilities for the properties of the reset signal and the PN Sequence Generator block.

## PN Sequence Generator

| Reset Signal Properties | PN Sequence Generator block | Reset Signal, Output Signal |
| :---: | :---: | :---: |
| Sample-based <br> Sample time $=1$ | Sample-based <br> Sample time $=1$ |  |
| Frame-based <br> Sample time $=1$ <br> Samples per frame $=2$ | Frame-based <br> Sample time = 1 <br> Samples per frame $=2$ |  |
| Sample-based <br> Sample time $=2$ <br> Samples per <br> frame $=1$ | Frame-based <br> Sample time $=1$ <br> Samples per <br> frame $=2$ |  |

In the first two cases, the PN sequence is reset at the fourth bit, because the fourth bit of the reset signal is a 1 and the Sample time is 1 . Note that in the second case, the frame sizes are 2 , and the reset occurs at the end of the second frame.

In the third case, the PN sequence is reset at the seventh bit. This is because the reset signal has Sample time 2, so the reset bit is first sampled at the seventh bit. With these settings, the reset always occurs at the beginning of a frame.

## Attributes of Output Signal

If the Frame-based outputs box is selected, the output signal is a frame-based column vector whose length is the Samples per frame parameter. Otherwise, the output signal is a one-dimensional scalar.

## Sequences of Maximum Length

If you want to generate a sequence of the maximum possible length for a fixed degree, $r$, of the generator polynomial, you can set

## PN Sequence Generator

Generator polynomial to a value from the following table. See [1] for more information about the shift-register configurations that these polynomials represent.

| r | Generator Polynomial | r | Generator Polynomial |
| :---: | :---: | :---: | :---: |
| 2 | $\left[\begin{array}{lll}2 & 1 & 0\end{array}\right]$ | 21 | $\left[\begin{array}{lll}21 & 19 & 0\end{array}\right]$ |
| 3 | $\left[\begin{array}{lll}3 & 2 & 0\end{array}\right]$ | 22 | [22 2100 |
| 4 | $\left[\begin{array}{lll}4 & 3 & 0\end{array}\right]$ | 23 | $\left[\begin{array}{llll}23 & 18 & 0\end{array}\right]$ |
| 5 | $\left[\begin{array}{lll}5 & 3 & 0\end{array}\right]$ | 24 | $\left[\begin{array}{llllll}24 & 23 & 22 & 17 & 0\end{array}\right]$ |
| 6 | $\left[\begin{array}{lll}6 & 5 & 0\end{array}\right]$ | 25 | $\left[\begin{array}{lll}25 & 22 & 0\end{array}\right]$ |
| 7 | $\left[\begin{array}{lll}7 & 6 & 0\end{array}\right]$ | 26 | $\left[\begin{array}{lllll}26 & 25 & 24 & 20 & 0\end{array}\right]$ |
| 8 | $\left[\begin{array}{lllll}8 & 6 & 5 & 4 & 0\end{array}\right]$ | 27 | $\left[\begin{array}{lllll}27 & 26 & 25 & 22 & 0\end{array}\right]$ |
| 9 | $\left[\begin{array}{lll}9 & 5 & 0\end{array}\right]$ | 28 | [ $\left.\begin{array}{lll}28 & 25 & 0\end{array}\right]$ |
| 10 | $\left[\begin{array}{lll}10 & 7 & 0\end{array}\right]$ | 29 | $\left[\begin{array}{lll}29 & 27 & 0\end{array}\right]$ |
| 11 | $\left[\begin{array}{lll}11 & 9 & 0\end{array}\right]$ | 30 | $\left[\begin{array}{llllll}30 & 29 & 28 & 7 & 0\end{array}\right]$ |
| 12 | $\left[\begin{array}{llllll}12 & 11 & 8 & 6 & 0\end{array}\right]$ | 31 | $\left[\begin{array}{lll}31 & 28 & 0\end{array}\right]$ |
| 13 | $\left[\begin{array}{llllll}13 & 12 & 10 & 9 & 0\end{array}\right]$ | 32 | $\left[\begin{array}{llllll}32 & 31 & 30 & 10 & 0\end{array}\right]$ |
| 14 | $\left[\begin{array}{llllll}14 & 13 & 8 & 4 & 0\end{array}\right]$ | 33 | $\left[\begin{array}{llll}33 & 20 & 0\end{array}\right]$ |
| 15 | $\left[\begin{array}{llll}15 & 14 & 0\end{array}\right]$ | 34 | $\left[\begin{array}{llllll}34 & 15 & 14 & 1 & 0\end{array}\right]$ |
| 16 | $\left[\begin{array}{llllll}16 & 15 & 13 & 4 & 0\end{array}\right]$ | 35 | $\left[\begin{array}{lll}35 & 2 & 0\end{array}\right]$ |
| 17 | $\left[\begin{array}{llll}17 & 14 & 0\end{array}\right]$ | 36 | $\left[\begin{array}{llll}36 & 11 & 0\end{array}\right]$ |
| 18 | $\left[\begin{array}{llll}18 & 11 & 0\end{array}\right]$ | 37 | [37 121210200$]$ |
| 19 | $\left[\begin{array}{lllllll}19 & 18 & 17 & 14 & 0\end{array}\right]$ | 38 | [38864rlll |
| 20 | $\left[\begin{array}{llll}20 & 17 & 0\end{array}\right]$ | 39 | [39 800 |
| 40 | $\left[\begin{array}{lllll}40 & 5 & 4 & 3 & 0\end{array}\right]$ | 47 | [47 14 0] |
| 41 | $\left[\begin{array}{lll}41 & 3 & 0\end{array}\right]$ | 48 | [48 28272710$]$ |

## PN Sequence Generator

| $\mathbf{r}$ | Generator <br> Polynomial | $\mathbf{r}$ | Generator Polynomial |
| :--- | :--- | :--- | :--- |
| 42 | $\left[\begin{array}{lllll}42 & 23 & 22 & 1 & 0\end{array}\right]$ | 49 | $\left[\begin{array}{llll}4 & 9 & 0\end{array}\right]$ |
| 43 | $\left[\begin{array}{lllll}43 & 6 & 4 & 3 & 0\end{array}\right]$ | 50 | $\left[\begin{array}{lllll}5 & 4 & 3 & 2 & 0\end{array}\right]$ |
| 44 | $\left[\begin{array}{lllll}44 & 6 & 5 & 2 & 0\end{array}\right]$ | 51 | $\left[\begin{array}{llllll}5 & 6 & 3 & 1 & 0\end{array}\right]$ |
| 45 | $\left[\begin{array}{llllll}45 & 4 & 3 & 1 & 0\end{array}\right]$ | 52 | $\left[\begin{array}{lllll}5 & 3 & 0\end{array}\right]$ |
| 46 | $\left[\begin{array}{llllll}46 & 21 & 10 & 1 & 0\end{array}\right]$ | 53 | $\left[\begin{array}{lllll}5 & 6 & 2 & 1 & 0\end{array}\right]$ |

## Example of PN Sequence Generation

This example clarifies the operation of the PN Sequence Generator block by comparing the output sequence from the library block with that generated from primitive Simulink blocks.

To open the model, enter doc_pnseq2 at the MATLAB command line.

## PN Sequence Generation



For the chosen generator polynomial, $p(z)=z^{6}+z+1$, the model generates a PN sequence of period 63, using both the library block and corresponding Simulink blocks. It shows how the two parameters, Initial states and Output mask vector (or scalar shift value), are interpreted in the latter schematic.

You can experiment with different initial states, by changing the value of Initial states prior to running the simulation. For all values, the two generated sequences are the same.

Using the PN Sequence Generator block allows you to easily generate PN sequences of large periods.
T. Source Block Parameters: PN Sequence Generator

PN Sequence Generator
Generate a pseudorandom noise (PN) sequence using a linear feedback shift register (LFSR). The LFSR is implemented using a simple shift register generator (SSRG, or Fibonacci) configuration.

The 'Generator polynomial' parameter values specify the shift register connections. Enter these values as either a binary vector or a descending-ordered polynomial. For the binary vector representation, the first and last elements of the vector must be 1 . For the descending-ordered polynomial representation, the last element of the vector must be 0 .

The 'Output mask source' may be from a dialog parameter or an input port. The 'Output mask vector' is a binary vector corresponding to the shift register states that are to be XORed to produce the output sequence values. Alternatively, you may enter an integer 'scalar shift value' to produce an equivalent advance or delay in the output sequence.

For variable-size output signals, the current output size is either specified from the 'oSiz' input or inherited from the 'Ref' input.

Parameters
Generator polynomial: $\left[\begin{array}{lllllll}1 & 0 & 0 & 0 & 0 & 1 & 1\end{array}\right]$
Initial states: $\left[\begin{array}{llllll}0 & 0 & 0 & 0 & 0 & 1\end{array}\right]$
Output mask source: Dialog parameter
Output mask vector (or scalar shift value): 0
$\square$ Output variable-size signals
Sample time: 1
$\square$ Frame-based outputsReset on nonzero inputEnable bit-packed outputs
Output data type: double

## Generator polynomial

Polynomial that determines the shift register's feedback connections.

## Initial states

Vector of initial states of the shift registers.

## Output mask source

Specifies how output mask information is given to the block.

- When you set this parameter to Dialog parameter, the field Output mask vector (or scalar shift value) is enabled for user input.
- When set this parameter to Input port, a Mask input port appears on the block icon. The Mask input port only accepts mask vectors.


## Output mask vector (or scalar shift value)

This field is available only when Output mask source is set to Dialog parameter.

Integer scalar or binary vector that determines the delay of the PN sequence from the initial time. If you specify the shift as a binary vector, the vector's length must equal the degree of the generator polynomial.

## Output variable-size signals

Select this check box if you want the output sequences to vary in length during simulation. The default selection outputs fixed-length signals.

## Maximum output size source

Specify how the block defines maximum output size for a signal.

- When you select Dialog parameter, the value you enter in the Maximum output size parameter specifies the maximum size of the output. When you make this selection, the oSiz input port specifies the current size of the output signal and the block output inherits sample time from the input signal. The input value must be less than or equal to the Maximum output size parameter.


## PN Sequence Generator

- When you select Inherit from reference port, the block output inherits sample time, maximum size, and current size from the variable-sized signal at the Ref input port.

This parameter only appears when you select Output variable-size signals. The default selection is Dialog parameter.

## Maximum output size

Specify a two-element row vector denoting the maximum output size for the block. The second element of the vector must be 1 For example, [101] gives a 10-by-1 maximum sized output signal. This parameter only appears when you select Output variable-size signals.

## Sample time

Period of each element of the output signal.

## Frame-based outputs

Determines whether the output is frame-based or sample-based.

## Samples per frame

The number of samples in a frame-based output signal. This field is active only if you select Frame-based outputs.

## Reset on nonzero input

When selected, you can specify an input signal that resets the internal shift registers to the original values of the Initial states parameter.

## Enable bit-packed outputs

When selected, the field Number of packed bits and the option Interpret bit-packed values as signed is enabled.

## Number of packed bits

Indicates how many bits to pack into each output data word (allowable range is 1 to 32).

## PN Sequence Generator

Interpret bit-packed values as signed
Indicates whether packed bits are treated as signed or unsigned integer data values. When selected, a 1 in the most significant bit (sign bit) indicates a negative value.

## Output data type

By default, this is set to double.
When Enable bit-packed outputs is not selected, the output data type can be specified as a double, boolean, or Smallest unsigned integer. When the parameter is set to Smallest unsigned integer, the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in the Hardware Implementation pane, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

When Enable bit-packed outputs is selected, the output data type can be specified as double or Smallest integer. When the parameter is set to Smallest integer, the output data type is selected based on Interpret bit-packed values as signed, Number of packed bits, and the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in the Hardware Implementation pane, the output data type is the ideal minimum $n$-bit size, i.e., sfix ( $n$ ) or ufix ( $n$ ), based on Interpret bit-packed values as signed. For all other selections, it is a signed or unsigned integer with the smallest available word length large enough to fit n bits.

See Also Kasami Sequence Generator, Scrambler<br>References<br>[1] Proakis, John G., Digital Communications, Third edition, New York, McGraw Hill, 1995.

## PN Sequence Generator

[2] Lee, J. S., and L. E. Miller, CDMA Systems Engineering Handbook, Artech House, 1998.
[3] Golomb, S.W., Shift Register Sequences, Aegean Park Press, 1967.

## Poisson Integer Generator

## Purpose

Generate Poisson－distributed random integers

## Library

Random Data Sources sublibrary of Comm Sources
Description The Poisson Integer Generator block generates random integers using a Poisson distribution．The probability of generating a nonnegative integer $k$ is

$$
\lambda^{k} \exp (-\lambda) /(k!)
$$

where $\lambda$ is a positive number known as the Poisson parameter．
You can use the Poisson Integer Generator to generate noise in a binary transmission channel．In this case，the Poisson parameter Lambda should be less than 1 ，usually much less．

## Attributes of Output Signal

The output signal can be a frame－based matrix，a sample－based row or column vector，or a sample－based one－dimensional array．These attributes are controlled by the Frame－based outputs，Samples per frame，and Interpret vector parameters as 1－D parameters．See ＂Sources and Sinks＂in Communications System Toolbox User＇s Guide for more details．

The number of elements in the Initial seed parameter becomes the number of columns in a frame－based output or the number of elements in a sample－based vector output．Also，the shape（row or column）of the Initial seed parameter becomes the shape of a sample－based two－dimensional output signal．

## Poisson Integer Generator

Dialog
Box


## Lambda

The Poisson parameter $\lambda$. If it is a scalar, then every element in the output vector shares the same Poisson parameter.

## Initial seed

The initial seed value for the random number generator.

## Sample time

The period of each sample-based vector or each row of a frame-based matrix.

## Frame-based outputs

Determines whether the output is frame-based or sample-based. This box is active only if Interpret vector parameters as 1-D is unchecked.

## Poisson Integer Generator

## Samples per frame

The number of samples in each column of a frame-based output signal. This field is active only if Frame-based outputs is checked.

## Interpret vector parameters as 1-D

If this box is checked, then the output is a one-dimensional signal. Otherwise, the output is a two-dimensional signal. This box is active only if Frame-based outputs is unchecked.

## Output data type

The output type of the block can be specified as a double, int8, uint8, int16, uint16, int32, or uint32. By default, the block sets this to double.

See Also Random Integer Generator; poissrnd (Statistics Toolbox)

## Puncture

## Purpose

Output elements which correspond to 1 s in binary Puncture vector

## Library

Description

Puncture
Sequence Operations
The Puncture block creates an output vector by removing selected elements of the input vector and preserving others. This block accepts an input signal that is a real or complex vector of length $K$. The block determines which elements to remove and preserve by using the binary Puncture vector parameter.
and mod is the modulus function (mod in MATLAB).

- If Puncture vector $(\mathrm{n})=0$, then the block removes the $n^{\text {th }}$ element of the input vector and does not include it as part of the output vector.
- If Puncture vector $(\mathrm{n})=1$, then the block preserves the $n^{\text {th }}$ element of the input vector as part of the output vector.

The input length, $K$, must be an integer multiple of the Puncture vector parameter length. The block repeats the puncturing pattern, as necessary, to include all input elements. The preserved elements appear in the output vector in the same order in which they appear in the input vector.

The input signal and the puncture vector are both column vectors.
The block accepts signals with the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.

## Tif Function Block Parameters: Puncture $X$ Puncture (mask) (link) <br> Output the elements which correspond to is in the binary Puncture vector. <br> The length of the input must be an integer multiple of the length of the Puncture vector parameter. The block repeats the Puncture vector, when necessary, to output all input elements.

## Parameters

Puncture vector:

## 110101].

Dialog Box

## Examples

If the Puncture vector parameter is the six-element vector $[1 ; 0 ; 1 ; 1 ; 1 ; 0]$, then the block:

- Removes the second and sixth elements from the group of six input elements.
- Sends the first, third, fourth, and fifth elements to the output vector.

The diagram below depicts the block's operation on an input vector of [ $1 ; 2 ; 3 ; 4 ; 5 ; 6]$, using this Puncture vector parameter.

## Puncture



See Also
Insert Zero

## QPSK Demodulator Baseband

## Purpose Demodulate QPSK-modulated data

## Library PM, in Digital Baseband sublibrary of Modulation

Description The QPSK Demodulator Baseband block demodulates a signal that was modulated using the quaternary phase shift keying method. The input
 is a baseband representation of the modulated signal.

The input must be a complex signal. This block accepts a scalar or column vector input signal. For information about the data types each block port supports, see "Supported Data Types" on page 2-747.

## Algorithm



Hard-Decision QPSK Demodulator Signal Diagram for Trivial Phase
Offset (odd multiple of $\frac{\pi}{4}$ )

## QPSK Demodulator Baseband



## Hard-Decision QPSK Demodulator Floating-Point Signal Diagram for Nontrivial Phase Offset



## Hard-Decision QPSK Demodulator Fixed-Point Signal Diagram for Nontrivial Phase Offset

The exact LLR and approximate LLR cases (soft-decision) are described in "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the Communications System Toolbox User's Guide.

## QPSK Demodulator Baseband

Dialog Box


## Phase offset (rad)

The phase of the zeroth point of the signal constellation.

## Constellation ordering

Determines how the block maps each integer to a pair of output bits.

## Output type

Determines whether the output consists of integers or bits.
If the Output type parameter is set to Integer and
Constellation ordering is set to Binary, then the block maps the point

$$
\exp (j \theta+j \Pi m / 2)
$$

to $m$, where $\theta$ is the Phase offset parameter and $m$ is $0,1,2$, or 3 .

## QPSK Demodulator Baseband

The reference page for the QPSK Modulator Baseband block shows the signal constellations for the cases when Constellation ordering is set to either Binary or Gray.

If the Output type is set to Bit, then the output contains pairs of binary values if Decision type is set to Hard decision. The most significant bit (i.e. the left-most bit in the vector), is the first bit the block outputs.

If the Decision type is set to Log-likelihood ratio or Approximate log-likelihood ratio, then the output contains bitwise LLR or approximate LLR values, respectively.

## Decision type

Specifies the use of hard decision, LLR, or approximate LLR during demodulation. This parameter appears when you select Bit from the Output type drop-down list. The output values for Log-likelihood ratio and Approximate log-likelihood ratio decision types are of the same data type as the input values. For integer output, the block always performs Hard decision demodulation.

See "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the Communications System Toolbox User's Guide for algorithm details.

## Noise variance source

This field appears when Approximate log-likelinood ratio or Log-likelihood ratio is selected for Decision type.

When set to Dialog, the noise variance can be specified in the Noise variance field. When set to Port, a port appears on the block through which the noise variance can be input.

## Noise variance

This parameter appears when the Noise variance source is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

## QPSK Demodulator Baseband

If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:

- Inf to - Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.


Data Types Pane for Hard-Decision

## QPSK Demodulator Baseband

## Output

For bit outputs, when Decision type is set to Hard decision, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, uint32, or boolean.

For integer outputs, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, or uint32.

When this parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is a floating-point type (single or double). If the input data type is fixed-point, the output data type will work as if this parameter is set to 'Smallest unsigned integer'.

When this parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box of the model.

If ASIC/FPGA is selected in the Hardware Implementation pane, and Output type is Bit, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

If ASIC/FPGA is selected in the Hardware Implementation pane, and Output type is Integer, the output data type is the ideal minimum two-bit size, i.e., ufix(2). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit two bits, usually corresponding to the size of a char (e.g., uint8).

## QPSK Demodulator Baseband

## Derotate factor

This parameter only applies when the input is fixed-point and
Phase offset is not an even multiple of $\frac{\pi}{4}$.
You can select Same word length as input or Specify word length, in which case you define the word length using an input field.


## Data Types Pane for Soft-Decision

For bit outputs, when Decision type is set to Log-likelinood ratio or Approximate log-likelihood ratio, the output data type is
inherited from the input (e.g., if the input is of data type double, the output is also of data type double).

## Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | • Single-precision floating point |
|  | • Signed fixed-point when: |
|  | - Output type is Integer |
|  | - Output type is Bit and Decision type is Hard-decision |
| Var | - Double-precision floating point |
|  | - Single-precision floating point |

## Pair Block

See Also M-PSK Demodulator Baseband, BPSK Demodulator Baseband, DQPSK Demodulator Baseband

## QPSK Modulator Baseband

Purpose | Modulate using quaternary phase shift keying method |
| :--- |
| Library |
| PM in Digital Baseband sublibrary of Modulation |

## QPSK Modulator Baseband



In the previous figure, the most significant bit (i.e. the left-most bit), is the first bit input to the block. For additional information about Gray mapping, see the M-PSK Modulator Baseband help page.

## Constellation Visualization

The QPSK Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the Communications System Toolbox User's Guide.

## QPSK Modulator Baseband

Dialog Box


## Phase offset (rad)

The phase of the zeroth point of the signal constellation.

## Constellation ordering

Determines how the block maps each pair of input bits or input integers to constellation symbols.

## Input type

Indicates whether the input consists of integers or pairs of bits.

## Output data type

The output data type can be set to double, single, Fixed-point, User-defined, or Inherit via back propagation.

Setting this parameter to Fixed-point or User-defined enables fields in which you can further specify details. Setting this parameter to Inherit via back propagation, sets the output data type and scaling to match the following block.

## QPSK Modulator Baseband

## Output word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed - point for the Output data type parameter.

## Set output fraction length to

Specify the scaling of the fixed-point output by either of the following methods:

- Choose Best precision to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose User-defined to specify the output scaling in the Output fraction length parameter.

This parameter is only visible when you select Fixed-point for the Output data type parameter or when you select User-defined and the specified output data type is a fixed-point data type.

## User-defined data type

Specify any signed built-in or signed fixed-point data type. You can specify fixed-point data types using the sfix, sint, sfrac, and fixdt functions from Fixed-Point Designer. This parameter is only visible when you select User-defined for the Output data type parameter.

## Output fraction length

For fixed-point output data types, specify the number of fractional bits or bits to the right of the binary point. This parameter is only visible when you select Fixed-point or User-defined for the Output data type parameter and User-defined for the Set output fraction length to parameter.

## QPSK Modulator Baseband

Supported
Data
Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Boolean when Input type is Bit |
|  | - 8-, 16-, and 32-bit signed integers |
|  | - 8-, 16-, and 32-bit unsigned integers |
|  | - ufix(1) when Input type is Bit |
|  | - ufix(2) when Input type is Integer |
| Output | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed point |

Pair Block QPSK Demodulator Baseband<br>See Also M-PSK Modulator Baseband, BPSK Modulator Baseband, DQPSK Modulator Baseband

Purpose
Decode quantization index according to codebook

## Library

Description

Idx Quantizing $\underset{\text { Decoder }}{\text { (U) }}$.

## Source Coding

The Quantizing Decoder block converts quantization indices to the corresponding codebook values. The Quantization codebook parameter, a vector of length N , prescribes the possible output values. If the input is an integer k between 0 and $\mathrm{N}-1$, then the output is the
$(\mathrm{k}+1)$ st element of Quantization codebook.

The input must be a discrete-time signal. This block processes each vector element independently. For information about the data types each block port supports, see the "Supported Data Type" on page 2-755 table on this page.

Note The Quantizing Encoder block also uses a Quantization codebook parameter. The first output of that block corresponds to the input of Quantizing Decoder, while the second output of that block corresponds to the output of Quantizing Decoder.

## Quantizing Decoder



## Quantizing Decoder

| Supported Data Type | Port | Supported Data Types |
| :---: | :---: | :---: |
|  | Idx | - Double-precision floating point <br> - Single-precision floating point <br> - 8-, 16-, and 32 -bit signed integers <br> - 8 -, 16 -, and 32 -bit unsigned integers |
|  | Q(U) | - Double-precision floating point <br> - Single-precision floating point |

## Pair Block Quantizing Encoder

See Also
Scalar Quantizer (Obsolete) (DSP System Toolbox documentation)

## Quantizing Encoder

## Purpose Quantize signal using partition and codebook <br> Library

Description
The Quantizing Encoder block quantizes the input signal according to the Partition vector and encodes the input signal according to the Codebook vector. This block processes each vector element independently. The input must be a discrete-time signal. This block processes each vector element independently. For information about the data types each block port supports, see the "Supported Data Type" on page 2-758 table on this page.

The first output is the quantization index. The second output is the quantized signal. The values for the quantized signal are taken from the Codebook vector.

The Quantization partition parameter, P , is a real vector of length n whose entries are in strictly ascending order. The quantization index (second output signal value) corresponding to an input value of x is

- 0 if $\mathrm{x} \mathrm{P}(1)$
- m if $\mathrm{P}(\mathrm{m})<\mathrm{x} \mathrm{P}(\mathrm{m}+1)$
- $n$ if $\mathrm{P}(\mathrm{n})<\mathrm{x}$

The Quantization codebook parameter, whose length is $\mathrm{n}+1$, prescribes a value for each partition in the quantization. The first element of Quantization codebook is the value for the interval between negative infinity and the first element of P . The second output signal from this block contains the quantization of the input signal based on the quantization indices and prescribed values.

You can use the function lloyds in Communications System Toolbox with a representative sample of your data as training data, to obtain appropriate partition and codebook parameters.

## Quantizing Encoder

> Function Block Parameters: Quantizing Encoder X Quantizing Encoder (mask) (link)

> Quantize the input signal using a partition and a codebook.
> The input signal is quantized according to the Quantization partition vector and encoded according to the Quantization codebook vector.

> The first output is the index from the Quantization codebook vector. The second output is the quantized signal. The values for the quantized signal are taken from the Quantization codebook vector.
Parameters
Quantization partition:

## [-.75-.25 .25.75]

Quantization codebook:
[-. 825 - . 50 . 5 . 825 ]
Index output data type: double
OK
Cancel
Help
Apply

Dialog Box

## Quantization partition

The vector of endpoints of the partition intervals.

## Quantization codebook

The vector of output values assigned to each partition.

## Index output data type

Select the output data type.

## Quantizing Encoder

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| U | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed-point |
| Idx | • Double-precision floating point |
|  | - 8-, $16-$, and 32 -bit signed |
|  | integers |
|  | - 8-, 16 -, and 32 -bit unsigned |
|  | integers |
| Q(U) | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed-point |

## Pair Block

Quantizing Decoder
See Also
Scalar Quantizer (Obsolete) (DSP System Toolbox documentation), lloyds (Communications System Toolbox documentation)

## Purpose

## Library

Description


Filter input signal, possibly downsampling, using raised cosine FIR filter

Comm Filters
The Raised Cosine Receive Filter block filters the input signal using a normal raised cosine FIR filter or a square root raised cosine FIR filter. It also downsamples the filtered signal if you set the Output mode parameter to Downsampling. The FIR Decimation block implements this functionality. The Raised Cosine Receive Filter block's icon shows the filter's impulse response.

## Characteristics of the Filter

Characteristics of the raised cosine filter are the same as in the Raised Cosine Transmit Filter block, except that the length of the filter's input response has a slightly different expression: 2 * N * Group delay + 1, where N is the value of the Input samples per symbol parameter (not the Upsampling factor parameter, as in the case of the Raised Cosine Transmit Filter block).

If the Filter gain parameter is chosen to be User-specified, then the passband gain of the filter is:

- $20 \log _{10}$ (Input samples per $\operatorname{symbol}(N) \times$ Linear amplitude filter gain) for a normal filter.
- $20 \log _{10}$ (Input samples per $\operatorname{symbol}(N) \times$ Linear amplitude filter gain) for a square root filter.


## Downsampling the Filtered Signal

To have the block downsample the filtered signal, set the Output mode parameter to Downsampling. By default, downsampling is on. If $L$ is the Downsampling factor parameter value, then the block retains $1 / \mathrm{L}$ of the samples, choosing them as follows:

- If the Sample offset parameter is zero, then the block selects the samples of the filtered signal indexed by $1, \mathrm{~L}+1,2 * \mathrm{~L}+1,3 * \mathrm{~L}+1$, etc.


## Raised Cosine Receive Filter

- If the Sample offset parameter is a positive integer less than L, then the block initially discards that number of samples from the filtered signal and downsamples the remaining data as in the previous case.

To preserve the entire filtered signal and avoid downsampling, set Output mode to None. This setting is appropriate, for example, when the output from the filter block forms the input to a timing phase recovery block such as Squaring Timing Recovery. The timing phase recovery block performs the downsampling in that case.

## Input Signals and Output Signals

This block accepts a column vector or matrix input signal. For information about the data types each block port supports, see the "Supported Data Type" on page 2-768 table on this page.

If you set Output mode to None, then the input and output signals share the same sampling mode, sample time, and vector length.

If you set Output mode to Downsampling and Downsampling factor is L , then L and the input sampling mode determine characteristics of the output signal:

## Single-Rate Processing

When you set the Rate options parameter to Enforce single-rate processing, the input and output of the block have the same sample rate. To genereate the output while maintaining the input sample rate, the block resamples the data in each column of the input such that the frame size of the output $\left(M_{o}\right)$ is $1 / L$ times that of the input $\left(M_{o}=M_{i} / L\right)$, In this mode, the input frame size, $M_{i}$, must be a multiple of $L$.

## Multirate Processing

When you set the Rate options parameter to Allow multirate processing, the input and output of the block are the same size, but the sample rate of the output is $K$ times slower than that of the input. When the block is in multirate processing mode, you must also specify a value for the Input processing parameter:

- When you set the Input processing parameter to Elements as channels (sample based), the block treats an $M$-by- $N$ matrix input as $M^{*} N$ independent channels, and processes each channel over time. The output sample period $\left(T_{s o}\right)$ is $L$ times longer than the input sample period ( $T_{s o}=L^{*} T_{s i}$ ), and the input and output sizes are identical.
- When you set the Input processing parameter to Columns as channels (frame based), the block treats an $M_{i}$-by- $N$ matrix input as $N$ independent channels. The block processes each column of the input over time by keeping the frame size constant $\left(M_{i}=M_{o}\right)$, and making the output frame period $\left(T_{f_{0}}\right) L$ times longer than the input frame period ( $T_{f o}=L^{*} T_{f i}$ ).


## Exporting Filter Coefficients to the MATLAB Workspace

To examine or manipulate the coefficients of the filter that this block designs, select Export filter coefficients to workspace. Then set the Coefficient variable name parameter to the name of a variable that you want the block to create in the MATLAB workspace. Running the simulation causes the block to create the variable, overwriting any previous contents in case the variable already exists.

## Latency

For information pertaining to the latency of the block, see details in FIR Decimation.

## Raised Cosine Receive Filter



Dialog Box

## Filter type

The type of raised cosine filter: Square root or Normal.

## Input samples per symbol

An integer greater than 1 representing the number of samples per symbol in the input signal.

## Group delay

A positive integer that represents the number of symbol periods between the start of the filter response and its peak. The default selection for this parameter is 5 .

## Rolloff factor

The rolloff factor for the filter, a real number between 0 and 1.

## Input processing

Specify how the block processes the input signal. You can set this parameter to one of the following options:

- Columns as channels (frame based) - When you select this option, the block treats each column of the input as a separate channel.
- Elements as channels (sample based) - When you select this option, the block treats each element of the input as a separate channel.

Note The Inherited (this choice will be removed - see release notes) option will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

## Rate options

Specify the method by which the block should filter and downsample the input signal. You can select one of the following options:

- Enforce single-rate processing - When you select this option, the block maintains the input sample rate and processes the signal by decreasing the output frame size by a factor of $L$. To select this option, you must set the Input processing parameter to Columns as channels (frame based).


## Raised Cosine Receive Filter

- Allow multirate processing - When you select this option, the block processes the signal such that the output sample rate is $L$ times slower than the input sample rate.


## Output mode

Determines whether or not the block downsamples the signal after filtering. Choices are Downsampling and None.

## Downsampling factor

The factor by which the block downsamples the signal after filtering. This field appears only if Output mode is set to Downsampling.

## Sample offset

The number of filtered samples the block discards before downsampling. This field appears only if Output mode is set to Downsampling.

## Filter gain

Determines how the block scales the filter coefficients. Choices are Normalized and User-specified.

## Linear amplitude filter gain

A positive scalar used to scale the filter coefficients. This field appears only if Filter gain is set to User-specified.

## Export filter coefficients to workspace

Select this check box to create a variable in the MATLAB workspace that contains the filter coefficients.

## Coefficient variable name

The name of the variable to create in the MATLAB workspace.
This field appears only if Export filter coefficients to workspace is selected.

## Visualize filter with FVTool

If you click this button, then MATLAB launches the Filter Visualization Tool, fvtool, to analyze the raised cosine filter whenever you apply any changes to the block's parameters. If you launch fvtool for the filter, and subsequently change parameters in the mask, fvtool will not update. You will need to launch a

## Raised Cosine Receive Filter

new fvtool in order to see the new filter characteristics. Also note that if you have launched fvtool, then it will remain open even after the model is closed.


## Rounding mode

Select the rounding mode for fixed-point operations. The block uses the Rounding mode when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. The filter coefficients do

## Raised Cosine Receive Filter

not obey this parameter; they always round to Nearest. For more information, see Rounding Modes in the DSP System Toolbox documentation or "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

## Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

## Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator).

See the Coefficients section of the FIR Decimation help page and "Filter Structure Diagrams" in DSP System Toolbox Reference Guide for illustrations depicting the use of the coefficient data types in this block:

See the Coefficients subsection of the Digital Filter help page for descriptions of parameter settings.

- When you select Same word length as input, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Specify word length, you are able to enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the coefficients, in bits. If applicable, you are able to enter separate fraction lengths for the numerator and denominator coefficients.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the coefficients. If applicable, you are able to enter separate slopes for the numerator and denominator coefficients. This block requires power-of-two slope and a bias of zero.
- The filter coefficients do not obey the Rounding mode and the Overflow mode parameters; they are always saturated and rounded to Nearest.


## Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" in DSP System Toolbox Reference Guide for illustrations depicting the use of the product output data type in this block:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the product output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.


## Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as product output, these characteristics match those of the product output.


## Raised Cosine Receive Filter

- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.


## Output

Choose how you specify the output word length and fraction length:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as accumulator, these characteristics match those of the accumulator.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.
Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.

Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | • Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed-point |
| Out | • Double-precision floating point |
|  | - Single-precision floating point |
|  | • Signed fixed-point |

Pair Block Raised Cosine Transmit Filter<br>See Also Gaussian Filter, rcosine, rcosflt

## Raised Cosine Transmit Filter

## Purpose Upsample and filter input signal using raised cosine FIR filter <br> Library <br> Comm Filters

Description The Raised Cosine Transmit Filter block upsamples and filters the input signal using a normal raised cosine FIR filter or a square root raised cosine FIR filter. The block's icon shows the filter's impulse response.

## Characteristics of the Filter

The Filter type parameter determines which type of filter the block uses; choices are Normal and Square root.

The impulse response of a normal raised cosine filter with rolloff factor R and symbol period T is

$$
h(t)=\frac{\sin (\pi t / T)}{(\pi t / T)} \cdot \frac{\cos (\pi R t / T)}{\left(1-4 R^{2} t^{2} / T^{2}\right)}
$$

The impulse response of a square root raised cosine filter with rolloff factor $R$ is

$$
h(t)=4 R \frac{\cos ((1+R) \pi t / T)+\frac{\sin ((1-R) \pi t / T)}{(4 R t / T)}}{\pi \sqrt{T}\left(1-(4 R t / T)^{2}\right)}
$$

The impulse response of a square root raised cosine filter convolved with itself is approximately equal to the impulse response of a normal raised cosine filter.
The Group delay parameter is the number of symbol periods between the start of the filter's response and the peak of the filter's response. The group delay and the upsampling factor, N , determine the length of the filter's impulse response, which is 2 * N * Group delay +1 .
The Rolloff factor parameter is the filter's rolloff factor. It must be a real number between 0 and 1 . The rolloff factor determines the excess

## Raised Cosine Transmit Filter

bandwidth of the filter. For example, a rolloff factor of .5 means that the bandwidth of the filter is 1.5 times the input sampling frequency.

The Filter gain parameter indicates how the block normalizes the filter coefficients. If you choose Normalized, then the block uses an automatic scaling:

- If Filter type is Normal, then the block normalizes the filter coefficients so that the peak coefficient equals 1.
- If Filter type is Square root, then the block normalizes the filter coefficients so that the convolution of the filter with itself produces a normal raised cosine filter whose peak coefficient equals 1 .

As a result, the output is scaled by $\sqrt{N}$. If the output of this block feeds the input to the AWGN Channel block, specify the AWGN signal power parameter to be $1 / \mathrm{N}$.

If the Filter gain parameter is chosen to be User-specified, then the passband gain of the filter is:

- $20 \log _{10}$ (Upsampling factor $(N) \times$ Linear amplitude filter gain) for a normal filter.
- $20 \log _{10}(\sqrt{\text { Upsampling factor }(N)} \times$ Linear amplitude filter gain) for a square root filter.


## Input Signals and Output Signals

The input must be a discrete-time signal. This block accepts a column vector or matrix input signal. For information about the data types each block port supports, see the "Supported Data Type" on page 2-779 table on this page.

The Rate options method and the value of the Upsampling factor (N) parameter determine the characteristics of the output signal:

## Raised Cosine Transmit Filter

## Single-Rate Processing

When you set the Rate options parameter to Enforce single-rate processing, the input and output of the block have the same sample rate. To generate the output while maintaining the input sample rate, the block resamples the data in each column of the input such that the frame size of the output $\left(M_{o}\right)$ is $N$ times larger than that of the input $\left(M_{o}=M_{i}{ }^{*} N\right)$.

## Multirate Processing

When you set the Rate options parameter to Allow multirate processing, the input and output of the block are the same size. However, the sample rate of the output is $N$ times faster than that of the input (i.e. the output sample time is $1 / \mathrm{N}$ times the input sample time). When the block is in multirate processing mode, you must also specify a value for the Input processing parameter:

- When you set the Input processing parameter to Elements as channels (sample based), the block treats an $M$-by- $L$ matrix input as $M^{*} L$ independent channels, and processes each channel over time. The output sample period ( $T_{s o}$ ) is $N$ times shorter than the input sample period ( $T_{s o}=T_{s i} / N$ ), while the input and output sizes remain identical.
- When you set the Input processing parameter to Columns as channels (frame based), the block treats an $M_{i}$-by- $L$ matrix input as $L$ independent channels. The block processes each column of the input over time by keeping the frame size constant $\left(M_{i}=M_{0}\right)$, while making the output frame period $\left(T_{f_{0}}\right) N$ times shorter than the input frame period ( $T_{f o}=T_{f i} / N$ ).


## Exporting Filter Coefficients to the MATLAB Workspace

To examine or manipulate the coefficients of the filter that this block designs, select Export filter coefficients to workspace. Then set the Coefficient variable name parameter to the name of a variable that you want the block to create in the MATLAB workspace. Running the simulation causes the block to create the variable, overwriting any previous contents in case the variable already exists.

## Raised Cosine Transmit Filter



Dialog Box

## Filter type

The type of raised cosine filter: Square root or Normal.
Group delay
A positive integer that represents the number of symbol periods between the start of the filter response and its peak. The default selection for this parameter is 5 .

## Rolloff factor

The rolloff factor for the filter, a real number between 0 and 1.

## Raised Cosine Transmit Filter

## Upsampling factor

An integer greater than 1 representing the number of samples per symbol in the filtered output signal.

## Input processing

Specify how the block processes the input signal. You can set this parameter to one of the following options:

- Columns as channels (frame based) - When you select this option, the block treats each column of the input as a separate channel.
- Elements as channels (sample based) - When you select this option, the block treats each element of the input as a separate channel.

Note The Inherited (this choice will be removed - see release notes) option will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

## Rate options

Specify the method by which the block should upsample and filter the input signal. You can select one of the following options:

- Enforce single-rate processing - When you select this option, the block maintains the input sample rate, and processes the signal by increasing the output frame size by a factor of $N$. To select this option, you must set the Input processing parameter to Columns as channels (frame based).
- Allow multirate processing - When you select this option, the block processes the signal such that the output sample rate is $N$ times faster than the input sample rate.


## Raised Cosine Transmit Filter

## Filter gain

Determines how the block scales the filter coefficients. Choices are Normalized and User-specified.

## Linear amplitude filter gain

A positive scalar used to scale the filter coefficients. This field appears only if Filter gain is set to User-specified.

## Export filter coefficients to workspace

Select this check box to create a variable in the MATLAB workspace that contains the filter coefficients.

## Coefficient variable name

The name of the variable to create in the MATLAB workspace. This field appears only if Export filter coefficients to workspace is selected.

## Visualize filter with FVTool

If you click this button, then MATLAB launches the Filter Visualization Tool, fvtool, to analyze the raised cosine filter whenever you apply any changes to the block's parameters. If you launch fvtool for the filter, and subsequently change parameters in the mask, fvtool will not update. You will need to launch a new fvtool in order to see the new filter characteristics. Also note that if you have launched fvtool, then it will remain open even after the model is closed.

## Raised Cosine Transmit Filter



## Rounding mode

Select the rounding mode for fixed-point operations. The block uses the Rounding mode when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. The filter coefficients do not obey this parameter; they always round to Nearest. For more information, see Rounding Modes in the DSP System Toolbox documentation or "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

## Raised Cosine Transmit Filter

## Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

## Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator). See "Filter Structure Diagrams" in DSP System Toolbox Reference Guide for illustrations depicting the use of the coefficient data types in this block:

- When you select Same word length as input, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Specify word length, you are able to enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the coefficients, in bits. If applicable, you are able to enter separate fraction lengths for the numerator and denominator coefficients.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the coefficients. If applicable, you are able to enter separate slopes for the numerator and denominator coefficients. This block requires power-of-two slope and a bias of zero.
- The filter coefficients do not obey the Rounding mode and the Overflow mode parameters; they are always saturated and rounded to Nearest.


## Raised Cosine Transmit Filter

## Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" in DSP System Toolbox Reference Guide for illustrations depicting the use of the product output data type in this block:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the product output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.


## Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as product output, these characteristics match those of the product output.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.


## Raised Cosine Transmit Filter

## Output

Choose how you specify the output word length and fraction length:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as accumulator, these characteristics match those of the accumulator.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.


## Lock scaling against changes by the autoscaling tool

Select this check box to prevent any fixed-point scaling you specify for this block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed-point |
| Out | - Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed-point |

Pair Block<br>Raised Cosine Receive Filter<br>See Also Gaussian Filter, rcosine, rcosflt

## Random Deinterleaver

Purpose $\quad$ Restore ordering of input symbols using random permutation
Library
Block sublibrary of Interleaving
Description The Random Deinterleaver block rearranges the elements of its input vector using a random permutation. The Initial seed parameter initializes the random number generator that the block uses to determine the permutation. If this block and the Random Interleaver block have the same value for Initial seed, then the two blocks are inverses of each other.

This block accepts a column vector input signal. The Number of elements parameter indicates how many numbers are in the input vector.

The block accepts the following data types int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.


## Random Integer Generator

## Purpose Generate integers randomly distributed in range [0, M-1] <br> Library <br> Random Data Sources sublibrary of Comm Sources

Description The Random Integer Generator block generates uniformly distributed random integers in the range [ $0, M-1$ ], where $M$ is the $\mathbf{M}$-ary number defined in the dialog box.
The M-ary number can be either a scalar or a vector. If it is a scalar, then all output random variables are independent and identically distributed (i.i.d.). If the M-ary number is a vector, then its length must equal the length of the Initial seed; in this case each output has its own output range.

If the Initial seed parameter is a constant, then the resulting noise is repeatable.

## Attributes of Output Signal

The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the Frame-based outputs, Samples per frame, and Interpret vector parameters as 1-D parameters. See "Sources and Sinks" in Communications System ToolboxUser's Guide for more details.

The number of elements in the Initial seed parameter becomes the number of columns in a frame-based output or the number of elements in a sample-based vector output. Also, the shape (row or column) of the Initial seed parameter becomes the shape of a sample-based two-dimensional output signal.

## Random Integer Generator



Dialog Box

## M-ary number

The positive integer, or vector of positive integers, that indicates the range of output values.

## Initial seed

The initial seed value for the random number generator. The vector length of the seed determines the length of the output vector.

## Sample time

The period of each sample-based vector or each row of a frame-based matrix.

## Frame-based outputs

Determines whether the output is frame-based or sample-based. This box is active only if Interpret vector parameters as 1-D is unchecked.

## Random Integer Generator

## Samples per frame

The number of samples in each column of a frame-based output signal. This field is active only if Frame-based outputs is checked.

## Interpret vector parameters as 1-D

If this box is checked, then the output is a one-dimensional signal. Otherwise, the output is a two-dimensional signal. This box is active only if Frame-based outputs is unchecked.

## Output data type

The output type of the block can be specified as a boolean, int8, uint8, int16, uint16, int32, uint32, single, or double. By default, the block sets this to double. Single outputs may lead to different results when compared with double outputs for the same set of parameters. For Boolean typed outputs, the M-ary number must be 2 .
randint (Communications Toolbox)

## Random Interleaver

## Purpose

Reorder input symbols using random permutation

## Library

Description

Random Interleaver

Block sublibrary of Interleaving
The Random Interleaver block rearranges the elements of its input vector using a random permutation. This block accepts a column vector input signal. The Number of elements parameter indicates how many numbers are in the input vector.

The block accepts the following data types: int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The output signal inherits its data type from the input signal.

The Initial seed parameter initializes the random number generator that the block uses to determine the permutation. The block is predictable for a given seed, but different seeds produce different permutations.


## Number of elements

The number of elements in the input vector.

## Random Interleaver

## Initial seed

The initial seed value for the random number generator.

## Pair Block Random Deinterleaver

See Also General Block Interleaver

## Rayleigh Noise Generator

## Purpose

Generate Rayleigh distributed noise

## Library

Description
Mamy
Rayleigh
Noise Generators sublibrary of Comm Sources

The Rayleigh Noise Generator block generates Rayleigh distributed noise. The Rayleigh probability density function is given by

$$
f(x)= \begin{cases}\frac{x}{\sigma^{2}} \exp \left(-\frac{x^{2}}{2 \sigma^{2}}\right) & x \geq 0 \\ 0 & x<0\end{cases}
$$

where $\sigma^{2}$ is known as the fading envelope of the Rayleigh distribution.
The block requires you to specify the Initial seed for the random number generator. If it is a constant, then the resulting noise is repeatable. The sigma parameter can be either a vector of the same length as the Initial seed, or a scalar. When sigma is a scalar, every element of the output signal shares that same value.

## Initial Seed

The Initial seed parameter initializes the random number generator that the Rayleigh Noise Generator block uses to add noise to the input signal. For best results, the Initial seed should be a prime number greater than 30. Also, if there are other blocks in a model that have an Initial seed parameter, you should choose different initial seeds for all such blocks.

You can choose seeds for the Rayleigh Noise Generator block using the Communications System Toolbox randseed function. At the MATLAB prompt, enter
randseed
This returns a random prime number greater than 30. Entering randseed again produces a different prime number. If you supply an integer argument, randseed always returns the same prime for that integer. For example, randseed (5) always returns the same answer.

## Rayleigh Noise Generator

## Attributes of Output Signal

The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the Frame-based outputs, Samples per frame, and Interpret vector parameters as 1-D parameters. See "Sources and Sinks" in the Control System Toolbox ${ }^{\text {TM }}$ documentation for more details.

The number of elements in the Initial seed parameter becomes the number of columns in a frame-based output or the number of elements in a sample-based vector output. Also, the shape (row or column) of the Initial seed parameter becomes the shape of a sample-based two-dimensional output signal.
 Box

## Sigma

Specify $\sigma$ as defined in the Rayleigh probability density function.

## Rayleigh Noise Generator

## Initial seed

The initial seed value for the random number generator.

## Sample time

The period of each sample-based vector or each row of a frame-based matrix.

## Frame-based outputs

Determines whether the output is frame-based or sample-based. This box is active only if Interpret vector parameters as 1-D is unchecked.

## Samples per frame

The number of samples in each column of a frame-based output signal. This field is active only if Frame-based outputs is checked.

## Interpret vector parameters as 1-D

If this box is checked, then the output is a one-dimensional signal. Otherwise, the output is a two-dimensional signal. This box is active only if Frame-based outputs is unchecked.

## Output data type

The output can be set to double or single data types.
See Also Multipath Rayleigh Fading Channel; raylrnd (Statistics Toolbox)
References [1] Proakis, John G., Digital Communications, Third edition, New York, McGraw Hill, 1995.

## Receiver Thermal Noise

Purpose
Library
Description

Temperature
290 K

Apply receiver thermal noise to complex baseband signal

RF Impairments
The Receiver Thermal Noise block simulates the effects of thermal noise on a complex, baseband signal. You can specify the amount of thermal noise in three ways, according to which Specification method you select:

- Noise temperature specifies the noise in degrees kelvin.
- Noise factor specifies the noise as $1+($ Noise temperature / 290).
- Noise figure specifies the noise as $10 * \log 10(1+($ Noise temperature / 290)). This is the decibel equivalent of Noise factor.

The following scatter plot shows the effect of the Receiver Thermal Noise block, with Specification method set to Noise figure and Noise figure (dB) set to 3.01 , on a signal modulated by 16-QAM.


## Receiver Thermal Noise

This plot is generated by the model described in "Illustrate RF Impairments That Distort a Signal" with the following parameter settings:

- Rectangular QAM Modulator Baseband
- Normalization method set to Average Power
- Average power (watts) set to 1e-12
- Receiver Thermal Noise
- Specification method set to Noise figure
- Noise figure (dB) set to 3.01


## Dialog

 Box

## Specification method

The method by which you specify the amount of noise. The choices are Noise temperature, Noise figure, and Noise factor.

## Receiver Thermal Noise

## Noise temperature (K)

Scalar specifying the amount of noise in degrees kelvin.

## Noise figure

Scalar specifying the amount of noise in decibels relative to a noise temperature of 290 degrees kelvin. A Noise figure setting of 0 dB indicates a noiseless system.

## Noise factor

Scalar specifying the amount of noise relative to a noise temperature of 290 degrees kelvin.

## Initial seed

The initial seed value for the random number generator that generates the noise.

See Also<br>Free Space Path Loss

## Rectangular QAM Demodulator Baseband

Purpose<br>Demodulate rectangular-QAM-modulated data<br>\section*{Library}<br>Description<br>MMW -<br>Rectangular QAM<br>AM, in Digital Baseband sublibrary of Modulation<br>The Rectangular QAM Demodulator Baseband block demodulates a signal that was modulated using quadrature amplitude modulation with a constellation on a rectangular lattice.

Note All values of power assume a nominal impedance of 1 ohm .

The signal constellation has M points, where M is the $\mathbf{M}$-ary number parameter. M must have the form $2^{\mathrm{K}}$ for some positive integer K . The block scales the signal constellation based on how you set the Normalization method parameter. For details, see the reference page for the Rectangular QAM Modulator Baseband block.

This block accepts a scalar or column vector input signal. For information about the data types each block port supports, see the "Supported Data Types" on page 2-804 table on this page.

The demodulator algorithm maps received input signal constellation values to $M$-ary integer $I$ and $Q$ symbol indices between 0 and $\sqrt{M}-1$ and then maps these demodulated symbol indices to formatted output values.

The integer symbol index computation is performed by first derotating and scaling the complex input signal constellation (possibly with noise) by a derotate factor and denormalization factor, respectively. These factors are derived from the Phase offset, Normalization method, and related parameters. These derotated and denormalized values are added to $\sqrt{\mathrm{M}}-1$ to translate them into an approximate range between 0 and $2 \times(\sqrt{\mathrm{M}}-1)$ (plus noise). The resulting values are then rescaled via a divide-by-two (or, equivalently, a right-shift by one bit for fixed-point operation) to obtain a range approximately between 0

## Rectangular QAM Demodulator Baseband

and $\sqrt{M}-1$ (plus noise) for I and Q. The noisy index values are rounded to the nearest integer and clipped, via saturation, and mapped to integer symbol values in the range [ $0 \mathrm{M}-1$ ]. Finally, based on other block parameters, the integer index is mapped to a symbol value that is formatted and cast to the selected Output data type.

The following figures contains signal flow diagrams for floating-point and fixed-point algorithm operation. The floating-point diagrams apply when the input signal data type is double or single. The fixed-point diagrams apply when the input signal is a signed fixed-point data type. Note that the diagram is simplified when Phase offset is a multiple of

$$
\frac{\pi}{2}, \text { and/or the derived denormalization factor is } 1 .
$$

## Rectangular QAM Demodulator Baseband



## Fixed Point



## Signal-Flow Diagrams with Trivial Phase Offset and Denormalization Factor Equal to 1

## Rectangular QAM Demodulator Baseband

Floating Point


Demodulator output

Fixed Point


## Signal-Flow Diagrams with Nontrivial Phase Offset and Nonunity Denormalization Factor

## Rectangular QAM Demodulator Baseband

## Dialog

Box


## M-ary number

The number of points in the signal constellation. It must have the form $2^{\mathrm{K}}$ for some positive integer K .

## Rectangular QAM Demodulator Baseband

## Normalization method

Determines how the block scales the signal constellation. Choices are Min. distance between symbols, Average Power, and Peak Power.

## Minimum distance

This parameter appears when Normalization method is set to Min. distance between symbols.

The distance between two nearest constellation points.
Average power, referenced to 1 ohm (watts)
The average power of the symbols in the constellation, referenced to 1 ohm . This field appears only when Normalization method is set to Average Power.

## Peak power, referenced to 1 ohm (watts)

The maximum power of the symbols in the constellation, referenced to 1 ohm . This field appears only when Normalization method is set to Peak Power.

Phase offset (rad)
The rotation of the signal constellation, in radians.

## Constellation ordering

Determines how the block assigns binary words to points of the signal constellation. More details are on the reference page for the Rectangular QAM Modulator Baseband block.

Selecting User-defined displays the field Constellation mapping, allowing for user-specified mapping.

## Constellation mapping

This parameter appears when User-defined is selected in the pull-down list Constellation ordering.

This is a row or column vector of size M and must have unique integer values in the range [ $0, \mathrm{M}-1$ ]. The values must be of data type double.

## Rectangular QAM Demodulator Baseband

The first element of this vector corresponds to the top-leftmost point of the constellation, with subsequent elements running down column-wise, from left to right. The last element corresponds to the bottom-rightmost point.

## Output type

Determines whether the block produces integers or binary representations of integers.

If set to Integer, the block produces integers.
If set to Bit, the block produces a group of K bits, called a binary word, for each symbol, when Decision type is set to Hard decision. If Decision type is set to Log-likelihood ratio or Approximate log-likelihood ratio, the block outputs bitwise LLR and approximate LLR, respectively.

## Decision type

This parameter appears when Bit is selected in the pull-down list Output type.

Specifies the use of hard decision, LLR, or approximate LLR during demodulation. See "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the Communications System Toolbox User's Guide for algorithm details.

## Noise variance source

This parameter appears when Approximate log-likelihood ratio or Log-likelihood ratio is selected for Decision type.

When set to Dialog, the noise variance can be specified in the Noise variance field. When set to Port, a port appears on the block through which the noise variance can be input.

## Noise variance

This parameter appears when the Noise variance source is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

## Rectangular QAM Demodulator Baseband

If you use the Simulink Coder rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:

- Inf to - Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.

## Rectangular QAM Demodulator Baseband



## Output

When the parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is of type single or double. Otherwise, the output data type will be as if this parameter is set to 'Smallest unsigned integer'.

## Rectangular QAM Demodulator Baseband

When the parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in the Hardware Implementation pane, the output data type is the ideal minimum size, i.e., ufix(1) for bit outputs, and ufix (ceil(log2(M))) for integer outputs. For all other selections, it is an unsigned integer with the smallest available word length large enough to fit the ideal minimum size, usually corresponding to the size of a char (e.g., uint8).

For integer outputs, this parameter can be set to Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, single, and double. For bit outputs, the options are Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, boolean, single, or double.

## Derotate factor

This parameter only applies when the input is fixed-point and Phase offset is not a multiple of $\frac{\pi}{2}$.

This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input.

## Denormalization factor

This parameter only applies when the input is fixed-point and the derived denormalization factor is nonunity (not equal to 1 ). This scaling factor is derived from Normalization method and other parameter values in the block dialog.

This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input. A best-precision fraction length is always used.

## Product output

This parameter only applies when the input is a fixed-point signal and there is a nonunity (not equal to 1 ) denormalized factor. It

## Rectangular QAM Demodulator Baseband

can be set to Inherit via internal rule or Specify word length, which enables a field for user input.

Setting to Inherit via internal rule computes the full-precision product word length and fraction length. Internal Rule for Product Data Types in DSP System Toolbox User's Guide describes the full-precision Product output internal rule.

Setting to Specify word length allows you to define the word length. The block computes a best-precision fraction length based on the word length specified and the pre-computed worst-case ( $\min / \max$ ) real world value Product output result. The worst-case Product output result is precomputed by multiplying the denormalized factor with the worst-case ( $\mathrm{min} / \mathrm{max}$ ) input signal range, purely based on the input signal data type.

The block uses the Rounding mode when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. For more information, see "Rounding Modes" in the DSP System Toolbox documentation or "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

## Sum

This parameter only applies when the input is a fixed-point signal. It can be set to Inherit via internal rule, Same as product output, or Specify word length, in which case a field is enabled for user input

Setting to Inherit via internal rule computes the full-precision sum word length and fraction length, based on the two inputs to the Sum in the fixed-point Hard Decision Algorithm signal flow diagram. The rule is the same as the fixed-point inherit rule of the internal Accumulator data type parameter in the Simulink Sum block.

## Rectangular QAM Demodulator Baseband

Setting to Specify word length allows you to define the word length. A best precision fraction length is computed based on the word length specified in the pre-computed maximum range necessary for the demodulated algorithm to produce accurate results. The signed fixed-point data type that has the best precision fully contains the values in the range $2 *(\sqrt{M}-1)$ for the specified word length.

Setting to Same as product output allows the Sum data type to be the same as the Product output data type (when Product output is used). If the Product output is not used, then this setting will be ignored and the Inherit via internal rule Sum setting will be used.

Supported
Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | • Single-precision floating point |
|  | • Signed fixed-point when M-ary number is an even power of 2 and: |
|  | $=$ Output type is Integer |
|  | - Output type is Bit and Decision type is Hard-decision |
| Var | • Double-precision floating point |
|  | - Single-precision floating point |
| Output | • Double-precision floating point |
|  | • Single-precision floating point |
|  | • Boolean when Output type is Bit |
|  | • 8-, 16-, and 32-bit signed integers |

## Rectangular QAM Demodulator Baseband

| Port | Supported Data Types |
| :--- | :--- |
|  | • 8-, 16-, and 32-bit unsigned integers |
|  | $\bullet$ ufix(1) in ASIC/FPGA when Output type is Bit |
|  | $\bullet u f i x\left(\log _{2} M\right)$ in ASIC/FPGA when Output type is Integer |

Pair Block Rectangular QAM Modulator Baseband
See Also General QAM Demodulator Baseband
References [1] Smith, Joel G., "Odd-Bit Quadrature Amplitude-Shift Keying," IEEE Transactions on Communications, Vol. COM-23, March 1975, 385-389.

## Rectangular QAM Modulator Baseband


#### Abstract

Purpose Modulate using rectangular quadrature amplitude modulation Library Description に—WM Rectangular QAM

AM, in Digital Baseband sublibrary of Modulation The Rectangular QAM Modulator Baseband block modulates using M -ary quadrature amplitude modulation with a constellation on a rectangular lattice. The output is a baseband representation of the modulated signal. This block accepts a scalar or column vector input signal. For information about the data types each block port supports, see "Supported Data Types" on page 2-812.


Note All values of power assume a nominal impedance of 1 ohm .

## Integer-Valued Signals and Binary-Valued Signals

When you set the Input type parameter to Integer, the block accepts integer values between 0 and $M-1 . M$ represents the $\mathbf{M}$-ary number block parameter.
When you set the Input type parameter to Bit, the block accepts binary-valued inputs that represent integers. The block collects binary-valued signals into groups of $K=\log _{2}(M)$ bits
where
$K$ represents the number of bits per symbol.
The input vector length must be an integer multiple of $K$. In this configuration, the block accepts a group of $K$ bits and maps that group onto a symbol at the block output. The block outputs one modulated symbol for each group of $K$ bits.

The Constellation ordering parameter indicates how the block assigns binary words to points of the signal constellation. Such assignments apply independently to the in-phase and quadrature components of the input:

## Rectangular QAM Modulator Baseband

- If Constellation ordering is set to Binary, the block uses a natural binary-coded constellation.
- If Constellation ordering is set to Gray and K is even, the block uses a Gray-coded constellation.
- If Constellation ordering is set to Gray and K is odd, the block codes the constellation so that pairs of nearest points differ in one or two bits. The constellation is cross-shaped, and the schematic below indicates which pairs of points differ in two bits. The schematic uses $M=128$, but suggests the general case.


> O Hollow vertical pairs of adjacent points differ by two bits

> Other pairs of adjacent points differ by one bit

For details about the Gray coding, see the reference page for the M-PSK Modulator Baseband block and the paper listed in References. Because the in-phase and quadrature components are assigned independently, the Gray and binary orderings coincide when $\mathrm{M}=4$.

## Constellation Size and Scaling

The signal constellation has M points, where M is the M -ary number parameter. M must have the form $2^{\mathrm{K}}$ for some positive integer K. The block scales the signal constellation based on how you set the Normalization method parameter. The following table lists the possible scaling conditions.

## Rectangular QAM Modulator Baseband

| Value of Normalization <br> Method Parameter | Scaling Condition |
| :--- | :--- |
| Min. distance between <br> symbols | The nearest pair of points in the <br> constellation is separated by the <br> value of the Minimum distance <br> parameter |
| Average Power | The average power of the symbols <br> in the constellation is the <br> Average power parameter |
| Peak Power | The maximum power of the <br> symbols in the constellation is the <br> Peak power parameter |

## Constellation Visualization

The Rectangular QAM Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the Communications System Toolbox User's Guide.

## Rectangular QAM Modulator Baseband



Dialog Box

## M-ary number

The number of points in the signal constellation. It must have the form $2^{\mathrm{K}}$ for some positive integer K .

## Input type

Indicates whether the input consists of integers or groups of bits.

## Constellation ordering

Determines how the block maps each symbol to a group of output bits or integer.

## Rectangular QAM Modulator Baseband

Selecting User-defined displays the field Constellation mapping, which allows for user-specified mapping.

## Constellation mapping

This parameter is a row or column vector of size M and must have unique integer values in the range [0, M-1]. The values must be of data type double.

The first element of this vector corresponds to the top-leftmost point of the constellation, with subsequent elements running down column-wise, from left to right. The last element corresponds to the bottom-rightmost point.

This field appears when User-defined is selected in the drop-down list Constellation ordering.

## Normalization method

Determines how the block scales the signal constellation. Choices are Min. distance between symbols, Average Power, and Peak Power.

## Minimum distance

The distance between two nearest constellation points. This field appears only when Normalization method is set to Min. distance between symbols.

## Average power, referenced to 1 ohm (watts)

The average power of the symbols in the constellation, referenced to 1 ohm. This field appears only when Normalization method is set to Average Power.
Peak power, referenced to 1 ohm (watts)
The maximum power of the symbols in the constellation, referenced to 1 ohm . This field appears only when Normalization method is set to Peak Power.
Phase offset (rad)
The rotation of the signal constellation, in radians.

# Rectangular QAM Modulator Baseband 

## Output data type

The output data type can be set to double, single, Fixed-point, User-defined, or Inherit via back propagation.

Setting this parameter to Fixed-point or User-defined enables fields in which you can further specify details. Setting this parameter to Inherit via back propagation, sets the output data type and scaling to match the following block.

## Output word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed-point for the Output data type parameter.

## User-defined data type

Specify any signed built-in or signed fixed-point data type. You can specify fixed-point data types using the sfix, sint, sfrac, and fixdt functions from Fixed-Point Designer software. This parameter is only visible when you select User-defined for the Output data type parameter.

## Set output fraction length to

Specify the scaling of the fixed-point output by either of the following methods:

- Choose Best precision to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose User-defined to specify the output scaling in the Output fraction length parameter.

This parameter is only visible when you select Fixed-point for the Output data type parameter or when you select User-defined and the specified output data type is a fixed-point data type.

## Output fraction length

For fixed-point output data types, specify the number of fractional bits, or bits to the right of the binary point. This parameter is only visible when you select Fixed-point or User-defined for

## Rectangular QAM Modulator Baseband

the Output data type parameter and User-defined for the Set output fraction length to parameter.

## Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | - Single-precision floating point |
|  | - Boolean when Input type is Bit |
|  | • 8-, 16-, 32-bit signed integers |
|  | - 8-, 16-, 32-bit unsigned integers |
|  | - ufix $\left(\log _{2} M\right)$ when Input type is Integer |
| Output | • Double-precision floating point |
|  | - Single-precision floating point |
|  | - Signed fixed-point |


| Pair Block | Rectangular QAM Demodulator Baseband |
| :--- | :--- |
| See Also | General QAM Modulator Baseband |
| References | [1] Smith, Joel G., "Odd-Bit Quadrature Amplitude-Shift Keying," IEEE <br> Transactions on Communications, Vol. COM-23, March 1975, 385-389. |

## Rectangular QAM TCM Decoder

## Purpose

Decode trellis-coded modulation data, modulated using QAM method

## Library

Description

Rectangular QAM TCM

TCM, in Digital Baseband sublibrary of Modulation
The Rectangular QAM TCM Decoder block uses the Viterbi algorithm to decode a trellis-coded modulation (TCM) signal that was previously modulated using a QAM signal constellation.

The M-ary number parameter represents the number of points in the signal constellation, which also equals the number of possible output symbols from the convolutional encoder. (That is, $\log _{2}$ (M-ary number) is the number of output bit streams from the convolutional encoder.)

The Trellis structure and M-ary number parameters in this block should match those in the Rectangular QAM TCM Encoder block, to ensure proper decoding.

## Input and Output Signals

This block accepts a column vector input signal containing complex numbers. For information about the data types each block port supports, see "Supported Data Types" on page 2-816.

If the convolutional encoder described by the trellis structure represents a rate $k / n$ code, then the Rectangular QAM TCM Decoder block's output is a binary column vector with a length of $k$ times the vector length of the input signal.

## Operation Modes

The block has three possible methods for transitioning between successive frames. The Operation mode parameter controls which method the block uses. This parameter also affects the range of possible values for the Traceback depth parameter, D.

- In Continuous mode, the block initializes all state metrics to zero at the beginning of the simulation, waits until it accumulates D symbols, and then uses a sequence of D symbols to compute each of the traceback paths. D can be any positive integer. At the end of


## Rectangular QAM TCM Decoder

each frame, the block saves its internal state metric for use with the next frame.

If you select Enable the reset input, the block displays another input port, labeled Rst. This port receives an integer scalar signal. Whenever the value at the Rst port is nonzero, the block resets all state metrics to zero and sets the traceback memory to zero.

- In Truncated mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In Terminated mode, the block treats each frame independently. The traceback path always starts at the all-zeros state. D must be less than or equal to the vector length of the input. If you know that each frame of data typically ends at the all-zeros state, then this mode is an appropriate choice.


## Decoding Delay

If you set Operation mode to Continuous, then this block introduces a decoding delay equal to Traceback depth*k bits, for a rate k/n convolutional code. The decoding delay is the number of zeros that precede the first decoded bit in the output.
The block incurs no delay for other values of Operation mode.

## Rectangular QAM TCM Decoder



Dialog Box


## Trellis structure

MATLAB structure that contains the trellis description of the convolutional encoder.

## M-ary number

The number of points in the signal constellation.

## Traceback depth

The number of trellis branches (equivalently, the number of symbols) the block uses in the Viterbi algorithm to construct each traceback path.

## Rectangular QAM TCM Decoder

## Operation mode

The operation mode of the Viterbi decoder. Choices are Continuous, Truncated, and Terminated.

## Enable the reset input port

When you select this check box, the block has a second input port labeled Rst. Providing a nonzero input value to this port causes the block to set its internal memory to the initial state before processing the input data. This option appears only if you set Operation mode to Continuous.

## Output data type

Select the data type for the block output signal as boolean or single. By default, the block sets this to double.

Supported Data Types

| Port | Supported Data Types |
| :--- | :--- |
| Input | • Double-precision floating point |
|  | • Single-precision floating point |

Pair Block Rectangular QAM TCM Encoder<br>See Also<br>General TCM Decoder, poly2trellis<br>References<br>[1] Biglieri, E., D. Divsalar, P. J. McLane and M. K. Simon, Introduction to Trellis-Coded Modulation with Applications, New York, Macmillan, 1991.<br>[2] Proakis, John G., Digital Communications, Fourth edition, New York, McGraw-Hill, 2001.

## Rectangular QAM TCM Encoder

## Purpose

Convolutionally encode binary data and modulate using QAM method

## Library

Description
TCM, in Digital Baseband sublibrary of Modulation
The Rectangular QAM TCM Encoder block implements trellis-coded modulation (TCM) by convolutionally encoding the binary input signal and mapping the result to a QAM signal constellation.

The M-ary number parameter is the number of points in the signal constellation, which also equals the number of possible output symbols from the convolutional encoder. (That is, $\log _{2}(\mathbf{M}$-ary number) is equal to n for a rate $\mathrm{k} / \mathrm{n}$ convolutional code.)

## Input Signals and Output Signals

If the convolutional encoder described by the trellis structure represents a rate $k / n$ code, then the Rectangular QAM TCM Encoder block's input must be a binary column vector with a length of $L^{*} k$ for some positive integer $L$.

The output from the Rectangular QAM TCM Encoder block is a complex column vector of length $L$.

## Specifying the Encoder

To define the convolutional encoder, use the Trellis structure parameter. This parameter is a MATLAB structure whose format is described in "Trellis Description of a Convolutional Code" in the Communications System Toolbox documentation. You can use this parameter field in two ways:

- If you want to specify the encoder using its constraint length, generator polynomials, and possibly feedback connection polynomials, then use a poly2trellis command within the Trellis structure field. For example, to use an encoder with a constraint length of 7 , code generator polynomials of 171 and 133 (in octal numbers), and a feedback connection of 171 (in octal), set the Trellis structure parameter to


## Rectangular QAM TCM Encoder

- If you have a variable in the MATLAB workspace that contains the trellis structure, then enter its name as the Trellis structure parameter. This way is faster because it causes Simulink to spend less time updating the diagram at the beginning of each simulation, compared to the usage in the previous bulleted item.

The encoder registers begin in the all-zeros state. You can configure the encoder so that it resets its registers to the all-zeros state during the course of the simulation. To do this, set the Operation mode to Reset on nonzero input via port. The block then opens a second input port, labeled Rst. The signal at the Rst port is a scalar signal. When it is nonzero, the encoder resets before processing the data at the first input port.

## Signal Constellations

The trellis-coded modulation technique partitions the constellation into subsets called cosets, so as to maximize the minimum distance between pairs of points in each coset. This block internally forms a valid partition based on the value you choose for the $\mathbf{M}$-ary number parameter.

The figures below show the labeled set-partitioned signal constellations that the block uses when M-ary number is 16, 32, and 64. For constellations of other sizes, see Biglieri, E., D. Divsalar, P. J. McLane and M. K. Simon, Introduction to Trellis-Coded Modulation with Applications, New York, Macmillan, 1991.

## Rectangular QAM TCM Encoder



## Rectangular QAM TCM Encoder



## Rectangular QAM TCM Encoder

Signal Constellation for 64-QAM

## Coding Gains

Coding gains of 3 to 6 decibels, relative to the uncoded case can be achieved in the presence of AWGN with multiphase trellis codes. For more information, see Biglieri, E., D. Divsalar, P. J. McLane and M. K. Simon, Introduction to Trellis-Coded Modulation with Applications, New York, Macmillan, 1991.

## Rectangular QAM TCM Encoder

Dialog Box


## Trellis structure

MATLAB structure that contains the trellis description of the convolutional encoder.

## Operation mode

In Continuous mode (default setting), the block retains the encoder states at the end of each frame, for use with the next frame.

In Truncated (reset every frame) mode, the block treats each frame independently. I.e., the encoder states are reset to all-zeros state at the start of each frame.

In Terminate trellis by appending bits mode, the block treats each frame independently. For each input frame, extra bits are used to set the encoder states to all-zeros state at the end of the frame. The output length is given by

## Rectangular QAM TCM Encoder

$y=n \cdot(x+s) / k$, where $x$ is the number of input bits, and $s=$ constraint length -1 (or, in the case of multiple constraint lengths, $s=$ sum(ConstraintLength(i)-1)). The block supports this mode for column vector input signals.

In Reset on nonzero input via port mode, the block has an additional input port, labeled Rst. When the Rst input is nonzero, the encoder resets to the all-zeros state.

## M-ary number

The number of points in the signal constellation.

## Output data type

The output type of the block can be specified as a single or double. By default, the block sets this to double.

## Pair Block Rectangular QAM TCM Decoder

See Also General TCM Encoder, poly2trellis
References [1] Biglieri, E., D. Divsalar, P. J. McLane and M. K. Simon, Introduction to Trellis-Coded Modulation with Applications, New York, Macmillan, 1991.
[2] Proakis, John G., Digital Communications, Fourth edition, New York, McGraw-Hill, 2001
[3] Ungerboeck, G., "Channel Coding with Multilevel/Phase Signals", IEEE Trans. on Information Theory, Vol IT28, Jan. 1982, pp. 55-67.

## Repeat

Library
Description

Purpose Resample input at higher rate by repeating values

Signal Operations
The Filter block is a DSP System Toolbox block. For more information, see the Repeat block reference page in the DSP System Toolbox documentation.

## Rician Noise Generator

## Purpose

Generate Rician distributed noise

## Library

Description
Mrnanan
Rician
Noise Generators sublibrary of Comm Sources The Rician probability density function is given by

The Rician Noise Generator block generates Rician distributed noise.

$$
f(x)=\left\{\begin{array}{cc}
\frac{x}{\sigma^{2}} I_{0}\left(\frac{m x}{\sigma^{2}}\right) \exp \left(-\frac{x^{2}+m^{2}}{2 \sigma^{2}}\right) & x \geq 0 \\
0 & x<0
\end{array}\right.
$$

where:

- $\sigma$ is the standard deviation of the Gaussian distribution that underlies the Rician distribution noise
- $\mathrm{m}^{2}=\mathrm{m}_{\mathrm{I}}{ }^{2}+\mathrm{m}_{\mathrm{Q}}{ }^{2}$, where $m_{\mathrm{I}}$ and $m_{\mathrm{Q}}$ are the mean values of two independent Gaussian components
- $I_{0}$ is the modified 0th-order Bessel function of the first kind given by

$$
I_{0}(y)=\frac{1}{2 \pi} \int_{-\pi}^{\pi} e^{y \cos t} d t
$$

Note that $m$ and $\sigma$ are not the mean value and standard deviation for the Rician noise.

You must specify the Initial seed for the random number generator. When it is a constant, the resulting noise is repeatable. The vector length of the Initial seed parameter should equal the number of columns in a frame-based output or the number of elements in a sample-based output. The set of numerical parameters above the Initial seed parameter in the dialog box can consist of vectors having the same length as the Initial seed, or scalars.

## Rician Noise Generator

## Initial Seed

The scalar Initial seed parameter initializes the random number generator that the block uses to generate its Rician-distributed complex random process. For best results, the Initial seed should be a prime number greater than 30. Also, if there are other blocks in a model that have an Initial seed parameter, you should choose different initial seeds for all such blocks.

You can choose seeds for the Rician Noise Generator block using the Communications System Toolbox randseed function. At the MATLAB prompt, enter

```
randseed
```

This returns a random prime number greater than 30. Entering randseed again produces a different prime number. If you supply an integer argument, randseed always returns the same prime for that integer. For example, randseed (5) always returns the same answer.

## Attributes of Output Signal

The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the Frame-based outputs, Samples per frame, and Interpret vector parameters as 1-D parameters. See "Sources and Sinks" in Communications System Toolbox User's Guide for more details.

The number of elements in the Initial seed and Sigma parameters becomes the number of columns in a frame-based output or the number of elements in a sample-based vector output. Also, the shape (row or column) of the Initial seed and Sigma parameters becomes the shape of a sample-based two-dimensional output signal.

## Rician Noise Generator



## Rician Noise Generator

## Sigma

The variable $\sigma$ in the Rician probability density function.

## Initial seed

The initial seed value for the random number generator.

## Sample time

The period of each sample-based vector or each row of a frame-based matrix.

## Frame-based outputs

Determines whether the output is frame-based or sample-based. This box is active only if Interpret vector parameters as 1-D is unchecked.

## Samples per frame

The number of samples in each column of a frame-based output signal. This field is active only if Frame-based outputs is checked.

## Interpret vector parameters as 1-D

If this box is checked, then the output is a one-dimensional signal. Otherwise, the output is a two-dimensional signal. This box is active only if Frame-based outputs is unchecked.

## Output data type

The output can be set to double or single data types.

See Also Multipath Rician Fading Channel<br>References [1] Proakis, John G., Digital Communications, Third edition, New York, McGraw Hill, 1995.

## RLS Decision Feedback Equalizer

## Purpose

## Library

Description


Equalize using decision feedback equalizer that updates weights with RLS algorithm

## Equalizers

The RLS Decision Feedback Equalizer block uses a decision feedback equalizer and the RLS algorithm to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the RLS algorithm to update the weights, once per symbol. When you set the Number of samples per symbol parameter to 1 , the block implements a symbol-spaced equalizer and updates the filter weights once for each symbol. When you set the Number of samples per symbol parameter to a value greater than 1 , the weights are updated once every $N^{\text {th }}$ sample, for a fractionally spaced equalizer.

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.

Set the Reference tap parameter so it is greater than zero and less than the value for the Number of forward taps parameter.

The port labeled Equalized outputs the result of the equalization process.

You can configure the block to have one or more of these extra ports:

- Mode input, as described in "Reference Signal and Operation Modes" in Communications System ToolboxUser's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.
- Weights output, as described in "Adaptive Algorithms" in Communications System ToolboxUser's Guide.


## RLS Decision Feedback Equalizer

## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Adaptive Algorithms" in Communications System Toolbox User's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to

## $1+($ Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

## RLS Decision Feedback Equalizer



Dialog Box

## Number of forward taps

The number of taps in the forward filter of the decision feedback equalizer.

## RLS Decision Feedback Equalizer

## Number of feedback taps

The number of taps in the feedback filter of the decision feedback equalizer.

## Number of samples per symbol

The number of input samples for each symbol.

## Signal constellation

A vector of complex numbers that specifies the constellation for the modulation.

## Reference tap

A positive integer less than or equal to the number of forward taps in the equalizer.

## Forgetting factor

The forgetting factor of the RLS algorithm, a number between 0 and 1.

## Inverse correlation matrix

The initial value for the inverse correlation matrix. The matrix must be N -by- N , where N is the total number of forward and feedback taps.

## Initial weights

A vector that concatenates the initial weights for the forward and feedback taps.

## Mode input port

When you select this check box, the block has an input port that allows you to toggle between training and decision-directed mode. For training, the mode input must be 1, and for decision directed, the mode must be 0 . For every frame in which the mode input is 1 or not present, the equalizer trains at the beginning of the frame for the length of the desired signal.

## Output error

When you select this check box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## RLS Decision Feedback Equalizer

## Output weights

When you select this check box, the block outputs the current forward and feedback weights, concatenated into one vector.

References [1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, Wiley, 1998.<br>[2] Haykin, Simon, Adaptive Filter Theory, Third Ed., Upper Saddle River, N.J., Prentice-Hall, 1996.<br>[3] Kurzweil, Jack, An Introduction to Digital Communications, New York, Wiley, 2000.<br>[4] Proakis, John G., Digital Communications, Fourth Ed., New York, McGraw-Hill, 2001.

See Also<br>RLS Linear Equalizer, LMS Decision Feedback Equalizer, CMA Equalizer

## RLS Linear Equalizer

Description


| Purpose | Equalize using linear equalizer that updates weights using RLS <br> algorithm |
| :--- | :--- |
| Library | Equalizers |

Equalize using linear equalizer that updates weights using RLS algorithm

Equalizers
The RLS Linear Equalizer block uses a linear equalizer and the RLS algorithm to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the RLS algorithm to update the weights, once per symbol. When you set the Number of samples per symbol parameter to 1, then the block implements a symbol-spaced (i.e. T-spaced) equalizer and updates the filter weights once for each symbol. When you set the Number of samples per symbol parameter to a value greater than 1 , the block updates the weights once every $N^{\text {th }}$ sample, for a fractionally spaced (i.e. T/N-spaced) equalizer.

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.

Set the Reference tap parameter so it is greater than zero and less than the value for the Number of taps parameter.

The port labeled Equalized outputs the result of the equalization process.
You can configure the block to have one or more of these extra ports:

- Mode input, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.


## RLS Linear Equalizer

- Weights output, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.


## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Adaptive Algorithms" in Communications System Toolbox User's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to
$1+($ Reference tap-1)/(Number of samples per symbol)
Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## RLS Linear Equalizer



## Box

## Number of taps

The number of taps in the filter of the linear equalizer.

## Number of samples per symbol

The number of input samples for each symbol.

## Signal constellation

A vector of complex numbers that specifies the constellation for the modulation.

## Reference tap

A positive integer less than or equal to the number of taps in the equalizer.

## Forgetting factor

The forgetting factor of the RLS algorithm, a number between 0 and 1.

## Inverse correlation matrix

The initial value for the inverse correlation matrix. The matrix must be N -by- N , where N is the number of taps.

## Initial weights

A vector that lists the initial weights for the taps.

## Mode input port

When you select this check box, the block has an input port that allows you to toggle between training and decision-directed mode. For training, the mode input must be 1, and for decision directed, the mode must be 0 . For every frame in which the mode input is 1 or not present, the equalizer trains at the beginning of the frame for the length of the desired signal.

## Output error

When you select this check box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## Output weights

When you select this check box, the block outputs the current weights.

Examples See the Adaptive Equalization example.
References [1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, Wiley, 1998.
[2] Haykin, Simon, Adaptive Filter Theory, Third Ed., Upper Saddle River, N.J., Prentice-Hall, 1996.
[3] Kurzweil, Jack, An Introduction to Digital Communications, New York, Wiley, 2000.
[4] Proakis, John G., Digital Communications, Fourth Ed., New York, McGraw-Hill, 2001.

See Also RLS Decision Feedback Equalizer, LMS Linear Equalizer, CMA Equalizer

## Purpose <br> Library <br> Description

Scramble input signal

Sequence Operations
The Scrambler block scrambles a scalar or column vector input signal. If you set the Calculation base parameter to $N$, then the input values must be integers between 0 and $\mathrm{N}-1$.

One purpose of scrambling is to reduce the length of strings of 0s or 1 s in a transmitted signal, since a long string of 0 s or 1 s may cause transmission synchronization problems. Below is a schematic of the scrambler. All adders perform addition modulo $N$.


At each time step, the input causes the contents of the registers to shift sequentially. The he Scramble polynomial parameter defines if each switch in the scrambler is on or off. Specify the polynomial by listing its coefficients in order of ascending powers of $z^{-1}$, where $p\left(z^{-1}\right)=1+p_{1} z^{-1}+$ $p_{2} z^{-2}+\ldots$, or by listing the powers of $z$ that appear in the polynomial with a coefficient of 1. For example $p=\left[\begin{array}{llllll}1 & 0 & 0 & 0 & 0 & 1\end{array} 1\right]$ and $p=\left[\begin{array}{lll}0 & -6 & -8\end{array}\right]$ both represent the polynomial $p\left(z^{-1}\right)=1+z^{-6}+z^{-8}$.
The Initial states parameter lists the states of the scrambler's registers when the simulation starts. The elements of this vector must be integers between 0 and $N-1$. The vector length of this parameter must equal the order of the scramble polynomial. (If the Scramble polynomial parameter is a vector that lists the coefficients in order, then the order of the scramble polynomial is one less than the vector length.)

## Scrambler

Function Block Parameters: Scrambler ..... x

    Scrambler (mask) (link)
    
    Scramble the input data using a linear feedback shift register whose configuration
    
    you specify using the 'Scramble polynomial' parameter.
    
    This block accepts a scalar or column vector input signal.
    
    Parameters
    
    Calculation base:
    
    4
    
    Scramble polynomial:
    
    [1:1101]
    
    Initial states:
    
    [012 123 ]
    

Dialog Box

## Calculation base

The calculation base N. The input and output of this block are integers in the range [ $0, \mathrm{~N}-1$ ].

## Scramble polynomial

A polynomial that defines the connections in the scrambler.

## Initial states

The states of the scrambler's registers when the simulation starts.
Pair Block Descrambler
See Also PN Sequence Generator

## Sign LMS Decision Feedback Equalizer

## Purpose

## Library

Description


Equalize using decision feedback equalizer that updates weights with signed LMS algorithm

## Equalizers

The Sign LMS Decision Feedback Equalizer block uses a decision feedback equalizer and an algorithm from the family of signed LMS algorithms to equalize a linearly modulated baseband signal through a dispersive channel.

The supported algorithms, corresponding to the Update algorithm parameter, are

- Sign LMS
- Sign Regressor LMS
- Sign Sign LMS

During the simulation, the block uses the particular signed LMS algorithm to update the weights, once per symbol. If the Number of samples per symbol parameter is 1 , then the block implements a symbol-spaced equalizer; otherwise, the block implements a fractionally spaced equalizer.

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.
Set the Reference tap parameter so it is greater than zero and less than the value for the Number of forward taps parameter.

The port labeled Equalized outputs the result of the equalization process.

You can configure the block to have one or more of these extra ports:

## Sign LMS Decision Feedback Equalizer

- Mode input, as described in "Reference Signal and Operation Modes" in Communications System Toolbox User's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.
- Weights output, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.


## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Adaptive Algorithms" in Communications System ToolboxUser's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to

## $1+($ Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

## Sign LMS Decision Feedback Equalizer

| T/ Function Block Parameters: Sign LMS Decision Feedback Equalizer X |  |  |  |  |
| :---: | :---: | :---: | :---: | :---: |
| Sign LMS Decision Feedback Equalizer (mask) (link) <br> Equalize a linearly modulated signal through a dispersive channel using a decision feedback equalizer and the signed least mean squares (LMS) algorithm. |  |  |  |  |
|  |  |  |  |  |
| Parameters <br> Update algorithm Sign LMS <br> Number of forward taps: |  |  |  |  |
|  |  |  |  |  |
|  |  |  |  |  |
| 4 |  |  |  |  |
| Number of feedback taps: |  |  |  |  |
| 4 |  |  |  |  |
| Number of samples per symbol: |  |  |  |  |
| 1 |  |  |  |  |
| Signal constellation: |  |  |  |  |
| $[-3+3 \mathrm{j}-3+\mathrm{j}-3 \mathrm{j}-3-3 \mathrm{j}-1+3 \mathrm{j}-1+\mathrm{j}-1-\mathrm{j}-1-3 \mathrm{j} 1+3 \mathrm{j} 1+\mathrm{j} 1-\mathrm{j} 1-3 \mathrm{j} 3+3 \mathrm{j} 3+\mathrm{j} 3-\mathrm{j} 3-3 \mathrm{j}]$ |  |  |  |  |
| Reference tap: |  |  |  |  |
| 2 |  |  |  |  |
| Step size: |  |  |  |  |
| 0.01 |  |  |  |  |
| Leakage factor: |  |  |  |  |
| 1 |  |  |  |  |
| Initial weights: |  |  |  |  |
| 0 |  |  |  |  |
| $\sqrt{ }$ Mode input port <br> V Output error <br> $\sqrt{ }$ Output weights |  |  |  |  |
| OK Cancel Help Apply |  |  |  |  |

Dialog
Box

## Update algorithm

The specific type of signed LMS algorithm that the block uses to update the equalizer weights.

## Sign LMS Decision Feedback Equalizer

## Number of forward taps

The number of taps in the forward filter of the decision feedback equalizer.

## Number of feedback taps

The number of taps in the feedback filter of the decision feedback equalizer.

## Number of samples per symbol

The number of input samples for each symbol.

- When you set this parameter to 1 , the filter weights are updated once for each symbol, for a symbol spaced (i.e. T-spaced) equalizer.
- When you set this parameter to a value greater than 1 , the weights are updated once every $N^{\text {th }}$ sample, for a T/N-spaced equalizer.


## Signal constellation

A vector of complex numbers that specifies the constellation for the modulation.

## Reference tap

A positive integer less than or equal to the number of forward taps in the equalizer.

## Step size

The step size of the signed LMS algorithm.

## Leakage factor

The leakage factor of the signed LMS algorithm, a number between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

## Initial weights

A vector that concatenates the initial weights for the forward and feedback taps.

## Sign LMS Decision Feedback Equalizer

## Mode input port

When you select this check box, the block has an input port that allows you to toggle between training and decision-directed mode. For training, the mode input must be 1 , for decision directed, the mode should be 0 . For every frame in which the mode input is 1 or not present, the equalizer trains at the beginning of the frame for the length of the desired signal.

## Output error

When you select this check box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## Output weights

When you select this check box, the block outputs the current forward and feedback weights, concatenated into one vector.

References [1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, Wiley, 1998.<br>[2] Kurzweil, Jack, An Introduction to Digital Communications, New York, Wiley, 2000.

See Also
Sign LMS Linear Equalizer, LMS Decision Feedback Equalizer

## Sign LMS Linear Equalizer

| Purpose | Equalize using linear equalizer that updates weights with signed LMS <br> algorithm |
| :--- | :--- |
| Library | Equalizers |
| Description | The Sign LMS Linear Equalizer block uses a linear equalizer and <br> an algorithm from the family of signed LMS algorithms to equalize |
| a linearly modulated baseband signal through a dispersive channel. |  |
| The supported algorithms, corresponding to the Update algorithm |  |
| parameter, are |  |

During the simulation, the block uses the particular signed LMS algorithm to update the weights, once per symbol. When you set the Number of samples per symbol parameter to 1, then the block implements a symbol-spaced equalizer and updates the filter weights once for each symbol. When you set the Number of samples per symbol parameter to a value greater than 1 , the weights are updated once every $N^{\text {th }}$ sample, for a $T / N$-spaced equalizer.

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.

Set the Reference tap parameter so it is greater than zero and less than the value for the Number of taps parameter.

The Equalized port outputs the result of the equalization process.
You can configure the block to have one or more of these extra ports:

## Sign LMS Linear Equalizer

- Mode input, as described in "Adaptive Algorithms" in Communications System ToolboxUser's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.
- Weights output, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.


## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Adaptive Algorithms" in Communications System Toolbox User's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to

## $1+($ Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## Sign LMS Linear Equalizer



Dialog

## Box

## Update algorithm

The specific type of signed LMS algorithm that the block uses to update the equalizer weights.

## Sign LMS Linear Equalizer

## Number of taps

The number of taps in the filter of the linear equalizer.

## Number of samples per symbol

The number of input samples for each symbol.

## Signal constellation

A vector of complex numbers that specifies the constellation for the modulation.

## Reference tap

A positive integer less than or equal to the number of taps in the equalizer.

## Step size

The step size of the signed LMS algorithm.

## Leakage factor

The leakage factor of the signed LMS algorithm, a number between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

## Initial weights

A vector that lists the initial weights for the taps.

## Mode input port

When you select this check box, the block has an input port that allows you to toggle between training and decision-directed mode. For training, the mode input must be 1 , for decision directed, the mode should be 0 . For every frame in which the mode input is 1 or not present, the equalizer trains at the beginning of the frame for the length of the desired signal.

## Output error

When you select this check box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## Sign LMS Linear Equalizer

## Output weights

When you select this check box, the block outputs the current weights.

## Examples See the Adaptive Equalization example.

References [1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, Wiley, 1998.
[2] Kurzweil, Jack, An Introduction to Digital Communications, New York, Wiley, 2000.

See Also<br>Sign LMS Decision Feedback Equalizer, LMS Linear Equalizer

## Squaring Timing Recovery

## Purpose

Recover symbol timing phase using squaring method

## Library

Description
Timing Phase Recovery sublibrary of Synchronization
The Squaring Timing Recovery block recovers the symbol timing phase of the input signal using a squaring method. This feedforward, non-data-aided method is similar to the conventional squaring loop. This block is suitable for systems that use linear baseband modulation types such as pulse amplitude modulation (PAM), phase shift keying (PSK) modulation, and quadrature amplitude modulation (QAM).

Typically, the input to this block is the output of a receive filter that is matched to the transmitting pulse shape. This block accepts a column vector input signal of type double or single. The input represents Symbols per frame symbols, using Samples per symbol samples for each symbol. Typically, Symbols per frame is approximately 100, Samples per symbol is at least 4, and the input signal is shaped using a raised cosine filter.

Note The block assumes that the phase offset is constant for all symbols in the entire input frame. If necessary, use the Buffer block to reorganize your data into frames over which the phase offset can be assumed constant. If the assumption of constant phase offset is valid, then a larger frame length yields a more accurate phase offset estimate.

The block estimates the phase offset for the symbols in each input frame and applies the estimate uniformly over the input frame. The block outputs signals containing one sample per symbol. Therefore, the size of each output equals the Symbols per frame parameter value. The outputs are as follows:

- The output port labeled Sym gives the result of applying the phase estimate uniformly over the input frame. This output is the signal value for each symbol, which can be used for decision purposes.


## Squaring Timing Recovery

- The output port labeled Ph gives the phase estimate for each symbol in the input frame. All elements in this output are the same nonnegative real number less than the Samples per symbol parameter value. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal.



## Symbols per frame

The number of symbols in each frame of the input signal.

## Samples per symbol

The number of input samples that represent each symbol. This must be greater than 1 .

## Squaring Timing Recovery

## Algorithm

## Examples

## References

See Also

This block uses a timing estimator that returns

$$
-\frac{1}{2 \pi} \arg \left(\sum_{m=0}^{\mathrm{LN}-1}\left|x_{m+1}\right|^{2} \exp (-\mathrm{j} 2 \pi m / \mathrm{N})\right)
$$

as the normalized phase between $-1 / 2$ and $1 / 2$, where $x$ is the input vector, L is the Symbols per frame parameter and N is the Samples per symbol parameter.

For more information about the role that the timing estimator plays in this block's algorithm, see "Feedforward Method for Timing Phase Recovery" in Communications System Toolbox User's Guide.

See "Squaring Timing Phase Recovery Example" in Communications System Toolbox User's Guide.
[1] Oerder, M. and H. Myer, "Digital Filter and Square Timing Recovery," IEEE Transactions on Communications, Vol. COM-36, No. 5, May 1988, pp. 605-612.
[2] Mengali, Umberto and Aldo N. D'Andrea, Synchronization Techniques for Digital Receivers, New York, Plenum Press, 1997.
[3] Meyr, Heinrich, Marc Moeneclaey, and Stefan A. Fechtel, Digital Communication Receivers, Vol 2, New York, Wiley, 1998.

Gardner Timing Recovery, Early-Late Gate Timing Recovery

## SSB AM Demodulator Passband

Purpose Demodulate SSB-AM-modulated data

Library
Description
WMWN
SSB AM

Analog Passband Modulation, in Modulation
The SSB AM Demodulator Passband block demodulates a signal that was modulated using single-sideband amplitude modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real scalar signals.
This block works only with real inputs of type double. This block does not work inside a triggered subsystem.

## SSB AM Demodulator Passband

Dialog Box

| Finnction Block Parameters: SSB AM De | Pas | X |
| :---: | :---: | :---: |
| SSB AM Demodulator Passband (mask) (link) |  |  |
| Demodulate a single-sideband amplitude modulated signal. |  |  |
| The input signal must be a scalar. |  |  |
| ParametersCarrier frequency $(\mathrm{Hz})$ : |  |  |
|  |  |  |
| 300 |  |  |
| Initial phase (rad): |  |  |
| 0 |  |  |
| Lowpass filter design method: Butterworth |  | $\checkmark$ |
| Filter order: |  |  |
| 4 |  |  |
| Cutoff frequency ( Hz ): |  |  |
| 300 |  |  |
| OK Cancel | Help | Apply |

## Carrier frequency ( Hz )

The carrier frequency in the corresponding SSB AM Modulator Passband block.

## Initial phase (rad)

The phase offset, $\theta$, of the modulated signal.
Lowpass filter design method
The method used to generate the filter. Available methods are Butterworth, Chebyshev type I, Chebyshev type II, and Elliptic.

## Filter order

The order of the lowpass digital filter specified in the Lowpass filter design method field .

## SSB AM Demodulator Passband

## Cutoff frequency

The cutoff frequency of the lowpass digital filter specified in the Lowpass filter design method field in Hertz.

## Passband ripple

Applies to Chebyshev type I and Elliptic filters only. This is peak-to-peak ripple in the passband in dB .

## Stopband ripple

Applies to Chebyshev type II and Elliptic filters only. This is the peak-to-peak ripple in the stopband in dB .

Pair Block<br>SSB AM Modulator Passband<br>See Also DSB AM Demodulator Passband, DSBSC AM Demodulator Passband

## SSB AM Modulator Passband

## Purpose

Modulate using single-sideband amplitude modulation

## Library

Description

SSB AM
Analog Passband Modulation, in Modulation
The SSB AM Modulator Passband block modulates using single-sideband amplitude modulation with a Hilbert transform filter. The output is a passband representation of the modulated signal. Both the input and output signals are real scalar signals.

SSB AM Modulator Passband transmits either the lower or upper sideband signal, but not both. To control which sideband it transmits, use the Sideband to modulate parameter.

If the input is $u(t)$ as a function of time $t$, then the output is

$$
u(t) \cos \left(f_{c} t+\theta\right) \mp u(t) \sin \left(f_{c} t+\theta\right)
$$

where:

- $f_{\mathrm{c}}$ is the Carrier frequency parameter.
- $\theta$ is the Initial phase parameter.
- $\hat{u}(t)$ is the Hilbert transform of the input $u(t)$.
- The minus sign indicates the upper sideband and the plus sign indicates the lower sideband.


## Hilbert Tranform Filter

This block uses the Analytic Signal block from the DSP System Toolbox Transforms block library.

The Analytic Signal block computes the complex analytic signal corresponding to each channel of the real M-by-N input, u

$$
y=u+j \mathrm{H}\{u\}
$$

where $j=\sqrt{-1}$ and $\mathrm{H}\}$ denotes the Hilbert transform. The real part of the output in each channel is a replica of the real input in that

## SSB AM Modulator Passband

channel; the imaginary part is the Hilbert transform of the input. In the frequency domain, the analytic signal retains the positive frequency content of the original signal while zeroing-out negative frequencies and doubling the DC component.

The block computes the Hilbert transform using an equiripple FIR filter with the order specified by the Filter order parameter, n. The linear phase filter is designed using the Remez exchange algorithm, and imposes a delay of $\mathrm{n} / 2$ on the input samples.

For best results, use a carrier frequency which is estimated to be larger than $10 \%$ of your input signal's sample rate. This is due to the implementation of the Hilbert transform by means of a filter.

In the following example, we sample a 10 Hz input signal at 8000 samples per second. We then designate a Hilbert Transform filter of order 100. Below is the response of the Hilbert Transform filter as returned by fvtool.

## SSB AM Modulator Passband



Note the bandwidth of the filter's magnitude response. By choosing a carrier frequency larger than $10 \%$ (but less than $90 \%$ ) of the input signal's sample time ( 8000 samples per second, in this example) or equivalently, a carrier frequency larger than 400 Hz , we ensure that the Hilbert Transform Filter will be operating in the flat section of the filter's magnitude response (shown in blue), and that our modulated signal will have the desired magnitude and form.
Typically, an appropriate Carrier frequency value is much higher than the highest frequency of the input signal. By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the Carrier frequency parameter.

## SSB AM Modulator Passband

This block works only with real inputs of type double. This block does not work inside a triggered subsystem.

Dialog Box


## Carrier frequency ( Hz )

The frequency of the carrier.

## Initial phase (rad)

The phase offset, $\theta$, of the modulated signal.

## Sideband to modulate

This parameter specifies whether to transmit the upper or lower sideband.

## Hilbert Transform filter order

The length of the FIR filter used to compute the Hilbert transform.

Pair Block<br>SSB AM Demodulator Passband

## SSB AM Modulator Passband

See Also<br>DSB AM Modulator Passband, DSBSC AM Modulator Passband; hilbiir (Communications Toolbox)<br>References<br>[1] Peebles, Peyton Z, Jr. Communication System Principles. Reading, Mass.: Addison-Wesley, 1976.

## Turbo Decoder

Purpose Decode input signal using parallel concatenated decoding scheme
Library
Convolutional sublibrary of Error Detection and Correction
Description


The Turbo Decoder block decodes the input signal using a parallel concatenated decoding scheme. The iterative decoding scheme uses the a posteriori probability (APP) decoder as the constituent decoder, an interleaver, and a deinterleaver.

The two constituent decoders use the same trellis structure and decoding algorithm.

## Block Diagram of Iterative Turbo Decoding



The previous block diagram illustrates that the APP decoders (labeled as SISO modules in the previous image) output an updated sequence of log-likelihoods of the encoder input bits, $\Pi(u ; O)$. This sequence is based on the received sequence of log-likelihoods of the channel (coded) bits, $п(c ; I)$, and code parameters.

The decoder block iteratively updates these likelihoods for a fixed number of decoding iterations and then outputs the decision bits. The interleaver (п) that the decoder uses is identical to the one the encoder uses. The deinterleaver $\left(\Pi^{-1}\right)$ performs the inverse permutation with respect to the interleaver. The decoder does not assume knowledge of the tail bits and excludes these bits from the iterations.

## Dimensions

This block accepts an $M$-by- 1 column vector input signal and outputs an $L$-by- 1 column vector signal. For a given trellis, $L$ and $M$ are related by:

$$
L=\frac{(M-2 \cdot n u m T a i l s)}{(2 \cdot n-1)}
$$

and

$$
M=L \cdot(2 \cdot n-1)+2 \cdot n u m T a i l s
$$

where
$M=$ decoder input length
$L=$ decoder output length
$n=\log 2$ (trellis.NumOutputSymbols), for a rate $1 / 2$ trellis, $n=2$
numTails $=\log 2($ trellis.numStates $) * n$

## Bit Stream Ordering

The bit ordering subsystem reorganizes the incoming data into the two log likelihood ratio (LLR) streams input to the constituent decoders. This subsystem reconstructs the second systematic stream and reorders the bits so that they match the two constituent encoder outputs at the transmitter. This ordering subsystem is the inverse of the reordering subsystem at the turbo encoder.

## Turbo Decoder

Dialog Box


## Trellis structure

Trellis structure of constituent convolutional code.

Specify the trellis as a MATLAB structure that contains the trellis description of the constituent convolutional code. Alternatively, use the poly2trellis function to create a custom trellis using the constraint length, code generator (octal), and feedback connection (octal).

## Turbo Decoder

The default structure is the result of poly2trellis(4, [13 15], 13).

## Interleaver indices

Specify the mapping that the Turbo encoder block uses to permute the input bits as a column vector of integers. The default is (64:-1:1).'. This mapping is a vector with the number of elements equal to $L$, the length of the output signal. Each element must be an integer between 1 and $L$, with no repeated values.

## Decoding algorithm

Specify the decoding algorithm that the constituent APP decoders use to decode the input signal as True APP, Max*, Max. When you set this parameter to:

- True APP - the block implements true a posteriori probability decoding
- Max* or Max - the block uses approximations to increase the speed of the computations.


## Number of scaling bits

Specify the number of bits which the constituent APP decoders must use to scale the input data to avoid losing precision during computations. The decoder multiplies the input by $2^{\wedge}$ Number of scaling bits and divides the pre-output by the same factor. The value for this parameter must be a scalar integer between 0 and 8 . This parameter only applies when you set Decoding algorithm to Max*. The default is 3 .

## Number of decoding iterations

Specify the number of decoding iterations the block uses. The default is 6 .

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | • Double |
|  | $\bullet$ Single |
| Out | $\bullet$ Double |

## Turbo Decoder

| Examples | For an example that uses the Turbo Encoder and Turbo Decoder blocks, <br> see the Parallel Concatenated Convolutional Coding: Turbo Codes <br> example. |
| :--- | :--- |
| Pair Block | Turbo Encoder |
| See Also | APP Decoder <br> General Block Deinterleaver <br> General Block Interleaver |
| References | [1] Berrou, C., A. Glavieux, and P. Thitimajshima. "Near Shannon <br> limit error correcting coding and decoding: turbo codes," Proceedings <br> of the IEEE International Conference on Communications, Geneva, |
|  | Switzerland, May 1993, pp. 1064-1070. |
|  | [2] Benedetto, S., G. Montorsi, D. Divsalar, and F. Pollara. "Soft-Input <br> Soft-utput Maximum A Posterior (MAP) Module to Decode Parallel <br> and Serial Concatenated Codes," Jet Propulsion Lab TDA Progress <br> Report, Vol. 42-27, Nov. 1996. |
|  | [3] Schlegel, Christian B. and Lance C. Perez. Trellis and Turbo |
| Coding, IEEE Press, 2004. |  |

## Purpose

Encode binary data using parallel concatenated encoding scheme

## Library

Description


Convolutional sublibrary of Error Detection and Correction
The Turbo Encoder block encodes a binary input signal using a parallel concatenated coding scheme. This coding scheme employs two identical convolutional encoders and one internal interleaver. Each constituent encoder is independently terminated by tail bits.

## Block Diagram of Parallel Concatenated Convolutional Code



The previous block diagram illustrates that the output of the Turbo Encoder block consists of the systematic and parity bits streams of the first encoder, and only the parity bit streams of the second encoder.

For a rate one-half constituent encoder, the block interlaces the three streams and multiplexes the tail bits to the end of the encoded data streams.

For more information about tail bits, see the terminate Operation mode on the Convolutional Encoder block reference page.

## Dimensions

This block accepts an $L$-by- 1 column vector input signal and outputs an $M$-by- 1 column vector signal. For a given trellis, $M$ and $L$ are related by:

$$
M=L \cdot(2 \cdot n-1)+2 \cdot n u m T a i l s
$$

and

$$
L=\frac{(M-2 \cdot n u m T a i l s)}{(2 \cdot n-1)}
$$

where
$L=$ encoder input length
$M=$ encoder output length
$n=\log 2$ (trellis.NumOutputSymbols), for a rate $1 / 2$ trellis, $n=2$ numTails $=\log 2($ trellis.numStates $) * n$

## Encoder Schematic for Rate 1/3 Turbo Code Example



The previous schematic shows the encoder configuration for a trellis specified by the default value of the Trellis structure parameter, poly2trellis(4, [13 15], 13). For an input vector length of 64 bits, the output of the encoder block is 204 bits. The first 192 bits correspond to the three 64 bit streams (systematic ( $\mathrm{X}_{\mathrm{k}}$ ) and parity $\left(\mathrm{Z}_{\mathrm{k}}\right)$ bit streams from the first encoder and the parity $\left(\mathrm{Z}_{\mathrm{k}}^{\prime}\right)$ bit stream of the second encoder), interlaced as per $\mathrm{X}_{\mathrm{k}}, \mathrm{Z}_{\mathrm{k}}, \mathrm{Z}_{\mathrm{k}}^{\prime}$. The last 12 bits correspond to the tail bits from the two encoders, when the switches are in the lower position corresponding to the dashed lines. The first group of six bits

## Turbo Encoder

are the tail bits from the first constituent encoder and the second group is from the second constituent encoder.

Due to the tail limits, the encoder output code rate is slightly less than 1/3.

Function Block Parameters: Turbo Encoder
Turbo Encoder (mask) (link)
Encode the binary data using a parallel concatenated coding scheme that employs the convolutional encoder as the constituent encoder.

Both the constituent encoders use the same trellis structure.
The block punctures the second systematic bit stream and appends the termination bits at the end of the encoded data bits.

Use the poly2trellis function to create a trellis using the constraint length, code generator (octal) and feedback connection (octal).

Parameters
Trellis structure:
poly2trellis(4, [13 15], 13)|
Interleaver indices:
[64:-1:1].'

## Trellis structure

Trellis structure of constituent convolutional code.
Specify the trellis as a MATLAB structure that contains the trellis description of the constituent convolutional code. Alternatively,
use the poly2trellis function to create a custom trellis using the constraint length, code generator (octal), and feedback connections (octal).

This block supports only rate 1 -by- $N$ trellises where $N$ is an integer.

The default structure is the result of poly2trellis(4, [13 15], 13).

## Interleaver indices

Specify the mapping that the block uses to permute the input bits as a column vector of integers. The default is $(64:-1: 1) . '$. This mapping is a vector with the number of elements equal to the length, $L$, of the input signal. Each element must be an integer between 1 and $L$, with no repeated values.

## Supported Data Type

## Examples

Pair Block
Turbo Decoder
See Also Convolutional Encoder
General Block Interleaver

## Turbo Encoder

References<br>[1] Berrou, C., A. Glavieux, and P. Thitimajshima. "Near Shannon limit error correcting coding and decoding: turbo codes," Proceedings of the IEEE International Conference on Communications, Geneva, Switzerland, May 1993, pp. 1064-1070.<br>[2] Benedetto, S., G. Montorsi, D. Divsalar, and F. Pollara. " Soft-Input Soft-Output Maximum A Posterior (MAP) Module to Decode Parallel and Serial Concatenated Codes,"Jet Propulsion Lab TDA Progress Report, Vol. 42-27, Nov. 1996.<br>[3] Schlegel, Christian B. and Lance C. Perez. Trellis and Turbo Coding, IEEE Press, 2004.<br>[4] 3GPP TS 36.212 v9.0.0, 3rd Generation partnership project; Technical specification group radio access network; Evolved Universal Terrestrial Radio Acess (E-UTRA); Multiplexing and channel coding (release 9), 2009-12.

## Uniform Noise Generator

## Purpose

Generate uniformly distributed noise between upper and lower bounds

## Library

Description
Mrancos
Uniform
Noise Generators sublibrary of Comm Sources
The Uniform Noise Generator block generates uniformly distributed noise. The output data of this block is uniformly distributed between the specified lower and upper bounds. The upper bound must be greater
than or equal to the lower bound.

You must specify the Initial seed in the simulation. When it is a constant, the resulting noise is repeatable.

If all the elements of the output vector are to be independent and identically distributed (i.i.d.), then you can use a scalar for the Noise lower bound and Noise upper bound parameters. Alternatively, you can specify the range for each element of the output vector individually, by using vectors for the Noise lower bound and Noise upper bound parameters. If the bounds are vectors, then their length must equal the length of the Initial seed parameter.

## Attributes of Output Signal

The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the Frame-based outputs, Samples per frame, and Interpret vector parameters as 1-D parameters.

The number of elements in the Initial seed parameter becomes the number of columns in a frame-based output or the number of elements in a sample-based vector output. Also, the shape (row or column) of the Initial seed parameter becomes the shape of a sample-based two-dimensional output signal.

## Uniform Noise Generator

Dialog Box


## Noise lower bound, Noise upper bound

The lower and upper bounds of the interval over which noise is uniformly distributed.

## Initial seed

The initial seed value for the random number generator.

## Sample time

The period of each sample-based vector or each row of a frame-based matrix.

## Uniform Noise Generator

## Frame-based outputs

Determines whether the output is frame-based or sample-based. This box is active only if Interpret vector parameters as 1-D is unchecked.

## Samples per frame

The number of samples in each column of a frame-based output signal. This field is active only if Frame-based outputs is checked.

## Interpret vector parameters as 1-D

If this box is checked, then the output is a one-dimensional signal. Otherwise, the output is a two-dimensional signal. This box is active only if Frame-based outputs is unchecked.

## Output data type

The output can be set to double or single data types.
Random Source (DSP System Toolbox documentation); rand (built-in MATLAB function)

## Unipolar to Bipolar Converter

## Purpose Map unipolar signal in range [0, M-1] into bipolar signal <br> Library <br> Utility Blocks

Description The Unipolar to Bipolar Converter block maps the unipolar input signal to a bipolar output signal. If the input consists of integers between 0
Unipolar to Bipolar Converter and $\mathrm{M}-1$, where M is the $\mathbf{M}$-ary number parameter, then the output consists of integers between -(M-1) and M-1. If M is even, then the output is odd. If M is odd, then the output is even. This block is only designed to work when the input value is within the set $\{0,1,2 \ldots(\mathrm{M}-1)\}$, where M is the $\mathbf{M}$-ary number parameter. If the input value is outside of this set of integers the output may not be valid.

The table below shows how the block's mapping depends on the Polarity parameter.

| Polarity Parameter Value | Output Corresponding to <br> Input Value of $\mathbf{k}$ |
| :--- | :--- |
| Positive | $2 \mathrm{k}-(\mathrm{M}-1)$ |
| Negative | $-2 \mathrm{k}+(\mathrm{M}-1)$ |

## Unipolar to Bipolar Converter

## Dialog

 Box

## M-ary number

The number of symbols in the bipolar or unipolar alphabet.

