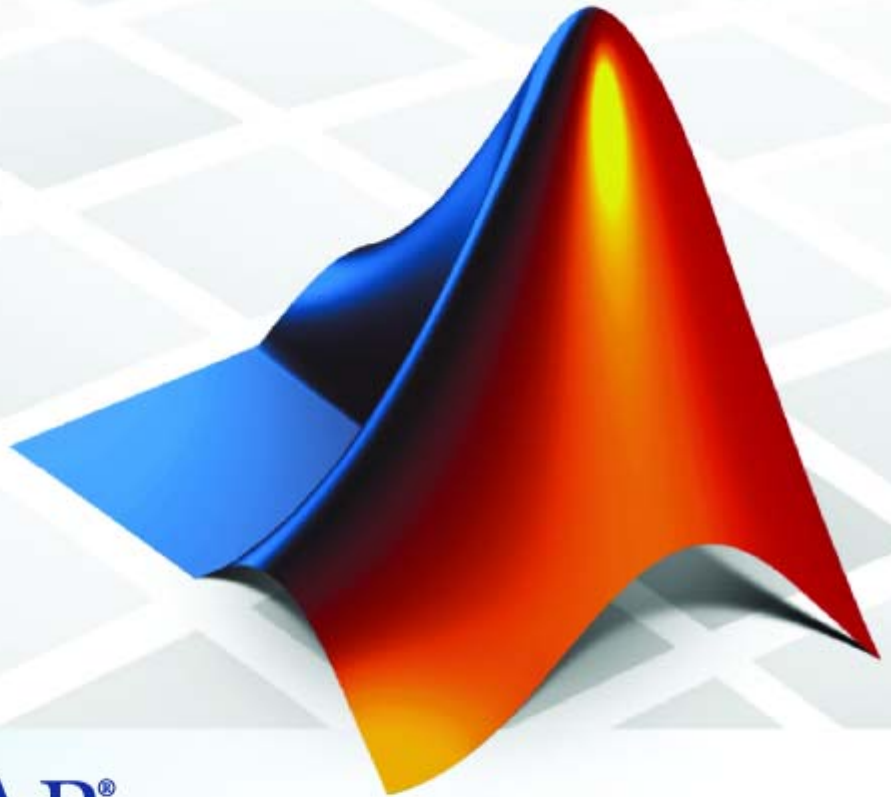


# Communications Blockset 3

## Reference



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March 2006	Online only	Revised for Version 3.3 (Release 2006a)
September 2006	Online only	Revised for Version 3.4 (Release 2006b)
March 2007	Online only	Revised for Version 3.5 (Release 2007a)



## Blocks — By Category

**1**

<b>Communications Sources</b> .....	<b>1-2</b>
Random Data Sources .....	<b>1-2</b>
Noise Generators .....	<b>1-2</b>
Sequence Generators .....	<b>1-2</b>
 <b>Communications Sinks</b> .....	 <b>1-3</b>
 <b>Source Coding</b> .....	 <b>1-3</b>
 <b>Error Detection and Correction</b> .....	 <b>1-4</b>
Block Coding .....	<b>1-4</b>
Convolutional Coding .....	<b>1-5</b>
Cyclic Redundancy Check Coding .....	<b>1-6</b>
 <b>Interleaving</b> .....	 <b>1-6</b>
Block Interleaving .....	<b>1-6</b>
Convolutional Interleaving .....	<b>1-7</b>
 <b>Modulation</b> .....	 <b>1-8</b>
Digital Baseband AM Sublibrary .....	<b>1-8</b>
Digital Baseband PM Sublibrary .....	<b>1-9</b>
Digital Baseband FM Sublibrary .....	<b>1-9</b>
Digital Baseband CPM Sublibrary .....	<b>1-10</b>
Digital Baseband TCM Sublibrary .....	<b>1-10</b>
Analog Passband Modulation .....	<b>1-11</b>
 <b>Communications Filters</b> .....	 <b>1-11</b>
 <b>Channels</b> .....	 <b>1-12</b>
 <b>RF Impairments</b> .....	 <b>1-12</b>

<b>Synchronization</b> .....	<b>1-13</b>
Carrier Phase Recovery .....	<b>1-13</b>
Timing Phase Recovery .....	<b>1-13</b>
Synchronization Components .....	<b>1-14</b>
<b>Equalizers</b> .....	<b>1-14</b>
<b>Sequence Operations</b> .....	<b>1-16</b>
<b>Utility Blocks</b> .....	<b>1-16</b>

**Blocks — Alphabetical List**

**2**

**Functions — Alphabetical List**

**3**

**Index**

# Blocks — By Category

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Communications Sources (p. 1-2)	Sources of random and nonrandom data
Communications Sinks (p. 1-3)	Error statistics and plotting
Source Coding (p. 1-3)	Quantization, companding, and differential coding
Error Detection and Correction (p. 1-4)	Block, convolutional, and CRC coding
Interleaving (p. 1-6)	Block and convolutional interleaving
Modulation (p. 1-8)	Digital baseband and analog passband modulation
Communications Filters (p. 1-11)	Filtering and pulse shaping
Channels (p. 1-12)	Modeling channel impairments
RF Impairments (p. 1-12)	Modeling impairments caused by the radio frequency components
Synchronization (p. 1-13)	Phase recovery methods and phase-locked loops
Equalizers (p. 1-14)	Adaptive and MLSE equalizers
Sequence Operations (p. 1-16)	Scrambling, puncturing, and other operations on sequences
Utility Blocks (p. 1-16)	Miscellaneous relevant blocks

## Communications Sources

These are the sublibraries of Communications Sources:

Random Data Sources (p. 1-2)

Noise Generators (p. 1-2)

Sequence Generators (p. 1-2)

### Random Data Sources

Bernoulli Binary Generator      Generate Bernoulli-distributed random binary numbers

Poisson Integer Generator      Generate Poisson-distributed random integers

Random Integer Generator      Generate integers randomly distributed in range  $[0, M-1]$

### Noise Generators

Gaussian Noise Generator      Generate Gaussian distributed noise with given mean and variance values

Rayleigh Noise Generator      Generate Rayleigh distributed noise

Rician Noise Generator      Generate Rician distributed noise

Uniform Noise Generator      Generate uniformly distributed noise between upper and lower bounds

### Sequence Generators

Barker Code Generator      Generate Barker Code

Gold Sequence Generator      Generate Gold sequence from set of sequences



Hadamard Code Generator	Generate Hadamard code from orthogonal set of codes
Kasami Sequence Generator	Generate Kasami sequence from set of Kasami sequences
OVSF Code Generator	Generate orthogonal variable spreading factor (OVSF) code from set of orthogonal codes
PN Sequence Generator	Generate pseudonoise sequence
Walsh Code Generator	Generate Walsh code from orthogonal set of codes

## Communications Sinks

Discrete-Time Eye Diagram Scope	Display multiple traces of modulated signal
Discrete-Time Scatter Plot Scope	Display in-phase and quadrature components of modulated signal constellation
Discrete-Time Signal Trajectory Scope	Plot modulated signal's in-phase component versus its quadrature component
Error Rate Calculation	Compute bit error rate or symbol error rate of input data

## Source Coding

A-Law Compressor	Implement A-law compressor for source coding
A-Law Expander	Implement A-law expander for source coding

Differential Decoder	Decode binary signal using differential coding
Differential Encoder	Encode binary signal using differential coding
Mu-Law Compressor	Implement $\mu$ -law compressor for source coding
Mu-Law Expander	Implement $\mu$ -law expander for source coding
Quantizing Decoder	Decode quantization index according to codebook
Quantizing Encoder	Quantize signal using partition and codebook

## Error Detection and Correction

These are the sublibraries of Error Detection and Correction:

- Block Coding (p. 1-4)
- Convolutional Coding (p. 1-5)
- Cyclic Redundancy Check Coding (p. 1-6)

### Block Coding

BCH Decoder	Decode BCH code to recover binary vector data
BCH Encoder	Create BCH code from binary vector data
Binary Cyclic Decoder	Decode systematic cyclic code to recover binary vector data

Binary Cyclic Encoder	Create systematic cyclic code from binary vector data
Binary Linear Decoder	Decode linear block code to recover binary vector data
Binary Linear Encoder	Create linear block code from binary vector data
Binary-Input RS Encoder	Create Reed-Solomon code from binary vector data
Binary-Output RS Decoder	Decode Reed-Solomon code to recover binary vector data
Hamming Decoder	Decode Hamming code to recover binary vector data
Hamming Encoder	Create Hamming code from binary vector data
Integer-Input RS Encoder	Create Reed-Solomon code from integer vector data
Integer-Output RS Decoder	Decode Reed-Solomon code to recover integer vector data
LDPC Decoder	Decode binary low-density parity-check code specified by parity-check matrix
LDPC Encoder	Encode binary low-density parity-check code specified by parity-check matrix

## **Convolutional Coding**

APP Decoder	Decode convolutional code using a posteriori probability (APP) method
Convolutional Encoder	Create convolutional code from binary data
Viterbi Decoder	Decode convolutionally encoded data using Viterbi algorithm

## Cyclic Redundancy Check Coding

CRC-N Generator	Generate CRC bits according to CRC method and append to input data frames
CRC-N Syndrome Detector	Detect errors in input data frames according to selected CRC method
General CRC Generator	Generate CRC bits according to generator polynomial and append to input data frames
General CRC Syndrome Detector	Detect errors in input data frames according to generator polynomial

## Interleaving

These are the sublibraries of Interleaving:

- Block Interleaving (p. 1-6)
- Convolutional Interleaving (p. 1-7)

## Block Interleaving

Algebraic Deinterleaver	Restore ordering of input symbols using algebraically derived permutation
Algebraic Interleaver	Reorder input symbols using algebraically derived permutation table
General Block Deinterleaver	Restore ordering of symbols in input vector
General Block Interleaver	Reorder symbols in input vector

Matrix Deinterleaver	Permute input symbols by filling matrix by columns and emptying it by rows
Matrix Helical Scan Deinterleaver	Restore ordering of input symbols by filling matrix along diagonals
Matrix Helical Scan Interleaver	Permute input symbols by selecting matrix elements along diagonals
Matrix Interleaver	Permute input symbols by filling matrix by rows and emptying it by columns
Random Deinterleaver	Restore ordering of input symbols using random permutation
Random Interleaver	Reorder input symbols using random permutation

## **Convolutional Interleaving**

Convolutional Deinterleaver	Restore ordering of symbols that were permuted using shift registers
Convolutional Interleaver	Permute input symbols using set of shift registers
General Multiplexed Deinterleaver	Restore ordering of symbols using specified-delay shift registers
General Multiplexed Interleaver	Permute input symbols using set of shift registers with specified delays
Helical Deinterleaver	Restore ordering of symbols permuted by helical interleaver
Helical Interleaver	Permute input symbols using helical array

## Modulation

These are the sublibraries of Modulation:

Digital Baseband AM Sublibrary  
(p. 1-8)

Digital Baseband PM Sublibrary  
(p. 1-9)

Digital Baseband FM Sublibrary  
(p. 1-9)

Digital Baseband CPM Sublibrary  
(p. 1-10)

Digital Baseband TCM Sublibrary  
(p. 1-10)

Analog Passband Modulation  
(p. 1-11)

### Digital Baseband AM Sublibrary

General QAM Demodulator Baseband	Demodulate QAM-modulated data
General QAM Modulator Baseband	Modulate using quadrature amplitude modulation
M-PAM Demodulator Baseband	Demodulate PAM-modulated data
M-PAM Modulator Baseband	Modulate using M-ary pulse amplitude modulation
Rectangular QAM Demodulator Baseband	Demodulate rectangular-QAM-modulated data
Rectangular QAM Modulator Baseband	Modulate using rectangular quadrature amplitude modulation

## Digital Baseband PM Sublibrary

BPSK Demodulator Baseband	Demodulate BPSK-modulated data
BPSK Modulator Baseband	Modulate using binary phase shift keying method
DBPSK Demodulator Baseband	Demodulate DBPSK-modulated data
DBPSK Modulator Baseband	Modulate using differential binary phase shift keying method
DQPSK Demodulator Baseband	Demodulate DQPSK-modulated data
DQPSK Modulator Baseband	Modulate using differential quaternary phase shift keying method
M-DPSK Demodulator Baseband	Demodulate DPSK-modulated data
M-DPSK Modulator Baseband	Modulate using M-ary differential phase shift keying method
M-PSK Demodulator Baseband	Demodulate PSK-modulated data
M-PSK Modulator Baseband	Modulate using M-ary phase shift keying method
OQPSK Demodulator Baseband	Demodulate OQPSK-modulated data
OQPSK Modulator Baseband	Modulate using offset quadrature phase shift keying method
QPSK Demodulator Baseband	Demodulate QPSK-modulated data
QPSK Modulator Baseband	Modulate using quaternary phase shift keying method

## Digital Baseband FM Sublibrary

M-FSK Demodulator Baseband	Demodulate FSK-modulated data
M-FSK Modulator Baseband	Modulate using M-ary frequency shift keying method

## Digital Baseband CPM Sublibrary

CPFSK Demodulator Baseband	Demodulate CPFSK-modulated data
CPFSK Modulator Baseband	Modulate using continuous phase frequency shift keying method
CPM Demodulator Baseband	Demodulate CPM-modulated data
CPM Modulator Baseband	Modulate using continuous phase modulation
GMSK Demodulator Baseband	Demodulate GMSK-modulated data
GMSK Modulator Baseband	Modulate using Gaussian minimum shift keying method
MSK Demodulator Baseband	Demodulate MSK-modulated data
MSK Modulator Baseband	Modulate using minimum shift keying method

## Digital Baseband TCM Sublibrary

General TCM Decoder	Decode trellis-coded modulation data, mapped using arbitrary constellation
General TCM Encoder	Convolutionally encode binary data and map using arbitrary constellation
M-PSK TCM Decoder	Decode trellis-coded modulation data, modulated using PSK method
M-PSK TCM Encoder	Convolutionally encode binary data and modulate using PSK method
Rectangular QAM TCM Decoder	Decode trellis-coded modulation data, modulated using QAM method
Rectangular QAM TCM Encoder	Convolutionally encode binary data and modulate using QAM method



## Analog Passband Modulation

DSB AM Demodulator Passband	Demodulate DSB-AM-modulated data
DSB AM Modulator Passband	Modulate using double-sideband amplitude modulation
DSBSC AM Demodulator Passband	Demodulate DSBSC-AM-modulated data
DSBSC AM Modulator Passband	Modulate using double-sideband suppressed-carrier amplitude modulation
FM Demodulator Passband	Demodulate FM-modulated data
FM Modulator Passband	Modulate using frequency modulation
PM Demodulator Passband	Demodulate PM-modulated data
PM Modulator Passband	Modulate using phase modulation
SSB AM Demodulator Passband	Demodulate SSB-AM-modulated data
SSB AM Modulator Passband	Modulate using single-sideband amplitude modulation

## Communications Filters

Gaussian Filter	Filter input signal, possibly downsampling, using Gaussian FIR filter
Ideal Rectangular Pulse Filter	Shape input signal using ideal rectangular pulses
Integrate and Dump	Integrate discrete-time signal, resetting to zero periodically

Raised Cosine Receive Filter	Filter input signal, possibly downsampling, using raised cosine FIR filter
Raised Cosine Transmit Filter	Upsample and filter input signal using raised cosine FIR filter
Windowed Integrator	Integrate over time window of fixed length

## Channels

AWGN Channel	Add white Gaussian noise to input signal
Binary Symmetric Channel	Introduce binary errors
Multipath Rayleigh Fading Channel	Simulate multipath Rayleigh fading propagation channel
Multipath Rician Fading Channel	Simulate multipath Rician fading propagation channel

## RF Impairments

Free Space Path Loss	Reduce amplitude of input signal by amount specified
I/Q Imbalance	Create complex baseband model of signal impairments caused by imbalances between in-phase and quadrature receiver components
Memoryless Nonlinearity	Apply memoryless nonlinearity to complex baseband signal
Phase Noise	Apply receiver phase noise to complex baseband signal

Phase/Frequency Offset	Apply phase and frequency offsets to complex baseband signal
Receiver Thermal Noise	Apply receiver thermal noise to complex baseband signal

## Synchronization

These are the sublibraries of Synchronization:

Carrier Phase Recovery (p. 1-13)  
Timing Phase Recovery (p. 1-13)  
Synchronization Components  
(p. 1-14)

### Carrier Phase Recovery

CPM Phase Recovery	Recover carrier phase using 2P-Power method
M-PSK Phase Recovery	Recover carrier phase using M-Power method

### Timing Phase Recovery

Early-Late Gate Timing Recovery	Recover symbol timing phase using early-late gate method
Gardner Timing Recovery	Recover symbol timing phase using Gardner's method
MSK-Type Signal Timing Recovery	Recover symbol timing phase using fourth-order nonlinearity method

Mueller-Muller Timing Recovery	Recover symbol timing phase using Mueller-Muller method
Squaring Timing Recovery	Recover symbol timing phase using squaring method

## Synchronization Components

Baseband PLL	Implement baseband phase-locked loop
Charge Pump PLL	Implement charge pump phase-locked loop using digital phase detector
Continuous-Time VCO	Implement voltage-controlled oscillator
Discrete-Time VCO	Implement voltage-controlled oscillator in discrete time
Linearized Baseband PLL	Implement linearized version of baseband phase-locked loop
Phase-Locked Loop	Implement phase-locked loop to recover phase of input signal

## Equalizers

CMA Equalizer	Equalize using constant modulus algorithm
LMS Decision Feedback Equalizer	Equalize using decision feedback equalizer that updates weights with LMS algorithm
LMS Linear Equalizer	Equalize using linear equalizer that updates weights with LMS algorithm

---

MLSE Equalizer	Equalize using Viterbi algorithm
Normalized LMS Decision Feedback Equalizer	Equalize using decision feedback equalizer that updates weights with normalized LMS algorithm
Normalized LMS Linear Equalizer	Equalize using linear equalizer that updates weights with normalized LMS algorithm
RLS Decision Feedback Equalizer	Equalize using decision feedback equalizer that updates weights with RLS algorithm
RLS Linear Equalizer	Equalize using linear equalizer that updates weights using RLS algorithm
Sign LMS Decision Feedback Equalizer	Equalize using decision feedback equalizer that updates weights with signed LMS algorithm
Sign LMS Linear Equalizer	Equalize using linear equalizer that updates weights with signed LMS algorithm
Variable Step LMS Decision Feedback Equalizer	Equalize using decision feedback equalizer that updates weights with variable-step-size LMS algorithm
Variable Step LMS Linear Equalizer	Equalize using linear equalizer that updates weights with variable-step-size LMS algorithm

## Sequence Operations

Deinterlacer	Distribute elements of input vector alternately between two output vectors
Derepeat	Reduce sampling rate by averaging consecutive samples
Descrambler	Descramble input signal
Insert Zero	Distribute input elements in output vector
Interlacer	Alternately select elements from two input vectors to generate output vector
Puncture	Output elements which correspond to 1s in binary Puncture vector
Scrambler	Scramble input signal

## Utility Blocks

Align Signals	Align two signals by finding delay between them
Bipolar to Unipolar Converter	Map bipolar signal into unipolar signal in range $[0, M-1]$
Bit to Integer Converter	Map vector of bits to corresponding vector of integers
Complex Phase Difference	Output phase difference between two complex input signals
Complex Phase Shift	Shift phase of complex input signal by second input value
Data Mapper	Map integer symbols from one coding scheme to another

Find Delay

Find delay between two signals

Integer to Bit Converter

Map vector of integers to vector of bits

Unipolar to Bipolar Converter

Map unipolar signal in range  $[0, M-1]$  into bipolar signal





# Blocks — Alphabetical List

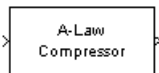
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# A-Law Compressor

**Purpose** Implement A-law compressor for source coding

**Library** 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



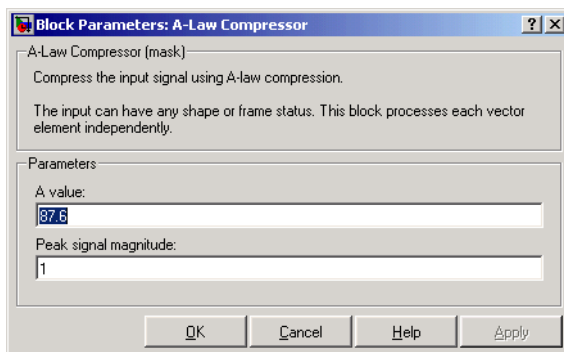
$$y = \begin{cases} \frac{A|x|}{1 + \log A} \operatorname{sgn}(x) & \text{for } 0 \leq |x| \leq \frac{V}{A} \\ \frac{V(1 + \log(A|x|/V))}{1 + \log A} \operatorname{sgn}(x) & \text{for } \frac{V}{A} < |x| \leq V \end{cases}$$

where  $A$  is the A-law parameter of the compressor,  $V$  is the peak signal magnitude for  $x$ ,  $\log$  is the natural logarithm, and  $\operatorname{sgn}$  is the signum function (sign in MATLAB®).

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.

## 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.

**Pair Block**

A-Law Expander

**See Also**

Mu-Law Compressor

**References**

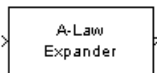
[1] Sklar, Bernard. *Digital Communications: Fundamentals and Applications*. Englewood Cliffs, N.J., Prentice-Hall, 1988.

# A-Law Expander

**Purpose** Implement A-law expander for source coding

**Library** Source Coding

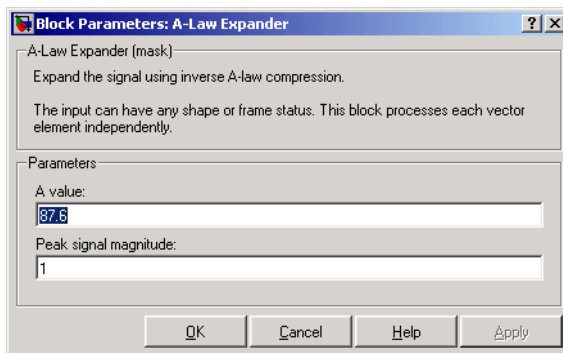
**Description** The A-Law Expander block recovers data that the A-Law Compressor block compressed. The formula for the A-law expander, shown below, is the inverse of the compressor function.



$$x = \begin{cases} \frac{y(1 + \log A)}{A} & \text{for } 0 \leq |y| \leq \frac{V}{1 + \log A} \\ \exp(|y|(1 + \log A) / V - 1) \frac{V}{A} \operatorname{sgn}(y) & \text{for } \frac{V}{1 + \log A} < |y| \leq V \end{cases}$$

The input can have any shape or frame status. This block processes each vector element independently.

## 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.

Match these parameters to the ones in the corresponding A-Law Compressor block.

**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.

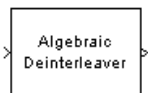
# Algebraic Deinterleaver

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**Purpose** Restore ordering of input symbols using algebraically derived permutation

**Library** Block sublibrary of Interleaving

## Description



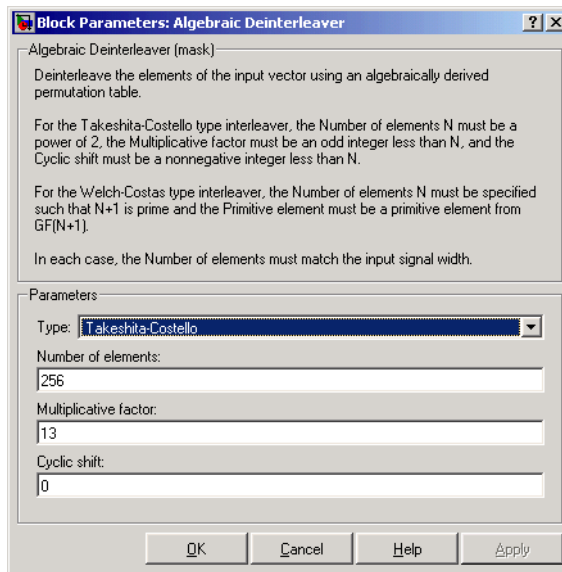
The Algebraic Deinterleaver block restores the original ordering of a sequence that was interleaved using the Algebraic Interleaver block. In typical usage, the parameters in the two blocks have the same values.

The **Number of elements** parameter,  $N$ , indicates how many numbers are in the input vector. If the input is frame-based, then it must be a column vector.

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.

The **Type** parameter indicates the algebraic method that the block uses to generate the appropriate permutation table. Choices are `Takehita-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.

## Dialog Box



### Type

The type of permutation table that the block uses for deinterleaving. Choices are Takeshita-Costello and Welch-Costas.

### Number of elements

The number of elements,  $N$ , in the input vector.

### Multiplicative factor

The factor used to compute the corresponding interleaver's cycle vector. This field appears only if **Type** is set to Takeshita-Costello.

### Cyclic shift

The amount by which the block shifts indices when creating the corresponding interleaver's permutation table. This field appears only if **Type** is set to Takeshita-Costello.

# Algebraic Deinterleaver

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## **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.

## **Pair Block**

Algebraic Interleaver

## **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.



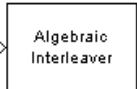
## Purpose

Reorder input symbols using algebraically derived permutation table

## Library

Block sublibrary of Interleaving

## Description



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. If the input is frame-based, then it must be a column vector.

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.

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 **Type** is set 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  $GF(N+1)$  can be expressed as  $A$  raised to some integer power.

In a Welch-Costas interleaver, the permutation maps the integer  $k$  to  $\text{mod}(A^k, N+1) - 1$ .

- If **Type** is set to Takeshita-Costello, then  $N$  must be  $2^m$  for some integer  $m$ . The **Multiplicative factor** parameter,  $h$ , must be an odd integer less than  $N$ . The **Cyclic shift** parameter,  $k$ , must be a nonnegative integer less than  $N$ .

A Takeshita-Costello interleaver uses a length- $N$  *cycle vector* whose  $n$ th element is

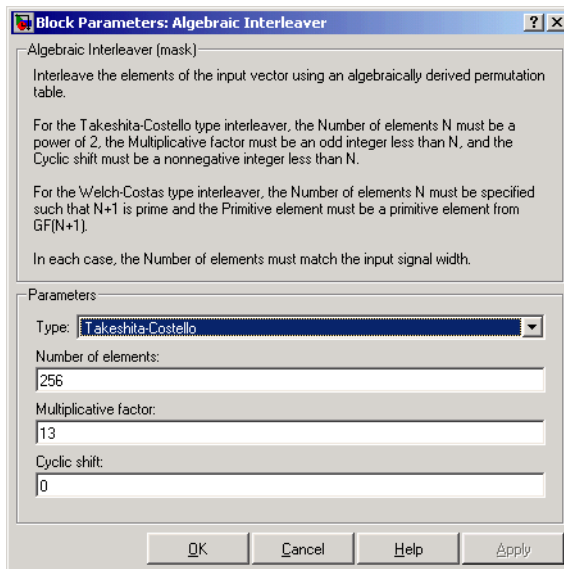
$$\text{mod}(k^{*(n-1)*n/2}, N)$$

for integers  $n$  between 1 and  $N$ . The block creates a permutation vector by listing, for each element of the cycle vector in ascending

# Algebraic Interleaver

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 the **Cyclic shift** parameter. (The block performs all computations on numbers and indices modulo  $N$ .)