## Polarity

A value of Positive causes the block to maintain the relative ordering of symbols in the alphabets. A value of Negative causes the block to reverse the relative ordering of symbols in the alphabets.

## Output Data Type

The type of bipolar signal produced at the block's output.
The block supports the following output data types:

- Inherit via internal rule
- Same as input
- double
- int8
- int16


## Unipolar to Bipolar Converter

- int32

When the parameter is set to its default setting, Inherit via internal rule, the block determines the output data type based on the input data type.

- If the input signal is floating-point (either single or double), the output data type is the same as the input data type.
- If the input data type is not floating-point:
- Based on the M-ary number parameter, an ideal signed integer output word length required to contain the range [-(M-1)M-1] is computed as follows:
ideal word length $=\operatorname{ceil}(\log 2(M))+1$

Note The +1 is associated with the need for the sign bit.

- The block sets the output data type to be a signed integer, based on the smallest word length (in bits) that can fit best the computed ideal word length.

Note The selections in the Hardware Implementation pane pertaining to word length constraints do not affect how this block determines output data types.

## Examples

If the input is $[0 ; 1 ; 2 ; 3]$, the $\mathbf{M}$-ary number parameter is 4 , and the Polarity parameter is Positive, then the output is $[-3 ;-1 ; 1 ; 3]$. Changing the Polarity parameter to Negative changes the output to [3; 1;-1;-3].

If the value for the $\mathbf{M}$-ary number is $2^{7}$ the block gives an output of int8.

## Unipolar to Bipolar Converter

If the value for the $\mathbf{M}$-ary number is $2^{7}+1$ the block gives an output of int16.

Pair Block Bipolar to Unipolar Converter

## Variable Step LMS Decision Feedback Equalizer

## Purpose

## Library

Description


Equalize using decision feedback equalizer that updates weights with variable-step-size LMS algorithm

Equalizers
The Variable Step LMS Decision Feedback Equalizer block uses a decision feedback equalizer and the variable-step-size LMS algorithm to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the variable-step-size LMS algorithm to update the weights, once per symbol. When you set the Number of samples per symbol parameter to 1 , then the block implements a symbol-spaced equalizer and updates the filter weights once for each symbol. When you set the Number of samples per symbol parameter to a value greater than 1 , the weights are updated once every $N^{\text {th }}$ sample, for a $T / N$-spaced equalizer.

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.
Set the Reference tap parameter so it is greater than zero and less than the value for the Number of forward taps parameter.

The port labeled Equalized outputs the result of the equalization process.
You can configure the block to have one or more of these extra ports:

- Mode input, as described in "Reference Signal and Operation Modes" in Communications System Toolbox User's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.


## Variable Step LMS Decision Feedback Equalizer

- Weights output, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.


## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Adaptive Algorithms" in Communications System Toolbox User's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to
$1+($ Reference tap -1$) /($ Number of samples per symbol)
Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

## Variable Step LMS Decision Feedback Equalizer



Dialog
Box

## Number of forward taps

The number of taps in the forward filter of the decision feedback equalizer.

## Variable Step LMS Decision Feedback Equalizer

## Number of feedback taps

The number of taps in the feedback filter of the decision feedback equalizer.

## Number of samples per symbol

The number of input samples for each symbol.

## Signal constellation

A vector of complex numbers that specifies the constellation for the modulation.

## Reference tap

A positive integer less than or equal to the number of forward taps in the equalizer.

## Initial step size

The step size that the variable-step-size LMS algorithm uses at the beginning of the simulation.

## Increment step size

The increment by which the step size changes from iteration to iteration

## Minimum step size

The smallest value that the step size can assume.

## Maximum step size

The largest value that the step size can assume.

## Leakage factor

The leakage factor of the variable-step-size LMS algorithm, a number between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

## Initial weights

A vector that concatenates the initial weights for the forward and feedback taps.

## Mode input port

When you select this check box, the block has an input port that enables you to toggle between training and decision-directed

## Variable Step LMS Decision Feedback Equalizer

mode. For training, the mode input must be 1, for decision directed, the mode should be 0 . The equalizer will train for the length of the Desired signal. If the mode input is not present, the equalizer will train at the beginning of every frame for the length of the Desired signal.

## Output error

When you select this check box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## Output weights

When you select this check box, the block outputs the current forward and feedback weights, concatenated into one vector.

References [1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, Wiley, 1998.

See Also<br>Variable Step LMS Linear Equalizer, LMS Decision Feedback Equalizer

## Variable Step LMS Linear Equalizer

## Purpose

## Library

Description


Equalize using linear equalizer that updates weights with variable-step-size LMS algorithm

## Equalizers

The Variable Step LMS Linear Equalizer block uses a linear equalizer and the variable-step-size LMS algorithm to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the variable-step-size LMS algorithm to update the weights, once per symbol. When you set the Number of samples per symbol parameter to 1 , then the block implements a symbol-spaced equalizer and updates the filter weights once for each symbol. When you set the Number of samples per symbol parameter to a value greater than 1 , the weights are updated once every $N^{\text {th }}$ sample, for a $T / N$-spaced equalizer.

## Input and Output Signals

The Input port accepts a column vector input signal. The Desired port receives a training sequence with a length that is less than or equal to the number of symbols in the Input signal. Valid training symbols are those symbols listed in the Signal constellation vector.
Set the Reference tap parameter so it is greater than zero and less than the value for the Number of taps parameter.

The Equalized port outputs the result of the equalization process.
You can configure the block to have one or more of these extra ports:

- Mode input, as described in "Reference Signal and Operation Modes" in Communications System Toolbox User's Guide.
- Err output for the error signal, which is the difference between the Equalized output and the reference signal. The reference signal consists of training symbols in training mode, and detected symbols otherwise.
- Weights output, as described in "Adaptive Algorithms" in Communications System Toolbox User's Guide.


## Variable Step LMS Linear Equalizer

## Decision-Directed Mode and Training Mode

To learn the conditions under which the equalizer operates in training or decision-directed mode, see "Adaptive Algorithms" in Communications System Toolbox User's Guide.

## Equalizer Delay

For proper equalization, you should set the Reference tap parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer output is equal to

## $1+($ Reference tap -1$) /($ Number of samples per symbol)

Since the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## Variable Step LMS Linear Equalizer

```
T
-Variable Step LMS Linear Equalizer (mask) (ink)
Equalize a linearly modulated signal through a dispersive channel using the variable step least mean squares (LMS) algorithm.
```


## Parameters <br> Number of taps:

## 4

Number of samples per symbol:
1
Signal constellation:
$[-3+3 \mathrm{j}-3+\mathrm{j}-3-\mathrm{j}-3-3 \mathrm{j}-1+3 \mathrm{j}-1+\mathrm{j}-1-\mathrm{j}-1-3 \mathrm{j} 1+3 \mathrm{j} 1+\mathrm{j} 1-\mathrm{j} 1-3 \mathrm{j} 3+3 \mathrm{j} 3+\mathrm{j} 3-\mathrm{j} 3-3 \mathrm{j}]$
Reference tap:
2
Initial step size:
.01
Increment step size:
.01
Minimum step size:
.001
Maximum step size:
.1
Leakage factor:
1
Initial weights:
0
$\sqrt{V}$ Mode input port
V Outputerror
V Output weights

Dialog Box


## Number of taps

The number of taps in the filter of the linear equalizer.

## Variable Step LMS Linear Equalizer

## Number of samples per symbol

The number of input samples for each symbol.

## Signal constellation

A vector of complex numbers that specifies the constellation for the modulation.

## Reference tap

A positive integer less than or equal to the number of taps in the equalizer.

## Initial step size

The step size that the variable-step-size LMS algorithm uses at the beginning of the simulation.

## Increment step size

The increment by which the step size changes from iteration to iteration

## Minimum step size

The smallest value that the step size can assume.

## Maximum step size

The largest value that the step size can assume.

## Leakage factor

The leakage factor of the LMS algorithm, a number between 0 and 1 . A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

## Initial weights

A vector that lists the initial weights for the taps.

## Mode input port

When you select this check box, the block has an input port that allows you to toggle between training and decision-directed mode. For training, the mode input must be 1, for decision directed, the mode should be 0 . For every frame in which the mode input is 1 or not present, the equalizer trains at the beginning of the frame for the length of the desired signal.

## Variable Step LMS Linear Equalizer

## Output error

When you select this check box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

## Output weights

When you select this check box, the block outputs the current weights.

Examples
References [1] Farhang-Boroujeny, B., Adaptive Filters: Theory and Applications, Chichester, England, Wiley, 1998.

See Also Variable Step LMS Decision Feedback Equalizer, LMS Linear Equalizer

## Viterbi Decoder

| Purpose | Decode convolutionally encoded data using Viterbi algorithm |
| :--- | :--- |
| Library | Convolutional sublibrary of Error Detection and Correction |

Description The Viterbi Decoder block decodes input symbols to produce binary output symbols. This block can process several symbols at a time for faster performance.

This block can output sequences that vary in length during simulation. For more information about sequences that vary in length, or variable-size signals, see "Variable-Size Signal Basics" in the Simulink documentation.

## Input and Output Sizes

If the convolutional code uses an alphabet of $2^{n}$ possible symbols, this block's input vector length is $L^{*} n$ for some positive integer $L$. Similarly, if the decoded data uses an alphabet of $2^{k}$ possible output symbols, this block's output vector length is $\mathrm{L}^{*} k$.

This block accepts a column vector input signal with any positive integer value for $L$. For variable-sized inputs, the $L$ can vary during simulation. The operation of the block is governed by the operation mode parameter."

For information about the data types each block port supports, see the "Supported Data Types" on page 2-907 table on this page.

## Input Values and Decision Types

The entries of the input vector are either bipolar, binary, or integer data, depending on the Decision type parameter.

| Decision type <br> Parameter | Possible <br> Entries in <br> Decoder Input | Interpretation <br> of Values | Branch metric <br> calculation |
| :--- | :--- | :--- | :--- |
| Unquantized | Real numbers | Positive real: <br> logical zero <br> Negative real: <br> logical one | Euclidean <br> distance |
| Hard Decision | 0,1 | $0:$ logical zero <br> $1:$ logical one | Hamming <br> distance |
| Soft Decision | Integers <br> between 0 and <br> $2^{\text {b }-1, ~ w h e r e ~} b$ is <br> the Number of <br> soft decision <br> bits parameter. | $0:$ most <br> confident <br> decision for <br> logical zero <br> $2^{\text {b }-1: ~ m o s t ~}$ <br> confident <br> decision for <br> logical one <br> Other values <br> represent | Hamming <br> distance |
|  |  | less confident <br> decisions. |  |

To illustrate the soft decision situation more explicitly, the following table lists interpretations of values for 3-bit soft decisions.

| Input Value | Interpretation |
| :--- | :--- |
| 0 | Most confident zero |
| 1 | Second most confident zero |
| 2 | Third most confident zero |
| 3 | Least confident zero |
| 4 | Least confident one |


| Input Value | Interpretation |
| :--- | :--- |
| 5 | Third most confident one |
| 6 | Second most confident one |
| 7 | Most confident one |

## Operation Modes for Inputs

The Viterbi decoder block has three possible methods for transitioning between successive input frames. The Operation mode parameter controls which method the block uses:

- In Continuous mode, the block saves its internal state metric at the end of each input, for use with the next frame. Each traceback path is treated independently.
- In Truncated mode, the block treats each input independently. The traceback path starts at the state with the best metric and always ends in the all-zeros state. This mode is appropriate when the corresponding Convolutional Encoder block has its Operation mode set to Truncated (reset every frame).
- In Terminated mode, the block treats each input independently, and the traceback path always starts and ends in the all-zeros state. This mode is appropriate when the uncoded message signal (that is, the input to the corresponding Convolutional Encoder block) has enough zeros at the end of each input to fill all memory registers of the feed-forward encoder. If the encoder has $k$ input streams and constraint length vector constr (using the polynomial description), "enough" means k*max (constr-1). For feedback encoders, this mode is appropriate if the corresponding Convolutional Encoder block has Operation mode set to Terminate trellis by appending bits.

Note When this block outputs sequences that vary in length during simulation and you set the Operation mode to Truncated or Terminated, the block's state resets at every input time step.

Use the Continuous mode when the input signal contains only one symbol.

## Traceback Depth and Decoding Delay

The Traceback depth parameter, D, influences the decoding delay. The decoding delay is the number of zero symbols that precede the first decoded symbol in the output.

- If you set the Operation mode to Continuous, the decoding delay consists of D zero symbols
- If the Operation mode parameter is set to Truncated or Terminated, there is no output delay and the Traceback depth parameter must be less than or equal to the number of symbols in each input.

If the code rate is $1 / 2$, a typical Traceback depth value is about five times the constraint length of the code.

## Reset Port

The reset port is usable only when the Operation mode parameter is set to Continuous. Selecting Enable reset input port gives the block an additional input port, labeled Rst. When the Rst input is nonzero, the decoder returns to its initial state by configuring its internal memory as follows:

- Sets the all-zeros state metric to zero.
- Sets all other state metrics to the maximum value.
- Sets the traceback memory to zero.

Using a reset port on this block is analogous to setting Operation mode in the Convolutional Encoder block to Reset on nonzero input via port.

The reset port supports double or boolean typed signals.

## Viterbi Decoder

## Fixed-Point Signal Flow Diagram

There are three main components to the Viterbi decoding algorithm. They are branch metric computation (BMC), add-compare and select (ACS), and traceback decoding (TBD). The following diagram illustrates the signal flow for a $k / n$ rate code.


## Viterbi Decoder

As an example of a BMC diagram, a $1 / 2$ rate, $n s d e c=3$ signal flow would be as follows.


$$
\begin{gathered}
W L=n s d e c+n-1 \\
n=2 \Rightarrow W L=4
\end{gathered}
$$

## Viterbi Decoder

The ACS component is generally illustrated as shown in the following diagram.


```
stMetNT(0, WL2, 0)
stMetFIMATH('floor', 'saturate')
```

Where WL2 is specified on the mask by the user.
In the flow diagrams above, inNT, bMetNT, stMetNT, and outNT are numerictype objects, and bMetFIMATH and stMetFIMATH, are fimath objects.

## Puncture Pattern Examples

For some commonly used puncture patterns for specific rates and polynomials, see the last three references.

Fixed-Point
Viterbi Decoding Examples

The following two example models showcase the fixed-point Viterbi decoder block used for both hard- and soft-decision convolutional decoding.

If you are reading this reference page in the MATLAB Help Browser, click Fixed-point Hard-Decision Viterbi Decoding and Fixed-point Soft-Decision Viterbi Decoding to open the models. These can also be found as doc_fixpt_vitharddec.mdl and doc_fixpt_vitsoftdec.mdl under help\toolbox\commm \examples.

Fixed-point Hard-Decision Viterbi Decoding


## Viterbi Decoder

## Fixed-point Soft-Decision Viterbi Decoding



The layout of the soft decision model example is also similar to the existing doc example on Soft-Decision Decoding, which can be found at help\toolbox\comm\examples\doc_softdecision.mdl

The purpose of this model is to highlight the fixed-point modeling attributes of the Viterbi decoder, using a familiar layout.

## Overview of the Simulations

The two simulations have a similar structure and have most parameters in common. A data source produces a random binary sequence that is convolutionally encoded, BPSK modulated, and passed through an AWGN channel.

The Convolutional encoder is configured as a rate $1 / 2$ encoder. For every 2 bits, the encoder adds another 2 redundant bits. To accommodate this, and add the correct amount of noise, the Eb/No (dB) parameter of the AWGN block is in effect halved by subtracting $10 * \log 10(2)$.

For the hard-decision case, the BPSK demodulator produces hard decisions, at the receiver, which are passed onto the decoder.

For the soft-decision case, the BPSK demodulator produces soft decisions, at the receiver, using the log-likelihood ratio. These soft outputs are 3 -bit quantized and passed onto the decoder.
After the decoding, the simulation compares the received decoded symbols with the original transmitted symbols in order to compute the bit error rate. The simulation ends after processing 100 bit errors or 1 e 6 bits, whichever comes first.

## Fixed-Point Modeling

Fixed-point modeling enables bit-true simulations which take into account hardware implementation considerations and the dynamic range of the data/parameters. For example, if the target hardware is a DSP microprocessor, some of the possible word lengths are 8,16 , or 32 bits, whereas if the target hardware is an ASIC or FPGA, there may be more flexibility in the word length selection.
To enable fixed-point Viterbi decoding, the block input must be of type ufix1 (unsigned integer of word length 1) for hard decisions. Based on this input (either a 0 or a 1 ), the internal branch metrics are calculated using an unsigned integer of word length = (number of output bits), as specified by the trellis structure (which equals 2 for the hard-decision example).
For soft decisions, the block input must be of type ufixN (unsigned integer of word length N ), where N is the number of soft-decision bits, to enable fixed-point decoding. The block inputs must be integers in the range 0 to $2^{\mathrm{N}-1}$. The internal branch metrics are calculated using an unsigned integer of word length $=(\mathrm{N}+$ number of output bits - 1$)$, as specified by the trellis structure (which equals 4 for the soft-decision example).

The State metric word length is specified by the user and usually must be greater than the branch metric word length already calculated. You can tune this to be the most suitable value (based on hardware and/or data considerations) by reviewing the logged data for the system.
Enable the logging by selecting Analysis > Fixed-Point Tool. In the Fixed-Point Setting GUI, set the Fixed-point instruments mode

## Viterbi Decoder

to Minimums, maximums and overflows, and rerun the simulation. If you see overflows, it implies the data did not fit in the selected container. You could either increase the size of the word length (if your hardware allows it) or try scaling the data prior to processing it. Based on the minimum and maximum values of the data, you are also able to determine whether the selected container is of the appropriate size.

Try running simulations with different values of State metric word length to get an idea of its effect on the algorithm. You should be able to narrow down the parameter to a suitable value that has no adverse effect on the BER results.

## Comparisons with Double-Precision Data

To run the same model with double precision data, Select Analysis > Fixed-Point Tool. In the Fixed-Point Tool GUI, select the Data type override to be Double. This selection overrides all data type settings in all the blocks to use double precision. For the Viterbi Decoder block, as Output type was set to Boolean, this parameter should also be set to double.

Upon simulating the model, note that the double-precision and fixed-point BER results are the same. They are the same because the fixed-point parameters for the model have been selected to avoid any loss of precision while still being most efficient.

## Comparisons Between Hard and Soft-Decision Decoding

The two models are set up to run from within BERTool to generate a simulation curve that compares the BER performance for hard-decision versus soft-decision decoding.
To generate simulation results for doc_fixpt_vitharddec.mdl, do the following:

1 Type bertool at the MATLAB command prompt.
2 Go to the Monte Carlo pane.
3 Set the Eb/No range to 2:5.

4 Set the Simulation model to doc_fixpt_vitharddec.mdl. Make sure that the model is on path.

5 Set the BER variable name to BER.
6 Set the Number of errors to 100, and the Number of bits to 1 e 6 .
7 Press Run and a plot generates.


To generate simulation results for doc_fixpt_vitsoftdec.mdl, just change the Simulation model in step 4 and press Run.

## Viterbi Decoder

Notice that, as expected, 3-bit soft-decision decoding is better than hard-decision decoding, roughly to the tune of 1.7 dB , and not 2 dB as commonly cited. The difference in the expected results could be attributed to the imperfect quantization of the soft outputs from the demodulator.

Dialog Box


## Trellis structure

MATLAB structure that contains the trellis description of the convolutional encoder. Use the same value here and in the corresponding Convolutional Encoder block.

## Punctured code

Select this check box to specify a punctured input code. The field, Punctured code, appears.

## Viterbi Decoder

## Puncture vector

Constant puncture pattern vector used at the transmitter (encoder). The puncture vector is a pattern of 1 s and 0 s . The 0 s indicate the punctured bits. When you select Punctured code, the Punctured vector field appears.

## Enable erasures input port

When you check this box, the decoder opens an input port labeled Era. Through this port, you can specify an erasure vector pattern of 1 s and 0 s , where the 1 s indicate the erased bits.

For these erasures in the incoming data stream, the decoder does not update the branch metric. The widths and the sample times of the erasure and the input data ports must be the same. The erasure input port can be of data type double or Boolean.

## Decision type

Specifies the use of Unquantized, Hard Decision, or Soft Decision for the branch metric calculation.

- Unquantized decision uses the Euclidean distance to calculate the branch metrics.
- Soft Decision and Hard Decision use the Hamming distance to calculate the branch metrics, where Number of soft decision bits equals 1 .


## Number of soft decision bits

The number of soft decision bits to represent each input. This field is active only when Decision type is set to Soft Decision.

## Error if quantized input values are out of range

Select this check box to throw an error when quantized input values are out of range. This check box is active only when Decision type is set to Soft Decision or Hard Decision.

## Traceback depth

The number of trellis branches to construct each traceback path.

## Operation mode

Method for transitioning between successive input frames:
Continuous, Terminated, and Truncated.

Note When this block outputs sequences that vary in length during simulation and you set the Operation mode to Truncated or Terminated, the block's state resets at every input time step.

## Enable reset input port

When you check this box, the decoder opens an input port labeled Rst. Providing a nonzero input value to this port causes the block to set its internal memory to the initial state before processing the input data.

## Delay reset action to next time step

When you select this option, the Viterbi Decoder block resets after decoding the encoded data. This option is available only when you set Operation mode to Continuous and select Enable reset input port. You must enable this option for HDL support.

## Viterbi Decoder

## Output data type



The output signal's data type can be double, single, boolean, int8, uint8, int16, uint16, int32, uint32, or set to 'Inherit via internal rule' or 'Smallest unsigned integer'.

When set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog
box of the model. If ASIC/FPGA is selected in the Hardware Implementation pane, the output data type is ufix(1). For all other selections, it is an unsigned integer with the smallest specified wordlength corresponding to the char value (e.g., uint8).

When set to 'Inherit via internal rule' (the default setting), the block selects double-typed outputs for double inputs, single-typed outputs for single inputs, and behaves similarly to the 'Smallest unsigned integer' option for all other typed inputs.

## Supported Data Types

| Port | Supported Data Types |
| :---: | :---: |
| Input | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean for Hard decision mode <br> - 8-, 16 -, and 32 -bit signed integers (for Hard decision and Soft decision modes) <br> - 8-, 16-, and 32 -bit unsigned integers (for Hard decision and Soft decision modes) <br> - ufix(n), where $n$ represents the Number of soft decision bits |
| Output | - Double-precision floating point <br> - Single-precision floating point <br> - Boolean <br> - 8-, 16-, and 32 -bit signed integers <br> - 8 -, 16-, and 32 -bit unsigned integers <br> - ufix(1) for ASIC/FPGA mode |

See Also<br>Convolutional Encoder, APP Decoder

## Viterbi Decoder

## References

[1] Clark, G. C. Jr. and J. Bibb Cain., Error-Correction Coding for Digital Communications, New York, Plenum Press, 1981.
[2] Gitlin, R. D., J. F. Hayes, and S. B. Weinstein, Data Communications Principles, New York, Plenum, 1992.
[3] Heller, J. A. and I. M. Jacobs, "Viterbi Decoding for Satellite and Space Communication," IEEE Transactions on Communication Technology, Vol. COM-19, October 1971, pp 835-848.
[4] Yasuda, Y., et. al., "High-rate punctured convolutional codes for soft decision Viterbi decoding," IEEE Transactions on Communications, Vol. COM-32, No. 3, pp 315-319, March 1984.
[5] Haccoun, D., and Begin, G., "High-rate punctured convolutional codes for Viterbi and sequential decoding," IEEE Transactions on Communications, Vol. 37, No. 11, pp 1113-1125, Nov. 1989.
[6] Begin, G., et.al., "Further results on high-rate punctured convolutional codes for Viterbi and sequential decoding," IEEE Transactions on Communications, Vol. 38, No. 11, pp 1922-1928, Nov. 1990.

## Purpose

Generate Walsh code from orthogonal set of codes

## Library

Description
Sequence Generators sublibrary of Comm Sources
Walsh codes are defined as a set of $N$ codes, denoted $W_{\mathrm{j}}$, for $j=0,1, \ldots$,

Walsh Code Generator
$N-1$, which have the following properties:

- $W_{\mathrm{j}}$ takes on the values +1 and -1 .
- $W_{\mathrm{j}}[0]=1$ for all $j$.
- $W_{\mathrm{j}}$ has exactly $j$ zero crossings, for $j=0,1, \ldots, N-1$.
- $W_{j} W_{k}^{T}= \begin{cases}0 & j \neq k \\ N & j=k\end{cases}$
- Each code Wjis either even or odd with respect to its midpoint.

Walsh codes are defined using a Hadamard matrix of order $N$. The Walsh Code Generator block outputs a row of the Hadamard matrix specified by the Walsh code index, which must be an integer in the range $[0, \ldots, N-1]$. If you set Walsh code index equal to an integer $j$, the output code has exactly $j$ zero crossings, for $j=0,1, \ldots, N-1$.

Note, however, that the indexing in the Walsh Code Generator block is different than the indexing in the Hadamard Code Generator block. If you set the Walsh code index in the Walsh Code Generator block and the Code index parameter in the Hadamard Code Generator block, the two blocks output different codes.

## Walsh Code Generator



Dialog Box

## Code length

Integer scalar that is a power of 2 specifying the length of the output code.

## Code index

Integer scalar in the range $[0,1, \ldots, \mathrm{~N}-1]$, where N is the Code length, specifying the number of zero crossings in the output code.

## Sample time

A positive real scalar specifying the sample time of the output signal.

## Frame-based outputs

When checked, the block outputs a frame-based signal. When cleared, the block outputs a [1] unoriented scalar.

## Samples per frame

The number of samples in a frame-based output signal. This field is active only if you select Frame-based outputs. If Samples per frame is greater than the Code length, the code is cyclically repeated.

## Output data type

The output type of the block can be specified as an int8 or double. By default, the block sets this to double.

## See Also

Hadamard Code Generator, OVSF Code Generator

## Windowed Integrator

## Purpose Integrate over time window of fixed length <br> Library <br> Comm Filters

Description The Windowed Integrator block creates cumulative sums of the input signal values over a sliding time window of fixed length. If the Integration period parameter is N and the input samples are denoted by $\mathrm{x}(1), \mathrm{x}(2), \mathrm{x}(3), \ldots$, then the nth output sample is the sum of the $\mathrm{x}(\mathrm{k})$ values for k between $\mathrm{n}-\mathrm{N}+1$ and n . In cases where $\mathrm{n}-\mathrm{N}+1$ is less than 1 , the block uses an initial condition of 0 to represent those samples.

## Input and Output Signals

This block accepts scalar, column vector, and $M$-by- $N$ matrix input signals. The block filters an $M$-by- $N$ input matrix as follows:

- When you set the Input processing parameter to Columns as channels (frame based), the block treats each column as a separate channel. In this mode, the block creates $N$ instances of the same filter, each with its own independent state buffer. Each of the $N$ filters process $M$ input samples at every Simulink time step.
- When you set the Input processing parameter to Elements as channels (sample based), the block treats each element as a separate channel. In this mode, the block creates $M^{*} N$ instances of the same filter, each with its own independent state buffer. Each filter processes one input sample at every Simulink time step.

The output dimensions always equal those of the input signal. For information about the data types each block port supports, see the "Supported Data Type" on page 2-918 table on this page.

## Dialog Box



## Integration period

The length of the interval of integration, measured in samples.

## Input processing

Specify how the block processes the input signal. You can set this parameter to one of the following options:

- Columns as channels (frame based) - When you select this option, the block treats each column of the input as a separate channel.


## Windowed Integrator

- Elements as channels (sample based) - When you select this option, the block treats each element of the input as a separate channel.

Note The Inherited (this choice will be removed - see release notes) option will be removed in a future release. See Frame-Based Processing in the Communications System Toolbox Release Notes for more information.

This parameter is available only when you set the Rate options parameter to Allow multirate processing.


## Rounding mode

Select the rounding mode for fixed-point operations. The block uses the Rounding mode when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. The filter coefficients do not obey this parameter; they always round to Nearest. For more information, see "Rounding Modes" in the DSP System Toolbox documentation or "Rounding Mode: Simplest" in the Fixed-Point Designer documentation.

## Windowed Integrator

## Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

## Coefficients

The block implementation uses a Direct-Form FIR filter with all tap weights set to one. The Coefficients parameter controls which data type represents the taps (i.e. ones) when the input data is a fixed-point signal.

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator). See "Filter Structure Diagrams" in DSP System Toolbox Reference Guide for illustrations depicting the use of the coefficient data types in this block:

- When you select Same word length as input, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Specify word length, you are able to enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the coefficients, in bits. If applicable, you are able to enter separate fraction lengths for the numerator and denominator coefficients.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the coefficients. If applicable, you are able to enter separate slopes for the
numerator and denominator coefficients. This block requires power-of-two slope and a bias of zero.
- The filter coefficients do not obey the Rounding mode and the Overflow mode parameters; they are always saturated and rounded to Nearest.


## Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" in DSP System Toolbox Reference Guide for illustrations depicting the use of the product output data type in this block:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the product output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.


## Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as product output, these characteristics match those of the product output.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.


## Windowed Integrator

- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.


## Output

Choose how you specify the output word length and fraction length:

- When you select Same as input, these characteristics match those of the input to the block.
- When you select Same as accumulator, these characteristics match those of the accumulator.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the output, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.


## Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.

## Supported Data Type

| Port | Supported Data Types |
| :--- | :--- |
| In | • Double-precision floating point |
|  | • Single-precision floating point |
|  | - Signed Fixed-point |
| Out | • Double-precision floating point |
|  | • Single-precision floating point |
|  | • Signed fixed-point |

Examples
If Integration period is 3 and the input signal is a ramp ( $1,2,3,4, \ldots$ ), then some of the sums that form the output of this block are as follows:

- $0+0+1=1$
- $0+1+2=3$
- $1+2+3=6$
- $2+3+4=9$
- $3+4+5=12$
- $4+5+6=15$
- etc.

The zeros in the first few sums represent initial conditions. With the Input processing parameter set to Elements as channels, then the values $1,3,6, \ldots$ are successive values of the scalar output signal. With the Input processing parameter set to Columns as channels, the values $1,3,6, \ldots$ are organized into output frames that have the same vector length as the input signal.

## See Also

Integrate and Dump, Discrete-Time Integrator (Simulink documentation)

## Windowed Integrator

Alphabetical List

## Purpose Adjacent Channel Power Ratio measurements

## Description <br> The ACPR System object measures adjacent channel power ratio (ACPR)

 of an input signal.
## Construction

H = comm.ACPR creates a System object, H, that measures adjacent channel power ratio (ACPR) of an input signal.

H = comm.ACPR(Name,Value) creates object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties

## NormalizedFrequency

Assume normalized frequency values
Specify whether the frequency values are normalized. If you set this property to true, the object assumes that frequency values are normalized (in the [-1 1] range). The default is false. If you set this property to false, the object assumes that frequency values are measured in Hertz.

## SampleRate

Sample rate of input signal
Specify the sample rate of the input signal, in samples per second, as a double-precision, positive scalar. The default is 1 e 6 samples per second. This property applies when you set the NormalizedFrequency property to false.

## MainChannelFrequency

Main channel center frequency
Specify the main channel center frequency as a double-precision scalar. The default is 0 Hz .

When you set the NormalizedFrequency property to true, you must specify the center frequency as a normalized value between -1 and 1 .

When you set the NormalizedFrequency property to false, you must specify the center frequency in Hertz. The object measures the main channel power in the bandwidth that you specify in the MainMeasurementBandwidth property. This measurement is taken at the center of the frequency that you specify in the MainMeasurementBandwidth property.

## MainMeasurementBandwidth

Main channel measurement bandwidth
Specify the main channel measurement bandwidth as a double-precision, positive scalar. The default is 50 e 3 Hz .

When you set the NormalizedFrequency property to true, you must specify the measurement bandwidth as a normalized value between 0 and 1.

When you set the NormalizedFrequency property to false, you must specify the measurement bandwidth in Hertz. The object measures the main channel power in the bandwidth that you specify in the MainMeasurementBandwidth property. This measurement is taken at the center of the frequency that you specify in the MainChannelFrequency property.

## AdjacentChannelOffset

Adjacent channel frequency offsets
Specify the adjacent channel offsets as a double-precision scalar or as a row vector comprising frequencies that define the location of adjacent channels of interest. The default is [-100e3 100e3] Hz.

When you set the NormalizedFrequency property to true, you must specify normalized frequency offset values between - 1 and 1. When you set the NormalizedFrequency property to false, you must specify frequency offset values in Hertz. The offset values indicate the distance between the main channel center
frequency and adjacent channel center frequencies. Positive offsets indicate adjacent channels to the right of the main channel center frequency. Negative offsets indicate adjacent channels to the left of the main channel center frequency.

## AdjacentMeasurementBandwidth

Adjacent channel measurement bandwidths
Specify the measurement bandwidth for each adjacent channel. The default is the scalar, 50 e 3 . The object assumes that each adjacent bandwidth is centered at the frequency defined by the corresponding frequency offset. You define this offset in the AdjacentChannelOffset property. Set this property to a double-precision scalar or row vector of length equal to the number of specified offsets in the AdjacentChannelOffset property.

When you set this property to a scalar, the object obtains all adjacent channel power measurements within equal measurement bandwidths. When you set the NormalizedFrequency property to true, you must specify normalized bandwidth values between 0 and 1. When you set the NormalizedFrequency property to false, you must specify the adjacent channel bandwidth values in Hertz.

## MeasurementFilterSource

Source of the measurement filter
Specify the measurement filter source as one of None | Property. The default is None. When you set this property to None the object does not apply filtering to obtain ACPR measurements. When you set this property to Property, the object applies a measurement filter to the main channel before measuring the average power. Each of the adjacent channel bands also receives a measurement filter. In this case, you specify the measurement filter coefficients in the MeasurementFilter property.

## MeasurementFilter

Measurement filter coefficients

Specify the measurement filter coefficients as a double-precision row vector containing the coefficients of an FIR filter in descending order of powers of $z$. Center the response of the filter at DC. The ACPR object automatically shifts and applies the filter response at each of the main and adjacent channel center frequencies before obtaining the average power measurements. The internal filter states persist and clear only when you call the reset method. This property applies when you set the MeasurementFilter property to Property. The default is 1 , which is an all-pass filter that has no effect on the measurements.

## SpectralEstimation

Spectral estimation control
Specify the spectral estimation control as one of Auto | Specify frequency resolution | Specify window parameters. The default is Auto.

When you set this property to Auto, the object obtains power measurements with a Welch spectral estimator with zero-percent overlap, a Hamming window, and a segment length equal to the length of the input data vector. In this setting, the spectral estimator set should achieve the maximum frequency resolution attainable with the input data length.

When you set this property to Specify frequency resolution, you specify the desired spectral frequency resolution, in normalized units or in Hertz, using the FrequencyResolution property. In this setting, the object uses the value in the FrequencyResolution property to automatically compute the size of the spectral estimator data window.

When you set this property to Specify window parameters, several spectral estimator properties become available so that you can control the Welch spectral estimation settings. These properties are: SegmentLength, OverlapPercentage, Window, and SidelobeAttenuation. Sidelobe attenuation applies only when you set the Window property to Chebyshev.

When you set the this property to Specify window parameters, the FrequencyResolution property does not apply, and you control the resolution using the above properties.

## Segmentlength

Segment length
Specify the segment length, in samples, for the spectral estimator as a numeric, positive, integer scalar. The default is 64 . The length of the segment allows you to make tradeoffs between frequency resolution and variance in the spectral estimates. A long segment length results in better resolution. A short segment length results in more averaging and a decrease in variance. This property applies when you set the SpectralEstimation property to Specify window parameters.

## OverlapPercentage

Overlap percentage
Specify the percentage of overlap between each segment in the spectral estimator as a double-precision scalar in the [0 100] interval. This property applies when you set the SpectralEstimation property to Specify window parameters. The default is 0 percent.

## Window

Window function
Specify a window function for the spectral estimator as one of Bartlett | Bartlett-Hanning | Blackman | Blackman-Harris | Bohman | Chebyshev | Flat Top | Hamming | Hann | Nuttall | Parzen | Rectangular | Triangular. The default is Hamming. A Hamming window has 42.5 dB of sidelobe attenuation. This attenuation may mask spectral content below this value, relative to the peak spectral content. Choosing different windows allows you to make tradeoffs between resolution and sidelobe attenuation. This property applies when you set the SpectralEstimation property to Specify window parameters.

## SidelobeAttenuation

Sidelobe attenuation for Chebyshev window
Specify the sidelobe attenuation, in decibels, for the Chebyshev window function as a double-precision, nonnegative scalar. The default is 100 dB . This property applies when you set the SpectralEstimation property to Specify window parameters and the Window property to Chebyshev.

## FrequencyResolution

Frequency resolution
Specify the frequency resolution of the spectral estimator as a double-precision scalar. The default is 10625 Hz .

When you set the NormalizedFrequency property to true, you must specify the frequency resolution as a normalized value between 0 and 1. When you set the NormalizedFrequency property to false, you must specify the frequency resolution in Hertz. The object uses the value in the FrequencyResolution property to calculate the size of the data window used by the spectral estimator. This property applies when you set the SpectralEstimation property to Specify frequency resolution.

## FFTLength

FFT length
Specify the FFT length that the Welch spectral estimator uses as one of Next power of $2 \mid$ Same as segment length | Custom. The default is Next power of 2 .

When you set this property to Custom, the CustomFFTLength property becomes available to specify the desired FFT length.

When you set this property to Next power of 2, the object sets the length of the FFT to the next power of 2. This length is greater than the spectral estimator segment length or 256 , whichever is greater.

When you set this property to Same as segment length, the object sets the length of the FFT. This length equals the spectral estimator segment length or 256, whichever is greater.

## CustomFFTLength

## Custom FFT length

Specify the number of FFT points that the spectral estimator uses as a numeric, positive, integer scalar. This property applies when you set the FFTLength property to Custom. The default is 256 .

## MaxHold

Max-hold setting control
Specify the maximum hold setting. The default is false.
When you set this property to true, the object compares two vectors. One vector compared is the current estimated power spectral density vector (obtained with the current input data frame). The object checks this vector against the previous maximum-hold accumulated power spectral density vector, (obtained at the previous call to the step method). The object stores the maximum values at each frequency bin and uses them to compute average power measurements. You clear the maximum-hold spectrum by calling the reset method on the object. When you set this property to false, the object obtains power measurements using instantaneous power spectral density estimates. This property is tunable.

## PowerUnits

Power units
Specify power measurement units as one of dBm \| dBW | Watts. The default is dBm.

When you set this property to dBm, or dBW, the step method outputs ACPR measurements in a dBc scale (adjacent channel power referenced to main channels power). If you set this property
to Watts, the step method outputs ACPR measurements in a linear scale.

## MainChannelPowerOutputPort

Enable main channel power measurement output
When you set this property to true, the step method outputs the main channel power measurement. The default is false. The main channel power is the power of the input signal measured in the band that you define with the MainChannelFrequency and MainMeasurementBandwidth properties. The step method returns power measurements in the units that you specify in the PowerUnits property.

## AdjacentChannelPowerOutputPort

Enable adjacent channel power measurements output
When you set this property to true, the step method outputs a vector of adjacent channel power measurements. The default is false. The adjacent channel powers correspond to the input signal's power measured in the bands that you define with the AdjacentChannelOffset and AdjacentMeasurementBandwidth properties. The step method returns power measurements in the units that you specify in the PowerUnits property.

## Methods

clone
getNumInputs
getNumOutputs
isLocked

Create ACPR measurement object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

| release | Allow property value and input <br> characteristics changes |
| :--- | :--- |
| reset | Reset states of ACPR <br> measurement object |
| step | Adjacent Channel Power Ratio <br> measurements |

## Examples

Measure ACPR of a 16-QAM signal with symbol rate of 3.84 Msps .

```
% Generate data with an alphabet size of 16 and modulate the data
x = randi([0 16-1],5000,1);
hMod = comm.RectangularQAMModulator(16);
y = step(hMod,x);
% Usample the data by L = 8 using a rectangular pulse shape
L = 8;
yPulse = rectpulse(y,L);
% Create an ACPR measurement object and measure the modulated signal
h = comm.ACPR(...
    'SampleRate', 3.84e6*8,...
    'MainChannelFrequency', 0,...
    'MainMeasurementBandwidth', 3.84e6,...
    'AdjacentChannelOffset', [-5e6 5e6],...
    'AdjacentMeasurementBandwidth', 3.84e6,...
    'MainChannelPowerOutputPort', true,...
    'AdjacentChannelPowerOutputPort', true);
[ACPR,mainChnlPwr,adjChnlPwr] = step(h,yPulse)
```


## Algorithms

```
Note The following conditions must be true, otherwise power measurements fall out of the Nyquist interval.
    \(\mid\) MainChannelFreq \(\left. \pm \frac{\text { MainChannelMeasBW }}{2} \right\rvert\,<F \max\)
    \(\left.\left\lvert\,(\) MainChannelFreq + AdjChannelOffset \() \pm \frac{\text { AdjChannelMeasBW }}{2}\right. \right\rvert\,<F \max\)
\(\mathrm{F}_{\text {max }}=\mathrm{Fs} / 2\) if NormalizedFrequency \(=\) false
\(\mathrm{F}_{\max }=1\) if NormalizedFrequency \(=\) true
```

See Also
comm.CCDF | comm.EVM | comm.MER

Purpose Create ACPR measurement object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a ACPR object $C$, with the same property values as H. The clone method creates a new unlocked object with uninitialized states.

Purpose $\quad$ Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.ACPR.getNumOutputs

Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn outputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the ACPR System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description
release (H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset states of ACPR measurement object |
| :--- | :--- |
| Syntax | $\operatorname{reset}(H)$ |
| Description | $\operatorname{reset}(H)$ resets the states of the ACPR object, H. |

```
Purpose Adjacent Channel Power Ratio measurements
Syntax \(\quad A=\operatorname{step}(H, X)\)
[A,MAINPOW] = step(H,X)
[A,ADJPOW] = step(H,X)
```


## Description

$A=\operatorname{step}(H, X)$ returns a vector of the adjacent channel power ratio, $A$, measured in the input data, $X$. The measurements are at the frequency bands that you specify with the MainChannelFrequency, MainMeasurementBandwidth, AdjacentChannelOffset, and AdjacentMeasurementBandwidth properties. Input X must be a double precision column vector. The length of the output vector, A, equals the number of adjacent channels that you specify in the AdjacentChannelOffset property.
[A,MAINPOW] = step $(H, X)$ returns the measured main channel power, MAINPOW, when you set the MainChannelPowerOutputPort property to true. The step method outputs the main channel power measured within the main channel frequency band of interest that you specify with the MainChannelFrequency and MainMeasurementBandwidth properties.
[A,ADJPOW] $=\operatorname{step}(H, X)$ returns a vector of the measured adjacent channel powers, ADJPOW, when you set the AdjacentChannelPowerOutputPort property to true. The adjacent channel powers are measured at the adjacent frequency bands of interest that you specify with the AdjacentChannelOffset and AdjacentMeasurementBandwidth properties. The length of the output vector, ADJPOW, equals the length of the vector that you specify in the AdjacentChannelOffset property. You can combine optional output arguments when you set their enabling properties. Optional outputs must be listed in the same order as the order of the enabling properties. For example,


#### Abstract

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.


## Purpose Adaptively adjust gain for constant signal-level output

Description

Construction

The comm.AGC System object creates an automatic gain controller (AGC) that adaptively adjusts its gain to achieve a constant signal level at the output.
$\mathrm{H}=$ comm.AGC creates an automatic gain controller (AGC) System object, H , that adaptively adjusts its gain to achieve a constant signal level at the output.

H = comm.AGC(Name, Value) creates an AGC object, H, with the specified property Name set to the specified Value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties

## DetectorMethod

Detector method
Specify the detector method as one of Rectifier | Square Law. The default is Rectifier.

When you set the DetectorMethod to Rectifier, the AGC detector outputs a voltage value proportional to the envelope amplitude of the output signal. The detector rectifies and then averages the input signal over the update period. The AGC adjusts the gain to obtain unity voltage at the output of the detector.

When you set the DetectorMethod to Square law, the AGC detector outputs a power value that is proportional to the square of the output voltage. The detector squares and then averages the input signal over the update period. The AGC adjusts the gain to obtain unity power at the output of the detector.

## LoopMethod

Loop method
Specify the loop method of the AGC as one of Linear | Logarithmic. The default is Linear.

When you set the LoopMethod to Linear, the AGC uses the direct value of the detector output to determine the gain value. Typically, a linear loop responds quickly to increases in the input signal level. However, the loop's response to decreases in the input signal level tends to be slow.

When you set the LoopMethod to Logarithmic, the AGC uses the logarithm of the detector output to determine the gain value. Logarithmic loops respond to decreases in the input signal level much more quickly than linear loops.

## UpdatePeriod

Period of gain updates in samples
Specify the period of the gain updates as a double- or single-precision, real, integer-valued scalar. The default is 100 .

The number of input samples must be an integer multiple of update period. Setting the period greater than 1 increases the speed of the AGC algorithm.

If you increase the update period, you may also need to increase the step size. Similarly, if you decrease the update period, you may also need to decrease the step size.

## StepSize

Step size for gain updates
Specify the step size for gain updates as a double- or single-precision, real, positive scalar. The default is 0.1.

If you increase the loop gain, the AGC responds to changes at the input signal level faster. However, gain pumping also increase.

If you increase the update period, you may also need to increase the step size. Similarly, if you decrease the update period, you may also need to decrease the step size.

## MaximumGain

Maximum gain in decibels

Specify the maximum gain of the AGC in decibels as a positive scalar. The default is 30 .

If the signal at the input of the AGC has a very low signal level, the AGC gain may increase rapidly. Use this property to limit the gain that the AGC applies to the input signal.

| Methods | clone |
| :--- | :--- |
|  | isLocked |
|  | release |
|  | reset |
|  | step |

## Examples Adaptively adjust the Received Signal Amplitude to Approximately 1 Volt

Modulate a QPSK signal of amplitude 4, set the received signal amplitude to approximately 1 volt using an AGC, and then plot the output.

Create a QPSK modulated signal of 10,000 symbols with an amplitude of 4 .

```
d = randi([0 3], 10000, 1);
hMod = comm.PSKModulator(4, 'PhaseOffset', pi/4);
x = 4*step(hMod, d);
```

Create an AGC System object to adjust the received signal amplitude to approximately 1 .

```
hAGC = comm.AGC;
y = step(hAGC, x);
```

After the AGC reaches steady state, plot the output.

```
figure;plot(x(1000:end), '*')
hold on;plot(y(1000:end), 'or'); grid on; axis square
legend('Input of AGC', 'Output of AGC')
```


## Compare the Performance of an AGC with a Rectifier Detector and a Square Law Detector

Modulate a QPSK signal, adjust the received signal level using a, and then plot the AGC response as a function of time.

Create a QPSK modulated signal of 10000 symbols with an amplitude of 4 .

```
d = randi([0 3], 300, 1);
hMod = comm.PSKModulator(4, 'PhaseOffset', pi/4);
x = 4*step(hMod, d);
```

Create two AGC System objects to adjust the received signal level. Use a rectifier detector and a square law detector, each with update period 10 .

```
hAGC1 = comm.AGC('DetectorMethod', 'Rectifier', 'UpdatePeriod', 10);
hAGC2 = comm.AGC('DetectorMethod', 'Square law', 'UpdatePeriod', 10);
y1 = step(hAGC1, x);
y2 = step(hAGC2, x);
Plot AGC response as a function of time.
```

```
figure;plot(abs(y1), 'b')
```

figure;plot(abs(y1), 'b')
hold on;plot(abs(y2), 'r'); grid on; axis square
hold on;plot(abs(y2), 'r'); grid on; axis square
xlabel('Time'); ylabel('Amplitude')
xlabel('Time'); ylabel('Amplitude')
legend('Rectifier detector', 'Square law detector')

```
legend('Rectifier detector', 'Square law detector')
```


## Plot the Effect of Step Size on AGC Performance

Create two AGC System objects to adjust the received signal level using two different step sizes with update period 10 .

Create a QPSK modulated signal of 10000 symbols with an amplitude of 4 .

```
d = randi([0 3], 1000, 1);
hMod = comm.PSKModulator(4, 'PhaseOffset', pi/4);
x = 4*step(hMod, d);
```

Use two AGC System objects to adjust the received signal level. Select a step size of 0.01 and 0.1 . Set the update period to 10 for both cases.

```
hAGC1 = comm.AGC('StepSize', 0.01, 'UpdatePeriod', 10);
hAGC2 = comm.AGC('StepSize', 0.1, 'UpdatePeriod', 10);
y1 = step(hAGC1, x);
y2 = step(hAGC2, x);
```

Plot AGC response as a function of time.

```
figure;plot(abs(y1), 'b')
hold on;plot(abs(y2), 'r'); grid on; axis square
xlabel('Time'); ylabel('Amplitude')
legend('Step size 0.01', 'Step size 0.1')
```


## Plot Effect of Maximum Gain on Burst Signals

Create a QPSK modulated burst with a length of 2000 symbols. Prepend the burst with an empty channel by concatenating an all zero vector. Add 20 dB additive white Gaussian noise.

Create a burst signal of 2,000 QPSK modulated symbols with an amplitude of 1 .

```
d = randi([0 3], 2000, 1);
hMod = comm.PSKModulator(4, 'PhaseOffset', pi/4);
x = [zeros(2000, 1); step(hMod, d)];
```

$r=\operatorname{awgn}(x, 20)$;
Create two AGC Sytem objects to adjust the received signal level using a maximum gain of 15 dB and 5 dB with an update period of 10 and a step size of 0.1.

```
hAGC1 = comm.AGC('MaximumGain', 15, 'UpdatePeriod', 10, 'StepSize', 0
hAGC2 = comm.AGC('MaximumGain', 5, 'UpdatePeriod', 10, 'StepSize', 0.
y1 = step(hAGC1, x);
y2 = step(hAGC2, x);
```

Plot the AGC response as a function of time. Limiting the maximum gain of the AGC enables the AGC to adjust faster when the packet arrives after the noise only signal.

```
figure;plot(abs(y1), 'b')
hold on;plot(abs(y2), 'r'); grid on; axis square
xlabel('Time'); ylabel('Amplitude')
legend('Maximum gain 15 dB', 'Maximum gain 12 dB')
```


## Algorithms Linear Loop AGC

In a linear loop AGC, the detector uses its output directly to generate an error signal. After applying a step size, the AGC passes the error signal to an integrator. The output of the integrator is used as the variable gain. Linear loop AGCs are limited by their decay, or slew, characteristics. In other words, they respond to input signal increases much more quickly than they respond to input signal decreases.


$$
\begin{aligned}
& y(n)=g(n) \cdot x(n) ; \\
& e(n)=A-z(m) ; \\
& g(n+1)=g(n)+K \cdot e(n) ;
\end{aligned}
$$

where
$A$ represents the reference value, which is 1
$K$ represents the step size
$e$ represents the error signal
$g$ represents the gain
$x$ represents the input signal
$y$ represents the output signal
$z$ represents the detector output

## Logarithmic Loop AGC

In a logarithmic loop AGC, the logarithm of the ratio of the detector output and the reference signal represents the error signal. A
logarithmic loop uses the exponential of the integrator output as the gain signal. Logarithmic-loop AGCs have the same response time to both increases or decreases to the input signal amplitude.


The logarithmic loop has longer attack and decay times. However, the gain pumping of the logarithmic loop is better than the linear loop.

$$
\begin{aligned}
& y(n)=e^{g(n)} \cdot x(n) \\
& e(n)=\ln (A)-\ln (z(m)) \\
& g(n+1)=g(n)+K \cdot e(n) ;
\end{aligned}
$$

where
$A$ represents the reference value, which is 1
$K$ represents the step size
$e$ represents the error signal
$g$ represents the gain
$x$ represents the input signal
$y$ represents the output signal
$z$ represents the detector output

## AGC Detector

Two AGC detectors are available:

## Rectifier type detector

$z=|y|$ when the detector represents a rectifier

$$
z(m)=\frac{1}{N} \sum_{n=m N}^{(m+1) N-1}|y(n)|
$$

where $N$ represents the update period

## Square law type detector

$\mathrm{z}=|\mathrm{y}|^{2}$ represents the square law detector

$$
z(m)=\frac{1}{N} \sum_{n=m N}^{(m+1) N-1}|y(n)|^{2}
$$

where $N$ represents the update period

## Performance Considerations

There are three performance criteria for AGCs:

- Attack time: The duration it takes the AGC to respond to an increase in the input amplitude.
- Decay time: The duration it takes the AGC to respond to a decrease in the input amplitude.
- Gain pumping: The variation in the gain value during steady-state operation.

Increasing the step size decreases the attack time and decay times, but it also increases gain pumping.

Purpose Create AGC object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates an AGC object $C$, with the same property values as H. The clone method creates a new unlocked object with uninitialized states.

The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the AGC System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description
release (H) releases system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset internal states of automatic gain controller |
| :--- | :--- |
| Syntax | reset $(H)$ |
| Description | reset $(H)$ resets the filter states of the automatic gain controller filter <br> object, $H$, to their initial values. |

Purpose Apply adaptive gain to input signal

## Syntax <br> Y = step(H,X)

Description $\quad Y=\operatorname{step}(H, X)$ applies an adaptive gain to the input $X$, to achieve a unity signal level at the output, Y . X must be a double or single precision column vector. The AGC determines the output signal level based on the DetectorMethod setting.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose

## Description

## Construction

## Properties

Deinterleave input symbols using algebraically derived permutation vector

The AlgebraicDeinterleaver object restores the original ordering of a sequence that was interleaved using the AlgebraicInterleaver object. In typical usage, the properties of the two objects have the same values.

H = comm.AlgebraicDeinterleaver creates a deinterleaver System object, H. This object restores the original ordering of a sequence from the corresponding algebraic interleaver object.

H = comm.AlgebraicDeinterleaver(Name, Value) creates an
Algebraic deinterleaver System object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Method

Algebraic method to generate permutation vector
Specify the algebraic method as one of Takeshita-Costello| Welch-Costas. The default is Takeshita-Costello. The algebraic interleaver performs all computations in modulo $N$, where $N$ equals the length you set in the Length property.

For the Welch-Costas method, the value of $(N+1)$ must be a prime number, where $N$ equals the value you specify in the Length property. You must set the PrimitiveElement property to an integer, $A$, between 1 and $N$. This integer represents a primitive element of the finite field $G F(N+1)$.

For the Takeshita-Costello method, you must set the Length property to a value equal to $2^{m}$, for any integer $m$. You must also set the MultiplicativeFactor property to an odd integer that is less than the value of the Length property. The CyclicShift property requires a nonnegative integer which is less than the value of the Length property. The Takeshita-Costello interleaver method uses a cycle vector of length $N$, which you
specify in the Length property. The cycle vector calculation
uses the equation, $\bmod \left(k \times(n-1) \times \frac{n}{2}, N\right)+1$, for any integer $n$, between 1 and $N$. The object creates an intermediate permutation
function using the relationship, $P(c(n))=c(n+1)$. You can shift the elements of the intermediate permutation vector to the left by the amount specified by the CyclicShift property. Doing so produces the interleaver's actual permutation vector.

## Length

Number of elements in input vector
Specify the number of elements in the input as a positive, integer, scalar. When you set the Method property to Welch-Costas, then the value of Length +1 must equal a prime number. When you set the Method property to Takeshita-Costello, then the value of the Length property requires a power of two. The default is 256 .

## MultiplicativeFactor

Cycle vector computation factor
Specify the factor the object uses to compute the interleaver's cycle vector as a positive, integer, scalar. This property applies when you set the Method property to Takeshita-Costello. The default is 13.

## CyclicShift

Amount of cyclic shift
Specify the amount by which the object shifts indices, when the object creates the final permutation vector, as a nonnegative, integer, scalar. The default is 0 . This property applies when you set the Method property to Takeshita-Costello.

## PrimitiveElement

Primitive element

Specify the primitive element as an element of order $N$ in the finite field $G F(N+1)$. $N$ is the value you specify in the Length property.
You can express every nonzero element of $G F(N+1)$ as the value of the PrimitiveElement property raised to some integer power. In a Welch-Costas interleaver, the permutation maps the integer $k$ to $\bmod \left(\mathrm{A}^{\mathrm{k}}, \mathrm{N}+1\right)-1$, where $A$ represents the value of the PrimitiveElement property. This property applies when you set the Method property to Welch-Costas. The default is 6 .

Methods<br>clone getNumInputs getNumOutputs isLocked release step

Create algebraic deinterleaver object with same property values
Number of expected inputs to step method
Number of outputs from step method
Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
Deinterleave input symbols using algebraically derived permutation vector

Examples Interleave and deinterleave data

```
hInt = comm.AlgebraicInterleaver('Length', 16);
hDeInt = comm.AlgebraicDeinterleaver('Length', 16);
data = randi(7, 16, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
[data, intData, deIntData]
```

Algorithms This object implements the algorithm, inputs, and outputs described on the Algebraic Deinterleaver block reference page. The object properties correspond to the block parameters.

See Also comm.AlgebraicInterleaver | comm.BlockInterleaver

## comm.AlgebraicDeinterleaver.clone

Purpose
Syntax $\quad C=$ clone $(H)$
Description
$C=$ clone (H) creates a AlgebraicDeinterleaver object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $N=$ getNumInputs $(H)$ method returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNuminputs ( $H$ ).

## comm.AlgebraicDeinterleaver.getNumOutputs

## Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the AlgebraicDeinterleaver System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose
Syntax release (H)
Description

Allow property value and input characteristics changes
release (H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Deinterleave input symbols using algebraically derived permutation <br> vector |
| :--- | :--- |
| Syntax | $Y=\operatorname{step}(H, X)$ |
| Description | $Y=\operatorname{step}(H, X)$ restores the original ordering of the sequence, $X$, that |
| was interleaved using an algebraic interleaver. An algebraically derived |  |
| permutation vector based on the algebraic method you specify in the |  |
| Method property forms the base of the output, $Y . X$ must be a column |  |
| vector of length specified by the Length property. $X$ can be numeric, |  |
| logical, or fixed-point (fi objects). $Y$ has the same data type as $X$. |  |

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose <br> Description

## Construction

## Properties

Permute input symbols using algebraically derived permutation vector

The AlgebraicInterleaver object rearranges the elements of its input vector using an algebraically derived permutation.

H = comm.AlgebraicInterleaver creates an interleaver System object, H , that permutes the symbols in the input signal. This permutation is based on an algebraically derived permutation vector.

H = comm.AlgebraicInterleaver(Name, Value) creates an algebraic interleaver object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Method

Algebraic method to generate permutation vector
Algebraic method to generate permutation vector
Specify the algebraic method as one of Takeshita-Costello| Welch-Costas. The default is Takeshita-Costello. The algebraic interleaver performs all computations in modulo $N$, where $N$ is the length you set in the Length property.

For the Welch-Costas method, the value of $(N+1)$ must be a prime number, where $N$ is the value you specify in the Length property. You must set the PrimitiveElement property to an integer, $A$, between 1 and $N$. This integer represents a primitive element of the finite field $G F(N+1)$.

For the Takeshita-Costello method, you must set the Length property to a value equal to $2^{m}$, for any integer $m$. You must also set the MultiplicativeFactor property to an odd integer which is less than the value of the Length property. In addition, you must set the CyclicShift property to a nonnegative integer which is less than the value of the Length property. The Takeshita-Costello interleaver method uses a cycle vector of
length $N$, which you specify in the Length property. The cycle vector calculation uses the equation, $\bmod \left(k \times(n-1) \times \frac{n}{2}, N\right)+1$, for any integer $n$, between 1 and $N$. The object creates an intermediate permutation function using the relationship, $P(c(n))=c(n+1)$. You can shift the elements of the intermediate permutation vector to the left by the amount specified by the CyclicShift property. Doing so produces the actual permutation vector of the interleaver.

## Length

Number of elements in input vector
Specify the number of elements in the input as a positive, integer, scalar. When you set the Method property to Welch-Costas, then the value of Length +1 must equal a prime number. When you set the Method property to Takeshita-Costello, then the value of the Length property requires a power of two. The default is 256 .

## MultiplicativeFactor

Cycle vector computation method
Specify the factor the object uses to compute the cycle vector for the interleaver as a positive, integer, scalar. This property applies when you set the Method property to Takeshita-Costello. The default is 13.

## CyclicShift

Amount of cyclic shift
Specify the amount by which the object shifts indices, when it creates the final permutation vector, as a nonnegative, integer, scalar. This property applies when you set the Method property to Takeshita-Costello. The default is 0 .

## PrimitiveElement

Primitive element

Specify the primitive element as an element of order $N$ in the finite field $G F(N+1) . N$ is the value you specify in the Length property. You can express every nonzero element of $G F(N+1)$ as the value of the PrimitiveElement property raised to an integer power. In a Welch-Costas interleaver, the permutation maps the integer $k$ to $\bmod \left(\mathrm{A}^{\mathrm{k}}, \mathrm{N}+1\right)-1$, where $A$ represents the value of the PrimitiveElement property. This property applies when you set the Method property to Welch-Costas. The default is 6 .

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>step

Create algebraic interleaver object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Permute input symbols using an algebraically derived permutation vector

Examples Interleave and deinterleave data

```
hInt = comm.AlgebraicInterleaver('Length', 16);
hDeInt = comm.AlgebraicDeinterleaver('Length', 16);
data = randi(7, 16, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
[data, intData, deIntData]
```

Algorithms | This object implements the algorithm, inputs, and outputs described on |
| :--- |
| the Algebraic Interleaver block reference page. The object properties |
| correspond to the block parameters. |

See Also $\quad$ comm.AlgebraicDeinterleaver

Purpose
Create algebraic interleaver object with same property values

## Syntax <br> C = clone(H)

Description

C = clone (H) creates a AlgebraicInterleaver object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $N=$ getNumInputs (H) method returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNuminputs ( $H$ ).

## comm.AlgebraicInterleaver.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked $(H)$ returns the locked status, TF of the AlgebraicInterleaver System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Permute input symbols using an algebraically derived permutation <br> vector |
| :--- | :--- |
| Syntax | $Y=\operatorname{step}(H, X)$ |
| Description | $Y=\operatorname{step}(H, X)$ permutes input sequence, $X$, and returns interleaved |
| sequence, $Y$. The object uses an algebraically derived permutation |  |
| vector, based on the algebraic method you specify in the Method |  |
| property, to form the output. The input $X$ must be a column vector of |  |
| length specified by the Length property. $X$ can be numeric, logical, or |  |
| fixed-point (fi objects). $Y$ has the same data type as $X$. |  |

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose <br> Description

## Construction

Decode convolutional code using the a posteriori probability method

The APPDecoder object performs a posteriori probability (APP) decoding of a convolutional code.
$\mathrm{H}=$ comm.APPDecoder creates an a posteriori probability (APP) decoder System object, $H$, that decodes a convolutional code using the APP method.

H = comm.APPDecoder(Name, Value) creates an APP decoder object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.APPDecoder(TRELLIS, Name, Value) creates an APP decoder object, H, with the TrellisStructure property set to TRELLIS, and the other specified properties set to the specified values.

## Properties

## TrellisStructure

Trellis structure of convolutional code
Specify trellis as a MATLAB structure that contains the trellis description of the convolutional code. The default is the result of poly2trellis(7, [171 133], 171). Use the istrellis function to check if a structure is a valid trellis structure.

## TerminationMethod

Termination method of encoded frame
Specify how the encoded frame is terminated as one of Truncated | Terminated. The default is Truncated. When you set this property to Truncated, the object assumes that the encoder stops after encoding the last symbol in the input frame. When you set this property to Terminated the object assumes that the encoder forces the trellis to end each frame in the all-zeros state by encoding additional symbols. If you use the comm. ConvolutionalEncoder System object to generate the
encoded frame, the TerminationMethod values of both encoder and decoder objects must match.

## Algorithm

Decoding algorithm
Specify the decoding algorithm that the object uses as one of True APP | Max* | Max. The default is Max*. When you set this property to True APP, the object implements true a posteriori probability decoding. When you set the property to any other value, the object uses approximations to increase the speed of the computations.

## NumScalingBits

Number of scaling bits
Specify the number of bits the decoder uses to scale the input data to avoid losing precision during the computations. The default is 3. The decoder multiplies the input by $2^{\text {NumScalingBits }}$ and divides the pre-output by the same factor. This property must be a scalar integer between 0 and 8 . This property applies when you set the Algorithm property to Max*.

## CodedBitLLROutputPort

Enable coded-bit LLR output
Set this property to false to disable the second output of the decoding step method. The default is true.

Methods<br>clone<br>getNumInputs<br>getNumOutputs

Create APP decoder object with same property values with same property values

Number of expected inputs to step method

Number of outputs from step method

| isLocked | Locked status for input attributes <br> and nontunable properties |
| :--- | :--- |
| release | Allow property value and input <br> characteristics changes |
| reset | Reset states of APP decoder object |
| step | Decode convolutional code using <br> the a posteriori probability <br> method |

## Examples Transmit a convolutionally encoded 8-PSK-modulated bit stream through an AWGN channel, then demodulate, decode using an APP decoder, and count errors.

## 1

Create the Convolutional encoder, PSK Modulator, and AWGN Channel System objects.
noiseVar = 2e-1;
frameLength = 300;
hConEnc = comm.ConvolutionalEncoder('TerminationMethod','Truncated');
hMod $=$ comm. PSKModulator('BitInput',true, 'PhaseOffset',0);
hChan $=$ comm.AWGNChannel('NoiseMethod', 'Variance', ...
'Variance' , noiseVar) ;

2
Demodulate using soft-decision decoding.

```
hDemod = comm.PSKDemodulator('BitOutput',true, 'PhaseOffset',0,
    'DecisionMethod', 'Approximate log-likelihood ratio',
    'Variance', noiseVar);
hAPPDec = comm.APPDecoder(...
    'TrellisStructure', poly2trellis(7, [171 133]), ...
    'Algorithm', 'True APP', 'CodedBitLLROutputPort', fals
hError = comm.ErrorRate;
```

3
Decode the convolutionally encoded data. Then, convert from soft-decision to hard-decision.

```
for counter = 1:5
        data = randi([[0 1],frameLength,1);
        encodedData = step(hConEnc, data);
        modSignal = step(hMod, encodedData);
        receivedSignal = step(hChan, modSignal);
        demodSignal = step(hDemod, receivedSignal);
    receivedSoftBits = step(hAPPDec, zeros(frameLength,1), -demodSignal)
receivedBits = double(receivedSoftBits > 0);
        errorStats = step(hError, data, receivedBits);
end
```

The APP decoder assumes a polarization of the soft inputs that is inverse to that of the demodulator soft outputs. Therefore, you must change the sign of demodulated signal, demodSignal.

4
Display the error rate information.

```
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
```

    errorStats(1), errorStats(2))
    
## Algorithms

## See Also

This object implements the algorithm, inputs, and outputs described on the APP Decoder block reference page. The object properties correspond to the block parameters.
comm.ConvolutionalEncoder | comm.ViterbiDecoder | poly2trellis

Purpose

Description

Create APP decoder object with same property values with same property values

Syntax $\quad C=$ clone $(H)$
C = clone (H) creates a APPDecoder object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked $(H)$ returns the locked status, TF of the APPDecoder System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.APPDecoder.reset

Purpose Reset states of APP decoder object

## Syntax reset (H)

Description reset (H) resets the states of the APPDecoder object, H.

## Purpose

Decode convolutional code using the a posteriori probability method

## Syntax

Description
[LUD,LCD] = step(H,LU,LC)
[LUD, LCD] = step( $\mathrm{H}, \mathrm{LU}, \mathrm{LC}$ ) performs APP decoding. The input LU is
the sequence of log-likelihoods of encoder input data bits. The input LC is the sequence of log-likelihoods of encoded bits. Negative soft inputs are considered to be zeros and positive soft inputs are considered to be ones. The outputs, LUD and LCD, are updated versions of the input LU and LC sequences and are obtained based on information about the encoder. The inputs must be of the same data type, which can be double or single precision. The output data type is the same as the input data type. If the convolutional code uses an alphabet of $2^{\wedge} \mathrm{N}$ symbols, the LC and LCD vector lengths are multiples of N. If the decoded data uses an alphabet of $2^{\wedge} \mathrm{K}$ output symbols, the LU and LUD vector lengths are multiples of K .

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

Description

## Construction

Properties

## Purpose Add white Gaussian noise to input signal

The AWGNChannel object adds white Gaussian noise to a real or complex input signal. When the input uses a real-valued signal, this object adds real Gaussian noise and produces a real output signal. When the input uses a complex signal, this object adds complex Gaussian noise and produces a complex output signal.

When the inputs to the object have a variable number of channels, the EbNo, EsNo, SNR, BitsPerSymbol, SignalPower, SamplesPerSymbol, and Variance properties must be scalars, when applicable.

H = comm.AWGNChannel creates an additive white Gaussian noise (AWGN) channel System object, H. This object then adds white Gaussian noise to a real or complex input signal.

H = comm.AWGNChannel(Name, Value) creates an AWGN channel object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## NoiseMethod

Method to specify noise level
Select the method to specify the noise level as one of Signal to noise ratio (Eb/No) | Signal to noise ratio (Es/No)| Signal to noise ratio (SNR) | Variance. The default is Signal to noise ratio (Eb/No).

## EbNo

Energy per bit to noise power spectral density ratio (Eb/No)
Specify the Eb/No ratio in decibels. You can set this property to a numeric, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (Eb/No). The default is 10 . This property is tunable.

## EsNo

Energy per symbol to noise power spectral density ratio (Es/No)
Specify the Es/No ratio in decibels. You can set this property to a numeric, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (Es/No). The default is 10 . This property is tunable.

## SNR

Signal to noise ratio (SNR)
Specify the SNR value in decibels. You can set this property to a numeric, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (SNR). The default is 10 . This property is tunable.

## BitsPerSymbol

Number of bits in one symbol
Specify the number of bits in each input symbol. You can set this property to a numeric, positive, integer scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (Eb/No). The default is 1 bit.

## SignalPower

Input signal power in Watts
Specify the mean square power of the input signal in Watts. You can set this property to a numeric, positive, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (Eb/No), Signal to noise ratio (Es/No), or Signal to noise ratio (SNR). The default is 1 . The object assumes a nominal impedance of $1 \Omega$. This property is tunable.

## SamplesPerSymbol

Number of samples per symbol
Specify the number of samples per symbol. You can set this property to a numeric, positive, integer scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (Eb/No) or Signal to noise ratio (Es/No). The default is 1 .

## VarianceSource

Source of noise variance
Specify the source of the noise variance as one of Property | Input port. The default is Property. Set this property to Input port to specify the noise variance value using an input to the step method. Set this property to Property to specify the noise variance value using the Variance property. This property applies when you set the NoiseMethod property to Variance.

## Variance

Noise variance
Specify the variance of the white Gaussian noise. You can set this property to a numeric, positive, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Variance and the VarianceSource property to Property. The default is 1. This property is tunable.

## RandomStream

Source of random number stream
Specify the source of random number stream as one of Global stream | mt19937ar with seed. The default value of this property is Global stream.

When you set this property to Global stream, the object uses the current global random number stream for normally distributed random number generation.

When you set this property to mt19937ar with seed, the object uses the mt19937ar algorithm for normally distributed random number generation. In this scenario, when you call the reset method, the object re-initializes the random number stream to the value of the Seed property.

## Seed

Initial seed of mt19937ar random number stream
Specify the initial seed of a mt19937ar random number generator algorithm as a double-precision, real, nonnegative integer scalar. The default value of this property is 67 .

This property applies when you set the RandomStream property to mt19937ar with seed. For each call to the reset method, the object re-initialize the mt19937ar random number stream to the Seed value.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release

Create AWGN channel object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

| reset | Reset states of the System object |
| :--- | :--- |
| step | Add white Gaussian noise to <br> input signal |

## Examples Add White Gaussian noise to an 8-PSK signal

Modulate an 8-PSK signal, add White Gaussian noise, and plot the signal to observe the effects of noise. Then, compare the results to a multi-channel response.

Create a PSK Modulator System object.
hMod = comm.PSKModulator;

Modulate the signal by calling the step method of the PSK modulator.

```
modData = step(hMod, randi([0 hMod.ModulationOrder-1], 1000, 1));
```

Add White Gaussian noise to the modulated signal by passing the signal through an AWGN channel.

```
hAWGN = comm.AWGNChannel('EbNo', 15, ...
'BitsPerSymbol', log2(hMod.ModulationOrder));
```

Transmit the signal through the AWGN channel by call the step method of the AWGN channel.
channelOutput $=$ step(hAWGN, modData);
Plot the noiseless and noisy data using scatter plots to observe the effects of noise.

```
scatterplot(modData)
scatterplot(channelOutput)
```

Use a multi-channel input with greater noise by setting the EbNo value to 10 .
hAWGN.EbNo = 10;

Modulate the signal by calling the step method.
modData $=$ step(hMod, randi([0 hMod.ModulationOrder-1], 1000, 1)); modData $=$ reshape (modData, 500, 2);

Obtain the signal at the output of the channel by calling the step method.

```
channelOutput = step(hAWGN, modData);
```

Plot the channel output to see the effects that the noise has on the signal.

```
scatterplot(channelOutput(:))
```


## Process two consecutive inputs with different numbers of samples

Create the AWGN System object with the EbNo ratio set to a 1 -by-2 vector by typing the following syntax at the MATLAB command line.

```
h = comm.AWGNChannel('EbNo', [10 5]);
```

Process the first input signal by calling the step method. This object processes 128 samples per channel over two channels.
step(h, ones(128, 2));
Process the second input signal by calling the step method. Without being released, this object processes an additional 256 samples per channel over two channels.
step (h, ones(256, 2))
In this example, the number of samples each channels processes can change. Because the number of channels remains fixed, you can specify the EbNo property value as a vector.

## Process two consecutive inputs with different numbers of samples and channels

This example shows how the value of a tunable property can change between two inputs of different sizes, as long as the input is a scalar.

Create an AWGN Channel System object, with the NoiseMethod property set to Signal to noise ratio and the SNR property value to 20 .
h = comm.AWGNChannel( ...
'NoiseMethod', 'Signal to noise ratio (SNR)',...
'SNR', 20);
Process the first input signal by calling the step method. The input has one channel.
step(h, ones(20, 1));
Process the second input signal by calling the step method. The input has two channels, and a different number of samples per channel.
step(h, ones(90, 2))
In this example, the number of channels can change, but the SNR property value must be a scalar.

## Process signals with a noise variance input

This example shows the noise variance input as a scalar or a row vector, with a length equal to the number of channels of the current signal input.

Create an AWGN Channel System object, with the NoiseMethod property set to Variance and the VarianceSource property set to Input port.

```
h = comm.AWGNChannel( ...
    'NoiseMethod', 'Variance', ...
    'VarianceSource', 'Input port');
```

Process the data by calling the step method. The object processes 128 samples through two channels. The variance input is a $1-b y-2$ vector.

```
step(h, ones(128, 2), [0.1, 0.2]);
```

Process the data by calling the step method. The object processes 150 samples through five channels. The variance input is a scalar.

```
step(h, ones(150, 5), 1)
```


## Process signals using a self-contained random stream for repeatability

This example shows how to produce the same outputs, after reset, when using a self-contained random stream.

Create an AWGN Channel System object, with the NoiseMethod property set to Variance, the and the RandomStream property set to mt19937ar with seed, and the Seed property set to 99 .

```
h = comm.AWGNChannel( ...
    'NoiseMethod', 'Variance', ...
    'RandomStream', 'mt19937ar with seed', ...
    'Seed', 99);
```

Process the data by calling the step method.

```
y1 = step(h, ones(8, 2));
```

Reset the AWGN Channel System object by calling the reset method. This resets the random data stream to the initial seed of 99 .

```
reset(h);
```

Process the same data by calling the step method.
y2 = step(h, ones(8, 2));
Compare the two signals.

```
isequal(y1, y2)
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the AWGN Channel block reference page. The object properties correspond to the block parameters, except for:

- The block uses a random number generator based on the V5 RANDN (Ziggurat) algorithm and an initial seed, set with the Initial seed parameter to initialize the random number generator. Every time the system that contains the block is run, the block generates the same sequence of random numbers. Similarly, on the object, when you set the RandomStream property to mt19937ar with seed, you can generate reproducible numbers by resetting the object.
When you set the RandomStream property to Global stream, this object uses the MATLAB default random stream to generate random numbers. To generate reproducible numbers using this object, you can reset the MATLAB default random stream using the following code.

```
reset(RandStream.getGlobalStream)
```

For more information, see help for RandStream.

- Sometimes, the input to the step method is complex. In such cases, if you try to match the block and object's random generator and seed by setting the random stream of MATLAB, the random numbers do not appear in the same order.
The object creates the random data as follows:

$$
\text { noise }=\operatorname{randn}(\text { lengthInput }, 1)+1 i \times \operatorname{randn}(\text { lengthInput }, 1)
$$

The block creates random data as follows:
randData $=$ randn $(2 \times$ lengthInput, 1$)$
noise $=\operatorname{randData}(1: 2: \mathrm{end})+1 i \times \operatorname{randData}(2: 2: \mathrm{end})$

- The Symbol period block parameter corresponds to the SamplesPerSymbol property.
- The Variance from mask and Variance from port block parameter options of the Mode parameter correspond to the VarianceSource property.

See Also comm.BinarySymmetricChannel

Purpose Create AWGN channel object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a AWGNChannel object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs (H)
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ )
Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn outputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the AWGNChannel System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset states of the AWGNChannel System object |
| :--- | :--- |
| Syntax | reset (H) |
| Description $\quad$reset (H) resets the states of the AWGNChannel object, H. <br> If you set the RandomStream property of H to Global stream, the <br> reset method only resets the filters. If you set RandomStream to <br> mt19937ar with seed, the reset method not only resets the filters <br> but also reinitializes the random number stream to the value of the <br> Seed property. |  |

## Purpose Add white Gaussian noise to input signal

$$
\begin{array}{ll}
\text { Syntax } & Y=\operatorname{step}(H, X) \\
& Y=\operatorname{step}(H, X, V A R)
\end{array}
$$

$Y=\operatorname{step}(H, X)$ adds white Gaussian noise to input $X$ and returns the result in $Y$. Depending on the value of the FrameBasedProcessing property, input $X$ can be a double or single precision data type scalar, vector, or matrix with real or complex values..
$Y=\operatorname{step}(H, X, V A R)$ uses input VAR as the variance of the white Gaussian noise. This applies when you set the NoiseMethod property to Variance and the VarianceSource property to Input port. Input VAR can be a numeric, positive scalar or row vector with a length equal to the number of channels.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose

Description

## Construction

## Properties

Generate Barker code
The BarkerCode object generates Barker codes to perform synchronization. Barker codes are subsets of PN sequences. They are short codes, with a length at most 13, which have low-correlation sidelobes. A correlation sidelobe is the correlation of a codeword with a time-shifted version of itself.

H = comm. BarkerCode creates a Barker code generator System object, H , that generates a Barker code of a specified length.

H = comm.BarkerCode(Name, Value) creates a Barker code generator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Length

Length of generated code
Specify the length of the Barker code as a numeric, integer scalar in the set $\{1,2,3,4,5,7,11,13\}$. The default is 7 . The codes that the object generates for a specified length are listed in the following table:

| LengthBarker code |  |
| :---: | :---: |
| 1 | [-1] |
| 2 | $\left[\begin{array}{ll}-1 & 1\end{array}\right]$ |
| 3 | $\left[\begin{array}{lll}-1 & -1 & 1\end{array}\right]$ |
| 4 | $\left[\begin{array}{llll}-1 & -1 & 1 & -1\end{array}\right]$ |
| 5 | $\left[\begin{array}{llllll}-1 & -1 & -1 & 1 & -1\end{array}\right]$ |
| 7 | $\left[\begin{array}{llllllll}-1 & -1 & -1 & 1 & 1 & -1 & 1\end{array}\right]$ |
| 11 | $\left[\begin{array}{lllllllllllll}-1 & -1 & -1 & 1 & 1 & 1 & -1 & 1 & 1 & -1 & 1\end{array}\right]$ |
| 13 | $\left[\begin{array}{llllllllllllllll}-1 & -1 & -1 & -1 & -1 & 1 & 1 & -1 & -1 & 1 & -1 & 1 & -1\end{array}\right]$ |

## SamplesPerFrame

Number of output samples per frame
Specify the number of Barker code samples that the step method outputs as a numeric, integer scalar. The default is 1 . If you set this property to a value of $M$, then the step method outputs $M$ samples of a Barker code sequence of length $N . N$ represents the length of the code that you specify in the Length property.

## OutputDataType

Data type of output
Specify the output data type as one of double | int8. The default is double.
$\left.\begin{array}{lll}\text { Methods } & \text { clone } & \begin{array}{l}\text { Create Barker code generator } \\ \text { object with same property values }\end{array} \\ \text { getNumInputs } & \begin{array}{l}\text { Number of expected inputs to } \\ \text { step method }\end{array} \\ \text { getNumOutputs } & \begin{array}{l}\text { Number of outputs from step } \\ \text { method }\end{array} \\ \text { isLocked } & \begin{array}{l}\text { Locked status for input attributes } \\ \text { and nontunable properties }\end{array} \\ \text { release } & \begin{array}{l}\text { Allow property value and input } \\ \text { characteristics changes }\end{array} \\ \text { Reset states of Barker code } \\ \text { generator object } \\ \text { Generate Barker code }\end{array}\right\}$

# Algorithms 

This object implements the algorithm, inputs, and outputs described on the Barker Code Generator block reference page. The object properties correspond to the block parameters, except:

- The block Sample time parameter does not have a corresponding property.
- The object only implements frame based outputs.


## See Also

comm.HadamardCode | comm.OVSFCode

## comm.BarkerCode.clone

Purpose Create Barker code generator object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a BarkerCode object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.BarkerCode.getNumInputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.BarkerCode.getNumOutputs

Purpose $\quad$ Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

# Purpose <br> Locked status for input attributes and nontunable properties 

Syntax TF $=$ isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the BarkerCode System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset states of Barker code generator object |
| :--- | :--- |
| Syntax | reset $(H)$ |
| Description | reset $(H)$ resets the states of the BarkerCode object, H. |

Purpose Generate Barker code

## Syntax $\quad Y=\operatorname{step}(H)$

Description $\quad Y=\operatorname{step}(H)$ outputs a frame of the Barker code in column vector $Y$. You specify the frame length with the SamplesPerFrame property. The output code is in a bi-polar format with 0 and 1 mapped to 1 and -1 , respectively.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Decode data using BCH decoder
Description The BCHDecoder object recovers a binary message vector from abinary BCH codeword vector. For proper decoding, the codeword andmessage length values in this object must match the properties in thecorresponding BCH Encoder block.
Construction H = comm.BCHDecoder creates a BCH decoder System object, H, that performs BCH decoding.

H = comm.BCHDecoder(Name, Value) creates a BCH decoder object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties <br> CodewordLength

Codeword length
Specify the codeword length of the BCH code as a double-precision, positive, integer scalar. The default is 15 . The values of the CodewordLength and MessageLength properties must produce a valid narrow-sense BCH code. For a full-length BCH code the value of the this property must take the form $2^{M}-1 . M$ is an integer, $3 \leq M \leq 16$, that corresponds to the degree of the primitive polynomial that you specify with PrimitivePolynomialSource and PrimitivePolynomial. If the this property is less than
$2^{M}-1$, the object assumes a shortened code.

## MessageLength

Message length
Specify the message length as a double-precision, positive, integer scalar. The default is 5 . The values of the CodewordLength and MessageLength properties must produce a valid narrow-sense BCH code.

## PrimitivePolynomialSource

Source of primitive polynomial
Specify the source of the primitive polynomial as one of Auto | Property. The default is Auto. When you set this property to Auto, the object uses a primitive polynomial of degree $M=$ ceil (log2(CodewordLength+1)). The result of fliplr(de2bi(primpoly(M))), sets the value for this polynomial. Set this property to Property to specify a polynomial using the PrimitivePolynomial property.

## PrimitivePolynomial

Primitive polynomial
Specify the primitive polynomial of order M, that defines the finite Galois field GF(2) as a double-precision, binary row vector with the coefficients of the polynomial in order of descending powers. This property applies when you set the PrimitivePolynomialSource property to Property. The default is fliplr(de2bi(primpoly(4)))
$=\left[\begin{array}{lllll}1 & 0 & 0 & 1 & 1\end{array}\right]$, which corresponds to the polynomial $x^{4}+x+1$.

## GeneratorPolynomialSource

Source of generator polynomial
Specify the source of the generator polynomial as one of Auto | Property. The default is Auto. When you set this property to Auto, the object chooses the generator polynomial automatically. The object calculates the generator polynomial based on the value of the PrimitivePolynomialSource property. When you set the PrimitivePolynomialSource property to Auto the object calculates the generator polynomial as bchgenpoly(CodewordLength $+S L$,MessageLength $+S L$ ). When you set the PrimitivePolynomialSource property to Property, the object computes generator polynomial as bchgenpoly (CodewordLength+SL, MessageLength+SL, PrimitivePolynomial). In both cases, $S L=$
( $2^{M}-1$ )-CodewordLength is the shortened length. and $M$ is the degree of the primitive polynomial that you specify with

PrimitivePolynomialSource and PrimitivePolynomial. Set this property to Property to specify a generator polynomial using the GeneratorPolynomial property.

## GeneratorPolynomial

Generator polynomial
Specify the generator polynomial as a binary, double-precision, row vector or as a binary Galois field row vector that represents the coefficients of the generator polynomial in order of descending powers. You must use CodewordLength-MessageLength+1 as the length of the generator polynomial. This property applies when you set the GeneratorPolynomialSource property to Property. The default is the result of bchgenpoly((15,5, [], 'double')), which corresponds to a 15,5 code.

When you use this object to generate code, you must set the generator polynomial to a binary, double precision row vector.

## CheckGeneratorPolynomial

Enable generator polynomial checking
Set this property to true to perform a generator polynomial check the first time you call the step method. The default is true. This check verifies that $x^{\text {CodewordLength }}+1$ is divisible by the generator polynomial specified in the GeneratorPolynomial property. For larger codes, disabling the check reduces processing time. As a best practice, perform the check at least once before setting this property to false. This property applies when you set the GeneratorPolynomialSource property to Property.

## PuncturePatternSource

Source of puncture pattern
Specify the source of the puncture pattern as one of None | Property. The default is None. Set this property to None to disable puncturing. Set this property to Property to decode punctured codewords based on a puncture pattern vector you specify in the PuncturePattern property.

## PuncturePattern

Puncture pattern vector
Specify the pattern that the object uses to puncture the encoded data as a double-precision, binary, column vector of length CodewordLength-MessageLength. Zeros in the puncture pattern vector indicate the position of the parity bits that the object punctures or excludes from each codeword. This property applies when you set the PuncturePatternSource property to Property. The default is [ones $(8,1)$; zeros $(2,1)]$.

## ErasuresInputPort

Enable erasures input
Set this property to true to specify a vector of erasures as a step method input. The default is false. The erasures vector is a double-precision or logical, binary, column vector that indicates which bits of the input codewords to erase or ignore. The length of the vector must equal the encoded data input, (that is, the length must be an integer multiple of (CodewordLength - number of punctures)). Values of 1 in the erasures vector correspond to erased bits in the same position of the (possibly punctured) input codewords. Set the this property to false to disable erasures.

## NumCorrectedErrorsOutputPort

Output number of corrected errors
Set this property to true so that the step method outputs the number of corrected errors. The default is true.

| Methods clone | Create BCH decoder object with <br> same property values |
| :--- | :--- | :--- |
| getNumInputs | Number of expected inputs to <br> step method |

getNumOutputs<br>isLocked<br>release<br>step

## Number of outputs from step

 methodLocked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Decode data using a BCH decoder

## Examples

## Algorithms

Transmit a BCH-encoded, 8-DPSK-modulated bit stream through an AWGN channel, then demodulate, decode, and count errors.

```
hEnc = comm.BCHEncoder;
hMod = comm.DPSKModulator('BitInput',true);
hChan = comm.AWGNChannel(...
    'NoiseMethod','Signal to noise ratio (SNR)','SNR',10);
hDemod = comm.DPSKDemodulator('BitOutput',true);
hDec = comm.BCHDecoder;
hError = comm.ErrorRate('ComputationDelay',3);
for counter = 1:20
    data = randi([0 1], 30, 1);
    encodedData = step(hEnc, data);
    modSignal = step(hMod, encodedData);
    receivedSignal = step(hChan, modSignal);
    demodSignal = step(hDemod, receivedSignal);
    receivedBits = step(hDec, demodSignal);
    errorStats = step(hError, data, receivedBits);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```

This object implements the algorithm, inputs, and outputs described on the BCH Decoder block reference. The object properties correspond to the block parameters.

## comm.BCHDecoder

See Also comm.BCHEncoder | comm.RSDecoder

Purpose Create BCH decoder object with same property values
Syntax $\quad C=$ clone $(H)$
 values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.BCHDecoder.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked $(H)$ returns the locked status, TF of the BCHDecoder System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Decode data using a BCH decoder<br>Syntax<br>Y = step (H,X)<br>[ $\mathrm{Y}, \mathrm{ERR}$ ] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$<br>Y = step( $\mathrm{H}, \mathrm{X}$, ERASURES)

## Description

$Y=\operatorname{step}(H, X)$ decodes input binary codewords in $X$ using a (CodewordLength,MessageLength) BCH decoder with the corresponding narrow-sense generator polynomial. The step method returns the estimated message in Y. This syntax applies when you set the NumCorrectedErrorsOutputPort property to false. The input X must be a numeric or logical column vector. $X$ must have an integer multiple of (CodewordLength - number of punctures) elements. Specify the number of punctures with the PuncturePatternSource and PuncturePattern properties. Each group of (CodewordLength - number of punctures) input elements represents one codeword to be decoded. The length of the output decoded data vector, Y , is an integer multiple of the message length specified in the MessageLength property.
[ $\mathrm{Y}, \mathrm{ERR}$ ] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$ returns the number of corrected errors in output ERR when you set the NumCorrectedErrorsOutputPort property to true. A non- negative value in the ith element of the ERR output vector denotes the number of corrected errors in the $i$-th input codeword. A value of -1 in the $i$-th element of the ERR output indicates that a decoding error occurred for the ith input codeword. A decoding error occurs when an input codeword has more errors than the error correction capability of the BCH code.

Y = step( $\mathrm{H}, \mathrm{X}$, ERASURES) uses ERASURES as the erasures pattern input when you set the ErasuresInputPort property to true. The object decodes the binary encoded data input, X , and treats as erasures the bits of the input codewords specified by the binary column vector, ERASURES. The length of ERASURES must equal the length of $X$, and its elements must be of data type double or logical. Values of 1 in the erasures vector correspond to erased bits in the same position of the (possibly punctured) input codewords.


#### Abstract

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.


## Purpose Encode data using BCH encoder

Description

The BCHEncoder object creates a BCH code with specified message and codeword lengths.

## Construction <br> H = comm. BCHEncoder creates a BCH encoder System object, H, that performs BCH encoding. <br> H = comm.BCHEncoder(Name, Value) creates a BCH encoder object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties

## CodewordLength

Codeword length
Specify the codeword length of the BCH code as a double-precision, positive, integer scalar. The default is 15. The values of the CodewordLength and MessageLength properties, must produce a valid narrow-sense BCH code. For a full-length BCH code the value of the this property must use the form $2^{M}-1$. In this case, $M$ is an integer, and $3 \leq M \leq 16$ corresponds to the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties. If the this property is less than $2^{M}-1$, the object assumes a shortened code form.

## MessageLength

Message length
Specify the message length as a double-precision, positive, integer scalar. The values of the CodewordLength and MessageLength properties must produce a valid narrow-sense BCH code. The default is 5 .

## PrimitivePolynomialSource

Source of primitive polynomial
Specify the source of the primitive polynomial as one of Auto | Property. The default is Auto. When you set this property to Auto, the object uses a primitive polynomial of degree $M=$ ceil (log2 (CodewordLength+1)). The result of fliplr(de2bi(primpoly (M))) sets the value for this polynomial. Set this property to Property to specify a polynomial using the PrimitivePolynomial property.

## PrimitivePolynomial

Primitive polynomial
Specify the primitive polynomial of order M, that defines the finite Galois field GF(2). Use a double-precision, binary row vector with the coefficients of the polynomial in order of descending powers. This property applies when you set the PrimitivePolynomialSource property to Property. The default is fliplr(de2bi(primpoly(4))) = $\left.\begin{array}{llll}1 & 0 & 0 & 1\end{array}\right]$, which corresponds to the polynomial $x^{4}+x+1$.

## GeneratorPolynomialSource

Source of generator polynomial
Specify the source of the generator polynomial as one of Auto | Property. The default is Auto. When you set this property to Auto, the object chooses the generator polynomial automatically. The object computes the generator polynomial based on the value of the PrimitivePolynomialSource property. When you set the PrimitivePolynomialSource property to Auto the object computes the generator polynomial as bchgenpoly(CodewordLength $+S L$,MessageLength $+S L$ ). When you set the PrimitivePolynomialSource property to 'Property', the object computes generator polynomial as bchgenpoly (CodewordLength $+S L$, MessageLength $+S L$, PrimitivePolynomial). In both cases, $S L=$ ( $2^{M}-1$ )-CodewordLength is the shortened length. $M$
indicates the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties. Set this property to Property to specify a generator polynomial using the GeneratorPolynomial property.

## GeneratorPolynomial

## Generator polynomial

Specify the generator polynomial for encoding as a binary, double-precision row vector or as a binary Galois row vector that represents the coefficients of the generator polynomial in order of descending powers. The length of the generator polynomial requires a value of CodewordLength-MessageLength +1 . This property applies when you set the GeneratorPolynomialSource property to Property. The default is the result of bchgenpoly(15,5, [], 'double'), which corresponds to a (15,5) code.

## CheckGeneratorPolynomial

Enable generator polynomial checking
Set this property to true to perform a generator polynomial check the first time you call the step method. The default is true. This check verifies that $x^{\text {CodewordLength }}+1$ is divisible by the generator polynomial specified in the GeneratorPolynomial property. For larger codes, disabling the check reduces processing time. As a best practice, perform the check at least once before setting this property to false. This property applies when you set the GeneratorPolynomialSource property to Property.

## PuncturePatternSource

Source of puncture pattern
Specify the source of the puncture pattern as one of None | Property. The default is None. Set this property to None, to disable puncturing. Set this property to Property to decode
punctured codewords. This decoding is based on a puncture pattern vector you specify in the PuncturePattern property.

## PuncturePattern

Puncture pattern vector
Specify the pattern that the object uses to puncture the encoded data. Use a double-precision, binary, column vector of length CodewordLength-MessageLength. Zeros in the puncture pattern vector indicate the position of the parity bits that the object punctures or excludes from each codeword. This property applies when you set the PuncturePatternSource property to Property. The default is [ones $(8,1)$; zeros $(2,1)]$.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| step |  |

Create BCH encoder object with same property values

Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Encode data using a BCH encoder

Examples Transmit a BCH-encoded, 8-DPSK-modulated bit stream through an AWGN channel, then demodulate, decode, and count errors.

```
hEnc = comm.BCHEncoder;
hMod = comm.DPSKModulator('BitInput',true);
hChan = comm.AWGNChannel(...
    'NoiseMethod','Signal to noise ratio (SNR)','SNR',10);
hDemod = comm.DPSKDemodulator('BitOutput',true);
```

```
hDec = comm.BCHDecoder;
hError = comm.ErrorRate('ComputationDelay',3);
for counter = 1:20
    data = randi([0 1], 30, 1);
    encodedData = step(hEnc, data);
    modSignal = step(hMod, encodedData);
    receivedSignal = step(hChan, modSignal);
    demodSignal = step(hDemod, receivedSignal);
    receivedBits = step(hDec, demodSignal);
    errorStats = step(hError, data, receivedBits);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the BCH Encoder block reference page. The object properties correspond to the block parameters.
comm. BCHDecoder | comm.RSEncoder

Purpose Create BCH encoder object with same property values
Syntax $\quad C=$ clone $(H)$
Description $\quad C=$ clone $(H)$ creates a $B C H E n c o d e r$ object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.BCHEncoder.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked ( H ) returns the locked status, TF of the BCHEncoder System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Encode data using a BCH encoder

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description
$Y=\operatorname{step}(H, X)$ encodes input binary data, $X$, using a (CodewordLength,MessageLength) BCH encoder with the corresponding narrow-sense generator polynomial and returns the result in vector $Y$. Input $X$ must be a numeric or logical column vector with length equal to an integer multiple of the message length stored in the MessageLength property. A group of MessageLength input elements represents one message word to be encoded. The length of the encoded data output vector, Y , is an integer multiple of (CodewordLength number of punctures). You specify the number of punctures with the PuncturePatternSource and PuncturePattern properties.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Convert vector of bits to vector of integers |
| :---: | :---: |
| Description | The BitToInteger object maps groups of bits in the input vector to integers in the output vector. |
| Construction | H = comm.BitToInteger creates a bit-to-integer converter System object, H , that maps a vector of bits to a corresponding vector of integer values. |
|  | H = comm.BitToInteger(Name, Value) creates a bit-to-integer converter object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value 1,...,NameN, ValueN). |
|  | H = comm.BitToInteger (NUMBITS, Name, Value) creates a bit-to-integer converter System object, H This object has the BitsPerInteger property set to NUMBITS and the other specified properties set to the specified values. |
| Properties | BitsPerInteger |
|  | Number of bits per integer |
|  | Specify the number of input bits that the object maps to each output integer. You can set this property to a scalar integer between 1 and 32. The default is 3 . |
|  | MSBFirst |
|  | Assume first bit of input bit words is most significant bit |
|  | Set this property to true to indicate that the first bit of the input bit words is the most significant bit (MSB). The default is true. You can set this property to false to indicate that the first bit of the input bit words is the least significant bit (LSB). |

## SignedIntegerOutput

Output signed integers

Set this property to true to generate signed integer outputs. The default is false. You can set this property to false to generate unsigned integer outputs.

When you set this property to false, the output values are integers between 0 and $\left(2^{N}\right)-1$. In this case, $N$ is the value you specified in the BitsPerInteger property.

When you set this property to true, the output values are integers between - $\left(2^{(N-1)}\right)$ and $\left(2^{(N-1)}\right)-1$.

## OutputDataType

Data type of output
Specify the output data type. The default is Full precision.
When you set the SignedIntegerOutput property to false, set this property as one of Full precision | Smallest integer | Same as input | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32.

When you set this property to Same as input, and the input data type is numeric or fixed-point (fi object), the output data has the same type as the input data.

When the input signal is an integer data type, you must have a Fixed-Point Designer user license to use this property in Smallest unsigned integer or Full precision mode.

When you set the SignedIntegerOutput property to true, specify the output data type as one of Full precision | Smallest integer | double | single | int8 | int16 | int32.

When you set this property to Full precision, the object determines the output data type based on the input data type. If the input data type is double or single precision, the output data has the same type as the input data. Otherwise, the property determines the output data type in the same way as when you set this property to Smallest unsigned integer.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| step |  |

Create bit to integer converter object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Convert vector of bits to vector of integers

## Examples Convert randomly generated 4-bit words to integers.

```
hBitToInt = comm.BitToInteger(4);
% Generate three 4-bit words
    bitData = randi([0 1],3*hBitToInt.BitsPerInteger,1);
    intData = step(hBitToInt,bitData);
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the Bit To Integer Converter block reference page. The object properties correspond to the block parameters.
comm.IntegerToBit | bi2de | bin2dec

Purpose Create bit to integer converter object with same property values

## Syntax <br> C = clone( H )

Description
$C=$ clone ( $H$ ) creates a BitToInteger object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.BitTolnteger.getNumInputs

Purpose $\quad$ Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description
$N=$ getNumInputs $(H)$ returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.BitTolnteger.getNumOutputs

Purpose $\quad$ Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.BitToInteger.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF = isLocked (H) |
| Description | TF = isLocked (H) returns the locked status, TF of the BitToInteger <br> System object. |
| The isLocked method returns a logical value that indicates whether <br> input attributes and nontunable properties for the object are locked. The <br> object performs an internal initialization the first time the step method <br> is executed. This initialization locks nontunable properties and input <br> specifications, such as dimensions, complexity, and data type of the <br> input data. After locking, the isLocked method returns a true value. |  |

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description
release (H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose
Convert vector of bits to vector of integers
Syntax $\quad Y=\operatorname{step}(H, X)$
Description
$Y=\operatorname{step}(H, X)$ converts binary input, $X$, to corresponding integers, $Y$. The input must be a scalar or a column vector and the data type can be numeric, numerictype $(0,1)$, or logical. The length of input $X$ must be an integer multiple of the value you specify in the BitsPerInteger property. The object outputs a column vector with a length equal to length(X)/BitsPerInteger. When you set the SignedIntegerOutput property to false, the object maps each group of bits to an integer between 0 and ( $2^{\text {BitsPerInteger })}$-1. A group of bits contains $N$ bits, where $N$ is the value of the BitsPerInteger property. If you set the SignedIntegerOutput property to true, the object maps each group of BitsPerInteger bits to an integer between - ( $2^{(\text {BitsPerInteger-1) })}$ and ( $2^{(\text {BitsPerInteger-1) })}$ ) 1 .

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Introduce binary errors

Description The BinarySymmetricChannel object introduces binary errors to the signal transmitted through this channel.

## Construction

H = comm. BinarySymmetricChannel creates a binary symmetric channel System object, $H$, that introduces binary errors to the input signal with a prescribed probability.

H = comm.BinarySymmetricChannel(Name, Value) creates a binary symmetric channel object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties

## ErrorProbability

Probability of binary error
Specify the probability of a binary error as a scalar with a value between 0 and 1. The default is 0.05 .

## ErrorVectorOutputPort

Enable error vector output
When you set this property to true, the step method outputs an error signal, ERR. This error signal, in vector form, indicates where errors were introduced in the input signal, $X$. A value of 1 at the $i$-th element of ERR indicates that an error was introduced at the $i$-th element of X. Set the property to false if you do not want the ERR vector at the output of the step method. The default is true.

## OutputDataType

Data type of output
Specify output data type as one of double | logical. The default is double.

| Methods | clone |
| :--- | :--- |
| getNumInputs |  |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| step |  |

Create binary symmetric channel object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Introduce binary errors

## Examples

## Algorithms

Add binary errors with a probability of 0.2 to a binary input signal.

```
H = comm.BinarySymmetricChannel('ErrorProbability',0.2);
data = randi([0 1], 10, 1);
[noisyData, err] = step(H, data);
[data noisyData err]
```

This object implements the algorithm, inputs, and outputs described on the Binary Symmetric Channel block reference page. The object properties correspond to the block parameters, except:This object uses the MATLAB default random stream to generate random numbers. The block uses a random number generator based on the V5 RANDN (Ziggurat) algorithm. An initial seed, set with the Initial seed parameter initializes the random number generator. For every system run that contains the block, the block generates the same sequence of random numbers. To generate reproducible numbers using this object, you can reset the MATLAB default random stream using the following code.
reset (RandStream.getGlobalStream)
For more information, see help for RandStream.

See Also comm. AWGNChannel

## comm.BinarySymmetricChannel.clone

Purpose
Syntax $\quad C=$ clone $(H)$
Description
$\mathrm{C}=$ clone $(\mathrm{H})$ creates a BinarySymmetricChannel object C , with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.BinarySymmetricChannel.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.BinarySymmetricChannel.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the BinarySymmetricChannel System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Introduce binary errors

| Syntax | $Y=\operatorname{step}(H, X)$ |
| :--- | :--- |
|  | $[Y, E R R]=\operatorname{step}(H, X)$ |

Description
$Y=\operatorname{step}(H, X)$ adds binary errors to the input signal $X$ and returns the modified signal, Y . The input signal can be a vector or matrix with numeric, logical, or fixed-point (fi objects) data type elements. The step method output, $Y$, has the same dimensions as the input, $X$. If $X$ input contains a non-binary value, $V$, the object considers it to be 1 when $\mathbf{a b s}(V)>0$. This syntax applies when you set the ErrorVectorOutputPort property to false.
[ $\mathrm{Y}, \mathrm{ERR}$ ] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$ returns the error signal vector, ERR. A value of 1 at the $i$-th element of ERR indicates that an error was introduced at the $i$-th element of X. The outputs, Y and ERR, have the same dimensions as the input, $X$. This syntax applies when you set the ErrorVectorOutputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

# Purpose Deinterleave input symbols using permutation vector <br> Description <br> <br> \section*{Construction} <br> <br> \section*{Construction} <br> H = comm. BlockDeinterleaver creates a block deinterleaver System object, H . This object restores the original ordering of a sequence that was interleaved using the block interleaver System object. <br> H = comm. BlockDeinterleaver(Name, Value) creates object, H, with the specified property set to the specified value. 

## Properties

## PermutationVector

## Permutation vector

Specify the mapping used to permute the input symbol as a column vector of integers. The default is $[5 ; 4 ; 3 ; 2 ; 1]$. The mapping is a column vector of integers where the number of elements is equal to the length, $N$, of the input to the step method. Each element must be an integer, between 1 and $N$, with no repeated values.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked

Create block deinterleaver object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

release<br>step

Allow property value and input characteristics changes
Deinterleave input symbols using permutation vector

Examples Interleave and deinterleave data.

```
hInt = comm.BlockInterleaver([3 4 1 2]');
hDeInt = comm.BlockDeinterleaver([3 4 1 2]');
data = randi(7, 4, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence, and restored
[data, intData, deIntData]
```

Interleave and deinterleave data with random interleaver.

```
permVec = randperm(7)'; % Random permutation vector
    hInt = comm.BlockInterleaver(permVec);
    hDeInt = comm.BlockDeinterleaver(permVec);
    data = randi(9, 7, 1);
    intData = step(hInt, data);
    deIntData = step(hDeInt, intData);
    % compare the original sequence, interleaved sequence, and restored
    % sequence
    [data, intData, deIntData]
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the General Block Deinterleaver block reference page. The object properties correspond to the block parameters.

See Also

comm.BlockInterleaver | comm.MatrixDeinterleaver

Purpose Create block deinterleaver object with same property values
Syntax $\quad C=$ clone $(H)$
Description $\quad C=$ clone $(H)$ creates a BlockDeinterleaver object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.BlockDeinterleaver.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the BlockDeinterleaver System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Deinterleave input symbols using permutation vector

## Syntax <br> Y $=\operatorname{step}(H, X)$

Description
$Y=\operatorname{step}(H, X)$ restores the original ordering of the sequence, $X$, that was interleaved using a block interleaver. The step method forms the output, Y, based on the mapping specified by the PermutationVector property as Output(PermutationVector $(k)$ )=Input $(k)$, for $k=1: N$, where $N$ is the length of the permutation vector. The input X must be a column vector of the same length, $N$. The data type of X can be numeric, logical, or fixed-point (fi objects). $Y$ has the same data type as $X$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose

Permute input symbols using permutation vector

Construction $H=$ comm.BlockInterleaver creates a block interleaver System object, $H$ This object permutes the symbols in the input signal based on a permutation vector.

H = comm.BlockInterleaver(Name, Value) creates object, H, with specified property set to the specified value.

## Properties

## PermutationVector

## Permutation vector

Specify the mapping used to permute the input symbols as an integer column vector. The default is $[5 ; 4 ; 3 ; 2 ; 1]$. The number of elements of the permutation vector property must equal the length of the input vector. The PermutationVector property indicates the indices, in order, of the input elements that form the output vector. The relationship Output $(k)=\operatorname{Input}($ PermutationVector $(k))$ describes this order. Each integer, $k$, must be between 1 and $N$, where $N$ is the number of elements in the permutation vector. The elements in the PermutationVector property must be integers between 1 and $N$ with no repetitions.

Methods<br>clone<br>getNumInputs<br>getNumOutputs

Create block interleaver object with same property values

Number of expected inputs to step method

Number of outputs from step method

| isLocked | Locked status for input attributes <br> and nontunable properties |
| :--- | :--- |
| release | Allow property value and input <br> characteristics changes |
| step | Permute input symbols using a <br> permutation vector |

## Examples Interleave and deinterleave data.

```
hInt = comm.BlockInterleaver([[3 4 4 1 2]');
hDeInt = comm.BlockDeinterleaver([[3 4 1 2]');
data = randi(7, 4, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence, and restored
[data, intData, deIntData]
```

Interleave and deinterleave data with random interleaver.

```
permVec = randperm(7)'; % Random permutation vector
    hInt = comm.BlockInterleaver(permVec);
    hDeInt = comm.BlockDeinterleaver(permVec);
    data = randi(9, 7, 1);
    intData = step(hInt, data);
    deIntData = step(hDeInt, intData);
    % compare the original sequence, interleaved sequence, and restored
    % sequence
    [data, intData, deIntData]
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the General Block Interleaver block reference page. The object properties correspond to the block parameters.

[^4]
# Purpose Create block interleaver object with same property values 

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a BlockInterleaver object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.BlockInterleaver.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the BlockInterleaver System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Permute input symbols using a permutation vector

## Syntax <br> Y = step(H,X)

$Y=\operatorname{step}(H, X)$ permutes input sequence, $X$, and returns interleaved sequence, $Y$. The step method forms the output $Y$, based on the mapping defined by the Permutationvector property as Output $(k)=\operatorname{Input}($ PermutationVector $(k)$ ), for $k=1: N$, where $N$ is the length of the PermutationVector property. The input X must be a column vector of length $N$. The data type of $X$ can be numeric, logical, or fixed-point (fi objects). Y has the same data type as X .

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose <br> Description

## Construction

Demodulate using BPSK method

The BPSKDemodulator object demodulates a signal that was modulated using the binary phase shift keying method. The input is a baseband representation of the modulated signal.

H = comm.BPSKDemodulator creates a demodulator System object, H, that demodulates the input signal using the binary phase shift keying (BPSK) method.

H = comm.BPSKDemodulator(Name,Value) creates a BPSK demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.BPSKDemodulator(PHASE,Name, Value) creates a BPSK demodulator object, H , with the PhaseOffset property set to PHASE, and the other specified properties set to the specified values.

## Properties

## PhaseOffset

Phase of zeroth point of constellation
Specify the phase offset of the zeroth point of the constellation, in radians, as a finite, real scalar. The default is 0 .

## DecisionMethod

Demodulation decision method
Specify the decision method the object uses as one of Hard decision | Log-likelihood ratio | Approximate log-likelihood ratio. The default is Hard decision.

## VarianceSource

Source of noise variance
Specify the source of the noise variance as one of Property | Input port. The default is Property. This property applies when
you set the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio.

## Variance

Noise variance
Specify the variance of the noise as a nonzero, real scalar. The default is 1 . If this value is very small (i.e., SNR is very high), log-likelihood ratio (LLR) computations can yield Inf or - Inf. This variance occurs because the LLR algorithm computes the exponential of very large or very small numbers using finite precision arithmetic. As a best practice in such cases, use approximate LLR because this option's algorithm does not compute exponentials. This property applies when you set the VarianceSource property to Property. This property is tunable.

## OutputDataType

## Data type of output

Specify the output data type as one of Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32 | logical. The default is Full precision. This property applies only when you set the DecisionMethod property to Hard decision. Thus, when you set the OutputDataType property to Full precision, and the input data type is single or double precision, the output data has the same data type as the input. If the input data is of a fixed-point type, then the output data type behaves as if you had set the OutputDataType property to Smallest unsigned integer. If you set the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is the same as that of the input. In this case, that data type can only be single or double precision.

When the input signal is an integer data type, you must have a Fixed-Point Designer user license to use this property in Smallest unsigned integer or Full precision mode.

## Fixed-Point Properties

## DerotateFactorDataType

Data type of derotate factor
Specify the derotate factor data type as one of Same word length as input | Custom. The default is Same word length as input. This property applies when you set the DecisionMethod property to Hard decision. The object uses the derotate factor in the computations only when certain conditions exist. The step method input must be of a fixed-point type, and the PhaseOffset property must have a value that is not a multiple of $\pi / 2$.