## Dialog Box



### Type

The type of permutation table that the block uses for interleaving.

### Number of elements

The number of elements,  $N$ , in the input vector.

### Multiplicative factor

The factor used to compute the interleaver's cycle vector. This field appears only if **Type** is set to Takeshita-Costello.

### 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.

**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.

**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.

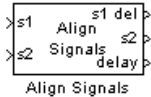
# Align Signals

---

**Purpose** Align two signals by finding delay between them

**Library** Utility Blocks

## Description



The Align Signals block aligns a signal with a delayed, and possibly distorted, version of itself. The block is particularly 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.

The input port labeled s1 receives the original signal, while the input port labeled s2 receives the delayed version of the signal. The two input signals must have the same sample times. The block calculates the delay between the two signal, 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 “Computing Delays” in the Communications Blockset 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 the **Disable recurring updates** check box, 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.

## 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.

## Examples

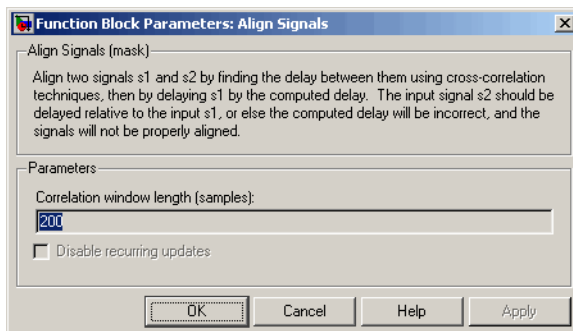
See the “Computing Delays” section of *Communications Blockset User's Guide* for an example that uses the Align Signals block in conjunction with the Error Rate Calculation block.

# Align Signals

---

See “Setting the Correlation Window Length” on page 2-188, on the reference page for the Find Delay block, for an example that illustrates how to set the correlation window length properly.

## Dialog Box



### Correlation window length

The number of samples the block uses to calculate the cross-correlations of the two signals.

### 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.

### 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.

## Algorithm

The Align Signals 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.

## See Also

Find Delay, Error Rate Calculation

## Purpose

Decode convolutional code using a posteriori probability (APP) method

## Library

Convolutional sublibrary of Channel Coding

## Description

The APP Decoder block performs a posteriori probability (APP) decoding of a convolutional code.



## Inputs and Outputs

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  $L(u)$  vectors have length  $Q*k$ . The integer  $Q$  is the number of frames that the block processes in each step.

The inputs can be either

- Sample-based vectors having the same dimension and orientation, with  $Q = 1$
- Frame-based column vectors 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.

## 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 Encoder” in the Communications 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 tail bits`, use the `Terminated` option in this block.

## 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



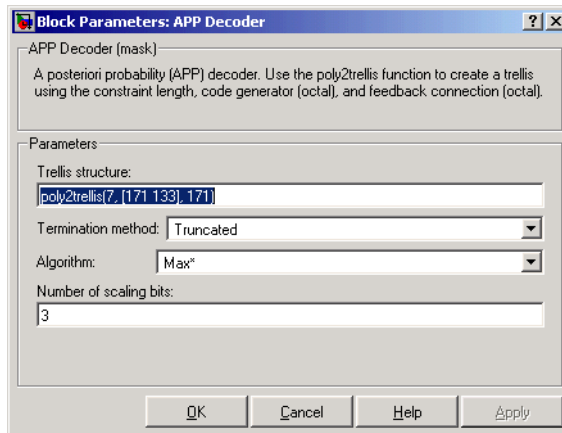
$$\log \sum_i \exp(a_i)$$

by other quantities. The Max option uses  $\max(a_i)$  as the approximation, while the Max\* option uses  $\max(a_i)$  plus a correction term given by

$$\ln(1 + \exp(-|a_{i-1} - a_i|)) \quad [3].$$

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  $(2^{\text{numScalingBits}})$  and divides the pre-output by the same factor). Use this parameter to avoid losing precision during the computations.

## Dialog Box



### Trellis structure

MATLAB structure that contains the trellis description of the convolutional encoder.

### Termination method

Either Truncated or Terminated. This parameter indicates how the convolutional encoder treats the trellis at the beginning and end of frames.

## Algorithm

Either True APP, Max\*, or Max.

## 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\*.

## See Also

Viterbi Decoder, Convolutional Encoder; poly2trellis  
(Communications Toolbox)

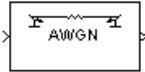
## References

- [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.

**Purpose** Add white Gaussian noise to input signal

**Library** Channels

## 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 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 uses the Signal Processing Blockset Random Source block to generate the noise. Random numbers are generated 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 whose length matches the number of channels in the input signal. For details on **Initial seed**, see the Random Source block reference page in the Signal Processing Blockset documentation set.

The signal inputs can only be of type `single` or `double`. The port data types are inherited from the signals that drive the block.

---

**Note** This block assumes a nominal impedance of 1 ohm.

---

## Frame-Based Processing and Input Dimensions

This block can process multichannel signals that are frame-based or sample-based. The guidelines below indicate how the block interprets your data, depending on the data's shape and frame status:

- If your input is a sample-based scalar, then the block adds scalar Gaussian noise to your signal.
- If your input is a sample-based vector or a frame-based row vector, then the block adds independent Gaussian noise to each channel.

# AWGN Channel

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- If your input is a frame-based column vector, then the block adds a frame of Gaussian noise to your single-channel signal.
- If your input is a frame-based m-by-n matrix, then the block adds a length-m frame of Gaussian noise independently to each of the n channels.

The input cannot be a sample-based m-by-n matrix if both m and n are greater than 1.

## 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 ( $E_b/N_0$ ), where the block calculates the variance from these quantities that you specify in the dialog box:
  - **$E_b/N_0$** , 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 ( $E_s/N_0$ ), where the block calculates the variance from these quantities that you specify in the dialog box:
  - **$E_s/N_0$** , 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
  - **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. If the first input signal is sample-based, then the variance input must be sample-based. If the first input signal is frame-based, then the variance input can be either frame-based with exactly one row, or sample-based.

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.

$$\text{NoiseVariance} = \frac{\text{SignalPower} \times \text{SymbolPeriod}}{\text{SampleTime} \times 10^{\frac{Es/No}{10}}}$$

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.

# AWGN Channel

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**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.

---

## Relationship Among $E_b/N_0$ , $E_s/N_0$ , and SNR Modes

For complex input signals, the AWGN Channel block relates  $E_b/N_0$ ,  $E_s/N_0$ , and SNR according to the following equations:

$$E_s/N_0 = (T_{\text{sym}}/T_{\text{samp}}) \cdot \text{SNR}$$

$$E_s/N_0 = E_b/N_0 + 10\log_{10}(k) \text{ in dB}$$

where

- $E_s$  = Signal energy (Joules)
- $E_b$  = Bit energy (Joules)
- $N_0$  = Noise power spectral density (Watts/Hz)
- $T_{\text{sym}}$  is the **Symbol period** parameter of the block in Es/No mode
- $k$  is the number of information bits per input symbol
- $T_{\text{samp}}$  is the inherited sample time of the block, in seconds

For real signal inputs, the AWGN Channel block relates  $E_s/N_0$  and SNR according to the following equation:

$$E_s/N_0 = 0.5 (T_{\text{sym}}/T_{\text{samp}}) \cdot \text{SNR}$$

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  $N_0/2$  Watts/Hz for real input signals, versus  $N_0$  Watts/Hz for complex signals.

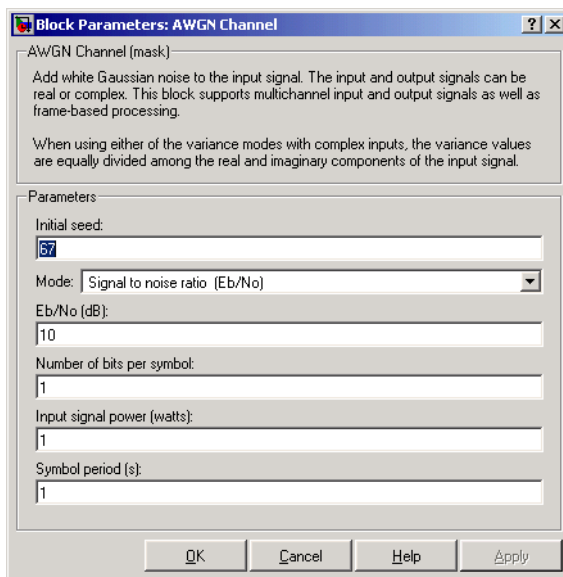
For more information about these quantities, see “Describing the Noise Level of an AWGN Channel” in the Communications Toolbox documentation.

## **Tuning Parameters in an RSim Executable (Real-Time Workshop)**

If you use the Real-Time Workshop® rapid simulation (RSim) target to build an RSim executable, then you can tune selected parameters 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 table below indicates, for different modes of the block, which parameters are tunable.

<b>Mode</b>	<b>Tunable Parameters</b>
Eb/No	<b>Eb/No, Input signal power</b>
Es/No	<b>Es/No, Input signal power</b>
SNR	<b>SNR, Input signal power</b>
Variance from mask	<b>Variance</b>

# AWGN Channel



## Dialog Box

### Initial seed

The seed for the Gaussian noise generator.

### 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.

### Eb/No (dB)

The ratio of bit energy per symbol to noise power spectral density, in decibels. This field appears only if **Mode** is set to Eb/No.

### Es/No (dB)

The ratio of signal energy per symbol to noise power spectral density, in decibels. This field appears only if **Mode** is set to Es/No.



**SNR (dB)**

The ratio of signal power to noise power, in decibels. This field appears only if **Mode** is set to SNR.

**Number of bits per symbol**

The number of bits in each input symbol. This field appears only if **Mode** is set to Eb/No.

**Input signal power (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.

**Symbol period (s)**

The duration of a channel symbol, in seconds. This field appears only if **Mode** is set to Eb/No or Es/No.

**Variance**

The variance of the white Gaussian noise. This field appears only if **Mode** is set to Variance from mask.

**Examples**

Many demonstration models and documentation examples use this block, including:

- Gray Coded 8-PSK demo, `commgraycode` (EbNo mode)
- Phase Noise Effects in 256-QAM demo, `commphasenoise` (EsNo mode)
- “Building a Frequency-Shift Keying Model” (EsNo mode)
- “Example: Using Raised Cosine Filters” (SNR mode)
- Discrete Multitone Signaling Demo, `commdmt` (Variance from mask mode)

**See Also**

Random Source (Signal Processing Blockset)

**Reference**

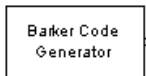
[1] Proakis, John G., *Digital Communications*, 4th Ed., McGraw-Hill, 2001.

# Barker Code Generator

**Purpose** Generate Barker Code

**Library** Sequence Generators sublibrary of Comm Sources

## Description



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,  $C_k$ , for a  $k$ -symbol shift of an  $N$ -bit code sequence,  $\{X_j\}$ , is given by

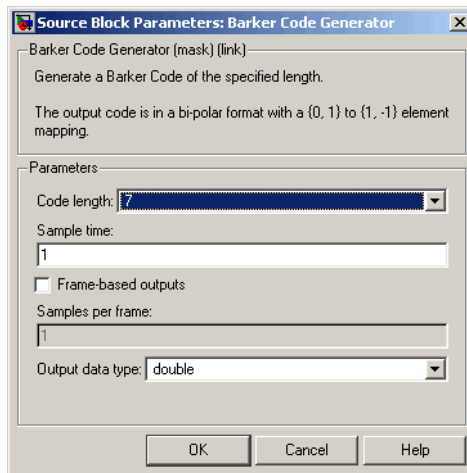
$$C_k = \sum_{j=1}^{N-k} X_j X_{j+k}$$

where  $X_j$  is an individual code symbol taking values +1 or -1 for  $j=1, 2, 3, \dots, N$ , and the adjacent symbols are assumed to be zero.

The Barker Code Generator block provides the codes listed in the following table:

Code length	Barker Code
1	[ -1 ]
2	[ -1 1 ] ;
3	[ -1 -1 1 ]
4	[ -1 -1 1 -1 ]
5	[ -1 -1 -1 1 -1 ]
7	[ -1 -1 -1 1 1 -1 1 ]
11	[ -1 -1 -1 1 1 1 -1 1 1 -1 1 ]
13	[ -1 -1 -1 -1 -1 1 1 -1 -1 1 -1 1 -1 ]

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “ Changing Source Block Parameters ” in the online Simulink documentation for details.

### Code length

The length of the Barker code.

### 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 the **Frame-based outputs** check box.

### 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

PN Sequence Generator

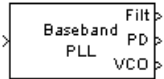
# Baseband PLL

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**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 functions such as `butter`, `cheby1`, and `cheby2` in Signal Processing Toolbox. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.

```
[num, den] = cheby2(3,40,100,'s')
```

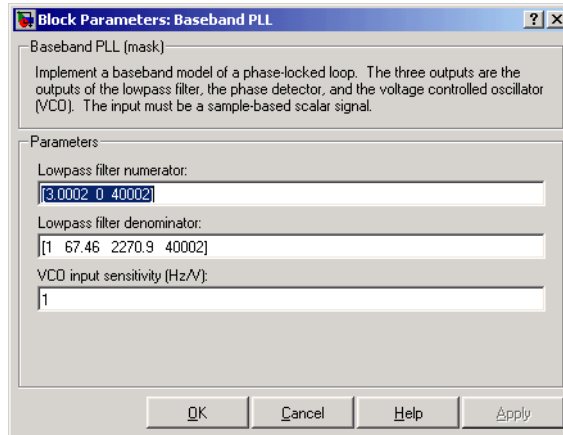
- A voltage-controlled oscillator (VCO). You specify the sensitivity of the VCO signal to its input using the **VCO input sensitivity** parameter. This parameter, measured in Hertz per volt, is a scale factor that determines how much the VCO shifts from its quiescent frequency.

The input signal represents the received signal. The input must be a sample-based scalar signal. The three output ports produce:

- The output of the filter
- The output of the phase detector
- The output of the VCO

This model is nonlinear; for a linearized version, use the Linearized Baseband PLL block.

## Dialog Box



### 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  $s$ .

### 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  $s$ .

### VCO input sensitivity (Hz/V)

This value scales the input to the VCO and, consequently, the shift from the VCO's quiescent frequency.

## 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 Blockset User's Guide*.

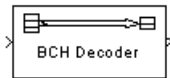
# BCH Decoder

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**Purpose** Decode BCH code to recover binary vector data

**Library** Block sublibrary of Channel Coding

## 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.

The input must be a frame-based column vector with an integer multiple of  $N$  elements. Each group of  $N$  input elements represents one codeword to be decoded. The values of  $N$  and  $K$  must produce a valid narrow-sense BCH code.

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 uses a shortened BCH code.

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 `bchenc` reference page in the Communications Toolbox documentation.

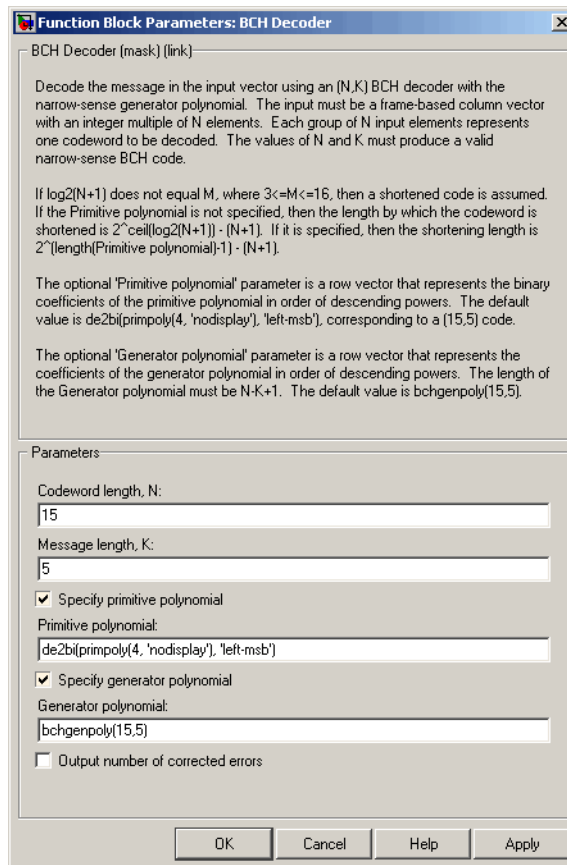
The primitive and generator polynomials may be specified in their respective fields, which appear after selecting their corresponding check boxes.

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 decoding of the codeword. A negative integer indicates that the block detected more errors than it could correct using the coding scheme.

The sample times of all input and output signals are equal.

This block supports double and boolean data types.

## Dialog Box



### Codeword length, N

The codeword length, which is also the vector length of the first input.

### Message length, K

The message length, which is also the vector length of the first output.

### Specify primitive polynomial

Selecting this check box enables the field **Primitive polynomial**.

## **Primitive polynomial**

This field is available only when **Specify primitive polynomial** is selected.

A row vector that represents the binary coefficients of the primitive polynomial in order of descending powers.

The default value is `de2bi(primpoly(4, 'nodisplay'), 'left-msb')`, corresponding to a (15,5) code.

## **Specify generator polynomial**

Selecting this check box enables the field **Generator polynomial**.

## **Generator polynomial**

This field is available only when **Specify generator polynomial** is selected.

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  $N-K+1$ .

The default value is `bchgenpoly(15,5)`.

## **Output number of corrected errors**

Selecting this check box gives the block an additional output port, which indicates the number of errors the block detected in the input codeword.

## **See Also**

`bchdec` (Communications Toolbox)

## **Pair Block**

BCH Encoder



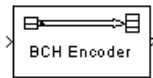
## Purpose

Create BCH code from binary vector data

## Library

Block sublibrary of Channel Coding

## Description



The BCH Encoder block creates a BCH code with message length  $K$  and codeword length  $N$ . You specify both  $N$  and  $K$  directly in the dialog box.

The input must be a frame-based column vector with an integer multiple of  $K$  elements. Each group of  $K$  input elements represents one message word to be encoded.

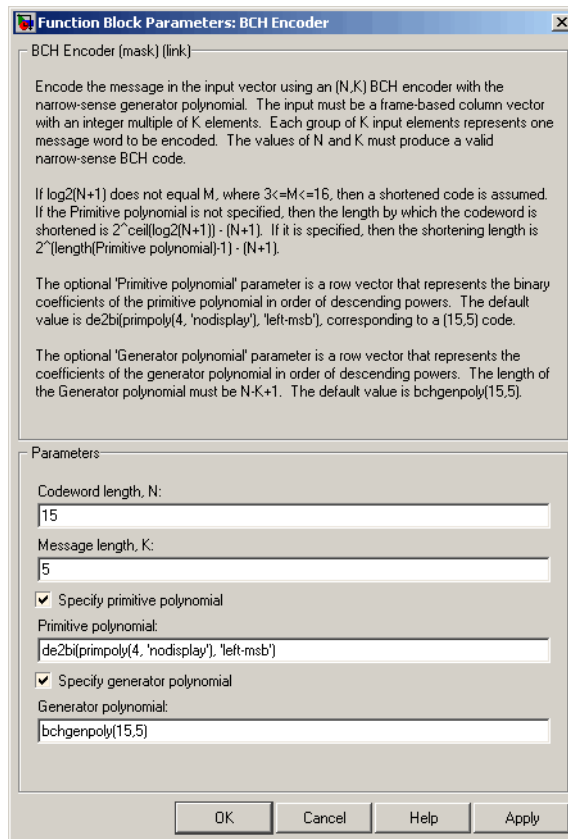
For a given codeword length  $N$ , only specific message lengths  $K$  are valid for a BCH code. For a full length BCH code,  $N$  must be of the form  $2^M-1$ , where  $3 \leq M \leq 16$ . If  $N$  is less than  $2^M-1$ , the block assumes a shortened BCH code.

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 `bchenc` reference page in the Communications Toolbox documentation.

This block supports double and boolean data types.

The primitive and generator polynomials may be specified in their respective fields, which appear after selecting their corresponding check boxes.

## Dialog Box



### Codeword length, N

The codeword length, which is also the output vector length.

### Message length, K

The message length, which is also the input vector length.

### Specify primitive polynomial

Selecting this check box enables the field **Primitive polynomial**.

## **Primitive polynomial**

This field is available only when **Specify primitive polynomial** is selected.

A row vector that represents the binary coefficients of the primitive polynomial in order of descending powers.

The default value is `de2bi(primpoly(4, 'nodisplay'), 'left-msb')`, corresponding to a (15,5) code.

## **Specify generator polynomial**

Selecting this check box enables the field **Generator polynomial**.

## **Generator polynomial**

This field is available only when **Specify generator polynomial** is selected.

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  $N-K+1$ .

The default value is `bchgenpoly(15,5)`.

## **Pair Block**

BCH Decoder

## **See Also**

`bchenc` (Communications Toolbox)

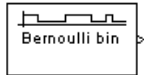
# Bernoulli Binary Generator

---

**Purpose** Generate Bernoulli-distributed random binary numbers

**Library** Random Data Sources sublibrary of Comm Sources

## Description



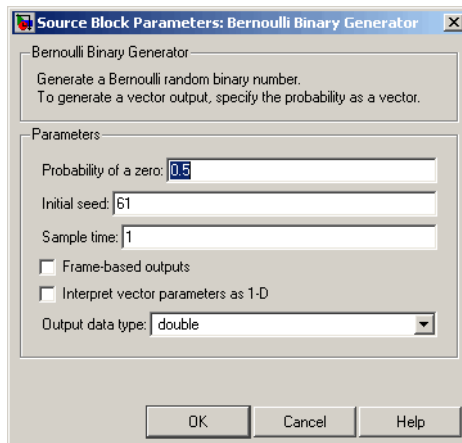
The Bernoulli Binary Generator block generates random binary numbers using a Bernoulli distribution. The Bernoulli distribution with parameter  $p$  produces zero with probability  $p$  and one with probability  $1-p$ . The Bernoulli distribution has mean value  $1-p$  and variance  $p(1-p)$ . The **Probability of a zero** parameter specifies  $p$ , and can be any real number between zero and one.

## 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 “Signal Attribute Parameters for Random Sources” in *Communications Blockset 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.

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

### Probability of a zero

The probability with which a zero output occurs.

### 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.

### 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.

# Bernoulli Binary 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.

## **See Also**

Random Integer Generator, Binary Symmetric Channel; randint (Communications Toolbox), rand (built-in MATLAB function)

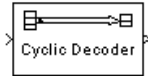
## Purpose

Decode systematic cyclic code to recover binary vector data

## Library

Block sublibrary of Channel Coding

## Description



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  $2^M - 1$  for some integer  $M$  greater than or equal to 3.

The input must contain exactly  $N$  elements. If it is frame-based, then it must be a column vector. The output is a vector of length  $K$ .

You can determine the systematic cyclic coding scheme in one of two ways:

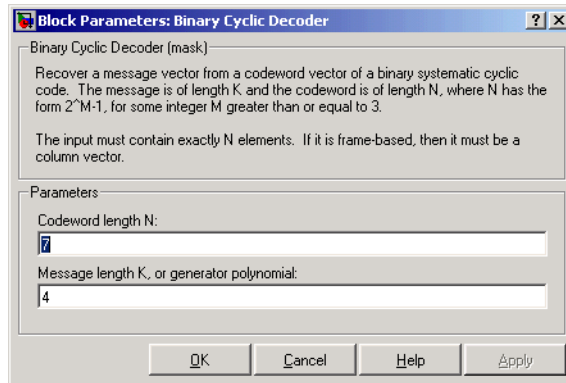
- To create an  $[N,K]$  code, enter  $N$  and  $K$  as the first and second dialog parameters, respectively. The block computes an appropriate generator polynomial, namely, `cyclpoly(N,K,'min')`.
- To create a code with codeword length  $N$  and a particular degree- $(N-K)$  binary *generator polynomial*, 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 `cyclpoly` function in Communications Toolbox.

This block supports double and boolean data types.

# Binary Cyclic Decoder

---

## Dialog Box



### Codeword length N

The codeword length N, which is also the input vector length.

### 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.

## Pair Block

Binary Cyclic Encoder

## See Also

cyclpoly (Communications Toolbox)



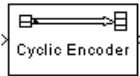
## Purpose

Create systematic cyclic code from binary vector data

## Library

Block sublibrary of Channel Coding

## 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  $2^M - 1$ , 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. The output is a vector of length  $N$ .

You can determine the systematic cyclic coding scheme in one of two ways:

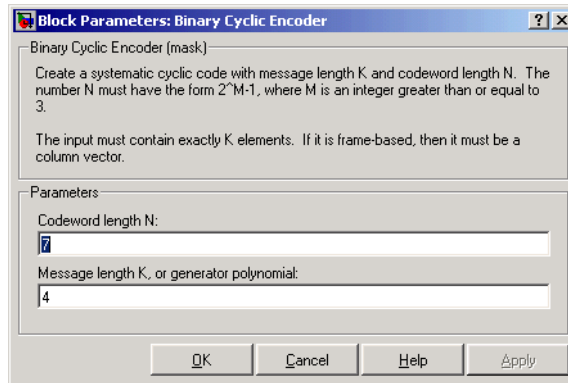
- To create an  $[N,K]$  code, enter  $N$  and  $K$  as the first and second dialog parameters, respectively. The block computes an appropriate generator polynomial, namely, `cyclpoly(N,K,'min')`.
- To create a code with codeword length  $N$  and a particular degree- $(N-K)$  binary *generator polynomial*, 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 `cyclpoly` function in Communications Toolbox.

This block supports double and boolean data types.

# Binary Cyclic Encoder

---

## Dialog Box



### Codeword length $N$

The codeword length, which is also the output vector length.

### 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.

## Pair Block

Binary Cyclic Decoder

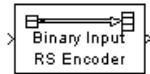
## See Also

cyclpoly (Communications Toolbox)

**Purpose** Create Reed-Solomon code from binary vector data

**Library** Block sublibrary of Channel Coding

## Description



The Binary-Input RS Encoder block creates a Reed-Solomon code with message length  $K$  and codeword length  $N$ . 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  $GF(2^M)$ , 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 2-44 below. The difference  $N-K$  must be an even integer.

The input and output are binary-valued signals that represent messages and codewords, respectively. The input must be a frame-based column vector whose length is an integer multiple of  $M*K$ . The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. The output is a frame-based column vector whose length is the same integer multiple of  $M*N$ , and whose data type is inherited from the input. For more information on representing data for Reed-Solomon codes, see the section “Integer Format (Reed-Solomon Only)” in *Communications Blockset User’s Guide*.