## CustomDerotateFactorDataType

Fixed-point data type of derotate factor
Specify the derotate factor fixed-point type as an unscaled, numerictype object with a Signedness of Auto. The default is numerictype([],16). This property applies when you set the DecisionMethod property to Hard decision and the DerotateFactorDataType property to Custom.

| Methods | clone | Create BPSK demodulator object <br> with same property values |
| :--- | :--- | :--- |
| constellation | getNumInputs | Calculate or plot ideal signal <br> constellation |
|  | getNumOutputs | Number of expected inputs to <br> step method |
| isLocked | Number of outputs from step <br> method |  |
|  | Locked status for input attributes <br> and nontunable properties |  |


| release | Allow property value and input <br> characteristics changes |
| :--- | :--- |
| step | Demodulate using BPSK method |

Examples Modulate and demodulate a signal using BPSK modulation.

```
    hMod = comm.BPSKModulator('PhaseOffset',pi/2);
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
        'Signal to noise ratio (SNR)','SNR',15);
    hDemod = comm.BPSKDemodulator('PhaseOffset',pi/2);
% Create an error rate calculator
    hError = comm.ErrorRate;
    for counter = 1:100
        % Transmit a 50-symbol frame
        data = randi([0 1],50,1);
        modSignal = step(hMod, data);
        noisySignal = step(hAWGN, modSignal);
        receivedData = step(hDemod, noisySignal);
        errorStats = step(hError, data, receivedData);
    end
    fprintf('Error rate = %f\nNumber of errors = %d\n', ...
        errorStats(1), errorStats(2))
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the BPSK Demodulator Baseband block reference page. The object properties correspond to the block parameters.

comm.BPSKModulator | comm.PSKDemodulator

Purpose Create BPSK demodulator object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a BPSKDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.BPSKDemodulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax $\quad y=$ constellation $(h)$ constellation(h)

Description $y=$ constellation( $h$ ) returns the numerical values of the constellation.
constellation(h) generates a constellation plot for the object.
Examples Calculate Ideal Signal Constellation for comm.BPSKDemodulator

Create a comm.BPSKDemodulator System object, and then calculate its ideal signal constellation.

Create a comm. BPSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.BPSKDemodulator

Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.BPSKDemodulator

Create a comm.BPSKDemodulator System object, and then plot the ideal signal constellation.

Create a comm.BPSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.BPSKDemodulator
Plot the ideal signal constellation by calling the constellation method. constellation(h)

## comm.BPSKDemodulator.getNuminputs

## Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.BPSKDemodulator.getNumOutputs

Purpose $\quad$ Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.BPSKDemodulator.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked $(H)$ |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the <br> BPSKDemodulator System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.BPSKDemodulator.release

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose <br> Demodulate using BPSK method

Syntax
Y $=\operatorname{step}(H, X)$
Y $=\operatorname{step}(H, X, V A R)$
$Y=\operatorname{step}(H, X)$ demodulates input data, $X$, with the BPSK demodulator System object, H, and returns Y. Input X must be a scalar or a column vector with double or single precision data type. When you set the DecisionMethod property to Hard decision, the data type of the input can also be signed integer, or signed fixed point (fi objects).

Y = step ( $\mathrm{H}, \mathrm{X}, \mathrm{VAR}$ ) uses soft decision demodulation and noise variance VAR. This syntax applies when you set the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio and the VarianceSource property to Input port. The data type of input VAR must be double or single precision.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Modulate using BPSK method

Description

## Construction

## Properties

The BPSKModulator object modulates using the binary phase shift keying method. The output is a baseband representation of the modulated signal.

H = comm.BPSKModulator creates a modulator System object, H, that modulates the input signal using the binary phase shift keying (BPSK) method.

H = comm.BPSKModulator(Name, Value) creates a BPSK modulator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.BPSKModulator(PHASE,Name, Value) creates a BPSK modulator object, H. The object's PhaseOffset property is set to PHASE, and the other specified properties are set to the specified values.

## PhaseOffset

Phase of zeroth point of constellation
Specify the phase offset of the zeroth point of the constellation, in radians, as a finite, real scalar. The default is 0 .

## OutputDataType

Data type of output
Specify the output data type as one of double | single | Custom. The default is double.

## Fixed-Point Properties

## CustomOutputDataType

Fixed-point data type of output
Specify the output fixed-point type as a numerictype object with a Signedness of Auto. The default is numerictype ([],16). This
property applies when you set the OutputDataType property to Custom.

| Methods | clone | Create BPSK modulator object <br> with same property values |
| :--- | :--- | :--- |
| constellation | Calculate or plot ideal signal <br> constellation |  |
| getNumInputs | Number of expected inputs to <br> step method |  |
| getNumOutputs | Number of outputs from step <br> method |  |
| isLocked | Locked status for input attributes <br> and nontunable properties |  |
| Examples | Allow property value and input <br> characteristics changes |  |
|  | release | Modulate using BPSK method |

## Algorithms

This object implements the algorithm, inputs, and outputs described on the BPSK Modulator Baseband block reference page. The object properties correspond to the block parameters.

## comm.BPSKModulator

See Also comm.BPSKDemodulator | comm.PSKModulator

Purpose Create BPSK modulator object with same property values
Syntax $\quad C=$ clone $(H)$
Description $\quad C=$ clone $(H)$ creates a BPSKModulator object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.BPSKModulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax $\quad y=$ constellation $(h)$
constellation(h)
Description $\quad y=$ constellation( $h$ ) returns the numerical values of the constellation.
constellation(h) generates a constellation plot for the object.

## Examples Calculate Ideal Signal Constellation for comm.BPSKModulator

Create a comm.BPSKModulator System object, and then calculate its ideal signal constellation.

Create a comm.BPSKModulator System object by entering the following at the MATLAB command line:
h = comm.BPSKModulator

Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Constellation View of Ideal Signal for comm.BPSKModulator

Create a comm.BPSKModulator System object, and then plot the ideal signal constellation .

Create a comm.BPSKModulator System object by entering the following at the MATLAB command line:
h = comm.BPSKModulator
Plot the ideal signal constellation by calling the constellation method.
constellation(h)

## comm.BPSKModulator.getNuminputs

## Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.BPSKModulator.getNumOutputs

Purpose $\quad$ Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.BPSKModulator.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF = isLocked (H) |
| Description | TF = isLocked (H) returns the locked status, TF of the BPSKModulator <br> System object. |
| The isLocked method returns a logical value that indicates whether <br> input attributes and nontunable properties for the object are locked. The <br> object performs an internal initialization the first time the step method <br> is executed. This initialization locks nontunable properties and input <br> specifications, such as dimensions, complexity, and data type of the <br> input data. After locking, the isLocked method returns a true value. |  |

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Modulate using BPSK method |
| :--- | :--- |
| Syntax | $Y=\operatorname{step}(H, X)$ |$\quad$| $Y=$ step $(H, X)$ modulates input data, $X$, with the BPSK modulator |
| :--- |
| System object, $H$. It returns the baseband modulated output, $Y$. The |
| input must be a column vector of bits. The data type of the input can be |
| numeric, logical, or unsigned fixed point of word length 1 (fi object). |

## Purpose Measure complementary cumulative distribution function

Description

## Construction

The CCDF object measures the probability of a signal's instantaneous power to be a specified level above its average power.
$H=$ comm.CCDF creates a complementary cumulative distribution function measurement (CCDF) System object, H, that measures the probability of a signal's instantaneous power to be a specified level above its average power.

H = comm.CCDF(Name, Value) creates a CCDF object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties

## NumPoints

Number of CCDF points
Specify the number of CCDF points that the object calculates. This property requires a numeric, positive, integer scalar. The default is 1000 . Use this property with the MaximumPowerLimit property to control the size of the histogram bins. The object uses these bins to estimate CCDF curves. This controls the resolution of the curves. All input channels must have the same number of CCDF points.

## MaximumPowerLimit

Maximum expected input signal power
Specify the maximum expected input signal power limit for each input channel. The default is 50 . Set this property to a numeric scalar or row vector length equal to the number of input channels. When you set the this property to a scalar, the object assumes that the signals in all input channels have the same expected maximum power. When you set this property to a row vector length equal to the number of input channels, the object assumes that the $i$-th element of the vector is the maximum expected
power for the signal at the $i$-th input channel. When you call the step method, the object displays the value of this property is in the units that you specify in the PowerUnits property. For each input channel, the object obtains CCDF results by integrating a histogram of instantaneous input signal powers. The object sets the bins of the histogram so that the last bin collects all power occurrences that are equal to, or greater than the power that you specify in this property. The object issues a warning if any input signal exceeds its specified maximum power limit. Use this property with the NumPoints property to control the size of the histogram bins that the object uses to estimate CCDF curves (such as control the resolution of the curves).

## PowerUnits

## Power units

Specify the power measurement units as one of dBm \| dBW | Watts. The default is dBm. The step method outputs power measurements in the units specified in the PowerUnits property. When you set this property to dBm or dBW , the step method outputs relative power values in a dB scale. When you set this property to Watts, the step method outputs relative power values in a linear scale. When you call the step method, the object assumes that the units of MaximumPowerLimit have the same value you specified in the PowerUnits property.

## AveragePowerOutputPort

Enable average power measurement output
When you set this property to true, the step method outputs running average power measurements. The default is false.

## PeakPowerOutputPort

Enable peak power measurement output
When you set this property to true, the step method outputs running peak power measurements. The default is false.

## PAPROutputPort

## Enable PAPR measurement output

When you set this property to true, the step method outputs running peak-to-average-power measurements. The default is false.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | getPercentileRelativePower |
|  | getProbability |
|  | isLocked |
|  | plot <br> release |
| reset |  |
| step |  |

Create CCDF measurement object with same property values
Number of expected inputs to step method

Number of outputs from step method

Get relative power value for a given probability

Get the probability for a given relative power value

Locked status for input attributes and nontunable properties

Plot CCDF curves
Allow property value and input characteristics changes

Reset states of CCDF measurement object

Measure complementary cumulative distribution function

Examples Obtain CCDF curves for 16-QAM and QPSK signals in AWGN.

```
hQAM = comm.RectangularQAMModulator(16);
hQPSK = comm.QPSKModulator;
hChan = comm.AWGNChannel('NoiseMethod',...
    'Signal to noise ratio (SNR)', 'SNR', 15);
```

```
    % Create a CCDF System object and request average power and peak
    % power measurement outputs
        hCCDF = comm.CCDF('AveragePowerOutputPort', true, ...
                'PeakPowerOutputPort', true);
% Modulate signals
    sQAM = step(hQAM,randi([0 16-1],20e3,1));
    sQPSK = step(hQPSK,randi([0 4-1],20e3,1));
% Pass signals through an AWGN channel
    hChan.SignalPower = 10;
    sQAMNoisy = step(hChan,sQAM);
    hChan.SignalPower = 1;
    sQPSKNoisy = step(hChan,sQPSK);
    % Obtain CCDF measurements
    [CCDFy,CCDFx,AvgPwr,PeakPwr] = step(hCCDF,[sQAMNoisy sQPSKNoisy])
    % plot CCDF curves using the plot method of the CCDF object
    plot(hCCDF)
    legend('16-QAM', 'QPSK')
See Also
comm.ACPR | comm.EVM | comm.MER
```

Purpose Create CCDF measurement object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a CCDF object $C$, with the same property values as H. The clone method creates a new unlocked object with uninitialized states.

Purpose $\quad$ Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description
$N=$ getNumInputs (H) method returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs $(H)$.

## comm.CCDF.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn outputs on or off are changed.

## Purpose

Syntax
Description

## Examples

Get relative power value for a given probability
$R$ = getPercentileRelativePower(H,P)
$R=$ getPercentileRelativePower( $H, P$ ) finds the relative power values, R. The power of the signal of interest is above its average power by R dB (if PowerUnits equals 'dBW', or 'dBm') or by a factor of R (in linear scale if PowerUnits equals 'Watts') with a probability P.
The method output R , is a column vector with the $i$-th element corresponding to the relative power for the $i$-th input channel. The method input $P$ can be a double precision scalar, or a vector with a number of elements equal to the number of input channels. If $P$ is a scalar, then all the relative powers in $R$ correspond to the same probability value specified in $P$. If $P$ is a vector, then the $i$-th element of R corresponds to a power value that occurs in the $i$-th input channel, with a probability specified in the $i$-th element of P .
For the $i$-th input channel, this method evaluates the inverse CCDF curve at probability value $P(i)$.

Obtain CCDF curves for a unit variance AWGN signal and a dual- one signal. The AWGN signal is RPW1 dB above its average power one percent of the time, and the dual-tone signal is RPW2 dB above its average power 10 percent of the time. This example finds the values of RPW1 and RPW2.

```
n = [0:5e3-1].';
s1 = randn(5e3,1); % AWGN signal
s2 = sin(0.01*pi*n)+sin(0.03*pi*n); % dual-tone signal
hCCDF = comm.CCDF; % create a CCDF object
step(hCCDF,[s1 s2]); % step the CCDF measurements
plot(hCCDF) % plot CCDF curves
legend('AWGN','Dual-tone')
RPW = getPercentileRelativePower(hCCDF,[1 10]);
RPW1 = RPW(1)
RPW2 = RPW(2)
```


## Purpose Get the probability for a given relative power value

```
Syntax P = getProbability(H,R)
```

Description $\quad P=$ getProbability ( $H, R$ ) finds the probability, $P$, of the power level of the signal of interest being R dBs (if PowerUnits equals ' dBW ', or ' dBm ') or Watts (if PowerUnits equals 'Watts') above its average power. P is a column vector with the $i$-th element corresponding to the probability value for the $i$-th input channel. Input R can be a double precision scalar or a vector with a number of elements equal to the number of input channels. If R is a scalar, then all the probability values in P correspond to the same relative power specified in R. If $R$ is a vector, then the ith element of P contains a probability value for the $i$-th channel and for the relative power specified in the $i$-th element of R .

For the $i$-th input channel, this method evaluates the CCDF curve at relative power value $\mathrm{R}(i)$

## Examples

Obtain CCDF curves for a unit variance AWGN signal and a dual- tone signal. Find the probability that the AWGN signal power is 5 dB above its average power and that the dual-tone signal power is 3 dB above its average power.

```
n = [0:5e3-1].';
s1 = randn(5e3,1); %AWGN signal
s2 = sin(0.01*pi*n)+sin(0.03*pi*n); % dual-tone signal
hCCDF = comm.CCDF;
step(hCCDF,[s1 s2]);
plot(hCCDF) % plot CCDF curves
legend('AWGN','Dual-tone')
P = getProbability(hCCDF,[5 3]) % get probabilities
```


## Purpose Locked status for input attributes and nontunable properties

Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the CCDF System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## Purpose Plot CCDF curves

## Syntax <br> D = plot(H)

Description $\quad D=\operatorname{plot}(H)$ plots CCDF measurements in the CCDF System object, $H$. The plot method returns the plot handles as an output, D. This method plots the same number of curves as there are input channels. The H input can be followed by parameter-value pairs to specify additional properties of the curves. For example, plot(H,LineWidth,2) will create curves with line widths of 2 points.

The comm. CCDF System object does not support C code generation for this method.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release (H) |
| Description | release (H) Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.CCDF.reset

Purpose Reset states of CCDF measurement object

## Syntax reset (H)

Description reset $(H)$ resets the states of the CCDF object, H.

Measure complementary cumulative distribution function
[CCDFY, CCDFX] = step( $\mathrm{H}, \mathrm{X}$ )
[CCDFY,CCDFX,AVG] = step(H,X)
[CCDFY,CCDFX,PEAK] = step(H,X)
[CCDFY,CCDFX,PAPR] = step $(H, X)$
[CCDFY, CCDFX] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$ updates CCDF, average power, and peak power measurements for input $X$ using the CCDF System object, H. It outputs the $y$-axis, CCDFY, and $x$-axis, CCDFX, CCDF points. X must be a double precision, $M$-by- $N$ matrix, where $M$ is the number of time samples and $N$ is the number of input channels. The step method outputs CCDFY as a matrix whose $i$-th column contains updated probability values measured from the $i$-th column of input matrix $X$. CCDFY contains the $y$-axis points of the CCDF curves of each channel. The step method outputs CCDFX as a matrix containing, in its $i$-th column, the corresponding updated instantaneous-to-average power ratios for the ith column of input matrix X. CCDFX contains the $x$-axis points of the CCDF curves of each channel. The object sets the number of rows in CCDFY and CCDFX equal to NumPoints property +1 . The probability values are percentages in the [ 0 100] interval. When you set the PowerUnits property to dBW or dBm , the relative powers are in dB scale. When you set the PowerUnits property to Watts, the relative powers are in linear scale. Measurements are updated each time you call the step method until you reset the object. You call the plot method to plot CCDF curves for each channel.
[CCDFY, CCDFX,AVG] = step $(\mathrm{H}, \mathrm{X})$ returns updated average power measurements, AVG, when you set the AveragePowerOutputPort property to true. The step method outputs AVG as a column vector with the ith element corresponding to an updated average power measurement for the signal available in the ith column of input matrix X . You specify the units for AVG in the PowerUnits property.
[CCDFY,CCDFX, PEAK] = step( $\mathrm{H}, \mathrm{X}$ ) returns updated peak power measurements, PEAK, when you set the PeakPowerOutputPort property to true. The step method outputs PEAK as a column vector with the ith element corresponding to an updated peak power measurement for the
signal available in the ith column of input matrix $X$. You specify the units for PEAK in the PowerUnits property.
[CCDFY, CCDFX, PAPR] = step(H,X) returns updated peak-to-average power ratio measurements, PAPR, when you set the PAPROutputPort property to true. The step methods outputs PAPR as a column vector with the ith element corresponding to an updated peak-to-average power ratio measurement for the signal available in the ith column of input matrix $X$. When you set the PowerUnits property to dBW or dBm , the method outputs PAPR in a dB scale. When you set the PowerUnits property to Watts, the method outputs PAPR in a linear scale. You can combine optional output arguments when you set their enabling properties. Optional outputs must be listed in the same order as the order of the enabling properties. For example,

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose

Display a constellation diagram for input signals

The ConstellationDiagram System object plots constellation diagrams and provides the ability to perform EVM and MER measurements.

## Construction $H=$ comm.ConstellationDiagram returns a System object, H, that

 displays real and complex-valued floating and fixed-point signals in the I/Q plane.H = comm.ConstellationDiagram(Name, Value, ...) returns a Constellation Diagram System object, H, with each specified property Name set to the specified Value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties

## Name

Caption to display on Constellation Diagram window
Specify the caption that the Constellation Diagram window displays as a string. The default value of this property is Constellation Diagram. This property is tunable.

## SamplesPerSymbol

Number of samples used to represent a symbol
Specify the number of samples that represent a symbol. The default value of this property is 1 . When the SamplesPerSymbol property is greater than 1 , the object downsamples and plots the input signal.

## SampleOffset

Number of samples to skip before plotting points
Specify the number of samples to skip when decimating the input signal. The default value of this property is 0 . This property is tunable. This value must be a nonnegative integer less than the number of samples per symbol.

## SymbolsToDisplay

The maximum number of symbols that can be displayed when input signal is long.
Always plot the latest SymbolsToDisplay symbols. The default value of this property is 256 . This property is tunable.

## ReferenceConstellation

The ideal constellation of the input signal
The object can display the ReferenceConstellation with its own marker. To obtain the signal quality measurement, you must set the ReferenceConstellation property to a valid value. The default value of this property is: $[0.7071+0.7071 \mathrm{i}-0.7071+0.7071 \mathrm{i}$ -0.7071-0.7071i 0.7070-0.7071i]. This property is tunable.

## ReferenceMarker

Specify the marker for reference display
The default value of this property is ' + '. This property is tunable.

## ReferenceColor

Specify the color for reference display constellation
The default value of this property is [ $\left.\begin{array}{lll}1 & 0 & 0\end{array}\right]$ (red). This property is tunable.

## ShowReferenceConstellation

Option to turn on the reference constellation
Set this property to true to show reference constellation on the display. The default value of this property is true. This property is tunable.

## Position

Scope window position in pixels
Specify the size and location of the scope window in pixels, as a four-element double vector of the form: [left bottom width height]. The default value of this property is dependent on the screen resolution, and is such that the window is positioned in the center
of the screen, with a width and height of 410 and 300 pixels respectively. This property is tunable.

## ShowGrid

Option to turn on grid
Set this property to true to turn on the grid or false to turn off the grid. The default value of this property is true. This property is tunable.

## ShowLegend

Option to turn on legend
Set this property to true to turn on the legend. The default is false. This property is tunable.

## ColorFading

Option to add color fading effect
When you set this property to true, the points in the display fade as the interval of time after they are first plotted increases. This is for animation that resembles an oscilloscope. The default value of this property is false. This property is tunable.

## Title

Display title
Specify the display title as a string. The default value of this property is an empty string. This property is tunable.

## XLimits

X -axis limits
Specify the x -axis limits as a two-element numeric vector: [xmin $x \max ]$. The default value of this property is [ -1.375 1.375]. This property is tunable.

## YLimits

Y -axis limits

Specify the y-axis limits as a two-element numeric vector: [ymin $y \max ]$. The default value of this property is [ -1.375 1.375]. This property is tunable

## XLabel

X -axis label
Specify the $x$-axis label as a string. The default value of this property is In-phase Amplitude. This property is tunable.

## YLabel

Y-axis label
Specify the $y$-axis label as a string. The default value of this property is Quadrature Amplitude. This property is tunable.

## MeasurementInterval

The measurement interval
When the input signal contains one sample per symbol and the reference constellation is provided, this System object can measure the signal quality in terms of EVM and MER. The measurement panel can be evoked by clicking on the Signal Quality button. This property specifies the window length for the measurement. The value of this property must be greater than one and less than or equal to the value of SymbolesToDisplay property. If the number of data input is less than MeasurementInterval, it will wait for more data before measurement can be calculated. The default value of this property is 2 . This property is tunable.

## EVMNormalization

EVM normalization
Specify the normalization method that the object uses in the EVM calculation as one of Average constellation power or Peak constellation power. The default value of this property is Average constellation power. This property is tunable.

## Methods

clone
hide
isLocked
release
reset
show
step

> Create scope object with same property values
> Hide scope window
> Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
Reset internal states of the scope object

Make scope window visible
Display constellation diagram of signal in scope figure

## Examples Plot Constellation with One Sample per Symbol

Create a 16-QAM modulator, transmit data using an AWGN channel, and plot the signal constellation.
Create a Rectangular QAM Modulator System object, hMod, and set the modulation order to 16 .
hMod = comm.RectangularQAMModulator('ModulationOrder', 16);
Transmit the modulated signal using an AWGN channel, hAWGN, with a signal-to-noise ratio of 20 .

```
hAWGN = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)
    'SNR', 20);
```

Create the constellation diagram scope.

```
hScope = comm.ConstellationDiagram('ReferenceConstellation', hMod.cons
```

Generate random data symbols as an input to the modulator.

```
d = randi([0 15], 100, 1);
```

Modulate the random data signal using the step method of the Rectangular QAM Modulator System object.

```
sym = step(hMod, d);
```

Transmit the modulated signal using the step method of the AWGN channel System object.
rcv = step(hAWGN, sym);
Plot the transmitted signal using the step method of the Constellation Diagram System object
step(hScope, rcv)

## Input signal has multiple samples per symbol, the ConstellationDiagram will decimate and plot one sample per symbol.

Input signal has multiple samples per symbol, the Constellation Diagram will decimate and plot one sample per symbol.

Create a Rectangular QAM Modulator System object, hMod, and set the modulation order to 16 .

```
hMod = comm.RectangularQAMModulator('ModulationOrder', 16);
```

Create the constellation diagram scope.

```
hScope = comm.ConstellationDiagram('SamplesPerSymbol', 8,...
    'SampleOffset', 1,...
    'ReferenceConstellation', hMod.constellation);
```

Transmit the modulated signal using an AWGN channel, hAWGN, with a signal-to-noise ratio of 30 .

```
hAWGN = comm.AWGNChannel('NoiseMethod', 'Signal to noise ratio (SNR)',..
    'SNR', 30);
```

Create a normalized upsampling filter.

```
hFilDesign = fdesign.pulseshaping(8,'Raised Cosine','Nsym,Beta', 8,0.!
    hFil = design(hFilDesign);
    hFil.Numerator = hFil.Numerator / max(hFil.Numerator);
    hInterp = dsp.FIRInterpolator('InterpolationFactor', hScope.Sa
                            'Numerator', hFil.Numerator);
```

Generate a random data stream, modulate the symbols, upsample the signal and transmit using an AWGN channel. Then, display the output.

```
d = randi([0 15], 100, 1);
sym = step(hMod, d);
xmt = step(hInterp, sym);
rcv = step(hAWGN, xmt);
step(hScope, rcv)
```

Signal
Display

To change the signal display settings, select View > Configuration Properties to bring up the Visuals-Constellation Properties dialog box. Then, modify the values for the Samples per symbol, Offset, Symbols to display and Reference Constellation parameters on the Main tab.

To communicate simulation data that corresponds to the current display, the scope uses the Frames indicator on the scope window. The following figure highlights important aspects of the Constellation Diagram window.

## comm.ConstellationDiagram



## Toolbar

Axes Control Buttons

| + | Tools > <br> Zoom In | N/A | When this tool is active, you can zoom in on the scope window. To do so, click in the center of your area of interest, or click and drag your cursor to draw a rectangular area of interest inside the scope window. |
| :---: | :---: | :---: | :---: |
| "Q" | Tools > <br> Zoom X | N/A | When this tool is active, you can zoom in on the $x$-axis. To do so, click inside the scope window, or click and drag your cursor along the $x$-axis over your area of interest. |
| ¢ | Tools > <br> Zoom Y | N/A | When this tool is active, you can zoom in on the $y$-axis. To do so, click inside the scope window, or click and drag your cursor along the $y$-axis over your area of interest. |
| 8in | Tools > Pan | N/A | When this tool is active, you can pan on the scope window. To do so, click in the center of your area of interest and drag your cursor to the left, right, up, |



|  |  |  | scope scales the axes each time the <br> simulation is stopped. |
| :--- | :--- | :--- | :--- |
| (B) | Tools $>$ <br> Measurements $>$ <br> Signal <br> Quality | N/A <br> Click this button to display Error <br> Vector Measurement (EVM) and <br> Modulation Error Ratio (MER) <br> measurement results. |  |

## Measurements Measurements Panel Buttons Panels <br> Each of the Measurements panels contains the following buttons that enable you to modify the appearance of the current panel.

| Button | Description |
| :--- | :--- |
| $\boldsymbol{\mp}$ | Move the current panel to the top. When you are displaying <br> more than one panel, this action moves the current panel <br> above all the other panels. |
| $\boldsymbol{\nabla}$ | Collapse the current panel. When you first enable a panel, by <br> default, it displays one or more of its panes. Click this button <br> to hide all of its panes to conserve space. After you click this <br> button, it becomes the expand button |
| $\boldsymbol{\nabla}$ | Expand the current panel. This button appears after you <br> click the collapse button to hide the panes in the current <br> panel. Click this button to display the panes in the current <br> panel and show measurements again. After you click this <br> button, it becomes the collapse button $\boldsymbol{\nabla}$ again. |
| $\boldsymbol{\pi}$ | Undock the current panel. This button lets you move the <br> current panel into a separate window that can be relocated <br> anywhere on your screen. After you click this button, it <br> becomes the dock button $\boldsymbol{\aleph}$ in the new window. |


| Button | Description |
| :--- | :--- |
| $\boldsymbol{\Delta}$ | Dock the current panel. This button appears only after you <br> click the undock button. Click this button to put the current <br> panel back into the right side of the Scope window. After you <br> click this button, it becomes the undock button $\boldsymbol{\pi}$ again. |
| $\mathbf{X}$ | Close the current panel. This button lets you remove the <br> current panel from the right side of the Scope window. |

Some panels have their measurements separated by category into a number of panes. Click the pane expand button to show each pane that is hidden in the current panel. Click the pane collapse button $\nabla$ to hide each pane that is shown in the current panel.

## Signal Quality Panel

The Signal Quality panel displays Error Vector Measurement (EVM) and Modulation Error Ratio (MER) measurement results.


You can choose to hide or display the Signal Quality panel. In the Scope menu, select Tools > Measurements > Signal Quality.

## Settings Pane

The Settings pane enables you to define the measurement interval and normalization method the scope uses when obtaining signal measurements.

- Measurement interval - Specify the duration of the EVM or MER measurement. For more information see MeasurementInterval.
- EVM normalization - For the EVM calculations, you may use one of two normalization methods: average constellation power or peak constellation power. The scope performs EVM calculations using the comm. EVM System object. For more information, see comm. EVM.


## Signal Quality Pane

The Signal Quality pane displays the calculation results.

- EVM - An error vector is a vector in the I-Q plane between the ideal constellation point and the actual point at the receiver. EVM is measured in two formats: root mean square (RMS) or normalized Peak. Typically, EVM is reported in decibels. For more information, see comm.EVM.
- MER - MER is the ratio of the average power of the error vector and the average power of the transmitted signal. The scope indicates the measurement result in decibels. For more information, see comm. MER.


## Visuals -

 Constellation Properties
## Main Pane

## Samples per symbol

Number of samples used to represent a symbol. This value must be a positive number. When the Measurements tool is on, you must set this property to 1 .

## Offset (samples)

Number of samples to skip before plotting points. The offset must be a nonnegative integer value less than the value of the samples per symbol.

## Symbols to display

The maximum number of symbols that can be displayed. Must be a positive integer value.

## Reference constellation

The ideal constellation of the input signal. When the Measurements tool is on, the reference constellation is used to detect the ideal signal input. Therefore, this property cannot be empty when the Measurements tool is on. (When the Measurements tool is not on, this property can be empty.)

## Display Pane

Show grid
Select this check box to turn on the grid.

## Color fading

When you set select this check box, the points in the display fade as the interval of time after they are first plotted increases. The default value of this property is false. This property is tunable.

## Show legend

Select this check box to display a legend for the graph.

## Show reference constellation

Select this check box to display the points comprising the reference constellation.

## Reference marker

Select the symbol that represents the points on the reference constellation.

## Reference color

Select the color of the points on the reference constellation. Refer to the following table for the binary values and their corresponding colors.

| Color | Binary Code |
| :--- | :--- |
| Black | 000 |
| Blue | 001 |


| Color | Binary Code |
| :--- | :--- |
| Green | 010 |
| Cyan | 011 |
| Red | 100 |
| Magenta | 101 |
| White | 111 |

## X-limits (Minimum)

Specify the minimum value of the x -axis.

## X-limits (Maximum)

Specify the maximum value of the x -axis.

## $\mathbf{Y}$-limits (Minimum)

Specify the minimum value of the $y$-axis.

## Y-limits (Maximum)

Specify the maximum value of the $y$-axis.

## Title

Specify a label that appears above the constellation diagram plot. By default, there is no title.

## X-axis label

Specify the text the scope displays along the x -axis

## Y-axis label

Specify the text the scope displays along the $y$-axis
In the Style dialog box, you can customize the style of displays. You are able to change the color of the figure containing the displays, the background and foreground colors of display axes, and properties of lines in a display. From the scope menu, select View > Style to open this dialog box.


## Properties

The Style dialog box allows you to modify the following properties of the scope figure:

## Figure color

Specify the color that you want to apply to the background of the scope figure. By default, the figure color is gray.

## Axes colors

Specify the color that you want to apply to the background of the axes for the active display.

## Properties for line

Specify the signal for which you want to modify the visibility, line properties, and marker properties.

## Visible

Specify whether the selected signal on the active display should be visible. If you clear this check box, the line disappears.

## Line

Specify the line style, line width, and line color for the selected signal on the active display.

## Marker

Specify marks for the selected signal on the active display to show at data points. This parameter is similar to the Marker property for the MATLAB Handle Graphics ${ }^{\circledR}$ plot objects. You can choose any of the marker symbols from the following table.

| Specifier | Marker Type |
| :--- | :--- |
| none | No marker (default) |
|  | Circle |
|  |  |
|  |  |
| $\square$ | Square |
| $\times$ | Cross |
| $\bullet$ | Point |
| + | Asterisk |
| $*$ | Diamond |
| $\diamond$ | Downward-pointing triangle |
| $\nabla$ | Left-pointing triangle |
| $\Delta$ | Right-pointing triangle |
| $\triangleleft$ |  |
| $\triangleright$ |  |


| Specifier | Marker Type |
| :--- | :--- |
| A | Five-pointed star (pentagram) |
|  | Six-pointed star (hexagram) |

## Tools: Plot Navigation Properties

## Properties

The Tools-Plot Navigation Properties dialog box appears as follows.

## Properties

$\square$
Axes scaling:
Do not allow Y -axis limits to shrink
Scale axes limits at stop

## Y-axis

Data range (\%): 80
Align:


Scale $X$-axis limits


## Axes Scaling

Specify when the scope should automatically scale the axes. You can select one of the following options:

- Manual - When you select this option, the scope does not automatically scale the axes. You can manually scale the axes in any of the following ways:
- Select Tools > Scale axes limits.
- Press the Scale Axes Limits toolbar button.
- When the scope figure is the active window, press $\mathbf{C t r l}$ and $\mathbf{A}$ simultaneously.
- Auto - When you select this option, the scope scales the axes as needed, both during and after simulation. Selecting this option shows the Do not allow Y-axis limits to shrink check box.
- After $N$ Updates - Selecting this option causes the scope to scale the axes after a specified number of updates. Selecting this option shows the Number of updates edit box.

This parameter is Tunable.
By default, this parameter is set to Manual.

## Do not allow Y-axis limits to shrink

When you select this parameter, the $y$-axis limits are only allowed to grow during axes scaling operations. If you clear this check box, the $y$-axis limits may shrink during axes scaling operations.

This parameter appears only when you select Auto for the Axis Scaling parameter. When you set the Axes Scaling parameter to Manual or After N Updates, the $y$-axis limits are allowed to shrink. Tunable.

## Number of updates

Specify as a positive integer the number of updates after which to scale the axes. This parameter appears only when you select After N Updates for the Axes Scaling parameter. Tunable.

## Scale axes limits at stop

Select this check box to scale the axes limits when the simulation stops. The $y$-axis are always scaled. The $x$-axis limits are only scaled if you also select the Scale X-axis limits check box.

## Y-axis Data range (\%)

Set the percentage of the $y$-axis that the scope should use to display the data when scaling the axes. Valid values are between 1 and 100. For example, if you set this parameter to 100 , the Scope scales the $y$-axis limits such that your data uses the entire $y$-axis range. If you then set this parameter to 30 , the scope increases the $y$-axis range such that your data uses only $30 \%$ of the $y$-axis range. Tunable.

## Y-axis Align

Specify where the scope should align your data with respect to the $y$-axis when it scales the axes. You can select Top, Center, or Bottom. Tunable.

## Scale X-axis limits

Check this box to allow the scope to scale the $x$-axis limits when it scales the axes. Tunable.

## X-axis Data range (\%)

Set the percentage of the $x$-axis that the Scope should use to display the data when scaling the axes. Valid values are between 1 and 100. For example, if you set this parameter to 100, the Scope scales the $x$-axis limits such that your data uses the entirex-axis range. If you then set this parameter to 30, the Scope increases the $x$-axis range such that your data uses only $30 \%$ of the $x$-axis range. Use the $x$-axis Align parameter to specify data placement with respect to the $x$-axis.

This parameter appears only when you select the Scale X-axis limits check box. Tunable.

## X-axis Align

Specify how the Scope should align your data with respect to the $x$-axis: Left, Center, or Right. This parameter appears only when you select the Scale X-axis limits check box. Tunable.

Purpose Create scope object with same property values
Syntax $\quad C=$ clone $(H)$
Description $\quad C=\operatorname{clone}(H)$ creates a scope object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.ConstellationDiagram.hide

| Purpose | Hide scope window |
| :--- | :--- |
| Syntax | hide (H) |
| Description | hide(H) hides the scope window associated with System object, H. |
| See Also | comm.ConstellationDiagram. show |

## comm.ConstellationDiagram.isLocked

Purpose Locked status for input attributes and nontunable properties

## Syntax isLocked (H)

Description isLocked $(\mathrm{H})$ returns the locked state of the scope object H .
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes
Syntax release(H)
Description
release $(H)$ releases system resources, such as memory, file handles, and hardware connections. This method lets you change any properties or input characteristics.
You should call the release method after calling the step method when there is no new data for the simulation. When you call the release method, the axes will automatically scale in the scope figure window. After calling the release method, any non-tunable properties can be set once again.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

See Also

In operation, the release method is similar to the mdlTerminate function.
Purpose Reset internal states of the scope object
Syntax ..... reset(H)
Description

reset ( H ) sets the internal states of the scope object H to their initial
values.

You should call the reset method after calling the step method when you want to clear the scope figure displays, prior to releasing system resources. This action enables you to start a simulation from the beginning. When you call the reset method, the displays will become blank again. In this sense, its functionality is similar to that of the MATLAB clf function. Do not call the reset method after calling the release method.

> Algorithms
> In operation, the reset method is similar to a consecutive execution of the mdlTerminate function and the mdlInitializeConditions function.

See Also

comm. ConstellationDiagram | comm. ConstellationDiagram.release
Purpose Make scope window visible
Syntax show(H)
Description show(H) makes the scope window associated with System object, H, visible.
See Also ..... comm.ConstellationDiagram.hide

Purpose
Display constellation diagram of signal in scope figure

## Syntax

step ( $\mathrm{H}, \mathrm{X}$ )
$\operatorname{step}(\mathrm{H}, \mathrm{X} 1, \mathrm{X} 2, \ldots, \mathrm{XN})$
Description
step $(H, X)$ displays the signal, $X$, in the scope figure.
step ( $\mathrm{H}, \mathrm{X} 1, \mathrm{X} 2, \ldots, \mathrm{XN}$ ) displays the signals $\mathrm{X} 1, \mathrm{X} 2, \ldots, \mathrm{XN}$ in the scope figure when you set the NumInputPorts property to N. In this case, X1, $\mathrm{X} 2, \ldots, \mathrm{XN}$ can have different data types and dimensions.

| Purpose | Restore ordering of symbols using shift registers |
| :---: | :---: |
| Description | The ConvolutionalDeinterleaver object recovers a signal that was interleaved using the convolutional Interleaver object. The parameters in the two blocks should have the same values. |
| Construction | H = comm. ConvolutionalDeinterleaver creates a convolutional deinterleaver System object, H. This object restores the original ordering of a sequence that was interleaved using the convolutional interleaver System object. |
|  | H = comm.ConvolutionalDeinterleaver(Name, Value) creates a convolutional deinterleaver System object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN). |
| Properties | NumRegisters |
|  | Number of internal shift registers |
|  | Specify the number of internal shift registers as a scalar, positive integer. The default is 6 . |
|  | RegisterLengthStep |
|  | Symbol capacity difference of each successive shift register |
|  | Specify the difference in symbol capacity of each successive shift register, where the last register holds zero symbols as a positive, scalar integer. The default is 2 . |
|  | InitialConditions |
|  | Initial conditions of shift registers |
|  | Specify the values that are initially stored in each shift register as a numeric scalar or vector, except the first shift register, which has zero delay. If you set this property to a scalar, then all shift registers, except the first one, store the same specified value. You can also set this property to a column vector with length equal to |

the value of the NumRegisters property. With this setting, the $i$-th shift register stores the $i$-th element of the specified vector. The value of the first element of this property is unimportant because the first shift register has zero delay.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
|  | release |
| reset |  |
| step |  |

> Create convolutional deinterleaver object with same property values
> Number of expected inputs to step method
> Number of outputs from step method
> Locked status for input attributes and nontunable properties
> Allow property value and input characteristics changes
> Reset states of the convolutional deinterleaver object
> Restore ordering of symbols using shift registers

Examples Interleave and deinterleave random data.

```
hInt = comm.ConvolutionalInterleaver('NumRegisters', 3, ...
    'RegisterLengthStep', 2, ...
    'InitialConditions', [ -1 -2 -3]');
hDeInt = comm.ConvolutionalDeinterleaver('NumRegisters', 3, ...
    'RegisterLengthStep', 2, ...
    'InitialConditions', [-1 -2 -3]');
data = (0:20)';
intrlvData = step(hInt, data);
deintrlvData = step(hDeInt, intrlvData);
```

```
% compare the original sequence, interleaved sequence and restored
[data, intrlvData, deintrlvData]
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the Convolutional Deinterleaver block reference page. The object properties correspond to the block parameters.

See Also

Purpose
Create convolutional deinterleaver object with same property values

## Syntax <br> C = clone(H)

Description
$C=$ clone (H) creates a ConvolutionalDeinterleaver object $C$, with the same property values as H. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.ConvolutionalDeinterleaver.getNumOutputs

## Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the ConvolutionalDeinterleaver System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of the convolutional deinterleaver object
Syntax ..... reset (H)
Description reset (H) resets the states of the ConvolutionalDeinterleaver object, H.

## Purpose

## Syntax

Description

Restore ordering of symbols using shift registers
$Y=\operatorname{step}(H, X)$
$Y=\operatorname{step}(H, X)$ restores the original ordering of the sequence, $X$, that was interleaved using a convolutional interleaver and returns Y. The input $X$ must be a column vector. The data type can be numeric, logical, or fixed-point (fi objects). $Y$ has the same data type as $X$. The convolutional deinterleaver object uses a set of $N$ shift registers, where $N$ is the value specified by the NumRegisters property. The object sets the delay value of the $k$-th shift register to the product of ( $k-1$ ) and RegisterLengthStep property value. With each new input symbol, a commutator switches to a new register and the new symbol shifts in while the oldest symbol in that register shifts out. When the commutator reaches the $N$-th register and the next new input occurs, it returns to the first register.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Convolutionally encode binary data <br> DescriptionThe ConvolutionalEncoder object encodes a sequence of binary input <br> vectors to produce a sequence of binary output vectors. |
| :--- | :--- |
| Construction | H = comm. ConvolutionalEncoder creates a System object, H, that <br> convolutionally encodes binary data. |
|  | H = comm. ConvolutionalEncoder (Name, Value) creates a <br> convolutional encoder object, H, with each specified property set to the <br> specified value. You can specify additional name-value pair arguments <br> in any order as (Name1, Value1,...,NameN, ValueN). |
|  | H = comm. ConvolutionalEncoder (TRELLIS, Name, Value) creates a <br> convolutional encoder object, H This object has the TrellisStructure <br> property set to TRELIIS, and the other specified properties set to the <br> specified values. |
| Properties | TrellisStructure |

Trellis structure of convolutional code
Specify the trellis as a MATLAB structure that contains the trellis description of the convolutional code. Use the istrellis function to check if a structure is a valid trellis structure. The default is the result of poly2trellis (7, [171 133]).

## TerminationMethod

Termination method of encoded frame
Specify how the encoded frame is terminated as one of Continuous | Truncated | Terminated. The default is Continuous. When you set this property to Continuous, the object retains the encoder states at the end of each input vector for use with the next input vector. When you set this property to Truncated, the object treats each input vector independently. The encoder states are reset at the start of each input vector. If you set the InitialStateInputPort property to false, the object resets its states to the all-zeros state. If you set the

InitialStateInputPort property to true, the object resets the states to the values you specify in the initial states step method input. When you set this property to Terminated, the object treats each input vector independently. For each input vector, the object uses extra bits to set the encoder states to all-zeros states at the end of the vector. For a rate $K / N$ code, the step
method outputs a vector with length $N \times(L+S) / K$, where $S=$ constraintLength-1 (or, in the case of multiple constraint lengths, $S=\operatorname{sum}($ constraintLength(i)-1)). $L$ is the length of the input to the step method.

## ResetInputPort

Enable encoder reset input
Set this property to true to enable an additional input to the step method. The default is false. When this additional reset input is a nonzero value, the internal states of the encoder reset to their initial conditions. This property applies when you set the TerminationMethod property to Continuous.

## DelayedResetAction

Delay output reset
Set this property to true to delay resetting the object output. The default is false. When you set this property to true, the reset of the internal states of the encoder occurs after the object computes the encoded data. When you set this property to false, the reset of the internal states of the encoder occurs before the object computes the encoded data. This property applies when you set the ResetInputPort property to true.

## InitialStatelnputPort

Enable initial state input
Set this property to true to enable a step method input that allows the specification of the initial state of the encoder for each
input vector. The default is false. This property applies when you set the TerminationMethod property to Truncated.

## FinalStateOutputPort

Enable final state output
Set this property to true to obtain the final state of the encoder via a step method output. The default is false. This property applies when you set the TerminationMethod property to Continuous or Truncated.

## PuncturePatternSource

Source of puncture pattern
Specify the source of the puncture pattern as one of None I Property. The default is None. When you set this property to None the object does not apply puncturing. When you set this property to Property, the object punctures the code. This puncturing is based on the puncture pattern vector that you specify in the PuncturePattern property. This property applies when you set the TerminationMethod property to Continuous or Truncated.

## PuncturePattern

Puncture pattern vector
Specify the puncture pattern used to puncture the encoded data as a column vector. The default is $[1 ; 1 ; 0 ; 1 ; 0 ; 1]$. The vector contains 1 s and 0 s , where the 0 indicates the punctured, or excluded, bits. This property applies when you set the TerminationMethod property to Continuous or Truncated and the PuncturePatternSource property to Property.

## Methods clone

getNumInputs

Create convolutional encoder object with same property values

Number of expected inputs to step method
getNumOutputs
isLocked
release
reset
step

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of the convolutional encoder object

Convolutionally encode binary
data

Examples Transmit a convolutionally encoded 8-DPSK-modulated bit stream.

```
hConEnc = comm.ConvolutionalEncoder;
hMod = comm.DPSKModulator('BitInput',true);
hChan = comm.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)',...
    'SNR',10);
hDemod = comm.DPSKDemodulator('BitOutput',true);
hDec = comm.ViterbiDecoder('InputFormat','Hard');
hError = comm.ErrorRate('ComputationDelay',3,'ReceiveDelay', 34);
for counter = 1:20
    data = randi([0 1],30,1);
    encodedData = step(hConEnc, data);
    modSignal = step(hMod, encodedData);
    receivedSignal = step(hChan, modSignal);
    demodSignal = step(hDemod, receivedSignal);
    receivedBits = step(hDec, demodSignal);
    errors = step(hError, data, receivedBits);
end
disp(errors)
```


## comm.ConvolutionalEncoder

Algorithms | This object implements the algorithm, inputs, and outputs described on |
| :--- |
| the Convolutional Encoder block reference page. The object properties |
| correspond to the block parameters, except: |

The operation mode Reset on nonzero input via port block
parameter corresponds to the ResetInputPort property.
See Also comm.ViterbiDecoder | comm.APPDecoder

Purpose
Create convolutional encoder object with same property values

## Syntax <br> C = clone(H)

Description

C = clone (H) creates a ConvolutionalEncoder object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.ConvolutionalEncoder.getNumOutputs

## Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the ConvolutionalEncoder System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of the convolutional encoder object

## Syntax reset (H)

Description reset (H) resets the states of the ConvolutionalEncoder object, H.

Convolutionally encode binary data

```
Y = step(H,X)
Y = step(H,X,INITSTATE)
Y = step(H,X,R)
[Y,FSTATE] = step(H,X)
```

$Y=\operatorname{step}(H, X)$ encodes the binary data, $X$, using the convolutional encoding that you specify in the TrellisStructure property. It returns the encoded data, $Y$. Both $X$ and $Y$ are column vectors of data type numeric, logical, or unsigned fixed point of word length 1 (fi object). When the convolutional encoder represents a rate $K / N$ code, the length of the input vector equals $K \times L$, for some positive integer, $L$. The step method sets the length of the output vector, Y , to $L \times N$.
$Y=\operatorname{step}(H, X$, INITSTATE) uses the initial state specified in the INITSTATE input when you set the TerminationMethod property to 'Truncated' and the InitialStateInputPort property to true. INITSTATE must be an integer scalar.
$Y=\operatorname{step}(H, X, R)$ resets the internal states of the encoder when you input a non-zero reset signal, R. R must be a double precision or logical scalar. This syntax applies when you set the TerminationMethod property to Continuous and the ResetInputPort property to true.
[ $\mathrm{Y}, \mathrm{FSTATE}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$ returns the final state of the encoder in the integer scalar output FSTATE when you set the FinalStateOutputPort property to true. This syntax applies when you set the TerminationMethod property to Continuous or Truncated.


#### Abstract

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.


| Purpose | Permute input symbols using shift registers with same property values |
| :--- | :--- |
| Description | The Convolutionalinterleaver object permutes the symbols in the <br> input signal. Internally, this class uses a set of shift registers. |
| Construction | H = comm. Convolutionalinterleaver creates a convolutional <br> interleaver System object, H. that permutes the symbols in the input <br> signal using a set of shift registers. <br> H = comm. Convolutionalinterleaver (Name, Value) creates <br> a convolutional interleaver System object, e. This object has <br> each specified property set to the specified value. You can <br> specify additional name-value pair arguments in any order as <br> (Name1, Value1,..,NameN,ValueN). |
| Properties $\quad$NumRegisters <br> Number of internal shift registers <br> Specify the number of internal shift registers as a scalar, positive <br> integer. The default is 6. |  |
| RegisterLengthStep |  |

Number of additional symbols that fit in each successive shift register
Specify the number of additional symbols that fit in each successive shift register as a positive, scalar integer. The default is 2 . The first register holds zero symbols.

## InitialConditions

Initial conditions of shift registers
Specify the values that are initially stored in each shift register as a numeric scalar or vector. You do not need to specify a value for the first shift register, which has zero delay. The default is 0 . The value of the first element of this property is unimportant because the first shift register has zero delay. If you set this property to a scalar, then all shift registers, except the first one, store the same
specified value. If you set it to a column vector with length equal to the value of the NumRegisters property, then the $i$-th shift register stores the $i$-th element of the specified vector.

```
Methods
clone
getNumInputs
getNumOutputs
isLocked
release
reset
step
Create convolutional interleaver object with same property values
Number of expected inputs to step method
getNumOutputs
isLocked
release
reset
step
Number of outputs from step method
Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes
Reset states of the convolutional interleaver object
Permute input symbols using shift registers
```


## Examples Interleave and deinterleave random data.

```
hInt = comm.ConvolutionalInterleaver('NumRegisters', 3, ...
```

hInt = comm.ConvolutionalInterleaver('NumRegisters', 3, ...
'RegisterLengthStep', 2, ...
'RegisterLengthStep', 2, ...
'InitialConditions', [ -1 -2 -3]');
'InitialConditions', [ -1 -2 -3]');
hDeInt = comm.ConvolutionalDeinterleaver('NumRegisters', 3, ...
hDeInt = comm.ConvolutionalDeinterleaver('NumRegisters', 3, ...
'RegisterLengthStep', 2, ...
'RegisterLengthStep', 2, ...
'InitialConditions', [-1 -2 -3]');
'InitialConditions', [-1 -2 -3]');
data = (0:20)';
data = (0:20)';
intrlvData = step(hInt, data);
intrlvData = step(hInt, data);
deintrlvData = step(hDeInt, intrlvData);
deintrlvData = step(hDeInt, intrlvData);
% compare the original sequence, interleaved sequence and restored
% compare the original sequence, interleaved sequence and restored
[data, intrlvData, deintrlvData]

```
[data, intrlvData, deintrlvData]
```


# Algorithms This object implements the algorithm, inputs, and outputs described on the Convolutional Interleaver block reference page. The object properties correspond to the block parameters. <br> See Also <br> comm. ConvolutionalDeinterleaver | comm.MultiplexedInterleaver 

Purpose Create convolutional interleaver object with same property values

## Syntax

Description
$\mathrm{C}=$ clone $(\mathrm{H})$ creates a Convolutionalinterleaver object C , with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.ConvolutionalInterleaver.getNumInputs

## Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose $\quad$ Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF = isLocked (H) |
| Description $\quad$TF = isLocked $(H)$ returns the locked status, TF of the <br> ConvolutionalInterleaver System object. |  |
| The isLocked method returns a logical value that indicates whether <br> input attributes and nontunable properties for the object are locked. The <br> object performs an internal initialization the first time the step method <br> is executed. This initialization locks nontunable properties and input <br> specifications, such as dimensions, complexity, and data type of the <br> input data. After locking, the isLocked method returns a true value. |  |

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of the convolutional interleaver object

## Syntax $\quad \operatorname{reset}(H)$

Description reset $(H)$ resets the states of the ConvolutionalInterleaver object, H .

Purpose $\quad$ Permute input symbols using shift registers

$$
\text { Syntax } \quad Y=\operatorname{step}(H, X)
$$

Description
$Y=\operatorname{step}(H, X)$ permutes input sequence, $X$, and returns interleaved sequence, $Y$. The input $X$ must be a column vector. The data type can be numeric, logical, or fixed-point (fi objects). Y has the same data type as X. The convolutional interleaver object uses a set of $N$ shift registers, where $N$ is the value specified by the NumRegisters property. The object sets the delay value of the $k$-th shift register to the product of ( $k-1$ ) and the RegisterLengthStep property value. With each new input symbol, a commutator switches to a new register and the new symbol shifts in while the oldest symbol in that register shifts out. When the commutator reaches the $N$-th register and the next new input occurs, it returns to the first register.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

Construction $H=$ comm.CPFSKDemodulator creates a demodulator System object, H .

## Purpose <br> Description

## Properties

Demodulate using CPFSK method and Viterbi algorithm
The CPFSKDemodulator object demodulates a signal that was modulated using the continuous phase frequency shift keying method. The input is a baseband representation of the modulated signal. This object demodulates the input continuous phase frequency shift keying (CPFSK) modulated data using the Viterbi algorithm.

H = comm.CPFSKDemodulator (Name, Value) creates a CPFSK demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.CPFSKDemodulator (M,Name, Value) creates a CPFSK demodulator object, H. This object has the ModulationOrder property set to M , and the other specified properties set to the specified values.

## ModulationOrder

Size of symbol alphabet
Specify the size of the symbol alphabet. The value of this property requires a power of two, real, integer scalar. The default is 4 .

## BitOutput

Output data as bits
Specify whether the output consists of groups of bits or integer values. The default is false.

When you set this property to false, the step method outputs a column vector of length equal to N/SamplesPerSymbol and with elements that are integers between -(ModulationOrder-1) and ModulationOrder-1. In this case, $N$, is the length of the input signal, which indicates the number of input baseband modulated symbols.

When you set this property to true, the step method outputs a binary column vector of length equal to $P \times$ (N/SamplesPerSymbol), where $P=\log 2$ (ModulationOrder). The output contains length $-P$ bit words. In this scenario, the object first maps each demodulated symbol to an odd integer value, $K$, between -(ModulationOrder-1) and ModulationOrder-1. The object then maps $K$ to the nonnegative integer ( $K+$ ModulationOrder -1 )/2. Finally, the object maps each nonnegative integer to a length- $P$ binary word, using the mapping specified in the SymbolMapping property.

## SymbolMapping

Symbol encoding
Specify the mapping of the modulated symbols as one of Binary | Gray. The default is Binary. This property determines how the object maps each demodulated integer symbol value (in the range 0 and ModulationOrder-1) to a $P$-length bit word, where $P=$ ModulationOrder(ModulationOrder).

When you set this property to Binary, the object uses a natural binary-coded ordering.

When you set this property to Gray, the object uses a Gray-coded ordering.
This property applies when you set the BitOutput property to true.

## ModulationIndex

Modulation index
Specify the modulation index. The default is 0.5 . The value of this property can be a scalar, $h$, or a column vector, $\left[h_{0}, h_{1}, \ldots . h_{\mathrm{H}-1}\right]$ where $\mathrm{H}-1$ represents the length of the column vector.
When $h_{\mathrm{i}}$ varies from interval to interval, the object operates in multi-h. When the object operates in multi-h, $h_{\mathrm{i}}$ must be a rational number.

## InitialPhaseOffset

Initial phase offset
Specify the initial phase offset of the input modulated waveform in radians as a real, numeric scalar. The default is 0 .

## SamplesPerSymbol

Number of samples per input symbol
Specify the expected number of samples per input symbol as a positive, integer scalar. The default is 8 .

## TracebackDepth

Traceback depth for Viterbi algorithm
Specify the number of trellis branches that the Viterbi algorithm uses to construct each traceback path as a positive, integer scalar. The default is 16 . The value of this property is also the value of the output delay. That value is the number of zero symbols that precede the first meaningful demodulated symbol in the output.

## OutputDataType

Data type of output
Specify the output data type as one of int8 | int16 | int32 | double, when you set the BitOutput property to false. The default is double.

When you set the BitOutput property to true, specify the output data type as one of logical \| double.

| Methods | clone | Create CPFSK demodulator <br> object with same property values |
| :--- | :--- | :--- |
| getNumInputs | Number of expected inputs to <br> step method |  |
|  | getNumOutputs | Number of outputs from step <br> method |


| isLocked | Locked status for input attributes <br> and nontunable properties |
| :--- | :--- |
| release | Allow property value and input <br> characteristics changes |
| reset | Reset states of CPFSK <br> demodulator object |
| step | Demodulate using CPFSK <br> method and Viterbi algorithm |

## Examples Modulate and demodulate a signal using CPFSK modulation with Gray

 mapping and bit inputs.```
hMod = comm.CPFSKModulator(8, 'BitInput', true, ...
    'SymbolMapping', 'Gray');
hAWGN = comm.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)','SNR',O);
hDemod = comm.CPFSKDemodulator(8, 'BitOutput', true, ...
    'SymbolMapping', 'Gray');
% Create an error rate calculator, account for the delay caused by th
% Viterbi algorithm.
delay = log2(hDemod.ModulationOrder)*hDemod.TracebackDepth;
hError = comm.ErrorRate('ReceiveDelay', delay);
for counter = 1:100
    % Transmit 100 3-bit words
    data = randi([0 1],300,1);
    modSignal = step(hMod, data);
    noisySignal = step(hAWGN, modSignal);
    receivedData = step(hDemod, noisySignal);
    errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```


# Algorithms This object implements the algorithm, inputs, and outputs described on the CPFSK Demodulator Baseband block reference page. The object properties correspond to the block parameters. <br> See Also comm.CPFSKModulator | comm.CPMModulator | comm.CPMDemodulator 

## comm.CPFSKDemodulator.clone

Purpose Create CPFSK demodulator object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a CPFSKDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.CPFSKDemodulator.getNuminputs

## Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs(H)

## comm.CPFSKDemodulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF = isLocked (H) |
| Description $\quad$TF = isLocked $(H)$ returns the locked status, TF of the <br> CPFSKDemodulator System object. |  |
| The isLocked method returns a logical value that indicates whether <br> input attributes and nontunable properties for the object are locked. The <br> object performs an internal initialization the first time the step method <br> is executed. This initialization locks nontunable properties and input <br> specifications, such as dimensions, complexity, and data type of the <br> input data. After locking, the isLocked method returns a true value. |  |

## comm.CPFSKDemodulator.release

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of CPFSK demodulator object
Syntax ..... reset (H)
Description reset $(\mathrm{H})$ resets the states of the CPFSKDemodulator object, H .

## comm.CPFSKDemodulator.step

Purpose Demodulate using CPFSK method and Viterbi algorithm

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description $\quad Y=\operatorname{step}(H, X)$ demodulates input data, $X$, with the CPFSK demodulator System object, $H$, and returns $Y$. Input $X$ must be a double or single precision, column vector with a length equal to an integer multiple of the number of samples per symbol specified in the SamplesPerSymbol property. Depending on the BitOutput property value, output $Y$ can be integer or bit valued.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Modulate using CPFSK method

## Construction <br> H = comm.CPFSKModulator creates a modulator System object, H. This

Description

## Properties

The CPFSKModulator object modulates using the continuous phase frequency shift keying method. The output is a baseband representation of the modulated signal. object modulates the input signal using the continuous phase frequency shift keying (CPFSK) modulation method.

H = comm.CPFSKModulator(Name, Value) creates a CPFSK modulator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.CPFSKModulator (M,Name, Value) creates a CPFSK modulator object, H . This object has the ModulationOrder property set to $M$, and the other specified properties set to the specified values.

## ModulationOrder

Size of symbol alphabet
Specify the size of the symbol alphabet. The value of this property requires a power of two, real, integer scalar. The default is 4 .

## BitInput

Assume bit inputs
Specify whether the input is bits or integers. The default is false. When you set this property to false, the step method input must be a double-precision or signed integer data type column vector. This vector comprises odd integer values between -(ModulationOrder-1) and ModulationOrder-1.

When you set this property to true, the step method input must be a column vector of $P$-length bit words, where $P$ $=\log 2$ (ModulationOrder). The input data must have a doubleprecision or logical data type. The object maps each
bit word to an integer K between 0 and ModulationOrder-1, using the mapping specified in the SymbolMapping property. The object then maps the integer $K$ to the intermediate value $2 K$-(ModulationOrder-1) and proceeds as in the case when you set the BitInput property to false.

## SymbolMapping

Symbol encoding
Specify the mapping of bit inputs as one of Binary | Gray. The default is Binary. This property determines how the object maps each input $P$-length bit word, where $P=\log 2$ (ModulationOrder), to an integer between 0 and ModulationOrder-1.

When you set this property to Binary, the object uses a natural binary-coded ordering.

When you set this property to Gray, the object uses a Gray-coded ordering.

This property applies when you set the BitInput property to true.

## ModulationIndex

Modulation index
Specify the modulation index. The default is 0.5 . The value of this property can be a scalar, $h$, or a column vector, $\left[h_{0}, h_{1}, \ldots . h_{\mathrm{H}-1}\right]$
where $\mathrm{H}-1$ represents the length of the column vector.
When $h_{\mathrm{i}}$ varies from interval to interval, the object operates in multi-h. When the object operates in multi-h, $h_{\mathrm{i}}$ must be a rational number.

## InitialPhaseOffset

Initial phase offset
Specify the initial phase of the modulated waveform in radians as a real, numeric scalar. The default is 0 .

## SamplesPerSymbol

Number of samples per output symbol
Specify the upsampling factor at the output as a real, positive, integer scalar. The default is 8 . The upsampling factor is the number of output samples that the step method produces for each input sample.

## OutputDataType

Data type of output
Specify output data type as one of double | single. The default is double.

| Methods | clone | Create CPFSK modulator object with same property values |
| :---: | :---: | :---: |
|  | getNumInputs | Number of expected inputs to step method |
|  | getNumOutputs | Number of outputs from step method |
|  | isLocked | Locked status for input attributes and nontunable properties |
|  | release | Allow property value and input characteristics changes |
|  | reset | Reset states of CPFSK modulator object |
|  | step | Modulate using CPFSK method |
| Examples | Modulate and dem mapping and bit in | using CPFSK modulation with Gray |
|  | $\begin{aligned} & \text { hMod }=\text { comm } \\ & \text { hAWGN }=\text { comr } \end{aligned}$ | ```(8, 'BitInput', true, ... olMapping', 'Gray'); 'NoiseMethod', ... al to noise ratio (SNR)','SNR``` |

```
    hDemod = comm.CPFSKDemodulator(8, 'BitOutput', true, ...
    'SymbolMapping', 'Gray');
% Create an error rate calculator, account for the delay caused by the
    delay = log2(hDemod.ModulationOrder)*hDemod.TracebackDepth;
    hError = comm.ErrorRate('ReceiveDelay', delay);
    for counter = 1:100
        % Transmit 100 3-bit words
        data = randi([0 1],300,1);
        modSignal = step(hMod, data);
        noisySignal = step(hAWGN, modSignal);
        receivedData = step(hDemod, noisySignal);
        errorStats = step(hError, data, receivedData);
    end
    fprintf('Error rate = %f\nNumber of errors = %d\n', ...
        errorStats(1), errorStats(2))
```

Algorithms<br>This object implements the algorithm, inputs, and outputs described on the CPFSK Modulator Baseband block reference page. The object properties correspond to the block parameters.<br>See Also comm.CPFSKDemodulator | comm.CPMModulator |<br>comm. CPMDemodulator

# Purpose <br> Create CPFSK modulator object with same property values 

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a CPFSKModulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.CPFSKModulator.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.CPFSKModulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the CPFSKModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.CPFSKModulator.release

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of CPFSK modulator object

## Syntax reset (H)

Description reset (H) resets the states of the CPFSKModulator object, H.

| Purpose | Modulate using CPFSK method |
| :--- | :--- |
| Syntax | $Y=\operatorname{step}(H, X)$ |
| Description | $Y=\operatorname{step}(H, X)$ modulates input data, $X$, with the CPFSK modulator <br> System object, $H$. It returns the baseband modulated output, $Y$. <br> Depending on the value of the Bit Input property, input $X$ can be an <br> integer or bit valued column vector with data types double, signed <br> integer, or logical. The length of output vector, $Y$, is equal to the number <br> of input samples times the number of samples per symbol specified in <br> the SamplesPerSymbol property. |

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Recover carrier phase of baseband CPM signal

## Construction

## Properties

H = comm.CPMCarrierPhaseSynchronizer creates a CPM carrier phase synchronizer System object, H. This object recovers the carrier phase of a baseband continuous phase modulation (CPM), minimum shift keying (MSK), continuous phase frequency shift keying (CPFSK), or Gaussian minimum shift keying (GMSK) modulated signal using the 2P-power method.
H = comm.CPMCarrierPhaseSynchronizer(Name, Value) creates a CPM carrier phase synchronizer object, H This object has each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).
H = comm.CPMCarrierPhaseSynchronizer(HALFPOW, Name, Value) creates a CPM carrier phase synchronizer object, H. This object has the $P$ property set to HALFPOW, and the other specified properties set to the specified values.
The CPMCarrierPhaseSynchronizer object recovers the carrier phase of the input signal using the $2 P$-Power method. This feedforward method is clock aided, but not data aided. The method is suitable for systems that use certain types of baseband modulation. These types include: continuous phase modulation (CPM), minimum shift keying (MSK), continuous phase frequency shift keying (CPFSK), and Gaussian minimum shift keying (GMSK).

## P

Denominator of CPM modulation index
Specify the denominator of the CPM modulation index of the input signal as a real positive scalar integer value of data type single or double. The default is 2 . This property is tunable.

## ObservationInterval

Number of symbols where carrier phase assumed constant

Specify the observation interval as a real positive scalar integer value of data type single or double. The default is 100 .

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
| isLocked |  |
| release |  |
| reset |  |
| step |  |

## Examples Recover carrier phase of a CPM signal using 2P-power method.

```
M = 16;
P = 2;
phOffset = 10 *pi/180; % in radians
numSamples = 100;
% Create CPM modulator System object
    hMod = comm.CPMModulator(M, 'InitialPhaseOffset',phOffset, ...
            'BitInput',true, 'ModulationIndex',1/P, 'SamplesPerSymbol',1);
% Create CPM carrier phase synchronizer System object
    hSync = comm.CPMCarrierPhaseSynchronizer(P,...
            'ObservationInterval',numSamples);
% Generate random binary data
            data = randi([0 1],numSamples*log2(M),1);
```

```
% Modulate random data and add carrier phase
        modData = step(hMod, data);
% Recover the carrier phase
    [recSig phEst] = step(hSync, modData);
    fprintf('The carrier phase is estimated to be %g degrees.\n', phEst);
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the CPM Phase Recovery block reference page. The object properties correspond to the block parameters.

See Also
comm.PSKCarrierPhaseSynchronizer | comm.CPMModulator

# Purpose <br> Create CPM carrier phase synchronizer object with same values 

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a CPMCarrierPhaseSynchronizer object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.CPMCarrierPhaseSynchronizer.getNumOutputs

| Purpose | Number of outputs from step method |
| :--- | :--- |
| Syntax | N = getNumOutputs (H) |
| Description | N = getNumOutputs $(H)$ returns the number of outputs, N, from the <br> step method. This value will change if any properties that turn inputs <br> on or off are changed. |

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the CPMCarrierPhaseSynchronizer System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release (H) |
| Description | release (H) Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset states of the CPM carrier phase synchronizer object |
| :--- | :--- |
| Syntax | $\operatorname{reset}(H)$ |
| Description | reset (H) resets the states of the CPMCarrierPhaseSynchronizer <br> object, H. |

# Purpose Recover carrier phase of baseband CPM signal <br> <br> Syntax <br> <br> Syntax <br> <br> [ $\mathrm{Y}, \mathrm{PH}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$ 

 <br> <br> [ $\mathrm{Y}, \mathrm{PH}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$}

Description
$[\mathrm{Y}, \mathrm{PH}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$ recovers the carrier phase of the input signal, $X$, and returns the phase corrected signal, $Y$, and the carrier phase estimate (in degrees), PH. X must be a complex scalar or column vector input signal of data type single or double.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Demodulate using CPM method and Viterbi algorithm

Description

## Construction

## Properties

The CPMDemodulator object demodulates a signal that was modulated using continuous phase modulation. The input is a baseband representation of the modulated signal.

H = comm.CPMDemodulator creates a demodulator System object, H. This object demodulates the input continuous phase modulated (CPM) data using the Viterbi algorithm.

H = comm.CPMDemodulator (Name, Value) creates a CPM demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.CPMDemodulator(M,Name, Value) creates a CPM demodulator object, H , with the ModulationOrder property set to M , and the other specified properties set to the specified values.

## ModulationOrder

Size of symbol alphabet
Specify the size of the symbol alphabet. The value of this property requires a power of two, real, integer scalar. The default is 4 .

## BitOutput

Output data as bits
Specify whether the output consists of groups of bits or integer values. The default is false.

When you set this property to false, the step method outputs a column vector of length equal to N/SamplesPerSymbol and with elements that are integers between -(ModulationOrder-1) and ModulationOrder-1. Here, $N$, is the length of the input signal which indicates the number of input baseband modulated symbols.

When you set this property to true, the step method outputs a binary column vector of length equal to $P \times$ (N/SamplesPerSymbol),
where $P=\log 2$ (ModulationOrder). The output contains length $-P$ bit words. In this scenario, the object first maps each demodulated symbol to an odd integer value, $K$, between -(ModulationOrder-1) and ModulationOrder -1 . The object then maps $K$ to the nonnegative integer ( $K+$ ModulationOrder -1 )/2. Finally, the object maps each nonnegative integer to a length- $P$ binary word, using the mapping specified in the SymbolMapping property.

## SymbolMapping

Symbol encoding
Specify the mapping of the demodulated symbols as one of Binary | Gray. The default is Binary. This property determines how the object maps each demodulated integer symbol value (in the range 0 and ModulationOrder-1) to a $P$-length bit word, where $P$ $=\log 2$ (ModulationOrder).

When you set this property to Binary, the object uses a natural binary-coded ordering.

When you set this property to Gray, the object uses a Gray-coded ordering.

This property applies when you set the BitOutput property to true.

## ModulationIndex

## Modulation index

Specify the modulation index. The default is 0.5 . The value of this property can be a scalar, $h$, or a column vector, $\left[h_{0}, h_{1}, \ldots . h_{\mathrm{H}-1}\right]$
where $\mathrm{H}-1$ represents the length of the column vector.
When $h_{\mathrm{i}}$ varies from interval to interval, the object operates in multi-h. When the object operates in multi-h, $h_{\mathrm{i}}$ must be a rational number.

## FrequencyPulse

Frequency pulse shape

Specify the type of pulse shaping that the modulator has used to smooth the phase transitions of the input modulated signal as one of Rectangular | Raised Cosine | Spectral Raised Cosine | Gaussian | Tamed FM. The default is Rectangular.

## MainLobeDuration

Main lobe duration of spectral raised cosine pulse
Specify, in number of symbol intervals, the duration of the largest lobe of the spectral raised cosine pulse. This value is the value that the modulatorused to pulse-shape the input modulated signal. The default is 1 . This property requires a real, positive, integer scalar. This property applies when you set the FrequencyPulse property to Spectral Raised Cosine.

## RolloffFactor

Rolloff factor of spectral raised cosine pulse
Specify the roll off factor of the spectral raised cosine pulse. This value is the value that the modulator used to pulse-shape the input modulated signal. The default is 0.2 . This property requires a real scalar between 0 and 1 . This property applies when you set the FrequencyPulse property to Spectral Raised Cosine.

## BandwidthTimeProduct

Product of bandwidth and symbol time of Gaussian pulse
Specify the product of bandwidth and symbol time for the Gaussian pulse shape. This value is the value that the modulator used to pulse-shape the input modulated signal. The default is 0.3 . This property requires a real, positive scalar. This property applies when you set the FrequencyPulse property to Gaussian.

## PulseLength

Pulse length
Specify the length of the frequency pulse shape in symbol intervals. The value of this property requires a real positive integer. The default is 1 .

## SymbolPrehistory

Symbol prehistory
Specify the data symbols used by the modulator prior to the first call to the step method. The default is 1 . This property requires a scalar or vector with odd integer elements between -(ModulationOrder-1) and (ModulationOrder-1). If the value is a vector, then its length must be one less than the value in the PulseLength property.

## InitialPhaseOffset

Initial phase offset
Specify the initial phase offset of the input modulated waveform in radians as a real, numeric scalar. The default is 0 .

## SamplesPerSymbol

Number of samples per input symbol
Specify the expected number of samples per input symbol as a positive, integer scalar. The default is 8 .

## TracebackDepth

Traceback depth for Viterbi algorithm
Specify the number of trellis branches that the Viterbi algorithm uses to construct each traceback path as a positive, integer scalar. The default is 16 . The value of this property is also the output delay, which is the number of zero symbols that precede the first meaningful demodulated symbol in the output.

## OutputDataType

Data type of output
Specify the output data type as one of int8 | int16 | int32 | double, when you set the BitOutput property to false. When you set the BitOutput property to true, specify the output data type as one of logical | double. The default is double.

## comm.CPMDemodulator

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>reset<br>step

Create CPM demodulator object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of CPM demodulator object

Demodulate using CPM method and Viterbi algorithm

## Examples Modulate and demodulate a signal using CPM modulation with Gray

 mapping and bit inputs.```
hMod \(=\) comm.CPMModulator(8, 'BitInput', true, ...
    'SymbolMapping', 'Gray');
hAWGN = comm.AWGNChannel('NoiseMethod', ...
    Signal to noise ratio (SNR)','SNR',0);
hDemod \(=\) comm.CPMDemodulator(8, 'BitOutput', true, ...
```

                            'SymbolMapping', 'Gray');
    \% Create an error rate calculator, account for the delay caused by the Vi
delay = log2(hDemod.ModulationOrder)*hDemod.TracebackDepth;
hError = comm.ErrorRate('ReceiveDelay', delay);
for counter = 1:100
\% Transmit 1003 -bit words
data = randi([0 1],300,1);
modSignal = step(hMod, data);
noisySignal = step(hAWGN, modSignal);
receivedData $=$ step(hDemod, noisySignal);

```
    errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the CPM Demodulator Baseband block reference page. The object properties correspond to the block parameters.

comm.CPMModulator | comm.CPFSKDemodulator |<br>comm.MSKDemodulator | comm.GMSKDemodulator

## comm.CPMDemodulator.clone

Purpose Create CPM demodulator object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a CPMDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.CPMDemodulator.getNuminputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.CPMDemodulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.CPMDemodulator.isLocked

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the CPMDemodulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset states of CPM demodulator object |
| :--- | :--- |
| Syntax | reset $(H)$ |
| Description | reset $(H)$ resets the states of the CPMDemodulator object, H. |

Purpose Demodulate using CPM method and Viterbi algorithm

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description $\quad Y=\operatorname{step}(H, X)$ demodulates input data, $X$, with the CPM demodulator System object, H, and returns Y. X must be a double or single precision, column vector with a length equal to an integer multiple of the number of samples per symbol specified in the SamplesPerSymbol property. Depending on the BitOutput property value, output $Y$ can be integer or bit valued.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Modulate using CPM method |
| :---: | :---: |
| Description | The CPMModulator object modulates using continuous phase modulation. The output is a baseband representation of the modulated signal. |
| Construction | H = comm.CPMModulator creates a modulator System object, H. This object modulates the input signal using the continuous phase modulation (CPM) method. |
|  | H = comm. CPMModulator (Name, Value) creates a CPM modulator object, H. This object has each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN). |
|  | H = comm.CPMModulator (M,Name, Value) creates a CPM modulator object, H , with the ModulationOrder property set to M and the other specified properties set to the specified values. |
| Properties | ModulationOrder |
|  | Size of symbol alphabet |
|  | Specify the size of the symbol alphabet. The value of this property must be a power of two, real, integer scalar. The default is 4 . |
|  | BitInput |
|  | Assume bit inputs |
|  | Specify whether the input is bits or integers. The default is false. |
|  | When you set this property to false, the step method input requires double-precision or signed integer data type column vector. This vector must comprise odd integer values between -(ModulationOrder-1) and ModulationOrder-1. |
|  | When you set this property to true, the step method input requires a column vector of $P$-length bit words, where $P$ $=\log 2$ (ModulationOrder). The input data must have a double-precision or logical data type. The object maps each |

bit word to an integer $K$ between 0 and ModulationOrder-1, using the mapping specified in the SymbolMapping property. The object then maps the integer $K$ to the intermediate value $2 K$-(ModulationOrder-1) and proceeds as in the case when BitInput is false.

## SymbolMapping

Symbol encoding
Specify the mapping of bit inputs as one of Binary | Gray. The default is Binary. This property determines how the object maps each input $P$-length bit word, where $P=\log 2$ (ModulationOrder), to an integer between 0 and ModulationOrder-1.

When you set this property to Binary, the object uses a natural binary-coded ordering.
When you set this property to Gray, the object uses a Gray-coded ordering.

This property applies when you set the BitInput property to true.

## ModulationIndex

Modulation index
Specify the modulation index. The default is 0.5 . The value of this property can be a scalar, $h$, or a column vector, $\left[h_{0}, h_{1}, \ldots . h_{\mathrm{H}-1}\right]$
where $\mathrm{H}-1$ represents the length of the column vector.
When $h_{\mathrm{i}}$ varies from interval to interval, the object operates in multi-h. When the object operates in multi-h, $h_{\mathrm{i}}$ must be a rational number.

## FrequencyPulse

Frequency pulse shape
Specify the type of pulse shaping that the modulator uses to smooth the phase transitions of the modulated signal. Choose from Rectangular | Raised Cosine | Spectral Raised Cosine | Gaussian | Tamed FM. The default is Rectangular.

## MainLobeDuration

Main lobe duration of spectral raised cosine pulse
Specify, in number of symbol intervals, the duration of the largest lobe of the spectral raised cosine pulse. The default is 1 . This property requires a real, positive, integer scalar. This property applies when you set the FrequencyPulse property to Spectral Raised Cosine.

## RolloffFactor

Rolloff factor of spectral raised cosine pulse
Specify the rolloff factor of the spectral raised cosine pulse. The default is 0.2 . This property requires a real scalar between 0 and 1. This property applies when you set the FrequencyPulse property to Spectral Raised Cosine.

## BandwidthTimeProduct

Product of bandwidth and symbol time of Gaussian pulse
Specify the product of bandwidth and symbol time for the Gaussian pulse shape. The default is 0.3 . This property requires a real, positive scalar. This property applies when you set the FrequencyPulse property to Gaussian.

## PulseLength

Pulse length
Specify the length of the frequency pulse shape in symbol intervals. The value of this property requires a real, positive integer. The default is 1 .

## SymbolPrehistory

Symbol prehistory
Specify the data symbols used by the modulator prior to the first call to the step method in reverse chronological order. The default is 1 . This property requires a scalar or vector with odd integer elements between -(ModulationOrder-1) and
(ModulationOrder-1). If the value is a vector, then its length must be one less than the value in the PulseLength property.

## InitialPhaseOffset

Initial phase offset
Specify the initial phase of the modulated waveform in radians as a real, numeric scalar. The default is 0 .

## SamplesPerSymbol

Number of samples per output symbol
Specify the upsampling factor at the output as a real, positive, integer scalar. The default is 8 . The upsampling factor is the number of output samples that the step method produces for each input sample.

## OutputDataType

Data type of output
Specify output data type as one of double | single. The default is double.

| Methods | clone | Create CPM modulator object <br> with same property values |
| :--- | :--- | :--- |
| getNumInputs | getNumOutputs | Number of expected inputs to <br> step method |
| isLocked | Number of outputs from step <br> method |  |
| release | Locked status for input attributes <br> and nontunable properties |  |
|  | Allow property value and input <br> characteristics changes |  |

```
reset
step
```

Reset states of CPM modulator object

Modulate using CPM method

## Examples

## Algorithms

See Also

```
Modulate and demodulate a signal using CPM modulation with Gray mapping and bit inputs.
```

```
    hMod = comm.CPMModulator(8, 'BitInput', true, ...
```

    hMod = comm.CPMModulator(8, 'BitInput', true, ...
    'SymbolMapping', 'Gray');
    'SymbolMapping', 'Gray');
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
    Signal to noise ratio (SNR)','SNR',O);
    Signal to noise ratio (SNR)','SNR',O);
    hDemod = comm.CPMDemodulator(8, 'BitOutput', true, ...
hDemod = comm.CPMDemodulator(8, 'BitOutput', true, ...
SymbolMapping', 'Gray');
% Create an error rate calculator, account for the delay caused by the
delay = log2(hDemod.ModulationOrder)*hDemod.TracebackDepth;
hError = comm.ErrorRate('ReceiveDelay', delay);
for counter = 1:100
% Transmit 100 3-bit words
data = randi([0 1],300,1);
modSignal = step(hMod, data);
noisySignal = step(hAWGN, modSignal);
receivedData = step(hDemod, noisySignal);
errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
errorStats(1), errorStats(2))

```

This object implements the algorithm, inputs, and outputs described on the CPM Modulator Baseband block reference page. The object properties correspond to the block parameters.

\footnotetext{
comm.CPMDemodulator | comm.CPFSKModulator | comm.MSKModulator | comm.GMSKModulator
}

\section*{comm.CPMModulator.clone}

Purpose Create CPM modulator object with same property values

\section*{Syntax \\ C = clone( H )}

Description \(\quad C=\) clone \((H)\) creates a CPMModulator object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.CPMModulator.getNuminputs}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.CPMModulator.getNumOutputs}

Purpose \(\quad\) Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs (H)
Description \(\quad N=\) getNum0utputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.CPMModulator.isLocked}

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the CPMModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

\section*{comm.CPMModulator.release}

Purpose Allow property value and input characteristics changes

\section*{Syntax \\ release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{ll} 
Purpose & Reset states of CPM modulator object \\
Syntax & reset \((H)\) \\
Description & reset \((H)\) resets the states of the CPMModulator object, H.
\end{tabular}

\section*{Purpose Modulate using CPM method}

\section*{Syntax \(\quad Y=\operatorname{step}(H, X)\)}

Description \(\quad Y=\operatorname{step}(H, X)\) modulates input data, \(X\), with the CPM modulator System object, H. It returns the baseband modulated output, Y. Depending on the value of the BitInput property, input \(X\) can be an integer or bit valued column vector with data types double, signed integer, or logical. The length of output vector, Y , is equal to the number of input samples times the number of samples per symbol specified in the SamplesPerSymbol property.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose \\ Detect errors in input data using CRC}

Description
Construction

The CRCDetector object computes checksums for its entire input frame.
H = comm. CRCDetector creates a cyclic redundancy code (CRC) detector System object, H. This object detects errors in the input data according to a specified generator polynomial.

H = comm.CRCDetector(Name,Value) creates a CRC detector object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm. CRCDetector (POLY, Name, Value) creates a CRC detector object, H. This object has the Polynomial property set to POLY, and the other specified properties set to the specified values.

\section*{Properties}

\section*{Polynomial}

Generator polynomial
Specify the generator polynomial as a binary or integer row vector, with coefficients in descending order of powers. The default is \(\left[\begin{array}{llllllllllllll}1 & 0 & 0 & 0 & 1 & 0 & 0 & 0 & 0 & 0 & 0 & 1 & 0 & 0\end{array} 0<01\right]\), which is equivalent to vector [ \(\left.\begin{array}{lll}16 & 12 & 5\end{array} 0\right]\). If you set this property to a binary vector, its length must equal the degree of the polynomial plus 1. If you set this property to an integer vector, its value must contain the powers of the nonzero terms of the polynomial. For example, \(\left[\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0\end{array}\right]\) and [ 8 2 0 ] represent the same polynomial, \(g(z)=z^{8}+z^{2}+1\). The following table lists commonly used generator polynomials.
\begin{tabular}{|c|c|}
\hline CRC methoc & Generator polynomial \\
\hline CRC-32 & \(\left[\begin{array}{lllllllllllllll}32 & 26 & 23 & 22 & 16 & 12 & 11 & 10 & 8 & 7 & 5 & 4 & 2 & 1 & 0\end{array}\right]\) \\
\hline CRC-24 & \(\left[\begin{array}{lllllll}24 & 23 & 14 & 12 & 8 & 0\end{array}\right.\) \\
\hline
\end{tabular}
\begin{tabular}{|c|c|}
\hline CRC method & Generator polynomial \\
\hline CRC-16 & \(\left[\begin{array}{llll}16 & 15 & 2 & 0\end{array}\right]\) \\
\hline Reverse CRC-16 & [ \(\left.\begin{array}{llll}16 & 14 & 1 & 0\end{array}\right]\) \\
\hline CRC-8 & \(\left[\begin{array}{llllll}8 & 7 & 6 & 4 & 2 & 0\end{array}\right]\) \\
\hline CRC-4 & \(\left[\begin{array}{lllll}4 & 3 & 2 & 1 & 0\end{array}\right]\) \\
\hline
\end{tabular}

\section*{InitialConditions}

Initial conditions of shift register
Specify the initial conditions of the shift register as a binary, double or single precision data type scalar or vector. The default is 0 . The vector length is the degree of the generator polynomial that you specify in the Polynomial property. When you specify initial conditions as a scalar, the object expands the value to a row vector of length equal to the degree of the generator polynomial.

\section*{DirectMethod}

Direct method (logical)
When you set this property to true, the object uses the direct algorithm for CRC checksum calculations. When you set this property to false, the object uses the non-direct algorithm for CRC checksum calculations. The default value for this property is false.

Refer to the Communications System Toolbox -> System Design -> Error Detection and Correction -> Cyclic Redundancy Check Coding -> CRC Algorithm section to learn more about the direct and non-direct algorithms.

\section*{ReflectInputBytes}

Reflect input bytes

Set this property to true to flip the input data on a bytewise basis prior to entering the data into the shift register. When you set this property to true, the input frame length divided by the ChecksumsPerFrame property value minus the degree of the generator polynomial, which you specify in the Polynomial property, must be an integer multiple of 8 . The default value of this property is false.

\section*{ReflectChecksums}

Reflect checksums before final XOR
When you set this property to true, the object flips the CRC checksums around their centers after the input data are completely through the shift register. The default value of this property is false.

\section*{FinalXOR}

Final XOR value
Specify the value with which the CRC checksum is to be XORed as a binary scalar or vector. The object applies the XOR operation just prior to appending the input data. The vector length is the degree of the generator polynomial that you specify in the Polynomial property. When you specify the final XOR value as a scalar, the object expands the value to a row vector with a length equal to the degree of the generator polynomial. The default value of this property is 0 , which is equivalent to no XOR operation.

\section*{ChecksumsPerFrame}

Number of checksums per input frame
Specify the number of checksums available at each input frame. The default is 1 . If the length of the input frame to the step method equals \(N\) and the degree of the generator polynomial equals \(P\), then \(N\)-CheckSumsPerFrame \(\times P\) must be divisible by ChecksumsPerFrame. The object sets the size of the message word as \(N\)-CheckSumsPerFrame \(\times P\), after the checksum bits have been removed from the input frame. This message word corresponds
to the first output of the step method. The step method then outputs a vector, with length equal to the value that you specify in the this property.

For example, you can set the input codeword size to 16 and the generator polynomial to a degree of 3 . Then, you can set the InitialConditions property to 0 and the this property to 2 When you do so, the system object:

1 Computes two checksums of size 3. One checksum comes from the first half of the received codeword, and the other from the second half of the received codeword.

2 Concatenates the two halves of the message word as a single vector of length 10. Then, outputs this vector through the first output of the step method.

3 Outputs a length 2 binary vector through the second output of the step method.

The vector values depend on whether the computed checksums are zero. A 1 in the \(i\)-th element of the vector indicates that an error occurred in transmitting the corresponding \(i\)-th segment of the input codeword.

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked
}

Create CRC detector object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties
\begin{tabular}{ll} 
release & \begin{tabular}{l} 
Allow property value and input \\
characteristics changes
\end{tabular} \\
reset & \begin{tabular}{l} 
Reset states of CRC detector \\
object
\end{tabular} \\
step & \begin{tabular}{l} 
Detect errors in input data using \\
CRC
\end{tabular}
\end{tabular}

\section*{Examples}

Algorithms
This object implements the algorithm, inputs, and outputs described on the CRC-N Syndrome Detector block reference page. The object properties correspond to the block parameters.

\section*{See Also}
comm. CRCGenerator

\section*{comm.CRCDetector.clone}

Purpose Create CRC detector object with same property values

\section*{Syntax \\ C = clone( H )}

Description \(\quad C=\) clone \((H)\) creates a CRCDetector object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.CRCDetector.getNumInputs}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description
\(N=\) getNumInputs( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.CRCDetector.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs (H)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the CRCDetector System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

\section*{comm.CRCDetector.release}

Purpose Allow property value and input characteristics changes

\section*{Syntax \\ release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{ll} 
Purpose & Reset states of CRC detector object \\
Syntax & \(\operatorname{reset}(H)\) \\
Description & reset \((H)\) resets the states of the CRCDetector object, H.
\end{tabular}

\section*{Purpose Detect errors in input data using CRC}

\section*{Syntax \(\quad[Y, E R R]=\operatorname{step}(H, X)\)}

Description
\([\mathrm{Y}, \mathrm{ERR}]=\operatorname{step}(\mathrm{H}, \mathrm{X})\) computes checksums for the entire input frame, \(X\). \(X\) must be a binary column vector and the data type can be double or logical. The step method outputs a row vector ERR, with size equal to the number of checksums that you specify in the CheckSumsPerFrame property. The elements of ERR are 0 if the checksum computation yields a zero value, and 1 otherwise. The method outputs Y , with the set of CheckSumsPerFrame message words concatenated after removing the checksums bits. The object sets the length of output Y as length \((\mathrm{X})-P \times\) CheckSumsPerFrame, where \(P\) is the order of the polynomial that you specify in the Polynomial property.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose \\ Description}

\section*{Construction}

Properties

Generate CRC code bits and append to input data

The CRCGenerator object generates cyclic redundancy code (CRC) bits for each input data frame and appends them to the frame. The input must be a binary column vector.

H = comm. CRCGenerator creates a cyclic redundancy code (CRC) generator System object, H. This object generates CRC bits according to a specified generator polynomial and appends them to the input data.

H = comm.CRCGenerator(Name, Value) creates a CRC generator object, \(H\), with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.CRCGenerator(POLY,Name, Value) creates a CRC generator object, H. This object has the Polynomial property set to POLY, and the other specified properties set to the specified values.

\section*{Polynomial}

Generator polynomial
Specify the generator polynomial as a binary or integer row vector, with coefficients in descending order of powers. The default is \(\left[\begin{array}{llllllllllllll}1 & 0 & 0 & 0 & 1 & 0 & 0 & 0 & 0 & 0 & 0 & 1 & 0 & 0\end{array} 001\right]\), which is equivalent to vector [16 1250 ]. If you set this property to a binary vector, its length must equal the degree of the polynomial plus 1. If you set this property to an integer vector, its value must contain the powers of the nonzero terms of the polynomial. For example, \(\left[\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0\end{array} 1\right.\) ] and [ 82200 represent the same
polynomial, \(g(z)=z^{8}+z^{2}+1\). The following table lists commonly used generator polynomials.
\begin{tabular}{|c|c|}
\hline CRC method & Generator polynomial \\
\hline CRC-32 & \(\left[\begin{array}{lllllllllllllll}32 & 26 & 23 & 22 & 16 & 12 & 11 & 10 & 8 & 7 & 5 & 4 & 2 & 1\end{array}\right]\) \\
\hline CRC-24 & \(\left[\begin{array}{llllll}24 & 23 & 14 & 12 & 8 & 0\end{array}\right]\) \\
\hline CRC-16 & \(\left[\begin{array}{llll}16 & 15 & 2 & 0\end{array}\right]\) \\
\hline Reversed CRC-16 & \[
d\left[\begin{array}{llll}
16 & 14 & 1 & 0
\end{array}\right]
\] \\
\hline CRC-8 & \(\left[\begin{array}{llllll}8 & 7 & 6 & 4 & 2 & 0\end{array}\right]\) \\
\hline CRC-4 & \(\left[\begin{array}{llllll}4 & 3 & 2 & 1 & 0\end{array}\right]\) \\
\hline
\end{tabular}

\section*{InitialConditions}

Initial conditions of shift register
Specify the initial conditions of the shift register as a scalar or vector with a binary, double- or single-precision data type. The default is 0 . The vector length must equal the degree of the generator polynomial that you specify in the Polynomial property. When you specify initial conditions as a scalar, the object expands the value to a row vector of length equal to the degree of the generator polynomial.

\section*{DirectMethod}

Direct method (logical)
When you set this property to true, the object uses the direct algorithm for CRC checksum calculations. When you set this property to false, the object uses the non-direct algorithm for CRC checksum calculations. The default value for this property is false.

Refer to the Communications System Toolbox -> System Design -> Error Detection and Correction -> Cyclic Redundancy Check Coding -> CRC Algorithm section to learn more about the direct and non-direct algorithms.

\section*{ReflectInputBytes}

\section*{Reflect input bytes}

Set this property to true to flip the input data on a bytewise basis prior to entering the data into the shift register. When you set this property to true, the input frame length divided by the ChecksumsPerFrame property value must be an integer multiple of 8 . The default value of this property is false.

\section*{ReflectChecksums}

Reflect checksums before final XOR
When you set this property to true, the object flips the CRC checksums around their centers after the input data are completely through the shift register. The default value of this property is false.

\section*{FinalXOR}

Final XOR value
Specify the value with which the CRC checksum is to be XORed as a binary scalar or vector. The object applies the XOR operation just prior to appending the input data. The vector length is the degree of the generator polynomial that you specify in the Polynomial property. When you specify the final XOR value as a scalar, the object expands the value to a row vector with a length equal to the degree of the generator polynomial. The default value of this property is 0 , which is equivalent to no XOR operation.

\section*{ChecksumsPerFrame}

Number of checksums per input frame
Specify the number of checksums that the object calculates for each input frame as a positive integer. The default is 1 . The integer must divide the length of each input frame evenly. The object performs the following actions:

1 Divides each input frame into ChecksumsPerFrame subframes of equal size.

2 Prefixes the initial conditions vector to each of the subframes.
3 Applies the CRC algorithm to each augmented subframe.
4 Appends the resulting checksums at the end of each subframe.
5 Outputs concatenated subframes.
For example, you can set an input frame size to 10 , the degree of the generator polynomial to 3, InitialConditions property set to 0 , and the ChecksumsPerFrame property set to 2 . When you do so, the object divides each input frame into two subframes of size 5 and appends a checksum of size 3 to each subframe. In this example, the output frame has a size \(10+2 \times 3=16\).
\(\left.\left.\begin{array}{lll}\text { Methods } & \text { clone } & \begin{array}{l}\text { Create CRC generator object with } \\ \text { same property values }\end{array} \\ \text { getNumInputs } & \begin{array}{l}\text { Number of expected inputs to } \\ \text { step method }\end{array} \\ \text { getNumOutputs } & \begin{array}{l}\text { Number of outputs from step } \\ \text { method }\end{array} \\ \text { LsLocked status for input attributes } \\ \text { and nontunable properties }\end{array}\right\} \begin{array}{l}\text { Allow property value and input } \\ \text { characteristics changes } \\ \text { Reset states of CRC generator } \\ \text { object } \\ \text { Generate CRC code bits and } \\ \text { append to input data }\end{array}\right\}\)
```

x = logical([[1 0
% Encode the message words using a CRC generator
hGen = comm.CRCGenerator([[1 0 0 1], 'ChecksumsPerFrame',2);
codeword = step(hGen, x);
% Add one bit error to each codeword
errorPattern = randerr(2,9,1).';
codewordWithError = xor(codeword, errorPattern(:));
% Decode messages with and without errors using a CRC decoder
hDetect = comm.CRCDetector([14 0 0 1], 'ChecksumsPerFrame',2);
[tx, err] = step(hDetect, codeword);
[tx1, err1] = step(hDetect, codewordWithError);
disp(err) % err is [0;0], no errors in transmitted message words
disp(err1) % err1 is [1;1], errors in both transmitted message wor

```

\section*{Algorithms}

See Also

This object implements the algorithm, inputs, and outputs described on the CRC-N Generator block reference page. The object properties correspond to the block parameters.
comm.CRCDetector

\section*{comm.CRCGenerator.clone}

Purpose Create CRC generator object with same property values

\section*{Syntax \\ C = clone(H)}

Description
\(C=\) clone \((H)\) creates a CRCGenerator object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.CRCGenerator.getNumInputs}

Purpose \(\quad\) Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.CRCGenerator.getNumOutputs}

Purpose \(\quad\) Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the CRCGenerator System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

\section*{Syntax release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{ll} 
Purpose & Reset states of CRC generator object \\
Syntax & reset \((H)\) \\
Description & reset \((H)\) resets the states of the CRCGenerator object, H.
\end{tabular}

Purpose Generate CRC code bits and append to input data

\section*{Syntax \\ \(Y=\operatorname{step}(H, X)\)}

Description \(\quad Y=\operatorname{step}(H, X)\) generates CRC checksums for an input message \(X\) and appends the checksums to \(X\). The input \(X\) must be a binary column vector and the data type can be double or logical. The length of output Y is length \((\mathrm{X})+P \times\) CheckSumsPerFrame, where \(P\) is the order of the polynomial that you specify in the Polynomial property.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Demodulate using DBPSK method}

Description

\section*{Construction}

The DBPSKDemodulator object demodulates a signal that was modulated using the differential binary phase shift keying method. The input is a baseband representation of the modulated signal.

H = comm. DBPSKDemodulator creates a demodulator System object, H.

This object demodulates the input signal using the differential binary phase shift keying (DBPSK) method.

H = comm.DBPSKDemodulator(Name, Value) creates a DBPSK demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.DBPSKDemodulator(PHASE, Name, Value) creates a DBPSK demodulator object, \(H\). This object has the PhaseRotation property set to PHASE and the other specified properties set to the specified values.

\section*{PhaseRotation}

Additional phase shift
Specify the additional phase difference between previous and current modulated bits in radians as a real scalar. The default is 0 . This value corresponds to the phase difference between previous and current modulated bits when the input is zero.

\section*{OutputDataType}

Data type of output
Specify output data type as one of Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32 | logical. The default is Full precision. When you set this property to Full precision, the output data type has the same data type as the input. In this case, that value must be a double- or single-precision data type.

\section*{comm.DBPSKDemodulator}

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release \\ reset \\ step
}

Create DBPSK demodulator object with same property values
Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of DBPSK demodulator object

Demodulate using DBPSK method

Examples Modulate and demodulate a signal using DBPSK modulation.
```

    hMod = comm.DBPSKModulator(pi/4);
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
                            'Signal to noise ratio (SNR)','SNR',15);
    hDemod = comm.DBPSKDemodulator(pi/4);
    % Create an error rate calculator, account for the one bit transient caus
hError = comm.ErrorRate('ComputationDelay',1);
for counter = 1:100
% Transmit a 50-symbol frame
data = randi([0 1],50,1);
modSignal = step(hMod, data);
noisySignal = step(hAWGN, modSignal);
receivedData = step(hDemod, noisySignal);
errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
errorStats(1), errorStats(2))

```

\title{
Algorithms This object implements the algorithm, inputs, and outputs described on the DBPSK Demodulator Baseband block reference page. The object properties correspond to the block parameters. \\ See Also comm.DBPSKModulator | comm.DQPSKModulator
}

Purpose Create DBPSK demodulator object with same property values

\section*{Syntax}

Description
C = clone (H) creates a DBPSKDemodulator object \(C\), with the same property values as \(H\). The clone method creates a new unlocked object with uninitialized states.

\section*{comm.DBPSKDemodulator.getNuminputs}

\section*{Purpose Number of expected inputs to step method}

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs(H)

\section*{comm.DBPSKDemodulator.getNumOutputs}

Purpose \(\quad\) Number of outputs from step method

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.DBPSKDemodulator.isLocked}

\section*{Purpose Locked status for input attributes and nontunable properties}
Syntax
TF = isLocked(H)

Description dTF = isLocked \((H)\) returns the locked status, TF of the DBPSKDemodulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

\section*{Syntax release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.DBPSKDemodulator.reset}
\begin{tabular}{ll} 
Purpose & Reset states of DBPSK demodulator object \\
Syntax & \(\operatorname{reset}(H)\) \\
Description & \(\operatorname{reset}(H)\) resets the states of the DBPSKDemodulator object, H.
\end{tabular}

Purpose Demodulate using DBPSK method

\section*{Syntax \(\quad Y=\operatorname{step}(H, X)\)}

Description \(\quad Y=\operatorname{step}(H, X)\) demodulates input data, \(X\), with the DBPSK demodulator System object, H, and returns Y. Input X must be a double or single precision data type scalar or column vector.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Modulate using DBPSK method}

Description
The DBPSKModulator object modulates using the differential binary phase shift keying method. The output is a baseband representation of the modulated signal.

\section*{Construction \(H=\) comm.DBPSKModulator creates a modulator System object, H. This} object modulates the input signal using the differential binary phase shift keying (DBPSK) method

H = comm.DBPSKModulator(Name, Value) creates a DBPSK modulator object, \(H\), with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm. DBPSKModulator(PHASE, Name, Value) creates a DBPSK modulator object, H. This object has the PhaseRotation property set to PHASE, and the other specified properties set to the specified values.

\section*{Properties}

PhaseRotation
Additional phase shift
Specify the additional phase difference between previous and current modulated bits in radians as a real scalar value. The default is 0 . This value corresponds to the phase difference between previous and current modulated bits when the input is zero.

\section*{OutputDataType}

Data type of output
Specify output data type as one of double | single. The default is double.

\section*{Algorithms \\ This object implements the algorithm, inputs, and outputs described on the DBPSK Modulator Baseband block reference page. The object properties correspond to the block parameters.}

\section*{Methods}

See Also
clone
getNumInputs
getNumOutputs
isLocked
release
reset
step

Create DBPSK modulator object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of DBPSK modulator object

Modulate using DBPSK method

\section*{Examples Modulate data using DBPSK modulation, and visualize the data in a scatter plot.}
```

% Create binary data symbols
data = randi([0 1], 96, 1);

```
\% Create a DBPSK modulator System object and set the phase rotation to pi
    hModulator = comm.DBPSKModulator(pi/4);
\% Modulate and plot the data
    modData \(=\) step(hModulator, data);
    scatterplot(modData)
comm.DBPSKDemodulator | comm. DQPSKModulator

\section*{comm.DBPSKModulator.clone}

Purpose
Syntax \(\quad C=\) clone \((H)\)
Description property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.DBPSKModulator.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNuminputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.DBPSKModulator.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNum0utputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description TF = isLocked (H) returns the locked status, TF of the DBPSKModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\title{
Purpose Reset states of DBPSK modulator object
}

\section*{Syntax reset (H)}

Description reset (H) resets the states of the DBPSKModulator object, H.
\begin{tabular}{|c|c|}
\hline Purpose & Modulate using DBPSK method \\
\hline Syntax & \(Y=\operatorname{step}(H, X)\) \\
\hline Description & \(\mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X})\) modulates input data, X , with the DBPSK modulator System object, H. It returns the baseband modulated output, Y. The input must be a numeric or logical data type column vector of bits. \\
\hline & Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object. \\
\hline
\end{tabular}

\section*{Purpose Descramble input signal}

Description

\section*{Construction}

\section*{Properties}

The Descrambler object descrambles a scalar or column vector input signal. The Descrambler object is the inverse of the Scrambler object. If you use the Scrambler object in a transmitter, then you use the Descrambler object in the related receiver.

H = comm. Descrambler creates a descrambler System object, H. This object descrambles the input data using a linear feedback shift register that you specify with the Polynomial property.

H = comm.Descrambler(Name, Value) creates a descrambler object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.Descrambler(N,POLY,COND,Name,Value) creates a descrambler object, \(H\). This object has the CalculationBase property set to N, the Polynomial property set to POLY, the InitialConditions property set to COND, and the other specified properties set to the specified values.

\section*{CalculationBase}

Range of input data
Specify calculation base as a positive, integer, scalar value. The step method input and output integers are in the range [ 0 , CalculationBase-1]. The default is 4 .

\section*{Polynomial}

Linear feedback shift register connections
Specify the polynomial that determines the shift register feedback connections. The default is [ \(\left.\begin{array}{lllll}1 & 1 & 1 & 0 & 1\end{array}\right]\). You can the generator polynomial as a numeric, binary vector that lists the coefficients of the polynomial in order of ascending powers of \(z^{-1}\), where \(p\left(z^{-1}\right)\) \(=1+p 1 z^{-1}+p 2 z^{-2}+\ldots\) is the generator polynomial. The first and last elements must be 1 . Alternatively, you can specify the
generator polynomial as a numeric vector. This vector contains the exponents of \(z^{-1}\) for the nonzero terms of the polynomial, in order of ascending powers of \(z^{-1}\). In this case, the first vector element must be 0. For example, both [ \(\begin{array}{llllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0\end{array} 1\) ] and
[ \(0-6-8\) - \(]\) ] specify the same polynomial \(p\left(z^{-1}\right)=1+z^{-6}+z^{-8}\).

\section*{InitialConditions}

Initial values of linear feedback shift register
Specify the initial values of the linear feedback shift register as an integer row vector with values in [0 CalculationBase-1]. The default is \(\left[\begin{array}{lll}0 & 1 & 2\end{array}\right]\). The length of this property vector must equal the order of the Polynomial property vector.
\begin{tabular}{ll} 
Methods & clone \\
& getNumInputs \\
& getNumOutputs \\
isLocked \\
& release \\
reset \\
step
\end{tabular}

Create descrambler object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of descrambler object
Descramble input signal

\section*{Examples}

Scramble and descramble random data with values in the range [0 7].
```

% Create scrambler and descrambler objects with calculation base
N = 8;
hSCR = comm.Scrambler(N, [1 0 1 1 0 1 0 1],...

```
```

    [0 3 2 2 5 1 7]);
    hDSCR = comm.Descrambler(N, [10 0 1 1 0 1 0 1],...
[0 3 2 2 5 1 7]);
for counter = 1:10
data = randi([0 N-1], 4, 1);
scrData = step(hSCR, data);
deScrData = step(hDSCR, scrData);
[data, scrData, deScrData]
end

```

Algorithms This object implements the algorithm, inputs, and outputs described on the Descrambler block reference page. The object properties correspond to the block parameters.

\author{
See Also comm.Scrambler | comm. PNSequence
}

Purpose Create descrambler object with same property values
Syntax \(\quad C=\) clone \((H)\)
Description
C = clone (H) creates a Descrambler object C, with the same property values as \(H\). The clone method creates a new unlocked object with uninitialized states.

\section*{comm.Descrambler.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.Descrambler.getNumOutputs}

\section*{Purpose Number of outputs from step method}

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description TF = isLocked (H) returns the locked status, TF of the Descrambler System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.Descrambler.reset}
Purpose Reset states of descrambler object

\section*{Syntax reset (H)}

Description reset \((H)\) resets the states of the Descrambler object, H.

\section*{Purpose Descramble input signal}

\section*{Syntax \\ Y = step (H,X)}

Description \(\quad Y=\operatorname{step}(H, X)\) descrambles input data, \(X\), and returns the result in \(Y\). \(X\) must be a double precision, logical, or integer column vector. The output \(Y\) is same data type and length as the input vector, \(X\).

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Decode binary signal using differential decoding}

Description

\section*{Construction}

The DifferentialDecoder object decodes the binary input signal. The output is the logical difference between the consecutive input element within a channel.

H = comm.DifferentialDecoder creates a differential decoder System object, H. This object decodes a binary input signal that was previously encoded using a differential encoder.

H = comm.DifferentialDecoder(Name, Value) creates object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

\section*{Properties}

\section*{InitialCondition}

Initial value used to generate initial output
Specify the initial condition as a real scalar. This property can have a logical, numeric, or fixed-point (embedded.fi object) data type. The default is 0 . The object treats nonbinary values as binary signals.

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release
}

Create differential decoder object with same property values
Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
\begin{tabular}{ll} 
reset & \begin{tabular}{l} 
Reset states of differential \\
decoder object
\end{tabular} \\
step & \begin{tabular}{l} 
Decode binary signal using \\
differential decoding
\end{tabular}
\end{tabular}

Examples Decode a differentially encoded signal.
\% Create Differential Encoder System object hdiffenc = comm.DifferentialEncoder;
\% Create Differential Decoder System object hdiffdec = comm.DifferentialDecoder;
\% Generate random binary data data \(=\) randi([0 1], 100, 1);
\% Encode data encdata = step(hdiffenc,data);
\% Decode data
recdata \(=\) step(hdiffdec, encdata); errors = biterr(data, recdata); fprintf(1, ['\nThere were \%d errors in the decoded signal '... 'out of \%d bits \(\left.\left.\backslash n^{\prime}\right], e r r o r s, ~ l e n g t h(d a t a)\right) ;\)

\section*{Algorithms}

See Also

This object implements the algorithm, inputs, and outputs described on the Differential Decoder block reference page. The object properties correspond to the block parameters, except:

The object only supports single channel, column vector inputs.

\author{
comm.DifferentialEncoder
}

\section*{comm.DifferentialDecoder.clone}

Purpose Create differential decoder object with same property values

\section*{Syntax \\ C = clone(H)}

Description \(\quad C=\) clone \((H)\) creates a DifferentialDecoder object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.DifferentialDecoder.getNumInputs}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.DifferentialDecoder.getNumOutputs}

Purpose Number of outputs from step method

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.DifferentialDecoder.isLocked}
\begin{tabular}{ll} 
Purpose & Locked status for input attributes and nontunable properties \\
Syntax & TF \(=\) isLocked (H) \\
Description & TF \(=\) isLocked \((H)\) returns the locked status, TF of the \\
& DifferentialDecoder System object.
\end{tabular}

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

\section*{comm.DifferentialDecoder.release}

Purpose Allow property value and input characteristics changes

\section*{Syntax release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of differential decoder object
Syntax reset (H)
Description reset \((H)\) resets the states of the DifferentialDecoder object, \(H\).

Purpose Decode binary signal using differential decoding

\section*{Syntax \(\quad Y=\operatorname{step}(H, X)\)}

Description \(\quad Y=\operatorname{step}(H, X)\) decodes the differentially encoded input data, \(X\), and returns the decoded data, \(Y\). The input \(X\) must be a column vector of data type logical, numeric, or fixed-point (embedded.fi objects). \(Y\) has the same data type as \(X\). The object treats non-binary inputs as binary signals. The object computes the initial output value by performing an Xor operation of the value in the InitialCondition property and the first element of the vector you input the first time you call the step method.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Encode binary signal using differential coding}

Description The DifferentialEncoder object encodes the binary input signal within a channel. The output is the logical difference between the current input element and the previous output element.

\section*{Construction \(H=\) comm.DifferentialEncoder creates a differential encoder System} object, H. This object encodes a binary input signal by calculating its logical difference with the previously encoded data.

H = comm.DifferentialEncoder(Name, Value) creates object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

\section*{Properties}

\section*{InitialCondition}

Initial value used to generate initial output
Specify the initial condition as a real scalar. This property can have a logical, numeric, or fixed-point (embedded.fi object) data type. The default is 0 . The object treats nonbinary values as binary signals.

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release
}

Create differential encoder object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
\begin{tabular}{ll} 
reset & \begin{tabular}{l} 
Reset states of differential \\
encoder object
\end{tabular} \\
step & \begin{tabular}{l} 
Encode binary signal using \\
differential coding
\end{tabular}
\end{tabular}

Examples Encode binary signal using differential coding.
```

% Create Differential Encoder System object
hdiffenc = comm.DifferentialEncoder;
% Generate random binary data
data = randi([0 1], 11, 1);
% Encode data
encdata = step(hdiffenc,data);

```

Algorithms This object implements the algorithm, inputs, and outputs described on the Differential Encoder block reference page. The object properties correspond to the block parameters, except:

The object only supports single channel, column vector inputs.
See Also
comm.DifferentialDecoder

Purpose Create differential encoder object with same property values
Syntax \(\quad C=\) clone \((H)\)
Description \(\quad C=\) clone \((H)\) creates a DifferentialEncoder object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.DifferentialEncoder.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.DifferentialEncoder.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNum0utputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description TF = isLocked \((H)\) returns the locked status, TF of the DifferentialEncoder System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.DifferentialEncoder.reset}

Purpose Reset states of differential encoder object

\section*{Syntax reset (H)}

Description reset \((H)\) resets the states of the DifferentialEncoder object, H.

\section*{Purpose Encode binary signal using differential coding}
\[
\text { Syntax } \quad Y=\operatorname{step}(H, X)
\]

Description \(\quad \mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X})\) encodes the binary input data, X , and returns the differentially encoded data, Y . The input X must be a column vector of data type logical, numeric, or fixed-point (embedded.fi objects). \(Y\) has the same data type as X . The object treats non-binary inputs as binary signals. The object computes the initial output value by performing an Xor operation of the value in the InitialCondition property and the first element of the vector you input the first time you call the step method.

> Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Generate variable frequency sinusoid
Description The DiscreteTimeVCO (voltage-controlled oscillator) object generates asignal whose frequency shift from the quiescent frequency property isproportional to the input signal. The input signal is interpreted as avoltage.
ConstructionH = comm.DiscreteTimeVCO creates a discrete-time voltage-controlledoscillator (VCO) System object, H. This object generates a sinusoidalsignal with the frequency shifted from the specified quiescent frequencyto a value proportional to the input signal.
H = comm.DiscreteTimeVCO(Name, Value) creates a discrete-timeVCO object, H , with each specified property set to the specified value.You can specify additional name-value pair arguments in any order as(Name1,Value1,...,NameN,ValueN).

\section*{Properties \\ OutputAmplitude}
Amplitude of output signal
Specify the amplitude of the output signal as a double- or single-precision, scalar value. The default is 1 . This property is tunable.

\section*{QuiescentFrequency}

Frequency of output signal when input is zero
Specify the quiescent frequency of the output signal in Hertz, as a double- or single-precision, real, scalar value. The default is 10. This property is tunable.

\section*{Sensitivity}

Sensitivity of frequency shift of output signal
Specify the sensitivity of the output signal frequency shift to the input as a double- or single-precision, real, scalar value. The default is 1 . This value scales the input voltage and, consequently,
the shift from the quiescent frequency value. The property measures Sensitivity in Hertz per volt. This property is tunable.

\section*{InitialPhase}

Initial phase of output signal
Specify the initial phase of the output signal, in radians, as a double or single precision, real, scalar value. The default is 0 .

\section*{SampleRate}

Sample rate of input
Specify the sample rate of the input, in Hertz, as a double- or single-precision, positive, scalar value. The default is 100.

\section*{Methods}
\begin{tabular}{ll} 
clone & \begin{tabular}{l} 
Create discrete-time VCO object \\
with same property values
\end{tabular} \\
getNumInputs & \begin{tabular}{l} 
Number of expected inputs to \\
step method
\end{tabular} \\
getNumOutputs & \begin{tabular}{l} 
Number of outputs from step \\
method
\end{tabular} \\
isLocked & \begin{tabular}{l} 
Locked status for input attributes \\
and nontunable properties
\end{tabular} \\
release & \begin{tabular}{l} 
Allow property value and input \\
characteristics changes
\end{tabular} \\
reset & \begin{tabular}{l} 
Reset states of discrete-time VCO \\
object
\end{tabular} \\
step & \begin{tabular}{l} 
Generate variable frequency \\
sinusoid
\end{tabular}
\end{tabular}

Examples Generate an FSK signal using a discrete time VCO.
```

% Create a SignalSource System object and generate random data
hreader = dsp.SignalSource;

```
```

hreader.Signal = randi([0 7],10,1);
% Rectangular pulse shaping
hreader.Signal = rectpulse(hreader.Signal, 100);
% Create a signal logger System object
hlogger = dsp.SignalSink;
% Create a discrete time VCO object and generate an FSK signal
hdvco = comm.DiscreteTimeVCO('OutputAmplitude',8, ...
'QuiescentFrequency',1);
while(~isDone(hreader))
sig = step(hreader);
y = step(hdvco,sig);
step(hlogger,y);
end
oscsig = hlogger.Buffer;
% Plot FSK signal
t = [0:length(oscsig)-1]'/hdvco.SampleRate;
plot(t,hreader.Signal,'--r', 'LineWidth',3); hold on;
plot(t,oscsig,'-b'); hold off;
xlabel('time (s)');
legend('Input Signal', 'FSK Signal');

```

\section*{Algorithms}

This object implements the algorithm, inputs, and outputs as described on the Discrete-Time VCO block reference page. However, this object and the corresponding block may not generate the exact same outputs for single-precision inputs or property values due to the following differences in casting strategies and arithmetic precision issues:
- The block always casts the result of intermediate mathematical operations to the input data type. The object does not cast intermediate results and MATLAB decides the data type. The object casts the final output to the input data type.
- You can specify the SampleRate object property in single-precision or double-precision. The block does not allow this.
- In arithmetic operations with more than two operands with mixed data types, the result may differ depending on the order of operation.

Thus, the following calculation may also contribute to the difference in the output of the block and the object:
input* sensitivity * sampleTime
- The block performs this calculation from left to right. However, since sensitivity * sampleTime is a one-time calculation, the object calculates this in the following manner:
input * (sensitivity * sampleTime)

\author{
See Also \\ comm.CPMCarrierPhaseSynchronizer \\ comm.PSKCarrierPhaseSynchronizer
}

Purpose Create discrete-time VCO object with same property values

\section*{Syntax \\ C = clone(H)}

Description \(\quad C=\) clone \((H)\) creates a DiscreteTimeVCO object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.DiscreteTimeVCO.getNumInputs}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( \(H\) )

\section*{comm.DiscreteTimeVCO.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.DiscreteTimeVCO.isLocked}
\begin{tabular}{ll} 
Purpose & Locked status for input attributes and nontunable properties \\
Syntax & TF \(=\) isLocked (H) \\
Description & TF \(=\) isLocked \((H)\) returns the locked status, TF of the \\
& DiscreteTimeVCO System object.
\end{tabular}

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

\section*{Syntax \\ release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{ll} 
Purpose & Reset states of discrete-time VCO object \\
Syntax & \(\operatorname{reset}(\mathrm{H})\) \\
Description & reset (H) resets the states of the DiscreteTimeVCO object, H.
\end{tabular}

Purpose Generate variable frequency sinusoid

\section*{Syntax \(\quad Y=\operatorname{step}(H, X)\)}

Description \(\quad Y=\operatorname{step}(H, X)\) generates a sinusoidal signal, \(Y\), with frequency shifted, from the value you specify in the QuiescentFrequency property, to a value proportional to the input signal, \(X\). The input, \(X\), must be a double or single precision, real, scalar value. The output, Y , has the same data type and size as the input, \(X\).

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Demodulate using M-ary DPSK method}

Description

\section*{Construction}

The DPSKDemodulator object demodulates a signal that was modulated using the M-ary differential phase shift keying method. The input is a baseband representation of the modulated signal. The input and output for this object are discrete-time signals. This object accepts a scalar-valued or column vector input signal.

H = comm. DPSKDemodulator creates a demodulator System object, H. This object demodulates the input signal using the \(M\)-ary differential phase shift keying (M-DPSK) method.

H = comm.DPSKDemodulator(Name, Value) creates an M-DPSK demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).
H = comm. DPSKDemodulator (M, PHASE, Name, Value) creates an M-DPSK demodulator object, H. This object has the ModulationOrder property set to M, the PhaseRotation property set to PHASE, and the other specified properties set to the specified values.

\section*{Properties}

\section*{ModulationOrder}

Number of points in signal constellation
Specify the number of points in the signal constellation as a positive, integer scalar value. The default is 8 .

\section*{PhaseRotation}

Additional phase shift
Specify the additional phase difference between previous and current modulated symbols in radians as a real scalar value. The default is \(\mathrm{pi} / 8\). This value corresponds to the phase difference between previous and current modulated symbols when the input is zero.

\section*{BitOutput}

Output data as bits
Specify whether the output consists of groups of bits or integer symbol values. The default is false. When you set this property to true the step method outputs a column vector of bit values. The length of this column vector is equal to \(\log 2\) (ModulationOrder) times the number of demodulated symbols.

When you set this property to false, the step method outputs a column vector. The length of this column vector is equal to that of the input data vector. The output contains integer symbol values between 0 and ModulationOrder-1.

\section*{SymbolMapping}

Constellation encoding
Specify how the object maps an integer or group of \(\log 2\) (ModulationOrder) bits to the corresponding symbol as one of Binary | Gray. The default is Gray. When you set this property to Gray, the object uses a Gray-encoded signal constellation. When you set this property to Binary, the input integer \(m\), between ( \(0 \leq m \leq\) ModulationOrder-1) maps to the current symbol. This mapping uses \(\exp (j \times\) PhaseRotation + \(j \times 2 \times \pi \times \mathrm{m} /\) ModulationOrder \() \times(\) previously modulated symbol \()\).

\section*{OutputDataType}

Data type of output
Specify the output data type as one of Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32. The default is Full precision. When you set this property to Full precision, the input data type is single or double precision, the output data is the same as that of the input. When you set the BitOutput property to true, logical data type becomes a valid option.
\begin{tabular}{ll} 
Methods & clone \\
& getNumInputs \\
& getNumOutputs \\
& isLocked \\
& release \\
reset \\
step
\end{tabular}

Create M-DPSK demodulator object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of M-DPSK demodulator object

Demodulate using M-ary DPSK method

Examples Modulate and demodulate a signal using 8-DPSK modulation.
```

    hMod = comm.DPSKModulator(8,pi/4);
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
                            'Signal to noise ratio (SNR)','SNR',20);
    hDemod = comm.DPSKDemodulator(8,pi/4);
    % Create an error rate calculator, account for the one symbol transier
hError = comm.ErrorRate('ComputationDelay',1);
for counter = 1:100
% Transmit a 50-symbol frame
data = randi([0 hMod.ModulationOrder-1],50,1);
modSignal = step(hMod, data);
noisySignal = step(hAWGN, modSignal);
receivedData = step(hDemod, noisySignal);
errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
errorStats(1), errorStats(2))

```

\section*{comm.DPSKDemodulator}
\begin{tabular}{ll} 
Algorithms & \begin{tabular}{l} 
This object implements the algorithm, inputs, and outputs described on \\
the M-DPSK Demodulator Baseband block reference page. The object \\
properties correspond to the block parameters.
\end{tabular} \\
See Also & \begin{tabular}{l} 
comm.DPSKModulator | comm. DBPSKDemodulator । \\
comm.DQPSKDemodulator
\end{tabular}
\end{tabular}

Purpose
Create M-DPSK demodulator object with same property values

\section*{Syntax \\ C = clone(H)}

Description

C = clone (H) creates a DPSKDemodulator object C, with the same property values as \(H\). The clone method creates a new unlocked object with uninitialized states.

\section*{comm.DPSKDemodulator.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNuminputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose Number of outputs from step method

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNum0utputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the DPSKDemodulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.DPSKDemodulator.reset}

Purpose Reset states of M-DPSK demodulator object

\section*{Syntax reset(H)}

Description reset \((H)\) resets the states of the DPSKDemodulator object, \(H\).

\section*{Purpose Demodulate using M-ary DPSK method}
\[
\text { Syntax } \quad Y=\operatorname{step}(H, x)
\]

Description \(\quad Y=\operatorname{step}(H, X)\) demodulates input data, \(X\), with the DPSK demodulator System object, H, and returns Y. Input X must be a double or single precision data type scalar or column vector. Depending on the BitOutput property value, output \(Y\) can be integer or bit valued.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Modulate using M-ary DPSK method}

Description

\section*{Construction}

\section*{Properties}

The DPSKModulator object modulates using the M-ary differential phase shift keying method. The output is a baseband representation of the modulated signal.

H = comm. DPSKModulator creates a modulator System object, H. This object modulates the input signal using the M-ary differential phase shift keying (M-DPSK) method.

H = comm. DPSKModulator(Name, Value) creates an M-DPSK modulator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.DPSKModulator (M, PHASE, Name, Value) creates an M-DPSK modulator object, H. This object has the ModulationOrder property set to M, the PhaseRotation property set to PHASE, and the other specified properties set to the specified values.

\section*{ModulationOrder}

Number of points in signal constellation
Specify the number of points in the signal constellation as a positive, integer scalar value. The default is 8 .

\section*{PhaseRotation}

Additional phase shift
Specify the additional phase difference between previous and current modulated symbols in radians as a real scalar value. The default is \(\mathrm{pi} / 8\). This value corresponds to the phase difference between previous and current modulated symbols when the input is zero.

\section*{BitInput}

Assume bit inputs

Specify whether the input is bits or integers. The default is false. When you set this property to true, the step method input must be a column vector of bit values whose length is an integer multiple of \(\log 2\) (ModulationOrder). This vector contains bit representations of integers between 0 and ModulationOrder-1. When you set this property to false, the step method input requires a column vector of integer symbol values between 0 and ModulationOrder-1.

\section*{SymbolMapping}

Constellation encoding
Specify how the object maps an integer or group of \(\log 2\) (ModulationOrder) input bits to the corresponding symbol as one of Binary | Gray. The default is Gray. When you set this property to Gray, the object uses a Gray-encoded signal constellation. When you set this property to Binary, the input integer \(m\), between ( \(0 \leq m \leq\) ModulationOrder-1) shifts the output phase. This shift is (PhaseRotation + \(2 \times \pi \times m /\) ModulationOrder) radians from the previous output phase. The output symbol uses \(\exp (j \times\) PhaseRotation + \(j \times 2 \times \pi \times \mathrm{m} /\) ModulationOrder \() \times(\) previously modulated symbol \()\).

\section*{OutputDataType}

Data type of output
Specify output data type as one of double | single. The default is double.

\section*{Methods}
clone
getNumInputs
getNumOutputs

Create M-DPSK modulator object with same property values
Number of expected inputs to step method
Number of outputs from step method

\section*{comm.DPSKModulator}
\begin{tabular}{ll} 
isLocked & \begin{tabular}{l} 
Locked status for input attributes \\
and nontunable properties
\end{tabular} \\
release & \begin{tabular}{l} 
Allow property value and input \\
characteristics changes
\end{tabular} \\
reset & \begin{tabular}{l} 
Reset states of M-DPSK \\
modulator object
\end{tabular} \\
step & \begin{tabular}{l} 
Modulate using M-ary DPSK \\
method
\end{tabular}
\end{tabular}
```

Examples Modulate data using 8-DPSK modulation and visualize the data in a scatter plot.

```
```

% Create binary data for 1000, 3 bit symbols

```
% Create binary data for 1000, 3 bit symbols
    data = randi([0 1],3000,1);
    data = randi([0 1],3000,1);
% Create an 8-DPSK modulator System object with bits as inputs,phase rota
    hModulator = comm.DPSKModulator(8,pi/4,'BitInput',true);
% Modulate and plot the data
    modData = step(hModulator, data);
    scatterplot(modData)
```

```
Algorithms
This object implements the algorithm, inputs, and outputs described on the M-DPSK Modulator Baseband block reference page. The object properties correspond to the block parameters.
```

See Also<br>comm.DPSKDemodulator | comm.DBPSKModulator |<br>comm.DQPSKModulator

## comm.DPSKModulator.clone

Purpose Create M-DPSK modulator object with same property values
Syntax $\quad C=$ clone $(H)$
Description $\quad C=$ clone $(H)$ creates a DPSKModulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.DPSKModulator.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNuminputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.DPSKModulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the DPSKModulator System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.DPSKModulator.reset

| Purpose | Reset states of M-DPSK modulator object |
| :--- | :--- |
| Syntax | $\operatorname{reset}(H)$ |
| Description | reset $(H)$ resets the states of the DPSKModulator object, H. |

[^5]
## Purpose Demodulate using DQPSK method

Description

## Construction

The DQPSKDemodulator object demodulates a signal that was modulated using the differential quaternary phase shift keying method. The input is a baseband representation of the modulated signal.

H = comm.DQPSKDemodulator creates a demodulator System object, H. This object demodulates the input signal using the differential quadrature phase shift keying (DQPSK) method.

H = comm.DQPSKDemodulator(Name, Value) creates a DQPSK demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm. DQPSKDemodulator(PHASE,Name, Value) creates a DQPSK demodulator object, H. This object has the PhaseRotation property set to PHASE and the other specified properties set to the specified values.

## Properties

## PhaseRotation

Additional phase shift
Specify the additional phase difference between previous and current modulated symbols in radians as a real scalar. The default is $\mathrm{pi} / 4$. This value corresponds to the phase difference between previous and current modulated symbols when the input is zero.

## BitOutput

Output data as bits
Specify whether the output consists of groups of bits or integer symbol values. The default is false. When you set this property to true the step method outputs a column vector of bit values with length equal to twice the number of demodulated symbols. When you set this property to false, the step method outputs a column vector, of length equal to the input data vector, that contains integer symbol values between 0 and 3 .

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of 2 bits to the corresponding symbol as one of Binary | Gray. The default is Gray. When you set this property to Gray, the object uses a Gray-encoded signal constellation. When you set this property to Binary, the integer $m$, between $0 \leq m \leq 3$ maps to the current
symbol as $\exp (j \times$ PhaseRotation $+j \times 2 \times \pi \times m / 4) \times($ previously modulated symbol).

## OutputDataType

Data type of output
Specify the output data type as one of Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16| int32|uint32. The default is Full precision. When you set this property to Full precision the output has the same data type as that of the input. In this case, the input data type is single- or double-precision value. When you set the BitOutput property to true, logical data type becomes a valid option.

## Methods

clone
getNumInputs
getNumOutputs
isLocked
release

Create DQPSK demodulator object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
reset
step

## Reset states of DQPSK demodulator object <br> Demodulate using DQPSK method

Examples Modulate and demodulate a signal using DQPSK modulation.

```
    hMod = comm.DQPSKModulator(pi/8);
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
        'Signal to noise ratio (SNR)','SNR',15);
    hDemod = comm.DQPSKDemodulator(pi/8);
% Create an error rate calculator, account for the one symbol transient
    hError = comm.ErrorRate('ComputationDelay',1);
    for counter = 1:100
% Transmit a 50-symbol frame
            data = randi([0 3],50,1);
            modSignal = step(hMod, data);
            noisySignal = step(hAWGN, modSignal);
            receivedData = step(hDemod, noisySignal);
            errorStats = step(hError, data, receivedData);
    end
    fprintf('Error rate = %f\nNumber of errors = %d\n', ...
        errorStats(1), errorStats(2))
```


# Algorithms <br> This object implements the algorithm, inputs, and outputs described on the DQPSK Demodulator Baseband block reference page. The object properties correspond to the block parameters. <br> See Also <br> comm.DQPSKModulator | comm.DPSKDemodulator | <br> comm. DBPSKDemodulator 

Purpose Create DQPSK demodulator object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a DQPSKDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.DQPSKDemodulator.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.DQPSKDemodulator.getNumOutputs

## Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.DQPSKDemodulator.isLocked

Purpose Locked status for input attributes and nontunable properties

## Syntax TF = isLocked (H)

Description TF = isLocked (H) returns the locked status, TF of the DQPSKDemodulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.DQPSKDemodulator.reset

Purpose Reset states of DQPSK demodulator object

## Syntax reset (H)

Description reset (H) resets the states of the DQPSKDemodulator object, H.

## Purpose Demodulate using DQPSK method

$$
\text { Syntax } \quad Y=\operatorname{step}(H, x)
$$

Description $\quad Y=\operatorname{step}(H, X)$ demodulates input data, $X$, with the DQPSK demodulator System object, H , and returns Y . Input X must be a single or double precision data type scalar or column vector. Depending on the BitOutput property value, output $Y$ can be integer or bit valued.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Modulate using DQPSK method

Description

## Properties

## Construction

The DQPSKModulator object modulates using the differential quaternary phase shift keying method. The output is a baseband representation of the modulated signal.

H = comm.DQPSKModulator creates a modulator System object, H. This object modulates the input signal using the differential quadrature phase shift keying (DQPSK) method.

H = comm.DQPSKModulator (Name, Value) creates a DQPSK modulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.DQPSKModulator(PHASE,Name, Value) creates a DQPSK modulator object, H. This object has the PhaseRotation property set to PHASE and the other specified properties set to the specified values.

## PhaseRotation

Additional phase shift
Specify the additional phase difference between previous and current modulated symbols in radians as a real scalar value. The default is pi/4. This value corresponds to the phase difference between previous and current modulated symbols when the input is zero.

## BitInput

Assume bit inputs
Specify whether the input is bits or integers. The default is false. When you set this property to true, the step method input must be a column vector of bit values. The length of this vector is an integer multiple of two. This vector contains bit representations of integers between 0 and 3 . When you set this property to false, the step method input must be a column vector of integer symbol values between 0 and 3 .

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of two input bits to the corresponding symbol as one of Binary \| Gray. The default is Gray. When you set this property to Gray, the object uses a Gray-encoded signal constellation. When you set this property to Binary, the input integer $m$, between $0 \leq m \leq 3$ shifts the output phase. This shift is (PhaseRotation $+2 \times \pi \times m / 4$ ) radians from the previous output phase. The output symbol is $\exp (j \times$ PhaseRotation $+j \times 2 \times \pi \times m / 4) \times($ previously modulated symbol).

## OutputDataType

Data type of output
Specify output data type as one of double | single. The default is double.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release

Create DQPSK modulator object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

| reset | Reset states of DQPSK modulator <br> object |
| :--- | :--- |
| step | Modulate using DQPSK method |

```
Examples Modulate data using DQPSK modulation and visualize the data in a scatter plot.
```

```
% Create binary data for 100, 4 bit symbols
```

% Create binary data for 100, 4 bit symbols
data = randi([0 1],400,1);
data = randi([0 1],400,1);
% Create a DQPSK modulator System object with bits as inputs,phase rotat
hModulator = comm.DQPSKModulator(pi/8,'BitInput',true);
% Modulate and plot the data
modData = step(hModulator, data);
scatterplot(modData)

```

Algorithms
This object implements the algorithm, inputs, and outputs described on the DQPSK Modulator Baseband block reference page. The object properties correspond to the block parameters.

See Also
comm.DQPSKDemodulator | comm.DPSKModulator |
comm.DBPSKModulator

Purpose Create DQPSK modulator object with same property values

\section*{Syntax \\ C = clone( H )}

Description \(C=\) clone \((H)\) creates a DQPSKModulator object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.DQPSKModulator.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.DQPSKModulator.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNum0utputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description TF = isLocked (H) returns the locked status, TF of the DQPSKModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

\section*{comm.DQPSKModulator.release}
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.DQPSKModulator.reset}
\begin{tabular}{ll} 
Purpose & Reset states of DQPSK modulator object \\
Syntax & \(\operatorname{reset}(H)\) \\
Description & \(\operatorname{reset}(H)\) resets the states of the DQPSKModulator object, H.
\end{tabular}
\begin{tabular}{ll} 
Purpose & \begin{tabular}{l} 
Modulate using DQPSK method \\
Syntax
\end{tabular} \\
Description & \begin{tabular}{l}
\(Y=\operatorname{step}(H, X)\) \\
\begin{tabular}{l} 
System object, \(H\). It returns the baseband modulated output, \(Y\). \\
Depending on the value of the Bit Input property, input \(X\) can be an \\
integer or bit valued column vector with numeric or logical data types.
\end{tabular} \\
\\
\\
\begin{tabular}{l} 
Note The object performs an initialization the first time the step \\
method is executed. This initialization locks nontunable properties and \\
input specifications, such as dimensions, complexity, and data type \\
of the input data. If you change a nontunable property or an input \\
specification, the System object issues an error. To change nontunable \\
properties or inputs, you must first call the release method to unlock \\
the object.
\end{tabular} \\
\hline
\end{tabular}
\end{tabular}

Purpose Recover symbol timing phase using early-late gate method

Description

\section*{Construction}

\section*{Properties}

H = comm.EarlyLateGateTimingSynchronizer creates a timing phase synchronizer System object, H. This object recovers the symbol timing phase of the input signal using the early-late gate method.

H = comm.EarlyLateGateTimingSynchronizer(Name, Value) creates an early-late gate timing synchronizer object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

\section*{SamplesPerSymbol}

Number of samples representing each symbol
Specify the number of samples that represent each symbol in the input signal as an integer-valued scalar greater than 1. The default is 4 .

\section*{ErrorUpdateGain}

Error update step size
Specify the step size for updating successive timing phase estimates as a positive real scalar value. Typically, this number is less than \(1 /\) SamplesPerSymbol, which corresponds to a slowly varying timing phase. The default is 0.05 . This property is tunable.

\section*{ResetInputPort}

Enable synchronization reset input
Set this property to true to enable resetting the timing phase recovery process based on an input argument value. When you set this property to true, you must specify a reset input value to
the step method. When the reset input is a nonzero value, the object restarts the timing phase recovery process. When you set this property to false, the object does not restart. The default is false.

\section*{ResetCondition}

Condition for timing phase recovery reset
Specify the conditions to reset the timing phase recovery process as one of Never | Every frame. The default is Never. When you set this property to Never, the phase recovery process never restarts. The object operates continuously, retaining information from one symbol to the next. When you set this property to Every frame, the timing phase recovery restarts at the start of each frame of data. In this case, each time the object calls the step method. This property applies when you set the ResetInputPort property to false.

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release \\ reset \\ step
}
\begin{tabular}{|c|c|}
\hline clone & Create early-late gate timing phase synchronizer object with same property values \\
\hline getNumInputs & Number of expected inputs to step method \\
\hline getNumOutputs & Number of outputs from step method \\
\hline isLocked & Locked status for input attributes and nontunable properties \\
\hline release & Allow property value and input characteristics changes \\
\hline reset & Reset states of early-late gate timing phase synchronizer \\
\hline step & Recover symbol timing phase using early-late gate method \\
\hline
\end{tabular}

Examples Recover timing phase using the early-late gate method.
```

% Initialize data
L = 16; M = 16; numSymb = 100; snrdB = 30;
R = 25; rollOff = 0.75; filtDelay = 3; g = 0.07; delay = 6.6498;
% Design raised cosine filters
txFiltSpec = fdesign.pulseshaping(L, 'Square root raised cosine', ..
'Nsym,Beta', 2*filtDelay, rollOff);
txFilterDesign = design(txFiltSpec);
txFilterDesign.Numerator = sqrt(L)*txFilterDesign.Numerator;
% Create System objects
hMod = comm.RectangularQAMModulator(M, ...
'NormalizationMethod', 'Average power');
hTxFilter = dsp.FIRInterpolator(L, txFilterDesign.Numerator);
hDelay = dsp.VariableFractionalDelay('MaximumDelay', L);
hChan = comm.AWGNChannel('NoiseMethod', ...
'Signal to noise ratio (SNR)', 'SNR', snrdB, ...
'SignalPower', 1/L);
hRxFilter = dsp.DigitalFilter(...
'TransferFunction', 'FIR (all zeros)', ...
'Numerator', txFilterDesign.Numerator);
hSync = comm.EarlyLateGateTimingSynchronizer(...
'SamplesPerSymbol', L, ...
'ErrorUpdateGain', g);
% Generate random data
data = randi([0 M-1], numSymb, 1);
% Modulate and filter transmitter data.
modData = step(hMod, data);
filterData = step(hTxFilter, modData);
% Introduce a random delay and add noise
delayedData = step(hDelay, filterData, delay);
chData = step(hChan, delayedData);
% Filter receiver data.

```

\section*{comm.EarlyLateGateTimingSynchronizer}
```

    rxData = step(hRxFilter, chData);
    % Estimate the delay from the received signal
[~, phase] = step(hSync, rxData);
fprintf(1, 'Actual Timing Delay: %f\n', delay);
fprintf(1, 'Estimated Timing Delay: %f\n', phase(end));

```

> Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the Early-Late Gate Timing Recovery block reference page. The object properties correspond to the block parameters, except:

The block Reset parameter corresponds to the ResetInputPort and ResetCondition properties.
comm.GardnerTimingSynchronizer | comm.MSKTimingSynchronizer

\section*{comm.EarlyLateGateTimingSynchronizer.clone}

Purpose \(\quad \begin{aligned} & \text { Create early-late gate timing phase synchronizer object with same } \\ & \text { property values }\end{aligned}\)

\section*{Syntax \\ C = clone(H)}

Description
C = clone(H) creates a EarlyLateGateTimingSynchronizer object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\title{
comm.EarlyLateGateTimingSynchronizer.getNumInputs
}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(N=\) getNumInputs \((H)\) method returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs \((H)\).

Purpose \(\quad\) Number of outputs from step method

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs (H) method returns a positive integer, \(N\), representing the number of outputs from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.EarlyLateGateTimingSynchronizer.isLocked}
\begin{tabular}{ll} 
Purpose & Locked status for input attributes and nontunable properties \\
Syntax & TF \(=\) isLocked \((H)\) \\
Description & TF \(=\) isLocked \((H)\) returns the locked status, TF of the \\
& EarlyLateGateTimingSynchronizer System object.
\end{tabular}

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

\section*{comm.EarlyLateGateTimingSynchronizer.release}

Purpose Allow property value and input characteristics changes

\section*{Syntax release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\author{
Purpose Reset states of early-late gate timing phase synchronizer
}

\section*{Syntax reset (H)}

Description reset (H) resets the states of early-late gate timing phase synchronizer for the EarlyLateGateTimingSynchronizer object H .

Purpose
Recover symbol timing phase using early-late gate method
Syntax
[Y,PHASE] \(=\operatorname{step}(H, X)\)
\([\mathrm{Y}, \mathrm{PHASE}]=\operatorname{step}(\mathrm{H}, \mathrm{X}, \mathrm{R})\)
[ Y, PHASE] \(=\operatorname{step}(\mathrm{H}, \mathrm{X})\) performs timing phase recovery and returns the time-synchronized signal, Y , and the estimated timing phase, PHASE, for input signal \(X\). The input \(X\) must be a double or single precision complex column vector. Ideally, it is when the timing phase estimate is zero and the input signal has symmetric Nyquist pulses. In this case, the timing error detector for the early-late gate method requires samples that span one symbol interval.
[ \(\mathrm{Y}, \mathrm{PHASE}]=\operatorname{step}(\mathrm{H}, \mathrm{X}, \mathrm{R})\) restarts the timing phase recovery process when you input a reset signal, \(R\), that is non-zero. \(R\) must be a double precision or logical scalar. This syntax applies when you set the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Compute bit or symbol error rate of input data
Description
Construction
The ErrorRate object compares input data from a transmitter with input data from a receiver and calculates the error rate as a running statistic. To obtain the error rate, the object divides the total number of unequal pairs of data elements by the total number of input data elements from one source.
H = comm.ErrorRate creates an error rate calculator System object, H. This object computes the error rate of the received data by comparing it to the transmitted data.
H = comm.ErrorRate(Name, Value) creates an error rate calculator object, \(H\), with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

\section*{ReceiveDelay}
Number of samples to delay transmitted signal
Specify the number of samples by which the received data lags behind the transmitted data. This value must be a real, nonnegative, double-precision, integer scalar. Use this property to align the samples for comparison in the transmitted and received input data vectors. Specify the delay in number of samples, regardless of whether the input is a scalar or a vector. The default is 0 .

\section*{ComputationDelay}
Computation delay
Specify the number of data samples that the object should ignore at the beginning of the comparison. This value must be a real, nonnegative, double-precision, integer scalar. Use this property to ignore the transient behavior of both input signals. The default is 0 .

\section*{Samples}

Samples to consider
Specify samples to consider as one of Entire frame | Custom | Input port. The property defines whether the object should consider all or only part of the input frames when computing error statistics. The default is Entire frame. Select Entire frame to compare all the samples of the RX frame to those of the TX frame. Select Custom or Input port to list the indices of the RX frame elements that the object should consider when making comparisons. When you set this property to Custom, you can list the indices as a scalar or a column vector of double-precision integers through the CustomSamples property. When you set this property to Input port, you can list the indices as an input to the step method.

\section*{CustomSamples}

Selected samples from frame
Specify a scalar or a column vector of double-precision, real, positive integers. This value lists the indices of the elements of the RX frame vector that the object uses when making comparisons. This property applies when you set the Samples property to Custom. The default is an empty vector, which specifies that all samples are used.

\section*{ResetInputPort}

Enable error rate reset input
Set this property to true to reset the error statistics via an input to the step method. The default is false.

\author{
Methods clone \\ getNumInputs \\ Create error rate calculator object with same property values \\ Number of expected inputs to step method
}
getNumOutputs
isLocked
release
reset
step

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of error rate calculator object

Compute bit or symbol error rate of input data

\section*{Examples Calculate BER between transmitted and received signal.}
```

% Use 8-DPSK modulation in an AWGN channel
hMod = comm.DPSKModulator('ModulationOrder',8,'BitInput',true);
hDemod = comm.DPSKDemodulator('ModulationOrder',8,'BitOutput',true
hAWGN = comm.AWGNChannel('NoiseMethod',...
'Signal to noise ratio (SNR)','SNR', 7);
% Create an error rate calculator, accounting for the three bit
% (i.e., one symbol) transient caused by the differential modulation
hError = comm.ErrorRate('ComputationDelay',3);
BER = zeros(10,1);
% Calculate BER for 10 frames
for i= 1:10
data = randi([0 1], 96, 1);
modData = step(hMod, data);
receivedSignal = step(hAWGN, modData);
receivedData = step(hDemod, receivedSignal);
errors = step(hError, data, receivedData);
BER(i) = errors(1);
end
disp(BER) % display BER for 10 frames

```

Algorithms

See Also alignsignals | finddelay

Purpose
Create error rate calculator object with same property values

\section*{Syntax \\ C = clone(H)}

Description
C = clone(H) creates a ErrorRate object C, with the same property values as \(H\). The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.ErrorRate.getNumOutputs}

\section*{Purpose Number of outputs from step method}

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description TF = isLocked (H) returns the locked status, TF of the ErrorRate System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\title{
Purpose Reset states of error rate calculator object
}

\section*{Syntax reset(H)}

Description reset \((H)\) resets the states of the ErrorRate object, H.

\section*{Purpose}

Compute bit or symbol error rate of input data
Syntax
\(Y=\operatorname{step}(H, T X, R X)\)
\(Y=\operatorname{step}(H, T X, R X, S E L)\)
\(Y=\operatorname{step}(H, T X, R X, R S T)\)
\(Y=\operatorname{step}(H, T X, R X)\) counts the number of differences between the transmitted data vector, TX, and received data vector, RX. The step method outputs a three-element vector consisting of the error rate, followed by the number of errors detected and the total number of samples compared. TX and RX inputs can be either scalars or column vectors of the same data type. Valid data types are single, double, integer or logical. If TX is a scalar and RX is a vector, or vice-versa, then the block compares the scalar with each element of the vector.
\(Y=\) step (H,TX, RX, SEL) calculates the errors based on selected samples from the input frame specified by the SEL input. SEL must be a real, double-precision integer-valued scalar or a column vector. The vector lists the indices of the elements of the RX input vector that the object should consider when making comparisons. This syntax applies when you set the Samples property to 'Input Port'.

Y = step( \(H, T X, R X, R S T)\) resets the error count whenever the input RST is non-zero. RST must be a real, double, or logical scalar. When you set the RST input to a nonzero value, the object clears its error statistics and then recomputes them based on the current TX and RX inputs. This syntax applies when you set the ResetInputPort property to true. You can combine optional input arguments when their enabling properties are set. Optional inputs must be listed in the same order as the order of the enabling properties. For example,

> Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Measure error vector magnitude}

Description

Construction

The Error Vector Magnitude EVM object is a measurement of modulator or demodulator performance in an impaired signal.

\section*{Properties}

\section*{Normalization}

EVM normalization method
Specify the normalization method that the object uses in the EVM calculation. Choose from Average reference signal power | Average constellation power | Peak constellation power. The default is Average reference signal power.

\section*{AverageConstellationPower}

Average constellation power
Specify the average constellation power (in watts) that the object uses to normalize the EVM measurements. Set this property to a positive, real scalar value with a data type of double, single, or integer. This property applies when you set the Normalization property to Average constellation power. The default is 1 .

\section*{PeakConstellationPower}

Peak constellation power
Specify the peak constellation power (in watts) that the object uses to normalize the EVM measurements. Set this property to a positive, real scalar value with a data type of double, single, or
integer. This property applies when you set the Normalization property to Peak constellation power. The default is 1 .

\section*{MaximumEVMOutputPort}

\section*{Enable maximum EVM measurement output}

When you set this property to true, the step method outputs maximum EVM measurements. The default is false. The maximum EVM output is the maximum EVM value measured in the current input frame.

\section*{XPercentileEVMOutputPort}

Enable \(X\)-percentile EVM measurement output
When you set this property to true, the step method outputs \(X\)-percentile EVM measurements. The default is false. After you set this property the \(X\)-percentile EVM measurements persist. These measurements are obtained based on all the input frames since the last reset.

\section*{XPercentileValue}
\(X\)-percentile value
Specify the \(X\)-percentile value (in percent) that the object uses to calculate the \(X\)-th percentile of the EVM measurements. The default is 95 . The \(X\)-th percentile is the EVM value below which \(X \%\) of all the computed EVM values lie. Set this property to a real scalar between 0 and 100, inclusive. This property can have a data type of double, single, or integer, and applies when you set the XPercentileEVMOutputPort property to true.

\section*{SymbolCountOutputPort}

Enable symbol count output
When you set this property to true, the step method outputs the number of accumulated symbols that the object uses to calculate the X-Percentile EVM measurements since the last reset. The default setting for this property is false. This property applies when you set the XPercentileEVMOutputPort property to true.

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release \\ reset \\ step
}

Create EVM measurement object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of EVM measurement object
Measure error vector magnitude

Examples Measure the EVM of a noisy 16-QAM modulated signal.
```

hMod = comm.RectangularQAMModulator(16);
hAWGN = comm.AWGNChannel('NoiseMethod',...
'Signal to noise ratio (SNR)',...
'SNR', 20, 'SignalPower', 10);
% Create an EVM object, output maximum and 90-percentile EVM
% measurements, and symbol count
hEVM = comm.EVM('MaximumEVMOutputPort',true,...
'XPercentileEVMOutputPort', true, 'XPercentileValue', 90,
'SymbolCountOutputPort', true);
% Generate modulated symbols and add noise
refsym = step(hMod, randi([0 15], 1000, 1));
rxsym = step(hAWGN, refsym);
% Calculate measurements
[RMSEVM,MaxEVM,PercentileEVM,NumSym] = step(hEVM,refsym,rxsym)

```

For an additional example that uses this object, see the EVM Measurements for a 802.15.4 ZigBee System example. To open this exampleopen this example file, enter edit EVMZigBeeDemo at the MATLAB command line.

\title{
Algorithms This object implements the algorithm, inputs, and outputs described on the EVM Measurement block reference page. The object properties correspond to the block parameters.
}

\author{
See Also \\ comm.MER | comm.ACPR | comm.CCDF
}

\title{
Purpose \\ Create EVM measurement object with same property values
}

\section*{Syntax \\ C = clone(H)}

Description
C = clone (H) creates a EVM object C, with the same property values as \(H\). The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose Number of outputs from step method

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNum0utputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description TF = isLocked (H) returns the locked status, TF of the EVM System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.EVM.reset}

\section*{Purpose Reset states of EVM measurement object}

\section*{Syntax reset (H)}

Description reset (H) resets the states of the EVM object, H.

arguments when you set their enabling properties. Optional outputs must be listed in the same order as the order of the enabling properties. For example,
[RMSEVM,MAXEVM,PEVM,NUMSYM] = step(H,REFSYM,RXSYM)

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Demodulate using M-ary FSK method}

Description

\section*{Construction}

\section*{Properties}

The FSKDemodulator object demodulates a signal that was modulated using the M-ary frequency shift keying method. The input is a baseband representation of the modulated signal. The input and output for this object are discrete-time signals.

H = comm.FSKDemodulator creates a demodulator System object, H. This object demodulates an M-ary frequency shift keying (M-FSK) signal using a noncoherent energy detector.

H = comm.FSKDemodulator (Name, Value) creates an M-FSK
demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.FSKDemodulator(M,FREQSEP,RS,Name, Value) creates an M-FSK demodulator object, H. This object has the ModulationOrder property set to M, the FrequencySeparation property set to FREQSEP, the SymbolRate property set to RS, and the other specified properties set to the specified values.

\section*{ModulationOrder}

Number of frequencies in modulated signal
Specify the number of frequencies in the modulated signal as a numeric, positive, integer scalar value that is a power of two. The default is 8 .

\section*{BitOutput}

Output data as bits
Specify whether the output is groups of bits or integer values. The default is false.

When you set this property to false, the step method outputs a column vector of length equal to \(N /\) SamplesPerSymbol. \(N\) is the length of the input data vector to the step method. The
elements of the output vector are integers between 0 and ModulationOrder-1. When you set this property to true, the step method outputs a column vector of length equal to \(\log 2(\) ModulationOrder \() \times(N / S a m p l e s P e r S y m b o l)\). The property's elements are bit representations of integers between 0 and ModulationOrder-1.

\section*{SymbolMapping}

Symbol encoding
Specify how the object maps an integer or group of \(\log 2\) (ModulationOrder) bits to the corresponding symbol as one of Binary | Gray. The default is Gray.

When you set this property to Gray, the object uses Gray-coded ordering.

When you set this property to Binary, the object uses natural binary-coded ordering.

For either type of mapping, the object maps the highest frequency to the integer 0 and maps the lowest frequency to the integer \(M-1\). In baseband simulation, the lowest frequency is the negative frequency with the largest absolute value.

\section*{FrequencySeparation}

Frequency separation between successive tones
Specify the frequency separation between successive symbols in the modulated signal in Hertz as a positive, real scalar value. The default is 6 Hz .

\section*{SamplesPerSymbol}

Number of samples per input symbol
Specify the number of samples per input symbol as a positive, integer scalar value. The default is 17 .

\section*{SymbolRate}

Symbol duration

Specify the symbol rate in symbols per second as a positive, double-precision, real scalar value. The default is 100 . To avoid output signal aliasing, specify an output sampling rate, \(F s=\) SamplesPerSymbol \(\times\) SymbolRate, which is greater than ModulationOrder \(\times\) FrequencySeparation. The symbol duration remain the same, regardless of whether the input is bits or integers.

\section*{OutputDataType}

Data type of output
Specify the output data type as one of logical | int8 | uint8 | int16 | uint16 | int32 | uint32 | double. The default is double. The logical type is valid only when you set the BitOutput property to false and the ModulationOrder property to two. When you set the BitOutput property to true, the output data requires a type of logical | double.

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release \\ reset \\ step
}

Create M-FSK demodulator object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
Reset states of M-FSK demodulator object
Demodulate using M-ary FSK method
```

Examples Modulate and demodulate a signal using 8-FSK modulation with a frequency separation of 100 Hz .

```
```

    hMod = comm.FSKModulator(8, 100);
    ```
    hMod = comm.FSKModulator(8, 100);
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
                            'Signal to noise ratio (SNR)','SNR',0);
                            'Signal to noise ratio (SNR)','SNR',0);
hDemod = comm.FSKDemodulator(8, 100);
hDemod = comm.FSKDemodulator(8, 100);
%Create an error rate calculator
%Create an error rate calculator
hError = comm.ErrorRate;
hError = comm.ErrorRate;
for counter = 1:100
for counter = 1:100
    % Transmit a 50-symbol frame
    % Transmit a 50-symbol frame
    data = randi([0 hMod.ModulationOrder-1],50,1);
    data = randi([0 hMod.ModulationOrder-1],50,1);
    modSignal = step(hMod, data);
    modSignal = step(hMod, data);
    noisySignal = step(hAWGN, modSignal);
    noisySignal = step(hAWGN, modSignal);
    receivedData = step(hDemod, noisySignal);
    receivedData = step(hDemod, noisySignal);
    errorStats = step(hError, data, receivedData);
    errorStats = step(hError, data, receivedData);
    end
    end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```

    errorStats(1), errorStats(2))
    ```

Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the M-FSK Demodulator Baseband block reference page. The object properties correspond to the block parameters, except:
- The Symbol set ordering parameter corresponds to the SymbolMapping property.
- The SymbolRate property replaces the block sample rate capability.
comm.FSKModulator | comm.CPFSKModulator |
comm. CPFSKDemodulator

Purpose Create M-FSK demodulator object with same property values

\section*{Syntax \(\quad C=\) clone \((H)\)}

Description \(\quad C=\) clone \((H)\) creates a FSKDemodulator object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.FSKDemodulator.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNuminputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.FSKDemodulator.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.FSKDemodulator.isLocked}

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax TF = isLocked (H)}

Description TF = isLocked (H) returns the locked status, TF of the FSKDemodulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.FSKDemodulator.reset}

Purpose Reset states of M-FSK demodulator object

\section*{Syntax reset(H)}

Description reset (H) resets the states of the FSKDemodulator object, H.

\section*{Purpose \\ Demodulate using M-ary FSK method}

\section*{Syntax \\ Y = step (H,X)}

Description
\(Y=\operatorname{step}(H, X)\) demodulates input data, X, with the FSK demodulator System object, H, and returns Y. X must be a double or single precision data type column vector of length equal to an integer multiple of the number of samples per symbol that you specify in the SamplesPerSymbol property. Depending on the BitOutput property value, output \(Y\) can be integer or bit valued.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Modulate using M-ary FSK method}

Description

Construction

H = comm.FSKModulator creates a modulator System object, H. This object modulates the input signal using the M-ary frequency shift keying (M-FSK) method.

H = comm.FSKModulator(Name, Value) creates an M-FSK modulator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.FSKModulator (M,FREQSEP,RS, Name, Value) creates an M-FSK modulator object, H. This object has the ModulationOrder property set to M, the FrequencySeparation property set to FREQSEP, the SymbolRate property set to RS, and the other specified properties set to the specified values.

\section*{Properties \\ ModulationOrder}

Number of frequencies in modulated signal
Specify the number of frequencies in the modulated signal as a numeric positive integer scalar value that is a power of two. The default is 8 .

\section*{BitInput}

Assume bit inputs
Specify whether the input is bits or integers. The default is false.
When you set this property to false, the step method input requires a numeric (except single precision data type) column vector of integer values between 0 and ModulationOrder-1. In this case, the input vector can also be of data type logical if ModulationOrder equals 2.

When you set this property to true, the step method input requires a double-precision or logical data type column vector of bit values. The length of this vector is an integer multiple of \(\log 2\) (ModulationOrder). This vector contains bit representations of integers between 0 and ModulationOrder-1.

\section*{SymbolMapping}

\section*{Symbol encoding}

Specify how the object maps an integer or group of log2(ModulationOrder) bits to the corresponding symbol as one of Binary | Gray. The default is Gray.

When you set this property to Gray, the object uses Gray-coded ordering.

When you set this property to Binary, the object uses natural binary-coded ordering. For either type of mapping, the object maps the highest frequency to the integer 0 and maps the lowest frequency to the integer \(M-1\). In baseband simulation, the lowest frequency is the negative frequency with the largest absolute value.

\section*{FrequencySeparation}

Frequency separation between successive tones
Specify the frequency separation between successive tones in the modulated signal in Hertz as a positive, real scalar value. The default is 6 Hz . To avoid output signal aliasing, specify an output sampling rate, \(F s=\) SamplesPerSymbol/SymbolRate, which is greater than ModulationOrder multiplied by FrequencySeparation.

\section*{ContinuousPhase}

Phase continuity
Specify if the phase of the output modulated signal is continuous or discontinuous. The default is true.

When you set this property to true, the modulated signal maintains continuous phase even when its frequency changes.

When you set this property to false, the modulated signal comprises portions of ModulationOrder sinusoids of different frequencies. In this case, a change in the input value can cause a discontinuous change in the phase of the modulated signal.

\section*{SamplesPerSymbol}

Number of samples per output symbol
Specify the number of output samples that the object produces for each integer or binary word in the input as a positive, integer scalar value. The default is 17 .

\section*{SymbolRate}

Symbol duration
Specify the symbol rate in symbols per second as a positive, double-precision, real scalar. The default is 100 . To avoid output signal aliasing, specify an output sampling rate, Fs \(=\) SamplesPerSymbol \(\times\) SymbolRate, which is greater than ModulationOrder \(\times\) FrequencySeparation. The symbol duration remain the same, regardless of whether the input is bits or integers.

\section*{OutputDataType}

Data type of output
Specify the output data type as one of double | single. The default is double.

\author{
Methods clone \\ getNumInputs \\ Create M-FSK modulator object with same property values \\ Number of expected inputs to step method
}

\author{
getNumOutputs \\ isLocked \\ release \\ reset \\ step
}

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of M-FSK modulator object

Modulate using M-ary FSK method

\section*{Examples}

\section*{Algorithms}

Modulate and demodulate a signal using 8-FSK modulation with a frequency separation of 100 Hz
```

hMod = comm.FSKModulator(8, 100);
hAWGN = comm.AWGNChannel('NoiseMethod', ...
'Signal to noise ratio (SNR)','SNR',O);
hDemod = comm.FSKDemodulator(8, 100);
%Create an error rate calculator
hError = comm.ErrorRate;
for counter = 1:100
% Transmit a 50-symbol frame
data = randi([0 hMod.ModulationOrder-1],50,1);
modSignal = step(hMod, data);
noisySignal = step(hAWGN, modSignal);
receivedData = step(hDemod, noisySignal);
errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
errorStats(1), errorStats(2))

```

This object implements the algorithm, inputs, and outputs described on the M-FSK Modulator Baseband block reference page. The object properties correspond to the block parameters, except:

\section*{comm.FSKModulator}
- The Symbol set ordering parameter corresponds to the SymbolMapping property.
- The SymbolRate property takes the place of the block sample rate capability.

See Also
comm.FSKDemodulator | comm.CPFSKModulator

\section*{Purpose Create M-FSK modulator object with same property values}

\section*{Syntax \\ C = clone( H )}

Description
\(C=\) clone \((H)\) creates a FSKModulator object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.FSKModulator.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.FSKModulator.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.FSKModulator.isLocked}

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description TF = isLocked (H) returns the locked status, TF of the FSKModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.FSKModulator.reset}

Purpose Reset states of M-FSK modulator object

\section*{Syntax reset (H)}

Description reset (H) resets the states of the FSKModulator object, H.
\begin{tabular}{ll} 
Purpose & Modulate using M-ary FSK method \\
Syntax & \(Y=\operatorname{step}(H, X)\) \\
Description & \begin{tabular}{l}
\(Y=\) step \((H, X)\) modulates input data, \(X\), with the FSK modulator \\
System object, \(H\). It returns the baseband modulated output, \(Y\). \\
Depending on the value of the BitInput property, input \(X\) can be an \\
integer or bit- valued column vector with numeric or logical data \\
types. The length of output vector, \(Y\), is equal to the number of input \\
samples times the number of samples per symbol you specify in the \\
SamplesPerSymbol property.
\end{tabular}
\end{tabular}

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Recover symbol timing phase using Gardner's method
Description
ConstructionH = comm.GardnerTimingSynchronizer creates a timing phasesynchronizer System object, H. This object recovers the symbol timingphase of the input signal using the Gardner method.
H = comm.GardnerTimingSynchronizer(Name, Value) creates anGardner timing synchronizer object, H, with each specified propertyset to the specified value. You can specify additional name-value pairarguments in any order as (Name1,Value1,...,NameN,ValueN).
Properties SamplesPerSymbol
Number of samples representing each symbol
Specify the number of samples that represent each symbol inthe input signal as an integer-valued scalar value greater than1. The default is 4 .

\section*{ErrorUpdateGain}
Error update step size
Specify the step size for updating successive timing phase estimates as a positive real scalar value. The default is 0.05 . Typically, this number is less than \(1 /\) SamplesPerSymbol, which corresponds to a slowly varying timing phase. This property is tunable.

\section*{ResetInputPort}
Enable synchronization reset input

Set this property to true to enable resetting the timing phase recovery process based on an input argument value. The default is false. When you set this property to true, you must specify a reset input value to the step method. When you specify a nonzero value as the reset input, the object restarts the timing phase recovery process. When you set this property to false, the object does not restart.

\section*{ResetCondition}

Condition for timing phase recovery reset
Specify the conditions to reset the timing phase recovery process as one of Never | Every frame. The default is Never. When you set this property to Never, the phase recovery process never restarts. The object operates continuously, retaining information from one symbol to the next. When you set this property to Every frame, the timing phase recovery restarts at the start of each frame of data. In this case, the restart occurs each time the object calls the step method. This property applies when you set the ResetInputPort property to false.

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release
}

Create Gardner timing phase synchronizer object with same with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes
reset
step

\section*{Examples Recover timing phase using the Gardner method.}
```

% Initialize some data
L = 16; M = 8; numSymb = 100; snrdB = 30;
R = 25; rollOff = 0.75; filtDelay = 3; g = 0.07; delay = 6.6498;
% Design raised cosine filters
txFiltSpec = fdesign.pulseshaping(L, 'Square root raised cosine', ...
'Nsym,Beta', 2*filtDelay, rollOff);
txFilterDesign = design(txFiltSpec);
txFilterDesign.Numerator = sqrt(L)*txFilterDesign.Numerator;
% Create System objects
hMod = comm.PSKModulator(M);
hTxFilter = dsp.FIRInterpolator(L, txFilterDesign.Numerator);
hDelay = dsp.VariableFractionalDelay('MaximumDelay', L);
hChan = comm.AWGNChannel(...
'NoiseMethod', 'Signal to noise ratio (SNR)', ...
'SNR', snrdB, 'SignalPower', 1/L);
hRxFilter = dsp.DigitalFilter('TransferFunction', 'FIR (all zeros)',
'Numerator', txFilterDesign.Numerator);
hSync = comm.GardnerTimingSynchronizer('SamplesPerSymbol', L, ...
'ErrorUpdateGain', g);
% Generate random data
data = randi([0 M-1], numSymb, 1);
% Modulate and filter transmitter data.
modData = step(hMod, data);
filterData = step(hTxFilter, modData);
% Introduce a random delay.
delayedData = step(hDelay, filterData, delay);
% Add noise
chData = step(hChan, delayedData);
% Filter receiver data.

```
```

    rxData = step(hRxFilter, chData);
    % Estimate the delay from the received signal
[~, phase] = step(hSync, rxData);
fprintf(1, 'Actual Timing Delay: %f\n', delay);
fprintf(1, 'Estimated Timing Delay: %f\n', phase(end));

```

\section*{Algorithms}

This object implements the algorithm, inputs, and outputs described on the Gardner Timing Recovery block reference page. The object properties correspond to the block parameters, except:

The Reset parameter corresponds to the ResetInputPort and ResetCondition properties.

\author{
See Also \\ comm. EarlyLateGateTimingSynchronizer | comm. MuellerMullerTimingSynchronizer
}

Purpose Create Gardner timing phase synchronizer object with same with same property values

\section*{Syntax \\ C = clone( H )}

Description
\(C=\) clone (H) creates a GardnerTimingSynchronizer object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.GardnerTimingSynchronizer.getNumInputs}

\section*{Purpose Number of expected inputs to step method}

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(N=\) getNumInputs \((H)\) method returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( \(H\) ).
Purpose Number of outputs from step method

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.GardnerTimingSynchronizer.isLocked}
\begin{tabular}{ll} 
Purpose & Locked status for input attributes and nontunable properties \\
Syntax & TF \(=\) isLocked \((H)\) \\
Description & \begin{tabular}{l} 
TF \(=\) isLocked \((H)\) returns the locked status, TF of the \\
GardnerTimingSynchronizer System object.
\end{tabular}
\end{tabular}

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

\section*{Syntax release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.GardnerTimingSynchronizer.reset}
Purpose Reset states of Gardner timing phase synchronizer object
Syntax ..... reset (H)
Description reset (H) resets the states of the GardnerTimingSynchronizer object, H.

Purpose Recover symbol timing phase using Gardner's method
Syntax \(\quad \begin{aligned} {[\mathrm{Y}, \text { PHASE }] } & =\operatorname{step}(H, X) \\ {[Y, \text { PHASE }] } & =\operatorname{step}(H, X, R)\end{aligned}\)
Description
[ \(\mathrm{Y}, \mathrm{PHASE}]=\operatorname{step}(\mathrm{H}, \mathrm{X})\) recovers the timing phase and returns the time-synchronized signal, \(Y\), and the estimated timing phase, PHASE, for input signal \(X\). The input \(X\) must be a double or single precision complex column vector. The length of \(X\) is \(N * K\), where \(N\) is an integer greater than or equal to two and K is the number of symbols. The output, Y , is the signal value for each symbol, which you use to make symbol decisions. Y is a column vector of length K with the same data type as X .
[ \(\mathrm{Y}, \mathrm{PHASE}\) ] = step \((\mathrm{H}, \mathrm{X}, \mathrm{R})\) restarts the timing phase recovery process when you input a reset signal, \(R\), that is non-zero. \(R\) must be a logical or double scalar. This syntax applies when you set the Reset InputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Demodulate using arbitrary QAM constellation}

Construction H = comm.GeneralQAMDemodulator creates a demodulator System

Description

\section*{Properties}

The GeneralQAMDemodulator object demodulates a signal that was modulated using quadrature amplitude modulation. The input is a baseband representation of the modulated signal. object, H. This object demodulates the input signal using a general quadrature amplitude modulation (QAM) method.

H = comm.GeneralQAMDemodulator (Name, Value) creates a general QAM demodulator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.GeneralQAMDemodulator(CONST,Name, Value) creates a general QAM demodulator object, H. This object has the Constellation property set to CONST, and the other specified properties set to the specified values.

\section*{Constellation}

Signal constellation
Specify the constellation points as a real or complex, double-precision data type vector. The default is
\(\exp (2 \times \pi \times 1 i \times(0: 7) / 8)\). The length of the vector determines the modulation order.

When you set the BitOutput property to false, the step method outputs a vector with integer values. These integers are between 0 and \(M-1\), where \(M\) is the length of this property vector. The length of the output vector equals the length of the input signal.
When you set the BitOutput property to true, the output signal contains bits. For bit outputs, the size of the signal constellation requires an integer power of two and the output length is an integer multiple of the number of bits per symbol.

\section*{BitOutput}

Output data as bits
Specify whether the output consists of groups of bits or integer symbol values. The default is false.

When you set this property to true the step method outputs a column vector of bit values with length equal to \(\log 2(M)\) times the number of demodulated symbols, where \(M\) is the length of the signal constellation specified in the Constellation property. The length \(M\) determines the modulation order.

When you set this property to false, the step method outputs a column vector, of length equal to the input data vector. The vector contains integer symbol values between 0 and \(M-1\).

\section*{DecisionMethod}

Demodulation decision method
Specify the decision method the object uses as one of Hard decision | Log-likelihood ratio | Approximate log-likelihood ratio. The default is Hard decision. When you set the BitOutput property to false the object always performs hard decision demodulation. This property applies when you set the BitOutput property to true.

\section*{VarianceSource}

Source of noise variance
Specify the source of the noise variance as one of Property | Input port. The default is Property. This property applies when you set the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio.

\section*{Variance}

Noise variance
Specify the variance of the noise as a nonzero, real scalar value. The default is 1 . If this value is very small (i.e., SNR
is very high), log-likelihood ratio (LLR) computations may yield Inf or -Inf. This result occurs because the LLR algorithm would compute the exponential of very large or very small numbers using finite-precision arithmetic. In such cases, using approximate LLR is recommended because its algorithm does not compute exponentials. This property applies when you set the VarianceSource property to Property. This property is tunable.

\section*{OutputDataType}

Data type of output
Specify the output data type as one of Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32. The default is Full precision.

This property applies only when you set the BitOutput property to false or when you set the BitOutput property to true and the DecisionMethod property to Hard decision or Approximate log-likelihood ratio. In this case, when you set the OutputDataType property to Full precision, the output data type is the same as that of the input when the input data has a single or double-precision data type.

When the input data is of a fixed-point type, the output data type works as if you had set the OutputDataType property to Smallest unsigned integer.

When the input signal is an integer data type, you must have a Fixed-Point Designer user license to use this property in Smallest unsigned integer or Full precision mode.

When you set the BitOutput property to true, and the DecisionMethod property to Hard Decision the data type logical becomes a valid option.

When you set the BitOutput property to true and the DecisionMethod property to Approximate log-likelihood ratio you may only set this property to Full precision | Custom.

If you set the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio, the output data has the same type as that of the input. In this case, that value can be only single or double precision.

\section*{Fixed-Point Properties}

\section*{FullPrecisionOverride}

Full precision override for fixed-point arithmetic
Specify whether to use full precision rules. If you set FullPrecisionOverride to true, which is the default, the object computes all internal arithmetic and output data types using full precision rules. These rules provide the most accurate fixed-point numerics. It also turns off the display of other fixed-point properties because they do not apply individually. These rules guarantee that no quantization occurs within the object. Bits are added, as needed, to ensure that no roundoff or overflow occurs. If you set FullPrecisionOverride to false, fixed-point data types are controlled through individual fixed-point property settings. For more information, see "Full Precision for Fixed-Point System Objects".

\section*{RoundingMethod}

Rounding of fixed-point numeric values
Specify the rounding method as one of Ceiling | Convergent | Floor | Nearest | Round | Simplest | Zero. The default is Floor. This property applies when the object is not in a full precision configuration. This property does not apply when you set BitOutput to true and DecisionMethod to Log-likelihood ratio.

\section*{OverflowAction}

Action when fixed-point numeric values overflow
Specify the overflow action as one of Wrap | Saturate. The default is Wrap. This property applies when the object is not in a
full precision configuration. This property does not apply when you set the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio.

\section*{ConstellationDataType}

Data type of signal constellation
Specify the constellation fixed-point data type as one of Same word length as input | Custom. The default is Same word length as input. This property does not apply when you set the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio.

\section*{CustomConstellationDataType}

Fixed-point data type of signal constellation
Specify the constellation fixed-point type as an unscaled numerictype object with a Signedness of Auto. The default is numerictype([],16). This property applies when you set the ConstellationDataType property to Custom.

\section*{Accumulator 1 DataType}

\section*{Data type of accumulator 1}

Specify the accumulator 1 fixed-point data type as one of Full precision | Custom. The default is Full precision. This property applies when you set the FullPrecisionOverride property to false. This property does not apply when you set the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio.

\section*{CustomAccumulator 1 DataType}

Fixed-point data type of accumulator 1
Specify the accumulator 1 fixed-point type as a scaled numerictype object with a Signedness of Auto. The default is numerictype([],32,30). This property applies when you set the Accumulator1DataType property to Custom.

\section*{ProductInputDataType}

Data type of product
Specify the product input fixed-point data type as one of Same as accumulator 1 | Custom. The default is Same as accumulator 1. This property applies when you set the FullPrecisionOverride property to false, the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio.

\section*{CustomProductInputDataType}

Fixed-point data type of product
Specify the product input fixed-point type as a scaled numerictype object with a Signedness of Auto. The default is numerictype([],32,30). This property applies when you set the FullPrecisionOverride property to false and the ProductInputDataType property to Custom.

\section*{ProductOutputDataType}

Data type of product output
Specify the product output fixed-point data type as one of Full precision | Custom. The default is Full precision. This property applies when you set the FullPrecisionOverride property to false, the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio.

\section*{CustomProductOutputDataType}

Fixed-point data type of product output
Specify the product output fixed-point type as a scaled numerictype object with a Signedness of Auto. The default is numerictype([],32,30). This property applies when you set the FullprecisionOverride property to false and the ProductOutputDataType property to Custom.

\section*{Accumulator2DataType}

Data type of accumulator 2
Specify the accumulator 2 fixed-point data type as one of Full precision | Custom. The default is Full precision. This property applies when you set the FullPrecisionOverride property to false, the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio.

\section*{CustomAccumulator2DataType}

Fixed-point data type accumulator 2
Specify the accumulator 2 fixed-point data type as a scaled numerictype object with a Signedness of Auto. The default is numerictype([],32,30). This property applies when you set the FullPrecisionOverride property to false and the Accumulator2DataType property to Custom.

\section*{Accumulator3DataType}

Data type of accumulator 3
Specify the accumulator 3 fixed-point data type as one of Full precision | Custom. The default is Full precision. This property applies when you set the FullPrecisionOverride property to false, the BitOutput property to true and the DecisionMethod property to Approximate log-likelihood ratio.

\section*{CustomAccumulator3DataType}

Fixed-point data type of accumulator 3
Specify the accumulator 3 fixed-point type as a scaled numerictype object with a Signedness of Auto. The default is numerictype ([],32,30). This property applies when you set the FullprecisionOverride property to false and the Accumulator3DataType property to Custom.

\section*{NoiseScalingInputDataType}

Data type of noise-scaling input

Specify the noise-scaling input fixed-point data type as one of Same as accumulator 3 | Custom. The default is Same as accumulator 3. This property applies when you set the FullprecisionOverride property to false, the BitOutput property to true and the DecisionMethod property to Approximate log-likelihood ratio.

\section*{CustomNoiseScalingInputDataType}

Fixed-point data type of noise-scaling input
Specify the noise-scaling input fixed-point type as a scaled numerictype object with a Signedness of Auto. The default is numerictype ([],32,30). This property applies when you set the FullprecisionOverride property to false and the NoiseScalingInputDataType property to Custom.

\section*{InverseVarianceDataType}

Data type of inverse noise variance
Specify the inverse noise variance fixed-point data type as one of Same word length as input | Custom. The default is Same word length as input. This property applies when you set the BitOutput property to true, the DecisionMethod property to Approximate log-likelihood ratio, and the VarianceSource property to Property.

\section*{CustomInverseVarianceDataType}

Fixed-point data type of inverse noise variance
Specify the inverse noise variance fixed-point type as a numerictype object with a Signedness of Auto. The default is numerictype([],16,8). This property applies when you set the InverseVarianceDataType property to Custom.

\section*{CustomOutputDataType}

Data type of output
Specify the output fixed-point type as a scaled numerictype object with a Signedness of Auto. The default is
numerictype([],32,30). This property applies when you set the FullPrecisionOverride property to false and the OutputDataType property to Custom.

\section*{Methods}

\author{
clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release \\ step
}

Create general QAM demodulator object with same property values
Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Demodulate using arbitrary QAM constellation

\section*{Examples}

Modulate and demodulate data using an arbitrary three-point constellation.
```

% Setup a three point constellation
c = [1 1i -1];
hQAMMod = comm.GeneralQAMModulator(c);
hAWGN = comm.AWGNChannel('NoiseMethod', ...
'Signal to noise ratio (SNR)','SNR',15, 'SignalPower',
hQAMDemod = comm.GeneralQAMDemodulator(c);
%Create an error rate calculator
hError = comm.ErrorRate;
for counter = 1:100
% Transmit a 50-symbol frame
data = randi([0 2],50,1);
modSignal = step(hQAMMod, data);

```

\section*{comm.GeneralQAMDemodulator}
```

    noisySignal = step(hAWGN, modSignal);
    receivedData = step(hQAMDemod, noisySignal);
    errorStats = step(hError, data, receivedData);
    end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
errorStats(1), errorStats(2))

```

\author{
Algorithms \\ This object implements the algorithm, inputs, and outputs described on the General QAM Demodulator Baseband block reference page. The object properties correspond to the block parameters. \\ See Also \\ comm. GeneralQAMModulator | comm.RectangularQAMDemodulator
}

Purpose
Create general QAM demodulator object with same property values

\section*{Syntax \\ C = clone(H)}

Description
\(C=\) clone \((H)\) creates a GeneralQAMDemodulator object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.GeneralQAMDemodulator.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.GeneralQAMDemodulator.getNumOutputs}

\section*{Purpose Number of outputs from step method}

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax \(\quad\) TF \(=\) isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the GeneralQAMDemodulator System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release (H) \\
Description & \begin{tabular}{l} 
release (H) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{ll} 
Purpose & Demodulate using arbitrary QAM constellation \\
Syntax & \(\left.\begin{array}{l}Y=\operatorname{step}(H, X) \\
Y\end{array}\right)\) \\
& \begin{tabular}{l}
\(Y=\operatorname{step}(H, X, V A R)\) \\
demodulator System object, \(H\), and returns \(Y\). Input \(X\) must be a scalar \\
or a column vector with double or single precision data type. When you \\
set the BitOutput property to true and the DecisionMethod property \\
to 'Log-likelihood ratio' the input data type must be single or double \\
precision. Depending on the BitOutput property value, output \(Y\) can be \\
integer or bit valued.
\end{tabular} \\
\begin{tabular}{l}
\(Y=\) step \((H, X, V A R)\) uses soft decision demodulation and noise variance
\end{tabular} \\
& \begin{tabular}{l} 
VAR. This syntax applies when you set the Bitoutput property to true, \\
the DecisionMethod property to Approximate log-likelihood ratio \\
or Log-likelihood ratio, and the VarianceSource property to 'Input \\
port'.
\end{tabular}
\end{tabular}

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose \\ Description}

\section*{Construction}

\section*{Properties}

Modulate using arbitrary QAM constellation

The GeneralQAMModulator object modulates using quadrature amplitude modulation. The output is a baseband representation of the modulated signal.

H = comm.GeneralQAMModulator creates a modulator System object, H. This object modulates the input signal using a general quadrature amplitude modulation (QAM) method.

H = comm.GeneralQAMModulator(Name,Value) creates a QAM modulator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.GeneralQAMModulator(CONST, Name, Value) creates a General QAM modulator object, H. This object has the Constellation property set to CONST, and the other specified properties set to the specified values.

\section*{Constellation}

Signal constellation
Specify the constellation points as a vector of real or complex
double-precision data type. The default is \(\exp (2 \times \pi \times 1 i \times(0: 7) / 8)\). The length of the vector determines the modulation order. The step method inputs requires integers between 0 and \(M-1\), where \(M\) indicates the length of this property vector. The object maps an input integer \(m\) to the \((m+1)^{\text {st }}\) value in the Constellation vector.

\section*{OutputDataType}

Data type of output
Specify the output data type as one of double | single | Custom. The default is double.

\section*{Fixed-Point Properties}

\section*{CustomOutputDataType}

Fixed-point data type of output
Specify the output fixed-point type as a numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the OutputDataType property to Custom.

\author{
Methods clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release \\ step
}

Examples Modulate data using an arbitrary 3-point constellation. Then, visualize the data in a scatter plot
```

    hQAMMod = comm.GeneralQAMModulator;
    % Setup a three point constellation
hQAMMod.Constellation = [1 1i -1];
data = randi([0 2],100,1);
modData = step(hQAMMod, data);
scatterplot(modData)

```

\title{
Algorithms This object implements the algorithm, inputs, and outputs described on the General QAM Modulator Baseband block reference page. The object properties correspond to the block parameters.
}

See Also comm.GeneralQAMDemodulator | comm.RectangularQAMModulator

\section*{comm.GeneralQAMModulator.clone}

Purpose Create general QAM modulator object with same property values

\section*{Syntax \\ C = clone( H )}

Description \(\quad C=\) clone \((H)\) creates a GeneralQAMModulator object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.GeneralQAMModulator.getNumInputs}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(\quad N=\) getNumInputs \((H)\) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.GeneralQAMModulator.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\title{
Purpose Locked status for input attributes and nontunable properties
}

Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the GeneralQAMModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

\section*{Syntax release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{Purpose Modulate using arbitrary QAM constellation}
\[
\text { Syntax } \quad Y=\operatorname{step}(H, X)
\]

Description
\(Y=\operatorname{step}(H, X)\) modulates input data, \(X\), with the general QAM modulator System object, H. It returns the baseband modulated output, Y. The input must be an integer scalar or an integer-valued column vector. The data type of the input can be numeric or unsigned fixed point of word length ceil( \(\log 2\) (ModulationOrder)) (fi object).

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Demodulate convolutionally encoded data mapped to arbitrary QAM} constellation

\section*{Description \\ The GeneralQAMTCMDemodulator object uses the Viterbi algorithm to decode a trellis-coded modulation (TCM) signal that was previously modulated using an arbitrary signal constellation.}

\section*{Construction}

H = comm.GeneralQAMTCMDemodulator creates a trellis-coded, general quadrature amplitude (QAM TCM) demodulator System object, H. This object demodulates convolutionally encoded data that has been mapped to an arbitrary QAM constellation.

H = comm.GeneralQAMTCMDemodulator (Name, Value) creates a general QAM TCM demodulator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.GeneralQAMTCMDemodulator(TRELLIS,Name, Value) creates a general QAM TCM demodulator object, H. This object has the TrellisStructure property set to TRELLIS, and the other specified properties set to the specified values.

\section*{Properties}

\section*{TrellisStructure}

Trellis structure of convolutional code
Specify trellis as a MATLAB structure that contains the trellis description of the convolutional code. Use the istrellis function to check if a structure is a valid trellis structure. The default is the value that results from poly2trellis([1 3], [100; 0 ( 5 2]).

\section*{TerminationMethod}

Termination method of encoded frame
Specify the termination method as one of Continuous | Truncated | Terminated. The default is Continuous.

When you set this property to Continuous, the object saves the internal state metric at the end of each frame. The next frame
uses the same state metric. The object treats each traceback path independently. If the input signal contains only one symbol, use Continuous mode.

When you set this property to Truncated, the object treats each input vector independently. The traceback path starts at the state with the best metric and always ends in the all-zeros state.

When you set this property to Terminated, the object treats each input vector independently, and the traceback path always starts and ends in the all-zeros state.

\section*{TracebackDepth}

Traceback depth for Viterbi decoder
Specify the scalar, integer number of trellis branches to construct each traceback path. The default is 21 . The Traceback depth parameter influences the decoding accuracy and delay. The decoding delay indicates the number of zero symbols that precede the first decoded symbol in the output.

When you set the TerminationMethod property to Continuous, the decoding delay consists of TracebackDepth zero symbols or TracebackDepth \(\times K\) zero bits for a rate \(K / N\) convolutional code.

When you set the TerminationMethod property to Truncated or Terminated, no output delay occurs and the traceback depth must be less than or equal to the number of symbols in each input vector.

\section*{ResetInputPort}

Enable demodulator reset input
Set this property to true to enable an additional input to the step method. The default is false. When this additional reset input is a nonzero value, the internal states of the encoder reset to their initial conditions. This property applies when you set the TerminationMethod property to Continuous.

\section*{Constellation}

Signal constellation
Specify a double- or single-precision complex vector. This vector lists the points in the signal constellation that were used to map the convolutionally encoded data. The constellation must be specified in set-partitioned order. See documentation for the General TCM Encoder block for more information on set-partitioned order. The length of the constellation vector must equal the number of possible input symbols to the convolutional decoder of the general QAM TCM demodulator object. This corresponds to \(2^{N}\) for a rate \(K / N\) convolutional code. The default corresponds to a set-partitioned order for the points of an 8-PSK signal constellation. This value is expressed as


\section*{OutputDataType}

Data type of output
Specify output data type as one of logical \| double. The default is double.
\begin{tabular}{lll} 
Methods & clone & \begin{tabular}{l} 
Create general QAM TCM \\
demodulator object with same \\
property values
\end{tabular} \\
getNumInputs & \begin{tabular}{l} 
Number of expected inputs to \\
step method
\end{tabular} \\
& getNumOutputs & \begin{tabular}{l} 
Number of outputs from step \\
method
\end{tabular} \\
isLocked & \begin{tabular}{l} 
Locked status for input attributes \\
and nontunable properties
\end{tabular} \\
& release & \begin{tabular}{l} 
Allow property value and input \\
characteristics changes
\end{tabular}
\end{tabular}
reset
step

Reset states of the general QAM TCM demodulator object
Demodulate convolutionally encoded data mapped to arbitrary QAM constellation

\section*{Examples}

\section*{Algorithms}

This object implements the algorithm, inputs, and outputs described on the General TCM Decoder block reference page. The object properties correspond to the block parameters.

\author{
See Also
}

Modulate and demodulate data using QAM TCM modulation with an arbitrary 4-point constellation.
```

% Define a trellis structure with binary inputs and 4-ary symbol outp
t = poly2trellis(7,[171 133]);
const = exp(pi*1i*[1 2 3 6]/4);
hMod = comm.GeneralQAMTCMModulator(t, 'Constellation', const);
hAWGN = comm.AWGNChannel('NoiseMethod', ...
'Signal to noise ratio (SNR)', ...
'SNR',5, 'SignalPower',0.875);
hDemod = comm.GeneralQAMTCMDemodulator(t, 'Constellation', const)
% Create an error rate calculator with delay in bits equal to Traceba
hError = comm.ErrorRate(...
'ReceiveDelay', hDemod.TracebackDepth*log2(t.numInputSymbo*
for counter = 1:10
data = randi([0 1],500,1);
modSignal = step(hMod, data);
noisySignal = step(hAWGN, modSignal);
receivedData = step(hDemod, noisySignal);
errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
errorStats(1), errorStats(2))

```
comm.GeneralQAMTCMModulator |
comm.RectangularQAMTCMDemodulator | comm.ViterbiDecoder

\section*{comm.GeneralQAMTCMDemodulator.clone}

Purpose \(\quad \begin{aligned} & \text { Create general QAM TCM demodulator object with same property } \\ & \text { values }\end{aligned}\)
Syntax \(\quad C=\) clone \((H)\)
Description \(\quad C=\) clone \((H)\) creates a GeneralQAMTCMDemodulator object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.GeneralQAMTCMDemodulator.getNumInputs}

\section*{Purpose Number of expected inputs to step method}

\section*{Syntax \\ N = getNumInputs( H )}

Description \(\quad N=\) getNumInputs \((H)\) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs(H)

\section*{comm.GeneralQAMTCMDemodulator.getNumOutputs}

Purpose \(\quad\) Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.GeneralQAMTCMDemodulator.isLocked}
\begin{tabular}{ll} 
Purpose & Locked status for input attributes and nontunable properties \\
Syntax & TF = isLocked (H) \\
Description & \begin{tabular}{l} 
TF = isLocked \((H)\) returns the locked status, TF of the \\
GeneralQAMTCMDemodulator System object.
\end{tabular} \\
\begin{tabular}{l} 
The isLocked method returns a logical value that indicates whether \\
input attributes and nontunable properties for the object are locked. The \\
object performs an internal initialization the first time the step method \\
is executed. This initialization locks nontunable properties and input \\
specifications, such as dimensions, complexity, and data type of the \\
input data. After locking, the isLocked method returns a true value.
\end{tabular}
\end{tabular}

Purpose Allow property value and input characteristics changes

\section*{Syntax release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{comm.GeneralQAMTCMDemodulator.reset}
\begin{tabular}{ll} 
Purpose & Reset states of the general QAM TCM demodulator object \\
Syntax & \(\operatorname{reset}(H)\) \\
Description & \(\operatorname{reset}(H)\) resets the states of the GeneralQAMTCMDemodulator object, H.
\end{tabular}

Purpose \(\begin{aligned} & \text { Demodulate convolutionally encoded data mapped to arbitrary QAM } \\ & \text { constellation }\end{aligned}\)
Syntax
\(Y=\operatorname{step}(H, X)\)
\(Y=\operatorname{step}(H, X, R)\)

\section*{Description}
\(\mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X})\) demodulates the general QAM modulated input data, \(X\), and uses the Viterbi algorithm to decode the resulting demodulated convolutionally encoded bits. X must be a complex double or single precision column vector. The step method outputs a demodulated binary column data vector, Y . When the convolutional encoder represents a rate \(K / N\) code, the length of the output vector equals \(K \times L\), where \(L\) is the length of the input vector, \(X\).
\(Y=\operatorname{step}(H, X, R)\) resets the decoder states of the general QAM TCM demodulator System object to the all-zeros state when you input a non-zero reset signal, R. R must be a double precision or logical scalar integer. This syntax applies when you set the Reset InputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Convolutionally encode binary data and map using arbitrary QAMconstellation
Description
Construction
Properties
H = comm.GeneralQAMTCMModulator creates a trellis-coded, general quadrature amplitude (QAM TCM) modulator System object, H. This object convolutionally encodes a binary input signal and maps the result using QAM modulation with a signal constellation specified in the Constellation property.
H = comm.GeneralQAMTCMModulator (Name, Value) creates a general QAM TCM modulator System object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).
H = comm.GeneralQAMTCMModulator(TRELLIS,Name,Value) creates a general QAM TCM modulator System object, H. This object has the TrellisStructure property set to TRELLIS, and the other specified properties set to the specified values.

\section*{TrellisStructure}
Trellis structure of convolutional code
Specify trellis as a MATLAB structure that contains the trellis description of the convolutional code. Use the istrellis function to check if a structure is a valid trellis structure. The default is


\section*{TerminationMethod}
Termination method of encoded frame
Specify the termination method as one of Continuous | Truncated | Terminated. The default is Continuous.

When you set this property to Continuous, the object retains the encoder states at the end of each input vector for use with the next input vector.
When you set this property to Truncated, the object treats each input vector independently. The encoder is reset to the all-zeros state at the start of each input vector.

When you set this property to Terminated, the object treats each input vector independently. For each input vector, the object uses extra bits to set the encoder to the all-zeros state at the end of the vector. For a rate \(K / N\) code, the step method
outputs the vector with length \(y=N \times(L+S) / K\), where \(S=\) constraintLength -1 . In the case of multiple constraint lengths, \(S\) \(=\operatorname{sum}(\) constraintLength(i)-1)). L represents the length of the input to the step method.

\section*{ResetInputPort}

Enable modulator reset input
Set this property to true to enable an additional input to the step method. The default is false. When this additional reset input is a nonzero value, the internal states of the encoder reset to their initial conditions. This property applies when you set the TerminationMethod property to Continuous.

\section*{Constellation}

\section*{Signal constellation}

Specify a double- or single-precision complex vector that lists the points in the signal constellation that were used to map the convolutionally encoded data. You must specify the constellation in set-partitioned order. See documentation for the General TCM Encoder block for more information on set-partitioned order. The length of the constellation vector must equal the number of possible input symbols to the convolutional decoder of the general QAM TCM demodulator object. This corresponds to \(2^{N}\) for a rate \(K / N\) convolutional code. The
default corresponds to a set-partitioned order for the points of an 8-PSK signal constellation. This value is expressed
\[
\exp \left(2 \times \pi \times j \times\left[\begin{array}{llllllll}
0 & 4 & 2 & 6 & 1 & 5 & 3 & 7
\end{array}\right] / 8\right)
\]

\section*{OutputDataType}

Data type of output
Specify the output data type as one of double | single. The default is double.
\begin{tabular}{lll} 
Methods & clone & \begin{tabular}{l} 
Create general QAM TCM \\
modulator object with same \\
property values
\end{tabular} \\
& getNumInputs & \begin{tabular}{l} 
Number of expected inputs to \\
step method
\end{tabular} \\
getNumOutputs & \begin{tabular}{l} 
Number of outputs from step \\
method
\end{tabular} \\
& isLocked & \begin{tabular}{l} 
Locked status for input attributes \\
and nontunable properties
\end{tabular} \\
release & \begin{tabular}{l} 
Allow property value and input \\
characteristics changes
\end{tabular} \\
reset states of the general QAM
\end{tabular}
```

% Use the trellis structure with generating polynomial [171 133] and 4-po
t = poly2trellis(7,[171 133]);
hMod = comm.GeneralQAMTCMModulator(t,...
'Constellation', exp(pi*1i*[1 2 3 6]/4));
% Modulate and plot the data
modData = step(hMod, data);
scatterplot(modData);

```

\author{
Algorithms \\ This object implements the algorithm, inputs, and outputs described on the General TCM Encoder block reference page. The object properties correspond to the block parameters. \\ See Also \\ comm.GeneralQAMTCMDemodulator | comm.GeneralQAMModulator | comm. PSKTCMModulator | comm. ConvolutionalEncoder
}

Purpose
Syntax \(\quad C=\) clone \((H)\)
Description
\(\mathrm{C}=\) clone \((\mathrm{H})\) creates a GeneralQAMTCMModulator object C , with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.GeneralQAMTCMModulator.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)
Purpose Number of outputs from step method

Syntax \(\quad N=\) getNumOutputs (H)
Description \(\quad N=\) getNum0utputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description tTF = isLocked (H) returns the locked status, TF of the GeneralQAMTCMModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

\section*{comm.GeneralQAMTCMModulator.release}
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\title{
Purpose Reset states of the general QAM TCM modulator object
}

\section*{Syntax reset (H)}

Description reset \((H)\) resets the states of the GeneralQAMTCMModulator object, H.

\section*{Purpose}

Syntax
\(Y=\operatorname{step}(H, X)\)
\(Y=\operatorname{step}(H, X, R)\) constellation

Convolutionally encode binary data and map using arbitrary QAM
\(Y=\operatorname{step}(H, X)\) convolutionally encodes and modulates the input data, \(X\), and returns the encoded and modulated data, Y . X must be of data type numeric, logical, or unsigned fixed point of word length 1 (fi object). When the convolutional encoder represents a rate \(K / N\) code, the length of the input vector, X , must be \(K \times L\), for some positive integer \(L\). The step method outputs a complex column vector, Y, of length \(L\).
\(Y=\operatorname{step}(H, X, R)\) resets the encoder of the general QAM TCM modulator object to the all-zeros state when you input a non-zero reset signal, R. R must be a double precision or logical scalar integer. This syntax applies when you set the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Demodulate using GMSK method and the Viterbi algorithm
Description The GMSKDemodulator object uses a Viterbi algorithm to demodulate asignal that was modulated using the Gaussian minimum shift keyingmethod. The input is a baseband representation of the modulatedsignal.
Construction H = comm.GMSKDemodulator creates a demodulator System object, H.This object demodulates the input Gaussian minimum shift keying(GMSK) modulated data using the Viterbi algorithm.
H = comm.GMSKDemodulator(Name, Value) creates a GMSKdemodulator object, H. This object has each specified property set to thespecified value. You can specify additional name-value pair argumentsin any order as (Name1,Value1,...,NameN,ValueN).
Properties BitOutput
Output data as bitsSpecify whether the output is groups of bits or integer values.The default is false.
When you set the BitOutput property to false, the step method outputs a column vector of length equal to \(N /\) SamplesPerSymbol. \(N\) is the length of the input signal, which is the number of input baseband modulated symbols. The elements of the output vector are 1 or 1 .
When you set the BitOutput property to true, the step method outputs a binary column vector of length equal to \(N /\) SamplesPerSymbol with bit values of 0 or 1 .

\section*{BandwidthTimeProduct}
Product of bandwidth and symbol time of Gaussian pulse
Specify the product of bandwidth and symbol time for the Gaussian pulse shape as a real, positive scalar. The default 0.3.

\section*{PulseLength}

\section*{Pulse length}

Specify the length of the Gaussian pulse shape in symbol intervals as a real positive integer. The default 4.

\section*{SymbolPrehistory}

Symbol prehistory
Specify the data symbols used by the modulator prior to the first call to the step method. The default is 1 . This property requires a scalar or vector with elements equal to -1 or 1 . If the value is a vector, its length must be one less than the value you set in the PulseLength property.

\section*{InitialPhaseOffset}

Initial phase offset
Specify the initial phase offset of the input modulated waveform in radians as a real, numeric scalar value. The default is 0 .

\section*{SamplesPerSymbol}

Number of samples per input symbol
Specify the expected number of samples per input symbol as a positive, integer scalar value. The default is 8 .

\section*{TracebackDepth}

Traceback depth for Viterbi algorithm
Specify the number of trellis branches that the Viterbi algorithm uses to construct each traceback path as a positive, integer scalar value. The value of this property is also the output delay, and the number of zero symbols that precede the first meaningful demodulated symbol in the output. The default is 16 .

\section*{OutputDataType}

Data type of output
Specify the output data type as one of int8 | int16 | int32 | double, when you set the BitOutput property to false.

\section*{comm.GMSKDemodulator}

When you set the BitOutput property to true, specify the output data type as one of logical | double. The default is double.
```

Methods

```
clone
getNumInputs
getNumOutputs
isLocked
release
reset
step

Create GMSK demodulator object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of the GMSK demodulator object

Demodulate using GMSK method and the Viterbi algorithm

\section*{Examples}
```

Modulate and demodulate a signal using GMSK modulation with bit inputs and an initial phase offset of pi/4.

```
```

    hMod = comm.GMSKModulator('BitInput', true, ...
    ```
    hMod = comm.GMSKModulator('BitInput', true, ...
    'InitialPhaseOffset', pi/4);
    'InitialPhaseOffset', pi/4);
hAWGN = comm.AWGNChannel('NoiseMethod', ...
hAWGN = comm.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)','SNR',O);
    'Signal to noise ratio (SNR)','SNR',O);
hDemod = comm.GMSKDemodulator('BitOutput', true, ...
hDemod = comm.GMSKDemodulator('BitOutput', true, ...
    'InitialPhaseOffset', pi/4);
    'InitialPhaseOffset', pi/4);
% Create an error rate calculator, account for the delay caused by the
% Create an error rate calculator, account for the delay caused by the
hError = comm.ErrorRate('ReceiveDelay', hDemod.TracebackDepth);
hError = comm.ErrorRate('ReceiveDelay', hDemod.TracebackDepth);
for counter = 1:100
for counter = 1:100
    % Transmit 100 3-bit words
    % Transmit 100 3-bit words
    data = randi([0 1],300,1);
    data = randi([0 1],300,1);
    modSignal = step(hMod, data);
```

    modSignal = step(hMod, data);
    ```
```

    noisySignal = step(hAWGN, modSignal);
    receivedData = step(hDemod, noisySignal);
    errorStats = step(hError, data, receivedData);
    end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
errorStats(1), errorStats(2))

```

\title{
Algorithms
}

This object implements the algorithm, inputs, and outputs described on the GMSK Demodulator Baseband block reference page. The object properties correspond to the block parameters.

See Also
comm.GMSKModulator | comm.CPMModulator | comm.CPMDemodulator

\section*{comm.GMSKDemodulator.clone}

Purpose Create GMSK demodulator object with same property values

\section*{Syntax \\ C = clone( H )}

Description \(\quad\) C \(=\) clone \((H)\) creates a GMSKDemodulator object \(C\), with the same property values as \(H\). The clone method creates a new unlocked object with uninitialized states.

\section*{comm.GMSKDemodulator.getNuminputs}

\section*{Purpose Number of expected inputs to step method}

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs(H)

\section*{comm.GMSKDemodulator.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\title{
Purpose Locked status for input attributes and nontunable properties
}
\[
\text { Syntax } \quad T F=\text { isLocked }(H)
\]

Description TF = isLocked (H) returns the locked status, TF of the GMSKDemodulator System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

\section*{Syntax \\ release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{ll} 
Purpose & Reset states of the GMSK demodulator object \\
Syntax & reset \((H)\) \\
Description & reset \((H)\) resets the states of the GMSKDemodulator object, H.
\end{tabular}

Purpose Demodulate using GMSK method and the Viterbi algorithm

\section*{Syntax \(\quad Y=\operatorname{step}(H, X)\)}

Description \(\quad Y=\operatorname{step}(H, X)\) demodulates input data, \(X\), with the GMSK demodulator object, H , and returns Y . X must be a double or single precision column vector with a length equal to an integer multiple of the number of samples per symbol you specify in the SamplesPerSymbol property.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Modulate using GMSK method
Description The GMSKModulator object modulates using the Gaussian minimumshift keying method. The output is a baseband representation of themodulated signal.
Construction H = comm.GMSKModulator creates a modulator System object, H. Thisobject modulates the input signal using the Gaussian minimum shiftkeying (GMSK) modulation method.
H = comm.GMSKModulator (Name, Value) creates a GMSK modulator object, H. This object has each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

\section*{Properties}

\section*{BitInput}
Assume input is bits
Specify whether the input is bits or integers. The default is false.
When you set the BitInput property to false, the step method input requires a double-precision or signed integer data type column vector with values of -1 or 1 .
When you set the BitInput property to true, step method input requires a double-precision or logical data type column vector of 0 s and 1 s .

\section*{BandwidthTimeProduct}
Product of bandwidth and symbol time of Gaussian pulse
Specify the product of the bandwidth and symbol time for the Gaussian pulse shape as a real, positive scalar value. The default is 0.3 .

\section*{PulseLength}
Pulse length

Specify the length of the Gaussian pulse shape in symbol intervals as a real, positive integer. The default is 4 .

\section*{SymbolPrehistory}

Symbol prehistory
Specify the data symbols the modulator uses prior to the first call to the step method in reverse chronological order. The default is 1. This property requires a scalar or vector with elements equal to -1 or 1 . If the value is a vector, then its length must be one less than the value in the PulseLength property.

\section*{InitialPhaseOffset}

Initial phase offset
Specify the initial phase of the modulated waveform in radians as a real, numeric scalar value. The default is 0 .

\section*{SamplesPerSymbol}

Number of samples per output symbol
Specify the upsampling factor at the output as a real, positive, integer scalar value. The default is 8 . The upsampling factor is the number of output samples that the step method produces for each input sample.

\section*{OutputDataType}

Data type of output
Specify output data type as one of double | single. The default is double.

\section*{Methods clone}
getNumInputs
Create GMSK modulator object with same property values
Number of expected inputs to step method
getNumOutputs
isLocked
release
reset
step

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of the GMSK modulator object

Modulate using GMSK method

\section*{Examples}

\section*{Algorithms}

Modulate and demodulate a signal using GMSK modulation with bit inputs and an initial phase offset of pi/4.
```

    hMod = comm.GMSKModulator('BitInput', true, 'InitialPhaseOffset',
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)','SNR',O);
    hDemod = comm.GMSKDemodulator('BitOutput', true, ...
                            'InitialPhaseOffset', pi/4);
    % Create an error rate calculator, account for the delay caused by the
hError = comm.ErrorRate('ReceiveDelay', hDemod.TracebackDepth);
for counter = 1:100
% Transmit 100 3-bit words
data = randi([0 1],300,1);
modSignal = step(hMod, data);
noisySignal = step(hAWGN, modSignal);
receivedData = step(hDemod, noisySignal);
errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
errorStats(1), errorStats(2))

```

This object implements the algorithm, inputs, and outputs described on the GMSK Modulator Baseband block reference page. The object properties correspond to the block parameters.

See Also comm.GMSKDemodulator | comm.CPMModulator | comm.CPMDemodulator

Purpose Create GMSK modulator object with same property values

\section*{Syntax \\ C = clone( H )}

Description
\(C=\) clone (H) creates a GMSKModulator object \(C\), with the same property values as \(H\). The clone method creates a new unlocked object with uninitialized states.

\section*{comm.GMSKModulator.getNuminputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs (H) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.GMSKModulator.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description TF = isLocked (H) returns the locked status, TF of the GMSKModulator System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\author{
Purpose Reset states of the GMSK modulator object
}

\section*{Syntax reset (H)}

Description reset \((H)\) resets the states of the GMSKModulator object, H.

\section*{Purpose Modulate using GMSK method}
\[
\text { Syntax } \quad Y=\operatorname{step}(H, X)
\]

Description \(\quad \mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X})\) modulates input data, X , with the GMSK modulator object, H . It returns the baseband modulated output in Y . Depending on the BitInput property value, input \(X\) can be a double precision, signed integer, or logical column vector. The length of vector \(Y\) is equal to the number of input samples times the number of samples per symbol that you specify in the SamplesPerSymbol property.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
\begin{tabular}{|c|c|}
\hline Purpose & Recover symbol timing phase using fourth-order nonlinearity method \\
\hline Description & The GMSKTimingSynchronizer object recovers the symbol timing phase of the input signal using a fourth-order nonlinearity method. This block implements a general non-data-aided feedback method. This timing synchronization is a non-data-aided feedback method that is independent of carrier phase recovery, but requires prior compensation for the carrier frequency offset. You can use this block for systems that use Gaussian minimum shift keying (GMSK) modulation. \\
\hline Construction & \begin{tabular}{l}
H = comm.GMSKTimingSynchronizer creates a timing phase synchronizer System object, H. This object recovers the symbol timing phase of the GMSK input signal using a fourth-order nonlinearity method. \\
H = comm.GMSKTimingSynchronizer(Name, Value) creates a GMSK timing synchronizer object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).
\end{tabular} \\
\hline Properties & \begin{tabular}{l}
SamplesPerSymbol \\
Number of samples representing each symbol \\
Specify the number of samples that represent each symbol in the input signal as an integer-valued scalar value greater than 1. The default is 4 .
\end{tabular} \\
\hline & ErrorUpdateGain \\
\hline & Error update step size \\
\hline & Specify the step size for updating successive timing phase estimates as a positive real scalar value. Typically, this number is less than \(1 /\) SamplesPerSymbol, which corresponds to a slowly varying timing phase. The default is 0.05 . This property is tunable. \\
\hline & ResetInputPort \\
\hline
\end{tabular}

Enable synchronization reset input
Set this property to true to enable resetting the timing phase recovery process based on an input argument value. The default is false.

When you set this property to true, you must specify a reset input value to the step method.

When you specify a nonzero value as the reset input, the object restarts the timing phase recovery process. When you set this property to false, the object does not restart.

\section*{ResetCondition}

Condition for timing phase recovery reset
Specify the conditions to reset the timing phase recovery process as one of Never | Every frame. The default is Never.

When you set this property to Never, the phase recovery process never restarts. The object operates continuously, retaining information from one symbol to the next.

When you set this property to Every frame, the timing phase recovery restarts at the start of each frame of data. In this case, the restart occurs at each step method call. This property applies when you set the ResetInputPort property to false.

\section*{Methods}
clone
getNumInputs
getNumOutputs
isLocked

Create GMSK timing phase synchronizer object with same property values
Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

\section*{comm.GMSKTimingSynchronizer}
\begin{tabular}{ll} 
release & \begin{tabular}{l} 
Allow property value and input \\
characteristics changes
\end{tabular} \\
reset & \begin{tabular}{l} 
Reset states of GMSK timing \\
phase synchronizer object
\end{tabular} \\
step & \begin{tabular}{l} 
Recover symbol timing phase \\
using fourth-order nonlinearity \\
method
\end{tabular}
\end{tabular}

\section*{Examples Recover timing phase of an MSK signal.}
```

% Create System objects
hMod = comm.GMSKModulator('BitInput', true, ...
'SamplesPerSymbol', 14);
timingOffset = 0.2; % Actual timing offset
hDelay = dsp.VariableFractionalDelay;
hSync = comm.GMSKTimingSynchronizer('SamplesPerSymbol', 14, ...
'ErrorUpdateGain', 0.05);
phEst = zeros(1, 10);
for i = 1:51
data = randi([0 1], 100, 1); % generate data
modData = step(hMod, data); % modulate data
% data impaired by timing offset error
impairedData = step(hDelay, modData, timingOffset*14);
% perform timing phase recovery
[y, phase] = step(hSync, impairedData);
phEst(i) = phase(1)/14;
end
figure, plot(0.2*ones(1, 50));
hold on; ylim([0 0.4])
plot(phEst, 'r'); legend( 'original', 'estimated')
title('Original and Estimated timing phases');

```

Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the MSK-Type Signal Timing Recovery block reference page. The object properties correspond to the block parameters, except:
- The object corresponds to the MSK-Type Signal Timing Recovery block with the Modulation type parameter set to GMSK.
- The Reset parameter corresponds to the ResetInputPort and ResetCondition properties.
comm.EarlyLateGateTimingSynchronizer |
comm. MuellerMullerTimingSynchronizer

\section*{comm.GMSKTimingSynchronizer.clone}

Purpose Create GMSK timing phase synchronizer object with same property values

\section*{Syntax \\ C = clone(H)}

Description
C = clone (H) creates a GMSKTimingSynchronizer object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.GMSKTimingSynchronizer.getNumInputs}

\section*{Purpose Number of expected inputs to step method}

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(N=\) getNumInputs \((H)\) method returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs \((H)\).

\section*{comm.GMSKTimingSynchronizer.getNumOutputs}

Purpose \(\quad\) Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.GMSKTimingSynchronizer.isLocked}
\begin{tabular}{ll} 
Purpose & Locked status for input attributes and nontunable properties \\
Syntax & TF \(=\) isLocked \((H)\) \\
Description & \begin{tabular}{l} 
TF \(=\) isLocked \((H)\) returns the locked status, TF of the \\
GMSKTimingSynchronizer System object.
\end{tabular}
\end{tabular}

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

\section*{comm.GMSKTimingSynchronizer.release}

Purpose Allow property value and input characteristics changes

\section*{Syntax \\ release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\title{
comm.GMSKTimingSynchronizer.reset
}
\begin{tabular}{ll} 
Purpose & Reset states of GMSK timing phase synchronizer object \\
Syntax & reset \((H)\) \\
Description & reset \((H)\) resets the states for the GMSKTimingSynchronizer object H.
\end{tabular}

Purpose Recover symbol timing phase using fourth-order nonlinearity method
Syntax \(\quad \begin{aligned} {[\mathrm{Y}, \mathrm{PHASE}] } & =\operatorname{step}(\mathrm{H}, \mathrm{X}) \\ {[\mathrm{Y}, \mathrm{PHASE}] } & =\operatorname{step}(\mathrm{H}, \mathrm{X}, \mathrm{R})\end{aligned}\)
Description
[ Y, PHASE] \(=\operatorname{step}(\mathrm{H}, \mathrm{X})\) performs timing phase recovery and returns the time-synchronized signal, \(Y\), and the estimated timing phase, PHASE, for input signal X . X must be a double or single precision complex column vector.
[ \(\mathrm{Y}, \mathrm{PHASE}\) ] = step( \(\mathrm{H}, \mathrm{X}, \mathrm{R}\) ) restarts the timing phase recovery process when you input a reset signal, R, that is non-zero. R must be a logical or double scalar. This syntax applies when you set the Reset InputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose \\ Generate Gold sequence}

Description

\section*{Construction}

\section*{Properties}

The GoldSequence object generates a Gold sequence. Gold sequences form a large class of sequences that have good periodic cross-correlation properties.

H = comm.GoldSequence creates a Gold sequence generator System object, H. This object generates a pseudo-random Gold sequence.

H = comm.GoldSequence(Name, Value) creates a Gold sequence generator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

\section*{FirstPolynomial}

Generator polynomial for first preferred PN sequence
Specify the polynomial that determines the feedback connections for the shift register of the first preferred PN sequence generator. The default is \(\left[\begin{array}{lllllll}1 & 0 & 0 & 0 & 0 & 1 & 1\end{array}\right]\). You can specify the generator polynomial as a numeric, binary vector that lists the coefficients of the polynomial in descending order of powers. The first and last elements must equal 1 , and the length of this vector requires a value of \(n+1\), where \(n\) is the degree of the generator polynomial. Alternatively, you can specify the generator polynomial as a numeric vector containing the exponents of \(z\) for the nonzero terms of the polynomial in descending order of powers. The last entry must be 0 . For example, \(\left[\begin{array}{lllllll}1 & 0 & 0 & 0 & 0 & 1 & 1\end{array}\right]\) and [8 2

0 ] represent the same polynomial, \(g(z)=z^{8}+z^{2}+1\). The degree of the first generator polynomial must equal the degree of the second generator polynomial specified in the SecondPolynomial property.

\section*{FirstInitialConditions}

Initial conditions for first PN sequence generator

Specify the initial conditions for the shift register of the first preferred PN sequence generator. The default is [0 0 0 0 0 1]. The initial conditions require a numeric, binary scalar, or a numeric, binary vector with length equal to the degree of the first generator polynomial specified in the FirstPolynomial property. If you set this property to a vector, each element of the vector corresponds to the initial value of the corresponding cell in the shift register. If you set this property to a scalar, the initial conditions of all shift register cells are the specified scalar value.

\section*{SecondPolynomial}

Generator polynomial for second preferred PN sequence
Specify the polynomial that determines the feedback connections for the shift register of the second preferred PN sequence generator. The default is [ \(\left.\begin{array}{lllllll}1 & 1 & 0 & 0 & 1 & 1 & 1\end{array}\right]\). You can specify the generator polynomial as a binary, numeric vector that lists the coefficients of the polynomial in descending order of powers. The first and last elements must equal 1, and the length of this vector requires a value of \(n+1\), where \(n\) is the degree of the generator polynomial. Alternatively, you can specify the generator polynomial as a numeric vector containing the exponents of \(z\) for the nonzero terms of the polynomial in descending order of powers. The last entry must be 0 . For example, \(\left[\begin{array}{ll}1 & 0\end{array}\right.\) \(00001001]\) and [ \(\left.\begin{array}{lllll}8 & 2 & 0\end{array}\right]\) represent the same polynomial, \(g(z)=z^{8}+z^{2}+1\). The degree of the second generator polynomial must equal the degree of the first generator polynomial specified in the FirstPolynomial property.

\section*{SecondInitialConditionsSource}

Source of initial conditions for second PN sequence
Specify the source of the initial conditions that determines the start of the second PN sequence as one of Property | Input port. The default is Property. When you set this property to Property, you can specify the initial conditions as a scalar or binary vector using the SecondInitialConditions property. When you set
this property to Input port, you specify the initial conditions as an input to the stepmethod. The object accepts a binary scalar or a binary vector input. The length of the input must equal the degree of the generator polynomial that the SecondPolynomial property specifies.

\section*{SecondInitialConditions}

Initial conditions for second PN sequence generator
Specify the initial conditions for the shift register of the second preferred PN sequence generator as a numeric, binary scalar, or as a numeric, binary vector. The length must equal the degree of the second generator polynomial. You set the second generator polynomial in the SecondPolynomial property.

When you set this property to a vector, each element of the vector corresponds to the initial value of the corresponding cell in the shift register. The default is [0 000011\(]\).

When you set this property to a scalar, the initial conditions of all shift register cells are the specified scalar value.

\section*{Index}

Index of output sequence of interest
Specify the index of the output sequence of interest from the set of available sequences as a scalar integer. The default is 0 . The scalar integer must be in the range \(\left[-2,2^{\mathrm{n}}-2\right.\) ], where \(n\) is the degree of the generator polynomials you specify in the FirstPolynomial and SecondPolynomial properties.

The index values -2 and -1 correspond to the first and second preferred PN sequences as generated by the FirstPolynomial and SecondPolynomial, respectively.

The set \(G(u, v)\) of available Gold sequences is defined by \(G(u, v)\) \(=\left\{u, v,\left(u\right.\right.\) xor \(\left.T^{v}\right)\), \(\left(u\right.\) xor \(\left.T^{2 v}\right), \ldots,\left(u\right.\) xor \(\left.\left.T^{(N-1) v)}\right)\right\}\). In this case, \(T\) represents the operator that shifts vectors cyclically to the left by one place, and \(u, v\) represent the two preferred PN sequences.

Also, \(G(u, v)\) contains \(N+2\) Gold sequences of period \(N\). You select the desired sequence from this set using the Index property.

\section*{Shift}

Sequence offset from initial time
Specify the offset of the Gold sequence from its starting point as a numeric, integer scalar value that can be positive or negative.
The default is 0 . The Gold sequence has a period of \(N=2^{n}-1\), where \(n\) is the degree of the generator polynomials specified in the FirstPolynomial and SecondPolynomial properties. The shift value is wrapped with respect to the sequence period.

\section*{VariableSizeOutput}

Enable variable-size outputs
Set this property to true to enable an additional input to the step method. The default is false. When you set this property to true, the enabled input specifies the output size of the Gold sequence used for the step. The input value must be less than or equal to the value of the MaximumOutputSize property.

When you set this property to false, the SamplesPerFrame property specifies the number of output samples.

\section*{MaximumOutputSize}

Maximum output size
Specify the maximum output size of the Gold sequence as a positive integer 2 -element row vector. The second element of the vector must be 1 . The default is [10 1].

This property applies when you set the VariableSizeOutput property to true.

\section*{SamplesPerFrame}

Number of output samples per frame

Specify the number of Gold sequence samples that the step method outputs as a numeric, integer scalar value. The default is 1. If you set this property to a value of \(M\), then the step method outputs \(M\) samples of a Gold sequence with a period of \(N=2^{n}-1\). The value of \(n\) represents the degree of the generator polynomials that you specify in the FirstPolynomial and SecondPolynomial properties.

\section*{ResetInputPort}

Enable generator reset input
Set this property to true to enable an additional reset input to the step method. The default is false. This input resets the states of the two shift registers of the Gold sequence generator to the initial conditions specified in the FirstInitialConditions and SecondInitialConditions properties.

\section*{OutputDataType}

Data type of output
Specify the output data type as one of double | logical | Smallest unsigned integer. The default is double.

You must have a Fixed-Point Designer user license to use this property in Smallest unsigned integer mode.

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked
}

Create Gold sequence generator object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties
\begin{tabular}{ll} 
release & \begin{tabular}{l} 
Allow property value and input \\
characteristics changes
\end{tabular} \\
reset & \begin{tabular}{l} 
Reset states of Gold sequence \\
generator object
\end{tabular} \\
step & Generate a Gold sequence
\end{tabular}

\author{
Examples \\ Get 10 samples of a Gold sequence of period \(2^{5}-1\). \\ ```
hgld = comm.GoldSequence('FirstPolynomial',[5 2 0],... \\ 'SecondPolynomial', [5 4 3 2 0],... \\ 'FirstInitialConditions', [0 0 0 0 1],... \\ 'SecondInitialConditions', [0 0 0 0 1],... \\ 'Index', 4, 'SamplesPerFrame', 10); \\ x = step(hgld)
```

}

Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the Gold Sequence Generator block reference page. The object properties correspond to the block parameters.

Purpose
Create Gold sequence generator object with same property values

## Syntax <br> C = clone(H)

Description
$C=$ clone $(H)$ creates a GoldSequence object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.GoldSequence.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.GoldSequence.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description $\quad$ TF $=$ isLocked $(H)$ returns the locked status, $T F$ of the GoldSequence System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of Gold sequence generator object

## Syntax reset (H)

Description reset $(H)$ resets the states of the GoldSequence object, H.
Purpose Generate a Gold sequence

Syntax

Description
$Y=\operatorname{step}(H)$
Y = step(H,RESET)
$Y=\operatorname{step}(H)$ outputs a frame of the Gold sequence in column vector Y. Specify the frame length with the SamplesPerFrame property. The object uses two PN sequence generators to generate a preferred pair of sequences with period $\mathrm{N}=2^{\wedge} \mathrm{n}-1$. Then the object XORs these sequences to produce the output Gold sequence. The value in n is the degree of the generator polynomials that you specify in the FirstPolynomial and SecondPolynomial properties.
Y = step( H, RESET) uses RESET as the reset signal when you set the Reset InputPort property to true. The data type of the RESET input must be double precision or logical. RESET can be a scalar value or a column vector with length equal to the number of samples per frame specified in the SamplesPerFrame property. When the RESET input is a non-zero scalar, the object resets to the initial conditions that you specify in the FirstInitialConditions and SecondInitialConditions properties. It then generates a new output frame. A column vector RESET input allows multiple resets within an output frame. A non-zero value at the ith element of the vector causes a reset at the ith output sample time.
Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Add white Gaussian noise to input signal with GPU |
| :--- | :--- |
| Description | The GPU AWGNChannel object adds white Gaussian noise to an input <br> signal using a graphics processing unit (GPU). |

Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.


## Construction

H = comm.gpu.AWGNChannel creates a GPU-based additive white Gaussian noise (AWGN) channel System object, H. This object adds white Gaussian noise to a real or complex input signal.
H = comm.gpu.AWGNChannel(Name, Value) creates a GPU-based AWGN channel object, $H$, with the specified property name set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.


## Properties

NoiseMethod
Method to specify noise level
Select the method to specify the noise level as one of Signal to noise ratio (Eb/No) | Signal to noise ratio (Es/No) | Signal to noise ratio (SNR) | Variance. The default is Signal to noise ratio (Eb/No).

## EbNo

Energy per bit to noise power spectral density ratio (Eb/No)
Specify the Eb/No ratio in decibels. Set this property to a numeric, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (Eb/No). The default is 10. This property is tunable.

## EsNo

Energy per symbol to noise power spectral density ratio (Es/No)

Specify the Es/No ratio in decibels. Set this property to a numeric, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (Es/No). The default is 10. This property is tunable.

## SNR

Signal to noise ratio (SNR)
Specify the SNR value in decibels. Set this property to a numeric, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (SNR). The default is 10 . This property is tunable.

## BitsPerSymbol

Number of bits in one symbol
Specify the number of bits in each input symbol. You can set this property to a numeric, positive, integer scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio ( $\mathrm{Eb} / \mathrm{No}$ ). The default is 1 bit.

## SignalPower

Input signal power in Watts
Specify the mean square power of the input signal in Watts. Set this property to a numeric, positive, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (Eb/No), Signal to noise ratio (Es/No) or Signal to noise ratio (SNR). The default is 1 Watt. The object assumes a nominal impedance of 1 Ohm . This property is tunable.

## SamplesPerSymbol

Number of samples per symbol

Specify the number of samples per symbol. Set this property to a numeric, positive, integer scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Signal to noise ratio (Eb/No) or Signal to noise ratio (Es/No). The default is 1 sample.

## VarianceSource

Source of noise variance
Specify the source of the noise variance as one of Property | Input port. The default is Property. Set VarianceSource to Input port to specify the noise variance value via an input to the step method. Set VarianceSource to Property to specify the noise variance value using the Variance property. This property applies when you set the NoiseMethod property to Variance.

## Variance

Noise variance
Specify the variance of the white Gaussian noise. You can set this property to a numeric, positive, real scalar or row vector with a length equal to the number of channels. This property applies when you set the NoiseMethod property to Variance and the VarianceSource property to Property. The default is 1. This property is tunable.

## RandomStream

Source of random number stream
Specify the source of random number stream. The only valid setting for this property is Global stream. The object generates the normally distributed random numbers from the current global random number stream.

## Seed

Initial seed of mt19937ar random number stream
The GPU version of the AWGN Channel System object does not use this property.

Methods<br>clone<br>isLocked<br>release<br>step

Create AWGN Channel object with same property values
Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes

Add white Gaussian noise to input signal

Algorithm

## Examples

Add AWGN to an 8-PSK signal.

```
hMod = comm.PSKModulator;
modData = step(hMod,randi([0 hMod.ModulationOrder-1],1000,1));
hAWGN = comm.gpu.AWGNChannel('EbNo',15, 'BitsPerSymbol', ...
    log2(hMod.ModulationOrder));
channelOutput = step(hAWGN, modData);
```

```
% Visualize the noiseless and noisy data in scatter plots scatterplot (modData) scatterplot (channelOutput)
```

See Also
comm. AWGNChannel

Purpose Create AWGN Channel object with same property values

## Syntax

Description
C = clone (H) creates a GPU AWGN Channel object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the GPU AWGN Channel System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.gpu.AWGNChannel.release

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release (H) release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose Add white Gaussian noise to input signal

Syntax
$Y=\operatorname{step}(H, X)$
Y = step(H,X,VAR)
$Y=\operatorname{step}(H, X)$ adds white Gaussian noise to input $X$ and returns the result in $Y$. The input $X$ can be a double or single precision data type scalar, vector, or matrix with real or complex values. The dimensions of input $X$ determine single or multichannel processing. For an M-by-N matrix input, $M$ represents the number of time samples per channel and N represents the number of channels. M and N can be equal to 1 . The object adds frames of length $M$ of Gaussian noise to each of the N channels independently.
$Y=\operatorname{step}(H, X, V A R)$ uses input VAR as the variance of the white Gaussian noise. This applies when you set the NoiseMethod property to Variance and the VarianceSource property to Input port. Input VAR can be a positive scalar or row vector with a length equal to the number of channels. VAR must be of the same data type as input $X$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Restore original ordering of block interleaved sequence with GPU |
| :--- | :--- |
| Description | The BlockDeinterleaver System object restores the original ordering <br> of a sequence that was interleaved using the block interleaver System <br> object. |

> Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.


## Construction

H = comm.gpu.BlockDeinterleaver creates a GPU-based block deinterleaver System object, H. This object restores the original ordering of a sequence that was interleaved using the BlockInterleaver System object

H = comm.gpu.BlockDeinterleaver(Name, Value) creates a GPU-based block deinterleaver object, H , with the specified property name set to the specified value.

H = comm.gpu.BlockDeinterleaver(PERMVEC) creates a GPU-based block deinterleaver object, H, with the PermutationVector property set to PERMVEC.

## Properties

## PermutationVector

## Permutation vector

Specify the mapping used to permute the input symbols as a column vector of integers. The default is $[5 ; 4 ; 3 ; 2 ; 1]$. The mapping is a vector where the number of elements is equal to the length, N , of the input to the step method. Each element must be an integer between 1 and $N$, with no repeated values.

| Methods | clone |
| :--- | :--- |
|  | isLocked |
|  | release |
|  | step |

Create Block Deinterleaver object with same property values

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Deinterleave input sequence
Algorithm
This object uses the same algorithm as the comm.BlockDeinterleaver System object. See Algorithms on the comm.BlockDeinterleaver help page for details.

## Examples Interleave and deinterleave data.

```
%Example 1: Interleave and deinterleave data
    hInt = comm.gpu.BlockInterleaver([3 4 1 2]');
    hDeInt = comm.gpu.BlockDeinterleaver([3 4 1 2]');
```

```
data = randi(7, 4, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence,
% and restored sequence
[data, intData, deIntData]
```

\%Example 2: Interleave and deinterleave data with random interleaver permVec $=$ randperm(7)'; \% Random permutation vector hInt = comm.gpu.BlockInterleaver(permVec); hDeInt = comm.gpu.BlockDeinterleaver(permVec); data $=$ randi(9, 7, 1);
intData $=$ step(hInt, data);
deIntData $=$ step(hDeInt, intData);
\% compare the original sequence, interleaved sequence,
\% and restored sequence
[data, intData, deIntData]

See Also
comm.gpu.BlockInterleaver | comm.BlockDeinterleaver

## Purpose Create Block Deinterleaver object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a GPU Block Deinterleaver object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.
The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked $(H)$ returns the locked status, TF of the GPU Block Deinterleaver System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Deinterleave input sequence

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description $\quad Y=\operatorname{step}(H, X)$ restores the original ordering of the sequence, $X$, that was interleaved using a block interleaver. The step method forms the output, Y, based on the mapping specified by the PermutationVector property as Output(PermutationVector(k))=Input(k), for $k=1: N$, where $N$ is the length of the permutation vector. The input $X$ must be a column vector of the same length, $N$. The data type of $X$ can be numeric, logical, or fixed-point (fi objects). Y has the same data type as X .

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose <br> Description

## Construction

Create block interleaved sequence with GPU

The GPU BlockInterleaver object permutes the symbols in the input signal using a graphics processing unit (GPU).

Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.

H = comm.gpu.BlockInterleaver creates a GPU-based block interleaver System object, H. This object permutes the symbols in the input signal based on a permutation vector.

H = comm.gpu.BlockInterleaver(Name, Value) creates a GPU-based block interleaver object, H , with the specified property Name set to the specified Value.

H = comm.gpu.BlockInterleaver(PERMVEC) creates a GPU-based block deinterleaver object, H, with the PermutationVector property set to PERMVEC.

## Properties

## PermutationVector

Permutation vector
Specify the mapping used to permute the input symbols as a column vector of integers. The default is $[5 ; 4 ; 3 ; 2 ; 1]$. The mapping is a vector where the number of elements is equal to the length, N , of the input to the step method. Each element must be an integer between 1 and $N$, with no repeated values.

## Methods <br> clone <br> isLocked <br> release <br> step

Block Interleaver object with same property values
Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes
Permute input symbols using a permutation vector

Algorithm

Examples

The GPU Block Interleaver System object uses the same algorithm as the comm. BlockInterleaver System object. See Algorithms on the comm. BlockInterleaver help page for details.

Interleave and deinterleave data.

```
%Example 1: Interleave and deinterleave data
    hInt = comm.gpu.BlockInterleaver([3 4 1 2]');
    hDeInt = comm.gpu.BlockDeinterleaver([3 4 1 2]');
    data = randi(7, 4, 1);
    intData = step(hInt, data);
    deIntData = step(hDeInt, intData);
```

```
    % compare the original sequence, interleaved sequence,
    % and restored sequence
    [data, intData, deIntData]
%Example 2: Interleave and deinterleave data with random interleaver
permVec = randperm(7)'; % Random permutation vector
hInt = comm.gpu.BlockInterleaver(permVec);
hDeInt = comm.gpu.BlockDeinterleaver(permVec);
data = randi(9, 7, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence,
% and restored sequence
[data, intData, deIntData]
```

See Also
comm.gpu.BlockDeinterleaver | comm.BlockInterleaver

Purpose Block Interleaver object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a GPU Block Interleaver object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.
The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked $(H)$ returns the locked status, TF of the GPU Block Interleaver System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H) release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose Permute input symbols using a permutation vector

$$
\text { Syntax } \quad Y=\operatorname{step}(H, X)
$$

Description $\quad Y=\operatorname{step}(H, X)$ permutes input sequence, $X$, and returns interleaved sequence, Y . The step method forms the output Y , based on the mapping defined by the PermutationVector property as Output ( $k$ )=Input (PermutationVector(k)), for $k=1: N$, where $N$ is the length of the PermutationVector property. The input $X$ must be a column vector of length $N$. The data type of $X$ can be numeric, logical, or fixed-point (fi objects). Y has the same data type as X .

> Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Convolutionally encode binary data with GPU |
| :--- | :--- |
| Description | The GPU ConvolutionalEncoder object encodes a sequence of binary <br> input vectors to produce a sequence of binary output vectors. |

Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.


## Construction

H = comm.gpu.ConvolutionalEncoder creates a System object, H, that convolutionally encodes binary data.

H = comm.gpu.ConvolutionalEncoder(Name, Value) creates a convolutional encoder object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.gpu.ConvolutionalEncoder(TRELLIS, Name, Value) creates a convolutional encoder object, H . This object has the TrellisStructure property set to TRELLIS, and the other specified properties set to the specified values.

## Properties

## TrellisStructure

Trellis structure of convolutional code
Specify the trellis as a MATLAB structure that contains the trellis description of the convolutional code. The default is the result of poly2trellis(7, [171 133]). Use the istrellis function to check if a structure is a valid trellis structure.

## TerminationMethod

Termination method of encoded frame
Specify how the encoded frame is terminated as one of Continuous | Truncated | Terminated. The default is Continuous.

When you set this property to Continuous, the object retains the encoder states at the end of each input vector for use with the next input vector.

When you set this property to Truncated, the object treats each input vector independently and resets its states to the all-zeros state.

When you set this property to Terminated, the object treats each input vector independently. For each input vector, the object uses extra bits to set the encoder states to the all-zeros state at the end of the vector. For a rate $K / N$ code, the step
method outputs a vector with length $N \times(L+S) / K$, where $S=$ constraintLength-1. In the case of multiple constraint lengths, $S$ $=\operatorname{sum}($ constraintLength(i)-1)). $L$ is the length of the input to the step method.

## ResetInputPort

Enable encoder reset input

You cannot reset this encoder object using an input port. The only valid property setting is false.

## DelayedResetAction

Delay output reset
You cannot reset this encoder object using an input port. The only valid property setting is false.

## InitialStatelnputPort

You cannot set the initial state of this encoder object. The only valid property setting is false.

## FinalStateOutputPort

You cannot output the final state of this encoder object. The only valid property setting is false.

## PuncturePatternSource

Source of puncture pattern
Specify the source of the puncture pattern as one of None | Property. The default is None. When you set this property to None the object does not apply puncturing. When you set this property to Property, the object punctures the code. This puncturing is based on the puncture pattern vector that you specify in the PuncturePattern property. This property applies when you set the TerminationMethod property to Continuous or Truncated.

## PuncturePattern

Puncture pattern vector
Specify the puncture pattern that the object uses to puncture the encoded data as a column vector. The default is $[1 ; 1 ; 0$; $1 ; 0 ; 1]$. The vector contains 1 s and 0 s , where 0 indicates a punctured, or excluded, bit. This property applies when you set the TerminationMethod property to Continuous or Truncated and the PuncturePatternSource property to Property.

## NumFrames

Number of independent frames present in the input and output data vectors.
Specify the number of independent frames contained in a single data input/output vector. The default value of this property is 1 . The objects segments the input vector into NumFrames segments and encodes them independently. The output contains NumFrames encoded segments. This property is applicable when you set the TerminationMethod to Terminated or Truncated.

| Methods | clone | Create convolutional encoder <br> object with same property values <br> Number of expected inputs to <br> step method |
| :--- | :--- | :--- |
| getNumInputs | Number of outputs from step <br> method |  |
| getNumOutputs | Locked status for input attributes <br> and nontunable properties |  |
| isLocked | Allow property value and input <br> characteristics changes |  |
| release | Reset states of the convolutional <br> encoder object |  |
| reset | Convolutionally encode binary <br> data |  |
| Examples | step | 8-PSK-Modulation With Convolutional Encoding |
|  | Transmit a Convolutionally Encoded, 8-PSK-Modulated Bit Stream |  |
| Through an AWGN Channel. |  |  |

Create a GPU-based PSK Modulator System object that accepts a bit input signal.
hMod = comm.gpu.PSKModulator('BitInput',true);
Create a GPU-based AWGN Channel System object with a signal-to-noise ratio of seven.

```
hChan = comm.gpu.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)',...
    'SNR', 7);
```

Create a GPU-based PSK Demodulator System object that outputs a column vector of bit values.
hDemod = comm.gpu.PSKDemodulator('BitOutput',true);
Create a GPU-based Viterbi Decoder System object that accepts an input vector of hard decision values, which are zeros or ones.

```
hDec = comm.gpu.ViterbiDecoder('InputFormat','Hard');
```

Create an Error Rate System object that ignores 3 data samples before makings comparisons. The received data lags behind the transmitted data by 34 samples.

```
hError = comm.ErrorRate('ComputationDelay',3,'ReceiveDelay', 34);
```

Run the simulation by using the step method to process data.

```
for counter = 1:20
    data = randi([0 1],30,1);
    encodedData = step(hConEnc, gpuArray(data));
    modSignal = step(hMod, encodedData);
    receivedSignal = step(hChan, modSignal);
    demodSignal = step(hDemod, receivedSignal);
    receivedBits = step(hDec, demodSignal);
    errors = step(hError, data, gather(receivedBits));
end
```

Display the errors.
disp(errors)

Algorithms This object implements the algorithm, inputs, and outputs described on the Convolutional Encoder block reference page. The object properties correspond to the block parameters.<br>See Also<br>comm.gpu.ViterbiDecoder | comm.gpu.ConvolutionalDeinterleaver | comm.gpu.ConvolutionalInterleaver |<br>comm. ConvolutionalEncoder

## comm.gpu.ConvolutionalEncoder.clone

Purpose Create convolutional encoder object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a GPU ConvolutionalEncoder object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ )
Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.gpu.ConvolutionalEncoder.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :---: | :---: |
| Syntax | TF = isLocked(H) |
| Description | TF = isLocked(H) returns the locked status, TF of the GPU ConvolutionalEncoder System object. |
|  | The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value. |

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release(H) Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of the convolutional encoder object
Syntax $\quad \operatorname{reset}(H)$

Description reset $(H)$ resets the states of the GPU ConvolutionalEncoder object, $H$.

Purpose Convolutionally encode binary data

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description $\quad Y=\operatorname{step}(H, X)$ encodes the binary data, $X$, using the convolutional encoding that you specify in the TrellisStructure property. It returns the encoded data, $Y$. Both $X$ and $Y$ are column vectors of data type single, double, or logical. When the convolutional encoder represents a rate $K / N$ code, the length of the input vector equals $K \times L$, for a positive integer, $L$. The step method sets the length of the output vector, Y, to $L \times N$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

# Purpose <br> Permute input symbols using shift registers with GPU <br> The GPU ConvolutionalInterleaver object permutes the symbols in the input signal using a graphics processing unit (GPU). Internally, this class uses a set of shift registers. 

> Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.


## Construction

H = comm.gpu.ConvolutionalInterleaver creates a GPU-based convolutional interleaver System object, H. This object permutes the symbols in the input signal using a set of shift registers.

H = comm.gpu.ConvolutionalInterleaver(Name, Value) creates a GPU-based convolutional interleaver System object, H, with the specified property Name set to the specified Value. You can
specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).
H = comm.gpu.ConvolutionalInterleaver ( $\mathrm{M}, \mathrm{B}, \mathrm{IC}$ ) creates a GPU-based convolutional interleaver System object H, with the NumRegisters property set to M, the RegisterLengthStep property set to B, and the InitialConditions property set to IC. M, B, and IC are value-only arguments. To specify a value-only argument, you must also specify all preceding value-only arguments.

## Properties NumRegisters

Number of internal shift registers
Specify the number of internal shift registers as a scalar, positive integer. The default is 6 .

## RegisterLengthStep

Number of additional symbols that fit in each successive shift register

Specify the number of additional symbols that fit in each successive shift register as a positive, scalar integer. The default is 2. The first register holds zero symbols.

## InitialConditions

Initial conditions of shift registers
Specify the values that are initially stored in each shift register as a numeric scalar or vector. You do not need to specify a value for the first shift register, which has zero delay. The default is 0 . The value of the first element of this property is unimportant because the first shift register has zero delay. If you set this property to a scalar, then all shift registers, except the first one, store the same specified value. If you set it to a column vector with length equal to the value of the NumRegisters property, then the $i$-th shift register stores the $i$-th element of the specified vector.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| reset |  |
| step |  |

## Examples Interleave and deinterleave random data

Interleave and deinterleave random data. Then, compare the original sequence, interleaved sequence and restored sequence.

Create a GPU-based Convolutional Interleaver with three internal shift registers capable of fitting two additional symbols. The initial value stored in each shift register is $\left[\begin{array}{lll}-1 & -2 & -3\end{array}\right]$.

```
hInt = comm.gpu.ConvolutionalInterleaver('NumRegisters', 3, ...
    'RegisterLengthStep', 2, ...
    'InitialConditions', [-1 -2 -3]');
```

Create a GPU-based Convolutional Deinterleaver with three internal shift registers capable of fitting two additional symbols. The initial value stored in each shift register is $\left[\begin{array}{lll}-1 & -2 & -3\end{array}\right]$.

```
hDeInt = comm.gpu.ConvolutionalDeinterleaver('NumRegisters', 3, ...
    'RegisterLengthStep', 2, ...
```

```
'InitialConditions', [-1 -2 -3]');
```

Copy numeric data to the GPU.

```
data = gpuArray((0:20)');
```

Run the simulation by using the step method to process data.

```
intrlvData = step(hInt, data);
deintrlvData = step(hDeInt, intrlvData);
```

Compare the original sequence, interleaved sequence and restored sequence.
[data, intrlvData, deintrlvData]
Algorithms on the Convolutional Interleaver block reference page. The object properties correspond to the block parameters.

See Also<br>comm. ConvolutionalInterleaver | comm.gpu.ConvolutionalDeinterleaver

This object implements the algorithm, inputs, and outputs described

Purpose
Create convolutional interleaver object with same property values
Syntax $\quad C=$ clone $(H)$
Description

C = clone (H) creates a GPU ConvolutionalInterleaver object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.
The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)
Purpose Number of outputs from step method

Syntax $\quad N=$ getNumOutputs $(H)$
Description $N=$ getNumOutputs (H) returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the GPU ConvolutionalInterleaver System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset states of the convolutional interleaver object |
| :--- | :--- |
| Syntax | $\operatorname{reset}(H)$ |
| Description | reset $(H)$ resets the states of the GPU ConvolutionalInterleaver <br> object, $H$. |

## Purpose Permute input symbols using shift registers

$$
\text { Syntax } \quad Y=\operatorname{step}(H, x)
$$

$Y=\operatorname{step}(H, X)$ permutes input sequence, $X$, and returns interleaved sequence, Y . The input X must be a column vector. The data type can be of type double, single, uint32, int32, or logical. $Y$ has the same data type as X . The convolutional interleaver object uses a set of $N$ shift registers, where $N$ is the value specified by the NumRegisters property. The object sets the delay value of the $k^{\text {th }}$ shift register to the product of ( $k-1$ ) and the RegisterLengthStep RegisterLengthStep property value. With each new input symbol, a commutator switches to a new register and the new symbol shifts in while the oldest symbol in that register shifts out. When the commutator reaches the $N^{\mathrm{th}}$ register and the next new input occurs, it returns to the first register.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Restore ordering of symbols using shift registers with GPU |
| :--- | :--- |
| Description | The GPU ConvolutionalDeinterleaver object recovers a signal that <br> was interleaved using the GPU-based convolutional interleaver object. <br> The parameters in the two blocks should have the same values. |

Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.


## Construction

H = comm.gpu.ConvolutionalDeinterleaver creates a GPU-based convolutional deinterleaver System object, H. This object restores the original ordering of a sequence that was interleaved using a convolutional interleaver.

H = comm.gpu.ConvolutionalDeinterleaver(Name, Value) creates a GPU-based convolutional deinterleaver System object, H, with
the specified property Name set to the specified Value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.gpu.ConvolutionalDeinterleaver(M,B,IC) creates a convolutional deinterleaver System object H, with the NumRegisters property set to M, the RegisterLengthStep property set to B , and the InitialConditions property set to IC. M, B, and IC are value-only arguments. To specify a value-only argument, you must also specify all preceding value-only arguments.

## Properties NumRegisters

Number of internal shift registers
Specify the number of internal shift registers as a scalar, positive integer. The default is 6 .

## RegisterLengthStep

Number of additional symbols that fit in each successive shift register

Specify the number of additional symbols that fit in each successive shift register as a positive, scalar integer. The default is 2. The first register holds zero symbols.

## InitialConditions

Initial conditions of shift registers
Specify the values that are initially stored in each shift register (except the first shift register, which has zero delay) as a numeric scalar or vector. The default is 0 . If you set this property to a scalar, then all shift registers, except the first one, store the same specified value. If you set it to a column vector with length equal to the value of the NumRegistersproperty, then the $i$-th shift register stores the $i$-th element of the specified vector. The value of the first element of this property is unimportant, since the first shift register has zero delay.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| reset |  |
| step |  |

Create convolutional deinterleaver object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of the convolutional deinterleaver object
Permute input symbols using shift registers

## Examples Interleave and Deinterleave Random Data

Interleave and deinterleave random data. Then, compare the original sequence, interleaved sequence and restored sequence.

Create a GPU-based Convolutional Interleaver with three internal shift registers capable of fitting two additional symbols. The initial value stored in each shift register is $\left[\begin{array}{lll}-1 & -2 & -3\end{array}\right]$.
hInt = comm.gpu.ConvolutionalInterleaver('NumRegisters', 3, ...
'RegisterLengthStep', 2, ...
'InitialConditions', [-1 -2 -3]');
Create a GPU-based Convolutional Deinterleaver with three internal shift registers capable of fitting two additional symbols. The initial value stored in each shift register is $\left[\begin{array}{lll}-1 & -2 & -3\end{array}\right]$.
hDeInt = comm.gpu.ConvolutionalDeinterleaver('NumRegisters', 3, ...

```
'RegisterLengthStep', 2, ...
'InitialConditions', [-1 -2 -3]');
```

Copy numeric data to the GPU.

```
data = gpuArray((0:20)');
```

Run the simulation by using the step method to process data.

```
intrlvData = step(hInt, data);
deintrlvData = step(hDeInt, intrlvData);
```

Compare the original sequence, interleaved sequence and restored sequence.
[data, intrlvData, deintrlvData]

## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the Convolutional Deinterleaver block reference page. The object properties correspond to the block parameters.
comm. ConvolutionalDeinterleaver | comm.gpu. ConvolutionalInterleaver

Purpose Create convolutional deinterleaver object with same property values

## Syntax <br> C = clone(H)

Description $\quad \mathrm{C}=$ clone $(\mathrm{H})$ creates a GPU ConvolutionalDeinterleaver object C , with the same property values as H . The clone method creates a new unlocked object with uninitialized states.
The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

## comm.gpu.ConvolutionalDeinterleaver.getNumInputs

## Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.gpu.ConvolutionalDeinterleaver.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF = isLocked (H) |
| Description $\quad$TF = isLocked $(H)$ returns the locked status, TF of the GPU <br> ConvolutionalDeinterleaver System object. <br> The isLocked method returns a logical value that indicates whether <br> input attributes and nontunable properties for the object are locked. The <br> object performs an internal initialization the first time the step method <br> is executed. This initialization locks nontunable properties and input <br> specifications, such as dimensions, complexity, and data type of the <br> input data. After locking, the isLocked method returns a true value. |  |

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release(H) Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose

Permute input symbols using shift registers

## Syntax

Y = step(H,X)
$Y=\operatorname{step}(H, X)$ restores the original ordering of the sequence, $X$, that was interleaved using a convolutional interleaver and returns Y. The input $X$ must be a column vector. The data type can be numeric, logical, or fixed-point (fi objects). Y has the same data type as $X$. The convolutional deinterleaver object uses a set of $N$ shift registers, where N represents the value specified by the NumRegisters property. The object sets the delay value of the $k^{\text {th }}$ shift register to the product of ( $k-1$ ) and the RegisterLengthStep property value. With each new input symbol, a commutator switches to a new register and the new symbol shifts in while the oldest symbol in that register shifts out. When the commutator reaches the $N$ th register and the next new input occurs, it returns to the first register.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Reset states of the convolutional deinterleaver object

## Syntax reset (H)

Description reset (H) resets the states of the GPU ConvolutionalDeinterleaver object, H .

## Purpose <br> Description

## Construction

Decode binary low-density parity-check data with GPU

The GPU LDPCDecoder object decodes a binary low-density parity-check code using a graphics processing unit (GPU).

> Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.
h = comm.gpu.LDPCDecoder creates a GPU-based binary low-densitycreates a GPU-based LDPC decoder object, h, with the ParityCheckMatrix property set to PARITY. parity-check (LDPC) decoder System object, h. This object performs LDPC decoding based on the specified parity-check matrix, where the object does not assume any patterns in the parity-check matrix.
h = comm.gpu.LDPCDecoder('PropertyName','ValueName') creates a GPU-based LDPC decoder object, h , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as ('PropertyName1','PropertyValue1',...,'PropertyNameN','PropertyValueN'). h = comm.gpu.LDPCDecoder(PARITY) creates a GPU-based LDPC decoder object, h, with the ParityCheckMatrix property set to PARITY.


## Properties ParityCheckMatrix

Parity-check matrix
Specify the parity-check matrix as a binary valued sparse matrix with dimension ( $N$-by- $K$ ) by $N$, where $N>K>0$. This property accepts numeric or logical data types. The upper bound for the value of $N$ is $\left(2^{31}\right)-1$. The default is the parity-check matrix of the half-rate LDPC code from the DVB-S. 2 standard, which is the result of dvbs2ldpc(1/2).

## OutputValue

Select output value format
Specify the output value format as one of Information part | Whole codeword. The default is Information part. When you set this property to Information part, the output contains only the message bits and is a multiple of $K$ length column vector, assuming an ( $N$-by- $K$ ) x $K$ parity check matrix. When you set this property to Whole codeword, the output contains the codeword bits and is an $N$ element column vector.

## DecisionMethod

Decision method
Specify the decision method used for decoding as one of Hard decision | Soft decision. The default is Hard decision. When you set this property to Hard decision, the output is decoded bits of logical data type. When you set this property to Soft decision, the output is log-likelihood ratios of single or double data type.

## IterationTerminationCondition

Condition for iteration termination
Specify the condition to stop the decoding iterations as one of Maximum iteration count | Parity check satisfied. The default is Maximum iteration count. When you set this property to Maximum iteration count, the object will iterate for the number of iterations you specify in the MaximumIterationCount property. When you set this property to Parity check satisfied, the object will determine if the parity checks are satisfied after each iteration and stops if all parity checks are satisfied.

## MaximumlterationCount

Maximum number of decoding iterations
Specify the maximum number of iterations the object uses as an integer valued numeric scalar. The default is 50 . This applies when you set the IterationTerminationCondition property to Maximum iteration count.

## NumlterationsOutputPort

Output number of iterations performed
Set this property to true to output the actual number of iterations the object performed. The default is false.

## FinalParityChecksOutputPort

Output final parity checks
Set this property to true to output the final parity checks the object calculated. The default is false.

Methods<br>clone<br>isLocked<br>release<br>step

Create GPU LDPC Decoder object with same property values
Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes

Process inputs using the object algorithm

Algorithm

## Examples

The GPU LDPC Decoder System object uses the same algorithm as the LDPC Decoder block. See Decoding Algorithm for details.

Transmit an LDPC-encoded, QPSK-modulated bit stream through an AWGN channel, then demodulate, decode, and count errors.

```
hEnc = comm.LDPCEncoder;
```

hEnc = comm.LDPCEncoder;
hMod = comm.PSKModulator(4, 'BitInput',true);
hMod = comm.PSKModulator(4, 'BitInput',true);
hChan = comm.AWGNChannel(...
hChan = comm.AWGNChannel(...
'NoiseMethod','Signal to noise ratio (SNR)','SNR',1);
'NoiseMethod','Signal to noise ratio (SNR)','SNR',1);
hDemod = comm.PSKDemodulator(4, 'BitOutput',true,...
hDemod = comm.PSKDemodulator(4, 'BitOutput',true,...
'DecisionMethod','Approximate log-likelihood ratio', ...
'DecisionMethod','Approximate log-likelihood ratio', ...
'Variance', 1/10^(hChan.SNR/10));
'Variance', 1/10^(hChan.SNR/10));
hDec = comm.gpu.LDPCDecoder;
hDec = comm.gpu.LDPCDecoder;
hError = comm.ErrorRate;
hError = comm.ErrorRate;
for counter = 1:10
for counter = 1:10
data = logical(randi([0 1], 32400, 1));
data = logical(randi([0 1], 32400, 1));
encodedData = step(hEnc, data);
encodedData = step(hEnc, data);
modSignal = step(hMod, encodedData);
modSignal = step(hMod, encodedData);
receivedSignal = step(hChan, modSignal);
receivedSignal = step(hChan, modSignal);
demodSignal = step(hDemod, receivedSignal);
demodSignal = step(hDemod, receivedSignal);
receivedBits = step(hDec, demodSignal);
receivedBits = step(hDec, demodSignal);
errorStats = step(hError, data, receivedBits);
errorStats = step(hError, data, receivedBits);
end
end
fprintf('Error rate = %1.2f\nNumber of errors = %d\n', ...

```
fprintf('Error rate = %1.2f\nNumber of errors = %d\n', ...
```

```
errorStats(1), errorStats(2))
```

See Also comm.LDPCEncoder | comm. LDPCDecoder

## comm.gpu.LDPCDecoder.clone

Purpose Create GPU LDPC Decoder object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a GPU LDPCDecoder object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the ACPR System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.gpu.LDPCDecoder.release

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H) releases system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Process inputs using the object algorithm |
| :--- | :--- |
| Syntax | $Y=\operatorname{step}(H, X)$ |
|  | $[Y, \operatorname{NUMITER}]=\operatorname{step}(H, X)$ |
|  | $[Y, \operatorname{PARITY}]=\operatorname{step}(H, X)$ |

## Description

$Y=\operatorname{step}(H, X)$ decodes input codeword, $X$, using an LDPC code that is based on an ( $N-K$ ) x $N$ parity-check matrix. You specify the parity-check matrix in the ParityCheckMatrix property. The input $X$ must be a column vector of type double or single. Each element is the log-likelihood ratio for a received bit (more likely to be 0 if the log-likelihood ratio is positive). This System object is capable of decoding multiple frames of input data simultaneously. The length of the input $X$ must be a multiple of $N$. The first $K$ elements of every $N$ elements correspond to the information part of a codeword. The decoded data output vector, $Y$, contains either only the message bits or the whole code word(s), based on the value of the OutputValue property.
[ $\mathrm{Y}, \mathrm{NUMITER}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$ returns the actual number of iterations the object performed when you set the NumIterationsOutputPort property to true. The step method outputs NUMITER as a double scalar.
[ Y, PARITY] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$ returns final parity checks the object calculated when you set the FinalParityChecksOutputPort property to true. The step method outputs PARITY as a logical vector of length ( $N-K$ ).

You can combine optional output arguments when you set their enabling properties. Optional outputs must be listed in the same order as the order of the enabling properties. For example,
[ $\mathrm{Y}, \mathrm{NUMITER}, \mathrm{PARITY}$ ] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$
Calling step on an object puts that object into a locked state. When locked, you cannot change non-tunable properties or any input characteristics (size, data type and complexity) without reinitializing (unlocking and relocking) the object.

> Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Demodulate using M-ary PSK method with GPU |
| :--- | :--- |
| Description | The GPU PSKDemodulator object demodulates an input signal using the <br> M-ary phase shift keying (M-PSK) method. |

Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.


## Construction

H = comm.gpu.PSKDemodulator returns a GPU-based demodulator System object, H. This object demodulates the input signal using the M-ary phase shift keying (M-PSK) method.

H = comm.gpu.PSKDemodulator (Name, Value) creates a GPU-based M-PSK demodulator object, H, with the specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN)

H = comm.gpu.PSKDemodulator(M, PHASE, Name, Value) creates a GPU-based M-PSK demodulator object, H , with the ModulationOrder property set to M, the PhaseOffset property set to PHASE and other specified property names set to the specified values. M and PHASE are value-only arguments. To specify a value-only argument, you must also specify all preceding value-only arguments. You can specify name-value pair arguments in any order.

## Properties ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation as a positive, integer scalar. The default is 8 .

## PhaseOffset

Phase of zeroth point of constellation
Specify the phase offset of the zeroth point of the constellation, in radians, as a real scalar. The default is $\pi / 8$.

## BitOutput

Output data as bits
Specify whether the output consists of groups of bits or integer symbol values. When you set this property to true, the step method outputs a column vector of bit values with length equal to $\log 2$ (ModulationOrder) times the number of demodulated symbols. When you set this property to false, the step method outputs a column vector, with a length equal to the input data vector that contains integer symbol values between 0 and ModulationOrder-1. The default is false.

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of $\log 2$ (ModulationOrder) bits to the corresponding symbol as Binary | Gray | Custom. The default is Gray. When you set
this property to Gray, the object uses a Gray-encoded signal constellation. When you set this property to Binary, the integer $\mathrm{m}(0 \leq \mathrm{m} \leq$ ModulationOrder-1) maps to the complex value. This value is represented as $\exp \left(\mathrm{j}^{*}\right.$ PhaseOffset + j*2*pi*m/ModulationOrder). When you set this property to Custom, the object uses the signal constellation defined in the CustomSymbolMapping property.

## CustomSymbolMapping

Custom constellation encoding
Specify a custom constellation symbol mapping vector. The default is $0: 7$. This property must be a row or column vector of size ModulationOrder with unique integer values in the range [ 0 , ModulationOrder-1]. The values must be of data type double. The first element of this vector corresponds to the constellation point at an angle of $0+$ PhaseOffset, with subsequent elements running counterclockwise. The last element corresponds to the constellation point at an angle of -п/ModulationOrder + PhaseOffset. This property applies when you set the SymbolMapping property to Custom.

## DecisionMethod

Demodulation decision method
Specify the decision method that the object uses as one of Hard decision | Log-likelihood ratio | Approximate log-likelihood ratio. The default is Hard decision. When you set DecisionMethod to false, the object always performs hard decision demodulation. This property applies when you set the BitOutput property to true.

## VarianceSource

Source of noise variance
Specify the source of the noise variance as one of Property | Input port. The default is Property. This property applies when you set the BitOutput property to true and the DecisionMethod property
to Log-likelihood ratio or Approximate log-likelihood ratio.

## Variance

Specify the variance of the noise as a positive, real scalar. The default is 1 . If this value is very small (i.e., SNR is very high), then log-likelihood ratio (LLR) computations may yield Inf or -Inf. This occurs because the LLR algorithm computes the exponential value of very large or very small numbers using finite precision arithmetic. In such cases, use approximate LLR is recommended because its algorithm does not compute exponentials. This property applies when you set the BitOutput property to true, the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio, and the VarianceSource property to Property. This property is tunable.

## OutputDataType

Data type of output
When you set this property to Full precision, the output signal inherits its data type from the input signal.

| Methods | clone |
| :--- | :--- |
| constellation |  |
| isLocked |  |
| release |  |
| step |  |

Create PSK demodulator object with same property values

Calculate or plot ideal signal constellation

Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes
Demodulate using M-ary PSK method
Algorithm
The GPU PSK Demodulator System object uses the same algorithm as the comm. PSKDemodulator Communications System Toolbox object. See Decoding Algorithm for details.

## Examples Transmit an LDPC-encoded, QPSK-modulated bit stream through an AWGN channel. Then demodulate, decode, and count errors.

## 16-PSK Modulation and Demodulation

Transmit an LDPC-encoded, QPSK-modulated bit stream through an AWGN channel.
Create a GPU-based PSK Modulator System object.
hMod = comm.gpu.PSKModulator(16, 'PhaseOffset',pi/16);
Create a GPU-based AWGN Channel System object with a signal-to-noise ratio of 15 .

```
hAWGN = comm.gpu.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)','SNR',15);
```

Create a GPU-based PSK Demodulator System object.

```
hDemod = comm.gpu.PSKDemodulator(16, 'PhaseOffset',pi/16);
```

Create an error rate calculator System object.

```
hError = comm.ErrorRate;
```

Transmit a frame of data containing 50 symbols.

```
for counter = 1:100
data = gpuArray.randi([0 hMod.ModulationOrder-1], 50, 1);
```

Run the simulation by using the step method to process data.

```
modSignal = step(hMod, data);
noisySignal = step(hAWGN, modSignal);
receivedData = step(hDemod, noisySignal);
```

```
errorStats = step(hError, gather(data), gather(receivedData));
end
```

Compute the error rate results.

```
fprintf('Error rate = %f\nNumber of errors = %d\n',...
    errorStats(1), errorStats(2))
```

See Also

comm. PSKDemodulator | comm.gpu.PSKModulator

Purpose Create PSK demodulator object with same property values
Syntax

C = clone(H)

Description $\quad C=$ clone $(H)$ creates a GPU PSK Demodulator object, $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.
The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

## comm.gpu.PSKDemodulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax $\quad y=$ constellation $(h)$
constellation(h)
Description $y=$ constellation( $h$ ) returns the numerical values of the constellation.
constellation(h) generates a constellation plot for the object.

## Examples Calculate Ideal Signal Constellation for comm.gpu.PSKDemodulator

Create a comm.gpu.PSKDemodulator System object, and then calculate its ideal signal constellation.

Create a comm.gpu.PSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.gpu.PSKDemodulator
Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.gpu.PSKDemodulator

Create a comm.gpu.PSKDemodulator System object, and then plot the ideal signal constellation.

Create a comm.gpu.PSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.gpu.PSKDemodulator
Plot the ideal signal constellation by calling the constellation method.
constellation(h)

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF = isLocked (H) |
| Description | TF = isLocked $(H)$ returns the locked status, TF of the GPU PSK <br> Demodulator System object. |
| The isLocked method returns a logical value that indicates whether <br> input attributes and nontunable properties for the object are locked. The <br> object performs an internal initialization the first time the step method <br> is executed. This initialization locks nontunable properties and input <br> specifications, such as dimensions, complexity, and data type of the <br> input data. After locking, the isLocked method returns a true value. |  |

## comm.gpu.PSKDemodulator.release

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release (H) release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose <br> Demodulate using M-ary PSK method

Syntax
Y = step (H,X)
Y = step(H,X,VAR)

Y = step ( $\mathrm{H}, \mathrm{X}$ ) demodulates data, X , with the GPU PSK Demodulator System object, H, and returns Y. Input X must be a scalar or a column vector with double- or single- precision data type. Depending on the BitOutput property value, output $Y$ can be integer or bit valued.
$Y=\operatorname{step}(H, X, V A R)$ uses soft decision demodulation and noise variance VAR. This syntax applies when you set the BitOutput property to true, the DecisionMethod property to Approximate log-likelihood ratio or Log-likelihood ratio, and the VarianceSource property to Input port. The data type of input VAR must be double or single precision.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Modulate using M-ary PSK method with GPU |
| :--- | :--- |
| Description $\quad$The GPU PSKModulator object modulates a signal using the M-ary <br> phase shift keying method implemented on a graphics processing unit <br> (GPU). The input is a baseband representation of the modulated signal. <br> The input and output for this object are discrete-time signals. This <br> object accepts a scalar-valued or column vector input signal. |  |

Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.

Construction
H = comm.gpu.PSKModulator returns a GPU-based demodulator System object, H. This object modulates the input signal using the M-ary phase shift keying (M-PSK) method with soft decision using the approximate log-likelihood ratio algorithm.

H = comm.gpu.PSKModulator (Name, Value) creates a GPU-based M-PSK modulator object, H , with the specified property Name set to the specified Value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN)

H = comm.gpu.PSKModulator(M,PHASE,Name,Value) creates a GPU-based M-PSK modulator object, H, with the ModulationOrder property set to M, the PhaseOffset property set to PHASE and other specified property Names set to the specified Values. M and PHASE are value-only arguments. To specify a value-only argument, you must also specify all preceding value-only arguments. You can specify name-value pair arguments in any order.

## Properties

ModulationOrder
Number of points in signal constellation
Specify the number of points in the signal constellation as a positive, integer scalar. The default is 8 .

## PhaseOffset

Phase of zeroth point of constellation
Specify the phase offset of the zeroth point of the constellation, in radians, as a real scalar. The default is $\pi / 8$.

## BitInput

Assume bit inputs
Specify whether the input is bits or integers. The default is false. When you set this property to true, the step method input must be a column vector of bit values whose length is an integer multiple of $\log 2$ (ModulationOrder). This vector contains bit representations of integers between 0 and ModulationOrder-1. The input data type can be numeric or logical. When you set the BitInput property to false, the step method input must be a column vector of integer symbol values between 0 and ModulationOrder-1. The data type of the input must be numeric.

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of $\log 2$ (ModulationOrder) bits to the corresponding symbol as one of Binary | Gray | Custom. The default is Gray. When you set this property to Gray, the object uses a Gray-encoded signal constellation. When you set this property to Binary, the integer $\mathrm{m}(0 \leq \mathrm{m} \leq$ ModulationOrder-1) maps to the complex value $\exp \left(\mathrm{j}^{*}\right.$ PhaseOffset $+\mathrm{j}{ }^{*} 2^{*} \mathrm{pi}{ }^{*} \mathrm{~m} /$ ModulationOrder). When you set this property to Custom, the object uses the signal constellation defined in the CustomSymbolmapping property.

## CustomSymbolMapping

Custom constellation encoding
Specify a custom constellation symbol mapping vector. This property must be a row or column vector of size ModulationOrder with unique integer values in the range [0, ModulationOrder-1]. The values must be of data type double. The first element of this vector corresponds to the constellation point at an angle of $0+$ PhaseOffset, with subsequent elements running counterclockwise. The last element corresponds to the constellation point at an angle of -п/ModulationOrder + PhaseOffset. This property applies when you set the SymbolMapping property to Custom. The default is $0: 7$.

## OutputDataType

Data type of output
Specify the output data type as one of double | single. The default is double.

Methods clone<br>constellation<br>Create PSK Modulator object with same property values<br>Calculate or plot ideal signal constellation

| isLocked | Locked status for input attributes <br> and nontunable properties |
| :--- | :--- |
| release | Allow property value and input <br> characteristics changes |
| step | Modulate using M-ary PSK <br> method with GPU |

## Algorithm

## Examples

The GPU PSK Modulator System object supports floating-point and integer input data types. This object uses the same algorithm as the comm. PSKModulator System object. See the Algorithms section of the comm. PSKModulator help page for details.

Modulate data using 16-PSK modulation and then visualize the data using a scatter plot.

```
% Create binary data for 24, 4-bit symbols
data = randi([0 1],96,1);
% Create a 16-PSK modulator System object with bits as inputs
% and Gray-coded signal constellation
hModulator = comm.gpu.PSKModulator(16,'BitInput',true);
% Change the phase offset to pi/16
hModulator.PhaseOffset = pi/16;
% Modulate and plot the data
modData = step(hModulator, data);
scatterplot(modData)
```

See Also comm.PSKDemodulator

## comm.gpu.PSKModulator.clone

Purpose Create PSK Modulator object with same property values

## Syntax

Description
C = clone(H) creates a GPU PSK Modulator object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

| Purpose | Calculate or plot ideal signal constellation |
| :---: | :---: |
| Syntax | $\begin{aligned} & y=\text { constellation(h) } \\ & \text { constellation(h) } \end{aligned}$ |
| Description | $y=$ constellation(h) returns the numerical values of the constellation. <br> constellation(h) generates a constellation plot for the object. |
| Examples | Calculate Ideal Signal Constellation for comm.gpu.PSKModulator |
|  | Create a comm.gpu.PSKModulator System object, and then calculate its ideal signal constellation. |
|  | Create a comm.gpu.PSKModulator System object by entering the following at the MATLAB command line: |
|  | $\mathrm{h}=$ comm.gpu.PSKModulator |
|  | Calculate and display the ideal signal constellation by calling the constellation method. |
|  | $\mathrm{a}=$ constellation(h) |
|  | Plot Ideal Signal Constellation for comm.gpu.PSKModulator |
|  | Create a comm.gpu.PSKModulator System object, and then plot the ideal signal constellation. |
|  | Create a comm.gpu.PSKModulator System object by entering the following at the MATLAB command line: |
|  | $\mathrm{h}=$ comm.gpu.PSKModulator |
|  | Plot the ideal signal constellation by calling the constellation method. constellation(h) |

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked(H) returns the locked status, TF of the GPU PSK Modulator System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Modulate using M-ary PSK method with GPU

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description $\quad Y=\operatorname{step}(H, X)$ modulates the input data, $X$, using the GPU-based PSK modulator System object, H. The object returns the baseband modulated output Y. Depending upon the value of the BitInput property, input $X$ can be an integer or bit-valued column vector with numeric or logical data types.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose <br> Description

Decode input signal using parallel concatenation decoding with GPU

The GPU Turbo Decoder System object decodes the input signal using a parallel concatenated decoding scheme. This scheme uses the a-posteriori probability (APP) decoder as the constituent decoder. Both constituent decoders use the same trellis structure and algorithm.

Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.


## Construction

H = comm.gpu.TurboDecoder creates a GPU-based turbo decoder System object, H. This object uses the a-posteriori probability (APP) constituent decoder to iteratively decode the parallel-concatenated convolutionally encoded input data.

H = comm.gpu.TurboDecoder (Name, Value) creates a GPU-based turbo decoder object, H , with the specified property name set to the specified value. Name must appear inside single quotes (' '). You can specify several name-value pair arguments in any order as Name1, Value1, ,NameN, ValueN.

H = comm.gpu.TurboDecoder(TRELLIS, INTERLVRINDICES, NUMITER) creates a GPU-based turbo decoder object, H. In this object, the TrellisStructure property is set to TRELLIS, the InterleaverIndices property set to INTERLVRINDICES, and the NumIterations property set to NUMITER.

## Properties

## TrellisStructure

Trellis structure of constituent convolutional code
Specify the trellis as a MATLAB structure that contains the trellis description of the constituent convolutional code. The default is the result of poly2trellis (4, [13 15], 13). Use the istrellis function to check if a structure is a valid trellis structure.

## InterleaverIndicesSource

Source of interleaver indices
Specify the source of the interleaver indices. The only valid setting for this property is Property.

## InterleaverIndices

Interleaver indices
Specify the mapping used to permute the input bits at the encoder as a column vector of integers. The default is (64:-1:1).' .. This mapping is a vector with the number of elements equal to the length, $L$, of the output of the step method. Each element must be an integer between 1 and $L$, with no repeated values.

## Algorithm

Decoding algorithm

Specify the decoding algorithm. This object implements true $a$ posteriori probability decoding. The only valid setting is True APP.

## NumScalingBits

Number of scaling bits
The GPU version of the Turbo Decoder does not use this property.

## Numlterations

Number of decoding iterations
Specify the number of decoding iterations used for each call to the step method. The default is 6 . The object iterates and provides updates to the log-likelihood ratios (LLR) of the uncoded output bits. The output of the step method is the hard-decision output of the final LLR update.

## NumFrames

Number of independent frames present in the input and output data vectors.

Specify the number of independent frames that a single data input/output vector contains. The default value of this property is 1. This object segments the input vector into NumFrames segments and decodes the segments independently. The output contains NumFrames decoded segments.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked

Create Turbo Decoder object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

release<br>reset<br>step

Allow property value and input characteristics changes
Reset states of the turbo decoder object

Decode input signal using parallel concatenated decoding scheme

## Examples Transmit and decode using turbo coding

Transmit turbo-encoded blocks of data over a BPSK-modulated AWGN channel. Then, decode using an iterative turbo decoder and display errors.

Define a noise variable, establish a frame length of 256 , and use the random stream property so that the results are repeatable.

```
noiseVar = 4; frmLen = 256;
s = RandStream('mt19937ar', 'Seed', 11);
intrlvrIndices = randperm(s, frmLen);
```

Create a Turbo Encoder System object. The trellis structure for the constituent convolutional code is poly2trellis(4, [13 15 17], 13). The InterleaverIndices property specifies the mapping the object uses to permute the input bits at the encoder as a column vector of integers.

```
hTEnc = comm.TurboEncoder('TrellisStructure', poly2trellis(4, ...
    [13 15 17], 13), 'InterleaverIndices', intrlvrIndices);
```

Create a BPSK Modulator System object.
hMod = comm.BPSKModulator;
Create an AWGN Channel System object.

```
hChan = comm.AWGNChannel('NoiseMethod', 'Variance', 'Variance', ...
    noiseVar);
```

Create a GPU-Based Turbo Decoder System object. The trellis structure for the constituent convolutional code is poly2trellis(4, [13 15 17], 13). The InterleaverIndicies property specifies the mapping the object uses to permute the input bits at the encoder as a column vector of integers.

```
hTDec = comm.gpu.TurboDecoder('TrellisStructure', poly2trellis(4, ...
    [13 15 17], 13), 'InterleaverIndices', intrlvrIndices, ...
    'NumIterations', 4);
Create an Error Rate System object.
```

```
hError = comm.ErrorRate;
```

```
hError = comm.ErrorRate;
```

Run the simulation by using the step method to process data.

```
for frmIdx = 1:8
    data = randi(s, [0 1], frmLen, 1);
    encodedData = step(hTEnc, data);
    modSignal = step(hMod, encodedData);
    receivedSignal = step(hChan, modSignal);
```

Convert the received signal to log-likelihood ratios for decoding.

```
receivedBits = step(hTDec, (-2/(noiseVar/2))*real(receivedSignal));
```

Compare original the data to the received data and then calculate the error rate results.

```
errorStats = step(hError, data, receivedBits);
end
fprintf('Error rate = %f\nNumber of errors = %d\nTotal bits = %d\n',
errorStats(1), errorStats(2), errorStats(3))
```


## Algorithms

This object implements the inputs and outputs described on the Turbo Decoder block reference page. The object properties correspond to the block parameters.

See Also comm.TurboEncoder | comm.TurboDecoder

## Purpose Create Turbo Decoder object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a GPU Turbo Decoder object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.
The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

The getNumInputs method returns a positive integer that is the number of expected inputs (not counting the object itself) to the step method. This value will change if you alter any properties that turn inputs on or off. You must call the step method with the number of input arguments equal to the result of getNumInputs $(H)$.

## comm.gpu.TurboDecoder.getNumOutputs

## Purpose Number of outputs from step method

Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.
The getNumOutputs method returns a positive integer that is the number of outputs from the step method. This value will change if you alter any properties that turn outputs on or off.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description Description
TF = isLocked(H) returns the locked status, TF of the TurboDecoder System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose
Syntax release (H)
Description

Allow property value and input characteristics changes
release (H) release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of the turbo decoder object

## Syntax reset (H)

Description reset (H) resets the states of the GPU TurboDecoder object, H.

## Purpose

Decode input signal using parallel concatenated decoding scheme

## Syntax

Description
$Y=\operatorname{step}(H, X)$
$Y=\operatorname{step}(H, X)$ decodes the input data, $X$, using the parallel
concatenated convolutional coding scheme. You specify this scheme using the TrellisStructure and InterleaverIndices properties. It returns the binary decoded data, $Y$. Both $X$ and $Y$ are column vectors of double-precision data type. When the constituent convolutional code represents a rate $1 / \mathrm{N}$ code, the step method sets the length of the output vector, Y , to ( $\mathrm{M}-2 *$ numTails)/( $2 * \mathrm{~N}-1$ ). M represents the input vector length and numTails is given by log2(TrellisStructure.numStates)*N. The output length, $L$, is the same as the length of the interleaver indices.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Decode convolutionally encoded data using Viterbi algorithm with GPU

The GPU ViterbiDecoder System object decodes input symbols to produce binary output symbols using a graphics processing unit (GPU). This object processes variable-size signals; however, variable-size signals cannot be applied for erasure inputs.

Note To use this object, you must install a Parallel Computing Toolbox license and have access to an appropriate GPU. For more about GPUs, see "GPU Computing" in the Parallel Computing Toolbox documentation.

A GPU-based System object accepts typical MATLAB arrays or objects that you create using the gpuArray class as an input to the step method. GPU-based System objects support input signals with doubleor single-precision data types. The output signal inherits its datatype from the input signal.

- If the input signal is a MATLAB array, then the output signal is also a MATLAB array. In this case, the System object handles data transfer between the CPU and GPU.
- If the input signal is a gpuArray, then the output signal is also a gpuArray. In this case, the data remains on the GPU. Therefore, when the object is given a gpuArray, calculations take place entirely on the GPU and no data transfer occurs. Invoking the step method with gpuArray arguments provides increased performance by reducing simulation time. For more information, see "Use gpuArray Data" in the Parallel Computing Toolbox documentation.


## Construction

H = comm.gpu.ViterbiDecoder creates a Viterbi decoder System object, H. This object uses the Viterbi algorithm to decode convolutionally encoded input data.
H = comm.gpu.ViterbiDecoder (Name, Value) creates a Viterbi decoder object, H, with the specified property Name set to the specified Value.

You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN.

H = comm.gpu.ViterbiDecoder(TRELLIS,Name, Value) creates a Viterbi decoder object, H, with the TrellisStructure property set to TRELLIS, and other specified property Names set to the specified Values.

## Properties

TrellisStructure
Trellis structure of convolutional code
Specify the trellis as a MATLAB structure that contains the trellis description of the convolutional code. Use the istrellis function to check if a structure is a valid trellis structure. This object supports rate $1 / 2,1 / 3$ and $1 / 4$ trellises from simple feedforward encoders. The default value is the result of poly2trellis(7, [171 133]).

## InputFormat

Input format
Specify the format of the input to the decoder as one of Unquantized | Hard \| Soft. The default is Unquantized.
When you set this property to Unquantized, the input must be a real vector of double or single precision unquantized soft values. The object considers negative numbers to be ones and positive numbers to be zeros. When you set this property to Hard, the input must be a vector of hard decision values, which are zeros or ones. The data type of the inputs can be double precision or single precision. When you set this property to Soft, the input must be a vector of quantized soft values represented as integers between 0 and $2^{\wedge}$ SoftInputWordLength- 1 . The data type of the inputs can be double precision or single precision.

## SoftInputWordLength

Soft input word length

Specify the number of bits used to represent each quantized soft input value as a positive, integer scalar. This property applies when you set the InputFormat property to Soft. The default is 4 bits.

## InvalidQuantizedInputAction

Action when input values are out of range
The only valid setting is Ignore which ignores out of range inputs.

## TracebackDepth

Traceback depth
Specify the number of trellis branches used to construct each traceback path as a positive, integer scalar less than or equal to 256. The traceback depth influences the decoding accuracy and delay. The number of zero symbols that precede the first decoded symbol in the output represent a decoding delay. When you set the TerminationMethod property to Continuous, the decoding delay consists of TracebackDepth zero symbols, or TracebackDepth zero bits for a rate $1 / \mathrm{N}$ convolutional code. When you set the TerminationMethod property to Truncated or Terminated, there is no output delay and TracebackDepth must be less than or equal to the number of symbols in each input. If the code rate is $1 / 2$, a typical traceback depth value is about five times the constraint length of the code. The default is 34 .

## TerminationMethod

Termination method of encoded frame
Specify TerminationMethod as one of Continuous | Truncated | Terminated. The default is Continuous. In Continuous mode, the object saves its internal state metric at the end of each frame for use with the next frame. The object treats each traceback path independently. Select Continuous mode when the input signal contains only one symbol. In Truncated mode, the object treats each frame independently. The traceback path starts at the state with the best metric and always ends in the all-zeros state. In

Terminated mode, the object treats each frame independently, and the traceback path always starts and ends in the all-zeros state.

## ResetInputPort

Enable decoder reset input
Set this property to true to enable an additional step method input. When the reset input is a non-zero value, the object resets the internal states of the decoder to initial conditions. This property applies when you set the TerminationMethod property to Continuous. The default is false.

## DelayedResetAction

Delay output reset
Delaying the output reset is not supported. The only valid setting is false.

## PuncturePatternSource

Source of puncture pattern
Specify the source of the puncture pattern as one of None | Property. The default is None. When you set this property to None the object assumes no puncturing. Set this property to Property to decode punctured codewords based on a puncture pattern vector specified via the PuncturePattern property.

## PuncturePattern

Puncture pattern vector
Specify puncture pattern used to puncture the encoded data. The default is $[1 ; 1 ; 0 ; 1 ; 0 ; 1]$. The puncture pattern is a column vector of ones and zeros, where the zeros indicate where to insert dummy bits. The puncture pattern must match the puncture pattern used by the encoder. This property applies when you set the PuncturePatternSource property to Property.

## ErasuresInputPort

Enable erasures input
Erasures are not supported. The only valid setting is false.

## OutputDataType

Data type of output
The only valid setting is Full precision which makes the output data type match the input data type.

## NumFrames

Number of independent frames present in the input and output data vectors.

Specify the number of independent frames contained in a single data input/output vector. The input vector will be segmented into NumFrames segments and decoded independently. The output will contain NumFrames decoded segments. The default value of this property is 1 . This property is applies when you set the TerminationMethod is set to Terminated or Truncated.

| Methods | clone | Create Viterbi Decoder object <br> with same property values |
| :---: | :---: | :--- |
| info | Display information about <br> GPU-based Viterbi Decoder <br> object |  |
| isLocked | release | Locked status for input attributes <br> and nontunable properties |
| reset | Allow property value and input <br> characteristics changes |  |
| step | Reset states of the GPU-based <br> Viterbi Decoder modulator object <br> Decode convolutionally encoded <br> data using Viterbi algorithm |  |

```
Examples Transmit a convolutionally encoded 8-DPSK-modulated bit stream
through an AWGN channel. Then, demodulate, decode using a Viterbi
decoder, and count errors.
hConEnc = comm.ConvolutionalEncoder;
hMod = comm.DPSKModulator('BitInput',true);
hChan = comm.gpu.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)', 'SNR',10);
hDemod = comm.DPSKDemodulator('BitOutput',true);
hDec = comm.gpu.ViterbiDecoder('InputFormat','Hard');
\% Delay in bits is TracebackDepth times the number of
\% bits per symbol
    delay \(=\) hDec.TracebackDepth*...
    log2(hDec.TrellisStructure.numInputSymbols);
hError = comm.ErrorRate('ComputationDelay',3,'ReceiveDelay', delay);
    for counter = 1:20
        data = randi([0 1],30,1);
        encodedData = step(hConEnc, data);
        modSignal = step(hMod, encodedData);
        receivedSignal = step(hChan, modSignal);
        demodSignal = step(hDemod, receivedSignal);
        receivedBits = step(hDec, demodSignal);
        errorStats = step(hError, data, receivedBits);
        end
fprintf('Error rate \(=\% f \backslash n N u m b e r ~ o f ~ e r r o r s ~=~ \% d \backslash n ', ~ . . . ~\)
errorStats(1), errorStats(2))
```


## References [1] Fettweis, G., H. Meyr. "Feedforward Architecture for Parallel

``` Viterbi Decoding," Journal of VLSI Signal Processing, Vol. 3, June 1991.
See Also comm.ViterbiDecoder
```


## comm.gpu.ViterbiDecoder.clone

Purpose Create Viterbi Decoder object with same property values

## Syntax <br> C = clone( H )

Description $\quad$ C $=$ clone $(H)$ creates a GPU Viterbi Decoder object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Display information about GPU-based Viterbi Decoder object

## Syntax

Description
$S=i n f o(0 B J)$
$S=$ info(OBJ) returns a structure, $S$, containing characteristic
information for the System object, OBJ. If OBJ has no characteristic information, S is empty. If OBJ has characteristic information, the fields of S vary depending on OBJ. For object specific details, refer to the help on the infoImpl method of that object.

## comm.gpu.ViterbiDecoder.isLocked

Purpose Locked status for input attributes and nontunable properties

## Syntax TF = isLocked (H)

Description TF = isLocked (H) returns the locked status, TF of the ACPR System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose $\quad$ Reset states of the GPU-based Viterbi Decoder modulator object

## Syntax reset (H)

Description reset (H) resets the states of the GPU-based ViterbiDecoder object, H.

## Purpose

Decode convolutionally encoded data using Viterbi algorithm
Syntax
$Y=\operatorname{step}(H, X)$
Y $=\operatorname{step}(H, X, R)$
$Y=\operatorname{step}(H, X)$ decodes encoded data, $X$, using the Viterbi algorithm and returns Y. X, must be a column vector with data type and values that depend on how you set the InputFormat property. If the convolutional code uses an alphabet of $2^{\wedge} \mathrm{N}$ possible symbols, the length of the input vector, X , must be $\mathrm{L} * \mathrm{~N}$ for some positive integer L. Similarly, if the decoded data uses an alphabet of $2^{\wedge} \mathrm{K}$ possible output symbols, the length of the output vector, Y , is $\mathrm{L} * \mathrm{~K}$.
$Y=\operatorname{step}(H, X, R)$ resets the internal states of the decoder when you input a non-zero reset signal, R. R must be a double precision, single precision or logical scalar. This syntax applies when you set the TerminationMethod property to Continuous and the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Generate Hadamard code

Generate Hadamard code
Construction
Properties

The HadamardCode object generates a Hadamard code from a Hadamard matrix, whose rows form an orthogonal set of codes. You can use orthogonal codes for spreading in communication systems in which the receiver is perfectly synchronized with the transmitter. In these systems, the despreading operation is ideal, because the codes decorrelate completely.

H = comm.HadamardCode creates a Hadamard code generator System object, H. This object generates Hadamard codes from a set of orthogonal codes.

H = comm. HadamardCode (Name, Value) creates a Hadamard code generator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Length

Length of generated code
Specify the length of the generated code as a numeric, integer scalar value with a power of two. The default is 64 .

## Index

Row index of Hadamard matrix
Specify the row index of the Hadamard matrix as a numeric, integer scalar value in the range [0, 1, ... , N-1]. $N$ is the value of the Length property. The default is 60 . An $N \times N$ Hadamard matrix, denoted as $P(N)$, is defined recursively as follows: $P(1)=[1] P(2 N)=[P(N) P(N) ; P(N)-P(N)]$ The $N \mathrm{x} N$ Hadamard matrix has the property that $P(N) \times P(N)^{\prime}=N \times$ eye $(N)$. The step method outputs code samples from the row of the Hadamard matrix that you specify in this property.

When you set this property to an integer $k$, the output code has exactly $k$ zero crossings, for $k=0,1, \ldots, N-1$.

## SamplesPerFrame

Number of output samples per frame
Specify the number of Hadamard code samples that the step method outputs as a numeric, positive, integer scalar value. The default is 1 .

When you set this property to a value of $M$, the step method outputs $M$ samples of a Hadamard code of length $N$. $N$ equals the length of the code that you specify in the Length property.

## OutputDataType

Data type of output
Specify the output data type as one of double | int8. The default is double.

| Methods | clone | Create Hadamard code generator <br> object with same property values |
| :--- | :--- | :--- |
| getNumInputs | getNumOutputs | Number of expected inputs to <br> step method |
| isLocked | Number of outputs from step <br> method |  |
| release | Locked status for input attributes <br> and nontunable properties |  |
| reset | Allow property value and input <br> characteristics changes |  |
| step | Reset states of Hadamard code <br> generator object <br> Generate Hadamard code |  |

## Examples

Generate 10 samples of a Hadamard code sequence with a length of 64 .

```
hHCode = comm.HadamardCode('SamplesPerFrame', 10);
```

```
seq = step(hHCode)
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the Hadamard Code Generator block reference page. The object properties correspond to the block parameters, except:

- The object does not have a property to select frame based outputs.
- The object does not have a property that corresponds to the Sample time parameter.


## See Also

comm.WalshCode | comm.OVSFCode

Purpose
Create Hadamard code generator object with same property values

## Syntax <br> C = clone(H)

Description

C = clone (H) creates a HadamardCode object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.HadamardCode.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.HadamardCode.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the HadamardCode System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release (H) |
| Description | release (H)Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

# Purpose Reset states of Hadamard code generator object 

## Syntax reset (H)

Description reset (H) resets the states of the HadamardCode object, H.

Purpose
Generate Hadamard code

## Syntax <br> Y = step(H)

$Y=$ step $(H)$ outputs a frame of the Hadamard code in column vector Y. Specify the frame length with the SamplesPerFrame property. The Hadamard code corresponds to one of the rows of an $N \mathrm{x} N$ Hadamard matrix, where $N$ is a nonnegative power of 2 , which you specify in the Length property. Use the Index property to choose the row of the Hadamard matrix. The step method outputs the code in a bi-polar format with 0 and 1 mapped to 1 and -1 , respectively.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Detect errors in input data using HDL-optimized CRC
Description
Construction
This hardware-friendly cyclic redundancy code (CRC) detector System object computes checksums for its entire input frame. The HDLCRCDetector System object is optimized for HDL code generation. Instead of frame processing, the System object processes data at the streaming mode. Control signals are added at both input and output for easy data synchronization.
H = comm.HDLCRCDetector creates an HDL-optimized CRC detector System object, H, that detects errors in the input data according to a specified generator polynomial.
H = comm. HDLCRCDetector(Name, Value, ) creates an HDL-optimized CRC detector System object, H, with additional options specified by one or more Name, Value pair arguments, where Name is a property name and Value is the corresponding value. Name must appear inside single quotes (' '). You can specify several name-value pair arguments in any order as Name1, Value1, ..., NameN, ValueN.
H = comm.HDLCRCDetector(POLY,Name, Value) creates an HDL-optimized CRC detector System object, H, with the Polynomial property set to POLY, and the other specified properties set to the specified values.

## Input Arguments

## POLY

Sets Polynomial property to POLY at System object construction

## Properties

## Polynomial

Specify the generator polynomial as a binary row vector, with coefficients in descending order of powers. If you set this property to a binary vector, its length must be equal to the degree of the polynomial plus 1. The default is $[10001000000100001$ ].

## FinalXORValue

The value with which the CRC checksum is to be XORed just prior to being appended to the input data. This property can be specified as a binary, double or single precision data type scalar or vector. The vector length is the degree of the generator polynomial that you specify in the Polynomial property. When you specify Final XOR Value as a scalar, the object expands the value to a row vector of length equal to the degree of the generator polynomial. The default is 0 .

## InitialState

Specify the initial conditions of the shift register as a binary, double or single precision data type scalar or vector. The vector length is the degree of the generator polynomial that you specify in the Polynomial property. When you specify initial conditions as a scalar, the object expands the value to a row vector of length equal to the degree of the generator polynomial. The default is 0 .

## ReflectCRCChecksum

A logical quantity that specifies whether the output CRC checksum should be flipped around its center after the input data is completely through the shift register. The default is false.

## ReflectInput

A logical quantity that specifies whether the input data should be flipped on a bytewise basis prior to entering the shift register. The default is false.

Methods clone<br>isLocked<br>release

Create HDLCRCDetector System object with same property values
Locked status for input attributes and nontunable properties

Allow property value and input characteristics change

| reset | Reset states of HDL CRC detector <br> object |
| :--- | :--- |
| step | Generate CRC checksums for <br> input message based on control <br> signals and appends checksums <br> to output message |
|  |  |

## Examples

Encode and decode a signal using an HDL-optimized CRC generator and detector.

## 1

Construct default polynomial with CRC length 16:
hGen = comm.HDLCRCGenerator;
hDet $=$ comm. HDLCRCDetector;
2
Run HDLCRCGenerator 10 steps:
numSteps = 10;

3
Assign control signals for all 10 steps:
startIn = logical([1 0 0 0 0 0 0 0 0 0]);
endIn = logical([0 1 0 0 0 0 0 0 0 0]);
validIn = logical([1 1 0 0 0 0 0 0 0 0]);

4
Assign 32 bit data to be encoded, in two 16 by 1 columns:
msg = randi([0 1],16,2);
5

Assign random input to the HDLCRCGenerator System object while it is processing msg:
randIn = randi([0, 1],16,numSteps-2);
dataIn = [msg randIn];

6
Run HDLCRCGenerator System object 10 steps:
\% Output data: dataOutGen
\% Output Control signals: startOutGen, endOutGen, validOutGen
for i = 1: numSteps
[dataOutGen(:,i),startOutGen(i),endOutGen(i), validOutGen(i)] = step(l dataIn(:,i), startIn(i), endIn(i), validIn(i)
end

7
Add noise to encoded message:
dataOutGen(2,4) = ~dataOutGen(2,4);
8
Run HDLCRCDetector System object 10 steps:
\% Output data: dataOut
\% Output Control signals: startOut, endOut, validOut,err
for i = 1:numSteps
[dataOut(:,i), startOut(i),endOut(i), validOut(i),err(i)] = step(hDet dataOutGen(:,i), startOutGen(i), endOutGen(i), validOutGen(i) end

Algorithms
Timing diagram for HDL-optimized CRC Detector


## Initial Delay

The HDLCRCGenerator System object introduces a latency on the output. This latency can be computed with the following equation:
initialdelay = 3 * CRC length/input data width + 2
See Also comm. HDLCRCGenerator | comm.CRCDetector |
Purpose Create HDLCRCDetector System object with same property values
Syntax C = clone(H)
Description $\mathrm{C}=\mathrm{clone}(\mathrm{H})$ creates another instance of the HDLCRCDetectorSystem object, H, with the same property values. The clone methodcreates a new unlocked object with uninitialized states.
Input
Arguments
Output ..... C
Arguments
See Alsocomm.HDLCRCDetector | comm.HDLCRCDetector.isLocked |comm.HDLCRCDetector.release | comm.HDLCRCDetector.reset |comm. HDLCRCDetector.step |
Purpose Locked status for input attributes and nontunable properties
SyntaxDescription$L=$ isLocked $(H)$ returns the locked status, $L$, of the HDL CRCDetector System object, H.
The isLocked method returns a logical value that indicates whetherinput attributes and nontunable properties for the object are locked. Theobject performs an internal initialization the first time the step methodis executed. This initialization locks nontunable properties and inputspecifications, such as dimensions, complexity, and data type of theinput data. After locking, the isLocked method returns a true value.
Input
Arguments
H
HDL CRC Detector System object
OutputArguments
See Also comm.HDLCRCDetector | comm.HDLCRCDetector.clone | comm.HDLCRCDetector.release | comm.HDLCRCDetector.reset | comm. HDLCRCGenerator.step |

# Purpose <br> Allow property value and input characteristics change <br> Syntax release (H) 

Description
release (H) releases system resources (such as memory, file handles or hardware connections) of the HDL CRC Detector System object, H, and allows all its properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Input <br> Arguments

See Also comm. HDLCRCDetector I comm. $\mathrm{HDLCRCDetector.clone} \mathrm{I} \begin{aligned} & \text { comm.HDLCRCDetector.isLocked | comm.HDLCRCDetector.reset I } \\ & \text { comm.HDLCRCDetector.step | }\end{aligned}$

Purpose Reset states of HDL CRC detector object

## Syntax reset (H)

Description reset (H) resets the internal states of the HDL CRC Detector System object, H , to their initial values

## Input <br> Arguments

## H

Instance of HDL CRC Detector System object

See Also<br>comm.HDLCRCDetector | comm.HDLCRCDetector.clone | comm.HDLCRCDetector.isLocked | comm.HDLCRCDetector.release | comm.HDLCRCDetector.step |

## Purpose

Syntax

## Description

## Input Arguments

Generate CRC checksums for input message based on control signals and appends checksums to output message

```
[Y,startOut,endOut,validOut,err] = step(H,X,startIn,endIn,
    validIn)
```

[Y,startOut, endOut, validOut,err] = step( $\mathrm{H}, \mathrm{X}$, startIn, endIn, validIn) computes CRC checksums for an input message $X$ based on the control signals and compares the computed checksum with input checksum. The output err is high if the two checksums are not equal.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## H

HDLCRCDetector System object

## X

Input message

- Must be a binary column vector.
- Data type can be double or logical.
- X can be part or all of the message to be encoded.
- The length of $X$ should be less than or equal to the CRC length, and the CRC length should be divisible by the length of $X$.
- The CRC length is the order of the polynomial that you specify in the Polynomial property.


## startln

Indicates the start of an input message. startIn is scalar with logical data type.
endln
Indicates the end of an input message. endIn is scalar with logical data type.

## validln

When validIn is high, input message is processed for CRC checksum computation. validIn is scalar with logical data type.

## Output $\quad \mathbf{Y}$

comm.HDLCRCDetector | comm.HDLCRCDetector.clone | comm.HDLCRCDetector.isLocked | comm.HDLCRCDetector.release | comm. HDLCRCDetector.reset I

## Purpose

Description

## Construction

Generate HDL-optimized CRC code bits and append to input data

This hardware-friendly CRC Generator System object, like the CRC Generator System object, generates cyclic redundancy code (CRC) bits. However, the HDL CRC Generator System object is optimized for HDL code generation. Instead of frame processing, the System object processes data at the streaming mode. Control signals are added at both input and output for easy data synchronization.

H=comm. HDLCRCGenerator creates an HDL-optimized cyclic redundancy code (CRC) generator System object, H. This object generates CRC bits according to a specified generator polynomial and appends them to the input data.

H = comm.HDLCRCGenerator(Name, Value) creates an HDL-optimized CRC generator System object, H, with additional options specified by one or more Name, Value pair arguments, where Name is a property name and Value is the corresponding value. Name must appear inside single quotes (' '). You can specify several name-value pair arguments in any order as Name1, Value1, ..., NameN, ValueN.

H = comm.HDLCRCGenerator(POLY,Name, Value) creates an HDL-optimized CRC generator System object, H, with the Polynomial property set to POLY, and the other specified properties set to the specified values.

## Input Arguments

## POLY

Sets Polynomial property to POLY at System object construction

## Properties

## Polynomial

Specify the generator polynomial as a binary row vector, with coefficients in descending order of powers. If you set this property to a binary vector, its length must be equal to the degree of the polynomial plus 1. The default is $[10001000000100001$ ].

## FinalXORValue

The value with which the CRC checksum is to be XORed just prior to being appended to the input data. This property can be specified as a binary, double or single precision data type scalar or vector. The vector length is the degree of the generator polynomial that you specify in the Polynomial property. When you specify Final XOR Value as a scalar, the object expands the value to a row vector of length equal to the degree of the generator polynomial. The default is 0 .

## InitialState

Specify the initial conditions of the shift register as a binary, double or single precision data type scalar or vector. The vector length is the degree of the generator polynomial that you specify in the Polynomial property. When you specify initial conditions as a scalar, the object expands the value to a row vector of length equal to the degree of the generator polynomial. The default is 0 .

## ReflectCRCChecksum

A logical quantity that specifies whether the output CRC checksum should be flipped around its center after the input data is completely through the shift register. The default is false.

## ReflectInput

A logical quantity that specifies whether the input data should be flipped on a bytewise basis prior to entering the shift register. The default is false.

Methods clone<br>isLocked<br>Create HDLCRCGenerator System object with same property values<br>Locked status for input attributes and nontunable properties

| release | Allow property value and input <br> characteristics change |
| :--- | :--- |
| reset | Reset states of CRC generator <br> object |
| step | Generate CRC checksums for <br> input message based on control <br> signals and appends checksums <br> to output message |

## Examples Encode signal using an HDL-optimized CRC generator.

```
% Using default polynomial with CRC length 16
hGen = comm.HDLCRCGenerator;
% run HDL CRC Generator 6 steps
numSteps = 6;
% Control signals for all 6 steps
startIn = logical([1 0 0 0 0 0]);
endIn = logical([0 1 0 0 0 0]);
validIn = logical([11 1 0 0 0 0]);
% 32 bit data to be encoded, in two 16 by 1 columns
msg = randi([0 1],16,2);
% random input to HDLCRCGenerator while it is processing the msg
randIn = randi([0, 1],16,numSteps-2);
dataIn = [msg randIn];
% Run HDL CRC Generator 6 steps
% Output data: dataOut
% Output Control signals: startOut, endOut, validOut
for i = 1: numSteps
[dataOut(:,i), startOut(i),endOut(i), validOut(i)] = step(hGen,...
    dataIn(:,i),startIn(i),endIn(i),validIn(i));
end
```


## Algorithms



## Initial Delay

The HDL CRC Generator System object introduces a latency on the output. This latency can be computed with the following equation:
initialdelay = (CRC length/input data width) + 2

## See Also

Purpose Create HDLCRCGenerator System object with same property values
Syntax C = clone(H)
Description C = clone (H) creates another instance of the HDLCRCGeneratorSystem object, H, with the same property values. The clone methodcreates a new unlocked object with uninitialized states.
Input
Arguments
Output ..... C
Arguments
See Alsocomm.HDLCRCGenerator | comm.HDLCRCGenerator.isLocked |comm.HDLCRCGenerator.release | comm.HDLCRCGenerator.reset |comm. HDLCRCGenerator.step |

[^6]
## Purpose

Allow property value and input characteristics change

## Syntax

Description

```
release(H)
```

release (H) releases system resources (such as memory, file handles or hardware connections) of the HDL CRC Generator System object, H, and allows all its properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Input <br> Arguments <br> H

See Also comm. HDLCRCGenerator I comm. HDLCRCGenerator.clone | $\begin{aligned} & \text { comm.HDLCRCGenerator.isLocked I comm. HDLCRCGenerator.reset | } \\ & \text { comm.HDLCRCGenerator.step | }\end{aligned}$
Instance of HDL CRC Generator System object

Purpose Reset states of CRC generator object

## Syntax <br> reset(H)

Description reset (H) resets the internal states of the HDL CRC Generator System object, H , to their initial values

## Input

Arguments

## H

Instance of HDL CRC Generator System object

See Also<br>comm.HDLCRCGenerator | comm.HDLCRCGenerator.clone | comm.HDLCRCGenerator.isLocked | comm.HDLCRCGenerator.release | comm.HDLCRCGenerator.step |

## Purpose

Syntax

## Description

## Input Arguments

Generate CRC checksums for input message based on control signals and appends checksums to output message

```
[Y,startOut,endOut,validOut] = step(H,X,startIn,endIn, validIn)
```

[Y,startOut,endOut, validOut] = step(H,X,startIn,endIn, validIn) generates CRC checksums for input message $X$ based on control signals and appends the checksums to $X$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## H

HDL CRC Generator System object

## X

## Input message

- Must be a binary column vector.
- Data type can be double or logical.
- X can be part or all of the message to be encoded.
- The length of $X$ should be less than or equal to the CRC length, and the CRC length should be divisible by the length of $X$.
- The CRC length is the order of the polynomial that you specify in the Polynomial property.


## startln

Indicates the start of an input message. startIn is scalar with logical data type.

## endln

Indicates the end of an input message. endIn is scalar with logical data type.

## validIn

When validIn is high, input message is processed for CRC checksum computation. validIn is scalar with logical data type.

## Output Arguments

See Also
comm.HDLCRCGenerator | comm.HDLCRCGenerator.clone | comm.HDLCRCGenerator.isLocked | comm.HDLCRCGenerator.release | comm.HDLCRCGenerator.reset |

## Purpose Decode data using a Reed-Solomon decoder

Description

Construction

The HDL-optimized HDLRSDecoder System object recovers a message vector from a Reed-Solomon codeword vector. For proper decoding, the property values for this object should match those in the corresponding HDLRSEncoder System object.

H = comm. HDLRSDecoder creates an HDL-optimized RS decoder System object, H, that performs Reed-Solomon (RS) decoding.

H = comm.HDLRSDecoder (Name, Value) creates an HDL-optimized RS decoder System object, H, with additional options specified by one or more Name, Value pair arguments, where Name is a property name and Value is the corresponding value. Name must appear inside single quotes (' '). You can specify several name-value pair arguments in any order as Name1, Value1, ... , NameN, ValueN.

H = comm. HDLRSDecoder ( $\mathrm{N}, \mathrm{K}$, Name, Value) creates an HDL-optimized RS decoder System object, H, with the CodewordLength property set to N , the MessageLength property set to K, and other specified property Names set to the specified Values.

## Properties

B
$B$ value for polynomial generation

## BSource

Source of B, the starting power for roots of the primitive polynomial
Specify the source of the B value as one of these values:

- Auto: $\mathrm{B}=0$
- Property

Default: Auto

## CodewordLength

Codeword length
Specify the codeword length of the RS code as a double-precision, positive, integer scalar value. The default is 7 .

If you set the PrimitivePolynomialSource property to Auto, CodewordLength must be in the range $3<$ CodewordLength $\leq$ $2^{16}-1$.

When you set the PrimitivePolynomialSource property to Property, CodewordLength must be in the range $3 \leq$ CodewordLength $\leq 2^{M}-1 . M$ is the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties. $M$ must be in the range $3 \leq M \leq 16$. The difference (CodewordLength -MessageLength) must be an even integer. The value of this property is rounded up to $2^{M}-1$.

If the value of this property is less than $2^{M}-1$, the object assumes a shortened RS code.

## MessageLength

Message length
Specify the message length as a double-precision, positive integer scalar value. The default is 3 . The difference (CodewordLength MessageLength) must be an even integer.

## NumErrorsOutputPort

Enable number of errors output
When you set this property to true, the step method outputs number of corrected errors. The number of corrected errors is not valid when errOut is asserted, since there were more errors than could be corrected. The default is false.

## PrimitivePolynomialSource

Source of primitive polynomial

Specify the source of the primitive polynomial as Auto | Property. The default is Auto.

When you set this property to Auto, the object uses a primitive polynomial of degree $M=\operatorname{ceil}(\log 2(C o d e w o r d L e n g t h+1))$, which is the result of fliplr(de2bi(primpoly $(M))$ ).

When you set this property to Property, you can specify a polynomial using the PrimitivePolynomial property.

## PrimitivePolynomial

Primitive polynomial
Specify the primitive polynomial that defines the finite field $\mathrm{GF}\left(2^{M}\right)$ corresponding to the integers that form messages and codewords. You must set this property to a double-precision, binary row vector that represents a primitive polynomial over GF(2) of degree $M$ in descending order of powers.

This property applies when you set the PrimitivePolynomialSource property to Property.

| Methods | clone | Create HDLRSDecoder System <br> object with same property values |
| :--- | :--- | :--- |
| isLocked | Locked status for input attributes <br> and nontunable properties |  |
|  | release | Allow property value and input <br> characteristics change |
| step | Perform Reed-Solomon decoding |  |

Examples
RS-encode and decode a DVD-II standard packet of random data.

## 1

Assign data and create System objects.
hHDLEnc = comm.HDLRSEncoder(204,188,'BSource','Property','B',0);

```
hHDLDec = comm.HDLRSDecoder(204,188,'BSource','Property','B',0);
dataIn = [randi([0,255],188,1,'uint8') ; zeros(1024-188,1)];
for ii = 1:1024
    [encOut(ii), startOut(ii), endOut(ii), validOut(ii)] = step(hHDLEnc,
        [decOut(ii), decStartOut(ii), decEndOut(ii), decValidOut(ii), decErrOut
end
```

2
Check results.
assert(all(dataIn(1:188) == decOut(decValidOut)'))
See Also
comm.RSDecoder | comm.HDLRSEncoder
Purpose Create HDLRSDecoder System object with same property values
Syntax C = clone(H)
Description C = clone (H) creates another instance of the HDLRSDecoder System object, H, with the same property values. The clone method creates a new unlocked object with uninitialized states.
Input
Arguments
Output
ArgumentsC
H
HDLRSDecoder System object
New instance of the HDLRSDecoder System object, H, withthe same property values. The new unlocked object containsuninitialized states.
See Alsocomm.HDLRSDecoder | comm.HDLRSDecoder.isLocked |comm.HDLRSDecoder.release | comm.HDLRSDecoder.step |
Purpose Locked status for input attributes and nontunable properties
Syntax

L = isLocked(H)Description
Input ..... H

## H

$\mathrm{L}=$ isLocked $(\mathrm{H})$ returns the locked status, L , of the HDLRSDecoderSystem object, H.The isLocked method returns a logical value that indicates whetherinput attributes and nontunable properties for the object are locked. Theobject performs an internal initialization the first time the step methodis executed. This initialization locks nontunable properties and inputspecifications, such as dimensions, complexity, and data type of theinput data. After locking, the isLocked method returns a true value.Arguments
Output

## L

ArgumentsSee Also

comm.HDLRSDecoder | comm.HDLRSDecoder.clone |
comm.HDLRSDecoder.release | comm.HDLRSDecoder.step
|

HDLRSDecoder System object

Logical value. Either 1 (true) or 0 (false).
$\mathrm{L}=$ isLocked $(\mathrm{H})$ returns the locked status, L , of the HDLRSDecoder System object, H. input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input input data. After locking, the isLocked method returns a true value.

## Purpose

Allow property value and input characteristics change

## Syntax

Description

Input
Arguments

See Also comm. HDLRSDecoder I comm. HDLRSDecoder.clone I
comm.HDLRSDecoder.isLocked | comm.HDLRSDecoder.step

Purpose Perform Reed-Solomon decoding
Syntax [Y, startOut, endOut, validOut, errout] = step (H,X,startIn, EndIn, validIn)

## Description

## Input <br> Arguments

[Y,startOut,endOut, validOut,errOut] =
step ( $H, X$, startIn, EndIn, validIn) decodes the input data, $X$, and returns the encoded data, Y, of HDLRSDecoder System object, H.

The step method for this object accepts fixed-point (fi) inputs for $X$.

Note Calling step on an object puts that object into a locked state. When locked, you cannot change nontunable properties or any input characteristics (size, data type and complexity) without reinitializing (unlocking and relocking) the object.

## H

Instance of HDLRSDecoder System object

## X

Message data
Must be an integer (uint8, uint16, uint32) or fi(). Doubles are allowed for simulation but not for HDL code generation.

## startln

Indicates the start of a frame of data. Boolean value.

## endln

Indicates the end of a frame of data. Boolean value.

## validln

Indicates that input data is valid. Boolean value.
OutputArguments
ExamplesRS-encode and decode a DVD-II standard packet of random data.
1
Assign data and create System objects.
hHDLEnc $=$ comm. HDLRSEncoder (204, 188,'BSource','Property','B', 0);
hHDLDec = comm.HDLRSDecoder(204,188,'BSource','Property','B',0);
dataIn = [randi([0,255],188,1,'uint8') ; zeros(1024-188,1)];
for ii = 1:1024
[encOut(ii), startOut(ii), endOut(ii), validOut(ii)] = step(hHDLEnc
[decOut(ii), decStartOut(ii), decEndOut(ii), decValidOut(ii), decEri
end

## 2

Check results.

## comm.HDLRSDecoder.step

assert(all(dataIn(1:188) == decOut(decValidOut)'))
See Also comm.HDLRSDecoder I comm.HDLRSDecoder.clone |
Purpose Encode data using a Reed-Solomon encoder
Description The HDL-optimized HDLRSEncoder System object creates aReed-Solomon code with message and codeword lengths you specify.
Construction H = comm.HDLRSEncoder returns a block encoder System object, H, thatperforms Reed-Solomon (RS) encoding in a streaming fashion for HDL.H = comm.HDLRSEncoder (Name, Value, ) creates an HDL-optimizedblock encoder System object, H, with additional options specified by oneor more Name, Value pair arguments, where Name is a property nameand Value is the corresponding value. Name must appear inside singlequotes (' '). You can specify several name-value pair arguments in anyorder as Name1, Value1, ..., NameN, ValueN.H = comm. HDLRSEncoder( $\mathrm{N}, \mathrm{K}$, Name, Value) creates an RSencoder object, H , with the CodewordLength property set to N , theMessageLength property set to K, and other specified property Name,Value pair arguments.
PropertiesB$B$ value for polynomial generation

## BSource

Source of B, the starting power for roots of the primitive polynomial
Specify the source of the B value as one of these values:

- Auto: B=0
- Property
Default: Auto


## CodewordLength

Codeword length

Specify the codeword length of the RS code as a double-precision, positive, integer scalar value. The default is 7 .

If you set the PrimitivePolynomialSource property to Auto, CodewordLength must be in the range $3<$ CodewordLength $\leq$ $2^{16}-1$.

When you set the PrimitivePolynomialSource property to Property, CodewordLength must be in the range $3 \leq$ CodewordLength $\leq 2^{M}-1 . M$ is the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties. $M$ must be in the range $3 \leq M \leq 16$. The difference (CodewordLength -MessageLength) must be an even integer. The value of this property is rounded up to $2^{M}-1$.

If the value of this property is less than $2^{M}-1$, the object assumes a shortened RS code.

## MessageLength

Message length
Specify the message length as a double-precision, positive integer scalar value. The default is 3 . The difference (CodewordLength MessageLength) must be an even integer.

## PrimitivePolynomialSource

Source of primitive polynomial
Specify the source of the primitive polynomial as Auto | Property. The default is Auto.

When you set this property to Auto, the object uses a primitive polynomial of degree $M=\operatorname{ceil}(\log 2(C o d e w o r d L e n g t h+1)$ ), which is the result of fliplr(de2bi(primpoly $(M))$ ).

When you set this property to Property, you can specify a polynomial using the PrimitivePolynomial property.

## PrimitivePolynomial

## Primitive polynomial

Specify the primitive polynomial that defines the finite field $\operatorname{GF}\left(2^{M}\right)$ corresponding to the integers that form messages and codewords. You must set this property to a double-precision, binary row vector that represents a primitive polynomial over GF(2) of degree $M$ in descending order of powers.

This property applies when you set the PrimitivePolynomialSource property to Property.

## PuncturePatternSource

Source of puncture pattern
Specify the source of the puncture pattern as None | Property. The default is None. If you set this property to None then the object does not apply puncturing to the code. If you set this property to Property then the object punctures the code based on a puncture pattern vector specified in the PuncturePattern property.

## PuncturePattern

Puncture pattern vector
Specify the pattern used to puncture the encoded data as a double-precision, binary column vector with a length of (CodewordLength-MessageLength). The default is [ones $(2,1)$; zeros(2,1)]. Zeros in the puncture pattern vector indicate the position of the parity symbols that are punctured or excluded from each codeword. This property applies when you set the PuncturePatternSource property to Property.

## Methods

clone
isLocked

Create HDLRSEncoder System object with same property values

Locked status for input attributes and nontunable properties

| release | Allow property value and input <br> characteristics change |
| :--- | :--- |
| step | Perform Reed-Solomon encoding |

```
Examples RS-encode a DVD-II standard packet of random data.
```

```
hHDLEnc = comm.HDLRSEncoder(204,188,'BSource','Property','B',0);
```

hHDLEnc = comm.HDLRSEncoder(204,188,'BSource','Property','B',0);
hRSEnc = comm.RSEncoder(204,188,···.
hRSEnc = comm.RSEncoder(204,188,···.
'GeneratorPolynomialSource','Property ',...
'GeneratorPolynomialSource','Property ',...
'GeneratorPolynomial', rsgenpoly(255,239, [],
'GeneratorPolynomial', rsgenpoly(255,239, [],
dataIn = [randi([0,255],188,1,'uint8') ; zeros(255-188,1)];
dataIn = [randi([0,255],188,1,'uint8') ; zeros(255-188,1)];
for ii = 1:255
for ii = 1:255
[dataOut(ii), startOut(ii), endOut(ii), validOut(ii)] = step(hHDLEr
[dataOut(ii), startOut(ii), endOut(ii), validOut(ii)] = step(hHDLEr
end
end
% Check the result:
% Check the result:
Y = step(hRSEnc,dataIn(1:188));
Y = step(hRSEnc,dataIn(1:188));
assert(all(Y == dataOut(validOut)'))

```
assert(all(Y == dataOut(validOut)'))
```

See Also comm.RSEncoder | comm.HDLRSDecoder
Purpose Create HDLRSEncoder System object with same property values
Syntax C = clone(H)
Description C = clone (H) creates another instance of the HDLRSEncoder Systemobject, H, with the same property values. The clone method creates anew unlocked object with uninitialized states.
Input
Arguments
Output ..... C
Arguments
H
HDLRSEncoder System object

New instance of the HDLRSEncoder System object, H, withthe same property values. The new unlocked object containsuninitialized states.
See Alsocomm.HDLRSEncoder | comm.HDLRSEncoder.isLocked |comm.HDLRSEncoder.release | comm.HDLRSEncoder.step |

| Purpose | Locked status for input attributes and nontunable properties |
| :---: | :---: |
| Syntax | L = isLocked ( H ) |
| Description | $\mathrm{L}=$ isLocked $(\mathrm{H})$ returns the locked status, L , of the HDLRSEncoder System object, H. |
|  | The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value. |
| Input Arguments | H HDLRSEncoder System object |
| Output <br> Arguments | L Logical value. Either 1 (true) or 0 (false). |
| See Also | comm. HDLRSEncoder \| comm.HDLRSEncoder.clone | <br> comm. HDLRSEncoder.release \| comm. HDLRSEncoder.step I |

## Purpose

Allow property value and input characteristics change

## Syntax

Description

Input
Arguments
$\begin{array}{ll}\text { See Also } & \begin{array}{l}\text { comm. HDLRSEncoder I comm. } \\ \\ \text { combLRSEncoder.clone I } \\ \text { l }\end{array}\end{array}$
Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## H

release (H) releases system resources (such as memory, file handles or hardware connections) of the HDLRSEncoder System object, H, and allows all its properties and input characteristics to be changed.

Instance of HDLRSEncoder System object

Purpose Perform Reed-Solomon encoding
$\begin{array}{cc}\text { Syntax } & \begin{array}{c}{[Y, \text { startOut, endOut, validOut }]} \\ \text { validIn })\end{array} \\ \text { Description } & \begin{array}{l}{[Y, \text { startOut, endOut, validOut }]=} \\ \text { step }(H, X, \text { startIn, EndIn, validIn }\end{array}\end{array}$

## Input <br> Arguments

 (unlocking and relocking) the object.
## H

Instance of HDLRSEncoder System object
step ( $\mathrm{H}, \mathrm{X}$, startIn, EndIn, validIn) decodes the input data, X , and returns the encoded data, Y, of HDLRSEncoder System object, H.

The step method for this object accepts fixed-point (fi) inputs for X .

Note Calling step on an object puts that object into a locked state. When locked, you cannot change nontunable properties or any input characteristics (size, data type and complexity) without reinitializing

## X

Message data
Must be an integer (uint8, uint16, uint32) or fi(). Doubles are allowed for simulation but not for HDL code generation.

## startln

Indicates the start of a frame of data. Boolean value.

## endln

Indicates the end of a frame of data. Boolean value.

## validln

Indicates that input data is valid. Boolean value.

## Output Arguments

## Examples

1
Assign data and create System objects.
hHDLEnc = comm.HDLRSEncoder(204,188,'BSource','Property','B',0);
hHDLDec = comm.HDLRSDecoder(204,188,'BSource','Property','B',0);
dataIn = [randi([0,255],188,1,'uint8') ; zeros(1024-188,1)];
for ii = 1:1024
[encOut(ii), startOut(ii), endOut(ii), validOut(ii)] = step(hHDLEnc
[decOut(ii), decStartOut(ii), decEndOut(ii), decValidOut(ii), decEr end

2
Check results.

```
assert(all(dataIn(1:188) == decOut(decValidOut)'))
```


## See Also

comm.HDLRSEncoder | comm.HDLRSEncoder.clone | comm.HDLRSEncoder.isLocked | comm.HDLRSEncoder.release |

Description

## Construction

## Properties

## Purpose Restore ordering of symbols using helical array

The HelicalDeinterleaver object permutes the symbols in the input signal by placing them in a row-by-row array and then selecting groups helically to send to the output port.

H = comm.HelicalDeinterleaver creates a helical deinterleaver System object, H. This object restores the original ordering of a sequence that was interleaved using the helical interleaver System object.

H = comm.HelicalDeinterleaver(Name,Value) creates a helical deinterleaver object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## NumColumns

Number of columns in helical array
Specify the number of columns in the helical array as a positive integer scalar value. The default is 6 .

## GroupSize

Size of each group of input symbols
Specify the size of each group of input symbols as a positive integer scalar value. The default is 4 .

## StepSize

Helical array step size
Specify number of rows of separation between consecutive input groups in their respective columns of the helical array. This property requires a positive integer scalar value. The default is 1 .

## InitialConditions

Initial conditions of helical array

Specify the value that is initially stored in the helical array as a numeric scalar value. The default is 0 .

```
Methods
clone
getNumInputs
getNumOutputs
isLocked
release
reset
step
```

clone
getNumInputs
getNumOutputs
isLocked
release
reset
step

Create helical deinterleaver object with same property values
Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes
Reset states of the helical deinterleaver object

Restore ordering of symbols using a helical array

```
Examples Interleave and deinterleave random data.
```

```
hInt = comm.HelicalInterleaver('GroupSize', 2, ...
```

hInt = comm.HelicalInterleaver('GroupSize', 2, ...
'NumColumns', 3, ...
'NumColumns', 3, ...
'InitialConditions', -1);
'InitialConditions', -1);
hDeInt = comm.HelicalDeinterleaver('GroupSize', 2, ...
hDeInt = comm.HelicalDeinterleaver('GroupSize', 2, ...
'NumColumns', 3, ...
'NumColumns', 3, ...
'InitialConditions', -1);
'InitialConditions', -1);
data = randi(7, 6, 1);
data = randi(7, 6, 1);
intData = step(hInt, data);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence, and restore
% compare the original sequence, interleaved sequence, and restore
[data, intData, deIntData]

```
[data, intData, deIntData]
```

```
Algorithms This object implements the algorithm, inputs, and outputs described on the Helical Deinterleaver block reference page. The object properties correspond to the block parameters.
See Also comm.HelicalInterleaver | comm.MultiplexedDeinterleaver
```

Purpose Create helical deinterleaver object with same property values
Syntax $\quad C=$ clone $(H)$
Description $\quad C=$ clone $(H)$ creates a HelicalDeinterleaver object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.HelicalDeinterleaver.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.HelicalDeinterleaver.getNumOutputs

## Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked $(H)$ returns the locked status, TF of the HelicalDeinterleaver System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.HelicalDeinterleaver.reset

Purpose Reset states of the helical deinterleaver object

## Syntax reset (H)

Description reset (H) resets the states of the HelicalDeinterleaver object, H.

## Purpose

Syntax
Description

Restore ordering of symbols using a helical array
Y $=\operatorname{step}(H, X)$
$Y=\operatorname{step}(H, X)$ restores the original ordering of the sequence, $X$, that was interleaved using a helical interleaver and returns Y . The input $X$ must be a column vector. The data type must be numeric, logical, or fixed-point (fi objects). $Y$ has the same data type as $X$. The helical deinterleaver object uses an array for its computations. If you set the NumColumns property of the object to $C$, then the array has $C$ columns and unlimited rows. If you set the GroupSize property to $N$, then the object accepts an input of length $C \times N$ and inserts it into the next $N$ rows of the array. The object also places the value of the InitialConditions property into certain positions in the top few rows of the array. This accommodates the helical pattern and also preserves the vector indices of symbols that pass through the HelicalInterleaver and HelicalDeinterleaver objects. The output consists of consecutive groups of $N$ symbols. The object selects the $k$-th output group in the array from column $k \bmod C$. This selection is of type helical because of the reduction modulo $C$ and because the first symbol in the $k$-th group is in row $1+(k-1) \times s$, where $s$ is the value for the StepSize property.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Permute input symbols using helical array

Description

## Construction

## Properties

The HelicalInterleaver object permutes the symbols in the input signal by placing them in an array in a helical arrangement and then sending rows of the array to the output port.

H = comm. HelicalInterleaver creates a helical interleaver System object, $H$. This object permutes the input symbols in the input signal by placing them in an array in a helical arrangement.

H = comm.HelicalInterleaver(Name, Value) creates a helical interleaver object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## NumColumns

Number of columns in helical array
Specify the number of columns in the helical array as a positive integer scalar value. The default is 6 .

## GroupSize

Size of each group of input symbols
Specify the size of each group of input symbols as a positive integer scalar value. The default is 4 .

## StepSize

Helical array step size
Specify the number of rows of separation between consecutive input groups in their respective columns of the helical array. This property requires as a positive integer scalar value. The default is 1 .

## InitialConditions

Initial conditions of helical array

Specify the value that is initially stored in the helical array as a numeric scalar value. The default is 0 .

| Methods | clone |
| :--- | :--- |
| getNumInputs |  |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| reset |  |
| step |  |

Create helical interleaver object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of the helical interleaver object

Permute input symbols using a helical array

Examples Interleave and deinterleave random data.

```
hInt = comm.HelicalInterleaver('GroupSize', 2, ...
    'NumColumns', 3, ...
    'InitialConditions', -1);
hDeInt = comm.HelicalDeinterleaver('GroupSize', 2, ...
    'NumColumns', 3, ...
    'InitialConditions', -1);
data = randi(7, 6, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence, and restor
[data, intData, deIntData]
```


## comm.HelicalInterleaver

[^7]
# Purpose Create helical interleaver object with same property values 

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a HelicalInterleaver object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.HelicalInterleaver.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNuminputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.HelicalInterleaver.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the HelicalInterleaver System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.Helicallnterleaver.reset

Purpose Reset states of the helical interleaver object

## Syntax reset (H)

Description reset $(H)$ resets the states of the HelicalInterleaver object, H.

Purpose Permute input symbols using a helical array
Syntax $\quad Y=\operatorname{step}(H, X)$
Description
$Y=\operatorname{step}(H, X)$ permutes input sequence, $X$, and returns interleaved sequence, $Y$. The input $X$ must be a column vector. The data type must be numeric, logical, or fixed-point (fi objects). Y has the same data type as $X$. The helical interleaver object places the elements of $X$ in an array in a helical fashion. If you set the NumColumns property of the object to $C$, then the array has $C$ columns and unlimited rows. If you set the GroupSize property to $N$, then the object accepts an input of length $C \times N$ and partitions the input into consecutive groups of $N$ symbols. The object places the $k$-th group in the array along column $k \bmod C$. This placement is of type helical because of the reduction modulo C and because the first symbol in the $k$-th group is in the row $1+(k-1) \times s$, where $s$ is the value for the StepSize property. Positions in the array that do not contain input symbols have default contents specified by the InitialConditions property. The object outputs $C \times N$ symbols from the array by reading the next $N$ rows sequentially.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Convert vector of integers to vector of bits

Description

## Construction

## Properties

The IntegerToBit object maps each integer (or fixed-point value) in the input vector to a group of bits in the output vector.

H = comm. IntegerToBit creates an integer-to-bit converter System object, H. This object maps a vector of integer-valued or fixed-point inputs to a vector of bits.

H = comm.IntegerToBit(Name, Value) creates an integer-to-bit converter object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.IntegerToBit(NUMBITS,Name, Value) creates an integer-to-bit converter object, H . This object has the BitsPerInteger property set to NUMBITS and the other specified properties set to the specified values.

## BitsPerInteger

Number of bits per integer
Specify the number of bits the System object uses to represent each input integer. You must set this property to a scalar integer between 1 and 32. The default is 3 .

## MSBFirst

Output bit words with first bit as most significant bit
Set this property to true to indicate that the first bit of the output bit words is the most significant bit (MSB). The default is true. Set this property to false to indicate that the first bit of the output bit words is the least significant bit (LSB).

## SignedIntegerinput

Assume inputs are signed integers

Set this property to true if the integer inputs are signed. The default is false. Set this property to false if the integer inputs are unsigned. If the SignedInteger Input property is false, the input values must be between 0 and $\left(2^{\wedge} \mathrm{N}\right)-1$. In this case, $N$ is the value you specified in the BitsPerInteger property. When you set this property to true, the input values must be between $-\left(2^{(N-1)}\right)$ and $\left(2^{(N-1)}\right)-1$.

## OutputDataType

Data type of output
Specify output data type as one of Full precision | Smallest unsigned integer | Same as input | double | single | int8 | uint8 | int16| uint16 | int32 | uint32 | logical. The default is Full precision.

When the input signal is an integer data type, you must have a Fixed-Point Designer user license to use this property in Smallest unsigned integer or Full precision mode.

When you set this property to Full precision, the object determines the output data type based on the input data type. If the input data type is double- or single-precision, the output data has the same data type as the input data. Otherwise, the output data type is determined in the same way as when you set this property to Smallest unsigned integer.

When you set this property to Same as input, and the input data type is numeric or fixed-point integer (fi object), the output data has the same data type as the input data.

Methods<br>clone<br>getNumInputs

Create an integer-to-bit converter object with same property values

Number of expected inputs to step method

getNumOutputs<br>isLocked<br>release<br>step

Number of outputs from step method
Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes

Convert vector of integers to vector of bits

## Examples

## Algorithms

See Also

Convert randomly generated integers to 4 -bit words.

```
hIntToBit = comm.IntegerToBit(4);
intData = randi([0 2^hIntToBit.BitsPerInteger-1],3,1);
bitData = step(hIntToBit,intData);
```

This object implements the algorithm, inputs, and outputs described on the Integer to Bit Converter block reference page. The object properties correspond to the block parameters.
comm.BitToInteger | de2bi | dec2bin

Purpose
Create an integer-to-bit converter object with same property values

## Syntax <br> C = clone(H)

Description
$\mathrm{C}=\mathrm{clone}(\mathrm{H})$ creates a IntegerToBit object C , with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.IntegerToBit.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.IntegerToBit.getNumOutputs

## Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the IntegerToBit System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Convert vector of integers to vector of bits

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description
$Y=\operatorname{step}(H, X)$ converts integer input, $X$, to corresponding bits, $Y$. The input must be scalar or a column vector and the data type can be numeric or fixed-point (fi objects). The output is a column vector with length equal to length $(\mathrm{X}) \times N$, where $N$ is the value of the BitsPerInteger property. If any input value is outside the range of $N$, the object issues an error. If the SignedIntegerInput property is false, the input values must be between 0 and $\left(2^{N}\right)$ - 1 . If you set the SignedInteger Input property to true, the input values must be between - $\left(2^{(N-1)}\right)$ and $\left(2^{(N-1)}\right)-1$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose

Integrate discrete-time signal with periodic resets

The IntegrateAndDumpFilter object creates a cumulative sum of the

## Construction

## Properties

discrete-time input signal, while resetting the sum to zero according to a fixed schedule. When the simulation begins, the object discards the number of samples specified in the Offset property. After this initial period, the object sums the input signal along columns and resets the sum to zero every Ninput samples, set by the integration period property. The reset occurs after the object produces output at that time step.

H = comm.IntegrateAndDumpFilter creates an integrate and dump filter System object, H. this object integrates over a number of samples in an integration period, and then resets at the end of that period.

H = comm. IntegrateAndDumpFilter(Name, Value) creates an integrate and dump filter object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.IntegrateAndDumpFilter(PERIOD,Name, Value)
creates an integrate and dump filter object, H. This object has the IntegrationPeriod property set to PERIOD and the other specified properties set to the specified values.

## IntegrationPeriod

Integration period
Specify the integration period, in samples, as a positive, integer scalar value greater than 1 . The integration period defines the length of the sample blocks that the object integrates between resets. The default is 8 .

## Offset

Number of offset samples
Specify a nonnegative, integer vector or scalar specifying the number of input samples that the object discards from each
column of input data at the beginning of data processing. Discarding begins when you call the step method for the first time. The default is 0 .

When you set the Offset property to a nonzero value, the object outputs one or more zeros during the initial period while discarding input samples.

When you specify this property as a vector of length $L$, the $i$-th element of the vector corresponds to the offset for the $i$-th column of the input data matrix, which has $L$ columns.

When you specify this property as a scalar value, the object applies the same offset to each column of the input data matrix. The offset creates a transient effect, rather than a persistent delay.

## DecimateOutput

## Decimate output

Specify whether the step method returns intermediate cumulative sum results or decimates intermediate results. The default is true.

When you set this property to true, the step method returns one output sample, consisting of the final integration value, for each block of IntegrationPeriod input samples. If the inputs are ( $K \times$ IntegrationPeriod) $\times L$ matrices, then the outputs are $K \times L$ matrices.

When you set this property to false, the step method returns IntegrationPeriod output samples, comprising the intermediate cumulative sum values, for each block of IntegrationPeriod input samples. In this case, inputs and outputs have the same dimensions.

## Fixed-Point Properties

## FullPrecisionOverride

Full precision override for fixed-point arithmetic

Specify whether to use full precision rules. If you set FullPrecisionOverride to true, which is the default, the object computes all internal arithmetic and output data types using full precision rules. These rules provide the most accurate fixed-point numerics. It also turns off the display of other fixed-point properties because they do not apply individually. These rules guarantee that no quantization occurs within the object. Bits are added, as needed, to ensure that no roundoff or overflow occurs. If you set FullPrecisionOverride to false, fixed-point data types are controlled through individual fixed-point property settings. For more information, see "Full Precision for Fixed-Point System Objects".

## RoundingMethod

Rounding of fixed-point numeric values
Specify the rounding method as one of Ceiling | Convergent | Floor | Nearest | Round | Simplest | Zero. The default is Floor. This property applies only if the object is not in full precision mode.

## OverflowAction

Action when fixed-point numeric values overflow
Specify the overflow action as one of Wrap | Saturate. The default is Wrap. This property applies only if the object is not in full precision mode.

## AccumulatorDataType

Data type of accumulator
Specify the accumulator data type as one of Full precision | Same as input | Custom. The default is Full precision. When you set this property to Full precision the object automatically calculates the accumulator output word and fraction lengths. Set this property to Custom to specify the accumulator data type using the CustomAccumulatorDataType property. This property applies when you set the FullPrecisionOverride property to false.

## CustomAccumulatorDataType

Fixed-point data type of accumulator
Specify the accumulator fixed-point type as a scaled numerictype object with a signedness of Auto. The default is numerictype([],32,30). This property applies when you set the FullPrecisionOverride property to false and the AccumulatorDataType property to Custom.

## OutputDataType

Data type of output
Specify the output fixed-point type as one of Same as accumulator | Same as input | Custom. The default is Same as accumulator. This property applies when you set the FullPrecisionOverride property to false.

## CustomOutputDataType

Fixed-point data type of output
Specify the output fixed-point type as a scaled numerictype object with a signedness of Auto. The default is numerictype ([], 32,30 ). This property applies when you set the FullPrecisionOverride property to false and the OutputDataType property to Custom.

| Methods | clone | Create integrate and dump filter <br> object with same property values |
| :--- | :--- | :--- |
| getNumInputs | Number of expected inputs to <br> step method |  |
| getNumOutputs | Number of outputs from step <br> method |  |
| isLocked | Locked status for input attributes <br> and nontunable properties |  |

## comm.IntegrateAndDumpFilter

| release | Allow property value and input <br> characteristics changes |
| :--- | :--- |
| step | Integrate discrete-time signal <br> with periodic resets |

Examples Integrate a signal specifying an integration period of 5 samples.

```
hInt = comm.IntegrateAndDumpFilter(5);
hInt.Offset = 3;
% Data matrix contains three columns (i.e. three channels)
data = reshape(1:30, 10, 3);
result = step(hInt, data)
```

Algorithms This object implements the algorithm, inputs, and outputs described on the Integrate and Dump block reference page. The object properties correspond to the block parameters, except:

The Output intermediate values parameter corresponds to the DecimateOutput property.

## comm.IntegrateAndDumpFilter.clone

Purpose Create integrate and dump filter object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a IntegrateAndDumpFilter object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.IntegrateAndDumpFilter.getNumInputs

## Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs(H)

## comm.IntegrateAndDumpFilter.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked $(H)$ |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the <br> IntegrateAndDumpFilter System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.IntegrateAndDumpFilter.release

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose Integrate discrete-time signal with periodic resets

$$
\text { Syntax } \quad Y=\operatorname{step}(H, X)
$$

Description $\quad \mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X})$ periodically integrates blocks of N samples from the input data, X , and returns the result in $\mathrm{Y} . \mathrm{N}$ is the number of samples that you specify in the IntegrationPeriod property. X is a column vector or a matrix and the data type is double, single or fixed-point (fi objects). X must have K*N rows for some positive integer K, with one or more columns. The object treats each column as an independent channel with integration occurring along every column. The dimensions of output $Y$ depend on the value you set for the DecimateOutput property.

> Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

Purpose Generate Kasami sequence

Description

Construction

H = comm.KasamiSequence creates a KasamiSequence System object, H. This object generates a Kasami sequence.

H = comm.KasamiSequence (Name, Value) creates a Kasami sequence generator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties

## Polynomial

Generator polynomial
Specify the polynomial that determines the shift register's feedback connections. The default is $\left[\begin{array}{lllllll}1 & 0 & 0 & 0 & 0 & 1 & 1\end{array}\right]$.

You can specify the generator polynomial as a binary numeric vector that lists the coefficients of the polynomial in descending order of powers. The first and last elements must equal 1. Specify the length of this vector as $n+1$, where $n$ is the degree of the generator polynomial and must be even.

Alternatively, you can specify the generator polynomial as a vector containing the exponents of $z$ for the nonzero terms of the polynomial in descending order of powers. The last entry must be 0. For example, [ $1000000010 c 1]$ and [8 200 ] represent the same polynomial, $g(z)=z^{8}+z^{2}+1$.

## InitialConditions

Initial conditions of shift register
Specify the initial values of the shift register as a binary numeric scalar or as binary numeric vector. The default is $\begin{array}{lllll}0 & 0 & 0 & 0 & 0\end{array}$

1]. Set the vector length equal to the degree of the generator polynomial.
When you set this property to a vector value, each element of the vector corresponds to the initial value of the corresponding cell in the shift register.

When you set this property to a scalar value, that value specifies the initial conditions of all the cells of the shift register. The scalar, or at least one element of the specified vector, requires a nonzero value for the object to generate a nonzero sequence.

## Index

Sequence index
Specify the index to select a Kasami sequence of interest from the set of possible sequences. The default is 0 . Kasami sequences have a period equal to $N=2^{n}-1$, where $n$ indicates a nonnegative, even integer equal to the degree of the generator polynomial that you specify in the Polynomial property.

There are two classes of Kasami sequences: those obtained from a small set and those obtained from a large set. You choose a Kasami sequence from the small set by setting this property to a numeric, scalar, integer value in the range $\left[0 . . .2^{n / 2}-2\right]$. You choose a sequence from the large set by setting this property to a numeric $1 \times 2$ integer vector $\left[k m\right.$ ] for $k$ in $\left[-2, \ldots, 2^{n}-2\right]$, and $m$ in $\left[-1, \ldots, 2^{n / 2}-2\right]$.

## Shift

Sequence offset from initial time
Specify the offset of the Kasami sequence from its starting point as a numeric, integer scalar value that can be positive or negative. The default is 0 . The Kasami sequence has a period of $N=2^{n}-1$, where $n$ is the degree of the generator polynomial that you specify in the Polynomial property. The shift value is wrapped with respect to the sequence period.

## VariableSizeOutput

Enable variable-size outputs
Set this property to true to enable an additional input to the step method. The default is false. When you set this property to true, the enabled input specifies the output size of the Kasami sequence used for the step. The input value must be less than or equal to the value of the MaximumOutputSize property.

When you set this property to false, the SamplesPerFrame property specifies the number of output samples.

## MaximumOutputSize

Maximum output size
Specify the maximum output size of the Kasami sequence as a positive integer 2 -element row vector. The second element of the vector must be 1 . The default is [10 1].

This property applies when you set the VariableSizeOutput property to true.

## SamplesPerFrame

Number of output samples per frame
Specify the number of Kasami sequence samples that the step method outputs as a numeric, positive, integer scalar value. The default value is 1 .

When you set this property to a value of $M$, then the step method outputs $M$ samples of a Kasami sequence that has a period of $N=$ $2^{n}-1$. The value $n$ equals the degree of the generator polynomial that you specify in the Polynomial property.

## ResetInputPort

Enable generator reset input
Set this property to true to enable an additional input to the step method. The default is false. The additional input resets the states of the Kasami sequence generator to the initial conditions that you specify in the InitialConditions property.

## OutputDataType

Data type of output
Specify the output data type as one of double | logical. The default is double.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
|  | release |
| reset |  |
| step |  |

Examples Generate 5 samples of a Kasami sequence of length 63.