The default value of  $M$  is the smallest integer that is greater than or equal to  $\log_2(N+1)$ , that is,  $\text{ceil}(\log_2(N+1))$ . You can change the value of  $M$  from the default by specifying the primitive polynomial for  $GF(2^M)$ , as described in “Specifying the Primitive Polynomial” on page 2-44 below. If  $N$  is less than  $2^M-1$ , the block uses a shortened Reed-Solomon code.

Each  $M*K$  input bits represent  $K$  integers between 0 and  $2^M-1$ . Similarly, each  $M*N$  output bits represent  $N$  integers between 0 and  $2^M-1$ . These integers in turn represent elements of the Galois field  $GF(2^M)$ .

An  $(N,K)$  Reed-Solomon code can correct up to  $\text{floor}((N-K)/2)$  symbol errors (*not* bit errors) in each codeword.

# Binary-Input RS Encoder

---

## Specifying the Primitive Polynomial

You can specify the primitive polynomial that defines the finite field  $GF(2^M)$ , 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  $GF(2)$  of degree  $M$ , in descending order of powers. For example, to specify the polynomial  $x^3+x+1$ , enter the vector `[1 0 1 1]`.

If you do not select **Specify primitive polynomial**, the block uses the default primitive polynomial of degree  $M = \text{ceil}(\log_2(N+1))$ . You can display the default polynomial by entering `primpoly(ceil(log2(N+1)))` at the MATLAB prompt.

## Restrictions on the M and the Codeword Length 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  $3 < N < 2^{16}-1$ .
- If you do select **Specify primitive polynomial**,  $N$  must lie in the range  $3 \leq N < 2^{16}-1$  and  $M$  must lie in the range  $3 \leq M \leq 16$ .

## 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  $2^M-1$ . The vector represents a polynomial, in descending order of powers, whose coefficients are elements of  $GF(2^M)$  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

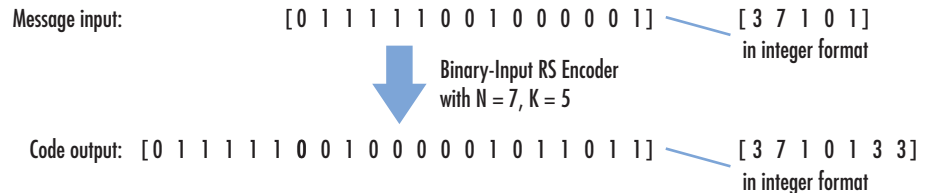
$$g(x) = (x+A^b)(x+A^{b+1})(x+A^{b+2})\dots(x+A^{b+N-K-1})$$

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(N1,K1)`, where  $N1=2^M-1$  and  $K1=K+(N1-N)$ , 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(N1,K1,poly)`.

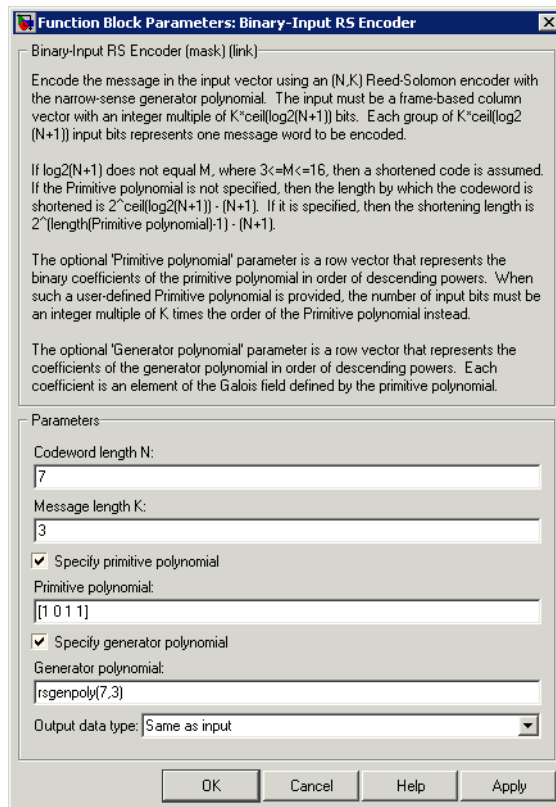
## Examples

Suppose  $M = 3$ ,  $N = 2^3-1 = 7$ , and  $K = 5$ . 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 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.



# Binary-Input RS Encoder

## Dialog Box



### Codeword length N

The codeword length. The output has vector length  $M \cdot N$ .

### Message length K

The message length. The input has vector length  $M \cdot K$ .

### Specify primitive polynomial

Selecting this check box enables the field **Primitive polynomial**.

### 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.

### **Specify generator polynomial**

Selecting this check box enables the field **Generator polynomial**.

### **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  $2^M-1$ , representing the generator polynomial in descending order of powers.

### **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`.

## **Pair Block**

Binary-Output RS Decoder

## **See Also**

Integer-Input RS Encoder

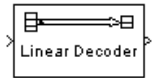
# Binary Linear Decoder

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**Purpose** Decode linear block code to recover binary vector data

**Library** Block sublibrary of Channel Coding

## Description



The Binary Linear Decoder block recovers a binary message vector from a binary codeword vector of a linear block code.

The **Generator matrix** parameter is the generator matrix for the block code. For proper decoding, this should match the **Generator matrix** parameter in the corresponding Binary Linear Encoder block. If  $N$  is the codeword length of the code, then **Generator matrix** must have  $N$  columns. If  $K$  is the message length of the code, then the **Generator matrix** parameter must have  $K$  rows.

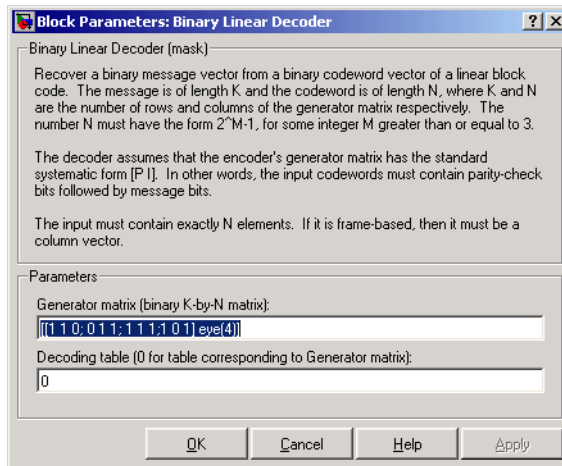
The input must contain exactly  $N$  elements. If it is frame-based, then it must be a column vector. The output is a vector of length  $K$ .

The decoder tries to correct errors, using the **Decoding table** parameter. If **Decoding table** is the scalar 0, then the block defaults to the table produced by the Communications Toolbox function `syndtable`. Otherwise, **Decoding table** must be a  $2^{N-K}$ -by- $N$  binary matrix. The  $r$ th row of this matrix is the correction vector 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.

This block supports double and boolean data types.



## Dialog Box



### Generator matrix

Generator matrix for the code; same as in Binary Linear Encoder block.

### Decoding table

Either a  $2^{N-K}$ -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.

## Pair Block

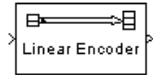
Binary Linear Encoder

# Binary Linear Encoder

**Purpose** Create linear block code from binary vector data

**Library** Block sublibrary of Channel Coding

## Description

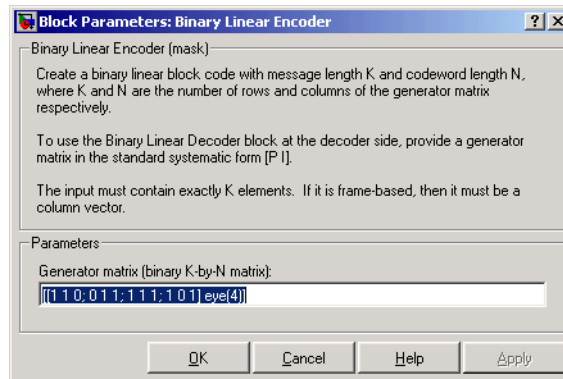


The Binary Linear Encoder block creates a binary linear block code using a generator matrix that you specify. If  $K$  is the message length of the code, then the **Generator matrix** parameter must have  $K$  rows. If  $N$  is the codeword length of the code, then **Generator matrix** must have  $N$  columns.

The input must contain exactly  $K$  elements. If it is frame-based, then it must be a column vector. The output is a vector of length  $N$ .

This block supports double and boolean data types.

## Dialog Box



### Generator matrix

A  $K$ -by- $N$  matrix, where  $K$  is the message length and  $N$  is the codeword length.

## Pair Block

Binary Linear Decoder

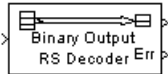
## Purpose

Decode Reed-Solomon code to recover binary vector data

## Library

Block sublibrary of Channel Coding

## Description



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  $N$ . 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  $GF(2^M)$ , 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 2-44. The difference  $N-K$  must be an even integer.

The input and output are binary-valued signals that represent messages and codewords, respectively. The input must be a frame-based column vector whose length is an integer multiple of  $M \cdot K$ . The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. The output is a frame-based column vector whose length is the same integer multiple of  $M \cdot N$ , and whose data type is inherited from the input. For more information on representing data for Reed-Solomon codes, see “Integer Format (Reed-Solomon Only)” in *Communications Blockset User’s Guide*.

The default value of  $M$  is  $\text{ceil}(\log_2(N+1))$ , 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  $GF(2^M)$ , as described in “Specifying the Primitive Polynomial” on page 2-44 below. If  $N$  is less than  $2^M-1$ , 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 2-44.

Each  $M \cdot K$  input bits represent  $K$  integers between 0 and  $2^M-1$ . Similarly, each  $M \cdot N$  output bits represent  $N$  integers between 0 and

# Binary-Output RS Decoder

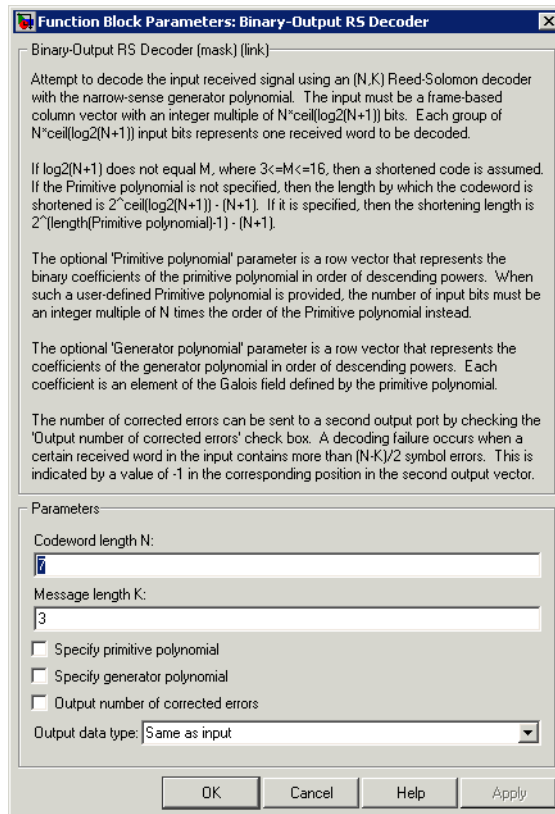
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$2^M - 1$ . These integers in turn represent elements of the Galois field  $GF(2^M)$ .

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  $\text{floor}((N - K) / 2)$  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 port for number of corrected errors**. This removes the block's second output port.

## Dialog Box



### Codeword length N

The codeword length. The input has vector length  $M \cdot N$ .

### Message length K

The message length. The first output has vector length  $M \cdot K$ .

### Specify primitive polynomial

Selecting this check box enables the field **Primitive polynomial**.

### Primitive polynomial

This field is available only when **Specify primitive polynomial** is selected.

# Binary-Output RS Decoder

---

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**

This field is available only when **Specify generator polynomial** is selected.

Integer row vector, whose entries are in the range from 0 to  $2^M-1$ , representing the generator polynomial in descending order of powers.

## **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.

## **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.

## **Algorithm**

This block uses the Berlekamp-Massey decoding algorithm. For information about this algorithm, see the references listed below.

## **Pair Block**

Binary-Input RS 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.

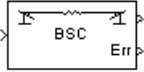
## **See Also**

Integer-Output RS Decoder

**Purpose** Introduce binary errors

**Library** Channels

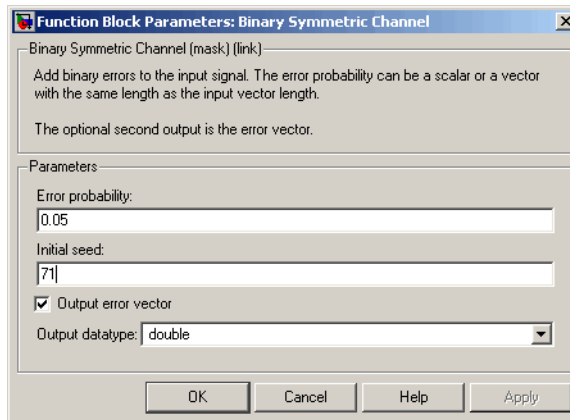
**Description** The Binary Symmetric Channel block introduces binary errors to the signal transmitted through this channel.



The input port is the transmitted binary signal. The input can be either a scalar, a sample-based vector, or a frame-based row vector. This block processes each vector element independently, and introduces an error in a given spot with probability **Error probability**. Random numbers are generated using the Ziggurat method, which is the same method used by the MATLAB randn function.

The first output port is the binary signal that has passed through the channel. The second output port is the vector of errors that were introduced. To suppress the second output port, clear the **Output error vector** check box.

## Dialog Box



### Error probability

The probability that a binary error will occur. The value of this parameter must be between zero and one.

# Binary Symmetric Channel

---

**Initial seed**

The initial seed value for the random number generator.

**Output error vector**

If this box is checked, then the block outputs the vector of errors.

**Output datatype**

You can set the output data type to double or boolean.

**See Also**

Bernoulli Binary Generator

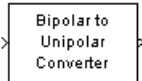


# Bipolar to Unipolar Converter

**Purpose** Map bipolar signal into unipolar signal in range  $[0, M-1]$

**Library** Utility Blocks

## Description

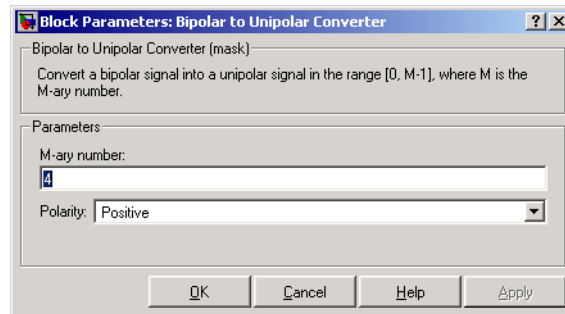


The Bipolar to Unipolar Converter block maps the bipolar input signal to a unipolar output signal. If the input consists of integers in the set  $\{-M+1, -M+3, -M+5, \dots, M-1\}$ , where  $M$  is the **M-ary number** parameter, then the output consists of integers between 0 and  $M-1$ .

The table below shows how the block's mapping depends on the **Polarity** parameter.

Polarity Parameter Value	Output Corresponding to Input Value of $k$
Positive	$(M-1+k)/2$
Negative	$(M-1-k)/2$

## Dialog Box



### M-ary number

The number of symbols in the bipolar or unipolar alphabet.

### Polarity

A value of **Positive** (respectively, **Negative**) causes the block to maintain (respectively, reverse) the relative ordering of symbols in the alphabets.

# Bipolar to Unipolar Converter

---

## Examples

If the input is [-3; -1; 1; 3], the **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].

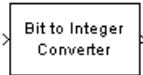
## Pair Block

Unipolar to Bipolar Converter

**Purpose** Map vector of bits to corresponding vector of integers

**Library** Utility Blocks

## Description



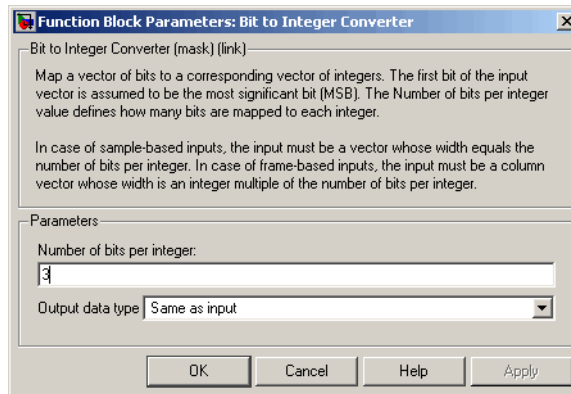
The Bit to Integer Converter block maps groups of bits in the input vector to integers in the output vector. If  $M$  is the **Number of bits per integer** parameter, then the block maps each group of  $M$  bits to an integer between 0 and  $2^M-1$ . As a result, the output vector length is  $1/M$  times the input vector length.

If the input is sample-based input, then it must be a vector whose length equals the **Number of bits per integer** parameter. If the input is frame-based, then it must be a column vector whose length is an integer multiple of **Number of bits per integer**.

The block interprets the first bit in each group as the most significant bit.

The block can accept the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, and double.

## Dialog Box



### 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 31.

# Bit to Integer Converter

---

## Output data type

The output data type can be set to `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, or `double`. The output can be of type `boolean` only if `M` is 2 and this field is set to `Same` as `input`.

## Examples

If the input is `[0; 1; 1; 1; 1; 1; 0; 1]` and the **Number of bits per integer** parameter is 4, then the output is `[7; 13]`. The block maps the first group of four bits (0, 1, 1, 1) to 7 and the second group of four bits (1, 1, 0, 1) to 13. Notice that the output length is one-fourth of the output length.

## Pair Block

Integer to Bit Converter

**Purpose** Demodulate BPSK-modulated data

**Library** PM, in Digital Baseband sublibrary of Modulation

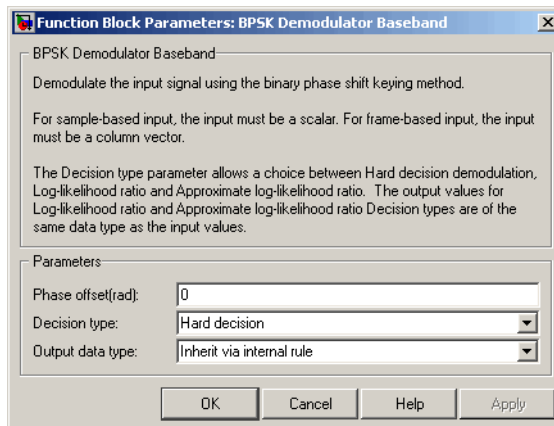
## Description



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. The input can be either a scalar or a frame-based column vector. The block can accept the data types `single` and `double`.

The input must be a discrete-time complex signal. The block maps the points  $\exp(j\theta)$  and  $-\exp(j\theta)$  to 0 and 1, respectively, where  $\theta$  is the **Phase offset** parameter.

## Dialog Box



### Phase offset (rad)

The phase of the zeroth point of the signal constellation.

### Decision type

Specifies the use of hard decision, LLR, or approximate LLR during demodulation. See “Exact LLR Algorithm” and “Approximate LLR Algorithm” for algorithm details.

# BPSK Demodulator Baseband

---

## 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.

When **Decision type** is set to Hard decision, output can be int8, uint8, int16, uint16, int32, uint32, boolean, single, or double. When **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is inherited from the input (i.e., if the input is of data type double, the output is also of data type double).

## 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.

## Noise variance

This field appears when **Noise variance source** is set to Dialog.

If this value is very small (i.e., SNR is very high), LLR computations may yield Inf or -Inf because the LLR algorithm would involve computing exponentials of very large or very small numbers using finite precision arithmetic. In such cases, use of approximate LLR is recommended, as its algorithm does not involve computing exponentials.

## Pair Block

BPSK Modulator Baseband

## See Also

M-PSK Demodulator Baseband, QPSK Demodulator Baseband, DBPSK Demodulator Baseband

**Purpose** Modulate using binary phase shift keying method

**Library** PM, in Digital Baseband sublibrary of Modulation

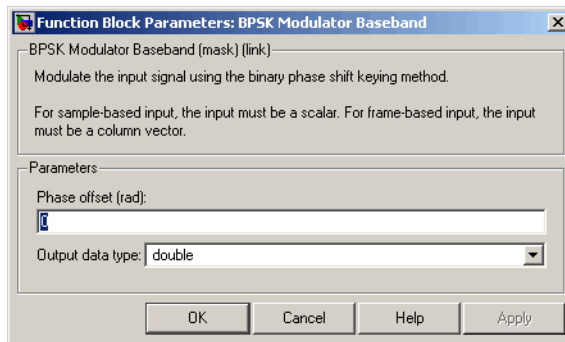
## Description



The BPSK Modulator Baseband block modulates using the binary phase shift keying method. The output is a baseband representation of the modulated signal. For both integer and bit inputs, this block can accept the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, and double.

The input must be a discrete-time binary-valued signal. If the input bit is 0 or 1, respectively, then the modulated symbol is  $\exp(j\theta)$  or  $-\exp(j\theta)$  respectively, where  $\theta$  is the **Phase offset** parameter.

## Dialog Box



### Phase offset (rad)

The phase of the zeroth point of the signal constellation.

### 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.

# BPSK Modulator Baseband

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## **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 Simulink Fixed Point. 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 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.

## **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.

## **Pair Block**

BPSK Demodulator Baseband

## **See Also**

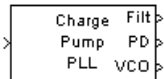
M-PSK Modulator Baseband, QPSK Modulator Baseband, DBPSK Modulator Baseband



**Purpose** Implement charge pump phase-locked loop using digital phase detector

**Library** Components sublibrary of Synchronization

## Description



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'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. 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.

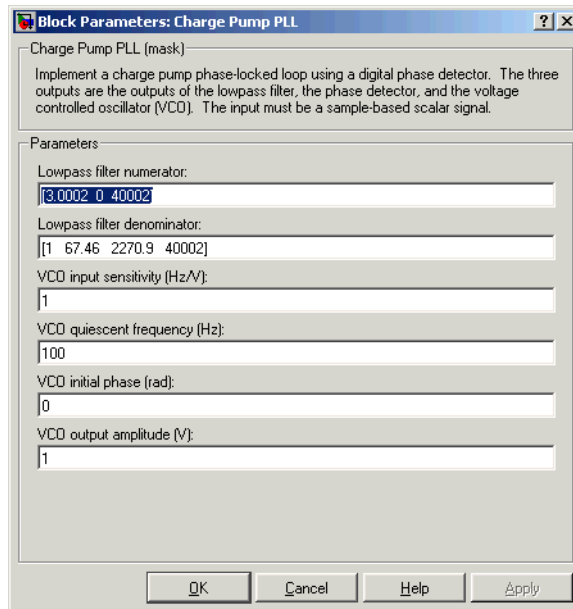
The input signal represents the received signal. The input must be a sample-based scalar signal. The three output ports produce:

- The output of the filter
- The output of the phase detector
- The output of the VCO

# Charge Pump PLL

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  $\pi$ . 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



### 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  $s$ .

### 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  $s$ .

**VCO input sensitivity (Hz/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 (Hz)**

The frequency of the VCO signal when the voltage applied to it is zero. This should match the 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**

Phase-Locked Loop

**References**

For more information about digital phase-locked loops, see the works listed in “Selected Bibliography for Synchronization” in *Communications Blockset User’s Guide*.

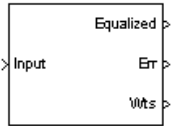
# CMA Equalizer

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**Purpose** Equalize using constant modulus algorithm

**Library** Equalizers

## 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.

## Input and Output Signals

The port labeled Input receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled Equalized outputs the result of the equalization process.

You can configure the block to have one or more of the extra ports listed in the table below.

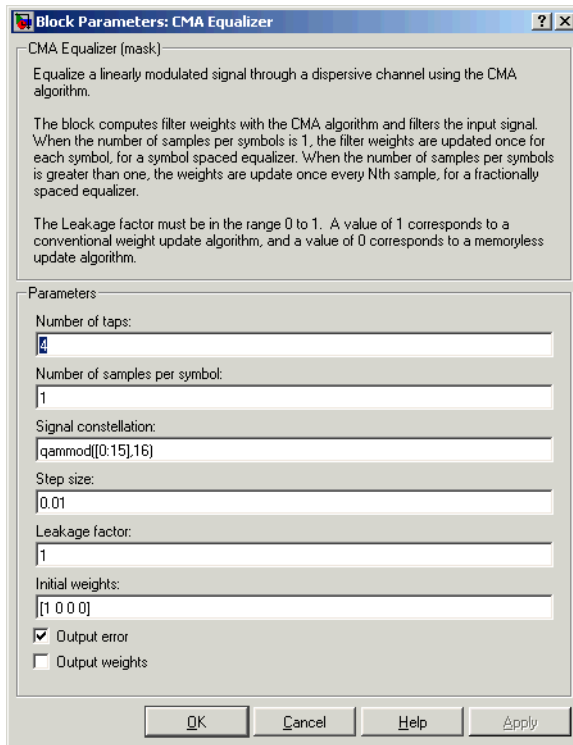
Port	Meaning	How to Enable
Err output	$y(R -  y ^2)$ , where $y$ is the equalized signal and $R$ is a constant related to the signal constellation	Check the <b>Output error</b> check box.
Wts output	A vector listing the weights after the block has processed either the current input frame or, in sample-based mode, the current input sample.	Check the <b>Output weights</b> check box.

## 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 blockset). If you need to determine the delay, you can use the Find Delay block.

# CMA Equalizer

## Dialog Box



### Number of taps

The number of taps in the filter of the 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.

### Step size

The step size of the CMA.

**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.

**Initial weights**

A vector that lists the initial weights for the taps.

**Output error**

If you check this box, the block outputs the error signal described in the table above.

**Output weights**

If you check this box, the block outputs the current weights.

**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

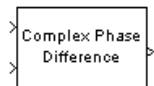
# Complex Phase Difference

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**Purpose** Output phase difference between two complex input signals

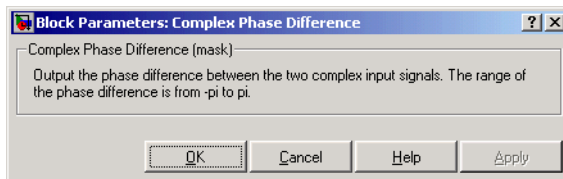
**Library** Utility Blocks

**Description** 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  $-\pi$  and  $\pi$ .



The input signals can have any size or frame status. This block processes each pair of elements independently.

## Dialog Box



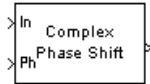
**See Also** Complex Phase Shift



**Purpose** Shift phase of complex input signal by second input value

**Library** Utility Blocks

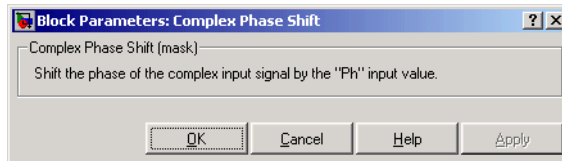
## Description



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.

## Dialog Box



**See Also** Complex Phase Difference

# Continuous-Time VCO

**Purpose** Implement voltage-controlled oscillator

**Library** Components sublibrary of Synchronization

## Description



The Continuous-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

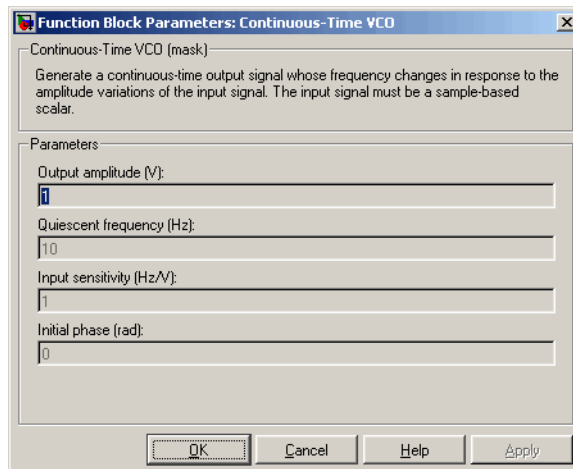
$$y(t) = A_c \cos\left(2\pi f_c t + 2\pi k_c \int_0^t u(\tau) d\tau + \phi\right)$$

where  $A_c$  is the **Output amplitude** parameter,  $f_c$  is the **Quiescent frequency** parameter,  $k_c$  is the **Input sensitivity** parameter, and  $\phi$  is the **Initial phase** parameter.

This block uses a continuous-time integrator to interpret the equation above.

The input and output signals are both sample-based scalars.

## Dialog Box



**Output amplitude**

The amplitude of the output.

**Quiescent frequency**

The frequency of the oscillator output when the input signal is zero.

**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.

**Initial phase**

The initial phase of the oscillator in radians.

**See Also**

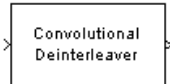
Discrete-Time VCO

# Convolutional Deinterleaver

**Purpose** Restore ordering of symbols that were permuted using shift registers

**Library** Convolutional sublibrary of Interleaving

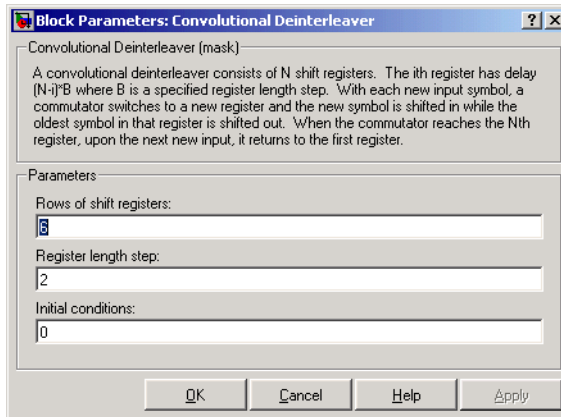
**Description** The Convolutional Deinterleaver block recovers a signal that was interleaved using the Convolutional Interleaver block. The parameters in the two blocks should have the same values.



The input can be either a scalar or a frame-based column vector. It can be real or complex. The sample times of the input and output signals are the same.

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.

## Dialog Box



### Rows of shift registers

The number of shift registers that the block uses internally.

### Register length step

The difference in symbol capacity of each successive shift register, where the last register holds zero symbols.

**Initial conditions**

The values that fill each shift register when the simulation begins.

**Examples**

For an example that uses this block, see “Example: Convolutional Interleavers”.

**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.

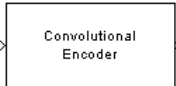
# Convolutional Encoder

---

**Purpose** Create convolutional code from binary data

**Library** Convolutional sublibrary of Channel Coding

**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.



## Input and Output Sizes

If the encoder takes  $k$  input bit streams (that is, can receive  $2^k$  possible input symbols), this block's input vector length is  $L \cdot k$  for some positive integer  $L$ . Similarly, if the encoder produces  $n$  output bit streams (that is, can produce  $2^n$  possible output symbols), this block's output vector length is  $L \cdot n$ .

The input can be a sample-based vector with  $L = 1$ , or a frame-based column vector with any positive integer for  $L$ .

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.

## Specifying the Encoder

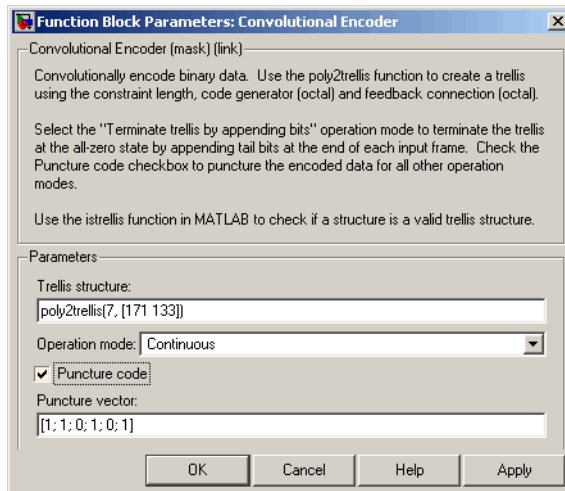
To define the convolutional encoder, use the **Trellis structure** parameter. This parameter is a MATLAB structure whose format is described in "Trellis Description of a Convolutional Encoder" in the Communications Toolbox documentation. You can use this parameter field in two ways:

- 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)
```

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.



## Dialog Box

### Trellis structure

MATLAB structure that contains the trellis description of the convolutional encoder.

# Convolutional Encoder

---

## Operation mode

In **Continuous** mode, 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 states 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 states 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  $s = \text{constraint length} - 1$  (or, in the case of multiple constraint lengths,  $s = \text{sum}(\text{ConstraintLength}(i) - 1)$ ). This mode is supported for frame based inputs only.

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.

## Puncture code

Selecting this option opens the field **Puncture vector**.

## Puncture vector

Vector used to puncture the encoded data. The puncture vector is a pattern of 1s and 0s where the 0s indicate the punctured bits. This field appears when the check box **Punctured code** is selected.

## Puncture Pattern Examples

For some commonly used puncture patterns for specific rates and polynomials, see the last three references.

## 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.
- [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.

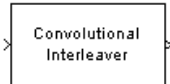
# Convolutional Interleaver

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**Purpose** Permute input symbols using set of shift registers

**Library** Convolutional sublibrary of Interleaving

## Description



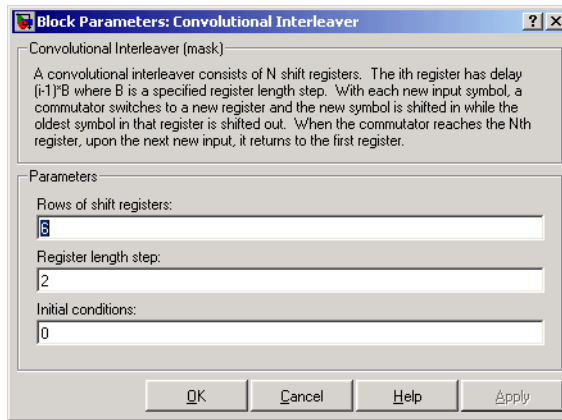
The Convolutional Interleaver block permutes the symbols in the input signal. Internally, it uses a set of shift registers. The delay value of the  $k$ th shift register is  $(k-1)$  times the **Register length step** parameter. The number of shift registers is the value of the **Rows of shift registers** 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.

The input can be either a scalar or a frame-based column vector. It can be real or complex. The sample times of the input and output signals are the same.

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.

## Dialog Box



### Rows of shift registers

The number of shift registers that the block uses internally.

### Register length step

The number of additional symbols that fit in each successive shift register, where the first register holds zero symbols.

### Initial conditions

The values that fill each shift register when the simulation begins.

## Examples

For an example that uses this block, see “Example: Convolutional Interleavers”.

## Pair Block

Convolutional Deinterleaver

## See Also

General Multiplexed Interleaver, Helical 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.

# Convolutional Interleaver

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[3] Ramsey, J. L. "Realization of Optimum Interleavers." *IEEE Transactions on Information Theory*, IT-16 (3), May 1970. 338-345.

## Purpose

Demodulate CPFSK-modulated data

## Library

CPM, in Digital Baseband sublibrary of Modulation

## Description



The CPFSK Demodulator Baseband block demodulates a signal that was modulated using the continuous phase frequency shift keying method. The input 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  $K$ .

The **Modulation index** parameter times  $\pi$  radians is the phase shift in the modulated signal due to the latest symbol, when that symbol is the integer 1. The **Phase offset** parameter is the initial phase of the modulated waveform.

### Traceback Length and Output Delays

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The **Traceback length** 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.

- If the input signal is sample-based, then the delay consists of  $D+1$  zero symbols.
- If the input signal is frame-based, then the delay consists of  $D$  zero symbols.

### Outputs and Symbol Sets

If the **Output type** parameter is set to Integer, then the block produces odd integers between  $-(M-1)$  and  $M-1$ .

If the **Output type** parameter is set 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

# CPFSK Demodulator Baseband

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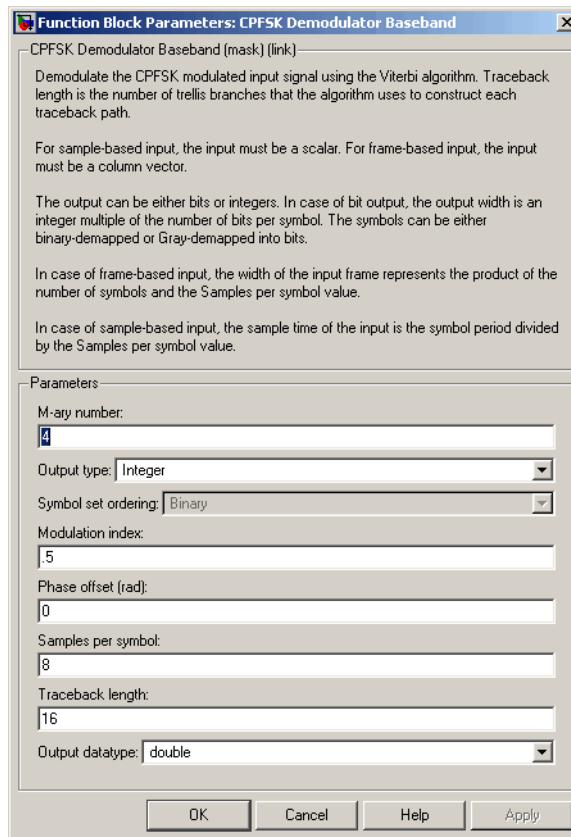
block maps each nonnegative integer to a binary word, using a mapping that depends on whether the **Symbol set ordering** parameter is set to Binary or Gray. For more information about Gray and binary coding, see “Binary-Valued and Integer-Valued Signals” in *Communications Blockset User’s Guide*.

The input can be either a scalar or a frame-based column vector and must be of type single or double.

## **Processing an Upsampled Modulated Signal**

The input signal can be an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor. It must be a positive integer. For more information, see “Upsampled Signals and Rate Changes” in *Communications Blockset User’s Guide*.

## Dialog Box



### M-ary number

The size of the alphabet.

### Output type

Determines whether the output consists of integers or groups of bits.

### 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.

# CPFSK Demodulator Baseband

---

## **Modulation index**

The number of half-revolutions of phase shift in the modulated signal after modulating the latest symbol of 1.

## **Phase offset (rad)**

The initial phase of the modulated waveform.

## **Samples per symbol**

The number of input samples that represent each modulated symbol.

## **Traceback length**

The number of trellis branches that the Viterbi Decoder block uses to construct each traceback path.

## **Output datatype**

The output data type can be boolean, int8, int16, int32, or double.

## **Pair Block**

CPFSK Modulator Baseband

## **See Also**

CPM Demodulator Baseband, Viterbi Decoder, M-FSK Demodulator Baseband

## **References**

[1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. *Digital Phase Modulation*. New York: Plenum Press, 1986.



**Purpose** Modulate using continuous phase frequency shift keying method

**Library** CPM, in Digital Baseband sublibrary of Modulation

## Description



The CPFSK Modulator Baseband block modulates using the continuous phase frequency shift keying method. The output 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  $K$ .

The **Modulation index** parameter times  $\pi$  radians is the phase shift due to the latest symbol when that symbol is the integer 1. The **Phase offset** parameter is the initial phase of the output waveform, measured in radians.

For the exact definitions of the rectangular pulse shape that this block uses, see the work by Anderson, Aulin, and Sundberg among the references listed below.

## Inputs and Symbol Sets

If the **Input type** parameter is set to Integer, then the block accepts odd integers between  $-(M-1)$  and  $M-1$ .

If the **Input type** parameter is set to Bit, then 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 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  $2k-(M-1)$  and proceeds as in the integer input mode. For more information, see “Binary-Valued and Integer-Valued Signals” in *Communications Blockset User’s Guide*.

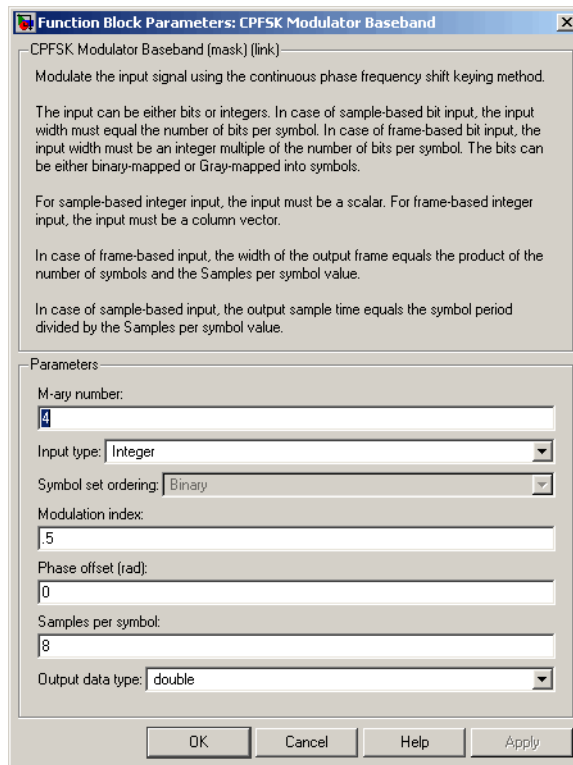
The input can be either a scalar or a frame-based column vector. If **Input type** is Bit, then the input can also be a vector of length  $K$ .

# CPFSK Modulator Baseband

## Upsampling the Modulated Signal

This block can output an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor. It must be a positive integer. For more information, see “Upsampled Signals and Rate Changes” in *Communications Blockset User’s Guide*.

### Dialog Box



The dialog box titled "Function Block Parameters: CPFSK Modulator Baseband" contains the following text and parameters:

CPFSK Modulator Baseband (mask) (link)  
Modulate the input signal using the continuous phase frequency shift keying method.

The input can be either bits or integers. In case of sample-based bit input, the input width must equal the number of bits per symbol. In case of frame-based bit input, the input width must be an integer multiple of the number of bits per symbol. The bits can be either binary-mapped or Gray-mapped into symbols.

For sample-based integer input, the input must be a scalar. For frame-based integer input, the input must be a column vector.

In case of frame-based input, the width of the output frame equals the product of the number of symbols and the Samples per symbol value.

In case of sample-based input, the output sample time equals the symbol period divided by the Samples per symbol value.

Parameters:

- M-ary number: 4
- Input type: Integer
- Symbol set ordering: Binary
- Modulation index: .5
- Phase offset (rad): 0
- Samples per symbol: 8
- Output data type: double

Buttons: OK, Cancel, Help, Apply

### M-ary number

The size of the alphabet.

### Input type

Indicates whether the input consists of integers or groups of bits.

**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.

**Modulation index**

The number of half-revolutions of phase shift due to the latest symbol when that symbol is the integer 1.

**Phase offset (rad)**

The initial phase of the output waveform.

**Samples per symbol**

The number of output samples that the block produces for each integer or binary word in the input.

**Output datatype**

The output data type can be single or double.

**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.

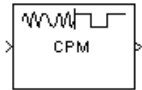
# CPM Demodulator Baseband

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**Purpose** Demodulate CPM-modulated data

**Library** CPM, in Digital Baseband sublibrary of Modulation

## Description



The CPM Demodulator Baseband block demodulates a signal that was modulated using continuous phase modulation. The input 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  $K$ .

The input can be either a scalar or a frame-based column vector and must be of type `single` or `double`.

The **Modulation index**, **Frequency pulse shape**, **Rolloff**, **BT product**, **Pulse length**, **Symbol prehistory**, and **Phase offset** parameters are as described on the reference page for the CPM Modulator Baseband block.

### Traceback Length and Output Delays

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The **Traceback length** 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.

- If the input signal is sample-based, then the delay consists of  $D+1$  zero symbols.
- If the input signal is frame-based, then the delay consists of  $D$  zero symbols.

### Outputs and Symbol Sets

If the **Output type** parameter is set to `Integer`, then the block produces odd integers between  $-(M-1)$  and  $M-1$ .

If the **Output type** parameter is set 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. For more information about Gray and binary coding, see “Binary-Valued and Integer-Valued Signals” in *Communications Blockset User’s Guide*.

## **Processing an Upsampled Modulated Signal**

The input signal can be an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor. It must be a positive integer. For more information, see “Upsampled Signals and Rate Changes” in *Communications Blockset User’s Guide*.

# CPM Demodulator Baseband

## Dialog Box

**Function Block Parameters: CPM Demodulator Baseband** [mask] [link]

Demodulate the CPM modulated input signal using the Viterbi algorithm. Traceback length is the number of trellis branches that the algorithm uses to construct each traceback path.

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. In case of bit output, the output width is an integer multiple of the number of bits per symbol. The symbols can be either binary-demapped or Gray-demapped into bits.

In case of frame-based input, the width of the input frame represents the product of the number of symbols and the Samples per symbol value.

In case of sample-based input, the sample time of the input is the symbol period divided by the Samples per symbol value.

Parameters

M-ary number:

Output type:

Symbol set ordering:

Modulation index:

Frequency pulse shape:

Pulse length (symbol intervals):

Symbol prehistory:

Phase offset (rad):

Samples per symbol:

Traceback length:

Output datatype:

OK Cancel Help Apply

### M-ary number

The size of the alphabet.

### Output type

Determines whether the output consists of integers or groups of bits.

**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.

**Modulation index**

The number of half-revolutions of phase shift in the modulated signal after modulating the latest symbol of 1.

**Frequency pulse shape**

The type of pulse shaping that the corresponding modulator uses to smooth the phase transitions of the modulated signal.

**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.

**Rolloff**

The rolloff factor of the raised cosine filter. This field appears only when **Frequency pulse shape** is set to Spectral Raised Cosine.

**BT product**

The product of bandwidth and time. This field appears only when **Frequency pulse shape** is set to Gaussian.

**Pulse length (symbol intervals)**

The length of the frequency pulse shape.

**Symbol prehistory**

The data symbols used by the modulator before the start of the simulation.

**Phase offset (rad)**

The initial phase of the modulated waveform.

**Samples per symbol**

The number of input samples that represent each modulated symbol.

# CPM Demodulator Baseband

---

## **Traceback length**

The number of trellis branches that the Viterbi Decoder block uses to construct each traceback path.

## **Output datatype**

The output data type can be boolean, int8, int16, int32, or double.

## **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.



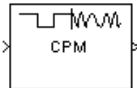
## Purpose

Modulate using continuous phase modulation

## Library

CPM, in Digital Baseband sublibrary of Modulation

## Description



The CPM Modulator Baseband block modulates using continuous phase modulation. The output 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  $K$ .

Continuous phase modulation uses pulse shaping to smooth the phase transitions of the modulated signal. Using the **Frequency pulse shape** parameter, you can choose these types of 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)

For the exact definitions of these pulse shapes, see the work by Anderson, Aulin, and Sundberg among the references listed below. Each pulse shape has a corresponding pulse duration. The **Pulse length** parameter measures this quantity in symbol intervals.

The **Modulation index** parameter times  $\pi$  radians is the phase shift due to the latest symbol when that symbol is the integer 1. The **Phase offset** parameter is the initial phase of the output waveform, measured in radians.

The **Symbol prehistory** parameter is a scalar or vector that specifies the data symbols used 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.

## Inputs and Symbol Sets

If the **Input type** parameter is set to Integer, then the block accepts odd integers between  $-(M-1)$  and  $M-1$ .

If the **Input type** parameter is set to Bit, then the block accepts groupings of  $K$  bits. Each grouping is called a binary *word*. The input vector length must be an integer multiple of  $K$ .

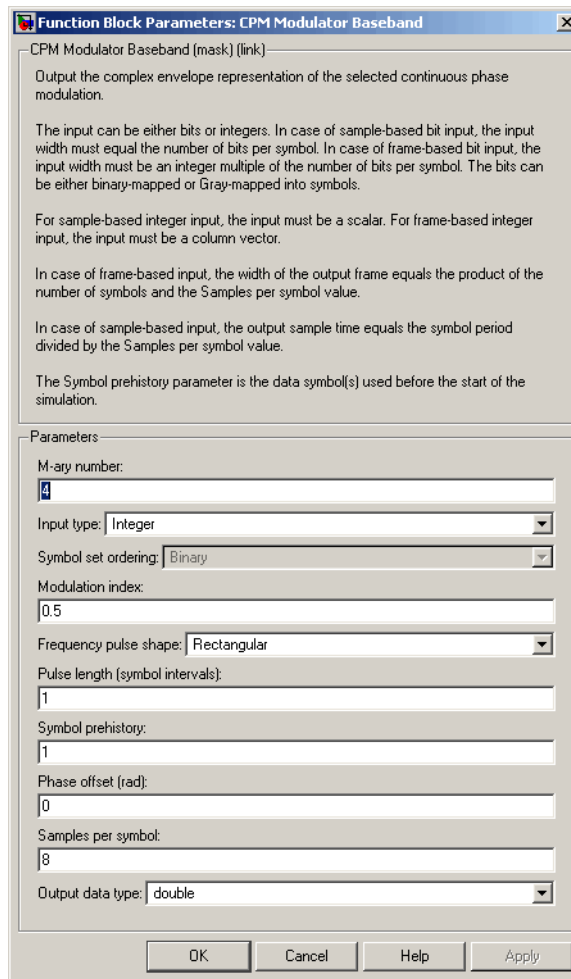
In binary input mode, the block maps each binary word to an integer between 0 and  $M-1$ , using a mapping that depends on whether the **Symbol set ordering** parameter is set to Binary or Gray. The block then maps the integer  $k$  to the intermediate value  $2k-(M-1)$  and proceeds as in the integer input mode. For more information, see “Binary-Valued and Integer-Valued Signals” in *Communications Blockset User’s Guide*.

The input can be either a scalar or a frame-based column vector. If **Input type** is Bit, then the input can also be a vector of length  $K$ .

## Upsampling the Modulated Signal

This block can output an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor. It must be a positive integer. For more information, see “Upsampled Signals and Rate Changes” in *Communications Blockset User’s Guide*.

## Dialog Box



Function Block Parameters: CPM Modulator Baseband

CPM Modulator Baseband (mask) (link)

Output the complex envelope representation of the selected continuous phase modulation.

The input can be either bits or integers. In case of sample-based bit input, the input width must equal the number of bits per symbol. In case of frame-based bit input, the input width must be an integer multiple of the number of bits per symbol. The bits can be either binary-mapped or Gray-mapped into symbols.

For sample-based integer input, the input must be a scalar. For frame-based integer input, the input must be a column vector.

In case of frame-based input, the width of the output frame equals the product of the number of symbols and the Samples per symbol value.

In case of sample-based input, the output sample time equals the symbol period divided by the Samples per symbol value.

The Symbol prehistory parameter is the data symbol(s) used before the start of the simulation.

Parameters:

M-ary number:

Input type:

Symbol set ordering:

Modulation index:

Frequency pulse shape:

Pulse length (symbol intervals):

Symbol prehistory:

Phase offset (rad):

Samples per symbol:

Output data type:

OK Cancel Help Apply

### M-ary number

The size of the alphabet.

### Input type

Indicates whether the input consists of integers or groups of bits.

# CPM Modulator Baseband

---

## **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.

## **Modulation index**

The number of half-revolutions of phase shift due to the latest symbol when that symbol is the integer 1.

## **Frequency pulse shape**

The type of pulse shaping that the block uses to smooth the phase transitions of the modulated signal.

## **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.

## **Rolloff**

The rolloff factor of the raised cosine filter. This field appears only when **Frequency pulse shape** is set to Spectral Raised Cosine.

## **BT product**

The product of bandwidth and time. This field appears only when **Frequency pulse shape** is set to Gaussian.

## **Pulse length (symbol intervals)**

The length of the frequency pulse shape.

## **Symbol prehistory**

The data symbols used before the start of the simulation, in reverse chronological order.

## **Phase offset (rad)**

The initial phase of the output waveform.

## **Samples per symbol**

The number of output samples that the block produces for each integer or binary word in the input.

**Output data type**

This block supports double and single data types.

**Pair Block**

CPM Demodulator Baseband

**See Also**

CPFSK Modulator Baseband, GMSK Modulator Baseband, MSK Modulator Baseband

**References**

[1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. *Digital Phase Modulation*. New York: Plenum Press, 1986.

# CPM Phase Recovery

---

## Purpose

Recover carrier phase using 2P-Power method

## Library

Carrier Phase Recovery sublibrary of Synchronization

## Description



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,  $h = K / P$ , then  $P$  is the number to which the name "2P-Power" refers.

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.

## Input and Outputs

The input signal must be a frame-based column vector or a sample-based scalar 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.

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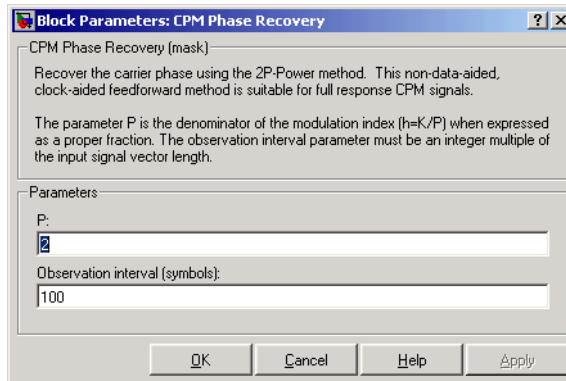
**Note** Because the block internally computes the argument of a complex number, the carrier phase estimate has an inherent ambiguity. The carrier phase estimate is between  $-90/P$  and  $90/P$  degrees and might differ from the actual carrier phase by an integer multiple of  $180/P$  degrees.

---

## 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 ( $h = K / P$ ) when expressed as a proper fraction.

### Observation interval

The number of symbols for which the carrier phase is assumed constant.

# CPM Phase Recovery

---

## Algorithm

If the symbols occurring during the observation interval are  $x(1)$ ,  $x(2)$ ,  $x(3)$ , ...,  $x(L)$ , then the resulting carrier phase estimate is

$$\frac{1}{2P} \arg \left\{ \sum_{k=1}^L (x(k))^{2P} \right\}$$

where the  $\arg$  function returns values between -180 degrees and 180 degrees.

## References

[1] Mengali, Umberto, and Aldo N. D'Andrea, *Synchronization Techniques for Digital Receivers*, New York, Plenum Press, 1997.

## See Also

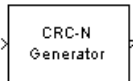
M-PSK Phase Recovery, CPM Modulator Baseband



**Purpose** Generate CRC bits according to CRC method and append to input data frames

**Library** CRC sublibrary of Error Detection and Correction

## Description



The CRC-N Generator block generates cyclic redundancy code (CRC) bits for each input data frame and appends them to the frame. 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.

CRC Method	Generator Polynomial	Number of Bits
CRC-32	$x^{32}+x^{26}+x^{23}+x^{22}+x^{16}+x^{12}+x^{11}+x^{10}+x^8+x^7+x^5+x^4+x^2+x+1$	32
CRC-24	$x^{24}+x^{23}+x^{14}+x^{12}+x^8+1$	24
CRC-16	$x^{16}+x^{15}+x^2+1$	16
Reversed CRC-16	$x^{16}+x^{14}+x+1$	16
CRC-8	$x^8+x^7+x^6+x^4+x^2+1$	8
CRC-4	$x^4+x^3+x^2+x+1$	4

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.

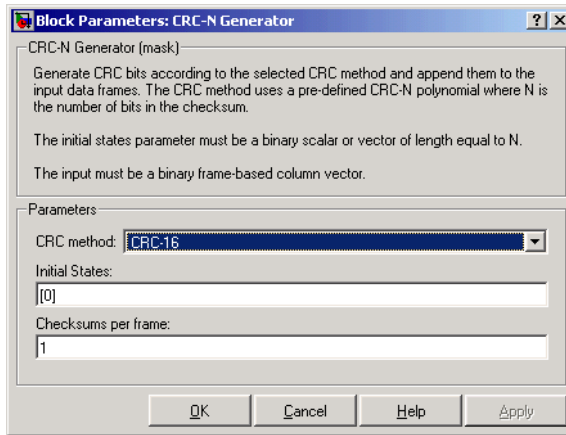
This block supports double and boolean data types. The output data type is inherited from the input.

# CRC-N Generator

## Signal Attributes

The General CRC Generator block has one input port and one output port. Both ports allow frame based binary column vectors only.

## Dialog Box



## CRC-N method

The generator polynomial for the CRC algorithm.

## 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.

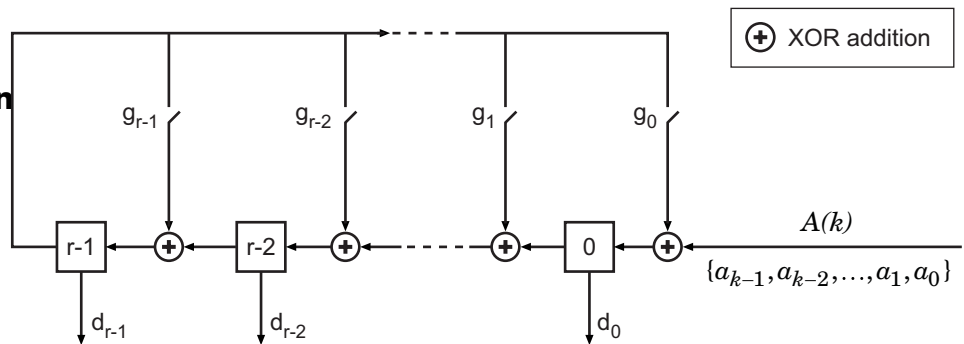
## Checksums per frame

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 Coding” in *Communications Blockset User’s Guide*.

## Schematic of the CRC Implementation



The above circuit divides the polynomial

$$a(x) = a_{k-1}x^{k-1} + a_{k-2}x^{k-2} + \dots + a_1x + a_0 \text{ by}$$

$$g(x) = g_{r-1}x^{r-1} + g_{r-2}x^{r-2} + \dots + g_1x + g_0, \text{ and returns the}$$

remainder  $d(x) = d_{r-1}x^{r-1} + d_{r-2}x^{r-2} + \dots + d_1x + d_0$ .

The input symbols  $\{a_{k-1}, a_{k-2}, \dots, a_2, a_1, a_0\}$  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.

## 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.

## Pair Block

CRC-N Syndrome Detector

# CRC-N Generator

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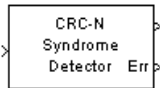
## **See Also**

General CRC Generator, General CRC Syndrome Detector

**Purpose** Detect errors in input data frames according to selected CRC method

**Library** CRC sublibrary of Error Detection and Correction

## Description



The CRC-N Syndrome Detector block computes checksums for its entire input frame. 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 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.

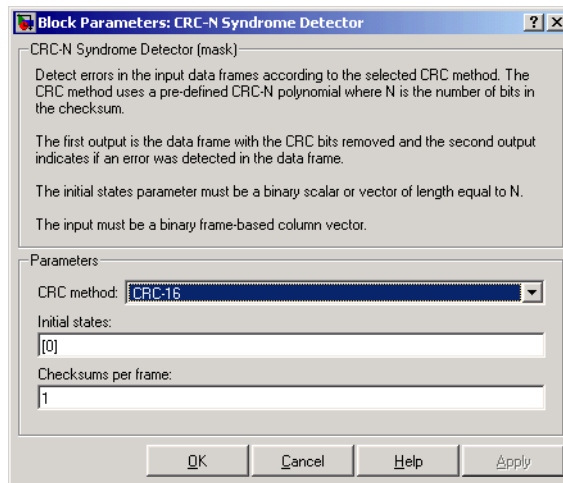
You specify the initial state of the internal shift register by the **Initial states** parameter. You specify the number of checksums that the block calculates for each input frame by the **Checksums per frame** parameter. For more detailed information, see the reference page for the General CRC Syndrome Detector block.

This block supports double and boolean data types. The output data type is inherited from the input.

# CRC-N Syndrome Detector

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## Dialog Box



### CRC-N method

The generator polynomial for the CRC algorithm.

### 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.

### 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 Coding” in *Communications Blockset 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.

**Pair Block** CRC-N Generator

**See Also** General CRC Generator, General CRC Syndrome Detector

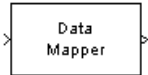
# Data Mapper

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**Purpose** Map integer symbols from one coding scheme to another

**Library** Utility Blocks

## 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.

The input can be either a scalar, a sample-based vector, or a frame-based column vector. The block can accept multichannel inputs and allows for input and output data types of double, single, int32, int16, int8, uint32, uint16, and uint8. If the input is double or single, then it must be non-negative in value. Note that although the block will provide outputs for non-integer valued inputs, the results will likely be meaningless.

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.



Binary to Gray Mode		Gray to Binary Mode	
Input	Output	Input	Output
0	0 (000)	0 (000)	0
1	1 (001)	1 (001)	1
2	3 (011)	2 (010)	3
3	2 (010)	3 (011)	2
4	6 (110)	4 (100)	7
5	7 (111)	5 (101)	6
6	5 (101)	6 (110)	4
7	4 (100)	7 (111)	5

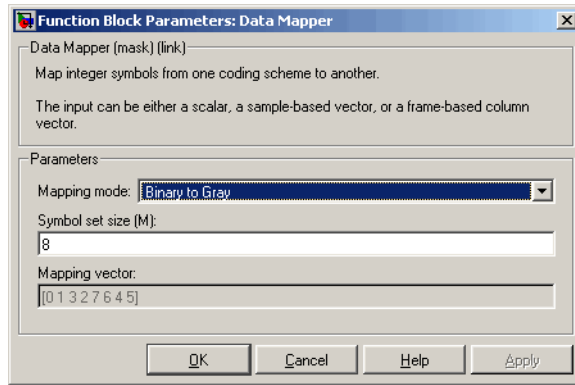
When you select the User Defined mode, you can use any arbitrary mapping by providing a vector to specify the output ordering. For example, the vector [1,5,0,4,2,3] defines the following mapping:

0 → 1  
1 → 5  
2 → 0  
3 → 4  
4 → 2  
5 → 3

When you select the Straight Through mode, the output equals the input.

# Data Mapper

## Dialog Box



### Mapping mode

The type of data mapping that the block performs.

### Symbol set size

Symbol set size of  $M$  restricts this block's inputs and outputs to integers in the range 0 to  $M-1$ .

### Mapping vector

A vector of length  $M$  that contains the integers from 0 to  $M-1$ . The order of the elements of this vector specifies the mapping of inputs to outputs. This field is active only when **Mapping mode** is set to User Defined.

# DBPSK Demodulator Baseband

## Purpose

Demodulate DBPSK-modulated data

## Library

PM, in Digital Baseband sublibrary of Modulation

## Description

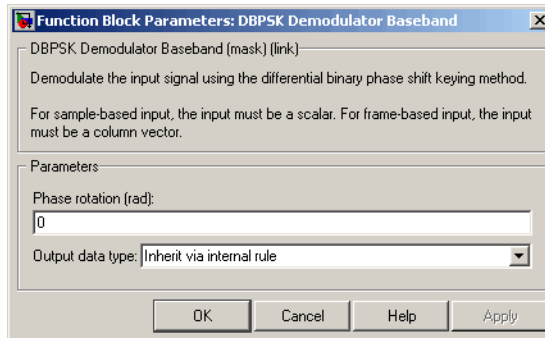


The DBPSK Demodulator Baseband block demodulates a signal that was modulated using the differential binary phase shift keying method. The input is a baseband representation of the modulated signal.

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  $\pi+\theta$ , 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.

The input can be either a scalar or a frame-based column vector. The block accepts input of data types `single` and `double`.

## Dialog Box



### Phase rotation (rad)

This phase difference between the current and previous modulated symbols 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

# DBPSK Demodulator Baseband

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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 both integer and bit inputs, this block can output the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, and `double`.

**Pair Block**      DBPSK Modulator Baseband

**See Also**      M-DPSK Demodulator Baseband, DQPSK Demodulator Baseband,  
BPSK Demodulator Baseband

**Purpose** Modulate using differential binary phase shift keying method

**Library** PM, in Digital Baseband sublibrary of Modulation

## Description

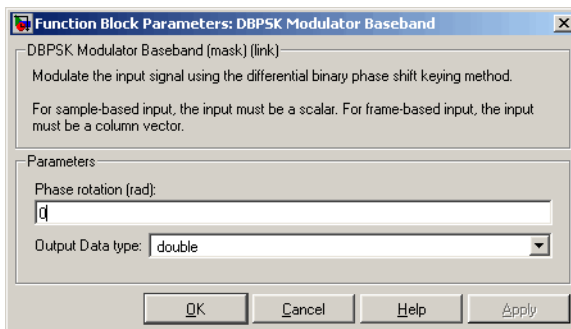


The DBPSK Modulator Baseband block modulates using the differential binary phase shift keying method. The output is a baseband representation of the modulated signal.