```
hks = comm.KasamiSequence('SamplesPerFrame', 5);
x = step(hks)
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the Kasami Sequence Generator block reference page. The object properties correspond to the block parameters, except:

- The object does not have a property to select frame based outputs.


## comm.KasamiSequence

- The object does not have a property that corresponds to the Sample time parameter.

See Also comm. PNSequence | comm. GoldSequence

Purpose
Create Kasami sequence generator object with same property values

## Syntax <br> C = clone(H)

Description

C = clone (H) creates a KasamiSequence object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.KasamiSequence.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.KasamiSequence.getNumOutputs

## Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the KasamiSequence System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

# Purpose Reset states of Kasami sequence generator object 

## Syntax reset (H)

Description reset (H) resets the states of the KasamiSequence object, H.

Purpose
Generate a Kasami sequence
Syntax
$Y=\operatorname{step}(H)$
$Y=\operatorname{step}(H, R E S E T)$
$Y=\operatorname{step}(H)$ outputs a frame of the Kasami sequence in column vector Y. Specify the frame length with the SamplesPerFrame property. The Kasami sequence has a period of $N=2^{n}-1$, where $n$ is the degree of the generator polynomial that you specify in the Polynomial property.
$Y=$ step ( $H$, RESET) uses RESET as the reset signal when you set the ResetInputPort property to true. The data type of the RESET input must be double precision or logical. RESET can be a scalar value or a column vector with a length equal to the number of samples per frame that you specify in the SamplesPerFrame property. When the RESET input is a non-zero scalar, the object resets to the initial conditions that you specify in the InitialConditions property. It then generates a new output frame. A column vector RESET input allows multiple resets within an output frame. A non-zero value at the $i$-th element of the vector causes a reset at the $i$-th output sample time.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Decode binary low-density parity-check code
DescriptionThe LDPCDecoder object decodes a binary low-density parity-check code.
Construction$\mathrm{h}=$ comm.LDPCDecoder creates a binary low-density parity-check(LDPC) decoder System object, $h$. This object performs LDPC decodingbased on the specified parity-check matrix, where the object does notassume any patterns in the parity-check matrix.
h = comm.LDPCDecoder('PropertyName','ValueName')
creates an LDPC encoder object, $h$, with each specifiedproperty set to the specified value. You can specifyadditional name-value pair arguments in any order as
('PropertyName1','PropertyValue1',...,'PropertyNameN','PropertyValueN').
$\mathrm{h}=$ comm.LDPCDecoder(PARITY) creates an LDPC decoder object, $h$,with the ParityCheckMatrix property set to PARITY.

## Properties <br> ParityCheckMatrix

Parity-check matrix
Specify the parity-check matrix as a binary valued sparse matrix $P$ with dimension ( $N$-by- $K$ ) by $N$, where $N>K>0$. Alternatively, you can specify a two-column, non-sparse integer index matrix I that defines the row and column indices of the 1 s in the parity-check matrix, such that $P=\operatorname{sparse}(I(:, 1), I(:, 2), 1)$.
This property accepts numeric data types. When you set this property to a sparse matrix, it also accepts a logical data type. The upper bound for the value of $N$ is $2^{31}-1$.
The default is the sparse parity-check matrix of the half-rate LDPC code from the DVB-S. 2 standard, which is the result of dvbs2ldpc(1/2).
To generate code, set this property to a non-sparse index matrix. For instance, you can obtain the index matrix for the DVB-S. 2 standard from dvbs2ldpc(R, 'indices') with the second input
argument explicitly specified to indices, where $R$ represents the code rate.

## OutputValue

Select output value format
Specify the output value format as one of 'Information part' | 'Whole codeword'. The default is 'Information part'. When you set this property to 'Information part', the output contains only the message bits and is a $K$ element column vector, assuming an ( $N$-by- $K$ ) $\mathrm{x} K$ parity check matrix. When you set this property to 'Whole codeword', the output contains the codeword bits and is an $N$ element column vector.

## DecisionMethod

Decision method
Specify the decision method used for decoding as one of 'Hard decision' | 'Soft decision'. The default is 'Hard decision'. When you set this property to 'Hard decision', the output is decoded bits of double or logical data type. When you set this property to 'Soft decision', the output is log-likelihood ratios of double data type.

## IterationTerminationCondition

Condition for iteration termination
Specify the condition to stop the decoding iterations as one of 'Maximum iteration count' | 'Parity check satisfied'. The default is 'Maximum iteration count'. When you set this property to 'Maximum iteration count', the object will iterate for the number of iterations you specify in the MaximumIterationCount property. When you set this property to 'Parity check satisfied', the object will determine if the parity checks are satisfied after each iteration and stops if all parity checks are satisfied.

## MaximumlterationCount

Maximum number of decoding iterations

Specify the maximum number of iterations the object uses as an integer valued numeric scalar. The default is 50 . This applies when you set the IterationTerminationCondition property to 'Maximum iteration count'.

## NumlterationsOutputPort

Output number of iterations performed
Set this property to true to output the actual number of iterations the object performed. The default is false.

## FinalParityChecksOutputPort

Output final parity checks
Set this property to true to output the final parity checks the object calculated. The default is false.

| Methods clone |  |
| :--- | :--- |
| isLocked |  |
|  | release |
| step |  |

Examples Transmit an LDPC-encoded, QPSK-modulated bit stream through an AWGN channel, then demodulate, decode, and count errors.

```
hEnc = comm.LDPCEncoder;
hMod = comm.PSKModulator(4, 'BitInput',true);
hChan = comm.AWGNChannel(...
    'NoiseMethod','Signal to noise ratio (SNR)','SNR',1);
hDemod = comm.PSKDemodulator(4, 'BitOutput',true,...
    'DecisionMethod','Approximate log-likelihood ratio', ...
```

```
    'Variance', 1/10^(hChan.SNR/10));
hDec = comm.LDPCDecoder;
hError = comm.ErrorRate;
for counter = 1:10
    data = logical(randi([0 1], 32400, 1));
    encodedData = step(hEnc, data);
    modSignal = step(hMod, encodedData);
    receivedSignal = step(hChan, modSignal);
    demodSignal = step(hDemod, receivedSignal);
    receivedBits = step(hDec, demodSignal);
    errorStats = step(hError, data, receivedBits);
end
fprintf('Error rate = %1.2f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the LDPC Decoder block reference page. The object properties correspond to the block parameters.
comm.LDPCEncoder | comm.BCHDecoder | comm.gpu.LDPCDecoder

Purpose Create LDPC Decoder object with same property values

## Syntax <br> C = clone(H)

Description
C = clone (H) creates an LDPC Decoder object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

# Purpose Locked status for input attributes and nontunable properties 

Syntax TF = isLocked (H)
Description Description
TF = isLocked (H) returns the locked status, TF of the LDPCEncode System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H) release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Encode input using LDPC coding scheme |
| :--- | :--- |
| Syntax | $Y=\operatorname{step}(H, X)$ |
|  | $[Y, \operatorname{NUMITER}]=\operatorname{step}(H, X)$ |
|  | $[Y, \operatorname{PARITY}]=\operatorname{step}(H, X)$ |

## Description

$Y=\operatorname{step}(H, X)$ decodes input codeword, $X$, using an LDPC code that is based on an ( $N-K$ ) x $N$ parity-check matrix. You specify the parity-check matrix in the ParityCheckMatrix property. Input $X$ must be a double column vector with length equal $N$. Each element is the log-likelihood ratio for a received bit (more likely to be 0 if the log-likelihood ratio is positive). The first $K$ elements correspond to the information part of a codeword. The decoded data output vector, Y, contains either only the message bits or the whole code word, based on the value of the OutputValue property.
$[\mathrm{Y}$, NUMITER] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$ returns the actual number of iterations the object performed when you set the NumIterationsOutputPort property to true. The step method outputs NUMITER as a double scalar.
[ $\mathrm{Y}, \mathrm{PARITY}$ ] = step $(\mathrm{H}, \mathrm{X})$ returns final parity checks the object calculated when you set the FinalParityChecksOutputPort property to true. The step method outputs PARITY as a double vector of length ( $N-K$ ). You can combine optional output arguments when you set their enabling properties. Optional outputs must be listed in the same order as the order of the enabling properties. For example, [ $\mathrm{Y}, \mathrm{NUMITER}, \mathrm{PARITY}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose <br> Description

Construction

Encode binary low-density parity-check code

The LDPCEncoder object encodes a binary low-density parity-check code.
$\mathrm{h}=$ comm.LDPCEncoder creates a binary low-density parity-check (LDPC) encoder System object, $h$. This object performs LDPC encoding based on the specified parity-check matrix.
h = comm.LDPCEncoder('PropertyName','ValueName') creates an LDPC encoder object, $h$, with each specified property set to the specified value.
$\mathrm{h}=$ comm.LDPCEncoder(PARITY) creates an LDPC encoder object, $h$, with the ParityCheckMatrix property set to PARITY.
output_args = function(input_args,Name, Value) with additional options specified by one or more Name, Value pair arguments. Name can also be a property name and Value is the corresponding value. Name must appear inside single quotes (' '). You can specify several name-value pair arguments in any order as Name1,Value1, ,NameN,ValueN.

## Properties ParityCheckMatrix

Parity-check matrix
Specify the parity-check matrix as a binary valued sparse matrix $P$ with dimension $(N$-by- $K$ ) by $N$, where $N>K>0$. Alternatively, you can specify a two-column, non-sparse integer index matrix I that defines the row and column indices of the 1 s in the parity-check matrix, such that $P=\operatorname{sparse}(I(:, 1), I(:, 2), 1)$.

This property accepts numeric data types. When you set this property to a sparse matrix, it also accepts a logical data type. The upper bound for the value of $N$ is $2^{31}-1$.

The default is the sparse parity-check matrix of the half-rate LDPC code from the DVB-S. 2 standard, which is the result of dvbs2ldpc(1/2).

To generate code, set this property to a non-sparse index matrix. For instance, you can obtain the index matrix for the DVB-S. 2 standard from dvbs2ldpc ( $R$, 'indices') with the second input argument explicitly specified to indices, where $R$ represents the code rate.

| Methods | clone |
| :--- | :--- |
| isLocked |  |
| release |  |
| step |  |

Copy Handle. To learn how handle classes affect copy operations, see Copying Semantics Objects in the MATLAB documentation.

## Examples

Transmit an LDPC-encoded, QPSK-modulated bit stream through an AWGN channel, then demodulate, decode, and count errors

```
hEnc = comm.LDPCEncoder;
hMod = comm.PSKModulator(4, 'BitInput',true);
hChan = comm.AWGNChannel(...
    'NoiseMethod','Signal to noise ratio (SNR)','SNR',1);
hDemod = comm.PSKDemodulator(4, 'BitOutput',true,...
    'DecisionMethod','Approximate log-likelihood ratio', ...
    'Variance', 1/10^(hChan.SNR/10));
hDec = comm.LDPCDecoder;
hError = comm.ErrorRate;
for counter = 1:10
    data = logical(randi([0 1], 32400, 1));
    encodedData = step(hEnc, data);
    modSignal = step(hMod, encodedData);
```

```
    receivedSignal = step(hChan, modSignal);
    demodSignal = step(hDemod, receivedSignal);
    receivedBits = step(hDec, demodSignal);
    errorStats = step(hError, data, receivedBits);
end
fprintf('Error rate = %1.2f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```

Algorithms correspond to the block parameters.<br>See Also<br>comm.LDPCDecoder | comm.BCHEncoder

This object implements the algorithm, inputs, and outputs described on the LDPC Encoder block reference page. The object properties

## Purpose Create LDPC Encoder object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates an LDPC Encoder object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.
The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked ( H ) returns the locked status, TF of the LDPCEncode System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Encode input using LDPC coding scheme

## Syntax <br> Y = step(H,X)

Description
$Y=\operatorname{step}(H, X)$ encodes input binary message, $X$, using an LDPC code that is based on an ( $N-K$ ) x $N$ parity-check matrix. You specify the parity-check matrix in the ParityCheckMatrix property. Input X must be a numeric or logical column vector with length equal $K$. The length of the encoded data output vector, Y , is $N$. It is a solution to the parity-check equation, with the first $K$ bits equal to the input, X .

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose

Filter input signal through LTE MIMO multipath fading channel

## Construction

## Properties

## SampleRate

Input signal sample rate (Hertz)
Specify the sample rate of the input signal in hertz as a double-precision, real, positive scalar. The default value of this property is 30.72 MHz , as defined in the LTE specification.

## Profile

Channel propagation profile
Specify the propagation conditions of the LTE multipath fading channel as one of EPA $5 \mathrm{~Hz} \mid$ EVA $5 \mathrm{~Hz} \mid$ EVA $70 \mathrm{~Hz} \mid$ ETU 70

Hz | ETU 300 Hz , which are supported in the LTE specification Release 10. The default value of this property is EPA 5 Hz .
This property defines the delay profile of the channel to be one of EPA, EVA, and ETU. This property also defines the maximum Doppler shift of the channel to be $5 \mathrm{~Hz}, 70 \mathrm{~Hz}$, or 300 Hz. The Doppler spectrum always has a Jakes shape in the LTE specification. The EPA profile has seven paths. The EVA and ETU profiles have nine paths.

The following tables list the delay and relative power per path associated with each profile.

Extended Pedestrian A Model (EPA)

| Excess tap delay [ns] | Relative power [db] |
| :---: | :---: |
| 0 | 0.0 |
| 30 | -1.0 |
| 70 | -2.0 |
| 90 | -3.0 |
| 110 | -8.0 |
| 190 | -17.2 |
| 410 | -20.8 |

Extended Vehicular A Model (EVA)

| Excess tap delay [ns] | Relative power [db] |
| :---: | :---: |
| 0 | 0.0 |
| 30 | -1.5 |
| 150 | -1.4 |
| 310 | -3.6 |
| 370 | -0.6 |


| Excess tap delay [ns] | Relative power [db] |
| :---: | :---: |
| 710 | -9.1 |
| 1090 | -7.0 |
| 1730 | -12.0 |
| 2510 | -16.9 |

Extended Typical Urban Model (ETU)

| Excess tap delay [ns] | Relative power [db] |
| :---: | :---: |
| 0 | -1.0 |
| 50 | -1.0 |
| 120 | -1.0 |
| 200 | 0.0 |
| 230 | 0.0 |
| 500 | 0.0 |
| 1600 | -3.0 |
| 2300 | -5.0 |
| 5000 | -7.0 |

## AntennaConfiguration

Antenna configuration
Specify the antenna configuration of the LTE MIMO channel as one of $1 \times 2|2 \times 2| 4 \times 2 \mid 4 \times 4$. These configurations are supported in the LTE specification Release 10. The default value of this property is $2 \times 2$.
The property value is in the format of $N_{\mathrm{t}}$-by $-N_{\mathrm{r}} . N_{\mathrm{t}}$ represents the number of transmit antennas and $N_{\mathrm{r}}$ represents the number of receive antennas.

## CorrelationLevel

Spatial correlation strength
Specify the spatial correlation strength of the LTE MIMO channel as one of Low | Medium | High. The default value of this property is Low. When you set this property to Low, the MIMO channel is spatially uncorrelated.

The transmit and receive spatial correlation matrices are defined from this property according to the LTE specification Release 10. See the Algorithms section for more information.

## AntennaSelection

Antenna selection
Specify the antenna selection scheme as one of Off | Tx|Rx|Tx and $R x$, where $T x$ represents transmit antennas and $R x$ represents receive antennas. When you select Tx and/or Rx, additional input(s) are required to specify which antennas are selected for signal transmission. The default value of this property is Off.

## RandomStream

Source of random number stream
Specify the source of random number stream as one of Global stream | mt19937ar with seed. The default value of this property is Global stream. When you set this property to Global stream, the current global random number stream is used for normally distributed random number generation. In this case, the reset method only resets the filters. If you set RandomStream to mt19937ar with seed, the object uses the mt19937ar algorithm for normally distributed random number generation. In this case, the reset method resets the filters and reinitializes the random number stream to the value of the Seed property.

## Seed

Initial seed of mt19937ar random number stream
Specify the initial seed of an mt19937ar random number generator algorithm as a double-precision, real, nonnegative integer scalar.

The default value of this property is 73. This property applies when you set the RandomStream property to mt19937ar with seed. The Seed reinitializes the mt19937ar random number stream in the reset method.

## NormalizePathGains

Normalize path gains (logical)
Set this property to true to normalize the fading processes so that the total power of the path gains, averaged over time, is 0 dB . The default value of this property is true. When you set this property to false, there is no normalization for path gains.

## NormalizeChannelOutputs

Normalize channel outputs (logical)
Set this property to true to normalize the channel outputs by the number of receive antennas. The default value of this property is true. When you set this property to false, there is no normalization for channel outputs.

## PathGainsOutputPort

Enable path gain output (logical)
Set this property to true to output the channel path gains of the underlying fading process. The default value of this property is false.

Methods<br>clone<br>getNumInputs<br>getNumOutputs

Create object with same property values

Number of expected inputs to step method

Number of outputs from step method

| isLocked | Locked status for input attributes <br> and nontunable properties |
| :--- | :--- |
| release | Allow property value and input <br> characteristics changes |
| reset | Reset states of the object |
| step | Filter input signal through LTE <br> MIMO multipath fading channel |

## Examples Configure Equivalent MIMO Channel System Object Using an LTE MIMO Channel System Object

Configure an equivalent MIMOChannel System Object using the LTEMIMOChannel System Object. Then, verify that the channel output and the path gain output from the two objects are the same.

Create a PSK Modulator System object to modulate randomly generated data.

```
hMod = comm.PSKModulator;
modData = step(hMod, randi([0 hMod.ModulationOrder-1],2e3,1));
```

Split modulated data into two spatial streams.

```
channelInput = reshape(modData, [2, 1e3]).';
```

Create an LTEMIMOChannel System object with a 2-by-2 antenna configuration and a medium correlation level.

```
hLTEChan = comm.LTEMIMOChannel(...
    'Profile', 'EVA 5Hz',...
    'AntennaConfiguration', '2x2',...
    'CorrelationLevel', 'Medium',...
    'RandomStream', 'mt19937ar with seed',...
    'Seed', 99,...
    'PathGainsOutputPort', true);
```

Filter the modulated data using the LTEMIMOChannel System object, hLTEChan.
[LTEChanOut, LTEPathGains] = step(hLTEChan, channelInput);
Create an equivalent MIMOChannel System object, hMIMOChan, using the properties of the LTEMIMOChannel System object, hLTEChan.

Note The KFactor, DirectPathDopplerShift and DirectPathInitialPhase properties only exist for the MIMOChannel System object. All other MIMOChannel System object properties also exist for the LTEMIMOChannel System object; however, some properties are hidden and read-only.

```
hMIMOChan = comm.MIMOChannel(...
    'SampleRate',
    'SampleRate',
    'AveragePathGains', hLTEChan.AveragePathGains,...
    'MaximumDopplerShift', hLTEChan.MaximumDopplerShift,
    'DopplerSpectrum',
    'NumTransmitAntennas',
    'NumReceiveAntennas',
    'TransmitCorrelationMatrix',
    'ReceiveCorrelationMatrix',
    'FadingDistribution',
    'RandomStream',
    'Seed',
    'NormalizePathGains',
    'NormalizeChannelOutputs',
    'PathGainsOutputPort',
        hLTEChan.DopplerSpectrum,...
hLTEChan.NumTransmitAntennas,...
hLTEChan.NumReceiveAntennas,...
hLTEChan.TransmitCorrelationMatrix,...
hLTEChan.ReceiveCorrelationMatrix,...
hLTEChan.FadingDistribution,...
hLTEChan.RandomStream, ...
hLTEChan.Seed,...
hLTEChan.NormalizePathGains,...
hLTEChan.NormalizeChannelOutputs,...
hLTEChan.PathGainsOutputPort);
```

Filter the modulated data using the equivalent hMIMOChan and use the step method to process data.
[MIMOChanOut, MIMOPathGains] = step(hMIMOChan, channelInput);

Verify that the channel output and the path gain output from the two objects are the same.
display(isequal(LTEChanOut, MIMOChanOut));
display(isequal(LTEPathGains, MIMOPathGains));

Note You can repeat the preceding process with AntennaConfiguration set to $4 \times 2$ or $4 \times 4$ and CorrelationLevel set to Medium or High for hLTEChan. If you do so, the resulting channel output and path gain output from the two objects are slightly different. This difference occurs because an LTE channel with such configurations has its spatial correlation matrix rounded to 4 -digit precision. See the LTE specification Release 10 for more details.

## Algorithms

This System object is a specialized implementation of the comm.MIMOChannel System object. For additional algorithm information, see the comm.MIMOChannel System object help page.

## Spatial Correlation Matrices

The following table defines the transmitter eNodeB correlation matrix.

|  | One Antenna | Two Antennas | Four <br> Antennas |
| :--- | :--- | :--- | :--- | :--- |
| eNodeB <br> Correlation | $R_{\mathrm{eNB}}=1$ |  |  |

The following table defines the receiver UE correlation matrix.

## comm.LTEMIMOChannel

|  | One Antenna | Two Antennas | Four <br> Antennas |
| :--- | :--- | :--- | :--- |
| UE Correlation | $R_{\mathrm{UE}}=1$ |  |  |
|  |  | $R_{U E}=\left(\begin{array}{ll}1 & \beta \\ \beta^{*} & 1\end{array}\right)$ |  |
|  |  |  | $R_{U E}=\left(\begin{array}{ccc}1 & \beta^{1 / 9} & \beta^{4 /} \\ \beta^{1 / 9^{*}} & 1 & \beta^{1 / 2} \\ \beta^{4 / 9^{*}} & \beta^{1 / 9^{*}} & 1 \\ \beta^{*} & \beta^{4 / 9^{*}} & \beta^{1 / 9} \\ \hline\end{array}\right.$ |
|  |  |  |  |

The following table describes the $R_{\text {spat }}$ channel spatial correlation matrix between the transmitter and receiver antennas.

| Tx-by-Rx Configuration | Correlation Matrix |
| :--- | :--- |
| 1-by-2 | $R_{\text {spat }}=R_{U E}=\left[\begin{array}{cc}1 & \beta \\ \beta^{*} & 1\end{array}\right]$ |
| 2-by-2 | $R_{\text {spat }}=R_{e N B} \otimes R_{U E}=\left[\begin{array}{cc}1 & \alpha \\ \alpha^{*} & 1\end{array}\right] \otimes\left[\begin{array}{cc}1 & \beta \\ \beta^{*} & 1\end{array}\right]=$ |


| Tx-by-Rx Configuration | Correlation Matrix |  |  |
| :---: | :---: | :---: | :---: |
| 4-by-2 |  |  |  |
|  | $R_{\text {spat }}=R_{e N B} \otimes R_{U E}=$ | $\left[\begin{array}{cc} 1 & \alpha^{1 / 9} \\ \alpha^{1 / 9} & 1 \\ \alpha^{4 / 9} & \alpha^{1 / 9} \\ \alpha^{*} & \alpha^{4 / 9} \end{array}\right.$ | $\alpha^{4 / 9}$ $\alpha^{1 / 9}$ 1 $1 / 9$ |
| 4-by-4 |  |  |  |
|  | $R_{\text {spat }}=R_{e N B} \otimes R_{U E}=$ | $\left[\begin{array}{cc}1 & \alpha^{1 / 9} \\ \alpha^{1 / 9} & 1 \\ \alpha^{4 / 9} & \alpha^{1 / 9} \\ \alpha^{*} & \alpha^{4 / 9}\end{array}\right.$ | $\alpha^{4 / 9}$ $\alpha^{1 / 9}$ 1 $1 / 9$ |

## Spatial Correlation Correction

| Low Correlation |  | Medium Correlation |  | High Correlation |  |
| :---: | :---: | :---: | :---: | :---: | :---: |
| $\alpha$ | $B$ | $\alpha$ | $B$ | $\alpha$ | $B$ |
| 0 | 0 | 0.3 | 0.9 | 0.9 | 0.9 |

To insure the correlation matrix is positive semi-definite after round-off to 4 digit precision, this System object uses the following equation:

$$
R_{\text {high }}=\left[R_{\text {spatial }}+a I_{n}\right] /(1+a)
$$

Where
a represents the scaling factor such that the smallest value is used to obtain a positive semi-definite result.
For the 4 -by- 2 high correlation case, $\alpha=0.00010$.
For the 4-by-4 high correlation case, $\alpha=0.00012$.
The object uses the same method to adjust the 4 -by- 4 medium correlation matrix to insure the correlation matrix is positive semi-definite after rounding to 4 digit precision with $\alpha=0.00012$.

## Selected Bibliography

[1] 3rd Generation Partnership Project, Technical Specification Group Radio Access Network, Evolved Universal Terrestrial Radio Access (E-UTRA), Base Station (BS) radio transmission and reception, Release 10, 2009-2010, 3GPP TS 36.104, Vol. 10.0.0.
[2] 3rd Generation Partnership Project, Technical Specification Group Radio Access Network, Evolved Universal Terrestrial Radio Access (E-UTRA), User Equipment (UE) radio transmission and reception, Release 10, 2010, 3GPP TS 36.101, Vol. 10.0.0.
[3] Oestges, C., and B. Clerckx. MIMO Wireless Communications: From Real-World Propagation to Space-Time Code Design, Academic Press, 2007.
[4] Correira, L. M. Mobile Broadband Multimedia Networks: Techniques, Models and Tools for 4G, Academic Press, 2006.
[5] Jeruchim, M., P. Balaban, and K. S. Shanmugan. Simulation of Communication Systems, Second Edition, New York, Kluwer Academic/Plenum, 2000.

See Also comm.mimochannel

## comm.LTEMIMOChannel.clone

Purpose Create LTEMIMOChannel object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates an LTEMIMOChannel object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## Purpose Number of expected inputs to step method

Syntax
N = getNumInputs( H )

Description
$N=$ getNumInputs $(H)$ returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

The getNumInputs method returns a positive integer that is the number of expected inputs (not counting the object itself) to the step method. This value will change if you alter any properties that turn inputs on or off. You must call the step method with the number of input arguments equal to the result of getNumInputs $(H)$.

## comm.LTEMIMOChannel.getNumOutputs

Purpose Number of outputs from step method

## Syntax

Description
$N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

The getNumOutputs method returns a positive integer that is the number of outputs from the step method. This value will change if you alter any properties that turn outputs on or off.

## comm.LTEMIMOChannel.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :---: | :---: |
| Syntax | TF = isLocked( H ) |
| Description | TF = isLocked (H) returns the locked status, TF of the LTEMIMOChannel System object. |
|  | The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value. |

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description
release (H) releases system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

# Purpose Reset states of the LTEMIMOChannel object 

## Syntax reset (H)

Description reset $(H)$ resets the states of the LTEMIMOChannel object, $H$.
If you set the RandomStream property of H to Global stream, the reset method only resets the filters. If you set RandomStream to mt19937ar with seed, the reset method not only resets the filters but also reinitializes the random number stream to the value of the Seed property.

# Purpose <br> Filter input signal through LTE MIMO multipath fading channel 

Syntax

Y = step (H, X)
[Y,PATHGAINS] = step(H,X)
$Y=\operatorname{step}(H, X)$ filters input signal $X$ through an LTE MIMO multipath fading channel and returns the result in $Y$. The input $X$ can be a double-precision data type scalar, vector, or 2-D matrix with real or complex values. X is of size $\mathrm{N}_{\mathrm{s}}$-by- $\mathrm{N}_{\mathrm{t}} . \mathrm{N}_{\mathrm{s}}$ represents the number of samples and $N_{t}$ represents the number of transmit antennas that must match the AntennaConfiguration property setting of $\mathrm{H} . \mathrm{Y}$ is the output signal of size $\mathrm{N}_{\mathrm{s}}$-by- $\mathrm{N}_{\mathrm{r}} . \mathrm{N}_{\mathrm{r}}$ represents the number of receive antennas that is specified by the AntennaConfiguration property of $\mathrm{H} . \mathrm{Y}$ is of double-precision data type with complex values.
[ Y, PATHGAINS $]=\operatorname{step}(\mathrm{H}, \mathrm{X})$ returns the LTE MIMO channel path gains of the underlying fading process in PATHGAINS. This applies when you set the PathGainsOutputPort property to true. PATHGAINS is of size $\mathrm{N}_{\mathrm{s}}$-by $-\mathrm{N}_{\mathrm{p}}$-by $-\mathrm{N}_{\mathrm{t}}$-by- $\mathrm{N}_{\mathrm{r}} . \mathrm{N}_{\mathrm{p}}$ represents the number of discrete paths of the channel implicitly defined by the Profile property of H . PATHGAINS is of double-precision data type with complex values.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Deinterleave input symbols using permutation matrix |
| :---: | :---: |
| Description | The MatrixDeinterleaver object performs block deinterleaving by filling a matrix with the input symbols column by column and then sending the matrix contents to the output port row by row. The number of rows and number of columns properties set the dimensions of the matrix that the object uses internally for computations. |
| Construction | H = comm.MatrixDeinterleaver creates a matrix deinterleaver System object, H. This object restores the original ordering of a sequence that was interleaved using the matrix interleaver object. |
|  | H = comm.MatrixDeinterleaver(Name, Value) creates a matrix deinterleaver object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN). |
|  | $H$ = comm.MatrixDeinterleaver $(N, M)$ creates a matrix deinterleaver object, H. This object has the NumRows property set to N, the NumColumns property set to M . |
| Properties | NumRows |
|  | Number of rows of permutation matrix |
|  | Specify the number of permutation matrix rows as a scalar, positive integer. The default is 3 . |
|  | NumColumns |
|  | Number of columns of permutation matrix |
|  | Specify the number of permutation matrix columns as a scalar, positive integer. The default is 4 . |

Methods<br>clone getNumInputs getNumOutputs isLocked release step

Create matrix deinterleaver object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Deinterleave input symbols using permutation matrix

## Examples Interleave and deinterleave data.

```
hInt = comm.MatrixInterleaver('NumRows', 2, ...
    'NumColumns', 5);
hDeInt = comm.MatrixDeinterleaver('NumRows', 2, ...
    'NumColumns', 5);
data = randi(7, 10, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
[data, intData, deIntData]
```


## Algorithms <br> This object implements the algorithm, inputs, and outputs described on the Matrix Deinterleaver block reference page. The object properties correspond to the block parameters.

## See Also

comm.MatrixInterleaver | comm.BlockDeinterleaver

Purpose
Create matrix deinterleaver object with same property values

## Syntax <br> C = clone(H)

Description

C = clone (H) creates a MatrixDeinterleaver object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.MatrixDeinterleaver.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked $(H)$ returns the locked status, TF of the MatrixDeinterleaver System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Deinterleave input symbols using permutation matrix

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description $\quad Y=\operatorname{step}(H, X)$ restores the original ordering of the sequence, $X$, that was interleaved using a block interleaver. The object fills a permutation matrix with the input symbols column by column and outputs the matrix contents row by row in the output, $Y$. The input $X$ must be a column vector of length equal to NumRows $\times$ NumColumns. The data type for X can be numeric, logical, or fixed-point (fi objects). Y has the same data type as $X$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose

Permute input symbols using permutation matrix

The MatrixInterleaver object performs block interleaving by filling a matrix with the input symbols row by row and then outputs the matrix contents column-by-column.

## Construction

## Properties

H = comm.MatrixInterleaver creates a matrix interleaver System object, H. This object permutes the input by filling a permutation matrix with the input symbols row by row. The object then outputs the matrix contents column by column.

H = comm.MatrixInterleaver(Name, Value) creates a matrix interleaver object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.MatrixInterleaver( $N$, M) creates a matrix interleaver object, H. This object has the NumRows property set to N, the NumColumns property set to M.

## NumRows

Number of rows of permutation matrix
Specify the number of permutation matrix rows as a scalar, positive integer. The default is 3 .

## NumColumns

Number of columns of permutation matrix
Specify the number of permutation matrix columns as a scalar, positive integer. The default is 4 .

## Methods

clone
getNumInputs

Create matrix interleaver object with same property values

Number of expected inputs to step method

getNumOutputs<br>isLocked<br>release<br>step

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Permute input symbols using permutation matrix

## Examples Interleave and deinterleave data

```
hInt = comm.MatrixInterleaver('NumRows', 2, ...
    'NumColumns', 5);
hDeInt = comm.MatrixDeinterleaver('NumRows', 2, ...
    'NumColumns', 5);
data = randi(7, 10, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence, and restored
[data, intData, deIntData]
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the Matrix Deinterleaver block reference page. The object properties correspond to the block parameters.
comm.MatrixDeinterleaver | comm.BlockInterleaver

# Purpose <br> Create matrix interleaver object with same property values 

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a MatrixInterleaver object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.MatrixInterleaver.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the MatrixInterleaver System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Permute input symbols using permutation matrix

## Syntax <br> Y = step( $\mathrm{H}, \mathrm{X}$ )

Description
$Y=\operatorname{step}(H, X)$ permutes input sequence, $X$, and returns interleaved sequence, $Y$. The object fills a permutation matrix with the input symbols row by row and outputs the matrix contents column by column. The input $X$ must be a column vector of length NumRows $\times$ NumColumns and the data type can be numeric, logical, or fixed-point (fi objects). $Y$ has the same data type as $X$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose

Deinterleave input symbols by filling a matrix along diagonals

The MatrixHelicalScanDeinterleaver object performs block deinterleaving by filling a matrix with the input symbols helically and then outputs the matrix contents row by row. The number of rows and number of columns properties represent the dimensions of the matrix that the object uses internally for computations.

## Construction

H = comm.MatrixHelicalScanDeinterleaver creates a matrix helical scan deinterleaver object, H. This object restores the original ordering of a sequence that was interleaved using the matrix helical scan interleaver System object.

H = comm.MatrixHelicalScanDeinterleaver(Name, Value) creates a matrix helical scan deinterleaver object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties

## NumRows

Number of rows of permutation matrix
Specify the number of rows in the permutation matrix as a scalar, positive integer. The default is 64 .

## NumColumns

Number of columns of permutation matrix
Specify the number of columns in the permutation matrix as a scalar, positive integer. The default is 64 .

## StepSize

Slope of diagonals
Specify slope as a scalar integer between 0 and the value you specify in the NumRows property. The default is 1. The slope value indicates the amount by which the row index increases as the column index increases by 1 . When you set the value of
this property to 0 , the object does not interleave and the output matches the input.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| step |  |

Create matrix helical scan deinterleaver object with same property values

Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Deinterleave input symbols by filling a matrix along diagonals

## Examples Interleave and deinterleave random data.

```
hInt = comm.MatrixHelicalScanInterleaver('NumRows', 4, ...
    'NumColumns', 4);
hDeInt = comm.MatrixHelicalScanDeinterleaver('NumRows', 4, ...
    'NumColumns', 4);
data = randi(7, 16, 1);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence and restored se
[data, intData, deIntData];
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the Matrix Helical Scan Deinterleaver block reference page. The object properties correspond to the block parameters.

See Also comm. MatrixHelicalScanInterleaver | comm.BlockDeinterleaver

## comm.MatrixHelicalScanDeinterleaver.clone

Purpose $\quad \begin{aligned} & \text { Create matrix helical scan deinterleaver object with same property } \\ & \text { values }\end{aligned}$
Syntax $\quad C=\operatorname{clone}(H)$
Description $\quad C=$ clone $(H)$ creates a MatrixHelicalScanDeinterleaver object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.MatrixHelicalScanDeinterleaver.getNuminputs

## Purpose Number of expected inputs to step method

## Syntax <br> $N$ = getNumInputs( H )

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)
Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked $(H)$ |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the |
|  | MatrixHelicalScanDeinterleaver System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose

Deinterleave input symbols by filling a matrix along diagonals

$$
\text { Syntax } \quad Y=\operatorname{step}(H, X)
$$

Description
$Y=\operatorname{step}(H, X)$ restores the original ordering of the sequence, $X$. The object fills a permutation matrix with the input symbols in a helical fashion and output the contents row by row, and returns Y . The input $X$ must be a NumRows $\times$ NumColumns long column vector and the data type can be numeric, logical, or fixed-point (fi objects). Y has the same data type as $X$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

Description

## Construction

## Properties

## Purpose Permute input symbols by selecting matrix elements along diagonals

The MatrixHelicalScanInterleaver object performs block interleaving by filling a matrix with the input symbols row by row and then outputs the matrix contents in a helical helically. The number of rows and number of columns properties are the dimensions of the matrix that the object uses internally for computations.

H = comm.MatrixHelicalScanInterleaver creates a matrix helical scan interleaver object, H. This object permutes the input by filling a permutation matrix with the input symbols row by row and then outputs the matrix contents helically.

H = comm.MatrixHelicalScanInterleaver(Name, Value) creates a matrix helical scan interleaver object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## NumRows

Number of rows of permutation matrix
Specify the number of rows in the permutation matrix as a scalar, positive integer. The default is 64 .

## NumColumns

Number of columns of permutation matrix
Specify the number of columns in the permutation matrix as a scalar, positive integer. The default is 64 .

## StepSize

Slope of diagonals
Specify slope as a scalar integer between 0 and the value you specify in the NumRows property. The slope value represents the amount by which the row index increases as the column index increases by 1 . When you set the value of this property to 0 , the
object does not interleave and the output matches the input. The default is 1 .

```
Methods
clone
getNumInputs
getNumOutputs
isLocked
release
step
\begin{tabular}{ll} 
clone & \begin{tabular}{l} 
Create matrix helical scan \\
interleaver object with same \\
property values
\end{tabular} \\
getNumInputs & \begin{tabular}{l} 
Number of expected inputs to \\
step method
\end{tabular} \\
getNumOutputs & \begin{tabular}{l} 
Number of outputs from step \\
method
\end{tabular} \\
isLocked & \begin{tabular}{l} 
Locked status for input attributes \\
and nontunable properties
\end{tabular} \\
release & \begin{tabular}{l} 
Allow property value and input \\
characteristics changes
\end{tabular} \\
step & \begin{tabular}{l} 
Permute input symbols by \\
selecting matrix elements along \\
diagonals
\end{tabular}
\end{tabular}
Examples Interleave and deinterleave random data.
```

```
hInt = comm.MatrixHelicalScanInterleaver('NumRows', 4, ...
```

hInt = comm.MatrixHelicalScanInterleaver('NumRows', 4, ...
'NumColumns', 4);
'NumColumns', 4);
hDeInt = comm.MatrixHelicalScanDeinterleaver('NumRows', 4, ...
hDeInt = comm.MatrixHelicalScanDeinterleaver('NumRows', 4, ...
'NumColumns', 4);
'NumColumns', 4);
data = randi(7, 16, 1);
data = randi(7, 16, 1);
intData = step(hInt, data);
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence and restorec
% compare the original sequence, interleaved sequence and restorec
[data, intData, deIntData]

```
[data, intData, deIntData]
```

Algorithms | This object implements the algorithm, inputs, and outputs described on |
| :--- |
| the Matrix Helical Scan Deinterleaver block reference page. The object |
| properties correspond to the block parameters. |

See Also $\quad$| comm.MatrixHelicalScanDeinterleaver \| comm.BlockInterleaver |
| :--- |

Purpose
Create matrix helical scan interleaver object with same property values

## Syntax <br> C = clone(H)

Description

C = clone (H) creates a MatrixHelicalScanInterleaver object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.MatrixHelicalScanInterleaver.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNuminputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.MatrixHelicalScanInterleaver.getNumOutputs

## Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax $\quad$ TF $=$ isLocked (H)
Description TF = isLocked $(H)$ returns the locked status, TF of the MatrixHelicalScanInterleaver System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Permute input symbols by selecting matrix elements along diagonals

## Syntax <br> $Y=\operatorname{step}(H, X)$

Description
$Y=\operatorname{step}(H, X)$ permutes input sequence, $X$, and returns interleaved sequence, Y . The input X must be a NumRows $\times$ NumColumns long column vector and the data type can be numeric, logical, or fixed-point (fi objects). $Y$ has the same data type as $X$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Apply memoryless nonlinearity to input signal |
| :---: | :--- |
| Description | The MemorylessNonlinearity object applies a memoryless nonlinearity <br> to a complex, baseband signal. You can use the object to model radio <br> frequency (RF) impairments to a signal at the receiver. |
| Construction | H = comm.MemorylessNonlinearity creates a memoryless nonlinearity <br> System object, H. This object models receiver radio frequency (RF) <br> impairments. |
| H = comm. MemorylessNonlinearity (Name, Value) creates a <br> memoryless nonlinearity object, H, with each specified property set <br> to the specified value. You can specify additional name-value pair <br> arguments in any order as (Name1,Value1,...,NameN,ValueN). |  |
| PropertiesMethod <br> Method used to model nonlinearity |  |
| Specify the nonlinearity method as one of Cubic polynomial <br> I Hyperbolic tangent \| Saleh model I Ghorbani model | <br> Rapp model. The default is Cubic polynomial. This property <br> is non-tunable. |  |
| InputScaling |  |
| Scale factor applied to input signal |  |
| Specify the scale factor in decibels. The object applies this |  |
| factor to the input signal as a real scalar value of double- or |  |
| single-precision data type. The default is 0. This property applies |  |
| when you set the Method property to Saleh model or Ghorbani |  |
| model. This property is tunable. |  |

the Method property to Cubic polynomial, Hyperbolic tangent, or Rapp model. This property is tunable.

## IIP3

Third-order input intercept point
Specify the third-order input intercept point (in decibels relative to a milliwatt) as a real scalar value of double- or single-precision data type. The default is 30 . This property applies when you set the Method property to Cubic polynomial or Hyperbolic tangent. This property is tunable.

## AMPMConversion

AM/PM conversion factor
Specify the AM/PM conversion factor (in degrees per decibel) as a real scalar value of double- or single-precision data type. The default is 10. This property applies when you set the Method property to Cubic polynomial or Hyperbolic tangent. This property is tunable.

## AMAMParameters

AM/AM conversion parameters
Specify the AM/AM conversion parameters that the object uses to compute the amplitude gain for an input signal as a real vector of double- or single-precision data type. The default is [2.1587 1.1517] for the Saleh model and [8.1081 1.54136 .5202 -0.0718] for the Ghorbani model.

This property applies when you set the Method property to Saleh model or Ghorbani model.

When you set the Method property to Saleh model, this property is a two-element vector that specifies alpha and beta values. Otherwise, this property is a four-element vector that specifies $\times 1$, $\mathrm{x} 2, \mathrm{x} 3$, and x 4 values. This property is tunable.

## AMPMParameters

AM/PM conversion parameters
Specify the AM/PM conversion parameters used to compute the phase change for an input signal as a real vector of double- or single-precision data type. The default is [4.0033 9.1040] for the Saleh model and [4.6645 $2.096510 .88-0.003$ ] for the Ghorbani model.

This property applies when you set the Method property to Saleh model or Ghorbani model.

When you set the Method property to Saleh model, this property is a two-element vector that specifies alpha and beta values. Otherwise, this property is a four-element vector that specifies y1, $\mathrm{y} 2, \mathrm{y} 3$, and y 4 values. This property is tunable.

## PowerLowerLimit

Lower input power limit
Specify the minimum input power (in decibels relative to a milliwatt) for which AM/PM conversion scales linearly with input power value. The default is 10 . Below this value, the phase shift resulting from AM/PM conversion is zero. You must set this property to a real scalar value of double- or single-precision data type. This property applies when you set the Method property to Cubic polynomial or Hyperbolic tangent. This property is tunable.

## PowerUpperLimit

Upper input power limit
Specify the maximum input power (in decibels relative to a milliwatt) for which AM/PM conversion scales linearly with input power value. The default is inf. Above this value, the phase shift resulting from AM/PM conversion is constant. You must set the PowerUpperLimit property to a real scalar value, which is greater than the PowerLowerLimit property and of doubleor single-precision data type. This property applies when you
set the Method property to Cubic polynomial or Hyperbolic tangent.This property is tunable.

## OutputScaling

Scale factor applied to output signal
Specify the scale factor (in decibels) that the object applies to the output signal as a real scalar value of double- or single-precision data type. The default is 0 . This property applies when you set the Method property to Saleh model or Ghorbani model. This property is tunable.

## Smoothness

Smoothness factor
Specify the smoothness factor as a real scalar value of doubleor single-precision data type. The default is 0.5 . This property applies when you set the Method property to Rapp model. This property is tunable.

## OutputSaturationLevel

Output saturation level
Specify the output saturation level as a real scalar value of double- or single-precision data type. This property applies when you set the Method property to Rapp model. The default is 1. This property is tunable.

Methods<br>clone<br>getNumInputs<br>getNumOutputs

Create memoryless nonlinearity object with same property values

Number of expected inputs to step method

Number of outputs from step method

| isLocked | Locked status for input attributes <br> and nontunable properties |
| :--- | :--- |
| release | Allow property value and input <br> characteristics changes |
| step | Apply memoryless nonlinearity to <br> input signal |
| Apply "Saleh model" nonlinearity to a 16-QAM signal. |  |

## Algorithms

This object implements the algorithm, inputs, and outputs described on the Memoryless Nonlinearity block reference page. The object properties correspond to the block parameters.

See Also
comm. PhaseNoise

Purpose Create memoryless nonlinearity object with same property values

## Syntax

Description
C = clone (H) creates a MemorylessNonlinearity object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

# comm.MemorylessNonlinearity.getNumInputs 

## Purpose Number of expected inputs to step method

## Syntax <br> N = getNumInputs( H )

Description
$N=$ getNumInputs $(H)$ returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs(H)

## comm.MemorylessNonlinearity.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.MemorylessNonlinearity.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked (H) |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the <br> MemorylessNonlinearity System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.MemorylessNonlinearity.release

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

# Purpose Apply memoryless nonlinearity to input signal <br> Syntax $\quad Y=\operatorname{step}(H, X)$ 

Description $\quad Y=\operatorname{step}(H, X)$ applies memoryless nonlinearity to the input, $X$, using the nonlinearity method you specify in the Method property, and returns the result in $Y$. The input $X$ must be a complex scalar or column vector of data type double or single precision. The output, Y , is of the same data type as the input, $X$.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Measure modulation error ratio
Description The Modulation Error Ratio (MER) MER object measures thesignal-to-noise ratio (SNR) in digital modulation applications. You canuse these types of measurements to determine system performance incommunications applications. For example, determining if an EDGEsystem conforms to 3GPP radio transmission standards requiresaccurate MER, Minimum MER, and 95th percentile for the MERmeasurements. The block measures all outputs in decibels.

## Construction

## Properties

H = comm.MER creates a modulation error ratio (MER) System object, H. This object measures the signal-to-noise ratio (SNR) in digital modulation applications.
$\mathrm{H}=$ comm. MER(Name, Value) creates an MER object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## MinimumMEROutputPort

Enable minimum MER measurement output
When you set this property to true, the step method outputs minimum MER measurements. The default is false. The step method outputs the minimum MER output as the minimum MER value measured in the current input frame.

## XPercentileMEROutputPort

Enable X-percentile MER measurement output
When you set this property to true, the step method outputs X-percentile MER measurements. The default is false. The X-percentile MER measurements persist. These measurements are based on all the input frames since the last reset.

## XPercentileValue

X-percentile value

Specify the X-percentile value (as a percentage) that the object uses to calculate the $x^{\text {th }}$ percentile of the MER measurements. The default is 95 . Set this property to a real scalar value between 0 and 100 , inclusive. This property can have a data type of double, single, or integer. This property applies when you set the XPercentileMEROutputPort property to true. The $x$-th percentile is the MER value above which $x \%$ of all the computed MER values lie.

## SymbolCountOutputPort

Enable symbol count output
When you set this property to true, the step method outputs the number of accumulated symbols that have been used to calculate the $x$-Percentile MER measurements since the last reset. The default is false. This property applies when you set the XPercentileMEROutputPort property to true.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>reset<br>step

Create MER measurement object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of MER measurement object
Measure modulation error ratio

```
Examples Measure MER of a noisy 16-QAM modulated signal.
    hMod \(=\) comm.RectangularQAMModulator(16);
    hAWGN = comm.AWGNChannel('NoiseMethod',...
    'Signal to noise ratio (SNR)',...
    'SNR', 20, 'SignalPower', 10);
\% Create an MER object, output minimum and 90-percentile MER, and symbol
\% count
    hMER = comm.MER('MinimumMEROutputPort', true, ...
    'XPercentileMEROutputPort', true,'XPercentileValue', 90,
    'SymbolCountOutputPort',true);
\% Generate modulated symbols and add noise
    refsym = step(hMod, randi([0 15], 1000, 1));
    rxsym = step(hAWGN, refsym);
\% Calculate measurements
    [MERdB,MinMER,PercentileMER,NumSym] = step(hMER,refsym,rxsym)
```

Measure MER of a noisy 16-QAM modulated signal.
hMod = comm.RectangularQAMModulator(16);
hAWGN = comm.AWGNChannel('NoiseMethod',..
'Signal to noise ratio (SNR)',...
'SNR', 20, 'SignalPower', 10);
\% Create an MER object, output minimum and 90-percentile MER, and symbol
\% count
hMER = comm.MER('MinimumMEROutputPort', true, ...
XPercentileMEROutputPort', true, 'XPercentileValue', 90,
'SymbolCountOutputPort',true);
\% Generate modulated symbols and add noise
refsym = step(hMod, randi([0 15], 1000, 1));
rxsym = step(hAWGN, refsym);
\% Calculate measurements
[MERdB,MinMER,PercentileMER,NumSym] = step(hMER,refsym,rxsym)
This object implements the algorithm, inputs, and outputs described on the MER Measurement block reference page. The object properties correspond to the block parameters.

## Algorithms

## See Also

comm.EVM | comm.ACPR | comm.CCDF

# Purpose <br> Create MER measurement object with same property values 

## Syntax <br> C = clone(H)

Description
C = clone (H) creates a MER object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax TF = isLocked (H)

Description TF = isLocked (H) returns the locked status, TF of the MER System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

# Purpose Reset states of MER measurement object 

## Syntax reset (H)

Description reset (H) resets the states of the MER object, H.
Purpose Measure modulation error ratio
Syntax

MERDB = step(H,REFSYM,RXSYM)
[MERDB,MINMER] = step(H,REFSYM,RXSYM)
[MERDB, PMER] = step(H,REFSYM,RXSYM)
[MERDB,NUMSYM] = step(H,REFSYM,RXSYM)

## Description

MERDB $=\operatorname{step}(\mathrm{H}$, REFSYM, RXSYM) outputs MER (in dB), MERDB, measured in the received signal, RXSYM, based on the reference signal, REFSYM. REFSYM, and RXSYM inputs are complex column vectors of equal dimensions and data type. The data type can be double, single, signed integer, or signed fixed point with power-of-two slope and zero bias. The step method outputs the MERDB measurement based solely on the current input frame. All outputs of this object are of data type double.
[MERDB,MINMER] = step(H,REFSYM,RXSYM) outputs the minimum MER (in dB), MINMER, measured in the received signal, RXSYM, when you set the MinimumMEROutputPort property to true. The step method outputs the MINMER measurement based on the reference signal, REFSYM. MINMER is the minimum MER value measured in the current input frame.
[MERDB, PMER] = step(H,REFSYM, RXSYM) outputs the percentile MER (in dB), PMER, measured in the received signal, RXSYM, when you set the XPercentileMEROutputPort property to true. The step method outputs the PMER measurement based on the reference signal, REFSYM. The object sets PMER equal to a value just smaller than the XPercentileValue percent of all the MER values. For example, if you set the XPercentileValue property to 95 , then $95 \%$ of all MER measurements are above the PMER value. The object calculates the persistent measurement PMER, using all the input frames since the last reset.
[MERDB, NUMSYM] = step(H,REFSYM,RXSYM) outputs the number of symbols, NUMSYM, used to calculate the X-Percentile MER measurements when you set the SymbolCountOutputPort property to true. You can combine optional output arguments when you set their enabling
properties. Optional outputs must be listed in the same order as the order of the enabling properties. For example,

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Filter input signal through MIMO multipath fading channel |
| :--- | :--- |
| Description $\quad$The MIMOChannel System object filters an input signal through a <br> multiple-input multiple-output (MIMO) multipath fading channel. <br> This object models both Rayleigh and Rician fading and employs the <br> Kronecker model for modeling the spatial correlation between the links. |  |
| The fading processing per link is per the Methodology for Simulating |  |
| Multipath Fading Channels section and assumes the same parameters |  |
| for all $N_{\mathrm{T}} \cdot N_{\mathrm{R}}$ links of the MIMO channel. Each link comprises all |  |
| multipaths for that link. |  | | H = comm. MIMOChannel creates a multiple-input multiple-output |
| :--- |
| (MIMO) frequency selective or frequency flat fading channel System |
| object, H. This object filters a real or complex input signal through the |
| multipath MIMO channel to obtain the channel impaired signal. |

## AveragePathGains

Average path gain vector (decibels)
Specify the average gains of the discrete paths in decibels as a double-precision, real, scalar or row vector. The default value of this property is 0 . AveragePathGains must have the same size as PathDelays.

## MaximumDopplerShift

Maximum Doppler shift (Hertz)
Specify the maximum Doppler shift for all channel paths in hertz as a double precision, real, nonnegative scalar. The default value of this property is 0.001 Hz .
The Doppler shift applies to all the paths of the channel. When you set the MaximumDopplerShift to 0 , the channel remains static for the entire input. You can use the reset method to generate a new channel realization.

The MaximumDopplerShift must be smaller than SampleRate $/ 10 / f_{c}$ for each path, where $f_{\mathrm{c}}$ represents the cutoff frequency factor of the path. For a Doppler spectrum type other than Gaussian and BiGaussian, the value of $f_{\mathrm{c}}$ is 1 . For these two Doppler spectrum types, $f_{\mathrm{c}}$ is dependent on the Doppler spectrum object properties. Refer to the algorithm section of this page for more details about how $f_{c}$ is defined.

## DopplerSpectrum

Doppler spectrum object(s)
Specify the Doppler spectrum shape for all channel paths as a single object from the Doppler spectrum package or a row vector of such objects. The default value of this property is a Jakes Doppler spectrum object.

The maximum Doppler shift value necessary to specify the Doppler spectrum/spectra is given by the MaximumDopplerShift
property. This property applies when the MaximumDopplerShift property value is greater than 0 .
If you assign a single Doppler spectrum object to DopplerSpectrum, all paths have the same specified Doppler spectrum. Select from the following:

- doppler.jakes
- doppler.flat
- doppler.rjakes(...)
- doppler.ajakes(...)
- doppler.rounded(...)
- doppler.bell(...)
- doppler.gaussian(...)
- doppler.bigaussian(...)

You can assign DopplerSpectrum a vector of Doppler spectrum objects, which can be chosen from any of those in the previous list. Each path has the Doppler spectrum specified by the corresponding Doppler spectrum object in the vector. In this case, the length of DopplerSpectrum must be equal to the length of PathDelays.

This object supports C code generation for the Jakes Doppler spectrum. You cannot explicitly specify the DopplerSpectrum property for code generation.

## TransmitCorrelationMatrix

Transmit correlation matrix (or 3D array)
Specify the spatial correlation of the transmitter as a double-precision, real or complex, 2D matrix or 3D array. The default value of this property is [10;01].

The first dimension of TransmitCorrelationMatrix determines the number of transmit antennas, $\mathrm{N}_{\mathrm{t}}$. This dimension must be a value between 1 and 8 , inclusive.

If the channel is frequency flat, i.e., PathDelays is a scalar, TransmitCorrelationMatrix is a 2D Hermitian matrix of size $\mathrm{N}_{\mathrm{t}}$-by- $\mathrm{N}_{\mathrm{t}}$. The main diagonal elements must be all ones. The off-diagonal elements must be real or complex numbers with a magnitude smaller than or equal to one.

If the channel is frequency selective, i.e., PathDelays is a row vector of length $\mathrm{N}_{\mathrm{p}}$, you can specify TransmitCorrelationMatrix as an $\mathrm{N}_{\mathrm{t}}$-by- $\mathrm{N}_{\mathrm{t}}$ matrix. In this case, each path has the same transmit spatial correlation matrix. Alternatively, you can specify the value as a 3 D array of size $\mathrm{N}_{\mathrm{t}}$-by- $\mathrm{N}_{\mathrm{t}}$-by- $\mathrm{N}_{\mathrm{p}}$. In this case, each path can have its own transmit spatial correlation matrix.

## ReceiveCorrelationMatrix

Receive correlation matrix (or 3D array)
Specify the spatial correlation of the receiver as a double-precision, real or complex, 2D matrix or 3D array. The default value of this property is [10;01].

The first dimension of ReceiveCorrelationMatrix determines the number of receive antennas, $\mathrm{N}_{\mathrm{r}}$. This dimension must be a value between 1 and 8 , inclusive.

If the channel is frequency flat, i.e., PathDelays is a scalar, ReceiveCorrelationMatrix is a 2D Hermitian matrix of size $\mathrm{N}_{\mathrm{r}}$-by- $\mathrm{N}_{\mathrm{r}}$. The main diagonal elements must be all ones. The off-diagonal elements must be real or complex numbers with a magnitude smaller than or equal to one.

If the channel is frequency selective, i.e., PathDelays is a row vector of length $\mathrm{N}_{\mathrm{p}}$, you can specify ReceiveCorrelationMatrix as an $\mathrm{N}_{\mathrm{r}}$-by- $\mathrm{N}_{\mathrm{r}}$ matrix. In this case, each path has the same receive spatial correlation matrix. Alternatively, you can specify the value as a 3 D array of size $\mathrm{N}_{\mathrm{r}}$-by- $\mathrm{N}_{\mathrm{r}}$-by- $\mathrm{N}_{\mathrm{p}}$. In this case, each path can have its own receive spatial correlation matrix.

## AntennaSelection

Optional transmit and/or receive antenna selection
Specify the antenna selection scheme as one of Off | Tx|Rx|Tx and $R x$. The default value of this property is Off.

Tx represents transmit antennas and Rx represents receive antennas. When you configure any antenna selection other than the default setting, the object requires one or more inputs to specify which antennas are selected for signal transmission.

## FadingDistribution

Rayleigh or Rician fading
Specify the fading distribution of the channel as one of Rayleigh| Rician. The default value of this property is Rayleigh, i.e., the channel is Rayleigh fading.

## KFactor

Rician K-factor scalar or vector (linear scale)
Specify the K factor of a Rician fading channel as a double-precision, real, positive scalar or nonnegative, nonzero row vector of the same length as PathDelays. This property applies when you set the FadingDistribution property to Rician. The default value of this property is 3.

If KFactor is a scalar, the first discrete path is a Rician fading process with a Rician K factor of KFactor. The remaining discrete paths are independent Rayleigh fading processes. If KFactor is a row vector, the discrete path corresponding to a positive element of the KFactor vector is a Rician fading process with a Rician K factor specified by that element. The discrete path corresponding to a zero-valued element of the KFactor vector is a Rayleigh fading process.

## DirectPathDopplerShift

Doppler shift(s) of line-of-sight component(s) (Hertz)

Specify the Doppler shifts for the line-of-sight components of a Rician fading channel in hertz as a double-precision, real scalar or row vector. The default value of this property is 0 . This property applies when you set the FadingDistribution property to Rician.

DirectPathDopplerShift must have the same size as KFactor. If DirectPathDopplerShift is a scalar, this value represents the line-of-sight component Doppler shift of the first discrete path. This path exhibits a Rician fading process. If DirectPathDopplerShift and KFactor are row vectors, the discrete path corresponding to a positive element of the KFactor vector is a Rician fading process. Its line-of-sight component Doppler shift is specified by the corresponding element of DirectPathDopplerShift.

## DirectPathInitialPhase

Initial phase(s) of line-of-sight component(s) (radians)
Specify the initial phases of the line-of-sight components of a Rician fading channel in radians as a double precision, real scalar or row vector. The default value of this property is 0 . This property applies when you set the FadingDistribution property to Rician.

DirectPathInitialPhase must have the same size as KFactor. If DirectPathInitialPhase is a scalar, this value represents the line-of-sight component initial phase of the first discrete path. This path exhibits a Rician fading process. If DirectPathInitialPhase and KFactor are row vectors, the discrete path corresponding to a positive element of the KFactor vector is a Rician fading process. Its line-of-sight component initial phase is specified by the corresponding element of DirectPathInitialPhase.

## RandomStream

Source of random number stream

Specify the source of random number stream as one of Global stream | mt19937ar with seed. The default value of this property is Global stream. If you set RandomStream to Global stream, the current global random number stream is used for normally distributed random number generation. In this case, the reset method only resets the filters. If you set RandomStream to mt19937ar with seed, the mt19937ar algorithm is used for normally distributed random number generation. In this case, the reset method not only resets the filters but also reinitializes the random number stream to the value of the Seed property.

## Seed

Initial seed of mt19937ar random number stream
Specify the initial seed of a mt19937ar random number generator algorithm as a double precision, real, nonnegative integer scalar. The default value of this property is 73 . This property applies when you set the RandomStream property to mt19937ar with seed. The Seed reinitializes the mt19937ar random number stream in the reset method.

## NormalizePathGains

Normalize path gains (logical)
Set this property to true to normalize the fading processes such that the total power of the path gains, averaged over time, is 0 dB . The default value of this property is true. When you set this property to false, there is no normalization on path gains. The average powers of the path gains are specified by the AveragePathGains property.

## NormalizeChannelOutputs

Normalize channel outputs (logical)
Set this property to true to normalize the channel outputs by the number of receive antennas. The default value of this property is true. When you set this property to false, there is no normalization for channel outputs.

## PathGainsOutputPort

Enable path gain output (logical)
Set this property to true to output the channel path gains of the underlying fading process. The default value of this property is false.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>reset<br>step

Create object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of the object
Filter input signal through MIMO multipath fading channel

## Examples Examine Spatial Correlation Characteristics of a 2-by-2 Rayleigh Fading Channel

Filter PSK modulated data through a 2-by-2 Rayleigh fading channel, and then examine the spatial correlation characteristics of the channel realization.

Create a PSK Modulator System object to modulate randomly generated data.
hMod = comm.PSKModulator;
modData $=$ step(hMod, randi([0 hMod.ModulationOrder-1],2e6,1));

Split modulated data into two spatial streams.

```
channelInput = reshape(modData, [2, 1e6]).';
```

Create a 2-by-2 MIMOChannel System object with two discrete paths. Each path has different transmit and receive correlation matrices, specified by the TransmitCorrelationMatrix and ReceiveCorrelationMatrix properties.

```
hMIMOChan = comm.MIMOChannel(...
    'SampleRate', 1000,...
    'PathDelays', [0, 1e-3],...
    'AveragePathGains', [3, 5],...
    'MaximumDopplerShift', 5,...
    'TransmitCorrelationMatrix', cat(3, eye(2), [1 0.1;0.1 1]),...
    'ReceiveCorrelationMatrix', cat(3, [1 0.2;0.2 1], eye(2)),...
    'RandomStream', 'mt19937ar with seed',...
    'Seed', 33,...
    'NormalizePathGains', false,...
    'PathGainsOutputPort', true);
```

Filter the modulated data using hMIMOChan and use the step method to process data.

```
[channelOutput, pathGains] = step(hMIMOChan, channelInput);
```

The transmit spatial correlation for the first discrete path at the first receive antenna is specified as an identity matrix in the TransmitCorrelationMatrix property. Confirm that the channel output pathGains exhibits the same statistical characteristics using the corrcoef function.

```
disp('Tx spatial correlation, first path, first Rx:');
disp(corrcoef(squeeze(pathGains(:,1,:,1))));
```

The transmit spatial correlation for the second discrete path at the second receive antenna is specified as [10.1;0.11] in the

TransmitCorrelationMatrix property. Confirm that the channel output pathGains exhibits the same statistical characteristics.

```
disp('Tx spatial correlation, second path, second Rx:');
disp(corrcoef(squeeze(pathGains(:,2,:,2))));
```

The receive spatial correlation for the first discrete path at the second transmit antenna is specified as [10.2;0.2 1] in the ReceiveCorrelationMatrix property. Confirm that the channel output pathGains exhibits the same statistical characteristics.

```
disp('Rx spatial correlation, first path, second Tx:');
disp(corrcoef(squeeze(pathGains(:,1,2,:))));
```

The receive spatial correlation for the second discrete path at the first transmit antenna is specified as an identity matrix in the ReceiveCorrelationMatrix property. Confirm that the channel output pathGains exhibits the same statistical characteristics.

```
disp('Rx spatial correlation, second path, first Tx:');
disp(corrcoef(squeeze(pathGains(:,2,1,:))));
```

Now enable transmit and receive antenna selection for the System object hMIMOChan. The input frame size is shortened to 100 .

```
release(hMIMOChan);
hMIMOChan.AntennaSelection = 'Tx and Rx';
modData = step(hMod,randi([0 hMod.ModulationOrder-1],1e2,1));
```

Select the first transmit and second receive antennas.
[channelOutput, pathGains] = step(hMIMOChan, modData, [1 0], [0 1]);
Confirm that the path gains MATLAB returns have NaN values for the unselected transmit-receive antenna pairs.

```
disp('Return 1 if the path gains for the second transmit antenna are Na\
disp(isequal(isnan(squeeze(pathGains(:,:,2,:))), ones(1e2, 2, 2)));
disp('Return 1 if the path gains for the first receive antenna are NaN:'
```

```
disp(isequal(isnan(squeeze(pathGains(:,:,:,1))), ones(1e2, 2, 2)));
```


## Algorithms

The fading processing per link is per the Methodology for Simulating Multipath Fading Channels section and assumes the same parameters for all $N_{\mathrm{T}} \cdot N_{\mathrm{R}}$ links of the MIMO channel. Each link comprises all multipaths for that link.

## The Kronecker Model

The Kronecker model assumes that the spatial correlations at the transmit and receive sides are separable. Equivalently, the direction of departure (DoD) and directions of arrival (DoA) spectra are assumed to be separable. The full correlation matrix can then be obtained as:

$$
R_{H}=E\left[R_{t} \otimes R_{r}\right]
$$

where:
The $\otimes$ symbol represents the Kronecker product.
$R_{\mathrm{t}}$ represents the correlation matrix at the transmit side, i.e.
$R_{t}=E\left\lfloor H^{H} H\right\rfloor$, of size $N_{\mathrm{t}}$-by- $N_{\mathrm{t}}$.
$R_{\mathrm{r}}$ represents the correlation matrix at the receive side, i.e.

$$
R_{r}=E\left[H H^{H}\right], \text { of size } N_{\mathrm{r}} \text {-by- } N_{\mathrm{r}}
$$

You can obtain a realization of the MIMO channel matrix as:

$$
H=R_{r}^{\frac{1}{2}} A R_{t}^{\frac{1}{2}}
$$

where: $A=$ unvec(a) is an $N_{\mathrm{r}}$-by- $N_{\mathrm{t}}$ matrix of i.i.d. complex Gaussian variables with zero mean and unit variance.

Note that the $\otimes$ symbol represents the Kronecker product.

## Cutoff Frequency Factor

The following information explains how this objects determines the cutoff frequency factor, $f_{c}$ :

- For any Doppler spectrum type, other than Gaussian and BiGaussian, $\mathrm{f}_{\mathrm{c}}$ equals 1 .
- For a Gaussian Doppler spectrum type, $\mathrm{f}_{\mathrm{c}}$ equals the Doppler spectrum object SigmaGaussian property value times sqrt( $2 \cdot \log (2)$ ).
- For a BiGaussian Doppler spectrum type:
- If the Doppler spectrum object GainGaussian1 (GainGaussian2) and CenterFreqGaussian2 (CenterFreqGaussian1) property values are both 0 , then $f_{c}$ equals the SigmaGaussian2 (SigmaGaussian1) property value times sqrt( $2 \cdot \log (2)$ ).
- If the CenterFreqGaussian1 and CenterFreqGaussian2 property values are both 0 and the SigmaGaussian1 and SigmaGaussian2 property values are the same, then $f_{c}$ is also equal to the SigmaGaussian2 property value times sqrt( $2 \cdot \log (2)$ ).
- In all other cases, $\mathrm{f}_{\mathrm{c}}$ equals 1.


## Antenna Selection

When the object is in antenna selection mode, it uses the following algorithms to process an input signal:

- The random path gains are always generated and keep evolving for each link, no matter whether the links is being selected or not. The path gain values for the non-selected links are marked as NaN in the path gain output.
- The spatial correlation only applies to the selected transmit and/or receive antennas, and the correlation coefficients are the corresponding entries in the transmit and/or receive correlation matrices. In other words, the spatial correlation matrix for the selected transmit/receive antennas is a submatrix of the

TransmitCorrelationMatrix/ReceiveCorrelationMatrix property value.

- The input filtering through the path gains only happens to the selected links. Channel output normalization happens over the number of selected receive antennas.


## Selected Bibliography

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[3] Kermoal, J. P., L. Schumacher, K. I. Pedersen, P. E. Mogensen, and F. Frederiksen. "A stochastic MIMO radio channel model with experimental validation." IEEE Journal on Selected Areas of Communications. Vol. 20, Number 6, 2002, pp. 1211-1226.
[4] Jeruchim, M., P. Balaban, and K. S. Shanmugan. Simulation of Communication Systems, Second Edition, New York, Kluwer Academic/Plenum, 2000.

See Also comm.LTEMIMOChannel

Purpose Create MIMOChannel object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a MIMOChannel object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

## Purpose Number of expected inputs to step method

Syntax
$N$ = getNumInputs( H )

Description
$N=$ getNumInputs $(H)$ returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ )

The getNumInputs method returns a positive integer that is the number of expected inputs (not counting the object itself) to the step method. This value will change if you alter any properties that turn inputs on or off. You must call the step method with the number of input arguments equal to the result of getNumInputs $(H)$.

## comm.MIMOChannel.getNumOutputs

Purpose Number of outputs from step method

## Syntax

Description
$N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

The getNumOutputs method returns a positive integer that is the number of outputs from the step method. This value will change if you alter any properties that turn outputs on or off.

| Purpose | Locked status for input attributes and nontunable properties |
| :---: | :---: |
| Syntax | TF = isLocked(H) |
| Description | TF = isLocked (H) returns the locked status, TF of the MIMOChannel System object. |
|  | The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value. |

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release (H) releases system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose Reset states of the MIMOChannel object

## Syntax reset (H)

Description reset $(H)$ resets the states of the MIMOChannel object, $H$.
If you set the RandomStream property of H to Global stream, the reset method only resets the filters. If you set RandomStream to mt19937ar with seed, the reset method not only resets the filters but also reinitializes the random number stream to the value of the Seed property.

Purpose
Filter input signal through MIMO multipath fading channel
Syntax

```
Y = step(H,X)
Y = step(H,X,SELTX)
Y = step(H,X,SELRX)
Y = step(H,X,SELTX,SELRX)
[Y,PATHGAINS] = step(H,X)
[Y,PATHGAINS] = step(H,X,SELTX/SELRX)
step(H,X,SELTX,SELRX)
```


## Description

$Y=\operatorname{step}(H, X)$ filters input signal $X$ through a MIMO fading channel and returns the result in $Y$. The input $X$ can be a double-precision data type scalar, vector, or 2-D matrix with real or complex values. X is of size $\mathrm{N}_{\mathrm{s}}-$ by $-\mathrm{N}_{\mathrm{t}}$, where $\mathrm{N}_{\mathrm{s}}$ represents the number of samples and $\mathrm{N}_{\mathrm{t}}$ represents the number of transmit antennas that is determined by the TransmitCorrelationMatrix property value of H . Y is the output signal of size $\mathrm{N}_{\mathrm{s}}$-by- $\mathrm{N}_{\mathrm{r}}$, where $\mathrm{N}_{\mathrm{r}}$ represents the number of receive antennas that is determined by the ReceiveCorrelationMatrix property value of H . Y is of double-precision data type with complex values.
$Y=\operatorname{step}(H, X$, SELTX) turns on selected transmit antennas for $X$ transmission. This syntax applies when you set the AntennaSelection property of H to Tx. SELTX is a numeric type binary-valued 1-by- $\mathrm{N}_{\mathrm{t}}$ row vector. In this row vector, the ones indicate the selected transmit antennas. X is size $\mathrm{N}_{\mathrm{s}}$-by- $\mathrm{N}_{\text {st }}$, where $\mathrm{N}_{\text {st }}$ represents the number of selected transmit antennas, i.e., the number of ones in SELTX. $Y$ is size $N_{s}-$ by $-N_{r}$.
$Y=\operatorname{step}(H, X, S E L R X)$ turns on selected receive antennas for $X$ transmission. This syntax applies when you set the AntennaSelection property of H to Rx . SELRX is a numeric type binary-valued 1-by- $\mathrm{N}_{\mathrm{r}}$ row vector, in which the ones indicate the selected receive antennas. $X$ is of size $N_{s}$-by $-N_{t}$. $Y$ is of size $N_{s}$-by- $N_{s r}$, where $N_{s r}$ represents the number of selected receive antennas, i.e., the number of ones in SELRX.
$Y=$ step ( $H, X$, SELTX, SELRX) turns on selected transmit and receive antennas for $X$ transmission. This syntax applies when you set the AntennaSelection property of H to Tx and Rx . X is of size $\mathrm{N}_{\mathrm{s}}$-by- $\mathrm{N}_{\mathrm{st}}$, and $Y$ is of size $\mathrm{N}_{\mathrm{s}}$-by $-\mathrm{N}_{\mathrm{sr}}$.
[ Y, PATHGAINS] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$ returns the MIMO channel path gains of the underlying fading process in PATHGAINS. This syntax applies when you set the PathGainsOutputPort property of $H$ to true. PATHGAINS is of size $\mathrm{N}_{\mathrm{s}}$-by- $\mathrm{N}_{\mathrm{p}}$-by- $\mathrm{N}_{\mathrm{t}}$-by- $\mathrm{N}_{\mathrm{r}}$, where $\mathrm{N}_{\mathrm{p}}$ represents the number of paths, i.e., the length of the PathDelays property value of $H$. PATHGAINS is of double-precision data type with complex values.

$$
\begin{aligned}
& {[\mathrm{Y}, \text { PATHGAINS }]=\mathrm{step}(\mathrm{H}, \mathrm{X}, \text { SELTX/SELRX }) \text { or }} \\
& \text { step }(\mathrm{H}, \mathrm{X}, \text { SELTX, SELRX) returns the MIMO channel path } \\
& \text { gains for antenna selection schemes. PATHGAINS is still of } \\
& \text { size } \mathrm{N}_{\mathrm{s}} \text {-by- } \mathrm{N}_{\mathrm{p}} \text {-by- } \mathrm{N}_{\mathrm{t}} \text {-by- } \mathrm{N}_{\mathrm{r}} \text { with NaN values for the unselected } \\
& \text { transmit-receive antenna pairs. }
\end{aligned}
$$

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Equalize using maximum likelihood sequence estimation

Description

Construction

The MLSEEqualizer object uses the Viterbi algorithm to equalize a linearly modulated signal through a dispersive channel. The object processes input frames and outputs the maximum likelihood sequence estimate (MLSE) of the signal. This processing uses an estimate of the channel modeled as a finite impulse response (FIR) filter.

H = comm.MLSEEqualizer creates a maximum likelihood sequence estimation equalizer (MLSEE) System object, H. This object uses the Viterbi algorithm and a channel estimate to equalize a linearly modulated signal that has been transmitted through a dispersive channel.

H = comm.MLSEEqualizer (Name, Value) creates an MLSEE object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.MLSEEqualizer(CHANNEL, Name, Value) creates an MLSEE object, H. This object has the Channel property set to CHANNEL, and the other specified properties set to the specified values.

## Properties

## ChannelSource

Source of channel coefficients
Specify the source of the channel coefficients as one of Input port | Property. The default is Property.

## Channel

Channel coefficients
Specify the channel as a numeric, column vector containing the coefficients of an FIR filter. The default is $[1 ; 0.7 ; 0.5 ; 0.3]$. The length of this vector determines the memory length of the channel. This must be a multiple of the samples per symbol, that you specify in the SamplesPerSymbol property. This property applies when you set the ChannelSource property to Property.

## Constellation

Input signal constellation
Specify the constellation of the input modulated signal as a complex vector. The default is $[1+1 \mathrm{i}-1+1 \mathrm{i}-1-1 \mathrm{i}$ 1-1i].

## TracebackDepth

Traceback depth of Viterbi algorithm
Specify the number of trellis branches (the number of symbols), the Viterbi algorithm uses to construct each traceback path. The default is 21 . The traceback depth influences the decoding accuracy and delay. The decoding delay represents the number of zero symbols that precede the first decoded symbol in the output. When you set the TerminationMethod property to Continuous, the decoding delay equals the number of zero symbols of this property. When you set the TerminationMethod property to Truncated, there is no output delay.

## TerminationMethod

Termination method of Viterbi algorithm
Specify the termination method of the Viterbi algorithm as one of Continuous | Truncated. The default is Truncated. When you set this property to Continuous, the object initializes the Viterbi algorithm metrics of all the states to 0 in the first call to the step method. Then, the object saves its internal state metric at the end of each frame, for use with the next frame. When you set this property to Truncated, the object resets at every frame. The Viterbi algorithm processes each frame of data independently, resetting the state metric at the end of each frame. The traceback path always starts at the state with the minimum metric. The initialization of the state metrics depends on whether you specify a preamble or postamble. If you set the PreambleSource property to None, the object initializes the metrics of all the states to 0 at the beginning of each data frame. If you set the PreambleSource property to Property, the object uses the preamble that you specify at the Preamble property, to initialize the state metrics at
the beginning of each data frame. When you specify a preamble, the traceback path ends at one of the states represented by that preamble. If you set the PostambleSource property to None, the traceback path starts at the state with the smallest metric. If you set the PostambleSource property to Property, the traceback path begins at the state represented by the postamble that you specify at the Postamble property. If the postamble does not decode to a unique state, the decoder identifies the smallest of all possible decoded states that are represented by the postamble. The decoder then begins traceback decoding at that state. When you set this property to Truncated, the step method input data signal must contain at least TracebackDepth symbols, not including an optional preamble.

## ResetInputPort

Enable equalizer reset input
Set this property to true to enable an additional input to the step method. The default is false. When this input is a nonzero, double-precision or logical scalar value, the object resets the states of the equalizer. This property applies when you set the TerminationMethod property to Continuous.

## PreambleSource

## Source of preamble

Specify the source of the preamble that is expected to precede the input signal. Choose from None | Property. The default is None. Set this property to Property to specify a preamble using the Preamble property. This property applies when you set the TerminationMethod property to Truncated.

## Preamble

Preamble that precedes input signals
Specify a preamble that is expected to precede the data in the input signal as an integer, row vector. The default is [0 32 1 ]. The values of the preamble should be between 0 and $M-1$,
where $M$ is the length of the signal constellation that you specify in the Constellation property. An integer value of $k-1$ in the vector corresponds to the $k$-th entry in the vector stored in the Constellation property. This property applies when you set the TerminationMethod property to Truncated and the PreambleSource property to Property.

## PostambleSource

Source of postamble
Specify the source of the postamble that is expected to follow the input signal. Choose from None | Property. The default is None. Set this property to Property to specify a postamble in the Postamble property. This property applies when you set the TerminationMethod property to Truncated.

## Postamble

Postamble that follows input signals
Specify a postamble that is expected to follow the data in the input signal as an integer row vector. The default is [ $\left.\begin{array}{llll}0 & 2 & 3 & 1\end{array}\right]$. The values of the postamble should be between 0 and $M-1$. In this case, $M$ indicates the length of the Constellation property. An integer value of $k-1$ in the vector corresponds to the $k$-th entry in the vector specified in the Constellation property. This property applies when you set the TerminationMethod property to Truncated and the PostambleSource property to Property. The default is [llll 0231$]$.

## SamplesPerSymbol

Number of samples per symbol
Specify the number of samples per symbol in the input signal as an integer scalar value. The default is 1 .

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>reset<br>step

Create MLSEE object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of MLSEE object
Equalize using maximum likelihood sequence estimation

## Examples

Equalize a QPSK signal transmitted through a dispersive channel.

```
    hMod = comm.QPSKModulator(0,'SymbolMapping','Binary');
    hDemod = comm.QPSKDemodulator(0,'SymbolMapping','Binary');
% Channel coefficients
    chCoeffs = [.986; .845; .237; .12345+.31i];
    hMLSEE = comm.MLSEEqualizer('TracebackDepth',10,...
                            'Channel',chCoeffs, 'Constellation',[11 1i -1 -1i]);
% Create an error rate calculator
    hError = comm.ErrorRate;
    for n = 1:50
        data= randi([0 3],100,1);
        modSignal = step(hMod, data);
% Introduce channel distortion.
            chanOutput = filter(chCoeffs,1,modSignal);
% Equalize the channel output and demodulate
    eqSignal = step(hMLSEE,chanOutput);
```

```
    demodData = step(hDemod,eqSignal);
% Compute BER
    errorStats = step(hError, data, demodData);
    end
    fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```


# Algorithms This object implements the algorithm, inputs, and outputs described on the MLSE Equalizer block reference page. The object properties correspond to the block parameters. 

See Also comm.ViterbiDecoder

Purpose Create MLSEE object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a MLSEEqualizer object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

Purpose $\quad$ Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ )

## comm.MLSEEqualizer.getNumOutputs

Purpose $\quad$ Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose
Syntax
Description

Locked status for input attributes and nontunable properties
TF = isLocked(H)
TF = isLocked (H) returns the locked status, TF of the MLSEEqualizer System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of MLSEE object
Syntax $\quad \operatorname{reset}(H)$

Description reset $(H)$ resets the states of the MLSEEqualizer object, $H$.

## Purpose <br> Equalize using maximum likelihood sequence estimation

Syntax<br>\section*{Description}

Y = step( $\mathrm{H}, \mathrm{X}$ )
Y = step(H,X,CHANNEL)
Y = step(H,X,RESET)
Y = step( $\mathrm{H}, \mathrm{X}, \mathrm{CHANNEL}$, RESET $)$
$Y=\operatorname{step}(H, X)$ equalizes the linearly modulated data input, $X$, using the Viterbi algorithm. The step method outputs $Y$, the maximum likelihood sequence estimate of the signal. Input X must be a column vector of data type double or single.
$\mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X}, \mathrm{CHANNEL})$ uses CHANNEL as the channel coefficients when you set the ChannelSource property to 'Input port'. The channel coefficients input, CHANNEL, must be a numeric, column vector containing the coefficients of an FIR filter in descending order of powers of z . The length of this vector is the channel memory, which must be a multiple of the samples per input symbol specified in the SamplesPerSymbol property.

Y = step (H,X,RESET) uses RESET as the reset signal when you set the TerminationMethod property to 'Continuous' and the ResetInputPort property to true. The object resets when RESET has a non-zero value. RESET must be a double precision or logical scalar. You can combine optional input arguments when you set their enabling properties. Optional inputs must be listed in the same order as the order of the enabling properties. For example,

```
Y = step(H,X,CHANNEL,RESET)
```


#### Abstract

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.


Purpose Demodulate using MSK method and the Viterbi algorithm
Construction
Properties

The MSKDemodulator object demodulates a signal that was modulated using the minimum shift keying method. The input is a baseband representation of the modulated signal. The initial phase offset property sets the initial phase of the modulated waveform.

H = comm.MSKDemodulator creates a demodulator System object, H. This object demodulates the input minimum shift keying (MSK) modulated data using the Viterbi algorithm.

H = comm.MSKDemodulator(Name, Value) creates an MSK demodulator object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## BitOutput

Output data as bits
Specify whether the output consists of groups of bits or integer values. The default is false.

When you set this property to false, the step method outputs a column vector with a length equal to $N /$ SamplesPerSymbol. $N$ represents the length of the input signal, which is the number of input baseband modulated symbols. The elements of the output vector are -1 or 1 .

When you set the BitOutput property to true, the step method outputs a binary column vector with a length equal to $N /$ SamplesPerSymbol. The vector elements are bit values of 0 or 1.

## InitialPhaseOffset

Initial phase offset
Specify the initial phase offset of the input modulated waveform in radians as a real, numeric scalar value. The default is 0 .

## SamplesPerSymbol

Number of samples per input symbol
Specify the expected number of samples per input symbol as a positive, integer scalar value. The default is 8 .

## TracebackDepth

Traceback depth for Viterbi algorithm
Specify the number of trellis branches that the Viterbi algorithm uses to construct each traceback path as a positive, integer scalar value. The default is 16 . The value of this property is also the output delay This value indicates number of zero symbols that precede the first meaningful demodulated symbol in the output.

## OutputDataType

Data type of output
Specify the output data type as one of int8 | int16 | int32 | double, when you set the BitOutput property to false. The default is double.

When you set the BitOutput property to true, specify the output data type as one of logical | double.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release

Create MSK demodulator object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
reset
step

> Reset states of the MSK demodulator object
> Demodulate using MSK method and the Viterbi algorithm

## Examples

Modulate and demodulate a signal using MSK modulation with bit inputs and an initial phase offset of $\mathrm{pi} / 4$.
hMod $=$ comm.MSKModulator('BitInput', true, 'InitialPhaseOffset', pi/4 hAWGN $=$ comm.AWGNChannel('NoiseMethod', ...
'Signal to noise ratio (SNR)','SNR',0);
hDemod $=$ comm.MSKDemodulator('BitOutput', true, ...
'InitialPhaseOffset', pi/4);
\% Create an error rate calculator, account for the delay caused by the Vi hError = comm.ErrorRate('ReceiveDelay', hDemod.TracebackDepth);
for counter = 1:100
\% Transmit 100 3-bit words data $=$ randi([0 1], 300, 1);
modSignal $=$ step (hMod, data);
noisySignal = step(hAWGN, modSignal);
receivedData = step(hDemod, noisySignal);
errorStats $=$ step(hError, data, receivedData);
end
fprintf('Error rate $=\% f \backslash n N u m b e r$ of errors $=\% d \backslash n ', \ldots$ errorStats(1), errorStats(2))

This object implements the algorithm, inputs, and outputs described on the MSK Demodulator Baseband block reference page. The object properties correspond to the block parameters.

See Also

# Purpose <br> Create MSK demodulator object with same property values 

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a MSKDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.MSKDemodulator.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.MSKDemodulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the MSKDemodulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.MSKDemodulator.release

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.MSKDemodulator.reset

Purpose Reset states of the MSK demodulator object

## Syntax reset (H)

Description reset (H) resets the states of the MSKDemodulator object, H.

# Purpose <br> Demodulate using MSK method and the Viterbi algorithm 

Syntax $\quad Y=\operatorname{step}(H, X)$
Description
$Y=\operatorname{step}(H, X)$ demodulates input data, $X$, with the MSK demodulator System object, H, and returns Y. X must be a double or single precision column vector with a length equal to an integer multiple of the number of samples per symbol you specify in the SamplesPerSymbol property.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Modulate using MSK method

Description

## Construction

The MSKModulator object modulates using the minimum shift keying method. The output is a baseband representation of the modulated signal. The initial phase offset property sets the initial phase of the output waveform, measured in radians.

H = comm.MSKModulator creates a modulator System object, H. This object modulates the input signal using the minimum shift keying (MSK) modulation method.

H = comm.MSKModulator(Name, Value) creates an MSK modulator object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties <br> BitInput

Assume bit inputs
Specify whether the input is bits or integers. The default is false.
When you set the BitInput property to false, the step method input must be a column vector with a double-precision or signed integer data type and of values equal to -1 or 1 .

When you set the BitInput property to true, the step method input requires double-precision or logical data type column vector of 0 s and 1 s .

## InitialPhaseOffset

Initial phase offset
Specify the initial phase of the modulated waveform in radians as a real, numeric scalar value. The default is 0 .

## SamplesPerSymbol

Number of samples per output symbol

Specify the upsampling factor at the output as a real, positive, integer scalar value. The default is 8 . The upsampling factor indicates the number of output samples that the step method produces for each input sample.

## OutputDataType

Data type of output
Specify output data type as one of double | single. The default is double.

| Methods | clone | Create MSK modulator object with same property values |
| :---: | :---: | :---: |
|  | getNumInputs | Number of expected inputs to step method |
|  | getNumOutputs | Number of outputs from step method |
|  | isLocked | Locked status for input attributes and nontunable properties |
|  | release | Allow property value and input characteristics changes |
|  | reset | Reset states of the MSK modulator object |
|  | step | Modulate using MSK method |
| Examples | Modulate and demodulate a signal using MSK modulation with bit inputs and an initial phase offset of $\mathrm{pi} / 4$ |  |
|  | $\begin{array}{r} \text { hMod = comm.MSKModulator('BitInput', true, } \ldots \\ \text { 'InitialPhaseOffset', pi/4); } \\ \text { hAWGN = comm.AWGNChannel('NoiseMethod', ... } \end{array}$ |  |

```
                    'InitialPhaseOffset', pi/4);
% Create an error rate calculator, account for the delay caused by the
    hError = comm.ErrorRate('ReceiveDelay', hDemod.TracebackDepth);
    for counter = 1:100
        % Transmit 100 3-bit words
        data = randi([0 1],300,1);
        modSignal = step(hMod, data);
        noisySignal = step(hAWGN, modSignal);
        receivedData = step(hDemod, noisySignal);
        errorStats = step(hError, data, receivedData);
    end
    fprintf('Error rate = %f\nNumber of errors = %d\n', ...
        errorStats(1), errorStats(2))
```

Algorithms This object implements the algorithm, inputs, and outputs described on the MSK Demodulator Baseband block reference page. The object properties correspond to the block parameters.
comm.MSKDemodulator | comm.CPMModulator | comm.CPMDemodulator

## Purpose Create MSK modulator object with same property values

## Syntax <br> C = clone(H)

Description
C = clone (H) creates a MSKModulator object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.MSKModulator.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNuminputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the MSKModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.MSKModulator.reset

Purpose Reset states of the MSK modulator object

## Syntax reset (H)

Description reset (H) resets the states of the MSKModulator object, H.

## Purpose <br> Modulate using MSK method

## Syntax <br> Y = step (H,X)

Description
$\mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X})$ modulates input data, X , with the MSK modulator object, $H$. It returns the baseband modulated output, Y. Depending on the value of the BitInput property, input $X$ can be a double precision, signed integer, or logical column vector. The length of output vector, Y , is equal to the number of input samples times the number of samples per symbol you specify in the SamplesPerSymbol property.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Recover symbol timing phase using fourth-order nonlinearity method |
| :---: | :---: |
| Description | The MSKTimingSynchronizer object recovers the symbol timing phase of the input signal using a fourth-order nonlinearity method. This object implements a general non-data-aided feedback method that is independent of carrier phase recovery. This method requires prior compensation for the carrier frequency offset. This object is suitable for systems that use baseband minimum shift keying (MSK) modulation. |
| Construction | H = comm.MSKTimingSynchronizer creates a timing phase synchronizer System object, H. This object recovers the symbol timing phase of the input signal using a fourth-order nonlinearity method. <br> H = comm.MSKTimingSynchronizer (Name, Value) creates an MSK timing synchronizer object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN). |
| Properties | SamplesPerSymbol |
|  | Number of samples representing each symbol |
|  | Specify the number of samples that represent each symbol in the input signal as an integer-valued scalar greater than 1 . The default is 4. |
|  | ErrorUpdateGain |
|  | Error update step size |
|  | Specify the step size for updating successive timing phase estimates as a positive, real scalar value. The default is 0.05 . Typically, this number is less than $1 /$ SamplesPerSymbol, which corresponds to a slowly varying timing phase. This property is tunable. |
|  | ResetInputPort |

Enable synchronization reset input

Set this property to true to enable resetting the timing phase recovery process based on an input argument value. The default is false.

When you set this property to true, you must specify a reset input value to the step method.

When the reset input is a nonzero value, the object restarts the timing phase recovery process. When you set this property to false, the object does not restart.

## ResetCondition

Condition for timing phase recovery reset
Specify the conditions to reset the timing phase recovery process as one of Never I Every frame. The default is Never.

When you set this property to Never, the phase recovery process never restarts. The object operates continuously, retaining information from one symbol to the next.

When you set this property to Every frame, the timing phase recovery restarts at the start of each frame of data. Thus, each time the object calls the step method. This property applies when you set the ResetInputPort property to false.

## Methods

clone<br>getNumInputs<br>getNumOutputs<br>isLocked

Create MSK timing phase synchronizer object with same property values
Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties
release
reset
step

Allow property value and input characteristics changes
Reset states of MSK timing phase synchronizer object

Recover symbol timing phase using fourth-order nonlinearity method

## Examples Recover timing phase of an MSK signal.

```
% Create System objects
    hMod = comm.MSKModulator('BitInput', true,...
        'SamplesPerSymbol', 14);
    timingOffset = 0.2; % Actual timing offset
    hDelay = dsp.VariableFractionalDelay;
    hSync = comm.MSKTimingSynchronizer('SamplesPerSymbol', 14, ...
        'ErrorUpdateGain', 0.05);
    phEst = zeros(1, 10);
    for i = 1:51
        data = randi([0 1], 100, 1); % generate data
        modData = step(hMod, data); % modulate data
% data impaired by timing offset error
        impairedData = step(hDelay, modData, timingOffset*14);
        % perform timing phase recovery
        [y, phase] = step(hSync, impairedData);
        phEst(i) = phase(1)/14;
    end
    figure, plot(0.2*ones(1, 50));
    hold on; ylim([0 0.4])
    plot(phEst, 'r'); legend( 'original', 'estimated')
    title('Original and Estimated timing phases');
```

Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the MSK-Type Signal Timing Recovery block reference page. The object properties correspond to the block parameters, except:

- The object corresponds to the MSK-Type Signal Timing Recovery block with the Modulation type parameter set to MSK.
- The Reset parameter corresponds to the ResetInputPort and ResetCondition properties.
comm.EarlyLateGateTimingSynchronizer |
comm.MuellerMullerTimingSynchronizer


## comm.MSKTimingSynchronizer.clone

Purpose Create MSK timing phase synchronizer object with same property values

## Syntax <br> C = clone(H)

Description
$C=$ clone $(H)$ creates a MSKTimingSynchronizer object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.MSKTimingSynchronizer.getNumInputs

## Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $N=$ getNumInputs $(H)$ method returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ ).

## comm.MSKTimingSynchronizer.getNumOutputs

Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.MSKTimingSynchronizer.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked $(H)$ |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the <br> MSKTimingSynchronizer System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.MSKTimingSynchronizer.release

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset states of MSK timing phase synchronizer object

## Syntax $\quad \operatorname{reset}(H)$

Description reset (H) resets the states of MSKTimingSynchronizer object, H.

Purpose
Recover symbol timing phase using fourth-order nonlinearity method

## Syntax

[Y,PHASE] = step(H,X)
[Y,PHASE] = step(H,X,R)
[ $\mathrm{Y}, \mathrm{PHASE}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$ recovers the timing phase and returns the time-synchronized signal, $Y$, and the estimated timing phase, PHASE, for input signal $X$. X must be a double or single precision complex column vector.
[ $\mathrm{Y}, \mathrm{PHASE}$ ] = $\operatorname{step}(\mathrm{H}, \mathrm{X}, \mathrm{R})$ restarts the timing phase recovery process when you input a reset signal, $R$, that is non-zero. $R$ must be a logical or double scalar. This syntax applies when you set the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Recover symbol timing phase using Mueller-Muller method |
| :---: | :---: |
| Description | The MuellerMullerTimingSynchronizer object recovers the symbol timing phase of the input signal using the Mueller-Muller method. This object implements a decision-directed, data-aided feedback method that requires prior recovery of the carrier phase. |
| Construction | H = comm.MuellerMullerTimingSynchronizer creates a timing synchronizer System object, H. This object recovers the symbol timing phase of the input signal using the Mueller-Muller method. |
|  | H = comm.MuellerMullerTimingSynchronizer(Name, Value) creates a Mueller-Muller timing recovery object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN). |
| Properties | SamplesPerSymbol |
|  | Number of samples representing each symbol |
|  | Specify the number of samples that represent each symbol in the input signal as an integer-valued scalar greater than 1. The default is 4 . |
|  | ErrorUpdateGain |
|  | Error update step size |
|  | Specify the step size for updating successive timing phase estimates as a positive real scalar value. The default is 0.05 . Typically, this number is less than 1 /SamplesPerSymbol, which corresponds to a slowly varying timing phase. This property is tunable. |
|  | ResetInputPort |
|  | Enable synchronization reset input |
|  | Set this property to true to enable resetting the timing phase recovery process based on an input argument value. The default is false. When you set this property to true, you must specify |

a reset input value to the step method. When the reset input is a nonzero value, the object restarts the timing phase recovery process. When you set this property to false, the object does not restart.

## ResetCondition

Condition for timing phase recovery reset
Specify the conditions to reset the timing phase recovery process as Never I Every frame. The default is Never. When you set this property to Never, the phase recovery process never restarts. The object operates continuously, retaining information from one symbol to the next. When you set this property to Every frame, the timing phase recovery restarts at the start of each frame of data. Thus, restart occurs each time the object calls the step method. This property applies when you set the ResetInputPort property to false.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>reset<br>step

Create Mueller-Muller timing phase synchronizer object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
Reset states of Mueller-Muller timing phase synchronizer
Recover symbol timing phase using Mueller-Muller method

Examples Recover timing phase using the Mueller-Muller method.

```
% Initialize some data
    L = 16; M = 2; numSymb = 100; snrdB = 30;
    R = 25; rollOff = 0.75; filtDelay = 3; g = 0.07; delay = 6.6498;
    % Design raised cosine filters
    txFiltSpec = fdesign.pulseshaping(L, 'Square root raised cosine',
        'Nsym,Beta', 2*filtDelay, rollOff);
    txFilterDesign = design(txFiltSpec);
    txFilterDesign.Numerator = sqrt(L)*txFilterDesign.Numerator;
% Create System objects
    hMod = comm.DPSKModulator(M, 'PhaseRotation', 0);
    hTxFilter = dsp.FIRInterpolator(L, txFilterDesign.Numerator);
    hDelay = dsp.VariableFractionalDelay('MaximumDelay', L);
    hChan = comm.AWGNChannel(...
                            'NoiseMethod', 'Signal to noise ratio (SNR)', ..
                            'SNR', snrdB, 'SignalPower', 1/L);
    hRxFilter = dsp.DigitalFilter('TransferFunction', 'FIR (all zeros)
                            'Numerator', txFilterDesign.Numerator);
    hSync = comm.MuellerMullerTimingSynchronizer('SamplesPerSymbol',
        'ErrorUpdateGain', g);
% Generate random data
    data = randi([0 M-1], numSymb, 1);
% Modulate and filter transmitter data
    modData = step(hMod, data);
    filterData = step(hTxFilter, modData);
% Introduce a random delay.
    delayedData = step(hDelay, filterData, delay);
% Add noise
    chData = step(hChan, delayedData);
% Filter the receiver data
```

```
    rxData = step(hRxFilter, chData);
% Estimate the delay from the received signal
    [~, phase] = step(hSync, rxData);
    fprintf(1, 'Actual Timing Delay: %f\n', delay);
    fprintf(1, 'Estimated Timing Delay: %f\n', phase(end));
```

Algorithms<br>This object implements the algorithm, inputs, and outputs described on the Mueller-Muller Timing Recovery block reference page. The object properties correspond to the block parameters, except:<br>The Reset parameter corresponds to the ResetInputPort and ResetCondition properties.<br>See Also<br>comm.EarlyLateGateTimingSynchronizer |<br>comm.GMSKTimingSynchronizer

Purpose

Syntax
Description

Create Mueller-Muller timing phase synchronizer object with same property values

C = clone(H)
C = clone (H) creates a MuellerMullerTimingSynchronizer object C , with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.MuellerMullerTimingSynchronizer.getNumInputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $N=$ getNumInputs (H) method returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ ).
Purpose Number of outputs from step method

Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax $\quad$ TF $=$ isLocked (H)
Description TF = isLocked $(H)$ returns the locked status, TF of the MuellerMullerTimingSynchronizer System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Reset states of Mueller-Muller timing phase synchronizer<br>\section*{Syntax reset (H)}<br>Description reset $(H)$ resets the states of the MuellerMullerTimingSynchronizer object, H .

## Purpose

Recover symbol timing phase using Mueller-Muller method
Syntax

Description
[ Y, PHASE] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$ [ Y, PHASE] $=\operatorname{step}(H, X, R)$
[ Y, PHASE] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$ performs timing phase recovery and returns the time-synchronized signal, Y , and the estimated timing phase, PHASE, for input signal $X$. The input $X$ must be a double or single precision complex column vector. The length of X is $\mathrm{N}^{*} \mathrm{~K}$, where N is the value you specify in the property SamplesPerSymbol and K is the number of symbols. The output, Y , is the signal value for each symbol, which you use to make symbol decisions. Y is a column vector of length K with the same data type as $X$.
[ $\mathrm{Y}, \mathrm{PHASE}]=\operatorname{step}(\mathrm{H}, \mathrm{X}, \mathrm{R})$ restarts the timing phase recovery process when you input a reset signal, R, that is non-zero. R must be a logical or double scalar. This syntax applies when you set the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Deinterleave input symbols using set of shift registers with specified <br> delays |
| :---: | :--- |
| Description $\quad$The MultiplexedDeinterleaver object restores the original ordering <br> of a sequence that was interleaved using the General Multiplexed <br> Interleaver object. |  |
| Construction $\quad$H = comm. MultiplexedDeinterleaver creates a multiplexed <br> deinterleaver System object, H. This object restores the original ordering <br> of a sequence that was interleaved using the multiplexed interleaver <br> System object. <br> H = comm. MultiplexedDeinterleaver (Name, Value) creates a <br> multiplexed deinterleaver object, H, with each specified property set <br> to the specified value. You can specify additional name-value pair <br> arguments in any order as (Name1,Value1,...,NameN,ValueN). |  |
| Properties | Delay <br> Interleaver delay |
| Specify the lengths of the shift registers as an integer column <br> vector. The default is [2;0;1; ; ; 10]. |  |
| InitialConditions |  |
| Initial conditions of shift registers |  |


| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
|  | release |
| reset |  |
| step |  |


| clone | Create multiplexed deinterleaver <br> object with same property values |
| :--- | :--- |
| getNumInputs | Number of expected inputs to <br> step method |
| getNumOutputs | Number of outputs from step <br> method |
| isLocked | Locked status for input attributes <br> and nontunable properties |
| release | Allow property value and input <br> characteristics changes |
| reset | Reset states of the multiplexed <br> deinterleaver object |
| step | Deinterleave input symbols <br> using a set of shift registers with <br> specified delays |

Examples Interleave a sequence, and then restore it.

```
hInt = comm.MultiplexedInterleaver('Delay', [1 0 2]');
hDeInt = comm.MultiplexedDeinterleaver('Delay', [1 0 2]');
data = (1:20)';
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence and restorec
display([data intData deIntData]);
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the General Multiplexed Deinterleaver block reference page. The object properties correspond to the block parameters.

See Also

Purpose Create multiplexed deinterleaver object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a MultiplexedDeinterleaver object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.MultiplexedDeinterleaver.getNumInputs

## Purpose Number of expected inputs to step method

## Syntax <br> N = getNumInputs( H )

Description $\quad N=$ getNumInputs $(H)$ returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)
Purpose $\quad$ Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked $(H)$ returns the locked status, TF of the MultiplexedDeinterleaver System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.MultiplexedDeinterleaver.reset

| Purpose | Reset states of the multiplexed deinterleaver object |
| :--- | :--- |
| Syntax | $\operatorname{reset}(H)$ |
| Description | $\operatorname{reset}(H)$ resets the states of the MultiplexedDeinterleaver object, H. |


#### Abstract

Purpose

Syntax Y = step( $\mathrm{H}, \mathrm{X}$ ) $Y=\operatorname{step}(H, X)$ restores the original ordering of the sequence, $X$, that was interleaved using a multiplexed interleaver and returns $Y$. The input $X$ must be a column vector. The data type for $X$ can be numeric, logical, or fixed-point (fi objects). Y has the same data type as $X$. The multiplexed deinterleaver object uses N shift registers, where N is the number of elements in the vector specified by the Delay property. When a new input symbol enters the deinterleaver, a commutator switches to a new register. The new symbol shifts in while the oldest symbol in that register is shifted out. When the commutator reaches the Nth register, upon the next new input, it returns to the first register. The multiplexed deinterleaver associated with a multiplexed interleaver has the same number of registers as the interleaver. The delay in a particular deinterleaver register depends on the largest interleaver delay minus the interleaver delay for the given register.


Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose

Permute input symbols using set of shift registers with specified delays

## Construction

## Properties

## Delay

Interleaver delay
Specify the lengths of the shift registers as an integer column vector. The default is $[2 ; 0 ; 1 ; 3 ; 10]$.

## InitialConditions

Initial conditions of shift registers
Specify the initial values in each shift register as a numeric scalar value or a column vector. The default is 0 . When you set this property to a column vector, the length must equal the value of the Delay property. This vector contains initial conditions, where the $i$-th initial condition is stored in the $i$-th shift register.

## Methods

getNumInputs
clone

Create multiplexed interleaver object with same property values

Number of expected inputs to step method
getNumOutputs
isLocked
release
reset
step

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of the multiplexed interleaver object

Permute input symbols using a set of shift registers with specified delays

## Examples Interleave a sequence, and then restore it.

```
hInt = comm.MultiplexedInterleaver('Delay', [1 0 2]');
hDeInt = comm.MultiplexedDeinterleaver('Delay', [1 0 2]');
data = (1:20)';
intData = step(hInt, data);
deIntData = step(hDeInt, intData);
% compare the original sequence, interleaved sequence, and restored
% sequence
[data, intData, deIntData]
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the General Multiplexed Interleaver block reference page. The object properties correspond to the block parameters.

See Also

comm.MultiplexedDeinterleaver | comm.ConvolutionalInterleaver

Purpose
Create multiplexed interleaver object with same property values

## Syntax <br> C = clone( H )

Description

C = clone (H) creates a MultiplexedInterleaver object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the MultiplexedInterleaver System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Reset states of the multiplexed interleaver object

## Syntax reset (H)

Description reset (H) resets the states of the MultiplexedInterleaver object, H.

## Purpose

## Syntax

Description

Permute input symbols using a set of shift registers with specified delays

Y $=\operatorname{step}(H, X)$
$Y=\operatorname{step}(H, X)$ permutes input sequence, $X$, and returns interleaved sequence, $Y$. The input $X$ must be a column vector and the data type can be numeric, logical, or fixed-point (fi objects). Y has the same data type as X . The multiplexed interleaver object consists of N registers, each with a specified delay. With each new input symbol, a commutator switches to a new register and the new symbol is shifted in while the oldest symbol in that register is shifted out. When the commutator reaches the Nth register, upon the next new input, it returns to the first register.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Demodulate using OQPSK method |
| :---: | :---: |
| Description | The OQPSKDemodulator object demodulates a signal that was modulated using the offset quadrature phase shift keying method. The input is a baseband representation of the modulated signal. |
| Construction | H = comm.OQPSKDemodulator creates a demodulator System object, H. This object demodulates the input signal using the offset quadrature phase shift keying (OQPSK) method. |
|  | H = comm.OQPSKDemodulator(Name, Value) creates an OQPSK demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN). |
|  | H = comm.OQPSKDemodulator(PHASE, Name, Value) creates an OQPSK demodulator object, H . This object has the PhaseOffset property set to PHASE and the other specified properties set to the specified values. |
| Properties | PhaseOffset |
|  | Phase of zeroth point of constellation from $\pi / 4$ |
|  | Specify the phase offset of the zeroth point of the constellation |
|  | shifted from $\pi / 4$, in radians, as a finite, real-valued scalar. The default is 0 . |
|  | BitOutput |
|  | Output data as bits |
|  | Specify whether the output consists of groups of bits or integer values. The default is false. When you set this property to true the step method outputs a column vector of bit values. The vector length must equal to twice the number of demodulated symbols. When you set this property to false, the step method outputs a column vector. The length of this vector equals to the number of demodulated symbols that contain integer values between 0 and |

3. The object produces one output demodulated symbol for each pair of input samples.

## OutputDataType

Data type of output
Specify output data type as Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32. The default is Full precision. When you set this property to Full precision, the step method output inherits the data type from the input. When the input is of single or double data, the step method outputs the same data type as the input. When the input data is of a fixed-point type, then the step method outputs the same data type as if you set the OutputDataType property to Smallest unsigned integer. When you set the BitOutput property to true, the logical data type becomes a valid option.

When the input signal is an integer data type, you must have a Fixed-Point Designer user license to use this property in Smallest unsigned integer or Full precision mode.

## Fixed-Point Properties

## DerotateFactorDataType

Data type of derotate factor
Specify derotate factor data type as one of Same word length as input | Custom. The default is Same word length as input. The object uses the derotate factor in the computations only when the step method input is of a fixed-point type and the

PhaseOffset property has a value that is not a multiple of $\pi / 2$.

## CustomDerotateFactorDataType

Fixed-point data type of derotate factor
Specify the derotate factor fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is
numerictype([],16). This property applies when you set the DerotateFactorDataType property to Custom.

## AccumulatorDataType

Data type of accumulator
Specify AccumulatorMode as one of Full precision | Same as input | Custom. The default is Full precision.

## CustomAccumulatorDataType

Fixed-point data type of accumulator
Specify the accumulator output fixed-point type as a scaled numerictype object with a signedness of Auto. The default is numerictype([],32,15). This property applies when you set the AccumulatorDataType property to Custom.

## AccumulatorRoundingMethod

Rounding of fixed-point numeric value of accumulator
Specify the accumulator rounding method as Ceiling | Convergent | Floor | Nearest | Round | Simplest | Zero. The default is Floor.

## AccumulatorOverflowAction

Action when fixed-point numeric value of accumulator overflows
Specify the accumulator overflow action as Wrap | Saturate. The default is Wrap.

## MappingDataType

Data type of mapping
Specify the mapping data type as Same as accumulator | Custom. The default is Same as accumulator.

## CustomMappingDataType

Fixed-point data type of mapping

Specify the mapping fixed-point type as a scaled numerictype object with a signedness of Auto. The default is numerictype([],32,15). This property applies when you set the MappingDataType property to Custom.

| Methods | clone |
| :--- | :--- |
|  | constellation |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
|  | release |
| reset |  |
| step |  |

Create OQPSK demodulator object with same property values
Calculate or plot ideal signal constellation

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset demodulator state
Demodulate using OQPSK method

Examples Modulate and demodulate a signal using OQPSK modulation with a constellation with pi/8 radians of phase offset.