The input must be a discrete-time binary-valued signal. The input can be either a scalar or a frame-based column vector. For both integer and bit inputs, the block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, and `double`. These 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  $\exp(j\theta)$  or  $-\exp(j\theta)$ , respectively.
- If a successive input bit is 0 or 1, respectively, then the modulated symbol is the previous modulated symbol multiplied by  $\exp(j\theta)$  or  $-\exp(j\theta)$ , respectively.

## Dialog Box



### Phase rotation (rad)

The phase difference between the previous and current modulated symbols when the input is zero.

# DBPSK Modulator Baseband

---

## **Output Data type**

The output data type can be either single or double. By default, the block sets this to double.

**Pair Block** DBPSK Demodulator Baseband

**See Also** DQPSK Modulator Baseband, BPSK Modulator Baseband

**Purpose** Distribute elements of input vector alternately between two output vectors

**Library** Sequence Operations

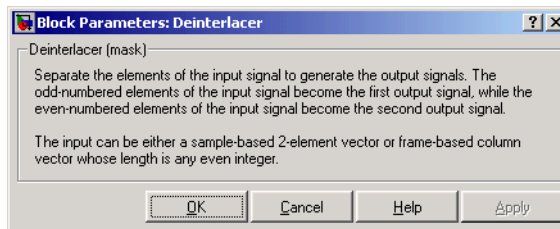
**Description** The Deinterlacer block accepts an input vector that has an even number of elements. The block alternately places the elements in each of two output vectors. As a result, each output vector size is half the input vector size. The output vectors have the same complexity and sample time of the input.



The input can be either a sample-based vector of length two, or a frame-based column vector whose length is any even integer. The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, `double`, and fixed-point. The data types of this output will be the same as that of the input signal.

This block can be useful for separating in-phase and quadrature information from a single vector into separate vectors.

## Dialog Box



**Examples** If the input vector is frame-based with value `[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 is the inverse of the example on the reference page for the Interlacer block.

If the input vector is frame-based with value `[1; 2; 3; 4; 5; 6]`, then the two output vectors are `[1; 3; 5]` and `[2; 4; 6]`.

**Pair Block** Interlacer

# Deinterlacer

---

## **See Also**

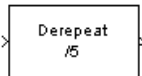
Demux (Simulink)



**Purpose** Reduce sampling rate by averaging consecutive samples

**Library** Sequence Operations

## Description



The Derepeat block resamples the discrete input at a rate  $1/N$  times the input sample rate by averaging  $N$  consecutive samples. This is one possible inverse of the Repeat block (Signal Processing Blockset). 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 input can have any shape or frame status. The block can accept the data types single and double. The data type of the output will be the same as that of the input signal.

This block will work within a triggered subsystem, as long as it is used in the single-rate mode.

### Sample-Based Operation

If the input is 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.

### Frame-Based Operation

If the input is frame-based, then 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.

# Derepeat

---

The **Framing** parameter determines how the block adjusts the rate at the output to accommodate the reduced number of samples. The two options are:

- Maintain input frame size

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.

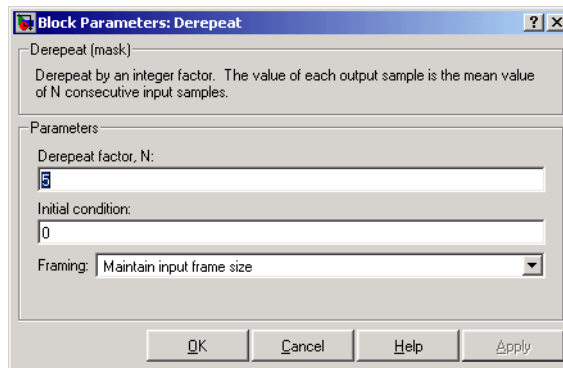
For example, if a single-channel input with a frame period of 1 second is derepeated by a factor of 4, then the output has a frame period of 4 seconds. The input and output frame sizes are equal.

- Maintain input frame rate

The block reduces the sampling rate by using a proportionally smaller frame *size* than the input. For derepetition by a factor of  $N$ , the output frame size is  $1/N$  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 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.

## Dialog Box



### Derepeat factor, N

The number of consecutive input samples to average in order to produce each output sample.

### Initial condition

The value with which to initialize the block.

### Framing

For frame-based operation, the method by which to reduce the amount of data. One method decreases the frame rate while maintaining frame size, while the other decreases the frame size while maintaining frame rate.

## See Also

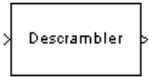
Repeat (Signal Processing Blockset), Downsample (Signal Processing Blockset)

# Descrambler

**Purpose** Descramble input signal

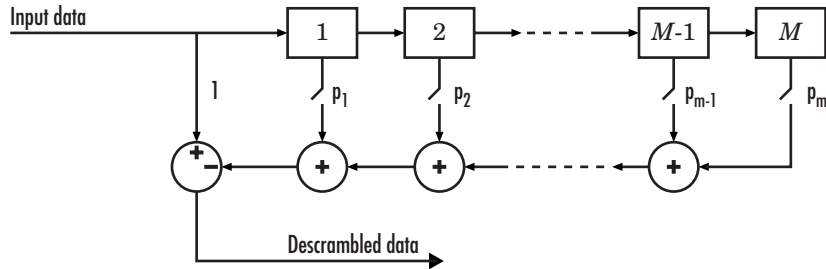
**Library** Sequence Operations

## Description

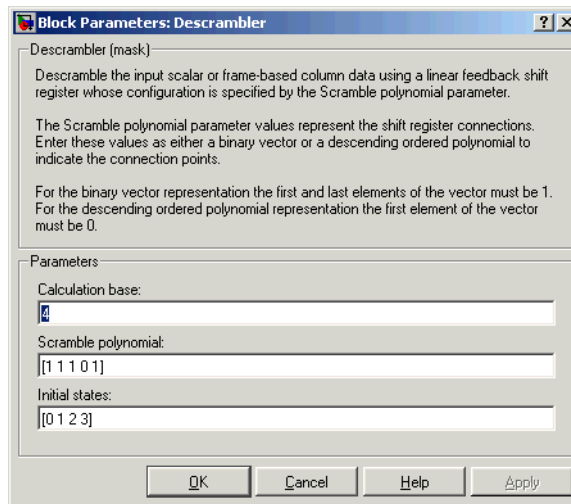


The Descrambler block descrambles the input signal, which must be a scalar or a frame-based column vector. The Descrambler block is the inverse of the Scrambler block. If you use the Scrambler block in the transmitter, then you should use the Descrambler block in the receiver.

Below is a schematic of the descrambler. All adders and the one subtractor operate modulo  $N$ , where  $N$  is the **Calculation base** parameter. The input values must be integers between 0 and  $N-1$ .



At each time step, the input causes the contents of the registers to shift sequentially. Each switch in the descrambler is on or off as defined by the **Scramble polynomial** parameter. 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.



## Dialog Box

### Calculation base

The calculation base  $N$ . The input and output of this block are integers in the range  $[0, N-1]$ .

### 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

Scrambler

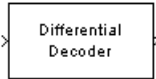
# Differential Decoder

---

**Purpose** Decode binary signal using differential coding

**Library** Source Coding

**Description** The Differential Decoder block decodes the binary input signal. The output is the logical difference between the present input and the previous input. More specifically, the block's input and output are related by



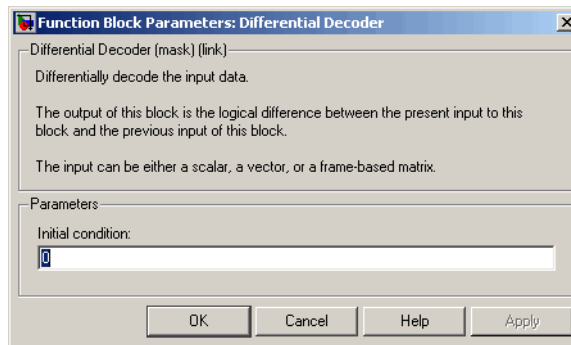
$$m(t_0) = d(t_0) \text{ XOR } \mathbf{Initial\ condition\ parameter\ value}$$

$$m(t_k) = d(t_k) \text{ XOR } d(t_{k-1})$$

where

- $d$  is the differentially encoded input.
- $m$  is the output message.
- $t_k$  is the  $k$ th time step.
- XOR is the logical exclusive-or operator.

The input can be either a scalar, a vector, or a frame-based matrix. This block processes each vector element independently.



## Dialog Box

### Initial condition

The logical exclusive-or of this value with the initial input value forms the initial output value.

## References

[1] Couch, Leon W., II, *Digital and Analog Communication Systems*, Sixth edition, Upper Saddle River, N. J., Prentice Hall, 2001.

## Pair Block

Differential Encoder

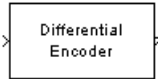
# Differential Encoder

---

**Purpose** Encode binary signal using differential coding

**Library** Source Coding

## Description



The Differential Encoder block encodes the binary input signal. The output is the logical difference between the present input and the previous output. More specifically, the input and output are related by

$$d(t_0) = m(t_0) \text{ XOR } \mathbf{Initial\ condition\ parameter\ value}$$

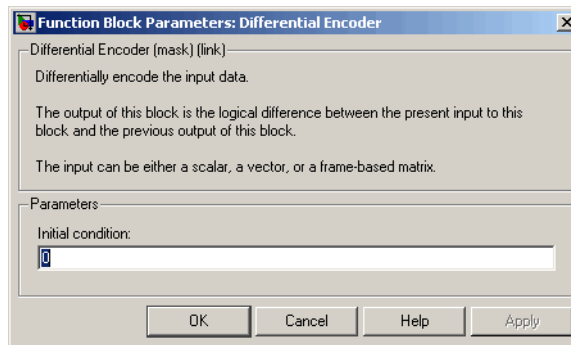
$$d(t_k) = d(t_{k-1}) \text{ XOR } m(t_k)$$

where

- $m$  is the input message
- $d$  is the differentially encoded output.
- $t_k$  is the  $k$ th time step.
- XOR is the logical exclusive-or operator.

The input can be either a scalar, a vector, or a frame-based matrix. This block processes each vector element independently.





## Dialog Box

### Initial condition

The logical exclusive-or of this value with the initial input value forms the initial output value.

## References

[1] Couch, Leon W., II, *Digital and Analog Communication Systems*, Sixth edition, Upper Saddle River, N. J., Prentice Hall, 2001.

## Pair Block

Differential Decoder

# Discrete-Time Eye Diagram Scope

**Purpose** Display multiple traces of modulated signal

**Library** Comm Sinks




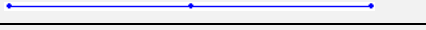

**Description** The Discrete-Time Eye Diagram Scope block displays multiple traces of a modulated signal to produce an eye diagram. You can use the block to reveal the modulation characteristics of the signal, such as pulse shaping or channel distortions.



The Discrete-Time Eye Diagram Scope block has one input port. The block accepts signal of type `double`, `single`, `boolean`, `base integer`, and `fixed-point data` types for input, but will cast it as `double`. The input signal must be a sample-based scalar in sample-based mode. The input must be a frame-based column vector or a scalar in frame-based mode.

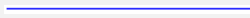



## Marker and Line Styles

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.


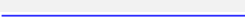
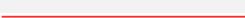
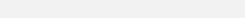
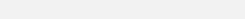
Marker Style	Parameter Symbol	Appearance
Plus	+	
Circle	o	
Asterisk	*	
Point	.	
Cross	x	

The **Line style** parameter specifies the style for lines in the eye diagram. The following lists some of the available line styles.

# Discrete-Time Eye Diagram Scope

Line Style	Appearance
Solid	
Dashed	
Dotted	
Dash-dot	

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).

Color	RGB Equivalent	Appearance
Black	(0,0,0)	
Blue	(0,0,255)	
Red	(255,0,0)	
Green	(0,255,0)	
Dark purple	(192,0,192)	

See the `line` function in the MATLAB documentation for more information about the available markers, colors, and line styles.

## Recommended Settings

The following table summarizes the recommended parameter settings for the Discrete-Time Eye Diagram Scope.

# Discrete-Time Eye Diagram Scope

Parameter	Recommended Setting
Samples per symbol	Same as the <b>Samples per symbol</b> setting in the modulator block, or the <b>Interpolation factor</b> setting in the interpolation block
Offset (samples)	0 to view the open part of the eye ( <b>Samples per symbol</b> )/2 to view the closed part of the eye
Symbols per trace	An integer between 1 and 4
Traces displayed	10 times the alphabet size of the modulator, M
New traces per display	Same as <b>Traces displayed</b> for greater speed A small positive integer for best animation
Marker	None or a point (.) to see where the samples are plotted
Line style	Solid dash (-)
Line color	Blue (b)
Duplicate points at trace boundary	Check <b>Duplicate points at trace boundary</b> for modulations such as PSK and QAM.  Clear to display the phase trees for MSK, CPFSK, GFSK, GMSK, and other continuous phase modulations.
Color fading	Check <b>Color fading</b> for animation that resembles an oscilloscope.  Clear for greater speed and animation that resembles a plot.

# Discrete-Time Eye Diagram Scope

Parameter	Recommended Setting
High quality rendering	Check <b>High quality rendering</b> for better animation.  Clear for greater speed.
Eye diagram to display	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.
Open at start of simulation	Check <b>Open at start of simulation</b> to view the signal at the start of simulation.  Clear to view the signal after convergence to steady state and for greater initial speed.
Y-axis minimum	Approximately 10% less than the expected minimum value of the signal
Y-axis maximum	Approximately 10% greater than the expected maximum value of the signal

## 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.

# Discrete-Time Eye Diagram Scope

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In addition to the standard MATLAB figure window menus (**File**, **Edit**, **Window**, **Help**), the Vector Scope window has an **Axes** and a **Channels** menu.

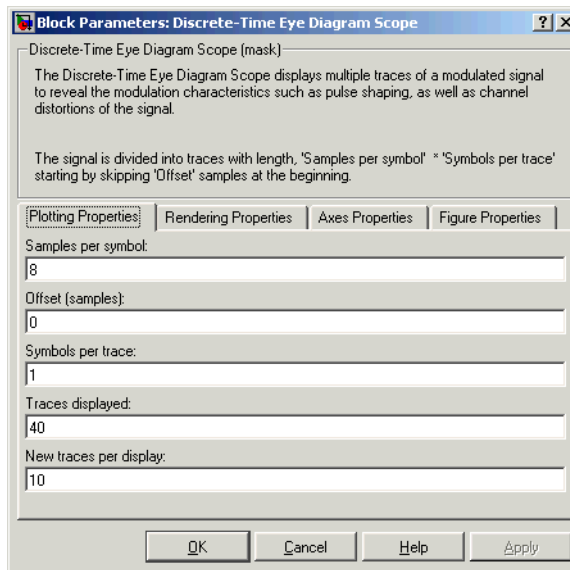
The properties listed in the **Axes** menu apply to all channels. Many of the parameters in this menu are also accessible through the block parameter dialog box. These are **Autoscale**, **Show grid**, **Frame #**, and **Save Position**. Below are descriptions of the other parameters listed in the **Axes** 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.

## Dialog Box



### **Samples per symbol**

Number of samples per symbol. Use with **Symbols per trace** to determine the number of samples per trace.

### **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.

### **Symbols per trace**

Positive integer specifying the number of symbols plotted per trace.

### **Traces displayed**

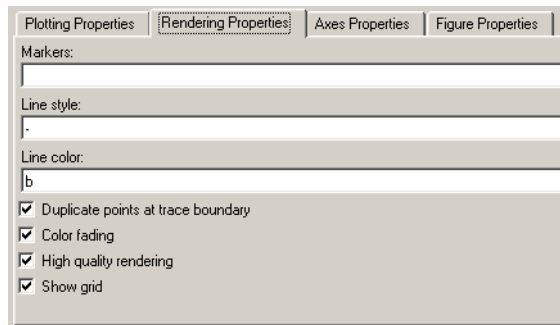
Number of traces plotted.

### **New traces per display**

Positive integer less than **Traces displayed**, specifying the number of new traces that appear in each display.

# Discrete-Time Eye Diagram Scope

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## Markers

The marker for points in the eye diagram. Tunable.

## Line style

The line style in the eye diagram. Tunable.

## Line color

The line color in the eye diagram. Tunable.

## Duplicate points at trace boundary

Check to enable duplicate points at the trace boundary. Clear to disable.

## Color fading

When selected, the points in the eye diagram fade as the interval of time after they are first plotted increases. Tunable.

## 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.

## Show grid

Toggles the scope grid on and off. Tunable.



# Discrete-Time Eye Diagram Scope

The screenshot shows the 'Axes Properties' tab of the scope's configuration window. It contains several input fields for defining the axes and labels:

- Y axis minimum: -1.5
- Y axis maximum: 1.5
- In-phase Y-axis label: In-phase Amplitude
- Quadrature Y-axis label: Quadrature Amplitude

## Y-axis minimum

Minimum signal value the scope displays. Tunable.

## Y-axis maximum

Maximum signal value the scope displays. Tunable.

## In-phase Y-axis label

Label for  $y$ -axis of the in-phase diagram. Tunable.

## Quadrature Y-axis label

Label for  $y$ -axis of the quadrature diagram. Tunable.

The screenshot shows the 'Figure Properties' tab of the scope's configuration window. It contains several options and input fields for defining the figure's appearance and behavior:

- Open scope at start of simulation
- Eye diagram to display: In-phase and Quadrature
- Trace number
- Scope position: get(0,'defaultfigureposition');
- Title: Eye Diagram

## 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.

# Discrete-Time Eye Diagram Scope

---

**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.

**Trace number**

Displays the number of the current trace in the input sequenced. Tunable.

**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.

**Title**

Title of eye diagram figure window. Tunable.

**Examples**

For documentation examples that use this block, see “Example: Viewing a Sinusoid” and “Example: Viewing a Modulated Signal”.

Also, the following Communications Blockset 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

## Purpose

Display in-phase and quadrature components of modulated signal constellation

## Library

Comm Sinks

## Description



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.

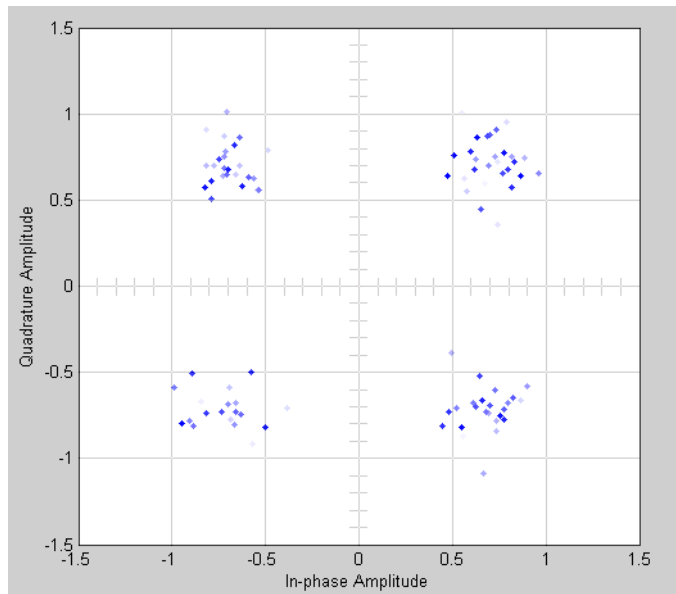
The Discrete-Time Scatter Plot Scope block has one input port. The input signal must be complex. The input signal must be complex. The block accepts signal of type `double`, `single`, base integer, and fixed-point for input, but will cast it as `double`. The input signal must be a sample-based scalar in sample-based mode. The input must be a frame-based column vector or a scalar in frame-based mode.

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.

# Discrete-Time Scatter Plot Scope

---



## 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.

## Recommended Settings

The following table summarizes the recommended parameter settings for the Discrete-Time Scatter Plot Scope.

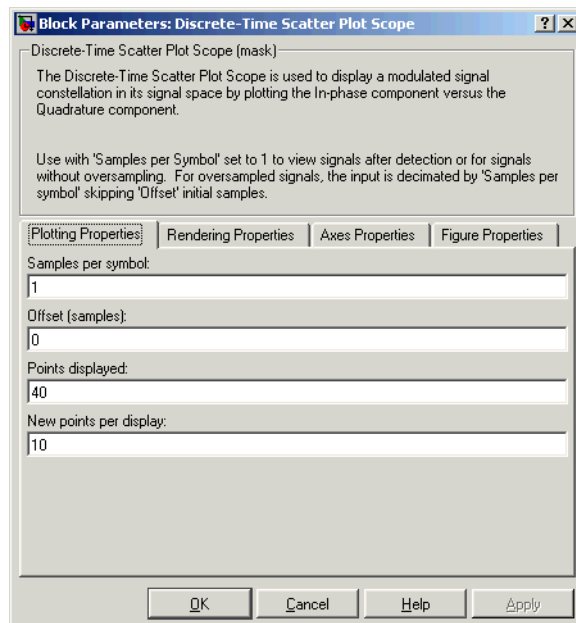
## Discrete-Time Scatter Plot Scope

Parameter	Recommended Setting
<b>Samples per symbol</b>	Same as the <b>Samples per symbol</b> setting in the modulator block, or the <b>Interpolation factor</b> setting in the interpolation block
<b>Points displayed</b>	10 times the alphabet size of the modulator
<b>New points per display</b>	Same as <b>Points displayed</b> for greater speed A small positive integer for best animation
<b>Line style</b>	Solid dash (-)
<b>Line color</b>	Blue (b)
<b>Color fading</b>	Check <b>Color fading</b> for animation that resembles an oscilloscope. Clear for greater speed and animation that resembles a plot.
<b>High quality rendering</b>	Check <b>High quality rendering</b> for higher quality rendering. Clear for greater speed.
<b>Open at start of simulation</b>	Check <b>Open at start of simulation</b> to view the signal at the start of simulation. Clear to view the signal after convergence to steady state and for greater initial speed.

# Discrete-Time Scatter Plot Scope

Parameter	Recommended Setting
X-axis minimum	Approximately 10% less than the expected minimum value of the signal
X-axis maximum	Approximately 10% greater than the expected maximum value of the signal

## Dialog Box



### Samples per symbol

Number of samples per symbol.

### Offset (samples)

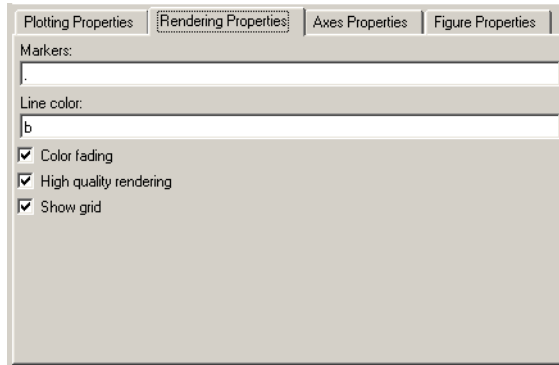
Nonnegative integer less than the number of samples per symbol, specifying the number of samples to skip before plotting points.

## Points displayed

Total number of points plotted.

## New points per display

Number of new points that appear in each display.



## Markers

Line markers used in the scatter plot. Tunable.

## Line color

The line color used in the scatter plot. Tunable.

## Color fading

When selected, the points in the scatter plot fade as the interval of time after they are first plotted increases. Tunable.

## High quality rendering

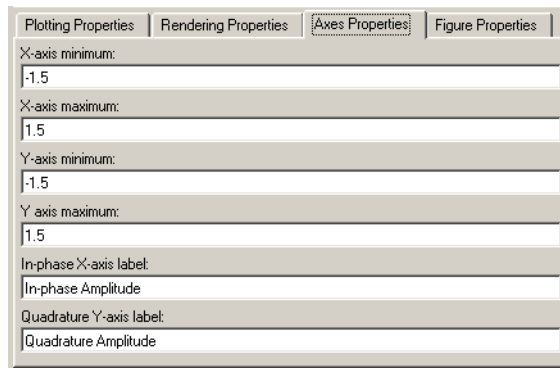
When selected, the block renders a slow, higher-quality picture with overwrite raster operations. When cleared, the block renders a fast, lower-quality picture with XOR raster operations. Tunable.

## Show grid

Toggles the scope grid on and off. Tunable.

# Discrete-Time Scatter Plot Scope

---



## **X-axis minimum**

Minimum value the scope displays on the  $x$ -axis. Tunable.

## **X-axis maximum**

Maximum value the scope displays on the  $x$ -axis. Tunable.

## **Y-axis minimum**

Minimum signal value the scope displays on the  $y$ -axis. Tunable.

## **Y-axis maximum**

Maximum signal value the scope displays on the  $y$ -axis. Tunable.

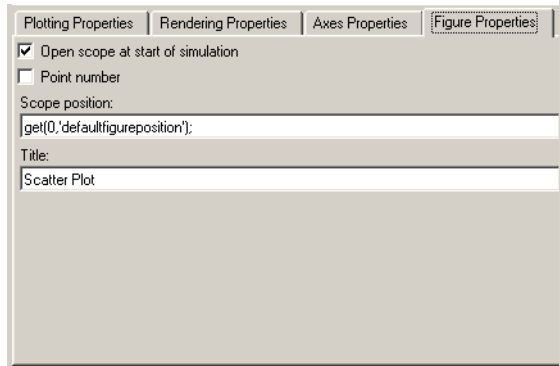
## **In-phase X-axis label**

Label for  $x$ -axis. Tunable.

## **Quadrature Y-axis label**

Label for  $y$ -axis. Tunable.





## 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.

## Point number

Displays the number of the current point in the input sequence.

Tunable.

## 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.

## Title

Title of scatter plot. Tunable.

## Examples

For documentation examples that use this block, see “Example: Viewing a Sinusoid” and “Example: Viewing a Modulated Signal”.

The following demos in Communications Blockset illustrate how to use the Discrete-Time Scatter Plot Scope block:

- Digital Video Broadcasting Model
- DS Spread Spectrum Example

# Discrete-Time Scatter Plot Scope

---

- HiperLAN/2
- Phase Noise Effects in 256 QAM
- Rayleigh Fading Channel

## **See Also**

Discrete-Time Eye Diagram Scope, Discrete-Time Signal Trajectory Scope, Real-Imag to Complex

# Discrete-Time Signal Trajectory Scope

## Purpose

Plot modulated signal's in-phase component versus its quadrature component

## Library

Comm Sinks

## Description



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. The input signal must be complex. The block accepts signal of type `double`, `single`, `base integer`, and `fixed-point` for input, but will cast it as `double`. The input signal must be a sample-based scalar in sample-based mode. The input must be a frame-based column vector or a scalar in frame-based mode.

### 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.

### Recommended Settings

The following table summarizes the recommended parameter settings for the Discrete-Time Signal Trajectory Scope.

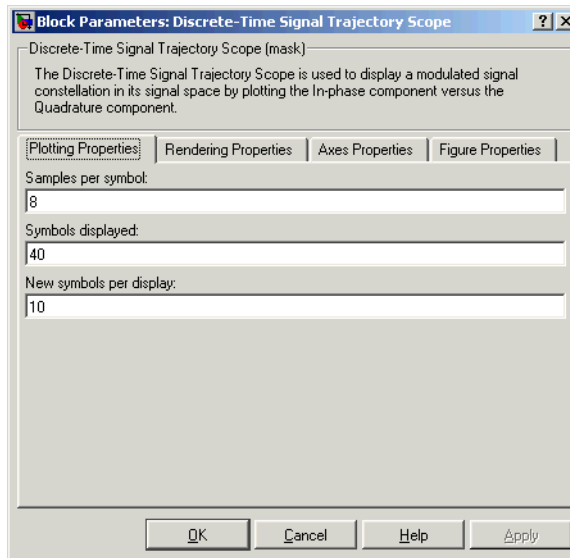
Parameter	Recommended Setting
<b>Samples per symbol</b>	Same as the <b>Samples per symbol</b> setting in the modulator block, or the <b>Interpolation factor</b> used in the interpolation block
<b>Symbols displayed</b>	10 times the alphabet size of the modulator, $M$

# Discrete-Time Signal Trajectory Scope

Parameter	Recommended Setting
New symbols per display	Same as <b>Symbols displayed</b> for greater speed  A small positive integer for best animation
Line style	Solid dash (-)
Line color	Blue (b)
Color fading	Check <b>Color fading</b> for animation that resembles an oscilloscope.  Clear for greater speed and animation that resembles a plot.
High quality rendering	Check <b>High quality rendering</b> for higher quality rendering.  Clear for greater speed.
Open at start of simulation	Check <b>Open at start of simulation</b> to view the signal at the start of simulation.  Clear to view the signal after convergence to steady state and for greater initial speed.
Y-axis minimum	Approximately 10% less than the expected minimum value of the signal
Y-axis maximum	Approximately 10% greater than the expected maximum value of the signal

# Discrete-Time Signal Trajectory Scope

## Dialog Box



### **Samples per symbol**

Number of samples per symbol.

### **Symbols displayed**

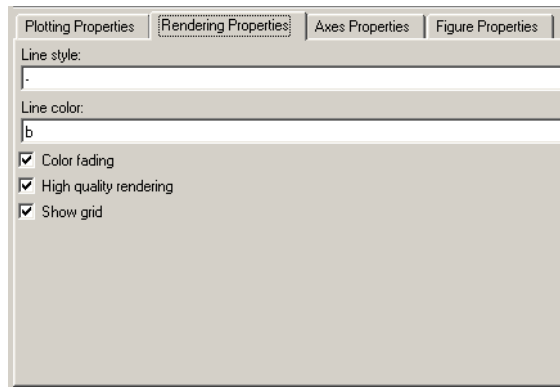
Total number of symbols plotted.

### **New symbols per display**

Number of new symbols that appear in each display.

# Discrete-Time Signal Trajectory Scope

---



## **Line markers**

The line markers used in the signal trajectory. Tunable.

## **Line color**

The line color used in the signal trajectory. Tunable.

## **Color fading**

When selected, the points in the signal trajectory fade as the interval of time after they are first plotted increases. Tunable.

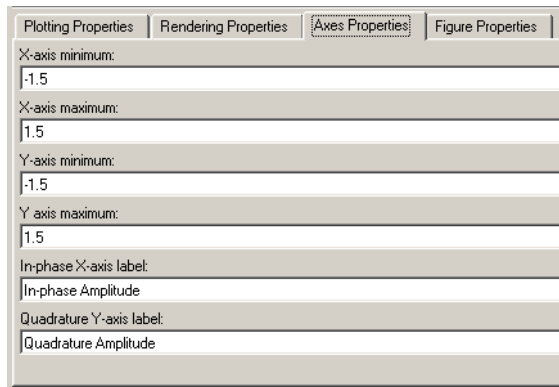
## **High quality rendering**

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.

## **Show grid**

Toggles the scope grid on and off. Tunable.

# Discrete-Time Signal Trajectory Scope



Property	Value
X-axis minimum:	-1.5
X-axis maximum:	1.5
Y-axis minimum:	-1.5
Y-axis maximum:	1.5
In-phase X-axis label:	In-phase Amplitude
Quadrature Y-axis label:	Quadrature Amplitude

## **X-axis minimum**

Minimum value the scope displays on the  $x$ -axis. Tunable.

## **X-axis maximum**

Maximum value the scope displays on the  $x$ -axis. Tunable.

## **Y-axis minimum**

Minimum signal value the scope displays on the  $y$ -axis. Tunable.

## **Y-axis maximum**

Maximum signal value the scope display on the  $y$ -axis. Tunable.

## **In-phase X-axis label**

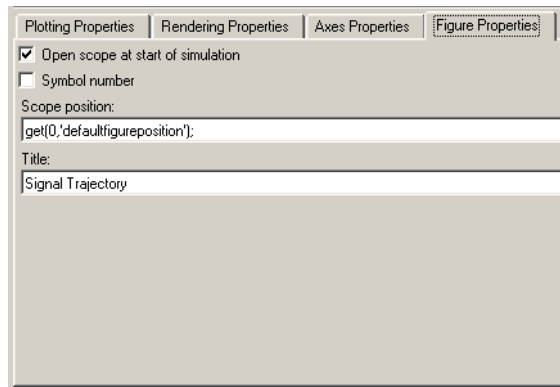
Label for  $x$ -axis. Tunable.

## **Quadrature Y-axis label**

Label for  $y$ -axis. Tunable.

# Discrete-Time Signal Trajectory Scope

---



## 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

## Symbol number

Displays the number of the current symbol in the input sequence. Tunable.

## 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.

## Title

Title of signal trajectory plot. Tunable.

## Examples

For documentation examples that use this block, see “Example: Viewing a Sinusoid” and “Example: Viewing a Modulated Signal”.

Also, the following demos in Communications Blockset illustrate how to use the Discrete-Time Signal Trajectory Scope:

- Filtered Offset QPSK vs. Filtered QPSK
- GMSK vs. MSK



# Discrete-Time Signal Trajectory Scope

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## **See Also**

Discrete-Time Eye Diagram Scope, Discrete-Time Scatter Plot Scope

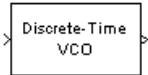
# Discrete-Time VCO

---

**Purpose** Implement voltage-controlled oscillator in discrete time

**Library** Components sublibrary of Synchronization

**Description**



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

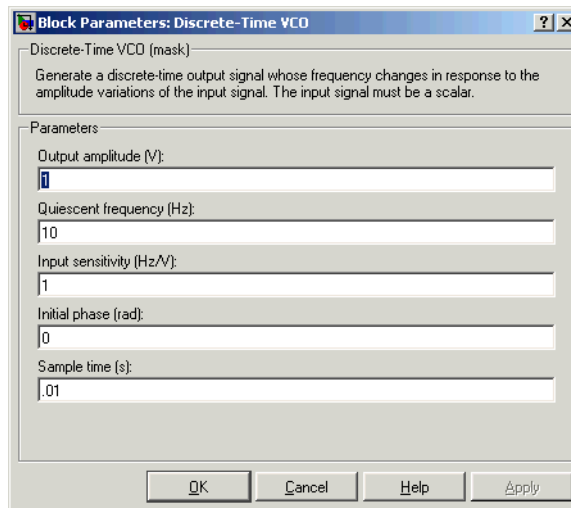
$$y(t) = A_c \cos\left(2\pi f_c t + 2\pi k_c \int_0^t u(\tau) d\tau + \varphi\right)$$

where  $A_c$  is the **Output amplitude**,  $f_c$  is the **Quiescent frequency**,  $k_c$  is the **Input sensitivity**, and  $\varphi$  is the **Initial phase**

This block uses a discrete-time integrator to interpret the equation above.

The input and output signals can be scalars of data type `single` or `double`. The data type of the output will be the same as that of the input signal.

## Dialog Box



### Output amplitude

The amplitude of the output.

### Quiescent frequency (Hz)

The frequency of the oscillator output when the input signal is zero.

### 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.

### Initial phase (rad)

The initial phase of the oscillator in radians.

### Sample time

The calculation sample time.

## See Also

Continuous-Time VCO

# DQPSK Demodulator Baseband

---

**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 the **Output type** parameter is set to Bit) in the block's output is the initial condition of zero because there is no previous symbol.

The input can be either a scalar or a frame-based column vector. The block accepts the input data types single and double.

## Outputs and Constellation Types

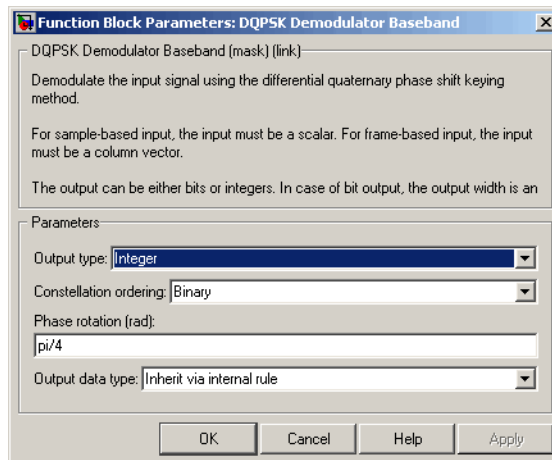
If the **Output type** parameter is set to Integer, then the block maps a phase difference of

$$\theta + \pi m/2$$

to  $m$ , where  $\theta$  is the **Phase rotation** parameter and  $m$  is 0, 1, 2, or 3.

If the **Output type** parameter is set 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.

## Dialog Box



### Output type

Determines whether the output consists of integers or pairs of bits.

### Constellation ordering

Determines how the block maps each integer to a pair of output bits. This field is active only when **Output type** is set to Bit.

### Phase rotation (rad)

This phase difference between the current and previous modulated symbols 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.

## Pair Block

DQPSK Modulator Baseband

# DQPSK Demodulator Baseband

---

## **See Also**

M-DPSK Demodulator Baseband, DBPSK Demodulator Baseband,  
QPSK Demodulator Baseband

**Purpose** Modulate using differential quaternary phase shift keying method

**Library** PM, in Digital Baseband sublibrary of Modulation

## Description



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 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`.

## Inputs and Constellation Types

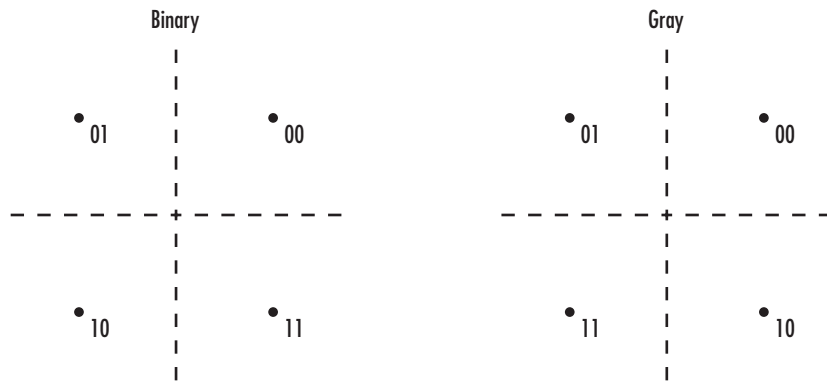
If the **Input type** parameter is set to `Integer`, then valid input values are 0, 1, 2, and 3. In this case, the input can be either a scalar or a frame-based column vector. If the first input is  $m$ , then the modulated symbol is

$$\exp(j\theta + j\pi m/2)$$

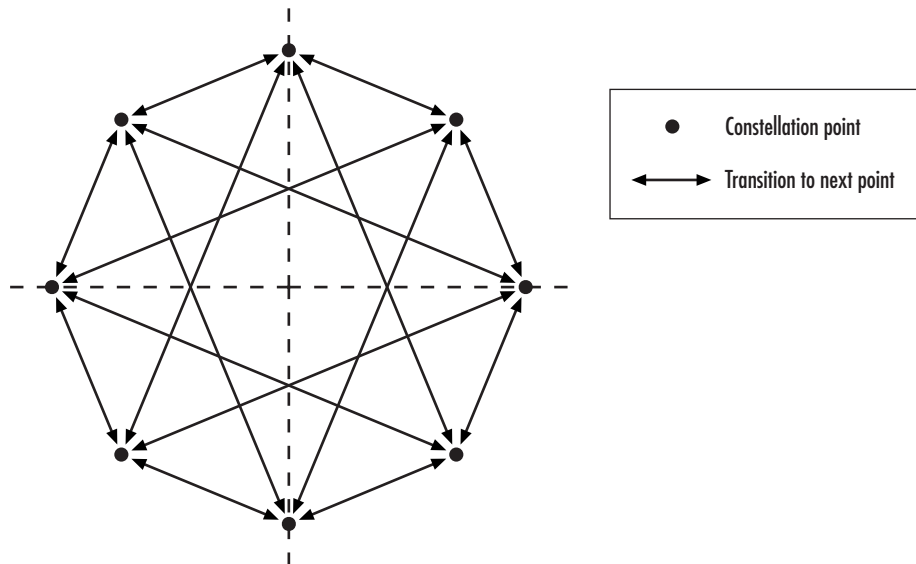
where  $\theta$  is the **Phase rotation** parameter. If a successive input is  $m$ , then the modulated symbol is the previous modulated symbol multiplied by  $\exp(j\theta + j\pi m/2)$ .

If the **Input type** parameter is set to `Bit`, then the input contains pairs of binary values. The input can be either a vector of length two or a frame-based column vector whose length is an even integer. The figure below shows the complex numbers by which the block multiplies the previous symbol to compute the current symbol, depending on whether the **Constellation ordering** parameter is set to `Binary` or `Gray`. The figure assumes that the **Phase rotation** parameter is set to  $\pi/4$ ; in other cases, the two schematics would be rotated accordingly.