```
    hMod = comm.OQPSKModulator(pi/8);
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)','SNR',6);
hDemod = comm.OQPSKDemodulator(pi/8);
% Create an error rate calculator, account for the one symbol delay
    hError = comm.ErrorRate('ReceiveDelay',1);
for counter = 1:100
% Transmit a 50-symbol frame
```

```
    data = randi([0 3],50,1);
    modSignal = step(hMod, data);
    noisySignal = step(hAWGN, modSignal);
    receivedData = step(hDemod, noisySignal);
    errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the OQPSK Demodulator Baseband block reference page. The object properties correspond to the block parameters.
comm.OQPSKModulator | comm.QPSKDemodulator

Purpose Create OQPSK demodulator object with same property values
Syntax $\quad C=$ clone $(H)$
Description $\quad C=$ clone $(H)$ creates a OQPSKDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.OQPSKDemodulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax $\quad y=$ constellation $(h)$
constellation(h)
Description $y=$ constellation( $h$ ) returns the numerical values of the constellation.
constellation(h) generates a constellation plot for the object.
Examples Calculate Ideal Signal Constellation for comm.OQPSKDemodulator

Create a comm.OQPSKDemodulator System object, and then calculate its ideal signal constellation.

Create a comm. OQPSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.OQPSKDemodulator

Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.OQPSKDemodulator

Create a comm.OQPSKDemodulator System object, and then plot the ideal signal constellation.

Create a comm.OQPSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.OQPSKDemodulator
Plot the ideal signal constellation by calling the constellation method. constellation(h)

## comm.OQPSKDemodulator.getNumInputs

## Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.OQPSKDemodulator.getNumOutputs

Purpose $\quad$ Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.OQPSKDemodulator.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked $(H)$ |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the |
|  | OQPSKDemodulator System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.OQPSKDemodulator.release

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Reset demodulator state

## Syntax $\quad \operatorname{reset}(H)$

Description reset (H) resets the states of the OQPSKDemodulator object, H.

Purpose Demodulate using OQPSK method

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description $\quad Y=\operatorname{step}(H, X)$ demodulates data, $X$, with the OQPSK demodulator object, H, and returns Y. Input X must be a double, single, or signed fixed-point data type scalar or column vector. The object produces one output symbol for each pair of input samples. When used with the OQPSK modulator object, the step method output has a one symbol delay as compared to the input of the modulator. Depending on the BitOutput property value, output $Y$ can be integer or bit valued.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Modulate using OQPSK method

Description

## Construction

The OQPSKModulator object modulates using the offset quadrature phase shift keying method. The output is a baseband representation of the modulated signal.

H = comm.OQPSKModulator creates a modulator System object, H. This object modulates the input signal using the offset quadrature phase shift keying (OQPSK) method.

H = comm.OQPSKModulator(Name, Value) creates an OQPSK modulator object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm. OQPSKModulator(PHASE, Name, Value) creates an OQPSK modulator object, H. This object has the PhaseOffset property set to PHASE and the other specified properties set to the specified values.

## Properties

## PhaseOffset

Phase of zeroth point of constellation from $\pi / 4$
Specify the phase offset of the zeroth point of the constellation
shifted from $\pi / 4$, in radians, as a finite, real-valued scalar. The default is 0 .

## BitInput

Assume input is bits
Specify whether the input is bits or integers. The default is false. When you set this property to true, the inputs are bit representations of integers between 0 and 3 . The input requires a column vector of bit values with length that is an integer multiple of two. When you set this property to false, the input requires a column vector of integer values between 0 and 3 .

## OutputDataType

Data type of output
Specify the output data type as double | single | Custom. The default is double.

## Fixed-Point Properties

## CustomOutputDataType

Fixed-point data type of output
Specify the output fixed-point type as a numerictype object with a signedness of Auto. The default is numerictype ([],16). This property applies when you set the OutputDataType property to Custom.

| Methods | clone | Create OQPSK modulator object <br> with same property values |
| :--- | :--- | :--- |
| constellation | Calculate or plot ideal signal <br> constellation |  |
| getNumInputs | Number of expected inputs to <br> step method |  |
| getNumOutputs | Number of outputs from step <br> method |  |
| isLocked | Locked status for input attributes <br> and nontunable properties |  |
| release | Allow property value and input <br> characteristics changes |  |
| Examples | reset | Reset modulator state <br> Modulate using OQPSK method |
|  | step | Modulate data using OQPSK, and visualize the modulated data in a |

\% Create binary data for 1000, 2-bit symbols data $=$ randi([0 1],2000,1);
\% Create an OQPSK modulator System object and accept bits as inputs ar hModulator = comm.OQPSKModulator(pi/16, 'BitInput',true);
\% Modulate and plot the data, ignore the first output symbol modData = step(hModulator, data); scatterplot(modData(2:end))

# Algorithms 

This object implements the algorithm, inputs, and outputs described on the OQPSK Modulator Baseband block reference page. The object properties correspond to the block parameters.

See Also

comm.OQPSKDemodulator | comm.QPSKModulator

Purpose Create OQPSK modulator object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a OQPSKModulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.OQPSKModulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax

y = constellation(h)

constellation(h)
Description $y=$ constellation( $h$ ) returns the numerical values of the
constellation.
constellation(h) generates a constellation plot for the object.

## Examples Calculate Ideal Signal Constellation for comm.OQPSKModulator

Create a comm.OQPSKModulator System object, and then calculate its ideal signal constellation.
Create a comm.OQPSKModulator System object by entering the following at the MATLAB command line:
h = comm.OQPSKModulator
Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.OQPSKModulator

Create a comm.OQPSKModulator System object, and then plot the ideal signal constellation.
Create a comm.OQPSKModulator System object by entering the following at the MATLAB command line:
h = comm.OQPSKModulator
Plot the ideal signal constellation by calling the constellation method.

```
constellation(h)
```


## comm.OQPSKModulator.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.OQPSKModulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the OQPSKModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.OQPSKModulator.reset

## Purpose Reset modulator state

## Syntax reset (H)

Description reset (H) resets the states of the OQPSKModulator object, H.

| Purpose | Modulate using OQPSK method |
| :--- | :--- |
| Syntax | $Y=\operatorname{step}(H, X)$ |
| Description | $Y=$ step $(H, X)$ modulates input data, $X$, with the OQPSK modulator <br> object, $H$, and returns baseband modulated output, Y. Depending on the <br> value of the BitInput property, input $X$ can be an integer or bit valued <br> column vector with numeric, logical, or fixed-point data types. |
|  | The OQPSK modulator object upsamples by a factor of two. The step <br> method outputs the length, $Y$, as $2 \times N$, where $N$ is the length of the <br> input, $X$. The step method outputs an initial condition of zero, which <br> is unrelated to the input values. |

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Construction

## Properties

Purpose Combine inputs using orthogonal space-time block code

Combine inputs using orthogonal space-time block code

The OSTBCCombiner object combines the input signal (from all of the receive antennas) and the channel estimate signal to extract the soft information of the symbols encoded by an OSTBC. The input channel estimate does not need to be constant and can vary at each call to the step method. The combining algorithm uses only the estimate for the first symbol period per codeword block. A symbol demodulator or decoder would follow the Combiner object in a MIMO communications system.

H = comm.OSTBCCombiner creates an orthogonal space-time block code (OSTBC) combiner System object, H. This object combines the input signal (from all of the receive antennas) with the channel estimate signal to extract the soft information of the symbols encoded by an OSTBC.

H = comm.OSTBCCombiner(Name, Value) creates an OSTBC Combiner object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm. OSTBCCombiner ( $\mathrm{N}, \mathrm{M}$, Name, Value) creates an OSTBC Combiner object, H. This object has the NumTransmitAntennas property set to N, the NumReceiveAntennas property set to N, and the other specified properties set to the specified values.

## NumTransmitAntennas

Number of transmit antennas
Specify the number of antennas at the transmitter as $2|3| 4$. The default is 2 .

## SymbolRate

Symbol rate of code
Specify the symbol rate of the code as $3 / 4 \mid 1 / 2$. The default is $3 / 4$. This property applies when the NumTransmitAntennas
property is greater than 2. For 2 transmit antennas, the symbol rate defaults to 1 .

## NumReceiveAntennas

Number of receive antennas
Specify the number of antennas at the receiver as a double-precision, real, scalar integer value from 1 to 8. The default is 1 .

## Fixed-Point Properties

## RoundingMethod

Rounding of fixed-point numeric values
Specify the rounding method as Ceiling | Convergent | Floor | Nearest | Round | Simplest | Zero. The default is Floor.

## OverflowAction

Action when fixed-point numeric values overflow
Specify the overflow action as one of Wrap \| Saturate. The default is Wrap. This property specifies the action to be taken in case of overflow. Such overflow occurs if the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result.

## ProductDataType

Data type of product
Specify the product data type as one of Full precision | Custom. The default is Full precision.

## CustomProductDataType

Fixed-point data type of product
Specify the product fixed-point type as a scaled numerictype object with a signedness of Auto. The default is numerictype([],32,16).

This property applies when you set the ProductDataType property to Custom.

## AccumulatorDataType

Data type of accumulator
Specify the accumulator data type as Full precision | Same as product | Custom. The default is Full precision.

## CustomAccumulatorDataType

Fixed-point data type of accumulator
Specify the accumulator fixed-point type as a scaled numerictype object with a signedness of Auto. The default is numerictype([],32,16). This property applies when you set the AccumulatorDataType property to Custom.

## EnergyProductDataType

Data type of energy product
Specify the complex energy product data type as one of Full precision | Same as product | Custom. The default is Full precision. This property sets the data type of the complex product in the denominator to calculate the total energy in the MIMO channel.

## CustomEnergyProductDataType

Fixed-point data type of energy product
Specify the energy product fixed-point type as a scaled numerictype object with a signedness of Auto. The default is numerictype([],32,16). This property applies when you set the EnergyProductDataType property to Custom.

## EnergyAccumulatorDataType

Data type of energy accumulator
Specify the energy accumulator data type as one of Full precision | Same as energy product | Same as accumulator
| Custom. The default is Full precision. This property sets the data type of the summation in the denominator to calculate the total energy in the MIMO channel.

## CustomEnergyAccumulatorDataType

Fixed-point data type of energy accumulator
Specify the energy accumulator fixed-point type as a scaled numerictype object with a signedness of Auto. The default is numerictype([],32,16). This property applies when you set the EnergyAccumulatorDataType property to Custom.

## DivisionDataType

Data type of division
Specify the division data type as one of Same as accumulator | Custom. The default is Same as accumulator. This property sets the data type at the output of the division operation. The setting normalizes diversity combining by the total energy in the MIMO channel.

## CustomDivisionDataType

Fixed-point data type of division
Specify the division fixed-point type as a scaled numerictype object with a signedness of Auto. The default is numerictype([], 32,16 ). This property applies when you set the DivisionDataType property to Custom.

## Methods

clone
getNumInputs
getNumOutputs

Create OSTBC combiner object with same property values

Number of expected inputs to step method

Number of outputs from step method

| isLocked | Locked status for input attributes <br> and nontunable properties |
| :--- | :--- |
| release | Allow property value and input <br> characteristics changes |
| step | Combine inputs using orthogonal <br> space-time block code |

## Examples

Encode and decode QPSK modulated data with OSTBC and calculate error.
\% Define system parameters numTx = 2; numRx = 1; Rs = 1e6; maxDopp = 30; ... numBits = 1024; SNR = 10;
\% Create modulator and encoder System objects hMod = comm.QPSKModulator(... 'BitInput', true,... 'SymbolMapping', 'Gray'); hDemod $=$ comm.QPSKDemodulator(...
'SymbolMapping', 'Gray',...
'BitOutput', true);
hOSTBCEnc $=$ comm.OSTBCEncoder(...
'NumTransmitAntennas', numTx); hOSTBCComb $=$ comm. OSTBCCombiner(..
'NumTransmitAntennas', numTx,...
'NumReceiveAntennas', numRx);
\% Create MIMO channel System object
hChan $=$ comm.MIMOChannel(...
'SampleRate', Rs,...
'MaximumDopplerShift', maxDopp,...
'NumTransmitAntennas', numTx,...
'NumReceiveAntennas', numRx,...
'ReceiveCorrelationMatrix', 1,...
'PathGainsOutputPort', true);
\% Create AWGN channel System object hAWGN = comm.AWGNChannel(...

```
    'NoiseMethod', 'Signal to noise ratio (SNR)',...
    'SNR', SNR,...
    'SignalPower', 1);
% Generate data
    data = randi([0 1], numBits, 1);
% Modulate data
    modData = step(hMod, data);
% Encode modulated data using OSTBC
    encData = step(hOSTBCEnc, modData);
% Transmit through Rayleigh and AWGN channels
    [chanOut, pathGains] = step(hChan, encData);
    rxSignal = step(hAWGN, chanOut);
% Decode and demodulate received signal
    decData = step(hOSTBCComb, rxSignal, squeeze(pathGains));
    receivedData = step(hDemod, decData);
% Compute number of bit errors in received data
    errors = biterr(data, receivedData);
    fprintf(1, ['\nThere were %d errors in the received signal ' ...
        'out of %d bits transmitted\n'], errors, length(data));
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the OSTBC Combiner block reference page. The object properties correspond to the block parameters.
comm.OSTBCEncoder

## comm.OSTBCCombiner.clone

Purpose Create OSTBC combiner object with same property values

## Syntax

Description
$C=$ clone $(H)$ creates a OSTBCCombiner object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.OSTBCCombiner.getNumInputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.OSTBCCombiner.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

| Purpose | Locked status for input attributes and nontunable properties |
| :---: | :---: |
| Syntax | TF = isLocked(H) |
| Description | TF = isLocked(H) returns the locked status, TF of the OSTBCCombiner System object. |
|  | The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value. |

## comm.OSTBCCombiner.release

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose

Combine inputs using orthogonal space-time block code

## Syntax

Description
$Y=\operatorname{step}(H, X, C E S T)$
$Y=\operatorname{step}(H, X, C E S T)$ combines the received data, $X$, and the channel estimate, CEST, to extract the symbols encoded by an OSTBC. Both $X$ and CEST are complex-valued and of the same data type, which can be double, single, or signed fixed point with power-of-two slope and zero bias. When the step method input $X$ has double or single precision, the output, Y , has the same data type as the input. The input channel estimate can remain constant or can vary during each codeword block transmission. The combining algorithm uses the estimate only for the first symbol period per codeword block.

The time domain length, $T /$ SymbolRate, must be a multiple of the codeword block length. $T$ is the output symbol sequence length in the time domain. Specifically, when you set the NumTransmitAntennas property to 2 , $T /$ SymbolRate must be a multiple of two. When you set the NumTransmitAntennas property greater than 2, T/SymbolRate must be a multiple of four. For an input of $T /$ SymbolRate rows by NumReceiveAntennas columns, the input channel estimate, CEST, must be a matrix of size $T /$ SymbolRateby NumTransmitAntennas by NumReceiveAntennas. In this case, the extracted symbol data, Y , is a column vector with $T$ elements. Input matrix size can be $F$ by $T /$ SymbolRate by NumReceiveAntennas, where $F$ is an optional dimension (typically frequency domain) over which the combining calculation is independent. In this case, the input channel estimate, CEST, must be a matrix of size F by T/SymbolRate by NumTransmitAntennas by NumReceiveAntennas. The extracted symbol data, Y, is an $F$ rows by $T$ columns matrix.


#### Abstract

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.


## Purpose Encode input using orthogonal space-time block code

Description

## Construction

The OSTBCEncoder object encodes an input symbol sequence using orthogonal space-time block code (OSTBC). The block maps the input symbols block-wise and concatenates the output codeword matrices in the time domain.

H = comm.OSTBCEncoder creates an orthogonal space-time block code (OSTBC) encoder System object, H. This object maps the input symbols block-wise and concatenates the output codeword matrices in the time domain.

H = comm.OSTBCEncoder(Name, Value) creates an OSTBC encoder object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm. OSTBCEncoder(N,Name, Value) creates an OSTBC encoder object, H. This object has the NumTransmitAntennas property set to N, and the other specified properties set to the specified values.

## Properties

NumTransmitAntennas
Number of transmit antennas
Specify the number of antennas at the transmitter as $2|3| 4$. The default is 2 .

## SymbolRate

Symbol rate of code
Specify the symbol rate of the code as one of $3 / 4$ | 1/2. The default is $3 / 4$. This property applies when you set the NumTransmitAntennas property to greater than 2. For 2 transmit antennas, the symbol rate defaults to 1 .

## Fixed-Point Properties

## OverflowAction

Action when fixed-point numeric values overflow
Specify the overflow action as one of Wrap \| Saturate. The default is Wrap. This property specifies the action to be taken in the case of an overflow. Such overflow occurs when the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| step |  |

getNumOutputs
isLocked
release
step

Create OSTBC encoder object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Encode input using orthogonal space-time block code

## Examples Encode BPSK modulated data with OSTBC.

```
% Generate random binary data
    data = randi([0 1], 8, 1);
```

\%Create BPSK Modulator and obtain modulated data
hMod = comm.BPSKModulator;
modData $=$ step(hMod, data);

> Algorithms This object implements the algorithm, inputs, and outputs described on the OSTBC Encoder block reference page. The object properties correspond to the block parameters.

When this object processes variable-size signals:

- If the input signal is a column vector, the first dimension can change, but the second dimension must remain fixed at 1.
- If the input signal is a matrix, both dimensions can change.

See Also
comm. OSTBCCombiner

Purpose Create OSTBC encoder object with same property values

## Syntax <br> C = clone(H)

Description
$C=$ clone $(H)$ creates a OSTBCEncoder object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.OSTBCEncoder.getNumInputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.OSTBCEncoder.getNumOutputs

Purpose $\quad$ Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.OSTBCEncoder.isLocked

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the OSTBCEncoder System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose

Encode input using orthogonal space-time block code

## Syntax

Description
Y = step(H,X)
$Y=\operatorname{step}(H, X)$ encodes the input data, $X$, using OSTBC encoder
object, H. The input is a complex-valued column vector or matrix of data type double, single, or signed fixed-point with power-of-two slope and zero bias. The step method output, Y , is the same data type as the input data. The time domain length, $T$, of X must be a multiple of the number of symbols in each codeword matrix. Specifically, when you set the NumTransmitAntennas property is 2 or the SymbolRate property is $1 / 2, T$ must be a multiple of two and when the SymbolRate property to $3 / 4, T$ must be a multiple of three. For a time or spatial domain input of $T$ rows by one column, the encoded output data, Y , is a (T/SymbolRate)-by-NumTransmitAntennas matrix. The input matrix size can be $F$ rows by $T$ columns, where $F$ is the additional dimension (typically the frequency domain) over which the encoding calculation is independent. In this case, the output is an $F$-by-(T/SymbolRate)-by-NumTransmitAntennas matrix.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Generate OVSF code

Description

Construction

## Properties

The OVSFCode object generates an OVSF code from a set of orthogonal codes. OVSF codes were first introduced for 3G communication systems. They are primarily used to preserve orthogonality between different channels in a communication system.
$\mathrm{H}=$ comm.OVSFCode creates an orthogonal variable spreading factor (OVSF) code generator System object, H. This object generates an OVSF code.

H = comm.OVSFCode(Name, Value) creates an OVSF code generator object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## SpreadingFactor

Length of generated code
Specify the length of the generated code as an integer scalar value with a power of two. The default is 64 .

## Index

Index of code of interest
Specify the index of the desired code from the available set of codes that have the spreading factor specified in the SpreadingFactor property. This property must be an integer scalar in the range 0 to SpreadingFactor-1. The default is 60.

OVSF codes are defined as the rows of an $n$-by- $n$ matrix, $C n$, where $n$ is the value specified in the SpreadingFactor property.

You can define the matrix $C n$ recursively as follows:
First, define $C 1=[1]$.
Next, assume that $C n$ is defined and let $C n(k)$ denote the $k$-th row of $C n$.

$$
\begin{aligned}
& \text { Then, } C 2 n=[C n(0) C n(0) ; C n(0)-C n(0) ; \ldots ; C n(n-1) C n(n-1) \text {; } \\
& C n(n-1)-C n(n-1)] \text {. } \\
& C n \text { is only defined for values of } \mathrm{n} \text { that are a power of } 2 \text {. Set the } \\
& \text { this property to a value of } k \text { to choose the } k \text {-th row of the } C \text { matrix } \\
& \text { as the code of interest. }
\end{aligned}
$$

## SamplesPerFrame

Number of output samples per frame
Specify the number of OVSF code samples that the step method outputs as a numeric, positive, integer scalar value. The default is 1 . If you set this property to a value of $M$, then the step method outputs $M$ samples of an OVSF code of length $N$. $N$ is the length of the OVSF code that you specify in the SpreadingFactor property.

## OutputDataType

Data type of output
Specify output data type as one of double | int8. The default is double.

| Methods | clone | Create OVSF code generator <br> object with same property values |
| :---: | :--- | :--- |
| getNumInputs | getNumOutputs | Number of expected inputs to <br> step method |
| isLocked | Number of outputs from step <br> method |  |
| release | Locked status for input attributes <br> and nontunable properties |  |
| reset | Allow property value and input <br> characteristics changes |  |
| step | Reset states of OVSF code <br> generator object <br> Generate OVSF code |  |

```
Examples Generate 10 samples of an OVSF code with a spreading factor of 64 .
hOVSF = comm.OVSFCode('SamplesPerFrame', 10,'SpreadingFactor',64);
seq \(=\) step(hOVSF)
```

Algorithms
This object implements the algorithm, inputs, and outputs described on the OVSF Code Generator block reference page. The object properties correspond to the block parameters, except:

- The object does not have a property to select frame based outputs.
- The object does not have a property that corresponds to the Sample time parameter.


## See Also

Purpose
Create OVSF code generator object with same property values

## Syntax <br> C = clone(H)

Description
C $=$ clone $(H)$ creates a OVSFCode object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs $(H)$ returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the OVSFCode System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.OVSFCode.reset

## Purpose Reset states of OVSF code generator object

## Syntax reset (H)

Description reset (H) resets the states of the OVSFCode object, H.

| Purpose | Generate OVSF code |
| :--- | :--- |
| Syntax | $Y=$ step (H) |
| Description | $Y=$ step $(H)$ outputs a frame of the OVSF code in column vector $Y$. <br> Specify the frame length with the SamplesPerFrame property. |
|  | Note The object performs an initialization the first time the step <br> method is executed. This initialization locks nontunable properties and <br> input specifications, such as dimensions, complexity, and data type <br> of the input data. If you change a nontunable property or an input <br> specification, the System object issues an error. To change nontunable <br> properties or inputs, you must first call the release method to unlock <br> the object. |

## Purpose Demodulate using M-ary PAM method

Description

## Construction

## Properties

The PAMDemodulator object demodulates a signal that was modulated using M-ary pulse amplitude modulation. The input is a baseband representation of the modulated signal.

H = comm. PAMDemodulator creates a demodulator System object, H. This object demodulates the input signal using the M-ary pulse amplitude modulation (M-PAM) method.

H = comm. PAMDemodulator (Name, Value) creates an M-PAM demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm. PAMDemodulator(M,Name, Value) creates an M-PAM demodulator object, H. This object has the ModulationOrder property set to $M$, and the other specified properties set to the specified values.

## ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation as a positive, integer scalar value. The default is 4 . When you set the BitOutput property to false, this value must be even. When you set the BitOutput property to true, this value requires an integer power of two.

## BitOutput

Output data as bits
Specify whether the output consists of groups of bits or integer symbol values. The default is false.

When you set this property to true the step method outputs a column vector of bit values with length equal to $\log 2$ (ModulationOrder) times the number of demodulated symbols.

When you set this property to false, the step method outputs a column vector, with length equal to the input data vector. This value contains integer symbol values between 0 and ModulationOrder-1.

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of log2(ModulationOrder) bits to the corresponding symbol as one of Binary | Gray. The default is Gray.

When you set this property to Gray, the object uses a Gray-encoded signal constellation.

When you set this property to Binary, the integer $m$, between $0 \leq m \leq$ (ModulationOrder-1) maps to the complex value $2 m$-ModulationOrder+1.

## NormalizationMethod

Constellation normalization method
Specify the method used to normalize the signal constellation as one of Minimum distance between symbols | Average power | Peak power. The default is Minimum distance between symbols.

## MinimumDistance

Minimum distance between symbols
Specify the distance between two nearest constellation points as a positive, real, numeric scalar value. The default is 2. This property applies when you set the NormalizationMethod property to Minimum distance between symbols.

## AveragePower

Average power of constellation
Specify the average power of the symbols in the constellation as a positive, real, numeric scalar value. The default is 1. This
property applies when you set the NormalizationMethod property to Average power.

## PeakPower

Peak power of constellation
Specify the maximum power of the symbols in the constellation as a positive, real, numeric scalar value. The default is 1 . This property applies when you set the NormalizationMethod property to Peak power.

## OutputDataType

Data type of output
Specify the output data type as one of Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32. The default is Full precision.

When you set this property to Full precision, and the input data type is single or double precision, the output data has the same data type that of the input.

When the input signal is an integer data type, you must have a Fixed-Point Designer user license to use this property in Smallest unsigned integer or Full precision mode.

When the input data is of a fixed-point type, the output data type behaves as if you had set the OutputDataType property to Smallest unsigned integer.

When you set the BitOutput property to true, then logical data type becomes a valid option.

## Fixed-Point Properties

## FullPrecisionOverride

Full precision override for fixed-point arithmetic
Specify whether to use full precision rules. If you set FullPrecisionOverride to true, which is the default, the object
computes all internal arithmetic and output data types using full precision rules. These rules provide the most accurate fixed-point numerics. It also turns off the display of other fixed-point properties because they do not apply individually. These rules guarantee that no quantization occurs within the object. Bits are added, as needed, to ensure that no roundoff or overflow occurs. If you set FullPrecisionOverride to false, fixed-point data types are controlled through individual fixed-point property settings. For more information, see "Full Precision for Fixed-Point System Objects".

## DenormalizationFactorDataType

Data type of denormalization factor
Specify the denormalization factor data type as one of Same word length as input | Custom. The default is Same word length as input.

## CustomDenormalizationFactorDataType

Fixed-point data type of denormalization factor
Specify the denormalization factor fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the DenormalizationFactorDataType property to Custom.

## ProductDataType

Data type of product
Specify the product data type as one of Full precision | Custom. The default is Full precision. When you set this property to Full precision the object calculates the full-precision product word and fraction lengths. This property applies when you set the FullPrecisionOverride property to false.

## CustomProductDataType

Fixed-point data type of product

Specify the product fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is numerictype([],32). This property applies when you set the FullPrecisionOverride property to false and the ProductDataType property to Custom.

## ProductRoundingMethod

Rounding of fixed-point numeric value of product
Specify the product rounding method as one of Ceiling Convergent | Floor | Nearest | Round | Simplest | Zero. The default is Floor. This property applies when the object is not in a full precision configuration

## ProductOverflowAction

Action when fixed-point numeric value of product overflows
Specify the product overflow action as one of Wrap | Saturate. The default is Wrap. This property applies when the object is not in a full precision configuration.

## SumDataType

Data type of sum
Specify the sum data type as one of Full precision | Same as product | Custom. The default is Full precision. When you set this property to Full precision, the object calculates the full-precision sum word and fraction lengths. This property applies when you set the FullPrecisionOverride property to false

## CustomSumDataType

Fixed-point data type of sum
Specify the sum fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is numerictype([],32). This property applies when you set the the FullPrecisionOverride property to false and the SumDataType property to Custom.

Methods<br>clone<br>constellation<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>step

Create M-PAM demodulator object with same property values

Calculate or plot ideal signal constellation

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Demodulate using M-ary PAM method

Examples Modulate and demodulate a signal using 16-PAM modulation.

```
hMod = comm.PAMModulator(16);
hAWGN = comm.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)', ...
    'SNR',20, 'SignalPower', 85);
hDemod = comm.PAMDemodulator(16);
%Create an error rate calculator
hError = comm.ErrorRate;
for counter = 1:100
    % Transmit a 50-symbol frame
    data = randi([0 hMod.ModulationOrder-1],50,1);
    modSignal = step(hMod, data);
    noisySignal = step(hAWGN, modSignal);
    receivedData = step(hDemod, noisySignal);
    errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
```


## comm.PAMDemodulator

```
errorStats(1), errorStats(2))
```

> Algorithms This object implements the algorithm, inputs, and outputs described on the M-PAM Demodulator Baseband block reference page. The object properties correspond to the block parameters.

See Also comm. PAMModulator

Purpose Create M-PAM demodulator object with same property values

## Syntax $\quad C=$ clone $(H)$

Description $\quad C=$ clone $(H)$ creates a PAMDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.PAMDemodulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax $\quad y=$ constellation $(h)$
constellation(h)
Description $\quad y=$ constellation( $h$ ) returns the numerical values of the constellation.
constellation(h) generates a constellation plot for the object.
Examples Calculate Ideal Signal Constellation for comm.PAMDemodulator

Create a comm. PAMDemodulator System object, and then calculate its ideal signal constellation.

Create a comm. PAMDemodulator System object by entering the following at the MATLAB command line:
h = comm.PAMDemodulator

Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.PAMDemodulator

Create a comm. PAMDemodulator System object, and then plot the ideal signal constellation.

Create a comm. PAMDemodulator System object by entering the following at the MATLAB command line:
h = comm.PAMDemodulator
Plot the ideal signal constellation by calling the constellation method.
constellation(h)

## comm.PAMDemodulator.getNuminputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs $(H)$ returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ )

## comm.PAMDemodulator.getNumOutputs

Purpose $\quad$ Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.PAMDemodulator.isLocked

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the PAMDemodulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.PAMDemodulator.release

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

# Purpose <br> Demodulate using M-ary PAM method 

Syntax $\quad Y=\operatorname{step}(H, X)$
Description
$Y=\operatorname{step}(H, X)$ demodulates data, $X$, with the M-PAM demodulator System object, H, and returns Y. Input X must be a scalar or column vector. The data type of the input can be double or single precision, signed integer, or signed fixed point (fi objects). Depending on the BitOutput property value, output $Y$ can be integer or bit valued.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Modulate using M-ary PAM method

Description

## Construction

## Properties

The PAMModulator object modulates using M-ary pulse amplitude modulation. The output is a baseband representation of the modulated signal. The M-ary number parameter, M, represents the number of points in the signal constellation and requires an even integer.

H = comm. PAMModulator creates a modulator System object, H. This object modulates the input signal using the M -ary pulse amplitude modulation (M-PAM) method.

H = comm.PAMModulator(Name, Value) creates an M-PAM modulator object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.PAMModulator(M,Name, Value) creates an M-PAM modulator object, H. This object has the ModulationOrder property set to M and the other specified properties set to the specified values.

## ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation as a positive, integer scalar value. The default is 4 . When you set the BitInput property to false, ModulationOrder must be even. When you set the BitInput property to true, ModulationOrder must be an integer power of two.

## BitInput

Assume bit inputs
Specify whether the input is in bits or integers. The default is false.

When you set this property to true, the step method input requires a column vector of bit values whose length is an integer
multiple of $\log 2$ (ModulationOrder). This vector contains bit representations of integers between 0 and ModulationOrder-1.

When you set this property to false, the step method input must be a column vector of integer symbol values between 0 and ModulationOrder-1.

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of $\log 2$ (ModulationOrder) input bits to the corresponding symbol as one of Binary | Gray. The default is Gray.

When you set this property to Gray, the object uses a Gray-encoded signal constellation.

When you set this property to Binary, the input integer $m$, between $0 \leq m \leq$ ModulationOrder-1) maps to the complex value $2 m$-ModulationOrder +1 .

## NormalizationMethod

Constellation normalization method
Specify the method used to normalize the signal constellation as one of Minimum distance between symbols | Average power | Peak power. The default is Minimum distance between symbols.

## MinimumDistance

Minimum distance between symbols
Specify the distance between two nearest constellation points as a positive, real, numeric scalar value. The default is 2. This property applies when you set the NormalizationMethod property to Minimum distance between symbols.

## AveragePower

Average power of constellation

Specify the average power of the symbols in the constellation as a positive, real, numeric scalar value. The default is 1 . This property applies when you set the NormalizationMethod property to Average power.

## PeakPower

Peak power of constellation
Specify the maximum power of the symbols in the constellation as a positive, real, numeric scalar value. The default is 1 . This property applies when you set the NormalizationMethod property to Peak power.

## OutputDataType

Data type of output
Specify the output data type as one of double | single | Custom. The default is double.

## Fixed-Point Properties

## CustomOutputDataType

Fixed-point data type of output
Specify the output fixed-point type as a numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the OutputDataType property to Custom.

| Methods | clone | Create PAM modulator object <br> with same property values |
| :--- | :--- | :--- |
| constellation | Calculate or plot ideal signal <br> constellation |  |
| getNumInputs | Number of expected inputs to <br> step method |  |


| getNumOutputs | Number of outputs from step <br> method |
| :--- | :--- |
| isLocked | Locked status for input attributes <br> and nontunable properties |
| release | Allow property value and input <br> characteristics changes |
| step | Modulate using M-ary PAM <br> method |

## Examples

## Algorithms

See Also

Modulate data using 16-PAM modulation, and visualize the data in a scatter plot.

```
% Create binary data for 100, 4-bit symbols
data = randi([0 1],400,1);
% Create a 16-PAM modulator System object with bits as inputs and
% Gray-coded signal constellation
hModulator = comm.PAMModulator(16,'BitInput',true);
% Modulate and plot the data
modData = step(hModulator, data);
constellation(hModulator)
```

This object implements the algorithm, inputs, and outputs described on the M-PAM Modulator Baseband block reference page. The object properties correspond to the block parameters.
comm. PAMDemodulator

## comm.PAMModulator.clone

Purpose Create PAM modulator object with same property values

## Syntax <br> C = clone(H)

Description
$\mathrm{C}=$ clone $(\mathrm{H})$ creates a PAMModulator object C , with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

| Purpose | Calculate or plot ideal signal constellation |
| :---: | :---: |
| Syntax | $\begin{aligned} & y=\text { constellation(h) } \\ & \text { constellation(h) } \end{aligned}$ |
| Description | $y=$ constellation(h) returns the numerical values of the constellation. <br> constellation(h) generates a constellation plot for the object. |
| Examples | Calculate Ideal Signal Constellation for comm.PAMModulator |
|  | Create a comm. PAMModulator System object, and then calculate its ideal signal constellation. |
|  | Create a comm. PAMModulator System object by entering the following at the MATLAB command line: |
|  | $\mathrm{h}=$ comm. PAMModulator |
|  | Calculate and display the ideal signal constellation by calling the constellation method. |
|  | $\mathrm{a}=$ constellation( h ) |
|  | Plot Ideal Signal Constellation for comm.PAMModulator |
|  | Create a comm. PAMModulator System object, and then plot the ideal signal constellation. |
|  | Create a comm. PAMModulator System object by entering the following at the MATLAB command line: |
|  | $\mathrm{h}=$ comm. PAMModulator |
|  | Plot the ideal signal constellation by calling the constellation method. constellation(h) |

## comm.PAMModulator.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNuminputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.PAMModulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description $\quad$ TF $=$ isLocked $(H)$ returns the locked status, $T F$ of the PAMModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release (H) |
| Description | release (H)Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Modulate using M-ary PAM method

## Syntax $\quad Y=\operatorname{step}(H, X)$

Description $\quad Y=\operatorname{step}(H, X)$ modulates input data, $X$, with the M-PAM modulator System object, H. It returns the baseband modulated output, Y. Depending on the value of the BitInput property, input $X$ can be an integer or bit valued column vector with numeric, logical, or fixed-point data types.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose <br> Description

## Construction

Apply phase and frequency offsets to input signal

The PhaseFrequencyOffset object applies phase and frequency offsets to an incoming signal.

H = comm. PhaseFrequencyOffset creates a phase and frequency offset System object, H. This object applies phase and frequency offsets to an input signal.

H = comm. PhaseFrequencyOffset(Name, Value) creates a phase and frequency offset object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Properties

## PhaseOffset

Phase offset
Specify the phase offset in degrees. The default is 0 . If the step method input is an $M$-by- $N$ matrix, the PhaseOffset property can be set to a numeric scalar, an $M$-by-1, or 1-by- $N$ numeric vector, or an $M$-by- $N$ numeric matrix.

When you set the PhaseOffset property to a scalar value, the object applies the constant specified phase offset to each column of the input matrix.

When you set this property to an $M$-by- 1 vector, the object applies time varying phase offsets, specified in the vector of this property, to each column of the input to the step method.

When you set this property to a 1 -by- $N$ vector, the object applies the $i$-th constant phase offset of this property to the $i$-th column of the input to the step method.

When you set this property to an $M$-by- $N$ matrix, the object applies the $i$-th time varying phase offsets, specified in the $i$-th column of this property, to the $i$-th column of the input to the step method. This property is tunable.

## FrequencyOffsetSource

Source of frequency offset
Specify the source of the frequency offset as one of Property | Input port. The default is Property. If you set this property to Property, you can specify the frequency offset using the FrequencyOffset property. If you set this property to Input port, you specify the frequency offset as a step method input.

## FrequencyOffset

Frequency offset
Specify the frequency offset in Hertz. The default is 0. If the step method input is an $M$-by- $N$ matrix, then the FrequencyOffset property is a numeric scalar, an $M$-by-1, or 1-by- $N$ numeric vector, or an $M$-by- $N$ numeric matrix.

This property applies when you set the FrequencyOffsetSource property to Property.

When you set this property to a scalar value, the object applies the constant specified frequency offset to each column of the input to the step method.

When you set this property to an $M$-by- 1 vector, the object applies time-varying frequency offsets. These offsets are specified in the property, to each column of the input to the step method.

When you set this property to a 1 -by- $N$ vector, the object applies the $i$-th constant frequency offset in this property to the $i$-th column of the input to the step method.

When you set this property to an $M$-by- $N$ matrix, the object applies the $i$-th time varying frequency offset. This offset is specified in the $i$-th column of this property and to the $i$-th column of input to the step method. This property is tunable.

## SampleRate

Sample rate

Specify the sample rate of the input samples in seconds as a double-precision, real, positive scalar value. The default is 1 .

SampleRate $=$ Input Vector Size $/$ Simulink Sample Time

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| step |  |

Create phase and frequency offset object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Apply phase and frequency offsets to input signal

## Examples Introduce phase offset to a rectangular 16-QAM signal.

```
    data = (0:15)';
    M = 16; % Modulation order
    hMod = comm.RectangularQAMModulator(M);
    hPFO = comm.PhaseFrequencyOffset('PhaseOffset', 20,'SampleRate',16
% Modulate data
    modData = step(hMod, data);
    scatterplot(modData);
    title(' Original Constellation');xlim([-5 5]);ylim([-5 5])
% Introduce phase offset
    impairedData = step(hPFO,modData);
    scatterplot(impairedData);
    title('Constellation after phase offset');xlim([-5 5]);ylim([-5 5)
```

Algorithms This object implements the algorithm, inputs, and outputs described on the Phase/Frequency Offset block reference page. The object properties correspond to the block parameters, except:

The object provides a SampleRate property, which you must specify. The block senses the sample time of the signal and therefore does not have a corresponding parameter.

See Also

comm.ThermalNoise | comm.PhaseNoise |
comm.MemorylessNonlinearity

Purpose
Create phase and frequency offset object with same property values

## Syntax <br> C = clone(H)

Description

C = clone (H) creates a PhaseFrequencyOffset object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.PhaseFrequencyOffset.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.PhaseFrequencyOffset.getNumOutputs

Purpose Number of outputs from step method<br>Syntax $\quad N=$ getNumOutputs $(H)$<br>Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the PhaseFrequencyOffset System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.


#### Abstract

Purpose Apply phase and frequency offsets to input signal Syntax Y = step( $\mathrm{H}, \mathrm{X}$ ) Y = step(H,X,FRQ) $Y=\operatorname{step}(H, X)$ applies phase and frequency offsets to input $X$, and returns $Y$. The input $X$ is a double or single precision matrix $X$, of dimensions MxN. M is the number of time samples in the input signals and N is number of channels. Both M and N can be equal to 1 . The object adds phase and frequency offsets independently to each column of $X$. The data type and dimensions of $X$ and $Y$ are the same. $Y=\operatorname{step}(H, X, F R Q)$ uses $F R Q$ as the frequency offset that the object applies to input $X$ when you set the FrequencyOffsetSource property to 'Input port'. When the $X$ input is an MxN matrix, the value for $F R Q$ can be a numeric scalar, an Mx 1 or 1 xN numeric vector, or an MxN numeric matrix. When the FRQ input is a scalar, the object applies a constant frequency offset, FRQ, to each column of $X$. When the FRQ input is an Mx1 vector, the object applies time varying frequency offsets, which are specified in the FRQ vector, to each column of $X$. When the FRQ input is a 1 xN vector, the object applies the ith constant frequency offset in FRQ to the ith column of $X$. When the FRQ input is an MxN matrix, the object applies the ith time varying frequency offsets, specified in the ith column of $F R Q$, to the ith column of $X$.


Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose <br> Description

## Construction

## Properties

Phase noise level
Specify the phase noise level in decibels relative to carrier per Hertz ( $\mathrm{dBc} / \mathrm{Hz}$ ) at a frequency offset specified by the FrequencyOffset property. The default is [-60-80]. This property requires a negative, real scalar or vector of data type double.

## FrequencyOffset

Frequency offset
Specify the frequency offset in Hertz as a nonnegative, real scalar or increasing vector of data type double. The default is [20 200].

## SampleRate

Sample rate

Specify the sample rate in Hertz as a positive, real scalar or vector of data type double. The default is 1024. The System object does not use this property when you specify Level and FrequencyOffset as scalars.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>step

Create phase noise object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Apply phase noise to a complex, baseband signal

## Examples

Add a phase noise vector and frequency offset vector to a 16-QAM signal. Then, plot the signal.

```
% Create 16-QAM modulator
    hMod = comm.RectangularQAMModulator(16, ...
            'NormalizationMethod','Average power', 'AveragePower',10);
% Create phase noise System object
    hPhNoise = comm.PhaseNoise('Level',[-60 -80], ...
            'FrequencyOffset',[20 200], ...
            'SampleRate',1024);
%Generate modulated symbols
    modData = step(hMod, randi([0 15], 1000, 1));
% Apply phase noise and plot the result
    y = step(hPhNoise, modData);
    scatterplot(y)
```


## Algorithms

This object implements the algorithm, inputs, and outputs described on the Phase Noise block reference page. The object properties correspond to the block parameters, except:

- The object respects the data types and does perform any casting other than casting the output to the input data type. The result of $\exp (1 i \times$ phase_noise $)$ is cast to the input data type first, before multiplying with the input signal. This order prevents the output (phase distorted) signal from being downcast to single precision if any of the properties are of data type single while the input data type is double precision.
- This object uses the MATLAB default random stream to generate random numbers. The block uses a random number generator based on the V5 RANDN (Ziggurat) algorithm. In addition, the block uses an initial seed, set with the Initial seed parameter to initialize the random number generator. Every time the system that contains the block runs, the block generates the same sequence of random numbers. To generate reproducible numbers using this object, reset the MATLAB default random stream using the following code.

```
reset(RandStream.getGlobalStream)
```

For more information, see help for RandStream.
See Also comm. PhaseFrequencyOffset | comm.MemorylessNonlinearity

## comm.PhaseNoise.clone

Purpose Create phase noise object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a PhaseNoise object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs (H)
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ )

## comm.PhaseNoise.getNumOutputs

Purpose $\quad$ Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.
Purpose Locked status for input attributes and nontunable properties

Syntax TF $=$ isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the PhaseNoise System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description
release (H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose

Apply phase noise to a complex, baseband signal

## Syntax

Description
$Y=\operatorname{step}(H, X)$
$Y=\operatorname{step}(H, X)$ adds phase noise with the specified level, at the
specified frequency offset, to the input $X$ and returns the result in $Y . X$ must be a complex scalar or column vector of data type double or single. The step method outputs, Y , with the same data type and dimensions as the input.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Generate a pseudo-noise (PN) sequence

## Construction

## Properties

The PNSequence object generates a sequence of pseudorandom binary numbers using a linear-feedback shift register (LFSR). This block implements LFSR using a simple shift register generator (SSRG, or Fibonacci) configuration. You can use a pseudonoise sequence in a pseudorandom scrambler and descrambler. You can also use one in a direct-sequence spread-spectrum system.

H = comm. PNSequence creates a pseudo-noise (PN) sequence generator System object, H. This object generates a sequence of pseudorandom binary numbers using a linear-feedback shift register (LFSR).

H = comm. PNSequence (Name, Value) creates a PN sequence generator object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

## Polynomial

Generator polynomial
Specify the polynomial that determines the shift register's feedback connections. The default is [ $\left.\begin{array}{lllllll}1 & 0 & 0 & 0 & 0 & 1 & 1\end{array}\right]$. You can specify the generator polynomial as a numeric, binary vector that lists the coefficients of the polynomial in descending order of powers. The first and last elements must equal 1, and the length of this vector must be $n+1$. The value $n$ indicates the degree of the generator polynomial. Alternatively, you can specify the generator polynomial as a numeric vector containing the exponents of $z$ for the nonzero terms of the polynomial in descending order of powers. The last entry must be 0 . For example, $\left[\begin{array}{lllllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0 & 1\end{array}\right]$
and $\left.\begin{array}{lll}8 & 2 & 0\end{array}\right]$ represent the same polynomial, $g(z)=z^{8}+z^{2}+1$.
The PN sequence has a period of $N=2^{n}-1$.

## InitialConditionsSource

Source of initial conditions

Specify the source of the initial conditions that determines the start of the PN sequence as one of Property | Input port. The default is Property. When you set this property to Property, the initial conditions can be specified as a scalar or binary vector using the InitialConditions property. When you set this property to Input port, you specify the initial conditions as an input to the step method. The object accepts a binary scalar or a binary vector input. The length of the input must equal the degree of the generator polynomial that the Polynomial property specifies.

## InitialConditions

Initial conditions of shift register
Specify the initial values of the shift register as a binary, numeric scalar or a binary, numeric vector. The default is $\left[\begin{array}{llll}0 & 0 & 0 & 0\end{array}\right.$ 1]. Set the vector length equal to the degree of the generator polynomial. If you set this property to a vector, each element of the vector corresponds to the initial value of the corresponding cell in the shift register. If you set this property to a scalar, the initial conditions of all the cells of the shift register are the specified scalar value. The scalar, or at least one element of the specified vector, must be nonzero for the object to generate a nonzero sequence.

## MaskSource

Source of mask to shift PN sequence
Specify the source of the mask that determines the shift of the PN sequence as one of Property | Input port. The default is Property. When you set this property to Property, the mask can be specified as a scalar or binary vector using the Mask property. When you set this property to Input port, the mask, which is an input to the step method, can only be specified as a binary vector. This vector must have a length equal to the degree of the generator polynomial specified in the Polynomial property.

## Mask

Mask to shift PN sequence

Specify the mask that determines how the PN sequence is shifted from its starting point as a numeric, integer scalar or as a binary vector. The default is 0 .

When you set this property to an integer scalar, the value is the length of the shift. A scalar shift can be positive or negative.

When the PN sequence has a period of $N=2^{n}-1$, where $n$ is the degree of the generator polynomial that you specify in the Polynomial property, the object wraps shift values that are negative or greater than $N$.

When you set this property to a binary vector, its length must equal the degree of the generator polynomial specified in the
Polynomial property. The mask vector that represents $m(z)=z^{D}$ modulo $g(z)$, where $g(z)$ is the generator polynomial, and the mask vector corresponds to a shift of $D$. For example, for a generator polynomial of degree of 4 , the mask vector corresponding to $D=2$ is [ $\left.\begin{array}{llll}0 & 1 & 0 & 0\end{array}\right]$, which represents the polynomial $m(z)=z^{2}$.

You can calculate the mask vector using the shift2mask function. This property applies when you set the MaskSource property to Property.

## VariableSizeOutput

Enable variable-size outputs
Set this property to true to enable an additional input to the step method. The default is false. When you set this property to true, the enabled input specifies the output size of the PN sequence used for the step. The input value must be less than or equal to the value of the MaximumOutputSize property.

When you set this property to false, the SamplesPerFrame property specifies the number of output samples.

## MaximumOutputSize

Maximum output size

Specify the maximum output size of the PN sequence as a positive integer 2 -element row vector. The second element of the vector must be 1 . The default is [10 1].
This property applies when you set the VariableSizeOutput property to true.

## SamplesPerFrame

Number of outputs per frame
Specify the number of PN sequence samples that the step method outputs as a numeric, positive, integer scalar value. The default is 1 . If you set this property to a value of $M$, then the step method outputs $M$ samples of a PN sequence that has a period of
$N=2^{n}-1$. The value $n$ represents the degree of the generator polynomial that you specify in the Polynomial property. If you set the BitPackedOutput property to false, the samples are bits from the PN sequence. If you set the BitPackedOutput property to true, then the output corresponds to SamplesPerFrame groups of bit-packed samples.

## ResetInputPort

Enable generator reset input
Set this property to true to enable an additional input to the step method. The default is false. This input resets the states of the PN sequence generator to the initial conditions specified in the InitialConditions property.

## BitPackedOutput

Output integer representations of bit-packed words
Set this property to true to enable bit-packed outputs. The default is false. In this case, the step method outputs a column vector of length $M$, which contains integer representations of bit words of length $P . M$ is the number of samples per frame specified in the SamplesPerFrame property. $P$ is the size of the bit-packed words
specified in the NumPackedBits property. The first bit from the left in the bit-packed word is considered the most significant bit.

## NumPackedBits

Number of bits per bit-packed word
Specify the number of bits to pack into each output data word as a numeric, integer scalar value between 1 and 32. The default is 8 . This property applies when you set the BitPackedOutput property to true.

## SignedOutput

Output signed bit-packed words
Set this property to true to obtain signed, bit-packed, output words. The default is false. In this case, a 1 in the most significant bit (sign bit) indicates a negative value. The property indicates negative numbers in a two's complement format. This property applies when you set the BitPackedOutput property to true.

## OutputDataType

Data type of output
Specify the output data type as one of double | logical | Smallest unsigned integer when the BitPackedOutput property is false. The default is double. Specify the output data type as double | Smallest unsigned integer when the BitPackedOutput property is set to true.

You must have a Fixed-Point Designer user license to use this property in Smallest unsigned integer mode.

## Methods clone

getNumInputs

Create PN sequence generator object with same property values

Number of expected inputs to step method

getNumOutputs<br>isLocked<br>release<br>reset<br>step

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of PN sequence generator object

Generate a pseudo-noise (PN) sequence

## Examples

Algorithms

See Also

Get 2 frames of 14 samples of a maximal length PN-sequence of period $2^{3}-1$ (i.e., get two periods of the sequence).

```
hpn = comm.PNSequence('Polynomial',[3 2 0], ...
    'SamplesPerFrame', 14, 'InitialConditions',[0 0 1]);
x1 = step(hpn);
x2 = step(hpn);
[x1 x2]
```

This object implements the algorithm, inputs, and outputs described on the PN Sequence Generator block reference page. The object properties correspond to the block parameters, except:

- The object does not have a property to select frame based outputs.
- The object does not have a property that corresponds to the Sample time parameter.
comm.KasamiSequence | comm.GoldSequence


## comm.PNSequence.clone

Purpose Create PN sequence generator object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a PNSequence object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.PNSequence.getNumInputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description
$N=$ getNumInputs $(H)$ returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.PNSequence.getNumOutputs

Purpose $\quad$ Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the PNSequence System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset states of PN sequence generator object |
| :--- | :--- |
| Syntax | $\operatorname{reset}(H)$ |
| Description | reset $(H)$ resets the states of the PNSequence object, H. |

## Purpose Generate a pseudo-noise (PN) sequence

## Syntax

Y = step(H)
Y = step(H,MASK)
Y = step(H,RESET)
Y = step(H,MASK,RESET)

## Description

$Y=$ step $(H)$ outputs a frame of the PN sequence in column vector Y. Specify the frame length with the SamplesPerFrame property. The PN sequence has a period of $\mathrm{N}=2^{\wedge} \mathrm{n}-1$, where n is the degree of the generator polynomial that you specify in the Polynomial property.

Y = step(H,MASK) uses MASK as the shift value when you set the MaskSource property to 'Input port'. MASK must be a numeric, binary vector with length equal to the degree of the generator polynomial specified in the Polynomial property. Refer to the Mask property help for details of the mask calculation.

Y = step( $H$, RESET) uses RESET as the reset signal when you set the ResetInputPort property to true. The data type of the RESET input must be double precision or logical. RESET can be a scalar value or a column vector with length equal to the number of samples per frame specified in the SamplesPerFrame property. When the RESET input is a non zero scalar, the object resets to the initial conditions that you specify in the InitialConditions property and then generates a new output frame. A column vector RESET input allows multiple resets within an output frame. A non-zero value at the ith element of the vector will cause a reset at the ith output sample time. You can combine optional input arguments when you set their enabling properties. Optional inputs must be listed in the same order as the order of the enabling properties. For example,

```
Y = step(H,MASK,RESET)
```


#### Abstract

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.


Purpose Recover carrier phase of baseband PSK signal

## Construction

## Properties

The PSKCarrierPhaseSynchronizer object recovers the carrier phase of the input signal using the M-Power method. This feedforward method is not data aided but is clock aided. You can use this method for systems that use baseband phase shift keying (PSK) modulation. The method is also suitable for systems that use baseband quadrature amplitude modulation (QAM). However, the results are less accurate than those for comparable PSK systems. The alphabet size for the modulation requires an even integer.
H = comm. PSKCarrierPhaseSynchronizer creates a PSK carrier phase synchronizer System object, H. This object recovers the carrier phase of a baseband phase shift keying (PSK) modulated signal using the M-power method.
H = comm.PSKCarrierPhaseSynchronizer(Name, Value) creates a PSK carrier phase synchronizer object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).
H = comm.PSKCarrierPhaseSynchronizer(M,Name, Value) creates a PSK carrier phase synchronizer object, H. This object has the ModulationOrder property set to M , and the other specified properties set to the specified values.

## ModulationOrder

Number of points in signal constellation
Specify the modulation order of the input signal as an even, positive, real scalar value. Choose a data type of single or double. The default is 2 . This property is tunable.

## ObservationInterval

Number of symbols where carrier phase assumed constant
Specify the observation interval as a real positive scalar integer value. Choose a data type of single or double. The default is 100 .

| Methods clone |  |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
| isLocked |  |
| release |  |
| reset |  |
| step |  |

Create PSK carrier phase synchronizer object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
Reset states of the PSK carrier phase synchronizer object

Recover baseband PSK signal's carrier phase

Examples Recover carrier phase of a 16-PSK signal using M-power method.

```
M = 16;
phOffset = 10 *pi/180; % in radians
numSamples = 100;
% Create PSK modulator System object
    hMod = comm.PSKModulator(M, phOffset, 'BitInput',false);
% Create PSK carrier phase synchronizer System object
    hSync = comm.PSKCarrierPhaseSynchronizer(M,...
                                    'ObservationInterval',numSamples);
% Generate random data
    data = randi([0 M-1],numSamples,1);
% Modulate random data and add carrier phase
    modData = step(hMod, data);
% Recover the carrier phase
    [recSig phEst] = step(hSync, modData);
fprintf('The carrier phase is estimated to be %g degrees.\n', phEst);
```


## comm.PSKCarrierPhaseSynchronizer

> Algorithms This object implements the algorithm, inputs, and outputs described on the M-PSK Phase Recovery block reference page. The object properties correspond to the block parameters.

> See Also comm.CPMCarrierPhaseSynchronizer | comm.PSKModulator

Purpose
Create PSK carrier phase synchronizer object with same property values
Syntax $\quad C=$ clone $(H)$
Description $\quad C=$ clone $(H)$ creates a PSKCarrierPhaseSynchronizer object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.PSKCarrierPhaseSynchronizer.getNumOutputs

## Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the PSKCarrierPhaseSynchronizer System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release (H) Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.PSKCarrierPhaseSynchronizer.reset

Purpose Reset states of the PSK carrier phase synchronizer object

## Syntax reset (H)

Description reset(H) resets the states of the PSKCarrierPhaseSynchronizer object, H .

## Purpose Recover baseband PSK signal's carrier phase

## Syntax <br> [ $\mathrm{Y}, \mathrm{PH}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$

Description
$[\mathrm{Y}, \mathrm{PH}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$ recovers the carrier phase of the input signal, $X$, and returns the phase corrected signal, $Y$, and the carrier phase estimate (in degrees), PH. X must be a complex scalar or column vector input signal of data type single or double.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Demodulate using M-ary PSK method
Description
ConstructionH = comm.PSKDemodulator creates a demodulator System object, H.This object demodulates the input signal using the M-ary phase shiftkeying (M-PSK) method.
H = comm.PSKDemodulator (Name, Value) creates an M-PSK demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).
H = comm. PSKDemodulator (M, PHASE, Name, Value) creates an M-PSK demodulator object, H. This object has the ModulationOrder property set to M, the PhaseOffset property set to PHASE, and the other specified properties set to the specified values. $M$ and PHASE are value-only arguments. To specify a value-only argument, you must also specify all preceding value-only arguments. You can specify name-value pair arguments in any order.

## Properties ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation as a positive, integer scalar value. The default is 8 .

## PhaseOffset

Phase of zeroth point of constellation
Specify the phase offset of the zeroth point of the constellation, in radians, as a real scalar value. The default is pi/8.

## BitOutput

## Output data as bits

Specify whether the output consists of groups of bits or integer symbol values. The default is false. When you set this property to true, the step method outputs a column vector of bit values. The length of this vector equals $\log 2$ (ModulationOrder) times the number of demodulated symbols. When you set this property to false, the step method outputs a column vector with a length equal to the input data vector. This vector contains integer symbol values between 0 and ModulationOrder-1.

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of $\log 2$ (ModulationOrder) bits to the corresponding symbol. Choose from Binary I Gray I Custom. The default is Gray. When you set this property to Gray, the object uses a Gray-encoded signal constellation. When you set this property to Binary, the integer $m$, between $0 \leq m \leq$ ModulationOrder-1) maps to the complex value $\exp (j \times$ PhaseOffset $+j \times 2 \times \pi \times m /$ ModulationOrder $)$. When you set this property to Custom, the object uses the signal constellation defined in the CustomSymbolMapping property.

## CustomSymbolMapping

Custom constellation encoding
Specify a custom constellation symbol mapping vector. The default is $0: 7$. This property requires a row or column vector with a size of ModulationOrder. This vector must have unique integer values in the range [ 0 , ModulationOrder-1]. The values must be of data type double. The first element of this vector corresponds to the constellation point at an angle of $0+$ PhaseOffset, with subsequent elements running counterclockwise. The last element corresponds to the constellation point at an angle of $-\pi /$ ModulationOrder + PhaseOffset. This property applies when you set the SymbolMapping property to Custom.

## DecisionMethod

Demodulation decision method

Specify the decision method the object uses as Hard decision | Log-likelihood ratio | Approximate log-likelihood ratio. The default is Hard decision. When you set the BitOutput property to false, the object always performs hard decision demodulation. This property applies when you set the BitOutput property to true.

## VarianceSource

Source of noise variance
Specify the source of the noise variance as one of Property | Input port. The default is Property. This property applies when you set the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio.

## Variance

## Noise variance

Specify the variance of the noise as a positive, real scalar value. The default is 1 . If this value is very small (i.e., SNR is very high), log-likelihood ratio (LLR) computations may yield Inf or -Inf. This result occurs because the LLR algorithm computes the exponential of very large or very small numbers using finite-precision arithmetic. In such cases, use approximate LLR instead because the algorithm for that option does not compute exponentials. This property applies when you set the BitOutput property to true, the DecisionMethod property to Log-likelihood ratio, or Approximate log-likelihood ratio, and the VarianceSource property to Property. This property is tunable.

## OutputDataType

Data type of output
Specify the output data type as Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32. The default is Full precision. This property applies when you set the BitOutput property to false. It
also applies when you set the BitOutput property to true and the DecisionMethod property to Hard decision. In this second case, when the OutputDataType property is set to Full precision, the input data type is single- or double-precision, the output data has the same data type as the input. . When the input data is of a fixed-point type, the output data type behaves as if you had set the OutputDataType property to Smallest unsigned integer.

When you set BitOutput to true and the DecisionMethod property to Hard Decision, then logical data type becomes a valid option. If you set the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio, the output data has the same data type as the input. In this case, the data type must be single- or double-precision.

## Fixed-Point Properties

## DerotateFactorDataType

Data type of derotate factor
Specify the derotate factor data type as Same word length as input | Custom. The default is Same word length as input. This property applies when you set the BitOutput property to false. It also applies when you set the BitOutput property to true and the DecisionMethod property to Hard decision. The object uses the derotate factor in the computations only when the ModulationOrder property is 2,4 , or 8 . The step method input must also have a fixed-point type, and the PhaseOffset property must have a nontrivial value. For ModulationOrder $=2$, the phase offset is trivial if that value is a multiple of $\pi / 2$. For ModulationOrder $=4$, the phase offset is trivial if that value is an even multiple of $\pi / 4$. For ModulationOrder $=8$, there are no trivial phase offsets.

## CustomDerotateFactorDataType

Fixed-point data type of derotate factor

Specify the derotate factor fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the DerotateFactorDataType property to Custom. The word length must be a value between 2 and 128 .

Methods<br>clone<br>constellation<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>step

Create PSK demodulator object with same property values
Calculate or plot ideal signal constellation

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Demodulate using M-ary PSK method

Examples Modulate and demodulate a signal using 16-PSK modulation.

```
hMod = comm.PSKModulator(16, 'PhaseOffset',pi/16);
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
            'Signal to noise ratio (SNR)','SNR',15);
    hDemod = comm.PSKDemodulator(16, 'PhaseOffset',pi/16);
    %Create an error rate calculator
    hError = comm.ErrorRate;
    for counter = 1:100
        % Transmit a 50-symbol frame
        data = randi([0 hMod.ModulationOrder-1],50,1);
        modSignal = step(hMod, data);
```

```
    noisySignal = step(hAWGN, modSignal);
    receivedData = step(hDemod, noisySignal);
    errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```


# Algorithms 

This object implements the algorithm, inputs, and outputs described on the M-PSK Demodulator Baseband block reference page. The object properties correspond to the block parameters.

See Also
comm.PSKModulator | comm.DPSKDemodulator

## comm.PSKDemodulator.clone

Purpose Create PSK demodulator object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a PSKDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.
Purpose Calculate or plot ideal signal constellation
Syntax

y = constellation(h)

constellation(h)
Description $y=$ constellation( $h$ ) returns the numerical values of the constellation.
constellation(h) generates a constellation plot for the object.

## Examples Calculate Ideal Signal Constellation for comm.PSKDemodulator

Create a comm. PSKDemodulator System object, and then calculate its ideal signal constellation.
Create a comm.PSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.PSKDemodulator
Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.PSKDemodulator

Create a comm. PSKDemodulator System object, and then plot the ideal signal constellation.
Create a comm.PSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.PSKDemodulator
Plot the ideal signal constellation by calling the constellation method.

```
constellation(h)
```


## comm.PSKDemodulator.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.PSKDemodulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked (H) returns the locked status, TF of the PSKDemodulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Demodulate using M-ary PSK method

Syntax $\quad$| $Y$ | $=\operatorname{step}(H, X)$ |
| ---: | :--- |
| $Y$ | $=\operatorname{step}(H, X, V A R)$ |

Description
$Y=\operatorname{step}(H, X)$ demodulates data, $X$, with the PSK demodulator System object, H, and returns Y. Input X must be a scalar or a column vector with double or single precision data type. If the value of the ModulationOrder property is less than or equal to 8 and you set BitOutput to false, or when you set the DecisionMethod property to Hard Decision and BitOutput to true, the object accepts an input with a signed integer data type or signed fixed point (fi objects). Depending on the BitOutput property value, output Y , can be integer or bit valued.
$Y=\operatorname{step}(H, X, V A R)$ uses soft decision demodulation and noise variance VAR. This syntax applies when you set the BitOutput property to true, the DecisionMethod property to Approximate log-likelihood ratio or Log-likelihood ratio, and the VarianceSource property to Input port. The data type of input VAR must be double or single precision.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Modulate using M-ary PSK method

Description

Construction $H=$ comm.PSKModulator creates a modulator System object, H. This object modulates the input signal using the M -ary phase shift keying (M-PSK) method.

H = comm.PSKModulator (Name, Value) creates an M-PSK modulator object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.PSKModulator(M, PHASE, Name, Value) creates an M-PSK modulator object, H . This object has the ModulationOrder property set to M, the PhaseOffset property set to PHASE, and the other specified properties set to the specified values.

## Properties

## ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation as a positive, integer scalar value. The default is 8 .

## PhaseOffset

Phase of zeroth point of constellation
Specify the phase offset of the zeroth point of the constellation, in radians, as a real scalar value. The default is pi/8.

## BitInput

Assume bit inputs
Specify whether the input is bits or integers. When you set this property to true, the step method input must be a column vector
of bit values. This vector must have a length that is an integer multiple of $\log 2$ (ModulationOrder). This vector contains bit representations of integers between 0 and ModulationOrder-1. When you set the BitInput property to false, the step method input must be a column vector of numeric data type integer symbol values. These values must be between 0 and ModulationOrder-1. The default is false.

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of $\log 2$ (ModulationOrder) input bits to the corresponding symbol as one of Binary | Gray | Custom. The default is Gray. When you set this property to Gray, the object uses a Gray-encoded signal constellation. When you set this property to Binary, the integer $m$, between $0 \leq m \leq$ ModulationOrder-1) maps to the complex value $\exp (j \times$ PhaseOffset $+j \times 2 \times \pi \times m /$ ModulationOrder $)$. When you set this property to Custom, the object uses the signal constellation defined in the CustomSymbolMapping property.

## CustomSymbolMapping

Custom constellation encoding
Specify a custom constellation symbol mapping vector. This property requires a row or column vector of size ModulationOrder and must have unique integer values in the range [ 0 , ModulationOrder-1]. The values must be of data type double. The first element of this vector corresponds to the constellation point at an angle of $0+$ PhaseOffset, with subsequent elements running counterclockwise. The last element corresponds to the constellation point at an angle of $-\pi$ /ModulationOrder + PhaseOffset. This property applies when you set the SymbolMapping property to Custom. The default is 0:7.

## OutputDataType

Data type of output

Specify the output data type as double | single | Custom. The default is double.

## Fixed-Point Properties

## CustomOutputDataType

Fixed-point data type of output
Specify the output fixed-point type as a numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the OutputDataType property to Custom.

## Methods

clone
constellation
getNumInputs
getNumOutputs
isLocked
release
step

Create PSK modulator object with same property values

Calculate or plot ideal signal constellation

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Modulate using M-ary PSK method

## Examples

Modulate data using 16-PSK modulation, and visualize the data in a scatter plot.

```
% Create binary data for 24, 4-bit symbols
    data = randi([0 1],96,1);
```

```
% Create a 16-PSK modulator System object with bits as inputs and Gray-c
    hModulator = comm.PSKModulator(16,'BitInput',true);
% Change the phase offset to pi/16
    hModulator.PhaseOffset = pi/16;
% Modulate and plot the data
    modData = step(hModulator, data);
    constellation(hModulator)
```

> Algorithms

This object implements the algorithm, inputs, and outputs described on the M-PSK Modulator Baseband block reference page. The object properties correspond to the block parameters.

See Also

comm.PSKDemodulator | comm. QPSKModulator

Purpose Create PSK modulator object with same property values
Syntax $\quad C=$ clone $(H)$
Description
$C=$ clone (H) creates a PSKModulator object C, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.PSKModulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax $\quad y=$ constellation $(h)$
constellation(h)
Description $\quad y=$ constellation( h ) returns the numerical values of the constellation.
constellation(h) generates a constellation plot for the object.

## Examples Calculate Ideal Signal Constellation for comm.PSKModulator

Create a comm. PSKModulator System object, and then calculate its ideal signal constellation.

Create a comm. PSKModulator System object by entering the following at the MATLAB command line:
h = comm.PSKModulator

Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.PSKModulator

Create a comm. PSKModulator System object, and then plot the ideal signal constellation.

Create a comm.PSKModulator System object by entering the following at the MATLAB command line:
h = comm.PSKModulator
Plot the ideal signal constellation by calling the constellation method.
constellation(h)

## comm.PSKModulator.getNumInputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.PSKModulator.getNumOutputs

Purpose $\quad$ Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.PSKModulator.isLocked

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the PSKModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
Purpose Modulate using M-ary PSK method
Syntax Y = step(H,X)
Description $\mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X})$ modulates input data, X , with the PSK modulatorSystem object, H. It returns the baseband modulated output, Y.Depending on the value of the BitInput property, input $X$ can be aninteger or bit valued column vector with numeric, logical, or fixed-pointdata types.
Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Demodulate convolutionally encoded data mapped to M-ary PSKconstellation
Description The PSKTCMDemodulator object uses the Viterbi algorithm to decode atrellis-coded modulation (TCM) signal that was previously modulatedusing a PSK signal constellation.
Construction H = comm. PSKTCMDemodulator creates a trellis-coded, M-ary phaseshift, keying (PSK TCM) demodulator System object, H. This objectdemodulates convolutionally encoded data that has been mapped toan M-PSK constellation.
H = comm.PSKTCMDemodulator(Name,Value) creates a PSK TCMdemodulator object, H , with each specified property set to the specifiedvalue. You can specify additional name-value pair arguments in anyorder as (Name1,Value1,...,NameN,ValueN).
H = comm.PSKTCMDemodulator(TRELLIS, Name, Value) createsa PSK TCM demodulator System object, H. This object has theTrellisStructure property set to TRELLIS and the other specifiedproperties set to the specified values.

## Properties <br> TrellisStructure

Trellis structure of convolutional code
Specify trellis as a MATLAB structure that contains the trellis description of the convolutional code. Use the istrellis function to check whether the trellis structure is valid. The default is the result of poly2trellis([13], [1 0 0; 0 5 2]).

## TerminationMethod

Termination method of encoded frame
Specify the termination method as one of Continuous \| Truncated | Terminated. The default is Continuous.
When you set this property to Continuous, the object saves the internal state metric at the end of each frame. The next frame
uses the same state metric. The object treats each traceback path independently. If the input signal contains only one symbol, use Continuous mode.

When you set this property to Truncated, the object treats each input vector independently. The traceback path starts at the state with the best metric and always ends in the all-zeros state.

When you set property to Terminated, the object treats each input vector independently, and the traceback path always starts and ends in the all-zeros state.

## TracebackDepth

Traceback depth for Viterbi decoder
Specify the scalar, integer number of trellis branches to construct each traceback path. The default is 21 . The traceback depth influences the decoding accuracy and delay. The decoding delay is the number of zero symbols that precede the first decoded symbol in the output.

When you set the TerminationMethod property to Continuous, the decoding delay consists of TracebackDepth zero symbols or TracebackDepth $\times K$ zero bits for a rate $K / N$ convolutional code.

When you set the TerminationMethod property to Truncated or Terminated, no output delay occurs and the traceback depth must be less than or equal to the number of symbols in each input vector.

## ResetInputPort

Enable demodulator reset input
Set this property to true to enable an additional input to the step method. The default is false. When this additional reset input is a nonzero value, the internal states of the encoder reset to initial conditions. This property applies when you set the TerminationMethod property to Continuous.

## ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation used to map the convolutionally encoded data as a positive, integer scalar value. The number of points must be 4,8 , or 16 . The default is 8. The ModulationOrder property value must equal the number of possible input symbols to the convolutional decoder of the PSK TCM demodulator object. The ModulationOrder property must equal $2^{N}$ for a rate $K / N$ convolutional code.

## OutputDataType

Data type of output
Specify output data type as logical | double. The default is double.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>reset<br>step

Create PSK TCM demodulator object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of the PSK TCM demodulator object

Demodulate convolutionally encoded data mapped to M-ary PSK constellation

```
Examples Modulate and demodulate data using 8-PSK TCM modulation.
```

```
% Define a trellis structure with 4-ary input symbols and 8-ary outpu
```

% Define a trellis structure with 4-ary input symbols and 8-ary outpu
t = poly2trellis([5 4],[23 35 0; 0 5 13]);
t = poly2trellis([5 4],[23 35 0; 0 5 13]);
hMod = comm.PSKTCMModulator(t,'ModulationOrder', 8);
hMod = comm.PSKTCMModulator(t,'ModulationOrder', 8);
hAWGN = comm.AWGNChannel('NoiseMethod', ...
hAWGN = comm.AWGNChannel('NoiseMethod', ...
'Signal to noise ratio (SNR)','SNR',6);
'Signal to noise ratio (SNR)','SNR',6);
hDemod = comm.PSKTCMDemodulator(t, 'ModulationOrder', 8, ...
hDemod = comm.PSKTCMDemodulator(t, 'ModulationOrder', 8, ...
'TracebackDepth',16);
'TracebackDepth',16);
% Create an error rate calculator with delay in bits equal to Traceba
% Create an error rate calculator with delay in bits equal to Traceba
hError = comm.ErrorRate('ReceiveDelay',...
hError = comm.ErrorRate('ReceiveDelay',...
hDemod.TracebackDepth*log2(t.numInputSymbols));
hDemod.TracebackDepth*log2(t.numInputSymbols));
for counter = 1:10
for counter = 1:10
% Transmit frames of 250 2-bit symbols
% Transmit frames of 250 2-bit symbols
data = randi([0 1],500,1);
data = randi([0 1],500,1);
modSignal = step(hMod, data);
modSignal = step(hMod, data);
noisySignal = step(hAWGN, modSignal);
noisySignal = step(hAWGN, modSignal);
receivedData = step(hDemod, noisySignal);
receivedData = step(hDemod, noisySignal);
errorStats = step(hError, data, receivedData);
errorStats = step(hError, data, receivedData);
end
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
errorStats(1), errorStats(2))

```
            errorStats(1), errorStats(2))
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the M-PSK TCM Decoder block reference page. The object properties correspond to the block parameters.

[^8]
## comm.PSKTCMDemodulator.clone

Purpose Create PSK TCM demodulator object with same property values

## Syntax <br> C = clone( H )

Description $\quad C=$ clone $(H)$ creates a PSKTCMDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.PSKTCMDemodulator.getNumInputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.PSKTCMDemodulator.getNumOutputs

Purpose $\quad$ Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.PSKTCMDemodulator.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked (H) |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the |
|  | PSKTCMDemodulator System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset states of the PSK TCM demodulator object |
| :--- | :--- |
| Syntax | $\operatorname{reset}(\mathrm{H})$ |
| Description | $\operatorname{reset}(\mathrm{H})$ resets the states of the PSKTCMDemodulator object, H. |

Purpose Demodulate convolutionally encoded data mapped to M-ary PSK constellation

```
Syntax
\(Y=\operatorname{step}(H, X)\)
Y \(=\operatorname{step}(H, X, R)\)
```

$Y=\operatorname{step}(H, X)$ demodulates the PSK modulated input data, $X$, and uses the Viterbi algorithm to decode the resulting demodulated, convolutionally encoded bits. X must be a complex, double or single precision column vector. The step method outputs a demodulated, binary data column vector, Y . When the convolutional encoder represents a rate $K / N$ code, the length of the output vector is $K \times L$, where $L$ is the length of the input vector, $X$.
$Y=\operatorname{step}(H, X, R)$ resets the decoder to the all-zeros state when you input a reset signal, $R$ that is non-zero. $R$ must be a double precision or logical, scalar integer. This syntax applies when you set the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose <br> Description

Convolutionally encode binary data and map using M-ary PSK constellation

The PSKTCMModulator object implements trellis-coded modulation (TCM) by convolutionally encoding the binary input signal and then mapping the result to a PSK signal constellation.

## Construction

H = comm.PSKTCMModulator creates a trellis-coded M-ary phase shift keying (PSK TCM) modulator System object, H. This object convolutionally encodes a binary input signal and maps the result to an M-PSK constellation.

H = comm. PSKTCMModulator(Name, Value) creates a PSK TCM encoder object, $H$, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.PSKTCMModulator(TRELLIS,Name, Value) creates a PSK TCM encoder object, H. This object has the TrellisStructure property set to TRELLIS and the other specified properties set to the specified values.

## Properties

## TrellisStructure

Trellis structure of convolutional code
Specify trellis as a MATLAB structure that contains the trellis description of the convolutional code. Use the istrellis function to check whether a trellis structure is valid. The default is the result of poly2trellis([1 3], [1 0 0; 0 5 2]).

## TerminationMethod

Termination method of encoded frame
Specify the termination method as one of Continuous | Truncated | Terminated. The default is Continuous.

When you set this property to Continuous, the object retains the encoder states at the end of each input vector for use with the next input vector.
When you set this property to Truncated, the object treats each input vector independently. The encoder is reset to the all-zeros state at the start of each input vector.

When you set this property to Terminated, the object treats each input vector independently. However, for each input vector, the object uses extra bits to set the encoder to the all-zeros state at the end of the vector. For a rate $K / N$ code, the step method outputs
the vector with a length given by $y=N \times(L+S) / K$, where $S=$ constraintLength -1 (or, in the case of multiple constraint lengths, $S=\operatorname{sum}($ constraintLength $(i)-1)) . L$ indicates the length of the input to the step method.

## ResetInputPort

Enable modulator reset input
Set this property to true to enable an additional input to the step method. The default is false. When this additional reset input is a nonzero value, the internal states of the encoder reset to initial conditions. This property applies when you set the TerminationMethod property to Continuous.

## ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation used to map the convolutionally encoded data as a positive integer scalar value equal to 4,8 , or 16 . The default is 8 . The value of the ModulationOrder property must equal the number of possible output symbols from the convolutional encoder of the PSK TCM modulator. Thus, the value for the ModulationOrder property must equal $2^{N}$ for a rate $K / N$ convolutional code.

## OutputDataType

Data type of output
Specify the output data type as one of double | single. The default is double.

```
Methods
clone
getNumInputs
getNumOutputs
isLocked
release
reset
step
Create PSK TCM modulator object with same property values
Number of expected inputs to step method
Number of outputs from step method
Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes
Reset states of the PSK TCM modulator object
Convolutionally encode binary data and map using M-ary PSK constellation
Examples Modulate data using 8-PSK TCM modulation.
\% Create binary data
data = randi([0 1], 1000,1);
\% Define a trellis structure with 4-ary input symbols and 8-ary output
t = poly2trellis([5 4],[23 35 0; 05 13]);
hMod = comm. PSKTCMModulator(t,'ModulationOrder', 8);
Modulate and plot the data
modData \(=\) step(hMod, data);
scatterplot(modData);
```

| Algorithms | This object implements the algorithm, inputs, and outputs described on <br> the M-PSK TCM Decoder block reference page. The object properties <br> correspond to the block parameters. |
| :--- | :--- |
| See Also | comm. PSKTCMDemodulator \| comm. GeneralQAMTCMModulator | <br> comm. RectangularQAMTCMModulator \| comm. ConvolutionalEncoder |

## comm.PSKTCMModulator.clone

Purpose Create PSK TCM modulator object with same property values<br>\section*{Syntax $\quad C=$ clone $(H)$}<br>Description $\quad C=$ clone $(H)$ creates a PSKTCMModulator object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.PSKTCMModulator.getNumInputs

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNuminputs $(H)$

Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.PSKTCMModulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties

## Syntax $\quad$ TF $=$ isLocked $(H)$

Description TF = isLocked $(H)$ returns the locked status, TF of the PSKTCMModulator System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release $(H)$ |
| Description | release $(H)$ Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.PSKTCMModulator.reset

Purpose Reset states of the PSK TCM modulator object
Syntax ..... reset (H)
Description reset $(\mathrm{H})$ resets the states of the PSKTCMModulator object, H .

## Purpose <br> Convolutionally encode binary data and map using M-ary PSK

 constellationSyntax<br>Y = step (H,X)<br>Y $=\operatorname{step}(H, X, R)$

## Description

$Y=\operatorname{step}(H, X)$ convolutionally encodes and modulates the input binary data column vector, $X$, and returns the encoded and modulated data, Y. X must be of data type numeric, logical, or unsigned fixed point of word length 1 (fi object). When the convolutional encoder represents a rate $K / N$ code, the length of the input vector, x , must be $K \times L$, for some positive integer $L$. The step method outputs a complex column vector, Y , of length $L$.
$Y=\operatorname{step}(H, X, R)$ resets the encoder of the PSK TCM modulator object to the all-zeros state when you input a reset signal, $R$, that is non-zero. $R$ must be a double precision or logical scalar integer. This syntax applies when you set the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Demodulate using QPSK method

Description

## Construction

The QPSKDemodulator object demodulates a signal that was modulated using the quaternary phase shift keying method. The input is a baseband representation of the modulated signal.

H = comm.QPSKDemodulator creates a demodulator System object, H. This object demodulates the input signal using the quadrature phase shift keying (QPSK) method.

H = comm.QPSKDemodulator (Name, Value) creates a QPSK demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.QPSKDemodulator (PHASE, Name, Value) creates a QPSK demodulator object, H. This object has the PhaseOffset property set to PHASE, and the other specified properties set to the specified values.

## Properties

## PhaseOffset

Phase of zeroth point in constellation
Specify the phase offset of the zeroth point in the constellation, in radians, as a real scalar value. The default is pi/4.

## BitOutput

Output data as bits
Specify whether the output consists of groups of bits or integer symbol values.

When you set this property to true, the step method outputs a column vector of bit values with length equal to twice the number of demodulated symbols.

When you set this property to false, the step method outputs a column vector with length equal to the input data vector. This vector contains integer symbol values between 0 and 3 . The default is false.

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of 2 bits to the corresponding symbol as one of Binary | Gray. The default is Gray.

When you set this property to Gray, the object uses a Gray-encoded signal constellation.

When you set this property to Binary, the integer m, between $0 \leq m \leq 3$ maps to the complex value $\exp (j \times$ PhaseOffset + $j \times 2 \pi \times m / 4)$.

## DecisionMethod

Demodulation decision method
Specify the decision method the object uses as Hard decision | Log-likelihood ratio | Approximate log-likelihood ratio. The default is Hard decision.

When you set the BitOutput property to false, the object always performs hard decision demodulation. This property applies when you set the BitOutput property to true.

## VarianceSource

Source of noise variance
Specify the source of the noise variance as one of Property | Input port. The default is Property. This property applies when you set the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio.

## Variance

Noise variance
Specify the variance of the noise as a positive, real scalar value. The default is 1 . If this value is very small (i.e., SNR is very
high), log-likelihood ratio (LLR) computations may yield Inf or - Inf. This result occurs because the LLR algorithm computes the exponential of very large or very small numbers using finite-precision arithmetic. In such cases, use approximate LLR is because that option's algorithm does not compute exponentials.

This property applies when you set the BitOutput property to true, the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio, and the VarianceSource property to Property.This property is tunable.

## OutputDataType

Data type of output
Specify the output data type as Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32. The default is Full precision.

This property applies when you set the BitOutput property to false. The property also applies when you set the BitOutput property to true and the DecisionMethod property to Hard decision. In this second case, when the OutputDataType property is set to Full precision, and the input data type is single or double precision, the output data has the same as that of the input.

When the input data is of a fixed-point type, the output data type behaves as if you had set the OutputDataType property to Smallest unsigned integer.

When you set BitOutput to true and the DecisionMethod property to Hard Decision, then logical data type becomes a valid option.

When you set the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is the same as that of the input. In this case, that data type can only be single or double precision.

## Fixed-Point Properties

## DerotateFactorDataType

Data type of derotate factor
Specify derotate factor data type as one of Same word length as input | Custom. The default is Same word length as input.

This property applies when you set the BitOutput property to false. The property also applies when you set the BitOutput property to true and the DecisionMethod property to Hard decision. The object uses the derotate factor in the computations only when the step method input is a fixed-point type and the PhaseOffset property has a value that is not an even multiple of pi/4.

## CustomDerotateFactorDataType

Fixed-point data type of derotate factor
Specify the derotate factor fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the DerotateFactorDataType property to Custom.

Methods<br>clone<br>constellation<br>getNumInputs<br>getNumOutputs<br>isLocked

Create QPSK demodulator object with same property values

Calculate or plot ideal signal constellation

Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

| release | Allow property value and input <br> characteristics changes |
| :--- | :--- |
| step | Demodulate using QPSK method |

Examples Modulate and demodulate a signal using QPSK modulation.

```
hMod = comm.QPSKModulator('PhaseOffset',pi/4);
hAWGN = comm.AWGNChannel('NoiseMethod',...
    'Signal to noise ratio (SNR)','SNR',15);
hDemod = comm.QPSKDemodulator('PhaseOffset',pi/4);
%Create an error rate calculator
hError = comm.ErrorRate;
for counter = 1:100
    % Transmit a 50-symbol frame
    data = randi([0 3],50,1);
    modSignal = step(hMod, data);
    noisySignal = step(hAWGN, modSignal);
    receivedData = step(hDemod, noisySignal);
    errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the QPSK Demodulator Baseband block reference page. The object properties correspond to the block parameters.
comm.QPSKModulator | comm.PSKDemodulator

# Purpose Create QPSK demodulator object with same property values 

## Syntax $\quad C=$ clone $(H)$

Description $\quad C=$ clone $(H)$ creates a QPSKDemodulator object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.QPSKDemodulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax $y=$ constellation(h) constellation(h)
Description $y=$ constellation(h) returns the numerical values of theconstellation.
constellation(h) generates a constellation plot for the object.
Examples Calculate Ideal Signal Constellation for comm.QPSKDemodulator
Create a comm. QPSKDemodulator System object, and then calculate its ideal signal constellation.
Create a comm. QPSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.QPSKDemodulator
Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.QPSKDemodulator

Create a comm.QPSKDemodulator System object, and then plot the ideal signal constellation.
Create a comm. QPSKDemodulator System object by entering the following at the MATLAB command line:
h = comm.QPSKDemodulator
Plot the ideal signal constellation by calling the constellation method. constellation(h)

## comm.QPSKDemodulator.getNumInputs

## Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs(H)

## comm.QPSKDemodulator.getNumOutputs

Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNum0utputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

# Purpose Locked status for input attributes and nontunable properties 

$$
\text { Syntax } \quad T F=\text { isLocked }(H)
$$

Description TF = isLocked (H) returns the locked status, TF of the QPSKDemodulator System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.QPSKDemodulator.release

Purpose Allow property value and input characteristics changes
Syntax release(H)
Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose <br> Demodulate using QPSK method

Syntax
Y $=\operatorname{step}(H, X)$
Y = step(H,X,VAR)
$Y=\operatorname{step}(H, X)$ demodulates input data, $X$, with the QPSK demodulator System object, H, and returns Y. Input X must be a scalar or a column vector with double or single precision data type. When you set the BitOutput property to false, or when you set the DecisionMethod property to Hard decision and the BitOutput property to true, the data type of the input can also be signed integer, or signed fixed point (fi objects). Depending on the BitOutput property value, output Y can be integer or bit valued.

Y = step ( $\mathrm{H}, \mathrm{X}, \mathrm{VAR}$ ) uses soft decision demodulation and noise variance VAR. This syntax applies when you set the BitOutput property to true, the DecisionMethod property to Approximate log-likelihood ratioor Log-likelihood ratio, and the VarianceSource property to Input port. The data type of input VAR must be double or single precision.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Modulate using QPSK method

Description

## Construction

The QPSKModulator object modulates using the quaternary phase shift keying method. The output is a baseband representation of the modulated signal.

H = comm.QPSKModulator creates a modulator System object, H. This object modulates the input signal using the quadrature phase shift keying (QPSK) method.

H = comm.QPSKModulator (Name, Value) creates a QPSK modulator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.QPSKModulator(PHASE, Name, Value) creates a QPSK modulator object, H. This object has the PhaseOffset property set to PHASE and the other specified properties set to the specified values.

## Properties

## PhaseOffset

Phase of zeroth point in constellation
Specify the phase offset of the zeroth point in the constellation, in radians, as a real scalar value. The default is $\mathrm{pi} / 4$.

## BitInput

Assume bit inputs
Specify whether the input is bits or integers. The default is false. When you set this property to true, the step method input must be a column vector of bit values. This vector must have a length that is an integer multiple of 2 . This vector contains bit representations of integers between 0 and 3 . When you set this property to false, the step method input must be a column vector of integer symbol values between 0 and 3 .

## SymbolMapping

Constellation encoding

Specify how the object maps an integer or a group of two input bits to the corresponding symbol as one of Binary | Gray. The default is Gray. When you set this property to Gray, the object uses a Gray-encoded signal constellation. When you set this property to Binary, the input integer $m$, between $0 \leq m \leq 3$, maps to the complex value $\exp (j \times$ PhaseOffset $+j \times 2 \times \pi \times m / 4)$.

## OutputDataType

Data type of output
Specify the output data type as one of double | single | Custom. The default is double.

## Fixed-Point Properties

## CustomOutpułDataType

Fixed-point data type of output
Specify the output fixed-point type as a numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the OutputDataType property to Custom.

Methods<br>clone<br>constellation<br>getNumInputs<br>getNumOutputs<br>isLocked

Create QPSK modulator object with same property values
Calculate or plot ideal signal constellation

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

| release | Allow property value and input <br> characteristics changes |
| :--- | :--- |
| step | Modulate using QPSK method |

Examples Modulate data using QPSK, and visualize the data in a scatter plot.

```
% Create binary data for 48, 2-bit symbols
    data = randi([0 1],96,1);
% Create a QPSK modulator System object with bits as inputs and Gray-coo
    hModulator = comm.QPSKModulator('BitInput',true);
% Change the phase offset to pi/16
    hModulator.PhaseOffset = pi/16;
% Modulate and plot the data
    modData = step(hModulator, data);
    scatterplot(modData)
```

Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the QPSK Modulator Baseband block reference page. The object properties correspond to the block parameters.
comm.QPSKDemodulator | comm.PSKModulator

Purpose Create QPSK modulator object with same property values
Syntax $\quad C=$ clone $(H)$
Description $\quad C=$ clone $(H)$ creates a QPSKModulator object $C$, with the same property values as $H$. The clone method creates a new unlocked object with uninitialized states.

## comm.QPSKModulator.constellation

## Purpose Calculate or plot ideal signal constellation

Syntax $\quad y=$ constellation $(h)$ constellation(h)

Description $\quad y=$ constellation( $h$ ) returns the numerical values of the constellation.
constellation(h) generates a constellation plot for the object.

## Examples Calculate Ideal Signal Constellation for comm.QPSKModulator

Create a comm.QPSKModulator System object, and then calculate its ideal signal constellation.

Create a comm.QPSKModulator System object by entering the following at the MATLAB command line:
h = comm.QPSKModulator

Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.QPSKModulator

Create a comm.QPSKModulator System object, and then plot the ideal signal constellation.

Create a comm.QPSKModulator System object by entering the following at the MATLAB command line:
h = comm.PSKModulator
Plot the ideal signal constellation by calling the constellation method.
constellation(h)

Purpose Number of expected inputs to step method

## Syntax $\quad N=$ getNumInputs $(H)$

Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ )

## comm.QPSKModulator.getNumOutputs

Purpose Number of outputs from step method

## Syntax $\quad N=$ getNumOutputs $(H)$

Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF = isLocked (H) |
| Description | TF = isLocked (H) returns the locked status, TF of the QPSKModulator <br> System object. |
| The isLocked method returns a logical value that indicates whether <br> input attributes and nontunable properties for the object are locked. The <br> object performs an internal initialization the first time the step method <br> is executed. This initialization locks nontunable properties and input <br> specifications, such as dimensions, complexity, and data type of the <br> input data. After locking, the isLocked method returns a true value. |  |

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Modulate using QPSK method |
| :--- | :--- |
| Syntax | $Y=$ step $(H, X)$ |$\quad$| $Y=$ step $(H, X)$ modulates input data, $X$, with the QPSK modulator |
| :--- |
| System object, H . It returns the baseband modulated output, $Y$. |
| Depending on the value of the Bit Input property, input $X$ can be an |
| integer or bit valued column vector with numeric, logical, or fixed-point |
| data types. |

## Purpose Demodulate using rectangular QAM method

Description

## Construction

## Properties

The RectangularQAMDemodulator object demodulates a signal that was modulated using quadrature amplitude modulation with a constellation on a rectangular lattice.

H = comm.RectangularQAMDemodulator creates a demodulator System object, H . This object demodulates the input signal using the rectangular quadrature amplitude modulation (QAM) method.

H = comm.RectangularQAMDemodulator(Name, Value) creates a rectangular QAM demodulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.RectangularQAMDemodulator(M,Name, Value) creates a rectangular QAM demodulator object, H. This object has the ModulationOrder property set to M, and the other specified properties set to the specified values.

## ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation as scalar value with a positive, integer power of two. The default is 16 .

## PhaseOffset

Phase offset of constellation
Specify the phase offset of the signal constellation, in radians, as a real scalar value. The default is 0 .

## BitOutput

Output data as bits
Specify whether the output consists of groups of bits or integer symbol values. When you set this property to true the step method outputs a column vector of bit values whose length
equals $\log 2$ (ModulationOrder) times the number of demodulated symbols. When you set this property to false, the step method outputs a column vector with a length equal to the input data vector. This vector contains integer symbol values between 0 and ModulationOrder-1. The default is false.

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of $\log 2(M o d u l a t i o n O r d e r) ~ b i t s ~ t o ~ t h e ~ c o r r e s p o n d i n g ~ s y m b o l ~ a s ~ o n e ~$ of Binary | Gray | Custom. The default is Gray. When you set this property to Gray, the object uses a Gray-coded signal constellation. When you set this property to Binary, the object uses a natural binary-coded constellation. When you set this property to Custom, the object uses the signal constellation defined in the CustomSymbolMapping property.

## CustomSymbolMapping

Custom constellation encoding
Specify a custom constellation symbol mapping vector. The default is $0: 15$. This property is a row or column vector with a size of ModulationOrder and with unique integer values in the range [ 0 , ModulationOrder-1]. The values must be of data type double. The first element of this vector corresponds to the top-leftmost point of the constellation, with subsequent elements running down column-wise, from left to right. The last element corresponds to the bottom-rightmost point. This property applies when you set the Symbolmapping property to Custom.

## NormalizationMethod

Constellation normalization method
Specify the method used to normalize the signal constellation as Minimum distance between symbols | Average power | Peak power. The default is Minimum distance between symbols.

## MinimumDistance

## comm.RectangularQAMDemodulator

Minimum distance between symbols
Specify the distance between two nearest constellation points as a positive, real, numeric scalar value. The default is 2 . This property applies when you set the NormalizationMethod property to Minimum distance between symbols.

## AveragePower

Average power of constellation
Specify the average power of the symbols in the constellation as a positive, real, numeric scalar value. The default is 1 . This property applies when you set the NormalizationMethod property to Average power.

## PeakPower

Peak power of constellation
Specify the maximum power of the symbols in the constellation as a positive, real, numeric scalar value. The default is 1 . This property applies when you set the NormalizationMethod property to Peak power.

## DecisionMethod

Demodulation decision method
Specify the decision method the object uses as Hard decision | Log-likelihood ratio | Approximate log-likelihood ratio. The default is Hard decision. When you set the BitOutput property to false the object always performs hard-decision demodulation. This property applies when you set the BitOutput property to true.

## VarianceSource

Source of noise variance
Specify the source of the noise variance as Property | Input port. The default is Property. This property applies when you set the BitOutput property to true and the DecisionMethod property
to Log-likelihood ratio or Approximate log-likelihood ratio.

## Variance

Noise variance
Specify the variance of the noise as a positive, real scalar value. The default is 1 . If this value is very small (i.e., SNR is very high), log-likelihood ratio (LLR) computations may yield Inf or -Inf. This result occurs because the LLR algorithm computes the exponential of very large or very small numbers using finite-precision arithmetic. In such cases, using approximate LLR is recommended because its algorithm does not compute exponentials. This property applies when you set the BitOutput property to true, the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio, and the VarianceSource property to Property. This property is tunable.

## OutputDataType

Data type of output
Specify the output data type as Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32. The default is Full precision.

This property applies only when you set the BitOutput property to false or when you set the BitOutput property to true and the DecisionMethod property to Hard decision. In this case, when the OutputDataType property is set to Full precision, and the input data type is single- or double-precision, the output data has the same data type as the input.
When the input data is of a fixed-point type, the output data type behaves as if you had set the OutputDataType property to Smallest unsigned integer.

When you set the BitOutput property to true and the DecisionMethod property to Hard Decision, then logical data type becomes a valid option.
When you set the BitOutput property to true and the DecisionMethod property to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is the same as that of the input. In this case, that data type can only be single- or double-precision.

## Fixed-Point Properties

## FullPrecisionOverride

Full precision override for fixed-point arithmetic
Specify whether to use full precision rules. If you set FullPrecisionOverride to true, which is the default, the object computes all internal arithmetic and output data types using full precision rules. These rules provide the most accurate fixed-point numerics. It also turns off the display of other fixed-point properties because they do not apply individually. These rules guarantee that no quantization occurs within the object. Bits are added, as needed, to ensure that no roundoff or overflow occurs. If you set FullPrecisionOverride to false, fixed-point data types are controlled through individual fixed-point property settings. For more information, see "Full Precision for Fixed-Point System Objects".

## DerotateFactorDataType

Data type of derotate factor
Specify the derotate factor data type as Same word length as input | Custom. The default is Same word length as input. This property applies when you set the BitOutput property to false, or when you set the BitOutput property to true and the DecisionMethod property to Hard decision. The object uses the derotate factor in the computations only when the step method
input is of a fixed-point type and the PhaseOffset property has a value that is not a multiple of $\pi / 2$.

## CustomDerotateFactorDataType

Fixed-point data type of derotate factor
Specify the derotate factor fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the DerotateFactorDataType property to Custom.

## DenormalizationFactorDataType

Data type of denormalization factor
Specify the denormalization factor data type as Same word length as input | Custom. The default is Same word length as input. This property applies when you set the BitOutput property to false or when you set the BitOutput property to true and the DecisionMethod property to Hard decision.

## CustomDenormalizationFactorDataType

Fixed-point data type of denormalization factor
Specify the denormalization factor fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the DenormalizationFactorDataType property to Custom.

## ProductDataType

Data type of product
Specify the product data type as Full precision | Custom. The default is Full precision. This property applies when you set the BitOutput property to false or when you set the BitOutput property to true and the DecisionMethod property to Hard decision.

## CustomProductDataType

## comm.RectangularQAMDemodulator

Fixed-point data type of product
Specify the product fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is numerictype([],32). This property applies when you set the ProductDataType property to Custom.

## ProductRoundingMethod

Rounding of fixed-point numeric value of product
Specify the product rounding method as Ceiling | Convergent | Floor | Nearest | Round | Simplest | Zero. The default is Floor. This property applies when the object is not in a full precision configuration, when you set the BitOutput property to false or when you set the BitOutput property to true and the DecisionMethod property to Hard decision.

## ProductOverflowAction

Action when fixed-point numeric value of product overflows
Specify the product overflow action as Wrap \| Saturate. The default is Wrap. This property applies when the object is not in a full precision configuration, when you set the BitOutput property to false or when you set the BitOutput property to true and the DecisionMethod property to Hard decision.

## SumDataType

Data type of sum
Specify the sum data type as Full precision | Same as product | Custom. The default is Full precision. This property applies when you set the FullPrecisionOverride property to false, when you set the BitOutput property to false or when you set the BitOutput property to true and the DecisionMethod property to Hard decision.

## CustomSumDataType

Fixed-point data type of sum

## comm.RectangularQAMDemodulator

Specify the sum fixed-point type as an unscaled numerictype object with a signedness of Auto. The default is numerictype([],32). This property applies when you set the FullPrecisionOverride property to false or when you set the SumDataType property Custom.

| Methods | clone |
| :--- | :--- |
|  | constellation |
|  | getNumInputs |
|  | getNumOutputs |
| isLocked |  |
| release |  |
| step |  |

Create rectangular QAM demodulator object with same property values
Calculate or plot ideal signal constellation

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes
Demodulate using rectangular QAM method

Examples Modulate and demodulate data using rectangular 8-QAM.

```
    hMod = comm.RectangularQAMModulator(8);
    hAWGN = comm.AWGNChannel('NoiseMethod',...
    'Signal to noise ratio (SNR)', 'SNR', 15, 'SignalPowe।
    hDemod = comm.RectangularQAMDemodulator(8);
% Create an error rate calculator
    hError = comm.ErrorRate;
    for counter = 1:100
% Transmit a 50-symbol frame
    data = randi([0 hMod.ModulationOrder-1],50,1);
```

```
        modData = step(hMod, data);
        receivedSignal = step(hAWGN, modData);
        receivedData = step(hDemod, receivedSignal);
        errorStats = step(hError, data, receivedData);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```

Algorithms

See Also
comm. RectangularQAMModulator | comm. GeneralQAMDemodulator

This object implements the algorithm, inputs, and outputs described on the Rectangular QAM Demodulator Baseband block reference page. The object properties correspond to the block parameters.

## comm.RectangularQAMDemodulator.clone

Purpose Create rectangular QAM demodulator object with same property values

## Syntax <br> C = clone(H)

Description
C = clone (H) creates a RectangularQAMDemodulator object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.RectangularQAMDemodulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax y = constellation(h) constellation(h)
Description $y=$ constellation(h) returns the numerical values of theconstellation.
constellation(h) generates a constellation plot for the object.
Examples Calculate Ideal Signal Constellation for comm.RectangularQAMDemodulator
Create a comm.RectangularQAMDemodulator System object, and then calculate its ideal signal constellation.
Create a comm. RectangularQAMDemodulator System object by entering the following at the MATLAB command line:
h = comm.RectangularQAMDemodulator
Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.RectangularQAMDemodulator

Create a comm.RectangularQAMDemodulator System object, and then plot the ideal signal constellation.
Create a comm.RectangularQAMDemodulator System object by entering the following at the MATLAB command line:
$\mathrm{h}=$ comm.RectangularQAMDemodulator
Plot the ideal signal constellation by calling the constellation method. constellation(h)

## comm.RectangularQAMDemodulator.getNumInputs

## Purpose Number of expected inputs to step method

## Syntax <br> N = getNumInputs( H )

Description
$N=$ getNumInputs $(H)$ returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs(H)

## comm.RectangularQAMDemodulator.getNumOutputs

Purpose $\quad$ Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.RectangularQAMDemodulator.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked $(H)$ |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the <br> RectangularQAMDemodulator System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.RectangularQAMDemodulator.release

Purpose Allow property value and input characteristics changes
Syntax release(H)
Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## Purpose Demodulate using rectangular QAM method

Syntax
Y = step (H,X)
Y = step( $\mathrm{H}, \mathrm{X}, \mathrm{VAR}$ )
$Y=\operatorname{step}(H, X)$ demodulates the input data, $X$, with the rectangular QAM demodulator System object, H, and returns, Y. Input X must be a scalar or a column vector with double or single precision data type. When ModulationOrder is an even power of two and you set the BitOutput property to false or, when you set the DecisionMethod to Hard decision and the BitOutput property to true, the data type of the input can also be signed integer, or signed fixed point (fi objects). Depending on the BitOutput property value, output $Y$ can be integer or bit valued.

Y = step( $\mathrm{H}, \mathrm{X}, \mathrm{VAR})$ uses soft decision demodulation and noise variance VAR. This syntax applies when you set the BitOutput property to true, the DecisionMethod property to Approximate log-likelihood ratioor Log-likelihood ratio, and the VarianceSource property to Input port. The data type of input VAR must be double or single precision.

> Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.
Purpose Modulate using rectangular QAM method
Description The RectangularQAMModulator object modulates using M-aryquadrature amplitude modulation with a constellation on a rectangularlattice. The output is a baseband representation of the modulatedsignal. This block accepts a scalar or column vector input signal.
Construction H = comm.RectangularQAMModulator creates a modulator object, H .This object modulates the input using the rectangular quadratureamplitude modulation (QAM) method.
H = comm.RectangularQAMModulator (Name, Value) creates a rectangular QAM modulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).
H = comm.RectangularQAMModulator (M, Name, Value) creates a rectangular QAM modulator object, H . This object has the ModulationOrder property set to M , and the other specified properties set to the specified values.

## Properties <br> ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation as scalar value that is a positive integer power of two. The default is 16 .

## PhaseOffset

Phase offset of constellation
Specify the phase offset of the signal constellation, in radians, as a real scalar value. The default is 0 .

## BitInput

Assume bit inputs
Specify whether the input is bits or integers. The default is false. When you set this property to true, the step method
input requires a column vector of bit values. The length of this vector must an integer multiple of $\log 2$ (ModulationOrder). This vector contains bit representations of integers between 0 and ModulationOrder-1. When you set this property to false, the step method input must be a column vector of integer symbol values between 0 and ModulationOrder-1.

## SymbolMapping

Constellation encoding
Specify how the object maps an integer or group of $\log 2$ (ModulationOrder) input bits to the corresponding symbol as Binary | Gray | Custom. The default is Gray. When you set this property to Gray, the System object uses a Gray-coded signal constellation. When you set this property to Binary, the object uses a natural binary-coded constellation. When you set this property to Custom, the object uses the signal constellation defined in the CustomSymbolMapping property.

## CustomSymbolMapping

## Custom constellation encoding

Specify a custom constellation symbol mapping vector. The default is $0: 15$. This property is a row or column vector with a size of ModulationOrder. This vector has unique integer values in the range [0, ModulationOrder-1]. These values must be of data type double. The first element of this vector corresponds to the top-leftmost point of the constellation, with subsequent elements running down column-wise, from left to right. The last element corresponds to the bottom-rightmost point. This property applies when you set the Symbolmapping property to Custom.

## NormalizationMethod

Constellation normalization method
Specify the method used to normalize the signal constellation as Minimum distance between symbols | Average power | Peak power. The default is Minimum distance between symbols.

## comm.RectangularQAMModulator

## MinimumDistance

Minimum distance between symbols
Specify the distance between two nearest constellation points as a positive, real, numeric scalar value. The default is 2 . This property applies when you set the NormalizationMethod property to Minimum distance between symbols.

## AveragePower

Average power of constellation
Specify the average power of the symbols in the constellation as a positive, real, numeric scalar value. The default is 1 . This property applies when you set the NormalizationMethod property to Average power.

## PeakPower

Peak power of constellation
Specify the maximum power of the symbols in the constellation as a positive real, numeric scalar value. The default is 1 . This property applies when you set the NormalizationMethod property to Peak power.

## OutputDataType

Data type of output
Specify the output data type as double | single | Custom. The default is double.

## Fixed-Point Properties

## CustomOutputDataType

Fixed-point data type of output
Specify the output fixed-point type as a numerictype object with a signedness of Auto. The default is numerictype([],16). This
property applies when you set the OutputDataType property to Custom.

Methods<br>clone<br>constellation<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>step

Create rectangular QAM modulator object with same property values
Calculate or plot ideal signal constellation
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Modulate using rectangular QAM method

## Examples

Modulate data using rectangular QAM modulation, and visualize the data in a scatter plot.

```
% Create binary data for 32, 3-bit symbols
    data = randi([0 1],96,1);
% Create a rectangular 8-QAM modulator System object with bits as inpI
    hModulator = comm.RectangularQAMModulator(8,'BitInput',true);
% Rotate the constellation by pi/4 radians
    hModulator.PhaseOffset = pi/4;
% Modulate and plot the data
    modData = step(hModulator, data);
    constellation(hModulator)
```


## comm.RectangularQAMModulator

$\begin{array}{ll}\text { Algorithms } & \begin{array}{l}\text { This object implements the algorithm, inputs, and outputs described on } \\ \text { the Rectangular QAM Modulator Baseband block reference page. The } \\ \text { object properties correspond to the block parameters. }\end{array} \\ \text { See Also } & \text { comm.RectangularQAMDemodulator I comm. GeneralQAMModulator }\end{array}$

## comm.RectangularQAMModulator.clone

Purpose
Syntax $\quad C=$ clone $(H)$
Description

Create rectangular QAM modulator object with same property values
$\mathrm{C}=\mathrm{clone}(\mathrm{H})$ creates a RectangularQAMModulator object C , with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.RectangularQAMModulator.constellation

Purpose Calculate or plot ideal signal constellation
Syntax $\quad y=$ constellation $(h)$
constellation(h)
Description $\quad y=$ constellation( $h$ ) returns the numerical values of the constellation.
constellation(h) generates a constellation plot for the object.

## Examples Calculate Ideal Signal Constellation for comm.RectangularQAMModulator

Create a comm.RectangularQAMModulator System object, and then calculate its ideal signal constellation.

Create a comm. RectangularQAMModulator System object by entering the following at the MATLAB command line:
h = comm.RectangularQAMModulator
Calculate and display the ideal signal constellation by calling the constellation method.
a = constellation(h)

## Plot Ideal Signal Constellation for comm.RectangularQAMModulator

Create a comm.RectangularQAMModulator System object, and then plot the ideal signal constellation.

Create a comm.RectangularQAMModulator System object by entering the following at the MATLAB command line:

```
h = comm.RectangularQAMModulator
```

Plot the ideal signal constellation by calling the constellation method.

```
constellation(h)
```


## comm.RectangularQAMModulator.getNuminputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.RectangularQAMModulator.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.RectangularQAMModulator.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked $(H)$ |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the <br> RectangularQAMModulator System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.RectangularQAMModulator.release

Purpose Allow property value and input characteristics changes
Syntax release(H)
Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.RectangularQAMModulator.step

$$
\begin{array}{ll}
\text { Purpose } & \text { Modulate using rectangular QAM method } \\
\text { Syntax } & Y=\operatorname{step}(H, X)
\end{array}
$$

Description $\quad \mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X})$ modulates input data, X , with the rectangular QAM modulator object, H . It returns the baseband modulated output, Y . Depending on the value of the BitInput property, input X can be an integer or bit valued column vector with numeric, logical, or fixed-point data types.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## comm.RectangularQAMTCMDemodulator

## Purpose Demodulate convolutionally encoded data mapped to rectangular QAM constellation

Description

Construction

The RectangularQAMTCMDemodulator object uses the Viterbi algorithm to decode a trellis-coded modulation (TCM) signal that was previously modulated using a rectangular QAM signal constellation.

H = comm.RectangularQAMTCMDemodulator creates a trellis-coded, rectangular, quadrature amplitude (QAM TCM) demodulator System object, H . This object demodulates convolutionally encoded data that has been mapped to a rectangular QAM constellation.

H = comm.RectangularQAMTCMDemodulator (Name, Value) creates a rectangular, QAM TCM, demodulator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).
H = comm.RectangularQAMTCMDemodulator(TRELLIS, Name, Value) creates a rectangular QAM TCM demodulator object, H. This object has the TrellisStructure property set to TRELLIS, and the other specified properties set to the specified values.

## Properties

## TrellisStructure

Trellis structure of convolutional code
Specify trellis as a MATLAB structure that contains the trellis description of the convolutional code. Use the istrellis function to check whether a structure is a valid trellis. The default is the result of poly2trellis([3 1 1], [ $5200 ; 0010 ; 0$ 0 1]).

## TerminationMethod

Termination method of encoded frame
Specify the termination method as Continuous | Truncated | Terminated. The default is Continuous.

When you set this property to Continuous, the object saves the internal state metric at the end of each frame. The next frame uses the same state metric. The object treats each traceback path independently. If the input signal contains only one symbol, you should use Continuous mode.

When you set this property to Truncated, the object treats each input vector independently. The traceback path starts at the state with the best metric and always ends in the all-zeros state.

When you set this property to Terminated, the object treats each input vector independently, and the traceback path always starts and ends in the all-zeros state.

## TracebackDepth

Traceback depth for Viterbi decoder
Specify the scalar, integer number of trellis branches to construct each traceback path. The default is 21 . The Traceback depth parameter influences the decoding accuracy and delay. The decoding delay is the number of zero symbols that precede the first decoded symbol in the output.

When you set the TerminationMethod property to Continuous, the decoding delay consists of TracebackDepth zero symbols or TracebackDepth $\times K$ zero bits for a rate $K / N$ convolutional code.

When you set the TerminationMethod property to Truncated or Terminated, no output delay occurs and the traceback depth must be less than or equal to the number of symbols in each input vector.

## ResetInputPort

Enable demodulator reset input
Set this property to true to enable an additional input to the step method. The default is false. When this additional reset input is a nonzero value, the internal states of the encoder reset to initial conditions. This property applies when you set the TerminationMethod property to Continuous.

## comm.RectangularQAMTCMDemodulator

## ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation used to map the convolutionally encoded data as a positive, integer scalar value. The number of points must be $4,8,16,32$, or 64 . The default is 16 . The ModulationOrder property value must equal the number of possible input symbols to the convolutional decoder of the rectangular QAM TCM demodulator object. The ModulationOrder must equal $2^{N}$ for a rate $K / N$ convolutional code.

## OutputDataType

Data type of output
Specify output data type as logical | double. The default is double.

Methods<br>clone<br>getNumInputs<br>getNumOutputs<br>isLocked<br>release<br>reset<br>step

Create rectangular QAM TCM demodulator object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of the rectangular QAM TCM demodulator object
Demodulate convolutionally encoded data mapped to rectangular QAM constellation

## comm.RectangularQAMTCMDemodulator

```
Examples Modulate and demodulate data using rectangular 16-QAM TCM
modulation.
    hMod = comm.RectangularQAMTCMModulator;
    hAWGN = comm.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)',...
    'SNR',5, 'SignalPower', 10);
    hDemod = comm.RectangularQAMTCMDemodulator('TracebackDepth',16);
% Create an error rate calculator with delay in bits equal to Traceba
    delay = hDemod.TracebackDepth* ...
            log2(hDemod.TrellisStructure.numInputSymbols);
    hErrorCalc = comm.ErrorRate('ReceiveDelay', delay);
    for counter = 1:10
        % Transmit frames of 200 3-bit symbols
        data = randi([0 1],600,1);
        modSignal = step(hMod, data);
        noisySignal = step(hAWGN, modSignal);
        receivedData = step(hDemod, noisySignal);
        errorStats = step(hErrorCalc, data, receivedData);
    end
    fprintf('Error rate = %f\nNumber of errors = %d\n', ...
        errorStats(1), errorStats(2))
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the Rectangular QAM TCM Decoder block reference page. The object properties correspond to the block parameters.