# DQPSK Modulator Baseband



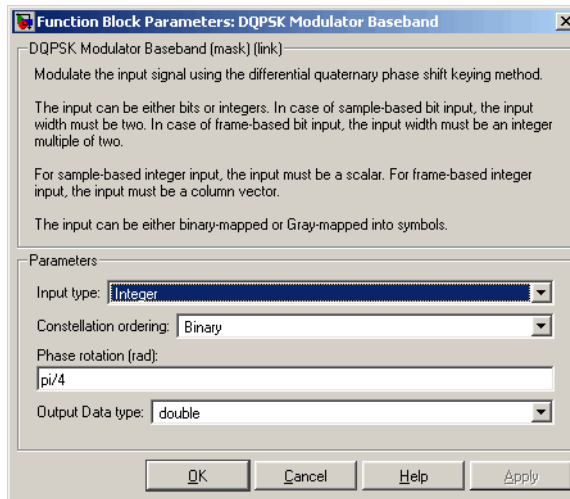
The figure below shows the signal constellation for the DQPSK modulation method when the **Phase rotation** parameter is  $\pi/4$ . The arrows indicate the four possible transitions from each symbol to the next symbol. The Binary and Gray options determine which transition is associated with each pair of input values.





More generally, if the **Phase rotation** parameter has the form  $\pi/k$  for some integer  $k$ , then the signal constellation has  $2k$  points.

## Dialog Box



### Input type

Indicates whether the input consists of integers or pairs of bits.

### Constellation ordering

Determines how the block maps each pair of input bits to a corresponding integer. This field is active only when **Input type** is set to Bit.

### 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.

## Pair Block

DQPSK Demodulator Baseband

# DQPSK Modulator Baseband

---

## **See Also**

M-DPSK Modulator Baseband, DBPSK Modulator Baseband, QPSK Modulator Baseband

**Purpose** Demodulate DSB-AM-modulated data

**Library** Analog Passband Modulation, in Modulation

## Description



The DSB AM Demodulator Passband block demodulates a signal that was 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 sample-based scalar signals.

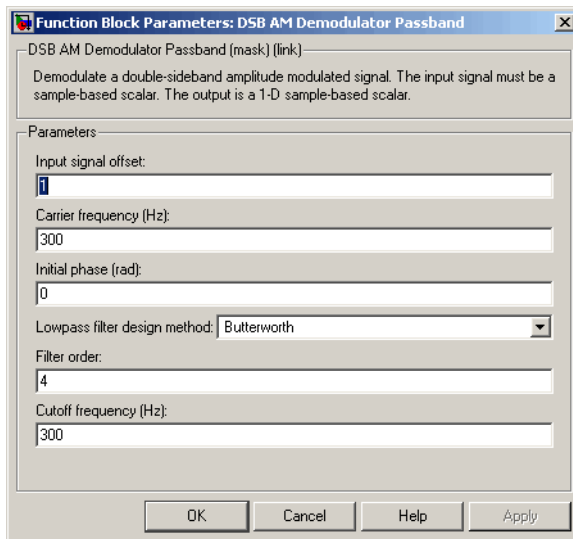
In the course of demodulating, this block uses a filter whose order, coefficients, passband ripple and stopband ripple are described by their respective lowpass filter parameters.

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 is not suited to be placed inside a triggered subsystem.

# DSB AM Demodulator Passband

## Dialog Block



### Input signal offset

The same as the **Input signal offset** parameter in the corresponding DSB AM Modulator Passband block.

### Carrier frequency (Hz)

The frequency of the carrier in the corresponding DSB AM Modulator Passband block.

### Initial phase (rad)

The initial phase of the carrier in radians.

### 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 .

# DSB AM Demodulator Passband

---

**Cutoff frequency (Hz)**

The cutoff frequency of the lowpass digital filter specified in the **Lowpass filter design method** field in Hertz.

**Passband ripple (dB)**

Applies to Chebyshev type I and Elliptic filters only. This is peak-to-peak ripple in the passband in dB.

**Stopband ripple (dB)**

Applies to Chebyshev type II and Elliptic filters only. This is the peak-to-peak ripple in the stopband in dB.

**Pair Block**

DSB AM Modulator Passband

# 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 using double-sideband amplitude modulation. The output is a passband representation of the modulated signal. Both the input and output signals are real sample-based scalar signals.

If the input is  $u(t)$  as a function of time  $t$ , then the output is

$$(u(t) + k) \cos(2\pi f_c t + \theta)$$

where:

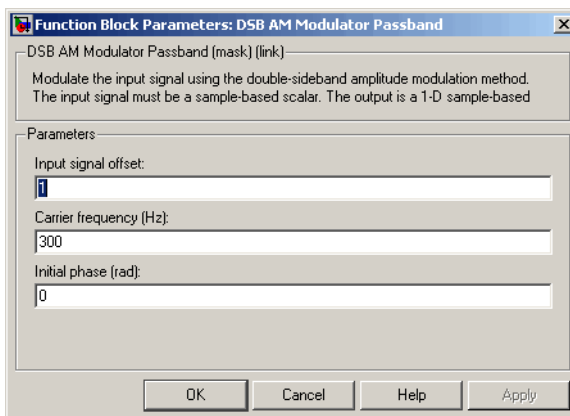
- $k$  is the **Input signal offset** parameter.
- $f_c$  is the **Carrier frequency** parameter.
- $\theta$  is the **Initial phase** parameter.

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 is not suited to be placed inside a triggered subsystem.

## Dialog Box



### 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.

### Carrier frequency (Hz)

The frequency of the carrier.

### Initial phase (rad)

The initial phase of the carrier.

## Pair Block

DSB AM Demodulator Passband

## See Also

DSBSC AM Modulator Passband, SSB AM Modulator Passband

# DSBSC AM Demodulator Passband

---

**Purpose** Demodulate DSBSC-AM-modulated data

**Library** Analog Passband Modulation, in Modulation

## Description



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 sample-based scalar signals.

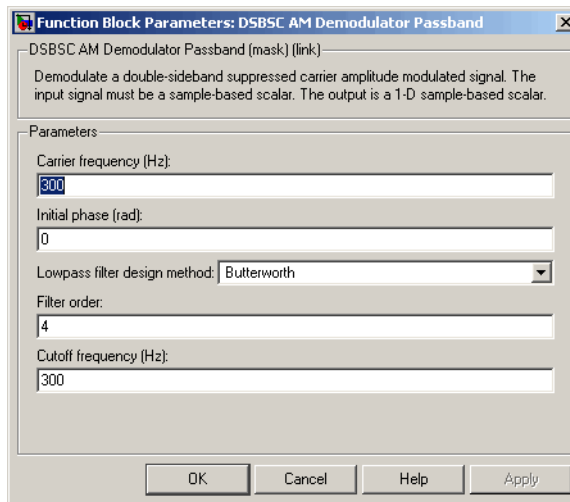
In the course of demodulating, this block uses a filter whose order, coefficients, passband ripple and stopband ripple are described by the their respective lowpass filter parameters.

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 is not suited to be placed inside a triggered subsystem.



## Dialog Box



### Carrier frequency (Hz)

The carrier frequency in the corresponding DSBSC AM Modulator Passband block.

### Initial phase (rad)

The initial phase of the carrier in radians.

### 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.

### Passband Ripple (dB)

Applies to Chebyshev type I and Elliptic filters only. This is peak-to-peak ripple in the passband in dB.

# DSBSC AM Demodulator Passband

---

## **Stopband Ripple (dB)**

Applies to Chebyshev type II and Elliptic filters only. This is the peak-to-peak ripple in the stopband in dB.

## **Pair Block**

DSBSC AM Modulator Passband

## **See Also**

DSB AM Demodulator Passband, SSB AM Demodulator Passband

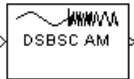
## Purpose

Modulate using double-sideband suppressed-carrier amplitude modulation

## Library

Analog Passband Modulation, in Modulation

## Description



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 sample-based scalar signals.

If the input is  $u(t)$  as a function of time  $t$ , then the output is

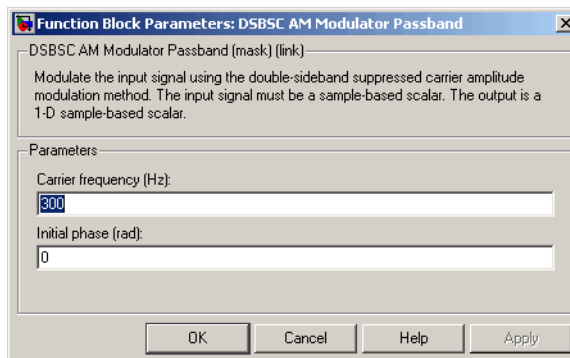
$$u(t)\cos(2\pi f_c t + \theta)$$

where  $f_c$  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.

This block works only with real inputs of type `double`. This block is not suited to be placed inside a triggered subsystem.

## Dialog Box



# DSBSC AM Modulator Passband

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**Carrier frequency (Hz)**

The frequency of the carrier.

**Initial phase (rad)**

The initial phase of the carrier in radians.

**Pair Block**

DSBSC AM Demodulator Passband

**See Also**

DSB AM Modulator Passband, SSB AM Modulator Passband

# Early-Late Gate Timing Recovery

## Purpose

Recover symbol timing phase using early-late gate method

## Library

Timing Phase Recovery sublibrary of Synchronization

## Description



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.

## 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. For best results, the input signal power should be normalized. The input must be a scalar or a frame-based column vector. The input uses  $N$  samples to represent each symbol, where  $N > 1$  is the **Samples per symbol** parameter. If the input is frame-based, then its vector length is  $N \cdot R$ , where  $R$  is a positive integer that indicates the number of symbols per frame. If the input is sample-based, then its sample time is  $1/N$  times the underlying symbol period.

If the **Reset** parameter is set 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. The sample time of the Rst input equals the symbol period if the input signal is sample-based, and the frame period if the input signal is frame-based.

Typically, **Samples per symbol** is at least 4 and the input signal is shaped using a raised cosine filter.

## 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:

# Early-Late Gate Timing Recovery

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- If the input signal is a frame-based column vector of length  $N \times R$ , then the Sym output is a frame-based column vector of length  $R$  having the same frame period.
- If the input signal is a sample-based scalar with sample time  $T/N$ , then the Sym output is a sample-based scalar with sample time  $T$ .
- The Ph output gives the phase estimate for each symbol in the input signal.

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 or frame period of the Ph output is the same as that of the Sym output.

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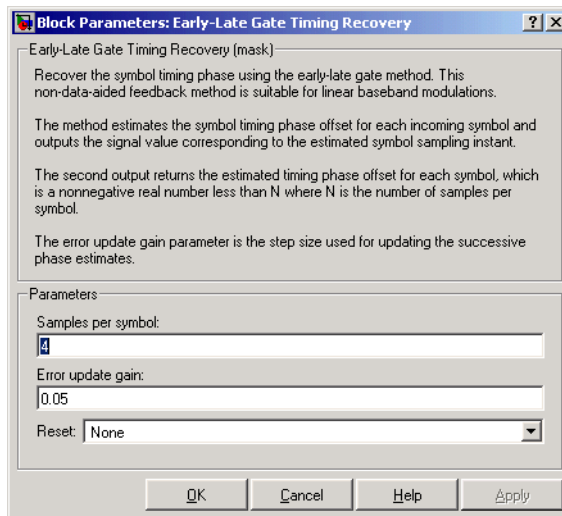
**Note** If the Ph output is very close to either zero or **Samples per symbol**, or if the actual timing phase offset in your input signal is very close to zero, then the block's accuracy might be compromised by small amounts of noise or jitter. The block works well when the timing phase offset is significant rather than very close to zero.

---

## Delays

This block incurs a delay of two symbols when the input is frame-based and three symbols when the input is sample-based.

## 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  $1/N$ , which corresponds to a slowly varying phase.

### 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 uses a timing error detector whose result for the  $k$ th symbol is  $e(k)$ , given by

# Early-Late Gate Timing Recovery

---

$$e(k) = a_I(k) + a_Q(k)$$

$$a_I(k) = y_I(kT + d_k) \{y_I(kT + T/2 + d_k) - y_I(kT - T/2 + d_{k-1})\}$$

$$a_Q(k) = y_Q(kT + d_k) \{y_Q(kT + T/2 + d_k) - y_Q(kT - T/2 + d_{k-1})\}$$

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  $k$ th 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 Blockset User's Guide*.

## 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



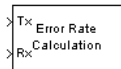
## Purpose

Compute bit error rate or symbol error rate of input data

## Library

Comm Sinks

## Description



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.

You can 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.

This block inherits the sample time of its inputs.

---

**Note** This block does not generate C code when the block parameter **Output data** is set to Workspace. Consider changing **Output data** to Port instead.

---

## 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 inputs can be either scalars or frame-based column vectors of fixed-point data type, or int8, uint8, int16, uint16, int32, uint32, boolean, single, or double. 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 Signal Processing Blockset Repeat block using the Maintain input frame rate option.)

# Error Rate Calculation

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If you check the **Reset port** box, then an additional inport appears, labeled Rst. The Rst input must be a sample-based scalar signal (of type double or boolean) and must have the same sampling rate as the Tx and Rx signals. When the Rst input is nonzero, the block clears its error statistics and then computes them anew.

If you set the **Computation mode** parameter to `Select samples from port`, then an additional inport appears, labeled Sel. The Sel input indicates which elements of a frame are relevant for the computation; this is explained further, in the last subbullet below. The Sel input can be either a sample-based column vector or a one-dimensional 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. You should leave the **Computation mode** parameter at its 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 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** Simulink requires that input signals have constant length throughout a simulation. 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 **Se1** signal with zeros. The Error Rate Calculation block ignores zeros in the **Se1** signal.

---

## Output Data

This block produces a vector of length three, whose entries correspond to:

- The error rate
- The total number of errors, that is, comparisons between unequal elements
- 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.

# Error Rate Calculation

---

If you plan to use this block along with the Real-Time Workshop, then you should not use the `Workspace` option; instead, use the `Port` option below 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.

## Delays

The **Receive delay** and **Computation delay** parameters implement two different types of delays for this block. One is useful when part of your model causes a lag in the received data, and the other is useful when you want to ignore the transient behavior of both input signals:

- The **Receive delay** parameter is the number of samples by which the received data lags behind the transmitted data. This parameter tells the block which samples "correspond" to each other and should be compared. The receive delay persists throughout the simulation.
- The **Computation delay** parameter tells the block to ignore the specified number of samples 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, you should set the **Receive delay** in the Error Rate Calculation block to 0.

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 that value.

---

**Note** The Version 1.4 Error Rate Calculation block considers a vector input to be a sample, whereas the current block considers a vector input to be a frame of multiple samples. For vector inputs of length  $n$ , a **Receive delay** of  $k$  in the Version 1.4 block is equivalent to a **Receive delay** of  $k*n$  in the current block.

---

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.

## 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 the **Stop simulation** check box. The block attempts to run the simulation until it detects **Target number of errors** errors. 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`.

## Examples

The figure below shows how the block compares pairs of elements and counts the number of error events. 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

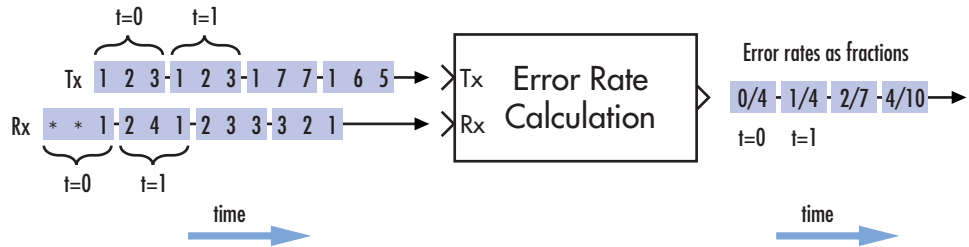
The input signals are both frame-based column vectors of length three. However, the schematic arranges each column vector horizontally and aligns pairs of vectors so as to reflect a receive delay of two samples. At each time step, the block compares elements of the Rx signal with those

# Error Rate Calculation

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  $t$ .

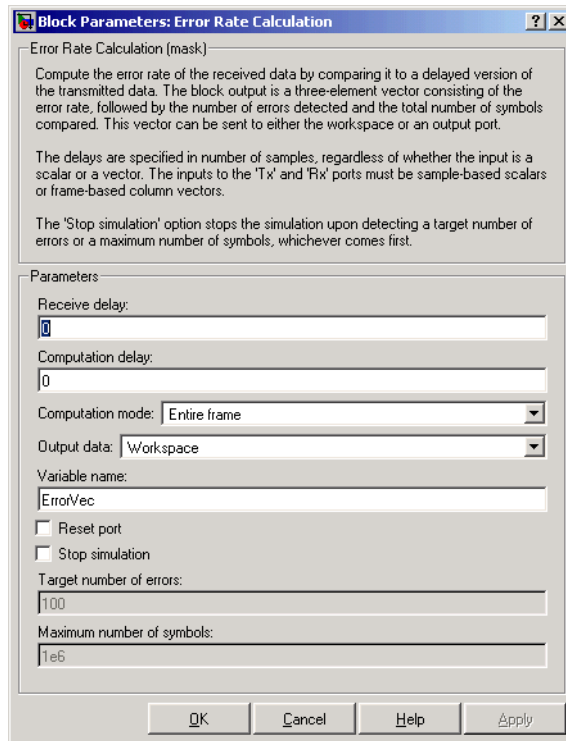


Note: Tx and Rx inputs are frame-based column vectors.

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.



# Error Rate Calculation



## Dialog Box

### 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.)

### Computation delay

Number of samples that the block should ignore at the beginning of the comparison.

### 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.



**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.

**Output data**

Either Workspace or Port, depending on where you want to send the output data.

**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.

**Reset port**

If you check this box, then an additional input port appears, labeled Rst.

**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.

**Target number of errors**

The simulation stops after detecting this number of errors. This field is active only if **Stop simulation** is checked.

**Maximum number of symbols**

The simulation stops after making this number of comparisons. This field is active only if **Stop simulation** is checked.

**See Also**

Align Signals, Find Delay

# Find Delay

---

**Purpose** Find delay between two signals

**Library** Utility Blocks

## Description



The Find Delay block finds the delay between a signal and a delayed, and possibly distorted, version of itself. The block is particularly 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. See “Computing Delays” for more information about signal delays.

The input port labeled sRef receives the original signal, while the input port labeled sDej receives the delayed version of the signal. The two input signals must have the same 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 output ports output signals of type double for double inputs, and uint32 for inputs of other non-double data types.

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 the **Disable recurring updates** check box, 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.

## 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.

## Examples

### 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 “Finding the Delay in a Model”. 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 “Computing Delays”.

### Finding the Delay to Help Align Words

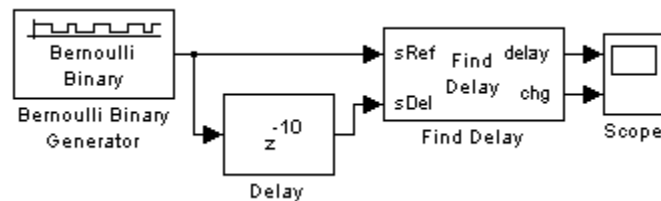
Another typical use of this block is to determine how to align the boundaries of frames with the boundaries of codewords or other types of

# Find Delay

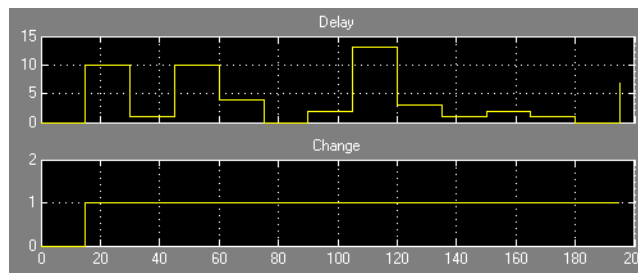
data blocks. “Manipulating Delays” describes when such alignment is necessary and also illustrates, in the “Aligning Words of a Block Code” discussion, how to use the Find Delay block to solve the problem.

## Setting the Correlation Window Length

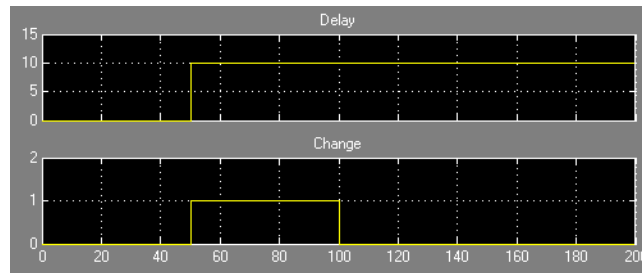
The next example illustrates how to tell when the **Correlation window length** is not sufficiently large.



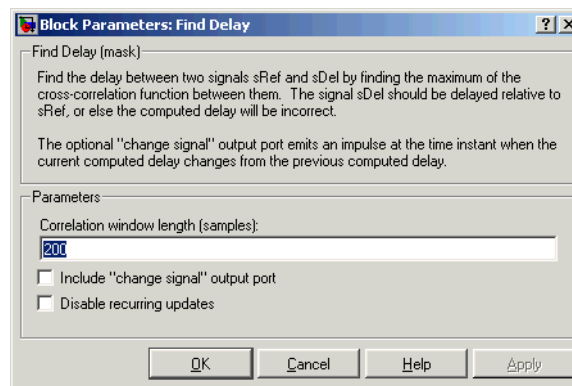
The model uses a Delay block to delay a signal by 10 samples, and uses the Find Delay block to compare the original signal with the delayed version. The model then displays the output of the Find Delay block in a scope. If the **Correlation window length** is 15, the scope shows that the calculated delay is not constant over time, as you can see below.



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.



## Dialog Box



### Correlation window length

The number of samples the block uses to calculate the cross-correlations of the two signals.

### 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.

### 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.

# Find Delay

---

## **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.

## **Algorithm**

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.

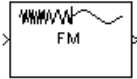
## **See Also**

Align Signals, Error Rate Calculation

**Purpose** Demodulate FM-modulated data

**Library** Analog Passband Modulation, in Modulation

## Description

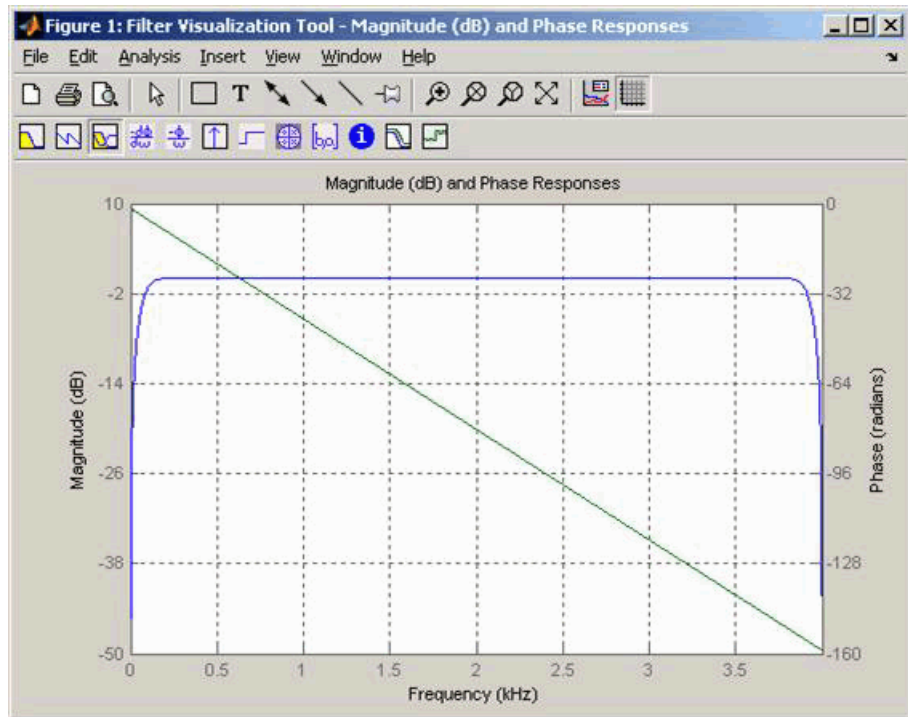


The FM Demodulator Passband block demodulates a signal that was modulated using frequency modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real sample-based scalar signals.

For best results, use a carrier frequency which is estimated to be larger than 10% of your input signal's sample time. This is due to the implementation of the Hilbert transform by means of a filter.

In the following example, we sample a 10Hz 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`.

# FM Demodulator Passband



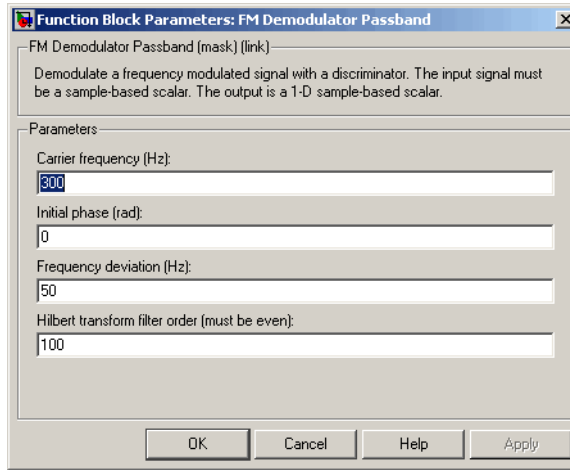
Note the bandwidth of the filter's magnitude response. By choosing a carrier frequency larger than 10% (but less than 90%) of the input signal's sample time (8000 samples per second, in this example) or equivalently, a carrier frequency larger than 400Hz, we ensure that the Hilbert Transform Filter will 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 is not suited to be placed inside a triggered subsystem.

## Dialog Box



### Carrier frequency (Hz)

The frequency of the carrier.

### Initial phase (rad)

The initial phase of the carrier in radians.

### Frequency deviation (Hz)

The frequency deviation of the carrier frequency in Hertz.  
Sometimes it is referred to as the "variation" in the frequency.

### Hilbert transform filter order

The length of the FIR filter used to compute the Hilbert transform.

## Pair Block

FM Modulator Passband

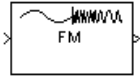
# FM Modulator Passband

---

**Purpose** Modulate using frequency modulation

**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 sample-based 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:

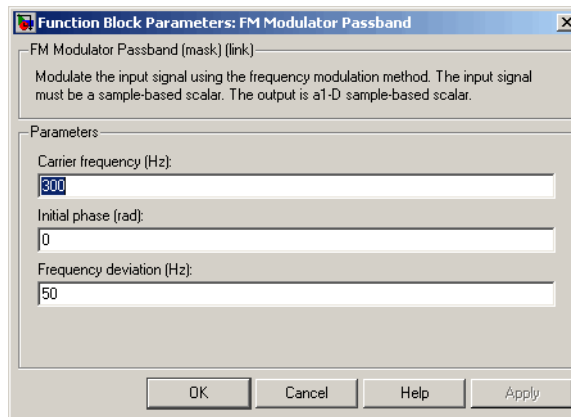
- $f_c$  is the **Carrier frequency** parameter.
- $\theta$  is the **Initial phase** parameter.
- $K_c$  is the **Modulation constant** parameter.

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 is not suited to be placed inside a triggered subsystem.

## Dialog Box



### **Carrier frequency (Hz)**

The frequency of the carrier.

### **Initial phase (rad)**

The initial phase of the carrier in radians.

### **Frequency deviation (Hz)**

The frequency deviation of the carrier frequency in Hertz.

Sometimes it is referred to as the "variation" in the frequency.

## Pair Block

FM Demodulator Passband

# Free Space Path Loss

**Purpose** Reduce amplitude of input signal by amount specified

**Library** RF Impairments

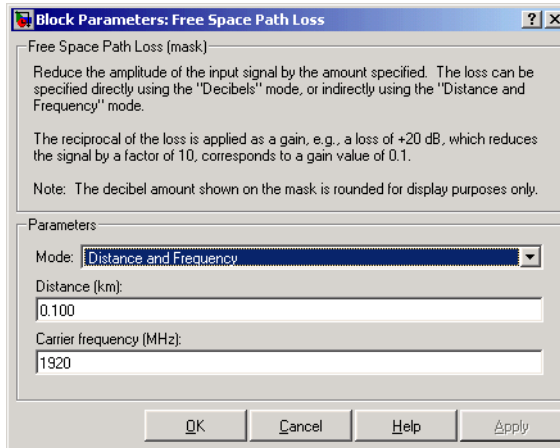
**Description** The Free Space Path Loss block simulates the loss of signal power due to the distance between transmitter and receiver. The block reduces the amplitude of the input signal by an amount that is determined in either of two ways:



- 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

The input to this block must be a complex signal.

## Dialog Box



**Mode** Method of specifying the amount by which the signal power is reduced. The choices are Decibels and Distance and Frequency.

## Loss

The signal loss in decibels. This parameter appears when you set **Mode** to Decibels.

## Distance

Distance between transmitter and receiver in kilometers. This parameter appears when you set **Mode** to Distance and Frequency.

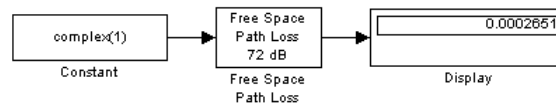
## Carrier frequency (MHz)

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 (MHz)** is set to 180



## See Also

Memoryless Nonlinearity

# Gardner Timing Recovery

---

**Purpose** Recover symbol timing phase using Gardner's method

**Library** Timing Phase Recovery sublibrary of Synchronization

## Description



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.

## 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. For best results, the input signal power should be less than 1. The input must be a scalar or a frame-based column vector. The input uses  $N$  samples to represent each symbol, where  $N > 1$  is the **Samples per symbol** parameter. If the input is frame-based, then its vector length is  $N \cdot R$ , where  $R$  is a positive integer that indicates the number of symbols per frame. If the input is sample-based, then its sample time is  $1/N$  times the underlying symbol period.

If the **Reset** parameter is set to 0n 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. The sample time of the *Rst* input equals the symbol period if the input signal is sample-based, and the frame period if the input signal is frame-based.

## 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:

- If the input signal is a frame-based column vector of length  $N \times R$ , then the Sym output is a frame-based column vector of length  $R$  having the same frame period.
- If the input signal is a sample-based scalar with sample time  $T/N$ , then the Sym output is a sample-based scalar with sample time  $T$ .
- 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 or frame period 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.

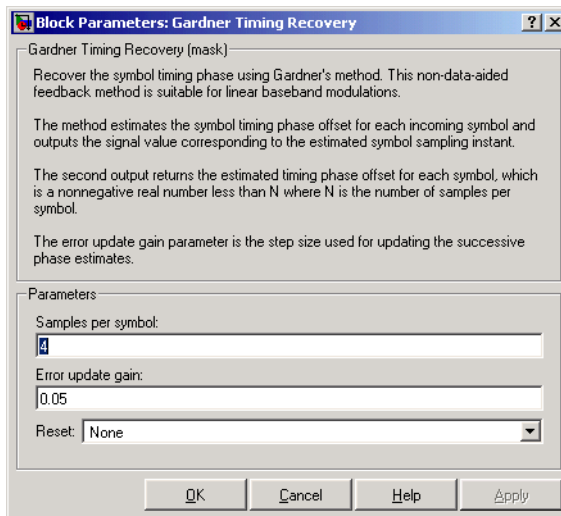
---

## Delays

This block incurs a delay of two symbols when the input is frame-based and three symbols when the input is sample-based.

# Gardner 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  $1/N$ , which corresponds to a slowly varying phase.

### 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 uses a timing error detector whose result for the  $k$ th symbol is  $e(k)$ , given by



$$e(k) = a_I(k) + a_Q(k)$$
$$a_I(k) = \{y_I((k-1)T + d_{k-1}) - y_I(kT + d_k)\} y_I(kT - T/2 + d_{k-1})$$
$$a_Q(k) = \{y_Q((k-1)T + d_{k-1}) - y_Q(kT + d_k)\} y_Q(kT - T/2 + d_{k-1})$$

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  $k$ th 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 Blockset User's Guide*.

## Examples

The `gardner_vfracdelay` demonstration model uses this block.

## References

- [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.

# Gardner Timing Recovery

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## **See Also**

Early-Late Gate Timing Recovery, Squaring Timing Recovery, Mueller-Muller Timing Recovery

**Purpose** Filter input signal, possibly downsampling, using Gaussian FIR filter

**Library** Comm Filters

## Description



The Gaussian Filter block filters the input signal using a Gaussian FIR filter. The block expects the input signal to be upsampled, so that the **Input samples per symbol** parameter, N, is at least 2. The block's icon shows the filter's impulse response."

## 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

$$\delta = \frac{\sqrt{\ln(2)}}{2\pi BT}$$

and B is the filter's 3-dB bandwidth. The **BT product** parameter is B times the input signal's symbol period. For a given BT product, the `gaussfir` function in Signal Processing Toolbox generates a filter that is half the bandwidth of the filter generated by the Communications Blockset 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  $2 * N * \text{Group delay} + 1$ .

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.

# Gaussian Filter

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- 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.

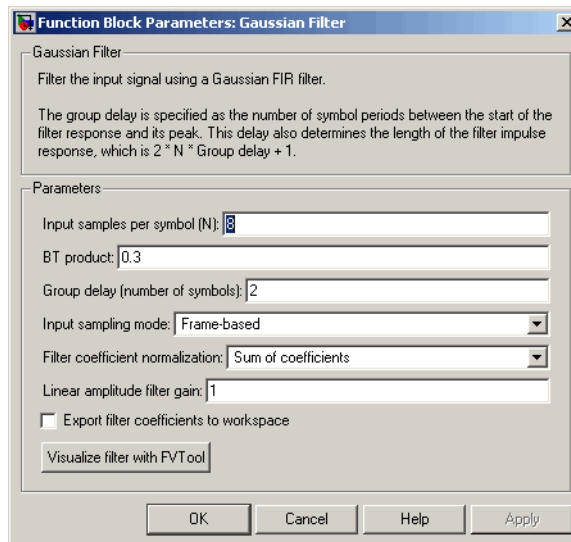
## Input and Output Signals

The input signal must be a scalar or a frame-based column vector. Set the **Input sampling mode** parameter according to whether the input is sample-based or frame-based. `double`, `single`, and fixed-point data types are supported.

## 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.

## Dialog Box



### Input samples per symbol

A positive integer representing the number of samples per symbol in the input signal.

### BT product

The product of the filter's 3-dB bandwidth and the input signal's symbol period

### Group delay

A positive integer that represents the number of symbol periods between the start of the filter response and its peak.

### Input sampling mode

The type of input signal: Frame-based or Sample-based.

### 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.

# Gaussian Filter

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## **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.

## **Export filter coefficients to workspace**

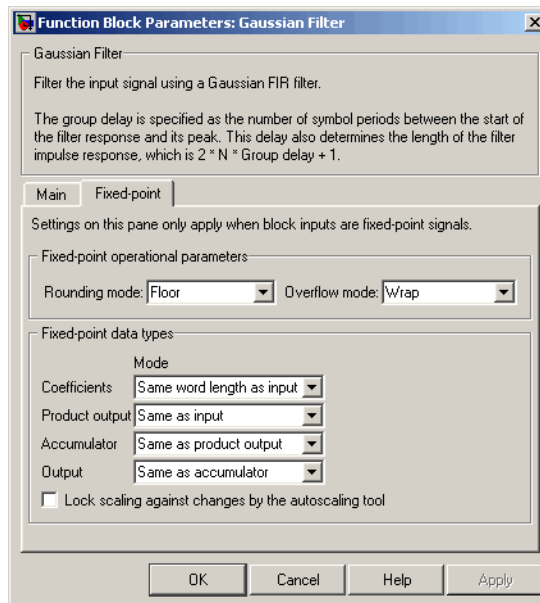
If you check this box, then the block creates a variable in the MATLAB workspace that contains the filter coefficients.

## **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 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.



## Rounding mode

Select the rounding mode for fixed-point operations. The filter coefficients do not obey this parameter; they always round to Nearest.

## 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 *Signal Processing Blockset 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 *Signal Processing Blockset 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.

## 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.

# Gaussian Filter

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## **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.

## **See Also**

Raised Cosine Receive Filter, `gaussfir`

## **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)

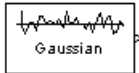
## Purpose

Generate Gaussian distributed noise with given mean and variance values

## Library

Noise Generators sublibrary of Comm Sources

## Description



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

$$f(x) = \left( (2\pi)^n \det K \right)^{-1/2} \exp \left( -(x - \mu)^T K^{-1} (x - \mu) / 2 \right)$$

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.

### 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

# Gaussian Noise Generator

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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 Blockset `randseed` function. At the MATLAB prompt, enter