[^9]
## comm.RectangularQAMTCMDemodulator.clone

| Purpose | Create rectangular QAM TCM demodulator object with same property <br> values |
| :--- | :--- |
| Syntax | $C=$ clone $(H)$ |
| Description | $C=$ clone $(H)$ creates a RectangularQAMTCMDemodulator object $C$, <br> with the same property values as $H$. The clone method creates a new <br> unlocked object with uninitialized states. |

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs (H) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( $H$ )

## comm.RectangularQAMTCMDemodulator.getNumOutputs

Purpose $\quad$ Number of outputs from step method
Syntax $\quad N=$ getNumOutputs $(H)$
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.RectangularQAMTCMDemodulator.isLocked

| Purpose | Locked status for input attributes and nontunable properties |
| :--- | :--- |
| Syntax | TF $=$ isLocked (H) |
| Description | TF $=$ isLocked $(H)$ returns the locked status, TF of the <br> RectangularQAMTCMDemodulator System object. |

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.RectangularQAMTCMDemodulator.release

Purpose Allow property value and input characteristics changes

## Syntax release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

| Purpose | Reset states of the rectangular QAM TCM demodulator object |
| :--- | :--- |
| Syntax | reset $(H)$ |
| Description | reset $(H)$ resets the states of the RectangularQAMTCMDemodulator <br> object, $H$. |

## comm.RectangularQAMTCMDemodulator.step

Purpose $\quad \begin{aligned} & \text { Demodulate convolutionally encoded data mapped to rectangular QAM } \\ & \text { constellation }\end{aligned}$
Syntax
$Y=\operatorname{step}(H, X)$
Y $=\operatorname{step}(H, X, R)$

## Description

$\mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X})$ demodulates the rectangular QAM modulated input data, $X$, and uses the Viterbi algorithm to decode the resulting demodulated, convolutionally encoded bits. X must be a complex, double or single precision column vector. The step method outputs a demodulated, binary data column vector, Y. When the convolutional encoder represents a rate $\mathrm{K} / \mathrm{N}$ code, the length of the output vector is $\mathrm{K} * \mathrm{~L}$, where L is the length of the input vector, X .
$Y=\operatorname{step}(H, X, R)$ resets the decoder to the all-zeros state when you input a reset signal, $R$ that is non-zero. $R$ must be a double precision or logical, scalar integer. This syntax applies when you set the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose

Description

## Construction

Convolutionally encode binary data and map using rectangular QAM constellation

The RectangularQAMTCMModulator object implements trellis-coded modulation (TCM) by convolutionally encoding the binary input signal and mapping the result to a rectangular QAM signal constellation.

H = comm. RectangularQAMTCMModulator creates a trellis-coded, rectangular, quadrature amplitude (QAM TCM) System object, H. This object convolutionally encodes a binary input signal and maps the result to a rectangular QAM constellation.

H = comm.RectangularQAMTCMModulator (Name, Value) creates a rectangular QAM TCM modulator object, H , with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.RectangularQAMTCMModulator(TRELLIS,Name, Value) creates a rectangular QAM TCM modulator object, H. This object has the TrellisStructure property set to TRELLIS and the other specified properties set to the specified values.

## Properties

## TrellisStructure

Trellis structure of convolutional code
Specify trellis as a MATLAB structure that contains the trellis description of the convolutional code. Use the istrellis function to check whether a structure is a valid trellis. The default is the result of poly2trellis([3 1 1], [ $5200 ; 0010 ; 000$ 1]).

## TerminationMethod

Termination method of encoded frame
Specify the termination method as Continuous | Truncated | Terminated. The default is Continuous.

When you set this property to Continuous, the object retains the encoder states at the end of each input vector for use with the next input vector.

When you set this property to Truncated, the object treats each input vector independently. The encoder is reset to the all-zeros state at the start of each input vector.

When you set this property to Terminated, the object treats each input vector independently. For each input vector, the object uses extra bits to set the encoder to the all-zeros state at the end of the vector. For a rate $K / N$ code, the step method outputs
the vector with a length given by $y=N \times(L+S) / K$, where $S=$ constraintLength -1 (or, in the case of multiple constraint lengths, $S=\operatorname{sum}($ constraintLength(i)-1)). $L$ is the length of the input to the step method.

## ResetlnputPort

Enable modulator reset input
Set this property to true to enable an additional input to the step method. The default is false. When you set the reset input to the step method to a nonzero value, the object resets the encoder to the all-zeros state. This property applies when you set the TerminationMethod property to Continuous.

## ModulationOrder

Number of points in signal constellation
Specify the number of points in the signal constellation used to map the convolutionally encoded data as a positive integer scalar value equal to $4,8,16,32$, or 64 . The default is 16 . The value of the ModulationOrder property must equal the number of possible output symbols from the convolutional encoder of the QAM TCM modulator. Thus, the value for the ModulationOrder property must equal $2^{N}$ for a rate $K / N$ convolutional code.

## OutputDataType

Data type of output
Specify the output data type as one of double | single. The default is double.


## comm.RectangularQAMTCMModulator

| Algorithms | This object implements the algorithm, inputs, and outputs described on <br> the Rectangular QAM TCM Encoder block reference page. The object <br> properties correspond to the block parameters. |
| :--- | :--- |
| See Also | comm. RectangularQAMTCMDemodulator \| <br> comm.GeneralQAMTCMModulator \| comm. ConvolutionalEncoder |

## comm.RectangularQAMTCMModulator.clone

Purpose

Syntax $\quad C=$ clone $(H)$
Description values

Create rectangular QAM TCM modulator object with same property

C = clone (H) creates a RectangularQAMTCMModulator object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.RectangularQAMTCMModulator.getNumInputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

# comm.RectangularQAMTCMModulator.getNumOutputs 

Purpose Number of outputs from step method<br>Syntax $\quad N=$ getNumOutputs $(H)$<br>Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

## comm.RectangularQAMTCMModulator.isLocked

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the RectangularQAMTCMModulator System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

## comm.RectangularQAMTCMModulator.release

| Purpose | Allow property value and input characteristics changes |
| :--- | :--- |
| Syntax | release (H) |
| Description | release (H) Release system resources (such as memory, file handles <br> or hardware connections) and allows all properties and input <br> characteristics to be changed. |

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

## comm.RectangularQAMTCMModulator.reset

Purpose Reset states of the rectangular QAM TCM modulator object

## Syntax reset (H)

Description reset (H) resets the states of the RectangularQAMTCMModulator object, H.

## Purpose

Syntax
$Y=\operatorname{step}(H, X)$
$Y=\operatorname{step}(H, X, R)$ constellation

Convolutionally encode binary data and map using rectangular QAM
$Y=\operatorname{step}(H, X)$ convolutionally encodes and modulates the input data numeric or logical column vector $X$, and returns the encoded and modulated data, $\mathrm{Y} . \mathrm{X}$ must be of data type numeric, logical, or unsigned fixed point of word length 1 (fi object). When the convolutional encoder represents a rate $K / N$ code, the length of the input vector, x , must be $K \times L$, for some positive integer $L$. The step method outputs a complex column vector, $Y$, of length $L$.
$Y=\operatorname{step}(H, X, R)$ resets the encoder of the rectangular QAM TCM modulator object to the all-zeros state when you input a non-zero reset signal, R. R must be a double precision or logical, scalar integer. This syntax applies when you set the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

| Purpose | Decode data using Reed-Solomon decoder |
| :---: | :---: |
| Description | The RSDecoder object recovers a message vector from a Reed-Solomon codeword vector. For proper decoding, the property values for this object should match those in the corresponding RS Encoder object. |
| Construction | H = comm. RSDecoder creates a block decoder System object, H. This object performs Reed-Solomon (RS) decoding. |
|  | H = comm.RSDecoder(Name, Value) creates an RS decoder object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN). |
|  | H = comm.RSDecoder ( $\mathrm{N}, \mathrm{K}$, Name, Value) creates an RS decoder object, H. This object has the CodewordLength property set to N, the MessageLength property set to N , and the other specified properties set to the specified values. |
| Properties | Bitlnput |
|  | Assume input is bits |
|  | Specify whether the input comprises bits or integers. The default is false. |
|  | When you set this property to false, the step method input data value must be a numeric, column vector of integers. The length of this vector must equal an integer multiple of (CodewordLength - number of punctures). You specify the number of punctures with the PuncturePatternSource and PuncturePattern properties. The CodewordLength property stores the codeword length value. The decoded data output result is a column vector of integers. The length of this vector equals an integer multiple of the message length you specify in the MessageLength property. Each symbol that forms the input codewords and output message is an integer between 0 and $2^{\mathrm{M}}-1$. These integers correspond to an element of the finite Galois field $\operatorname{GF}\left(2^{\mathrm{M}}\right) . M$ is the degree of the primitive polynomial that you specify with |

the PrimitivePolynomialSource and PrimitivePolynomial properties.

When you set this property to true, the input encoded data value must be a numeric column vector of bits. The length equal to an integer multiple of (CodewordLength - number of punctures) $\times M$. You specify the number of punctures with PuncturePatternSource and PuncturePattern properties. The decoded data output result is a column vector of bits. The length equals an integer multiple of MessageLength $\times M$. A group of $M$ bits represents an integer between 0 and $2^{M}-1$ that belongs to the finite Galois field $\operatorname{GF}\left(2^{M}\right) . M$ is the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties.

When you set BitInput to false and ErasuresInputPort to true, the erasures input, ERASURES, must be set to a length equal to the encoded data input vector. Values of 1 in the ERASURES vector correspond to erased symbols in the same position as the input codewords, and values of 0 correspond to nonerased symbols.

When you set this property to true and ErasuresInputPort to true, ERASURES, requires a length of $1 / M$ times the length of the input encoded data vector. $M$ corresponds to the degree of the primitive polynomial. Values of 1 in the ERASURES vector correspond to erased symbols and values of 0 correspond to nonerased symbols. In this case, a symbol corresponds to $M$ bits.

## CodewordLength

Codeword length
Specify the codeword length of the RS code as a double-precision, positive, integer scalar value. The default is 7 .

If you set the PrimitivePolynomialSource property to Auto, CodewordLength must be in the range $3 \ll$ CodewordLength $\leq 2^{16}-1$.

If you set the PrimitivePolynomialSource property to Property, CodewordLength must be in the range $3 \leq$ CodewordLength $\leq 2^{M}-1$. $M$ is the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties. $M$ must be in the range $3 \leq M$ $\leq 16$. The difference (CodewordLength - MessageLength) must be an even integer.

For a full-length RS code the value of the CodewordLength property requires the form $2^{M}-1$. If this property is less than $2^{M}-1$, the object assumes a shortened RS code.

## MessageLength

Message length
Specify the message length as a double-precision, positive, integer scalar value. The default is 3 . The difference (CodewordLength -MessageLength) must be an even integer.

## PrimitivePolynomialSource

Source of primitive polynomial
Specify the source of the primitive polynomial as Auto | Property. The default is Auto.

When you set this property to Auto, the object uses a primitive polynomial of degree $M=$ ceil(log2(CodewordLength+1)), which is the result of fliplr(de2bi(primpoly $(M))$ ).
When you set this property to Property you specify a polynomial using the PrimitivePolynomial property.

## PrimitivePolynomial

Primitive polynomial
Specify the primitive polynomial that defines the finite field GF $\left(2^{M}\right)$ corresponding to the integers that form messages and codewords. The default is the result of fliplr(de2bi(primpoly(3))), which is [ $\left.\begin{array}{llll}1 & 0 & 1 & 1\end{array}\right]$ or the polynomial $x^{3}+x+1$. You must set this
property to a double-precision, binary, row vector that represents a primitive polynomial over GF(2) of degree $M$ in descending order of powers. If CodewordLength is less than $2^{M}-1$, the object uses a shortened RS code. This property applies when you set the PrimitivePolynomialSource property to Property.

## GeneratorPolynomialSource

## Source of generator polynomial

Specify the source of the generator polynomial as Auto | Property. The default is Auto.

When you set this property to Auto, the object automatically chooses the generator polynomial. The object calculates the generator polynomial based on the value of the PrimitivePolynomialSource property.

When you set the PrimitivePolynomialSource property to Auto the object calculates the generator polynomial as rsgenpoly(CodewordLength+SL,MessageLength+SL).

When you set the PrimitivePolynomialSource property to Property, the object calculates generator polynomial as rsgenpoly(CodewordLength $+S L$,MessageLength $+S L$, PrimitivePolynomial). In both cases, $S L=\left(2^{M}-1\right)$ -CodewordLength is the shortened length, and $M$ is the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties. When you set this property to Property, you can specify a generator polynomial using this property.

## GeneratorPolynomial

Generator polynomial
Specify the generator polynomial for the RS code as a double-precision, integer row vector or as a Galois field row vector whose entries are in the range from 0 to $2^{\mathrm{M}}-1$ and represent a generator polynomial in descending order of powers. The length of the generator polynomial must be

CodewordLength-MessageLength+1. This property applies when you set the GeneratorPolynomialSource property to Property.

The default is the result of rsgenpoly (7,3, [], [], 'double'), which corresponds to $\left[\begin{array}{lllll}1 & 3 & 1 & 2 & 3\end{array}\right]$.

When you use this object to generate code, you must set the generator polynomial to a double-precision, integer row vector.

## CheckGeneratorPolynomial

Enable generator polynomial checking
Set this property to true to perform a generator polynomial check. The default is true. This check verifies that $\times$ codewordLength +1 is divisible by the generator polynomial you specify in the GeneratorPolynomial property.

For larger codes, disabling the check accelerates processing time. You should perform the check at least once before setting this property to false. This property applies when you set the GeneratorPolynomialSource property to Property.

## PuncturePatternSource

Source of puncture pattern
Specify the source of the puncture pattern as None | Property. The default is None. If you set this property to None then the object does not apply puncturing to the code. If you set this property to Property then the object punctures the code based on a puncture pattern vector specified in the PuncturePattern property.

## PuncturePattern

Puncture pattern vector
Specify the pattern used to puncture the encoded data as a double-precision, binary column vector of length (CodewordLength-MessageLength). The default is [ones $(2,1)$; zeros $(2,1)]$. Zeros in the puncture pattern vector indicate the position of the parity symbols that are punctured or excluded
from each codeword. This property applies when you set the PuncturePatternSource property to Property.

## ErasuresInputPort

Enable erasures input
Set this property to true to specify a vector of erasures as an input to the step method. The default is false. The erasures input must be a double-precision or logical binary column vector that indicates which symbols of the input codewords to erase.

When you set BitInput to true, the erasures vector length must equal $1 / M$ times the length of the input encoded data vector, where $M$ corresponds to the degree of the primitive polynomial. Values of 1 in the erasures vector correspond to erased symbols in the same position of the bit-packed input codewords. Values of 0 correspond to nonerased symbols.

When you set BitInput to false, the erasures vector length must equal the input encoded data vector. Values of 1 in the erasures vector correspond to erased symbols in the same position of the input codewords. Values of 0 correspond to nonerased symbols.

When this property is set to false the object assumes no erasures.

## NumCorrectedErrorsOutputPort

Enable number of corrected errors output
Set this property to true to obtain the number of corrected errors as an output to the step method. The default is true. A nonnegative value in the $i$-th element of the error output vector, denotes the number of corrected errors in the $i$-th input codeword. A value of -1 in the $i$-th element of the error output vector indicates that a decoding error occurred for that codeword. A decoding error occurs when an input codeword has more errors than the error correction capability of the RS code.

## OutputDataType

Data type of output

Specify the output data type as Same as input | double | logical. The default is Same as input. This property applies when you set the BitInput property to true.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
|  | release |
| step |  |

## Examples

Transmit an RS-encoded, 8-DPSK-modulated symbol stream through an AWGN channel. Then, demodulate, decode, and count errors.

```
hEnc = comm.RSEncoder;
hMod = comm.DPSKModulator('BitInput',false);
hChan = comm.AWGNChannel(...
    'NoiseMethod','Signal to noise ratio (SNR)','SNR',10);
hDemod = comm.DPSKDemodulator('BitOutput',false);
hDec = comm.RSDecoder;
hError = comm.ErrorRate('ComputationDelay',3);
for counter = 1:20
    data = randi([0 7], 30, 1);
    encodedData = step(hEnc, data);
    modSignal = step(hMod, encodedData);
    receivedSignal = step(hChan, modSignal);
```

```
    demodSignal = step(hDemod, receivedSignal);
    receivedSymbols = step(hDec, demodSignal);
    errorStats = step(hError, data, receivedSymbols);
end
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```


## Algorithms

See Also

This object implements the algorithm, inputs, and outputs described on the Binary-Output RS Decoder and Integer-Output RS Decoder block reference pages. The object properties correspond to the block parameters, except:

The BitInput property allows you to select between the Binary-Output RS Decoder and Integer-Output RS Decoder algorithms.

Purpose Create RS decoder object with same property values

## Syntax <br> C = clone(H)

Description $\quad C=$ clone $(H)$ creates a RSDecoder object $C$, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

## comm.RSDecoder.getNumInputs

Purpose Number of expected inputs to step method
Syntax $\quad N=$ getNumInputs $(H)$
Description $\quad N=$ getNumInputs ( $H$ ) returns a positive integer, $N$, representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

## comm.RSDecoder.getNumOutputs

Purpose Number of outputs from step method
Syntax $\quad N=$ getNumOutputs (H)
Description $\quad N=$ getNumOutputs $(H)$ returns the number of outputs, $N$, from the step method. This value will change if any properties that turn inputs on or off are changed.

# Purpose <br> Locked status for input attributes and nontunable properties 

Syntax TF $=$ isLocked (H)
Description
TF = isLocked (H) returns the locked status, TF of the RSDecoder System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

## Syntax <br> release(H)

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose
Decode data using a Reed-Solomon decoder
Syntax
[ $\mathrm{Y}, \mathrm{ERR}$ ] $=\operatorname{step}(\mathrm{H}, \mathrm{X})$
$Y=\operatorname{step}(H, X)$
$Y=\operatorname{step}(H, X, E R A S U R E S)$

## Description

$[\mathrm{Y}, \mathrm{ERR}]=\operatorname{step}(\mathrm{H}, \mathrm{X})$ decodes the encoded input data, X , into the output vector $Y$ and returns the number of corrected errors in output vector ERR. The value of the BitInput property determines whether X is a vector of integers or bits with a numeric, logical, or fixed-point data type. The PuncturePatternSource and PuncturePattern properties affect the expected length of X . The MessageLength property affects the length of $Y$. This syntax applies when you set the NumCorrectedErrorsOutputPort property to true.
$Y=\operatorname{step}(H, X)$ decodes the encoded data, $X$, into the output vector $Y$. This syntax applies when you set the NumCorrectedErrorsOutputPort property to false.

Y = step( $\mathrm{H}, \mathrm{X}$, ERASURES) uses the binary column input vector, ERASURES, to erase the symbols of the input codewords. The elements in ERASURES must be of data type double or logical. Values of 1 in the ERASURES vector correspond to erased symbols, and values of 0 correspond to non-erased symbols. This syntax applies when you set the ErasuresInputPort property to true. See the ErasuresInputPort property help for more information.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

## Purpose Encode data using Reed-Solomon encoder

The RSEncoder object creates a Reed-Solomon code with message and codeword lengths you specify.

Construction H = comm.RSEncoder creates a block encoder System object, H. This

Description

## Properties

 object performs Reed-Solomon (RS) encoding.H = comm.RSEncoder(Name, Value) creates an RS encoder object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.RSEncoder(N, K, Name, Value) creates an RS encoder object, H. This object has the CodewordLength property set to N, the MessageLength property set to K, and the other specified properties set to the specified values.

## BitInput

Assume input is bits
Specify whether the input comprises bits or integers. The default is false.

When you set this property to false, the step method input data value must be a numeric, column vector of integers. The length equals an integer multiple of the message length value stored in the MessageLength property. Each group of MessageLength input elements represents one message word the object will encode.

The step method outputs an encoded data output vector. The output result is a column vector of integers. The length is an integer multiple of (CodewordLength - number of punctures). You specify the number of punctures with the PuncturePatternSource and PuncturePattern properties. Each symbol that forms the input message and output codewords is an integer between 0 and $2^{M}-1$. These integers correspond to an element of the finite Galois field $\operatorname{GF}\left(2^{M}\right) . M$ is the degree of the primitive polynomial
that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties.

When you set this property to true, the input value must be a numeric, column vector of bits with an integer multiple of MessageLength $\times M$ bits. Each group of MessageLength $\times M$ input bits represents one message word the object will encode. The encoded data output result is a column vector of bits. The length of this vector equals an integer multiple of (CodewordLength number of punctures) $\times M$. You specify the number of punctures with the PuncturePatternSource and PuncturePattern properties. A group of $M$ bits represents an integer between 0 and $2^{M}-1$ that belongs to the finite Galois field $G F\left(2^{M}\right) . M$ is the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties.

## CodewordLength

## Codeword length

Specify the codeword length of the RS code as a double-precision, positive, integer scalar value. The default is 7 .
If you set the PrimitivePolynomialSource property to Auto, CodewordLength must be in the range $3<$ CodewordLength $\leq$ $2^{16}-1$.

When you set the PrimitivePolynomialSource property to Property, CodewordLength must be in the range $3 \leq$ CodewordLength $\leq 2^{M}-1 . M$ is the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties. $M$ must be in the range $3 \leq M \leq 16$. The difference (CodewordLength -MessageLength) must be an even integer. For a full-length RS code, the value of this property requires the form $2^{M}-1$.

If the value of this property is less than $2^{M}-1$, the object assumes a shortened RS code.

## MessageLength

Message length
Specify the message length as a double-precision, positive integer scalar value. The default is 3 . The difference (CodewordLength MessageLength) must be an even integer.

## PrimitivePolynomialSource

Source of primitive polynomial
Specify the source of the primitive polynomial as Auto | Property. The default is Auto.

When you set this property to Auto, the object uses a primitive polynomial of degree $M=\operatorname{ceil}(\log 2(C o d e w o r d L e n g t h+1))$, which is the result of fliplr(de2bi(primpoly $(M))$ ).

When you set this property to Property, you can specify a polynomial using the PrimitivePolynomial property.

## PrimitivePolynomial

Primitive polynomial
Specify the primitive polynomial that defines the finite field $\mathrm{GF}\left(2^{M}\right)$ corresponding to the integers that form messages and codewords. You must set this property to a double-precision, binary row vector that represents a primitive polynomial over GF(2) of degree $M$ in descending order of powers.

If CodewordLength is less than $2^{M}-1$, the object uses a shortened RS code. The default is the result of fliplr(de2bi(primpoly(3))),
which is [1011] or the polynomial $x^{M}+x+1$.
This property applies when you set the PrimitivePolynomialSource property to Property.

## GeneratorPolynomialSource

Source of generator polynomial
Specify the source of the generator polynomial as Auto | Property. The default is Auto.

When you set this property to Auto, the object automatically chooses the generator polynomial. The object calculates the generator polynomial based on the value of the PrimitivePolynomialSource property.

When you set the PrimitivePolynomialSource property to Auto the object calculates the generator polynomial as rsgenpoly(CodewordLength $+S L$,MessageLength $+S L$ ).

When you set the PrimitivePolynomialSource property to Property, the object computes generator polynomial as rsgenpoly(CodewordLength $+S L$,MessageLength $+S L$, PrimitivePolynomial). In both cases, $S L=$ $\left(2^{M}-1\right)$-CodewordLength is the shortened length, and $M$ is the degree of the primitive polynomial that you specify with the PrimitivePolynomialSource and PrimitivePolynomial properties.

When you set this property to Property, you can specify a generator polynomial using the GeneratorPolynomial property.

## GeneratorPolynomial

Generator polynomial
Specify the generator polynomial for the RS code as a double-precision, integer row vector or as a Galois row vector whose entries are in the range from 0 to $2^{M}-1$ and represent a generator polynomial in descending order of powers. Each coefficient is an element of Galois field $\operatorname{GF}\left(2^{M}\right)$ represented in integer format. The default is the result of rsgenpoly ( $7,3,[],\left[\right.$,'double'), which evaluates to a $\operatorname{GF}\left(2^{3}\right)$ array with elements $\left[\begin{array}{llll}1 & 3 & 1 & 2\end{array}\right]$. This property applies when you set the GeneratorPolynomialSource property to Property.

## CheckGeneratorPolynomial

Enable generator polynomial checking
Set this property to true to perform a generator polynomial check. The default is true. This check verifies that $x^{\text {codewordLength }}$
+1 is divisible by the generator polynomial specified in the GeneratorPolynomial property. For larger codes, disabling the check speeds up processing. You should perform the check at least once before setting this property to false. This property applies when you set the GeneratorPolynomialSource property to Property.

## PuncturePatternSource

Source of puncture pattern
Specify the source of the puncture pattern as None | Property. The default is None. If you set this property to None then the object does not apply puncturing to the code. If you set this property to Property then the object punctures the code based on a puncture pattern vector specified in the PuncturePattern property.

## PuncturePattern

Puncture pattern vector
Specify the pattern used to puncture the encoded data as a double-precision, binary column vector with a length of (CodewordLength-MessageLength). The default is [ones $(2,1)$; zeros $(2,1)]$. Zeros in the puncture pattern vector indicate the position of the parity symbols that are punctured or excluded from each codeword. This property applies when you set the PuncturePatternSource property to Property.

## OutputDataType

Data type of output
Specify the output data type as Same as input | double | logical. The default is Same as input. This property applies when you set the BitInput property to true.

| Methods | clone |
| :--- | :--- |
|  | getNumInputs |
|  | getNumOutputs |
|  | isLocked |
| release |  |
| step |  |

Create RS encoder object with same property values
Number of expected inputs to step method
Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Encode data using a Reed-Solomon encoder

Examples
Transmit an RS-encoded, 8-DPSK-modulated symbol stream through an AWGN channel. Then, demodulate, decode, and count errors.

```
hEnc = comm.RSEncoder;
hMod = comm.DPSKModulator('BitInput',false);
hChan = comm.AWGNChannel(...
    'NoiseMethod','Signal to noise ratio (SNR)','SNR',10)
hDemod = comm.DPSKDemodulator('BitOutput',false);
hDec = comm.RSDecoder;
hError = comm.ErrorRate('ComputationDelay',3);
for counter = 1:20
    data = randi([0 7], 30, 1);
    encodedData = step(hEnc, data);
    modSignal = step(hMod, encodedData);
    receivedSignal = step(hChan, modSignal);
    demodSignal = step(hDemod, receivedSignal);
    receivedSymbols = step(hDec, demodSignal);
    errorStats = step(hError, data, receivedSymbols);
end
```

```
fprintf('Error rate = %f\nNumber of errors = %d\n', ...
    errorStats(1), errorStats(2))
```Algorithms

This object implements the algorithm, inputs, and outputs described on the Binary-Input RS Encoder and Integer-Input RS Encoder block reference pages. The object properties correspond to the block parameters, except for:

The BitInput property allows you to select between the Binary-Input RS Encoder and Integer-Input RS Encoder algorithms.

\author{
See Also
}
comm.RSDecoder | comm.BCHEncoder

Purpose Create RS encoder object with same property values
Syntax \(\quad C=\) clone \((H)\)
Description \(\quad C=\) clone (H) creates a RSEncoder object \(C\), with the same property values as \(H\). The clone method creates a new unlocked object with uninitialized states.

\section*{comm.RSEncoder.getNumInputs}

Purpose Number of expected inputs to step method

\section*{Syntax \(\quad N=\) getNumInputs \((H)\)}

Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.RSEncoder.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNum0utputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the RSEncoder System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) Release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

Purpose Encode data using a Reed-Solomon encoder

\section*{Syntax \\ Y = step(H,X)}

Description
\(Y=\operatorname{step}(H, X)\) encodes the numeric column input data vector, \(X\), and returns the encoded data, Y . The value of the BitInput property determines whether \(X\) is a vector of integers or bits with a numeric, logical, or fixed-point data type. The MessageLength property affects the expected length of \(X\). The PuncturePatternSource and PuncturePattern properties affect the length of \(Y\).

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Scramble input signal}

Description

\section*{Construction}

\section*{Properties}

The Scrambler object scrambles a scalar or column vector input signal.
H = comm. Scrambler creates a scrambler System object, H. This object scrambles the input data using a linear feedback shift register that you specify with the Polynomial property.

H = comm.Scrambler(Name, Value) creates a scrambler object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.Scrambler(N, POLY,COND, Name, Value) creates a scrambler object, H. This object has the CalculationBase property set to N, the Polynomial property set to POLY, the InitialConditions property set to COND, and the other specified properties set to the specified values.

\section*{CalculationBase}

Range of input data
Specify calculation base as a positive, integer, scalar value. Set the calculation base property to one greater than the number of input values. The step method input and output integers are in the range [ 0 , CalculationBase- 1 ]. The default is 4 .

\section*{Polynomial}

Linear feedback shift register connections
Specify the polynomial that determines the shift register feedback connections. The default is [ \(\left.\begin{array}{lllll}1 & 1 & 1 & 0 & 1\end{array}\right]\). You can the generator polynomial as a numeric, binary vector that lists the coefficients of the polynomial in order of ascending powers of \(z^{-1}\), where \(p\left(z^{-1}\right)=1\) \(+p 1 z^{-1}+p 2 z^{-2}+\ldots\) is the generator polynomial. The first and last elements must be 1. Alternatively, you can specify the generator polynomial as a numeric vector. This vector must contain the exponents of \(z^{-1}\) for the nonzero terms of the polynomial, in order of ascending powers of \(z^{-1}\). In this case, the first vector element
must be 0. For example, both \(\left[\begin{array}{lllllllll}1 & 0 & 0 & 0 & 0 & 0 & 1 & 0 & 1\end{array}\right]\) and \(\left[\begin{array}{ll}0 & -6\end{array}\right.\)
-8] specify the same polynomial \(p\left(z^{-1}\right)=1+z^{-6}+z^{-8}\).

\section*{InitialConditions}

Initial values of linear feedback shift register
Specify the initial values of the linear feedback shift register as an integer row vector with values in [0 CalculationBase-1]. The default is \(\left[\begin{array}{llll}0 & 1 & 2 & 3\end{array}\right]\). The length of this property vector must equal the order of the Polynomial property vector.
\begin{tabular}{ll} 
Methods & clone \\
& getNumInputs \\
& getNumOutputs \\
& isLocked \\
& release \\
reset \\
step
\end{tabular}

Create scrambler object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties

Allow property value and input characteristics changes

Reset states of scrambler object
Scramble input signal

\section*{Examples}

Scramble and descramble random data with values in the range [0 7].
```

% Create scrambler and descrambler objects with calculation base of
N = 8;
hSCR = comm.Scrambler(N, [1 0 1 1 0 1 0 1],...
[0 3 2 2 5 1 7]);
hDSCR = comm.Descrambler(N, [11 0 1 1 0 1 0 1],...
[0 3 2 2 5 1 7]);

```
```

for counter = 1:10
data = randi([0 N-1], 4, 1);
scrData = step(hSCR, data);
deScrData = step(hDSCR, scrData);
[data, scrData, deScrData]
end

```

\title{
Algorithms This object implements the algorithm, inputs, and outputs described on the Scrambler block reference page. The object properties correspond to the block parameters.
}

See Also
comm.Descrambler | comm. PNSequence

\section*{comm.Scrambler.clone}

Purpose Create scrambler object with same property values

\section*{Syntax \\ C = clone( H )}

Description \(\quad C=\) clone \((H)\) creates a Scrambler object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.Scrambler.getNumInputs}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(\quad N=\) getNumInputs \((H)\) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs ( \(H\) )

\section*{comm.Scrambler.getNumOutputs}

Purpose Number of outputs from step method
Syntax \(\quad N=\) getNumOutputs \((H)\)
Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\title{
Purpose \\ Locked status for input attributes and nontunable properties
}

Syntax TF \(=\) isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the Scrambler System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

\section*{Syntax \\ release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{ll} 
Purpose & Reset states of scrambler object \\
Syntax & \(\operatorname{reset}(H)\) \\
Description & reset \((H)\) resets the states of the Scrambler object, H.
\end{tabular}

Purpose Scramble input signal

\section*{Syntax}

Description
\(Y=\operatorname{step}(H, X)\) scrambles input data, \(X\), and returns the result in \(Y . X\) must be a double precision, logical, or integer column vector. The output \(Y\) is same data type and length as the input vector.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Decode input using sphere decoder}

Description

\section*{Construction}

The Sphere Decoder System object decodes the symbols sent over Nt antennas using the sphere decoding algorithm.

H = comm. SphereDecoder creates a System object, H. This object uses the sphere decoding algorithm to find the maximum-likelihood solution for a set of received symbols over a MIMO channel with \(N_{\mathrm{t}}\) transmit antennas and \(N_{\mathrm{r}}\) receive antennas.

H = comm.SphereDecoder (Name, Value) creates a sphere decoder object, H , with the specified property name set to the specified value. Name must appear inside single quotes ("). You can specify several name-value pair arguments in any order as Name1,Value1,...,NameN,ValueN.

H = comm.SphereDecoder (CONSTELLATION,BITTABLE) creates a sphere decoder object, H , with the Constellation property set to CONSTELLATION, and the BitTable property set to BITTABLE.

\section*{Properties}

\section*{Constellation}

Signal constellation per transmit antenna
Specify the constellation as a complex column vector containing the constellation points to which the transmitted bits are mapped. The default setting is a QPSK constellation with an average power of 1 . The length of the vector must be a power of two. The object assumes that each transmit antenna uses the same constellation.

\section*{BitTable}

Bit mapping used for each constellation point.
Specify the bit mapping for the symbols that the Constellation property specifies as a numerical matrix. The default is [0 0; \(01 ; 10 ; 11]\), which matches the default Constellation property value.

The matrix size must be [ConstellationLength bitsPerSymbol]. ConstellationLength represents the length of the Constellation property. bitsPerSymbol represents the number of bits that each symbol encodes.

\section*{InitialRadius}

Initial search radius of the decoding algorithm.
Specify the initial search radius for the decoding algorithm as either Infinity | ZF Solution. The default is Infinity.

When you set this property to Infinity, the object sets the initial search radius to Inf.

When you set this property to ZF Solution, the object sets the initial search radius to the zero-forcing solution. This calculation uses the pseudo-inverse of the input channel when decoding. Large constellations and/or antenna counts can benefit from the initial reduction in the search radius. In most cases, however, the extra computation of the ZF Solution will not provide a benefit.

\section*{DecisionType}

Specify the decoding decision method as either Soft | Hard. The default is Soft.

When you set this property to Soft, the decoder outputs log-likelihood ratios (LLRs), or soft bits.

When you set this property to Hard, the decoder converts the soft LLRs to bits. The hard-decision output logical array follows the mapping of a zero for a negative LLR and one for all other values.

\footnotetext{
Methods clone
isLocked

Create object with same property
values
Locked status for input attributes and nontunable properties
}
release
step

Allow property value and input characteristics changes
Decode received symbols using sphere decoding algorithm

\section*{Examples Decode using a sphere decoder}

Modulate a set of bits using 16-QAM constellation. Transmit the signal as two parallel streams over a MIMO channel. Then, decode using a sphere decoder, with perfect channel knowledge.

Define the modulation order, number of bits to transmit, and the noise variance. Variance is directly related to noise power. The greater the variance value, the smaller the signal-to-noise ratio.
```

M = 16;
nBits = 1e3*log2(M);
noiseVariance = 1e-2;
symMap = [11 10 14 15 9 8 12 13 1 0 4 5 3 2 6 7];

```

Create a Rectangular QAM modulator System object with the Bit Input property set to true, the NormalizationMethod set to Average Power, the SymbolMapping set to Binary, and CustomSymbolMapping set to symMap.
```

hMod = comm.RectangularQAMModulator('BitInput', true, ...
'ModulationOrder', M, 'NormalizationMethod', 'Average power',
'SymbolMapping', 'Custom', 'CustomSymbolMapping', symMap);

```

Convert the decimal value of the symbol map to binary bits using the left bit as the most significant bit (msb).
```

BitTable = de2bi(symMap, log2(M), 'left-msb');

```

Create a MIMO Channel System object with the RandomStream property set to mt19937ar with seed and PathGainsOutputPort set to true and a default configuration of 2-by-2.
```

hMIMO = comm.MIMOChannel('RandomStream', 'mt19937ar with seed',...
'PathGainsOutputPort', true);

```

Create an AWGN Channel System object with the NoiseMethod property set to Variance, VarianceSource set to Property, and Variance set to noiseVariance.
```

hAWGN = comm.AWGNChannel('NoiseMethod', 'Variance',...
'VarianceSource', 'Property', 'Variance', 'noiseVariance');

```

Create a Sphere Decoder System object that processes bits using hard-decision decoding.
```

hSpDec = comm.SphereDecoder('Constellation', constellation(hMod),...
'BitTable', BitTable, 'DecisionType', 'Hard');

```

Create an error rate System object.
hBER = comm.ErrorRate;
Generate a random data stream.
```

data = randi([0 1], nBits, 1);

```

Modulate the data by calling the step method of the Rectangular QAM Modulator System object, hMod.
```

yMod = step(hMod, data);

```

Split the modulated data stream into two, and then transmit over a 2-by-2 MIMO fading channel.
```

yTx = reshape(yMod, [], 2);
[yFad, yPG] = step(hMIMO, yTx);

```

Add noise to the received signal by calling the step method of the AWGN System object, hAWGN.
```

yRec = step(hAWGN, yFad);

```

Decode the received signal.
```

rxBits = step(hSpDec, yRec, squeeze(yPG));

```

Calculate and then display the bit error rate results.
```

ber = step(hBER, data, double(rxBits(:)));
disp(ber(1));

```

\section*{LTE PDSCH Processing Using Sphere Decoding}

See this example which simulates LTE PDSCH spatial multiplexing with sphere decoding.

\section*{Algorithm}

This object implements a soft-output max-log APP MIMO detector by means of a soft-output Schnorr-Euchner sphere decoder (SESD), implemented as single tree search (STS) tree traversal. The algorithm assumes the same constellation and bit table on all of the transmit antennas. Given as inputs, the received symbol vector and the estimated channel matrix, the algorithm outputs the log-likelihood ratios (LLRs) of the transmitted bits.

The algorithm assumes a MIMO system model with \(N_{t}\) transmit antennas and \(N_{r}\) receive antennas where \(N_{t}\) symbols are simultaneously sent, which is express as:
\[
y=H s+n
\]
where the received symbols \(y\) are a function of the transmitted symbol vector \(s\), the MIMO channel matrix, \(H\), and the thermal noise, \(n\).

The goal of the MIMO detector is to find the maximum-likelihood (ML) solution, for which it holds that
\[
\hat{s}_{M L}=\arg \min \|y-H s\|^{2}
\]
where O is the complex-valued constellation from which the \(N_{t}\) elements of \(s\) are chosen.

Soft detection additionally delivers, for each bit, estimates on how reliable the estimate is. For each of the sent bits, denoted as \(x_{j, b}\) (the b-th bit of the \(j\)-th symbol), the reliability of the estimate is calculated by means of the log-likelihood ratio (LLR), which is denoted as \(L\) and is calculated as using the max-log approximation:
\[
L\left(x_{i, j}\right)=\underbrace{\min _{s \in x_{j, b}^{(0)}}\|y-H s\|^{2}}_{\lambda^{M L}}-\underbrace{\min _{s \in x_{j, b}^{(1)}}\|y-H s\|^{2}}_{\lambda_{j, b}^{M L}}
\]
where \(x_{j, b}^{(0)}\) and \(x_{j, b}^{(1)}\) are the disjoint sets of vector symbols that have the \(b\)-th bit in the label of the \(j\)-th scalar symbol equal to 0 and 1 , respectively. The symbol \(\lambda\) denotes the distance calculated as norm squared. The two terms can be expressed as the difference of:

1 The distance to the ML solution \(\hat{s}_{M L}\), denoted as \(\lambda^{M L}\).
2 The distance \(\lambda_{j, b}^{\overline{M L}}\) to the counter-hypothesis, which denotes the binary complement of the b-th bit in the binary label of the j-th entry of \(\hat{s}_{M L}\), i.e., the minimum of the symbol set \(x_{j, b}^{\left(x_{j, b}^{\overline{M L}}\right)}\), which contains all of the possible vectors for which the b-th bit of the j -th entry is flipped compared to the same entry of \(\hat{s}_{M L}\).

Thus, depending on whether \(x_{j, b}^{\left(x_{j, b}^{M L}\right)}\) is zero or one, the LLR for the bit \(x_{j, b}\) is expressed as
\[
L\left(x_{j, b}\right)= \begin{cases}\lambda^{M L}-\lambda_{j, b}^{\overline{M L}} & , x_{j, b}^{M L}=0 \\ \lambda_{j, b}^{\overline{M L}}-\lambda^{M L} & , x_{j, b}^{M L}=1\end{cases}
\]

The design of a decoder thus aims at efficiently finding \(\hat{s}_{M L}, \lambda^{M L}\), and \(\lambda_{j, b}^{\overline{M L}}\).
This search can be converted into a tree search by means of the sphere decoding algorithms. To this end, the channel matrix is decomposed into \(H=Q R\) by means of a QR decomposition. Left-multiplying \(y\) by \(Q^{\mathrm{H}}\), the problem can be reformulated as
\[
\begin{aligned}
& \lambda^{M L}=\underset{s \in o}{\arg \min }\|\bar{y}-R s\|^{2} \\
& \left.\lambda_{j, b}^{\overline{M L}}=\underset{s \in x_{j, b}}{\arg \min } \| \overline{x_{j, b}}\right)
\end{aligned}
\]
from which the triangular structure of \(R\) can be exploited to arrange a tree structure where each of the leaf nodes corresponds to a possible s vector and the partial distances to the nodes in the tree can be calculated cumulatively adding to the partial distance of the parent node.
In the STS algorithm, the \(\lambda^{M L}\) and \(\lambda_{j, b}^{\overline{M L}}\) metrics are searched concurrently. The main idea is to have a list containing the metric \(\lambda^{M L}\), along with the corresponding bit sequence \(x^{M L}\) and the metrics \(x_{j, b}^{\left(x_{j, b}^{M L}\right)}\) of all counter-hypotheses. Then, we search the sub-tree originating from a given node only if the result can lead to an update of either \(\lambda^{M L}\) or \(\lambda_{j, b}^{\overline{M L}}\).
The STS algorithm flow can be summarized as:

1 If when reaching a leaf node, a new ML hypothesis is found \(\left(d(x)<\lambda^{M L}\right)\), all \(\lambda_{j, b}^{\overline{M L}}\) for which \(x_{j, b}=x_{j, b}^{\overline{M L}}\) are set to \(\lambda^{M L}\) which now turns into a valued counter-hypothesis. Then, \(\lambda^{M L}\) is set to the current distance \(\mathrm{d}(\mathrm{x})\).

2 If the current partial distance \(\mathrm{d}(\mathrm{x})\) satisfies \(d(x) \geq \lambda^{M L}\), only the counter-hypotheses have to be checked. For all \(j\) and \(b\) for which \(\left(d(x)<\lambda^{M L}\right)\) and \(x_{j, b}=x_{j, b}^{\overline{M L}}\) the decoder updates \(\lambda_{j, b}^{\overline{M L}}\) to be \(d(x)\).

3 A sub-tree is pruned if the partial distance of the node is bigger than the current \(\lambda_{j, b}^{\overline{M L}}\) which may be affected when traversing the subtree.

4 The algorithm finalizes once all of the tree nodes have been visited once or pruned.

\section*{Selected Bibliography}
[1] Studer, C., M. Wenk, A. Burg, and H. Bölcskei. "Soft-output MIMO detection algorithms: Performance and implementation aspects" Proceedings of the 40th Asilomar Conference on signals, Systems, and Computers, October 2006.
[2] Cho, Y. S., et.al. "MIMO-OFDM Wireless communications with MATLAB," IEEE Press, 2011.
[3] Hochwald, B.M., S. ten Brink. "Achieving near-capacity on a multiple-antenna channel", IEEE Transactions on Communications, Vol. 51, No. 3, Mar 2003, pp. 389-399.
[4] Agrell, E., T. Eriksson, A. Vardy, K. Zeger. "Closest point search in lattices", IEEE Transactions on Information Theory, Vol. 48, No. 8, Aug 2002, pp. 2201-2214.

Purpose Create SphereDecoder object with same property values
Syntax \(\quad C=\) clone \((H)\)
Description \(\quad C=\) clone \((H)\) creates a SphereDecoder object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.
The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked \((H)\)}

Description \(\quad\) TF \(=\) isLocked (H) returns the locked status, TF of the SphereDecoder System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
Purpose Allow property value and input characteristics changes
Syntax ..... release(H)
Description release (H) releases system resources (such as memory, file handlesor hardware connections) and allows all properties and inputcharacteristics to be changed.
Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{ll} 
Purpose & Decode received symbols using sphere decoding algorithm \\
Syntax & \(Y=\operatorname{step}(H\), RXSYMBOLS, CHAN \()\) \\
Description & \begin{tabular}{l} 
Y = step (H, RXSYMBOLS, CHAN) decodes the received symbols, \\
RXSYMBOLS, using the sphere decoding algorithm. The algorithm can \\
be employed to decode Ns channel realizations in one call, where in each \\
channel realization, Nr symbols are received.
\end{tabular}
\end{tabular}

The inputs are:
RXSYMBOLS: a [Ns Nr] complex double matrix containing the received symbols.

CHAN: a [Ns Nt Nr] or [1 Nt Nr] complex double matrix representing the fading channel coefficients of the flat-fading MIMO channel. For the [ Ns Nt Nr ] case, the object applies each channel matrix to each Nr symbol set. For the block fading case, i.e., when the size of CHAN is [1 Nt Nr ], the same channel is applied to all of the received symbols.

The output \(Y\), which depends on the setting of the DecisionType property, is a double matrix containing the Log-Likelihood Ratios (LLRs) of the decoded bits or the bits themselves. For both cases, the size of the output is [Ns*bitsPerSymbol Nt], where bitsPerSymbol represents the number of bits per transmitted symbol, as determined by the BitTable property.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Add receiver thermal noise}

Description

\section*{Construction}

\section*{Properties}

The ThermalNoise object simulates the effects of thermal noise on a complex, baseband signal.

H = comm. ThermalNoise creates a receiver thermal noise System object, H. This object adds thermal noise to the complex, baseband input signal.

H = comm. ThermalNoise(Name, Value) creates a receiver thermal noise object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

\section*{NoiseMethod}

Method to specify noise level
Select the method to specify the noise level as Noise temperature | Noise figure | Noise factor. The default is Noise temperature.

\section*{NoiseTemperature}

\section*{Noise temperature}

Specify the noise temperature in degrees Kelvin as a numeric, nonnegative, real scalar value. The default is 290 . This property applies when you set the NoiseMethod property to Noise temperature.

\section*{NoiseFigure}

Noise figure
Specify the noise figure in decibels relative to a noise temperature of 290 K . You must set this property to a numeric, nonnegative, real scalar value. This property applies when you set the NoiseMethod property to Noise figure. The default is 3.01 dB , which corresponds to a noise temperature of \(290 \times\left(10^{\text {(NoiseFigure/10) }}-1\right)\). This value approximates 290 K .

\section*{NoiseFactor}

Noise factor
Specify the noise factor as a factor relative to a noise temperature of 290 K . You must set this property to a numeric, real scalar value greater than or equal to 1 . This property applies when you set the NoiseMethod property to Noise factor. The default is 2, which corresponds to a noise temperature of \(290 \times\) (NoiseFactor-1) \(=\) 290 K.

\section*{SampleRate}

Sample time
Specify the sample rate of the input samples in Hz as a numeric, real, positive scalar. The default is 1 . The object computes the variance of the noise added to the input signal as (kT*SampleRate). The value \(k\) is Boltzmann's constant and \(T\) is the noise temperature specified explicitly or implicitly via one of the noise methods.

\author{
Methods \\ clone \\ getNumInputs \\ getNumOutputs \\ isLocked \\ release \\ step
}

Create thermal noise object with same property values

Number of expected inputs to step method

Number of outputs from step method

Locked status for input attributes and nontunable properties
Allow property value and input characteristics changes
Add receiver thermal noise

\section*{Examples}

\section*{Algorithms}

\section*{See Also}

Add thermal noise with a noise temperature of 290 K to QPSK data.
```

        hTNoise = comm.ThermalNoise('NoiseTemperature',290);
    % Create a modulator and obtain complex baseband signal
hMod = comm.QPSKModulator;
data = randi([0 3],32,1);
modData = step(hMod,data);
% Add noise to signal
noisyData = step(hTNoise,modData);

```

This object implements the algorithm, inputs, and outputs described on the Receiver Thermal Noise block reference page. The object properties correspond to the block parameters, except:
- This object uses the MATLAB default random stream to generate random numbers. The block uses a random number generator based on the V5 RANDN (Ziggurat) algorithm. The block also uses an initial seed, set with the Initial seed parameter to initialize the random number generator. Ever time the system that contains the block runs, the block generates the same sequence of random numbers. To generate reproducible numbers using this object, you can reset the MATLAB default random stream using the following code.
```

reset(RandStream.getGlobalStream)

```

For more information, see help for RandStream.
- The object provides a SampleRate property, which needs to be specified. The block senses the sample time of the signal and therefore does not have a corresponding parameter.

Purpose Create thermal noise object with same property values

\section*{Syntax \\ C = clone(H)}

Description \(\quad C=\) clone \((H)\) creates a ThermalNoise object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.ThermalNoise.getNumInputs}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)
Purpose Number of outputs from step method

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

Purpose Locked status for input attributes and nontunable properties
Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the ThermalNoise System object.
The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

\section*{Syntax release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{Purpose Add receiver thermal noise}

\section*{Syntax \(\quad Y=\operatorname{step}(H, X)\)}

Description \(\quad Y=\operatorname{step}(H, X)\) adds thermal noise to the complex, baseband input signal, \(X\), and outputs the result in \(Y\). The input signal \(X\) must be a complex, double or single precision data type column vector or scalar.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

Description

\section*{Construction}

\section*{Properties}

\section*{Purpose Decode input signal using parallel concatenated decoding scheme}

The Turbo Decoder System object decodes the input signal using a parallel concatenated decoding scheme that employs the a-posteriori probability (APP) decoder as the constituent decoder. Both constituent decoders use the same trellis structure and algorithm.

H = comm.TurboDecoder creates a System object, H. This object uses the \(a\)-posteriori probability (APP) constituent decoder to iteratively decode the parallel-concatenated convolutionally encoded input data.

H = comm.TurboDecoder(Name, Value) creates a turbo decoder object, H , with the specified property name set to the specified value. Name must appear inside single quotes (' ' ). You can specify several name-value pair arguments in any order as Name1, Value1, , NameN, ValueN.

H = comm.TurboDecoder(TRELLIS, INTERLVRINDICES, NUMITER) creates a turbo decoder object, H, with the TrellisStructure property set to TRELLIS, the InterleaverIndices property set to INTERLVRINDICES, and the NumIterations property set to NUMITER.

\section*{TrellisStructure}

Trellis structure of constituent convolutional code
Specify the trellis as a MATLAB structure that contains the trellis description of the constituent convolutional code. Use the istrellis function to check if a structure is a valid trellis structure. The default is the result of poly2trellis (4, [13 15], 13).

\section*{InterleaverIndicesSource}

Source of interleaver indices
Specify the source of the interleaver indices as one of Property | Input port. When you set this property to Input port, the object uses the interleaver indices specified as an input to the step method. When you set this property to Property, the object uses the interleaver indices that you specify in the

InterleaverIndices property. When you set this property to Input port, the object processes variable-size signals.

Default: Property

\section*{InterleaverIndices}

Interleaver indices
Specify the mapping used to permute the input bits at the encoder as a column vector of integers. This mapping is a vector with the number of elements equal to length, \(L\), of the output of the step method. Each element must be an integer between 1 and \(L\), with no repeated values.

Default: (64:-1:1).'.

\section*{Algorithm}

Decoding algorithm
Specify the decoding algorithm that the object uses for decoding as one of True APP | Max* | Max. When you set this property to True APP, the object implements true \(a\)-posteriori probability decoding. When you set this property to any other value, the object uses approximations to increase the speed of the computations.
Default: True APP

\section*{NumScalingBits}

Number of scaling bits
Specify the number of bits the constituent decoders use to scale the input data to avoid losing precision during the computations. The constituent decoders multiply the input by 2 NumScalingBits and divide the pre-output by the same factor. The NumScalingBits property must be a scalar integer between 0 and 8 . This property applies when you set the Algorithm property to Max*.
Default: 3

\section*{Numlterations}

Number of decoding iterations
Specify the number of decoding iterations used for each call to the step method. The object iterates and provide updates to the log-likelihood ratios (LLR) of the uncoded output bits. The output of the step method is the hard-decision output of the final LLR update.

Default: 6

\author{
Methods \\ clone \\ isLocked \\ release \\ step
}

Create Turbo Decoder object with same property values
Locked status (logical)
Allow property value and input characteristics changes

Decode input signal using parallel concatenated decoding scheme

\section*{Examples}

Transmit turbo-encoded blocks of data over a BPSK-modulated AWGN channel. Then, decode using an iterative turbo decoder and display errors.
```

noiseVar= 4; frmLen = 256;
s = RandStream('mt19937ar', 'Seed', 11);
intrlvrIndices = randperm(s, frmLen);
hTEnc = comm.TurboEncoder('TrellisStructure', poly2trellis(4, ...
[13 15 17], 13), 'InterleaverIndices', intrlvrIndices);
hMod = comm.BPSKModulator;
hChan = comm.AWGNChannel('NoiseMethod', 'Variance', 'Variance', nois
hTDec = comm.TurboDecoder('TrellisStructure', poly2trellis(4, ...
[13 15 17], 13), 'InterleaverIndices', intrlvrIndices, ...
'NumIterations', 4);
hError = comm.ErrorRate;

```
```

for frmIdx = 1:8
data = randi(s, [0 1], frmLen, 1);
encodedData = step(hTEnc, data);
modSignal = step(hMod, encodedData);
receivedSignal = step(hChan, modSignal);
% Convert received signal to log-likelihood ratios for decodir
receivedBits = step(hTDec, (-2/(noiseVar/2))*real(receivedSis
errorStats = step(hError, data, receivedBits);
end
fprintf('Error rate = %f\nNumber of errors = %d\nTotal bits = %d\r
errorStats(1), errorStats(2), errorStats(3))

```

\section*{Algorithms}

This object implements the algorithm, inputs, and outputs described on the Turbo Decoder block reference page. The object properties correspond to the block parameters.

See Also comm. TurboEncoder | comm.APPDecoder

\section*{comm.TurboDecoder.clone}

Purpose Create Turbo Decoder object with same property values

\section*{Syntax}

Description
C = clone(H) creates a Turbo Decoder object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.
\begin{tabular}{ll} 
Purpose & Locked status (logical) \\
Syntax & TF = isLocked (H) \\
Description & \begin{tabular}{l} 
Description
\end{tabular} \\
& \begin{tabular}{l} 
TF \(=\) isLocked (H) returns the locked status, TF of the TurboDecoder \\
System object.
\end{tabular} \\
& \begin{tabular}{l} 
The isLocked method returns a logical value that indicates whether \\
input attributes and nontunable properties for the object are locked. The \\
object performs an internal initialization the first time the step method \\
is executed. This initialization locks nontunable properties and input \\
specifications, such as dimensions, complexity, and data type of the \\
input data. After locking, the isLocked method returns a true value.
\end{tabular} \\
\end{tabular}

Purpose Allow property value and input characteristics changes

\section*{Syntax \\ release(H)}

Description
release (H) release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.

\section*{Purpose}

Decode input signal using parallel concatenated decoding scheme
\(Y=\operatorname{step}(H, X)\)
\(Y=\operatorname{step}(H, X, \quad\) INTERLVRINDICES \()\)
\(Y=\operatorname{step}(H, X)\) decodes the input data, \(X\), using the parallel concatenated convolutional coding scheme that you specify using the TrellisStructure and InterleaverIndices properties. It returns the binary decoded data, \(Y\). Both \(X\) and \(Y\) are column vectors of double precision data type. When the constituent convolutional code represents a rate \(1 / \mathrm{N}\) code, the step method sets the length of the output vector, Y , to (M-2*numTails)/(2*N-1), where M represents the input vector length and numTails is given by log2(TrellisStructure.numStates)*N. The output length, \(L\), is the same as the length of the interleaver indices.
\(Y=\operatorname{step}(H, X\), INTERLVRINDICES \()\) uses the INTERLVRINDICES specified as an input. INTERLVRINDICES is a column vector containing integer values from 1 to \(L\) with no repeated values. The lengths of the INTERLVRINDICES input and the \(Y\) output are the same.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Encode input signal using parallel concatenated encoding scheme}

Description

\section*{Construction}

The Turbo Encoder System object encodes a binary input signal using a parallel concatenated coding scheme. This coding scheme uses two identical convolutional encoders and appends the termination bits at the end of the encoded data bits.

H = comm. TurboEncoder creates a System object, H, that encodes binary data using a turbo encoder.

H = comm.TurboEncoder(Name, Value) creates a turbo encoder object, H , with the specified property name set to the specified value. Name must appear inside single quotes (' ' ). You can specify several name-value pair arguments in any order as Name1, Value1, , NameN, ValueN.

H = comm.TurboEncoder(TRELLIS, INTERLVRINDICES) creates a turbo encoder object, H. In this construction, the TrellisStructure property is set to TRELLIS, and the InterleaverIndices property is set to INTERLVRINDICES.

\section*{Properties}

\section*{TrellisStructure}

Trellis structure of constituent convolutional code
Specify the trellis as a MATLAB structure that contains the trellis description of the constituent convolutional code. Use the istrellis function to check if a structure is a valid trellis structure. The default is the result of poly2trellis(4, [13 15], 13).

\section*{InterleaverIndicesSource}

Source of interleaver indices
Specify the source of the interleaver indices as one of Property | Input port. When you set this property to Input port, the object uses the interleaver indices specified as an input to the step method. When you set this property to Property, the object uses the interleaver indices that you specify in the

InterleaverIndices property. When you set this property to Input port, the object processes variable-size signals.

Default: Property

\section*{InterleaverIndices}

Interleaver indices
Specify the mapping used to permute the input bits at the encoder as a column vector of integers. This mapping is a vector with the number of elements equal to the length of the input for the step method. Each element must be an integer between 1 and \(L\), with no repeated values.

Default: (64:-1:1).'.
\(\left.\left.\begin{array}{lll}\text { Methods } & \text { clone } & \begin{array}{l}\text { Create Turbo Encoder object with } \\
\text { same property values }\end{array} \\
\text { isLocked } & \text { Locked status for input attributes } \\
\text { and nontunable properties }\end{array}\right\} \begin{array}{l}\text { Allow property value and input } \\
\text { characteristics changes }\end{array}\right\}\)\begin{tabular}{l} 
Encode input signal using parallel \\
concatenated coding scheme
\end{tabular}

Examples Transmit turbo-encoded blocks of data over a BPSK-modulated AWGN channel. Then, decode the data using an iterative turbo decoder and display errors.
```

noiseVar= 4; frmLen = 256;
s = RandStream('mt19937ar', 'Seed', 11);
intrlvrIndices = randperm(s, frmLen);
hTEnc = comm.TurboEncoder('TrellisStructure', poly2trellis(4, ..
[13 15 17], 13), 'InterleaverIndices', intrlvrIndices);

```
```

hMod = comm.BPSKModulator;
hChan = comm.AWGNChannel('NoiseMethod', 'Variance', 'Variance', nois
hTDec = comm.TurboDecoder('TrellisStructure', poly2trellis(4, ...
[13 15 17], 13), 'InterleaverIndices', intrlvrIndices, ...
'NumIterations', 4);
hError = comm.ErrorRate;
for frmIdx = 1:8
data = randi(s, [0 1], frmLen, 1);
encodedData = step(hTEnc, data);
modSignal = step(hMod, encodedData);
receivedSignal = step(hChan, modSignal);
% Convert received signal to log-likelihood ratios for decoding
receivedBits = step(hTDec, (-2/(noiseVar/2))*real(receivedSignal
errorStats = step(hError, data, receivedBits);
end
fprintf('Error rate = %f\nNumber of errors = %d\nTotal bits = %d\n',
errorStats(1), errorStats(2), errorStats(3))

```

\title{
Algorithms This object implements the algorithm, inputs, and outputs described on the Turbo Encoder block reference page. The object properties correspond to the block parameters.
}

See Also comm. TurboDecoder | comm.ConvolutionalEncoder

Purpose Create Turbo Encoder object with same property values
Syntax \(\quad C=\) clone \((H)\)
Description \(\quad C=\) clone \((H)\) creates a Turbo Encoder object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.
The clone method creates an instance of an object. The property values, but not internal states, are copied into the new instance of the object.

Purpose Locked status for input attributes and nontunable properties

\section*{Syntax \(\quad\) TF \(=\) isLocked (H)}

Description Description
TF = isLocked(H) returns the locked status, TF of the TurboEncoder System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.
\begin{tabular}{ll} 
Purpose & Allow property value and input characteristics changes \\
Syntax & release \((H)\) \\
Description & \begin{tabular}{l} 
release \((H)\) release system resources (such as memory, file handles \\
or hardware connections) and allows all properties and input \\
characteristics to be changed.
\end{tabular}
\end{tabular}

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{|c|c|}
\hline Purpose & Encode input signal using parallel concatenated coding scheme \\
\hline Syntax & \[
\begin{aligned}
& Y=\operatorname{step}(H, X) \\
& Y=\operatorname{step}(H, X, \text { INTERLVRINDICES })
\end{aligned}
\] \\
\hline Description & \begin{tabular}{l}
\(Y=\operatorname{step}(H, X)\) encodes the input data, \(X\), using the parallel concatenated convolutional coding scheme that you specify using the TrellisStructure and InterleaverIndices properties. It returns the binary decoded data, \(Y\). Both \(X\) and \(Y\) are column vectors of numeric, logical, or unsigned fixed point with word length 1 (fi object). When the constituent convolutional encoder represents a rate \(1 / \mathrm{N}\) code, the step method sets the length of the output vector, Y, to L* (2*N-1)+2*numTails where \(L\) represents the input vector length and numTails is given by \(\log _{2}(\) TrellisStructure.numStates)*N. The tail bits, due to the termination, are appended at the end after the input bits are encoded. \\
Y = step( \(\mathrm{H}, \mathrm{X}\), INTERLVRINDICES) uses the INTERLVRINDICES specified as an input. INTERLVRINDICES is a column vector containing integer values from 1 to \(L\) with no repeated values. The length of the data input X and the INTERLVRINDICES input must be the same.
\end{tabular} \\
\hline
\end{tabular}

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose}

Decode convolutionally encoded data using Viterbi algorithm

The ViterbiDecoder object decodes input symbols to produce binary output symbols. This object can process several symbols at a time for faster performance. This object processes variable-size signals; however, variable-size signals cannot be applied for erasure inputs.

\section*{Construction}

H = comm. ViterbiDecoder creates a Viterbi decoder System object, H. This object uses the Viterbi algorithm to decode convolutionally encoded input data.

H = comm.ViterbiDecoder(Name,Value) creates a Viterbi decoder object, \(H\), with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

H = comm.ViterbiDecoder(TRELLIS, Name, Value) creates a Viterbi decoder object, H. This object has the TrellisStructure property set to TRELLIS and the other specified properties set to the specified values.

\section*{Properties}

\section*{TrellisStructure}

Trellis structure of convolutional code
Specify the trellis as a MATLAB structure that contains the trellis description of the convolutional code. The default is the result of poly2trellis(7, [171 133]). Use the istrellis function to verify whether a structure is a valid trellis.

\section*{InputFormat}

Input format
Specify the format of the input to the decoder as Unquantized | Hard \| Soft. The default is Unquantized.

When you set this property to Unquantized, the input must be a real vector of double- or single-precision soft values that are unquantized. The object considers negative numbers to be 1 s and positive numbers to be 0s.

When you set this property to Hard, the input must be a vector of hard decision values, which are 0 s or 1 s . The data type of the inputs can be double-precision, single-precision, logical, 8-, 16-, and 32 -bit signed integers. You can also use 8 -, 16 -, and 32 -bit unsigned integers.
When you set this property to Soft, the input requires a vector of quantized soft values represented as integers between 0 and
\(2^{\text {Sofinpu WowLecesh }}-1\). The data type of the inputs can be double-precision, single-precision, logical, 8 -, 16 -, and 32 -bit signed integers. You can also use 8 -, 16 -, and 32 -bit unsigned integers. Alternately, you can specify the data type as an unsigned and unscaled fixed point object (fi) with a word length equal to the word length that you specify in the SoftInputWordLength property. The object considers negative numbers to be 0s and positive numbers to be 1 s .

\section*{SoftInputWordLength}

Soft input word length
Specify the number of bits to represent each quantized soft input value as a positive, integer scalar value. The default is 4 bits. This property applies when you set the InputFormat property to Soft.

\section*{InvalidQuantizedInputAction}

Action when input values are out of range
Specify the action the object takes when input values are out of range as Ignore | Error. The default is Ignore. Set this property to Error so that the object generates an error when the quantized input values are out of range. This property applies when you set the InputFormat property to Hard or Soft.

\section*{TracebackDepth}

Traceback depth
Specify the number of trellis branches to construct each traceback path as a numeric, integer scalar value. The default is 34 . The
traceback depth influences the decoding accuracy and delay. The number of zero symbols that precede the first decoded symbol in the output represent a decoding delay.

When you set the TerminationMethod property to Continuous, the decoding delay consists of TracebackDepth zero symbols or TracebackDepth \(\times K\) zero bits for a rate \(K / N\) convolutional code.

When you set the TerminationMethod property to Truncated or Terminated, there is no output delay. In this case, TracebackDepth must be less than or equal to the number of
symbols in each input. If the code rate is \(1 / 2\), a typical traceback depth value is about five times the constraint length of the code.

\section*{TerminationMethod}

Termination method of encoded frame
Specify the termination method as Continuous | Truncated | Terminated. The default is Continuous.

In Continuous mode, the object saves the internal state metric at the end of each frame for use with the next frame. The object treats each traceback path independently.

In Truncated mode, the object treats each frame independently. The traceback path starts at the state with the best metric and always ends in the all-zeros state. In Terminated mode, the object treats each frame independently, and the traceback path always starts and ends in the all-zeros state.

\section*{ResetInputPort}

Enable decoder reset input
Set this property to true to enable an additional step method input. The default is false. When the reset input is a nonzero value, the object resets the internal states of the decoder to initial conditions. This property applies when you set the TerminationMethod property to Continuous.

\section*{DelayedResetAction}

Reset on nonzero input via port
Set this property to true to delay resetting the object output. The default is false. When you set this property to true, the reset of the internal states of the decoder occurs after the object computes the decoded data. When you set this property to false, the reset of the internal states of the decoder occurs before the object computes the decoded data. This property applies when you set the ResetInputPort property to true.

\section*{PuncturePatternSource}

Source of puncture pattern
Specify the source of the puncture pattern as None | Property. The default is None.

When you set this property to None, the object assumes no puncturing. Set this property to Property to decode punctured codewords based on a puncture pattern vector specified via the PuncturePattern property.

\section*{PuncturePattern}

Puncture pattern vector
Specify puncture pattern to puncture the encoded data. The default is \([1 ; 1 ; 0 ; 1 ; 0 ; 1]\). The puncture pattern is a column vector of 1 s and 0 s . The 0 s indicate the position to insert dummy bits. The puncture pattern must match the puncture pattern used by the encoder. This property applies when you set the PuncturePatternSource property to Property.

\section*{ErasuresInputPort}

Enable erasures input
Set this property to true to specify a vector of erasures as a step method input. The default is false. The erasures input must be a double-precision or logical, binary, column vector. This vector indicates which symbols of the input codewords to erase. Values
of 1 indicate erased bits. The decoder does not update the branch metric for the erasures in the incoming data stream.

The lengths of the step method erasure input and the step method data input must be the same. When you set this property to false, the object assumes no erasures.

\section*{OutputDataType}

Data type of output
Specify the data type of the output as Full precision | Smallest unsigned integer | double | single | int8 | uint8 | int16 | uint16 | int32 | uint32 | logical. The default is Full precision.

When the input signal is an integer data type, you must have a Fixed-Point Designer user license to use this property in Smallest unsigned integer or Full precision mode.

\section*{Fixed-Point Properties}

\section*{StateMetricDataType}

Data type of state metric
Specify the state metric data type as Full precision | Custom. The default is Full precision.

When you set this property to Full precision, the object sets the state metric fixed-point type to numerictype ([],16). This property applies when you set the InputFormat property to Hard or Soft.

When you set the InputFormat property to Hard, the step method data input must be a column vector. This vector comprises unsigned, fixed point numbers (fi objects) of word length 1 to enable fixed-point Viterbi decoding. Based on this input (either a 0 or a 1), the object calculates the internal branch metrics using an unsigned integer of word length \(L\). In this case, \(L\) indicates the number of output bits as specified by the trellis structure.

\begin{abstract}
When you set the InputFormat property to Soft, the step method data input must be a column vector. This vector comprises unsigned, fixed point numbers (fi objects) of word length N . \(N\) indicates the number of soft-decision bits specified in the SoftInputWordLength property.
\end{abstract}

The step method data inputs must be integers in the range 0 to \(2^{N}-1\). The object calculates the internal branch metrics using an unsigned integer of word length \(L=(N+\) Nout -1\()\). In this case, Nout represents the number of output bits as specified by the trellis structure.

\section*{CustomStateMetricDataType}

Fixed-point data type of state metric
Specify the state metric fixed-point type as an unscaled, numerictype object with a signedness of Auto. The default is numerictype([],16). This property applies when you set the StateMetricDataType property to Custom.
\begin{tabular}{lll} 
Methods & clone & \begin{tabular}{l} 
Create Viterbi decoder object with \\
same property values
\end{tabular} \\
getNumInputs & getNumOutputs & \begin{tabular}{l} 
Number of expected inputs to \\
step method
\end{tabular} \\
isLocked & \begin{tabular}{l} 
Number of outputs from step \\
method
\end{tabular} \\
release & \begin{tabular}{l} 
Locked status for input attributes \\
and nontunable properties
\end{tabular} \\
reset & \begin{tabular}{l} 
Allow property value and input \\
characteristics changes
\end{tabular} \\
step & \begin{tabular}{l} 
Reset states of the Viterbi decoder \\
object
\end{tabular} \\
Decode convolutionally encoded \\
data using Viterbi algorithm
\end{tabular}
```

Examples Transmit a convolutionally encoded 8-DPSK-modulated bit stream through an AWGN channel. Then, demodulate, decode using a Viterbi decoder, and count errors.

```
```

hConEnc = comm.ConvolutionalEncoder;

```
hConEnc = comm.ConvolutionalEncoder;
hMod = comm.DPSKModulator('BitInput',true);
hMod = comm.DPSKModulator('BitInput',true);
hChan = comm.AWGNChannel('NoiseMethod', ...
hChan = comm.AWGNChannel('NoiseMethod', ...
    'Signal to noise ratio (SNR)', 'SNR',10);
    'Signal to noise ratio (SNR)', 'SNR',10);
hDemod = comm.DPSKDemodulator('BitOutput',true);
hDemod = comm.DPSKDemodulator('BitOutput',true);
hDec = comm.ViterbiDecoder('InputFormat','Hard');
hDec = comm.ViterbiDecoder('InputFormat','Hard');
% Delay in bits is TracebackDepth times the number of bits per symbo:
% Delay in bits is TracebackDepth times the number of bits per symbo:
    delay = hDec.TracebackDepth*...
    delay = hDec.TracebackDepth*...
                                    log2(hDec.TrellisStructure.numInputSymbols);
                                    log2(hDec.TrellisStructure.numInputSymbols);
    hError = comm.ErrorRate('ComputationDelay',3,'ReceiveDelay',delay)
    hError = comm.ErrorRate('ComputationDelay',3,'ReceiveDelay',delay)
    for counter = 1:20
    for counter = 1:20
        data = randi([0 1],30,1);
        data = randi([0 1],30,1);
        encodedData = step(hConEnc, data);
        encodedData = step(hConEnc, data);
        modSignal = step(hMod, encodedData);
        modSignal = step(hMod, encodedData);
        receivedSignal = step(hChan, modSignal);
        receivedSignal = step(hChan, modSignal);
        demodSignal = step(hDemod, receivedSignal);
        demodSignal = step(hDemod, receivedSignal);
        receivedBits = step(hDec, demodSignal);
        receivedBits = step(hDec, demodSignal);
        errorStats = step(hError, data, receivedBits);
        errorStats = step(hError, data, receivedBits);
        end
        end
        fprintf('Error rate = %f\nNumber of errors = %d\n', ...
        fprintf('Error rate = %f\nNumber of errors = %d\n', ...
        errorStats(1), errorStats(2))
```

        errorStats(1), errorStats(2))
    ```

\section*{Algorithms}

See Also

This object implements the algorithm, inputs, and outputs described on the Viterbi Decoder block reference page. The object properties correspond to the block parameters, except:
- The Decision type parameter corresponds to the InputFormat property.
- The Operation mode parameter corresponds to the TerminationMethod property.
comm. ConvolutionalEncoder | comm.APPDecoder

\section*{comm.ViterbiDecoder.clone}

Purpose Create Viterbi decoder object with same property values

\section*{Syntax \\ C = clone(H)}

Description
C = clone(H) creates a ViterbiDecoder object C, with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.ViterbiDecoder.getNumlnputs}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.ViterbiDecoder.getNumOutputs}

Purpose \(\quad\) Number of outputs from step method

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{comm.ViterbiDecoder.isLocked}

\section*{Purpose Locked status for input attributes and nontunable properties}

Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the ViterbiDecoder System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

\section*{comm.ViterbiDecoder.release}

Purpose Allow property value and input characteristics changes

\section*{Syntax \\ release(H)}

Description release(H)Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You can use the release method on a System object in code generated from MATLAB, but once you release its resources, you cannot use that System object again.
\begin{tabular}{ll} 
Purpose & Reset states of the Viterbi decoder object \\
Syntax & reset \((H)\) \\
Description & reset \((H)\) resets the states of the ViterbiDecoder object, H.
\end{tabular}
```

Purpose Decode convolutionally encoded data using Viterbi algorithm
Syntax $\quad Y=\operatorname{step}(H, X)$
$Y=\operatorname{step}(H, X, E R A S U R E S)$
Y = step(H,X,R)

```

\section*{Description}
\(Y=\operatorname{step}(H, X)\) decodes encoded data, \(X\), using the Viterbi algorithm and returns Y . X , must be a column vector with data type and values that depend on how you set the InputFormat property. If the convolutional code uses an alphabet of \(2^{N}\) possible symbols, the length of the input vector, X, must be \(L \times N\) for some positive integer \(L\). Similarly, if the decoded data uses an alphabet of \(2^{K}\) possible output symbols, the length of the output vector, Y , is \(L \times K\).
\(\mathrm{Y}=\operatorname{step}(\mathrm{H}, \mathrm{X}\), ERASURES) uses the binary column input vector, ERASURES, to erase the symbols of the input codewords. The elements in ERASURES must be of data type double or logical. Values of 1 in the ERASURES vector correspond to erased symbols, and values of 0 correspond to non-erased symbols. The lengths of the \(X\) and ERASURES inputs must be the same. This syntax applies when you set the ErasuresInputPort property to true.
\(Y=\operatorname{step}(H, X, R)\) resets the internal states of the decoder when you input a non-zero reset signal, R. R must be a double precision or logical scalar. This syntax applies when you set the TerminationMethod property to Continuous and the ResetInputPort property to true.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose Generate Walsh code from orthogonal set of codes \\ Description \\ The WalshCode object generates a Walsh code from an orthogonal set of codes.}

Construction H = comm. WalshCode creates a Walsh code generator System object, H.

Properties This object generates a Walsh code from a set of orthogonal codes.

H = comm. WalshCode(Name, Value) creates a Walsh code generator object, H, with each specified property set to the specified value. You can specify additional name-value pair arguments in any order as (Name1,Value1,...,NameN,ValueN).

\section*{Length}

Length of generated code
Specify the length of the generated code as a numeric, integer scalar value that is a power of two. The default is 64 .

\section*{Index}

Index of code of interest
Specify the index of the desired code from the available set of codes as a numeric, integer scalar value in the range \([0,1, \ldots\) , \(N-1] . N\) is the value of the Length property. The default is 60 . The number of zero crossings in the generated code equals the value of the specified index.

\section*{SamplesPerFrame}

Number of output samples per frame
Specify the number of Walsh code samples that the step method outputs as a numeric, positive, integer scalar value. The default is 1 . If you set this property to a value of \(M\), then the step method outputs \(M\) samples of a Walsh code of length \(N . N\) is the length of the code that you specify in the Length property.

\section*{OutputDataType}

\section*{Data type of output}

Specify the output data type as double | int8. The default is double.
\begin{tabular}{|c|c|c|}
\hline Methods & clone & Create Walsh code generator object with same property values \\
\hline & getNumInputs & Number of expected inputs to step method \\
\hline & getNumOutputs & Number of outputs from step method \\
\hline & isLocked & Locked status for input attributes and nontunable properties \\
\hline & release & Allow property value and input characteristics changes \\
\hline & reset & Reset states of Walsh code generator object \\
\hline & step & Generate Walsh code from orthogonal set of codes \\
\hline Examples & Generate 10 samp
\[
\begin{aligned}
& \text { hwc }=\text { comm. } \\
& \text { seq }=\text { step }
\end{aligned}
\] & \begin{tabular}{l}
Walsh code sequence. \\
lesPerFrame', 10);
\end{tabular} \\
\hline Algorithms & \begin{tabular}{l}
This object implem the Walsh Code G correspond to the \\
- The object does \\
- The object does time parameter
\end{tabular} & \begin{tabular}{l}
\(m\), inputs, and outputs described on erence page. The object properties except: \\
ty to select frame based outputs. \\
ty that corresponds to the Sample
\end{tabular} \\
\hline
\end{tabular}

See Also comm. HadamardCode | comm.OVSFCode

\section*{comm.WalshCode.clone}

Purpose Create Walsh code generator object with same property values

\section*{Syntax \\ C = clone( H )}

Description \(\quad C=\) clone \((H)\) creates a WalshCode object \(C\), with the same property values as H . The clone method creates a new unlocked object with uninitialized states.

\section*{comm.WalshCode.getNumInputs}

Purpose Number of expected inputs to step method
Syntax \(\quad N=\) getNumInputs \((H)\)
Description \(\quad N=\) getNumInputs ( \(H\) ) returns a positive integer, \(N\), representing the number of expected inputs to the step method. This value will change if any properties that turn inputs on or off are changed. The step method must be called with a number of input arguments equal to the result of getNumInputs (H)

\section*{comm.WalshCode.getNumOutputs}

Purpose Number of outputs from step method

\section*{Syntax \(\quad N=\) getNumOutputs \((H)\)}

Description \(\quad N=\) getNumOutputs \((H)\) returns the number of outputs, \(N\), from the step method. This value will change if any properties that turn inputs on or off are changed.

\section*{Purpose Locked status for input attributes and nontunable properties}

Syntax TF = isLocked (H)
Description TF = isLocked (H) returns the locked status, TF of the WalshCode System object.

The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

Purpose Allow property value and input characteristics changes

\section*{Syntax \\ release(H)}

Description release(H) Release system resources (such as memory, file handles or hardware connections) and allows all properties and input characteristics to be changed.

Note You cannot use the release method on System objects in code generated from MATLAB.
\begin{tabular}{ll} 
Purpose & Reset states of Walsh code generator object \\
Syntax & \(\operatorname{reset}(H)\) \\
Description & reset \((H)\) resets the states of the WalshCode object, H.
\end{tabular}

\section*{Purpose Generate Walsh code from orthogonal set of codes}

\section*{Syntax \(\quad Y=\operatorname{step}(H)\)}

Description
\(Y=\operatorname{step}(H)\) outputs a frame of the Walsh code in column vector \(Y\). Specify the frame length with the SamplesPerFrame property. The Walsh code corresponds to a row of an \(N \mathrm{x} N\) Hadamard matrix, where \(N\) is a nonnegative power of 2 that you specify in the Length property. Use the Index property to choose the row of the Hadamard matrix. The output code is in a bi-polar format with 0 and 1 mapped to 1 and - 1 respectively.

Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

\section*{Purpose \\ Description}

Base class for System objects
matlab.System is the base class for System objects. In your class definition file, you must subclass your object from this base class (or from another class that derives from this base class). Subclassing allows you to use the implementation and service methods provided by this base class to build your object. You use this syntax as the first line of your class definition file to directly inherit from the matlab. System base class, where ObjectName is the name of your object:
classdef ObjectName < matlab.System

Note You must set Access=protected for each matlab.System method you use in your code.
\begin{tabular}{lll} 
Methods & cloneImpl & Copy System object \\
getDiscreteStateImpl & getNumInputsImpl & \begin{tabular}{l} 
Discrete state property values \\
Number of input arguments \\
passed to step and setup methods
\end{tabular} \\
& getNumOutputsImpl & \begin{tabular}{l} 
Number of outputs returned by \\
method
\end{tabular} \\
& isInactivePropertyImpl & \begin{tabular}{l} 
Active or inactive flag for \\
properties
\end{tabular} \\
loadObjectImpl & \begin{tabular}{l} 
Load saved System object from \\
MAT file
\end{tabular} \\
& processInputSizeChangeImpl & \begin{tabular}{l} 
Action when input size changes
\end{tabular} \\
processTunedPropertiesImpl & \begin{tabular}{l} 
Action when tunable properties \\
change
\end{tabular} \\
releaseImpl & Release resources
\end{tabular}

\section*{matlab.System}
\begin{tabular}{ll} 
resetImpl & Reset System object states \\
saveObjectImpl & Save System object in MAT file \\
setProperties & \begin{tabular}{l} 
Set property values from \\
name-value pair inputs
\end{tabular} \\
setupImpl & Initialize System object \\
stepImpl & \begin{tabular}{l} 
System output and state update \\
equations
\end{tabular} \\
validateInputsImpl & Validate inputs to step method \\
validatePropertiesImpl & Validate property values
\end{tabular}

\section*{Attributes \\ In addition to the attributes available for MATLAB objects, you can} apply the following attributes to any property of a custom System object.
\begin{tabular}{ll} 
Nontunable & \begin{tabular}{l} 
After an object is locked (after step or setup \\
has been called), use Nontunable to prevent \\
a user from changing that property value. \\
By default, all properties are tunable. The \\
Nontunable attribute is useful to lock a
\end{tabular} \\
property that has side effects when changed. \\
This attribute is also useful for locking a \\
property value assumed to be constant during \\
processing. You should always specifiy \\
properties that affect the number of input or \\
output ports as Nontunable.
\end{tabular}
\begin{tabular}{ll} 
PositiveInteger & \begin{tabular}{l} 
Use PositiveInteger to limit the property \\
value to a positive integer value.
\end{tabular} \\
DiscreteState & \begin{tabular}{l} 
Use DiscreteState to mark a property so it \\
will display its state value when you use the \\
getDiscreteState method.
\end{tabular}
\end{tabular}

To learn more about attributes, see "Property Attributes" in the MATLAB Object-Oriented Programming documentation.

\section*{Examples}

Create a simple System object, AddOne, which subclasses from matlab. System. You place this code into a MATLAB file, AddOne.m.
```

classdef AddOne < matlab.System
%ADDONE Compute an output value that increments the input by one

```
    methods (Access=protected)
        \% stepImpl method is called by the step method.
        function \(y\) = stepImpl(~,x)
            \(y=x+1 ;\)
        end
    end
end

To use this object, create an instance of AddOne, provide an input, and use the step method:
hAdder = AddOne;
\(\mathrm{x}=1\);
\(y=\operatorname{step}(\) hAdder,\(x)\)

Assign the Nontunable attribute to the InitialValue property, which you define in your class definition file.
```

properties (Nontunable)
InitialValue

```
end
See Also matlab.system.StringSet | matlab.system.StringSet | matlab.system.StringSet | matlab.system.StringSet
| matlab.system.mixin.FiniteSource |
matlab.system.mixin.FiniteSource |
matlab.system.mixin.FiniteSource |
matlab.system.mixin.FiniteSource

\section*{How To}
- "Object-Oriented Programming"
- Class Attributes
- Property Attributes
- "Method Attributes"
- "Define Basic System Objects"
- "Define Basic System Objects"
- "Define Basic System Objects"
- "Define Basic System Objects"
- "Define Property Attributes"
- "Define Property Attributes"
- "Define Property Attributes"
- "Define Property Attributes"

\section*{Purpose}

Copy System object

\section*{Syntax}

Description

Input
Arguments
Examples
Use the cloneImpl method to copy a System object
```

methods (Access=protected)
function obj2 = cloneImpl(obj1)
s = saveObject (obj1);
obj2 = loadObject(s);
end
end

```
See Also \begin{tabular}{l} 
save0bjectImpl | save0bjectImpl | saveObjectImpl | \\
saveObjectImpl | loadObjectImpl | loadObjectImpl | \\
loa0bjectImpl | saveObjectImpl
\end{tabular}

How To . "Clone System Object"
- "Clone System Object"
- "Clone System Object"
- "Clone System Object"
\begin{tabular}{|c|c|}
\hline Purpose & Discrete state property values \\
\hline Syntax & \(s\) = getDiscreteStateImpl(obj) \\
\hline \multirow[t]{2}{*}{Description} & \begin{tabular}{l}
s = getDiscreteStateImpl(obj) returns a struct The field names of the struct are the object's Discre names. To restrict or change the values returned by method, you can override this getDiscreteStateIm \\
getDiscreteStatesImpl is called by the getDiscre which is called by the setup method.
\end{tabular} \\
\hline & Note You must set Access=protected for this met \\
\hline Input Arguments & obi \({ }^{\text {System object handle }}\) \\
\hline Output Arguments & s Struct of state values. \\
\hline Examples & ```
methods (Access=protected)
    function s = getDiscreteState(obj)
    end
end
``` \\
\hline See Also & setupImpl | setupImpl| setupImpl| setupImpl \\
\hline How To & \begin{tabular}{l}
- "Define Property Attributes" \\
- "Define Property Attributes" \\
- "Define Property Attributes" \\
- "Define Property Attributes"
\end{tabular} \\
\hline
\end{tabular}

\section*{matlab.System.getNumInputsImpl}

Purpose Number of input arguments passed to step and setup methods
Syntax num = getNumInputsImpl(obj)

Description

Input obi
Arguments
Output
Arguments
num
num = getNumInputsImpl(obj) returns the number of inputs num (excluding the System object handle) expected by the step method. The default implementation returns 1 , which requires one input from the user, in addition to the System object handle. To specify a value other than 1 , you must use include the getNumInputsImpl method in your class definition file.
getNumInputsImpl is called by the getNumInputs method and by the setup method if the number of inputs has not been determined already.

Note You must set Access=protected for this method.
Do not set any object properties in this getNumInputsImpl method.

System object handle

Number of inputs expected by the step method for the specified object.

Default: 1
Examples Specify the number of inputs (2, in this case) expected by the step method.
```

methods (Access=protected)
function num = getNumInputsImpl(obj)
num = 2;
end

```
end

Specify that the step method will not accept any inputs.
```

methods (Access=protected)
function num = getNumInputsImpl(~)
num = 0;
end
end

```
See Also \begin{tabular}{ll} 
setupImpl | setupImpl | setupImpl | setupImpl | stepImpl \\
| stepImpl | stepImpl | stepImpl | getNumOutputsImpl | \\
getNumOutputsImpl | getNumOutputsImpl | getNumOutputsImpl
\end{tabular}

How To . "Change Number of Step Inputs or Outputs"
- "Change Number of Step Inputs or Outputs"
- "Change Number of Step Inputs or Outputs"
- "Change Number of Step Inputs or Outputs"

\section*{matlab.System.getNumOutputsImpl}

Purpose \(\quad\) Number of outputs returned by step method
Syntax num = getNumOutputsImpl (obj)
Description num = getNumOutputsImpl (obj) returns the number of outputs from the step method. The default implementation returns 1 output. To specify a value other than 1 , you must use include the getNumOutputsImpl method in your class definition file.
getNumOutputsImpl is called by the getNumOutputs method, if the number of outputs has not been determined already.

Note You must set Access=protected for this method.
Do not set any object properties in this getNumOutputsImpl method.

\section*{Input obi}

Arguments
Output
Arguments

\section*{num}

Number of outputs to be returned by the step method for the specified object.

\section*{Examples}

Specify the number of outputs (2, in this case) returned from the step method.
```

methods (Access=protected)
function num = getNumOutputsImpl(obj)
num = 2;
end
end

```

\section*{matlab.System.getNumOutputsImpl}

Specify that the step method does not return any outputs.
methods (Access=protected)
function num \(=\) getNumOutputsImpl(~) num = 0;
end
end
See Also
stepImpl | stepImpl | stepImpl | stepImpl | getNumInputsImpl | getNumInputsImpl | getNumInputsImpl | getNumInputsImpl | setupImpl | setupImpl \| setupImpl | setupImpl

How To . "Change Number of Step Inputs or Outputs"
- "Change Number of Step Inputs or Outputs"
- "Change Number of Step Inputs or Outputs"
- "Change Number of Step Inputs or Outputs"

\section*{matlab.System.isInactivePropertyImpl}
\begin{tabular}{ll} 
Purpose & Active or inactive flag for properties \\
Syntax & flag = isInactivePropertyImpl (obj, prop)
\end{tabular}

Description

Input obi
Arguments

\section*{Output \\ Arguments}

Examples

\section*{prop}

\section*{flag}
flag \(=\) isInactivePropertyImpl(obj, prop) specifies whether a public, non-state property is inactive for the current object configuration. An inactive property is a property that is not relevant to the object, given the values of other properties. Inactive properties are not shown if you use the disp method to display object properties. If you attempt to use public access to directly access or use get or set on an inactive property, a warning occurs.
isInactiveProperty is called by the disp method and by the get and set methods.

Note You must set Access=protected for this method.

System object handle

Public, non-state property name

Logical scalar value indicating whether the input property prop is inactive for the current object configuration.

Display the InitialValue property only when the UseRandomInitialValue property value is false.
```

methods (Access=protected)

```
    function flag = isInactivePropertyImpl(obj, propertyName)
        if strcmp(propertyName,'InitialValue')
            flag = obj.UseRandomInitialValue;
```

        else
            flag = false;
        end
        end
    end

```
See Also setProperties | setProperties | setProperties | setProperties
How To . "Hide Inactive Properties"
- "Hide Inactive Properties"
- "Hide Inactive Properties"
- "Hide Inactive Properties"

\section*{matlab.System.processTunedPropertiesImpl}
Purpose Action when tunable properties change
Syntax processTunedPropertiesImpl(obj)
Description processTunedPropertiesImpl(obj) specifies the actions to performwhen one or more tunable property values change. This method iscalled as part of the next call to the step method after a tunableproperty value changes. A property is tunable only if its Nontunableattribute is false, which is the default.processTunedPropertiesImpl is called by the step method.
Note You must set Access=protected for this method.

Tips

Input
Arguments

\section*{Examples}

Use this method when a tunable property affects a different property value. For example, two property values determine when to calculate a lookup table. You want to perform that calculation when either property changes. You also want the calculation to be done only once if both properties change before the next call to the step method.
obj
System object handle
Use processTunedPropertiesImpl to recalculate the lookup table if the value of either the NumNotes or MiddleC property changes.
```

methods (Access=protected)
function processTunedPropertiesImpl(obj)
% Generate a lookup table of note frequencies
obj.pLookupTable = obj.MiddleC * (1+log(1:obj.NumNotes)/log(12));
end
end

```
See Also \begin{tabular}{l} 
validatePropertiesImpl | validatePropertiesImpl | \\
validatePropertiesImpl | validatePropertiesImpl \\
setProperties | setProperties | setProperties | setProperties
\end{tabular}
Purpose Release resources
Syntax releaseImpl(obj)

Description
releaseImpl(obj) releases any resources used by the System object, such as file handles. This method also performs any necessary cleanup tasks. To release resources for a System object, you must use releaseImpl instead of a destructor.
releaseImpl is called by the release method. releaseImpl is also called when the object is deleted or cleared from memory, or when all references to the object have gone out of scope.

Note You must set Access=protected for this method.

\section*{Input \\ ob}

Arguments
System object handle
Examples Use the releaseImpl method to close a file.
```

methods (Access=protected)
function releaseImpl(obj)
fclose(obj.pFileID);
end
end

```

\section*{See Also \\ resetImpl| resetImpl| resetImpl|resetImpl}
How To

- "Release System Object Resources"
- "Release System Object Resources"
- "Release System Object Resources"
- "Release System Object Resources"
\begin{tabular}{|c|c|}
\hline Purpose & Reset System object states \\
\hline Syntax & resetImpl(obj) \\
\hline \multirow[t]{2}{*}{Description} & \begin{tabular}{l}
resetImpl(obj) defines the state reset equations for the System obj Typically you reset the states to a set of initial values. \\
reset Impl is called by the reset method. It is also called by the set method, after the setupimpl method.
\end{tabular} \\
\hline & Note You must set Access=protected for this method. \\
\hline Input Arguments & \begin{tabular}{l}
obi \\
System object handle
\end{tabular} \\
\hline Examples & Use the reset method to reset the counter pCount property to zero.
```

methods (Access=protected)
function resetImpl(obj)
obj.pCount = 0;
end
end

``` \\
\hline See Also & releaseImpl | releaseImpl| releaseImpl| releaseImpl \\
\hline How To & \begin{tabular}{l}
- "Reset Algorithm State" \\
- "Reset Algorithm State" \\
- "Reset Algorithm State" \\
- "Reset Algorithm State"
\end{tabular} \\
\hline
\end{tabular}

\section*{matlab.System.setProperties}

Purpose
Syntax

\section*{Description}

Input
Arguments

Set property values from name-value pair inputs
setProperties(obj, numargs, name1, value1, name2, value2,...) setProperties(obj, numargs, arg1,..., argm, name1, value1, name2, value2,...)
setProperties(obj, numargs, name1, value1, name2, value2, ...) provides the name-value pair inputs to the System object constructor. Use this syntax if every input must specify both name and value.

Note To allow standard name-value pair handling at construction, define setProperties for your System object.
setProperties(obj, numargs, arg1,..., argm, name1, value1, name2, value2, ...) provides the value-only inputs, followed by the name-value pair inputs to the System object during object construction. Use this syntax if you want to allow users to specify one or more inputs by their values only.

\section*{obi}

System objectSystem object handle

\section*{numargs}

Number of inputs passed in by the object constructor

\section*{name*}

Name of property

\section*{value*}

Value of the property

\section*{arg*}

Value of property (for value-only input to the object constructor)
```

Examples Set up the object so users can specify property values via name-value pairs when constructing the object.
methods
function obj = MyFile(varargin)
setProperties(obj, nargin, varargin\{:\});
end
end

```

How To
- "Set Property Values at Construction Time"
- "Set Property Values at Construction Time"
- "Set Property Values at Construction Time"
- "Set Property Values at Construction Time"

\section*{matlab.System.setupImpl}

Purpose Initialize System object
Syntax setupImpl(obj,input1, input2,...)
Description
setupImpl(obj,input1, input2,...) sets up a System object. To acquire resources for a System object, you must use setupImpl instead of a constructor. setupImpl executes the first time the step method is called on an object after that object has been created. It also executes the next time step is called after an object has been released. . The number of inputs must match the number of inputs defined in the getNumInputsImpl method. You pass the inputs into setupImpl to use the input sizes, datatypes, etc. in the one-time calculations.
setupImpl is called by the setup method, which is done automatically as the first subtask of the step method on an unlocked System object.

Note You must set Access=protected for this method.

Tips

Input
Arguments
obj
System object handle

\section*{input*}

Inputs to the setup method
Open a file for writing using the setupImpl method.
```

methods (Access=protected)
function setupImpl(obj,data)
obj.pFileID = fopen(obj.Filename, 'wb');
if obj.pFileID < 0

```
```

            error('Opening the file failed');
        end
    end
    end

```
\begin{tabular}{|c|c|}
\hline See Also & ```
validatePropertiesImpl | validatePropertiesImpl |
validatePropertiesImpl | validatePropertiesImpl |
validateInputsImpl | validateInputsImpl | validateInputsImpl
| validateInputsImpl | setProperties | setProperties |
setProperties | setProperties
``` \\
\hline
\end{tabular}

How To . "Initialize Properties and Setup One-Time Calculations"
- "Initialize Properties and Setup One-Time Calculations"
- "Initialize Properties and Setup One-Time Calculations"
- "Initialize Properties and Setup One-Time Calculations"
- "Set Property Values at Construction Time"
- "Set Property Values at Construction Time"
- "Set Property Values at Construction Time"
- "Set Property Values at Construction Time"

\section*{matlab.System.stepImpl}

\section*{Purpose System output and state update equations}
```

Syntax
[output1,output2,...] = stepImpl(obj,input1,input2,...)

```

Description
[output1,output2,...] = stepImpl(obj,input1,input2,...) defines the algorithm to execute when you call the step method on the specified object obj. The step method calculates the outputs and updates the object's state values using the inputs, properties, and state update equations.
stepImpl is called by the step method.

Note You must set Access=protected for this method.

Tips

Input
Arguments

\section*{obi}

System object handle

\section*{input*}

Inputs to the step method

\section*{output}

Output returned from the step method.
Use the stepImpl method to increment two numbers.
```

methods (Access=protected)
function [y1,y2] = stepImpl(obj,x1,x2)
y1 = x1 + 1;
y2 = x2 + 1;
end

```
\begin{tabular}{|c|c|}
\hline See Also & \begin{tabular}{l}
getNumInputsImpl | getNumInputsImpl| getNumInputsImpl | \\
getNumInputsImpl| getNumOutputsImpl | getNumOutputsImpl| \\
getNumOutputsImpl| getNumOutputsImpl| validateInputsImpl | \\
validateInputsImpl | validateInputsImpl | validateInputsImpl
\end{tabular} \\
\hline
\end{tabular}

How To . "Define Basic System Objects"
- "Define Basic System Objects"
- "Define Basic System Objects"
- "Define Basic System Objects"
- "Change Number of Step Inputs or Outputs"
- "Change Number of Step Inputs or Outputs"
- "Change Number of Step Inputs or Outputs"
- "Change Number of Step Inputs or Outputs"

\section*{matlab.System.validateInputsImpl}
Purpose Validate inputs to step method
Syntax validateInputsImpl(obj,input1,input2,...)
Description validateInputsImpl(obj,input1,input2,...) validates inputs to the step method at the beginning of initialization Validation includes checking data types, complexity, cross-input validation, and validity of inputs controlled by a property value.
validateInputsImpl is called by the setup method before setupImpl. validateInputsImpl executes only once.

Note You must set Access=protected for this method.
```

Input
Arguments
obj
System object handle
input*
Inputs to the setup method

```
```

Examples Validate that the input is numeric.

```
Examples Validate that the input is numeric.
methods (Access=protected)
methods (Access=protected)
    function validateInputsImpl(~,x)
    function validateInputsImpl(~,x)
        if ~isnumeric(x)
        if ~isnumeric(x)
            error('Input must be numeric');
            error('Input must be numeric');
        end
        end
    end
    end
end
```

end

```

See Also
validatePropertiesImpl | validatePropertiesImpl | validatePropertiesImpl | validatePropertiesImpl | setupImpl| setupImpl | setupImpl | setupImpl

How To
- "Validate Property and Input Values"
- "Validate Property and Input Values"
- "Validate Property and Input Values"
- "Validate Property and Input Values"

\section*{matlab.System.validatePropertiesImpl}
\begin{tabular}{ll} 
Purpose & Validate property values \\
Syntax & validatePropertiesImpl(obj)
\end{tabular}

Description validatePropertiesImpl(obj) validates interdependent or interrelated property values at the beginning of object initialization, such as checking that the dependent or related inputs are the same size.
validatePropertiesImpl is the first method called by the setup method. validatePropertiesImpl also is called before the processTunablePropertiesImpl method.

Note You must set Access=protected for this method.

\section*{Input \\ Arguments \\ obi}

System object handle
Examples
Validate that the useIncrement property is true and that the value of the increment property is greater than zero.
```

methods (Access=protected)
function validatePropertiesImpl(obj)
if obj.useIncrement \&\& obj.increment < 0
error('The increment value must be positive');
end
end
end

```
See Also \(\quad\)\begin{tabular}{l} 
processTunedPropertiesImpl | processTunedPropertiesImpl | \\
processTunedPropertiesImpl | processTunedPropertiesImpl \\
| setupImpl | setupImpl | setupImpl | setupImpl | \\
\\
\\
\\
\\
\\
\\
validateInputsImpl | validateInputsImpl | validateInputsImpl |
\end{tabular}

\section*{How To}
- "Validate Property and Input Values"
- "Validate Property and Input Values"
- "Validate Property and Input Values"
- "Validate Property and Input Values"

\section*{matlab.System.loadObjectlmpl}
Purpose Load saved System object from MAT file
Syntax loadObjectImpl(obj)
Description loadObjectImpl(obj) loads a saved System object, obj, from a MAT file. Your loadObjectImpl method should correspond to your saveObjectImpl method to ensure that all saved properties and data are loaded.
Input ..... obj
System object handle
Examples Load a saved System object. In this case, the object contains a child object, protected and private properties, and a discrete state.
```

methods(Access=protected)
function loadObjectImpl(obj, s, wasLocked)
% Load child System objects
obj.child = matlab.System.loadObject(s.child);

```
            \% Save protected \& private properties
            obj.protected = s.protected;
            obj.pdependentprop = s.pdependentprop;
            \% Save state only if locked when saved
            if wasLocked
                obj.state = s.state;
            end
            \% Call base class method
            loadObjectImpl@matlab.System(obj, s,wasLocked);
        end
    end
See Also saveObjectImpl

\author{
How To \\ - "Load System Object" \\ - "Load System Object" \\ - "Load System Object" \\ - "Load System Object" \\ - "Save System Object" \\ - "Save System Object" \\ - "Save System Object" \\ - "Save System Object"
}

\section*{matlab.System.processInputSizeChangeImpl}
\begin{tabular}{|c|c|}
\hline Purpose & Action when input size changes \\
\hline Syntax & processInputSizeChangeImpl(obj,input1, ...,inputN) \\
\hline \multirow[t]{2}{*}{Description} & \begin{tabular}{l}
processInputSizeChangeImpl(obj,input1, ...,inputN) specifies the actions to perform when any step method input changes size (after the first call to step). \\
processInputSizeChangeImpl is called by the step method before the stepImpl method is called.
\end{tabular} \\
\hline & Note You must set Access=protected for this method. \\
\hline Tips & Use this method when property values or settings depend on the size of the input. For example, you may want to reset some or all of the states in the object when the input sizes change. \\
\hline \multirow[t]{4}{*}{Input Arguments} & obi \\
\hline & System object handle \\
\hline & input \(1, \ldots\), inputN \\
\hline & Inputs to the System object step method. \\
\hline \multirow[t]{3}{*}{Examples} & Use processInputSizeChangeImpl to have the object reset if the input size changes. In this case, ResetOnSizeChange is a property of the object. If ResetOnSizeChange is true, then reset is called what an input size changes. \\
\hline & ```
methods (Access=protected)
    function processInputSizeChangeImpl(obj, ~)
        if obj.ResetOnSizeChange
            reset(obj);
``` \\
\hline & \[
\begin{aligned}
& \text { end } \\
& \text { end }
\end{aligned}
\] \\
\hline
\end{tabular}
end
\begin{tabular}{ll} 
See Also & \begin{tabular}{l} 
resetImpl | resetImpl | resetImpl | resetimpl | stepImpl | \\
stepImpl | stepImpl | stepImpl
\end{tabular} \\
How To & - "Process Input Size Change" \\
& - "Process Input Size Change" \\
& - "Process Input Size Change" \\
& - "Process Input Size Change"
\end{tabular}
\begin{tabular}{ll} 
Purpose & Save System object in MAT file \\
Syntax & saveObjectImpl(obj)
\end{tabular}

Description

\section*{Input \\ Arguments}

Examples
saveObjectImpl(obj) defines what System object obj property and state values are saved in a MAT file when a user calls save on that object. save calls saveObject, which then calls saveObjectImpl. If you do not define a saveObjectImpl method for your System object class, only public properties are saved. To save any private or protected properties or state information, you must define a saveObjectImpl in your class definition file.

You should save the state of an object only if the object is locked. When the user loads that saved object, it loads in that locked state.

To save child object information, you use the associated saveObject method within the saveObjectImpl method.
End users can use load, which calls loadObjectImpl to load a System object into their workspace.

\section*{obi}

System object handle

Define what is saved for the System object. Call the base class version of saveObjectImpl to save public properties. Then, save any child System objects and any protected and private properties. Finally, save the state, if the object is locked.
```

methods(Access=protected)
function s = saveObjectImpl(obj)
s = saveObjectImpl@matlab.System(obj);
s.child = matlab.System.saveObject(obj.child);
s.protected = obj.protected;
s.pdependentprop = obj.pdependentprop;
if isLocked(obj)
s.state = obj.state;

```

\section*{end \\ end \\ end}

See Also
loadObjectImpl
How To . "Save System Object"
- "Save System Object"
- "Save System Object"
- "Save System Object"
- "Load System Object"
- "Load System Object"
- "Load System Object"
- "Load System Object"

\section*{matlab.system.mixin.FiniteSource}

\section*{Purpose Finite source mixin class}

Description

Methods

See Also matlab.System | matlab.System | matlab.System | matlab.System
Tutorials . "Define Finite Source Objects"
- "Define Finite Source Objects"
- "Define Finite Source Objects"
- "Define Finite Source Objects"

How To
- "Object-Oriented Programming"
- Class Attributes
- Property Attributes

\title{
matlab.system.mixin.FiniteSource.isDonelmpl
}
\begin{tabular}{|c|c|}
\hline Purpose & End-of-data flag \\
\hline Syntax & status = isDoneImpl(obj) \\
\hline \multirow[t]{2}{*}{Description} & status = isDoneImpl(obj) indicates if an end-of-data condition has occurred. The isDone method should return false when data from a finite source has been exhausted, typically by having read and output all data from the source. You should also define the result of future reads from an exhausted source in the isDoneImpl method. \\
\hline & isDoneImpl is called by the isDone method. \\
\hline \multirow[t]{2}{*}{Input Arguments} & obi \\
\hline & System object handle \\
\hline \multirow[t]{2}{*}{Output Arguments} & status \\
\hline & Logical value, true or false, that indicates if an end-of-data condition has occurred or not, respectively. \\
\hline \multirow[t]{5}{*}{Examples} & Set up isDoneImpl so the isDone method checks whether the object has completed eight iterations. \\
\hline & methods (Access=private) \\
\hline & ```
function bdone = isDoneImpl(obj)
    bdone = obj.NumIters==8;
``` \\
\hline & end \\
\hline & end \\
\hline \multirow[t]{4}{*}{See Also} & matlab.system.mixin.FiniteSource | \\
\hline & matlab.system.mixin.FiniteSource | \\
\hline & matlab.system.mixin.FiniteSource | \\
\hline & matlab.system.mixin.FiniteSource \\
\hline \multirow[t]{2}{*}{How To} & - "Define Finite Source Objects" \\
\hline & - "Define Finite Source Objects" \\
\hline
\end{tabular}

\section*{matlab.system.mixin.FiniteSource.isDonelmpl}
- "Define Finite Source Objects"
- "Define Finite Source Objects"

\section*{Purpose \\ Description}

Examples

Set of valid string values
matlab.system.StringSet defines a list of valid string values for a property. This class validates the string in the property and enables tab completion for the property value. A StringSet allows only predefined or customized strings as values for the property.

A StringSet uses two linked properties, which you must define in the same class. One is a public property that contains the current string value. This public property is displayed to the user. The other property is a hidden property that contains the list of all possible string values. This hidden property should also have the transient attribute so its value is not saved to disk when you save the System object.
The following considerations apply when using StringSets:
- The string property that holds the current string can have any name.
- The property that holds the StringSet must use the same name as the string property with the suffix "Set" appended to it. The string set property is an instance of the matlab. system. StringSet class.
- Valid strings, defined in the StringSet, must be declared using a cell array. The cell array cannot be empty nor can it have any empty strings. Valid strings must be unique and are case-insensitive.
- The string property must be set to a valid StringSet value.

Set the string property, Flavor, and the StringSet property, FlavorSet, in this example.
```

properties

```
    Flavor='Chocolate';
end
properties (Hidden, Transient)
    FlavorSet = ...
        matlab.system.StringSet(\{'Vanilla', 'Chocolate'\});
end

\author{
See Also matlab.System | matlab.System | matlab.System | matlab.System How To \\ - "Object-Oriented Programming" \\ - Class Attributes \\ - Property Attributes \\ - "Limit Property Values to Finite String Set" \\ - "Limit Property Values to Finite String Set" \\ - "Limit Property Values to Finite String Set" \\ - "Limit Property Values to Finite String Set"
}

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[^0]:    Integer-Output RS Decoder HDL Optimized Block Mask, Expanded View

[^1]:    Pair Block
    See Also

    References
    M-DPSK Modulator Baseband
    DBPSK Demodulator Baseband, DQPSK Demodulator Baseband, M-PSK Demodulator Baseband
    [1] Pawula, R. F., "On M-ary DPSK Transmission Over Terrestrial and Satellite Channels," IEEE Transactions on Communications, Vol. COM-32, July 1984, 752-761.

[^2]:    See Also EVM Measurement
    References [1] Digital Video Broadcasting (DVB): Measurement guidelines for DVB systems, DVB (ETSI) Standard ETR290, May 1997.

[^3]:    Algorithm

[^4]:    comm.BlockDeinterleaver | comm.MatrixInterleaver

[^5]:    Purpose
    Modulate using M-ary DPSK method

    ## Syntax <br> Y = step (H,X)

    Description $\quad Y=\operatorname{step}(H, X)$ modulates input data, $X$, with the DPSK modulator System object, H. It returns the baseband modulated output, Y. Depending on the value of the BitInput property, input $X$ can be an integer or bit valued column vector with numeric or logical data types.

    Note The object performs an initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. If you change a nontunable property or an input specification, the System object issues an error. To change nontunable properties or inputs, you must first call the release method to unlock the object.

[^6]:    Purpose Locked status for input attributes and nontunable properties

    Syntax L = isLocked (H)
    Description

    Input
    Arguments
    Output
    Arguments
    See Also Generator System object, H.

    ## H

    L
    $\mathrm{L}=$ isLocked $(\mathrm{H})$ returns the locked status, L , of the HDL CRC

    The isLocked method returns a logical value that indicates whether input attributes and nontunable properties for the object are locked. The object performs an internal initialization the first time the step method is executed. This initialization locks nontunable properties and input specifications, such as dimensions, complexity, and data type of the input data. After locking, the isLocked method returns a true value.

    HDL CRC Generator System object

    Logical value. Either 1 (true) or 0 (false).
    comm. HDLCRCGenerator | comm.HDLCRCGenerator.clone | comm.HDLCRCGenerator.release | comm.HDLCRCGenerator.reset | comm. HDLCRCGenerator.step |

[^7]:    Algorithms This object implements the algorithm, inputs, and outputs described on the Helical Interleaver block reference page. The object properties correspond to the block parameters.

    See Also comm.HelicalDeinterleaver | comm.MultiplexedInterleaver

[^8]:    comm.PSKTCMModulator | comm.GeneralQAMTCMDemodulator | comm.RectangularQAMTCMDemodulator | comm.ViterbiDecoder

[^9]:    comm.RectangularQAMTCMModulator | comm.GeneralQAMTCMDemodulator | comm.ViterbiDecoder