```
randseed
```

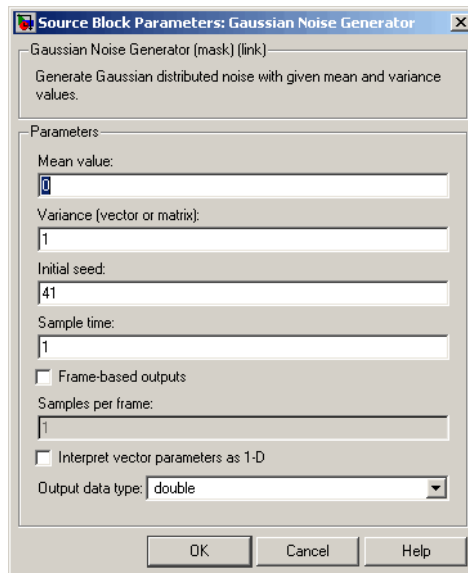
This returns a random prime number greater than 30. Entering `randseed` again produces a different prime number. If you supply an integer argument, `randseed` always returns the same prime for that integer. For example, `randseed(5)` always returns the same answer.

## 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 “Signal Attribute Parameters for Random Sources” in *Communications Blockset 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.

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

### Mean value

The mean value of the random variable output.

### Variance

The covariance among the output random variables.

### 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.

# Gaussian Noise Generator

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## **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**

Random Source (Signal Processing Blockset), AWGN Channel, rand (built-in MATLAB function), randseed

**Purpose** Restore ordering of symbols in input vector

**Library** Block sublibrary of Interleaving

## Description



The General Block Deinterleaver block rearranges the elements of its input vector without repeating or omitting any elements. The input can be real or complex. If the input contains  $N$  elements, then the **Elements** parameter is a vector of length  $N$  that indicates the indices, in order, of the output elements that came from the input vector. That is, for each integer  $k$  between 1 and  $N$ ,

$$\text{Output}(\mathbf{Elements}(k)) = \text{Input}(k)$$

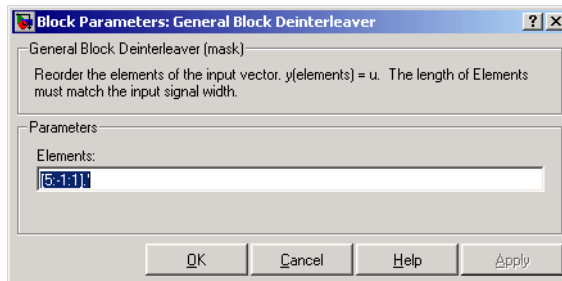
The **Elements** parameter must contain unique integers between 1 and  $N$ .

If the input is frame-based, then both it and the **Elements** parameter must be column vectors.

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.

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.

## Dialog Box



# General Block Deinterleaver

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## Elements

A vector of length N that lists the indices of the output elements that came from the input vector.

## Examples

This example reverses the operation in the example on the General Block Interleaver block reference page. If **Elements** is [4, 1, 3, 2] and the input to the General Block Deinterleaver block is [1;40;59;32], then the output of the General Block Deinterleaver block is [40;32;59;1].

## Pair Block

General Block Interleaver

## See Also

perms (MATLAB function)



**Purpose** Reorder symbols in input vector

**Library** Block sublibrary of Interleaving

## Description



The General Block Interleaver block rearranges the elements of its input vector without repeating or omitting any elements. The input can be real or complex. If the input contains  $N$  elements, then the **Elements** parameter is a vector of length  $N$  that indicates the indices, in order, of the input elements that form the length- $N$  output vector; that is,

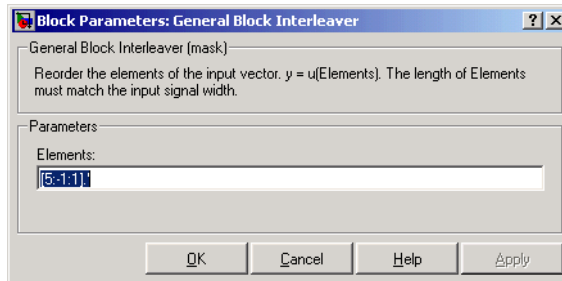
$$\text{Output}(k) = \text{Input}(\mathbf{Elements}(k))$$

for each integer  $k$  between 1 and  $N$ . The contents of **Elements** must be integers between 1 and  $N$ , and must have no repetitions.

If the input is frame-based, then both it and the **Elements** parameter must be column vectors.

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.

## Dialog Box



## Elements

A vector of length  $N$  that lists the indices of the input elements that form the output vector.

# General Block Interleaver

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## Examples

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].

## Pair Block

General Block Deinterleaver

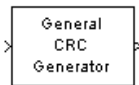
## See Also

perms (MATLAB function)

**Purpose** Generate CRC bits according to generator polynomial and append to input data frames

**Library** CRC sublibrary of Error Correction and Detection

## Description



The General CRC Generator block generates cyclic redundancy code (CRC) bits for each input data frame and appends them to the frame. 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, [1 1 0 1] 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, [3 2 0] 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
- 3 Applies the CRC algorithm to each augmented subframe

# General CRC Generator

4 Appends the resulting checksums at the end of each subframe

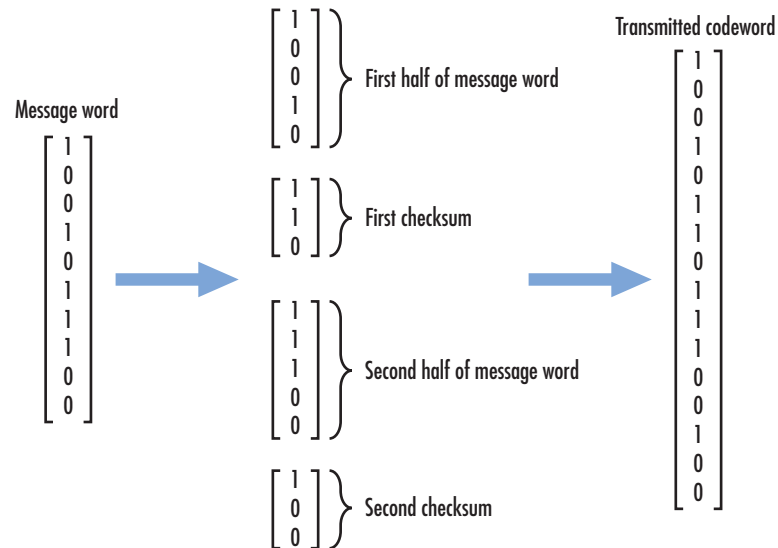
5 Outputs concatenated subframes

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 output data type is inherited from the input.

## Example

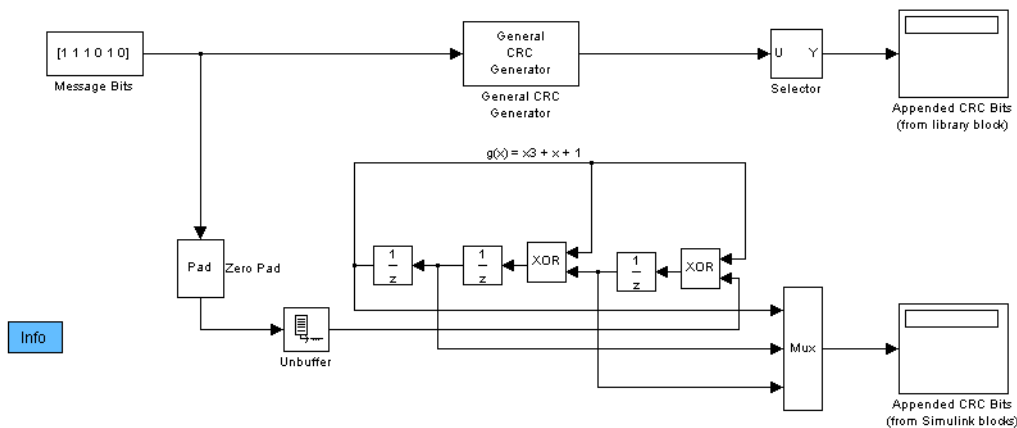
Suppose the size of the input frame is 10, the degree of the generator polynomial is 3, **Initial states** is [0], and **Checksums per frame** is 2. The block divides each input frame into two subframes of size 5 and appends a checksum of size 3 to each subframe, as shown below. The initial states are 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  $5 + 3 + 5 + 3 = 16$ .



## 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. The model is located at: `matlabroot/help/toolbox/commblocks/commblocks_examples/doc_crcgen`.

### 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.

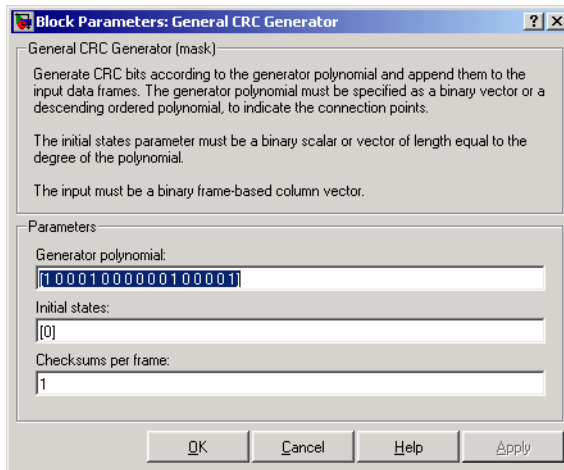
Using the General CRC Generator block allows you to easily specify the generator polynomial (especially for higher order polynomials).

### Signal Attributes

The General CRC Generator block has one input port and one output port. Both ports allow only frame-based binary column vectors.

# General CRC Generator

## Dialog Box



### Generator polynomial

A binary or integer row vector specifying the generator polynomial, in descending order of powers.

### Initial states

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.

### Checksums per frame

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 Coding” in *Communications Blockset User’s Guide*.



# General CRC Generator

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## **See Also**

CRC-N Generator, CRC-N Syndrome Detector

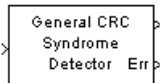


# General CRC Syndrome Detector

**Purpose** Detect errors in input data frames according to generator polynomial

**Library** CRC sublibrary of Error Correction and Detection

## Description



The General CRC Syndrome Detector block computes checksums for its entire input frame. 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 block's parameter settings should agree with those in the General CRC Generator block.

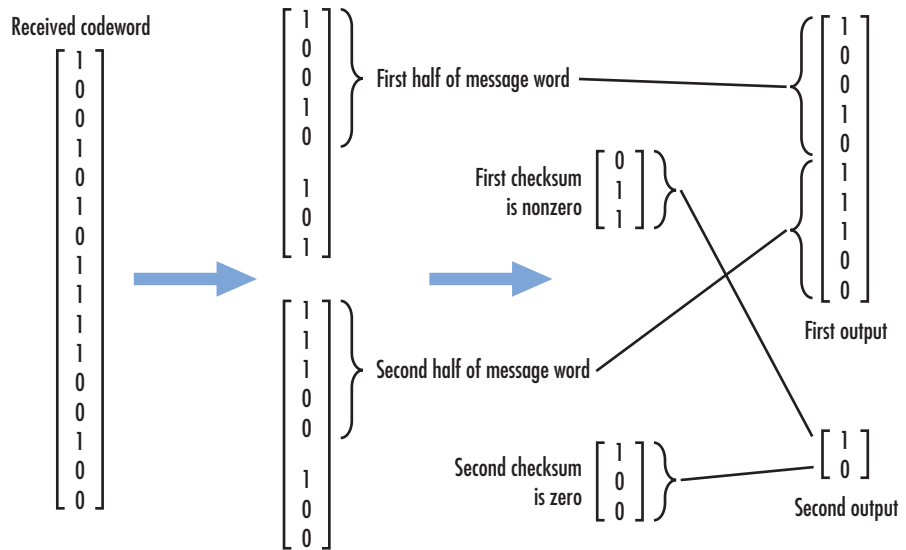
You specify the number of checksums the block calculates for each frame by the **Checksums per frame** parameter. If the **Checksums per frame** value is  $k$ , the size of the input frame is  $n$ , and the degree of the generator polynomial is  $r$ , then  $k$  must divide  $n - k*r$ , which is the size of the message word.

This block supports double and boolean data types. The output data type is inherited from the input.

## 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 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.

# General CRC Syndrome Detector

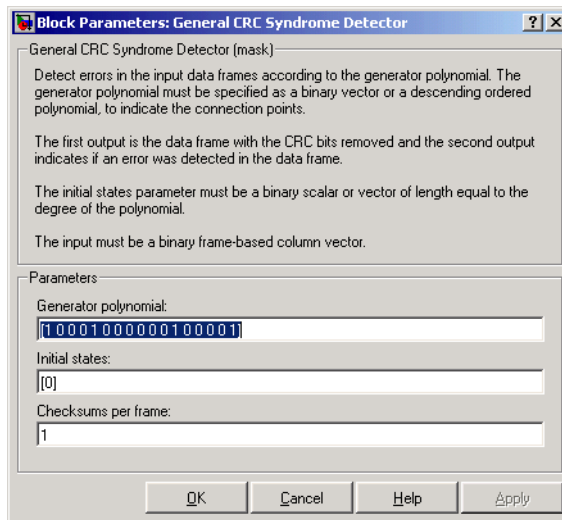


## Signal Attributes

The General CRC Syndrome Detector block has one input port and two output ports. All ports allow frame-based binary column vectors only.

# General CRC Syndrome Detector

## Dialog Box



### Generator polynomial

A binary or integer row vector specifying the generator polynomial, in descending order of powers.

### 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.

### 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 Coding” in *Communications Blockset User’s Guide*.

## References

[1] Sklar, Bernard. *Digital Communications: Fundamentals and Applications*. Englewood Cliffs, N.J., Prentice-Hall, 1988.

# General CRC Syndrome Detector

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[2] Wicker, Stephen B., *Error Control Systems for Digital Communication and Storage*, Upper Saddle River, N.J., Prentice Hall, 1995.

**Pair Block**      General CRC Generator

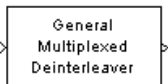
**See Also**      CRC-N Generator, CRC-N Syndrome Detector

# General Multiplexed Deinterleaver

**Purpose** Restore ordering of symbols using specified-delay shift registers

**Library** Convolutional sublibrary of Interleaving

**Description** The General Multiplexed Deinterleaver block restores the original ordering of a sequence that was interleaved using the General Multiplexed Interleaver block.

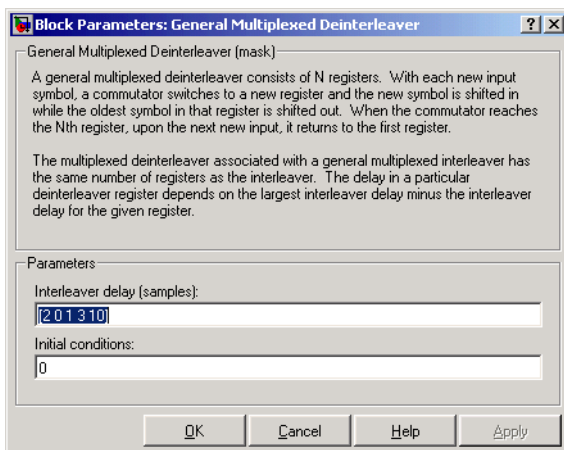


In typical usage, the parameters in the two blocks have the same values. As a result, the **Interleaver delay** parameter,  $V$ , specifies the delays for each shift register in the corresponding *interleaver*, so that the delays of the deinterleaver's shift registers are actually  $\max(V) - V$ .

The input can be either a scalar or a frame-based column vector. It can be real or complex. The input and output signals share the same sample time.

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.

## Dialog Box



# General Multiplexed Deinterleaver

---

## **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.

## **Initial conditions**

The values that fill each shift register when the simulation begins.

## **Pair Block**

General Multiplexed Interleaver

## **See Also**

Convolutional Deinterleaver, Helical Deinterleaver

## **References**

[1] Heegard, Chris and Stephen B. Wicker. *Turbo Coding*. Boston: Kluwer Academic Publishers, 1999.

# General Multiplexed Interleaver

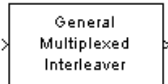
## Purpose

Permute input symbols using set of shift registers with specified delays

## Library

Convolutional sublibrary of Interleaving

## Description



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.

The input can be either a scalar or a frame-based column vector. It can be real or complex. The input and output signals share the same sample time.

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.

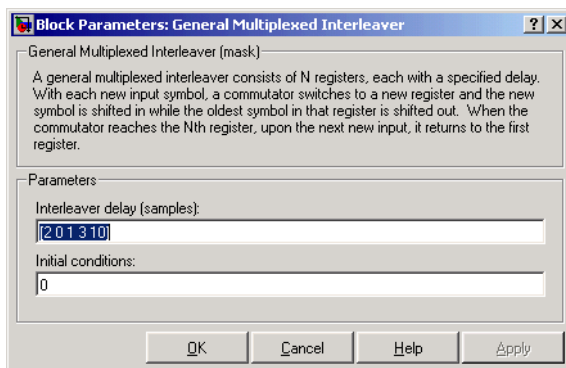
The **Interleaver delay** parameter is a column vector whose entries indicate how many symbols can fit into each shift register. The length of the vector is the number of shift registers. (In sample-based mode, it can also be a row vector.)

The **Initial conditions** parameter indicates 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.

# General Multiplexed Interleaver

---

## Dialog Box



### Interleaver delay (samples)

A vector that lists the number of symbols that fit in each shift register. The length of this vector is the number of shift registers.

### Initial conditions

The values that fill each shift register when the simulation begins.

## Pair Block

General Multiplexed Deinterleaver

## See Also

Convolutional Interleaver, Helical Interleaver

## References

[1] Heegard, Chris and Stephen B. Wicker. *Turbo Coding*. Boston: Kluwer Academic Publishers, 1999.



# General QAM Demodulator Baseband

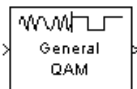
## Purpose

Demodulate QAM-modulated data

## Library

AM, in Digital Baseband sublibrary of Modulation

## Description

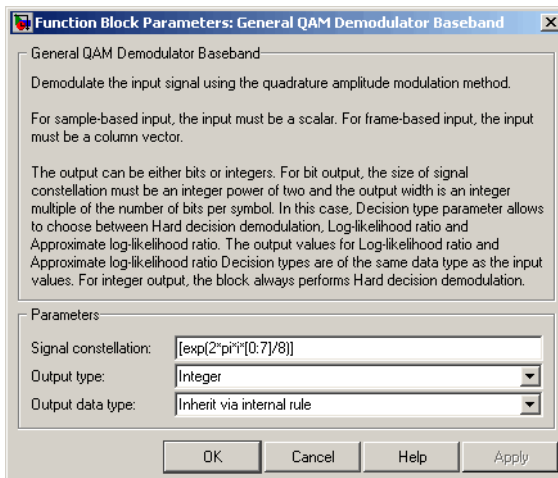


The General QAM Demodulator Baseband block demodulates a signal that was modulated using quadrature amplitude modulation. The input is a baseband representation of the modulated signal.

The input must be a discrete-time complex signal. The **Signal constellation** parameter defines the constellation by listing its points in a vector of complex numbers. The block maps the  $m$ th point in the **Signal constellation** vector to the integer  $m-1$ .

The input can be either a scalar or a frame-based column vector and must be of data types `single` or `double`.

## Dialog Box



### Signal constellation

A real or complex vector that lists the constellation points.

### Output type

Determines whether the block produces integers or binary representations of integers.

# General QAM Demodulator Baseband

---

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 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” for algorithm details.

## **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's output can be set to Inherit via internal rule (default setting), Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, single, and double.

For bit outputs, when **Decision type** is set to Hard decision, output can be set to Inherit via internal rule, Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, boolean, single, or double. When **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is inherited from the input (i.e., if the input is of data type double, the output is also of data type 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 block. 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`).

## 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.

## Noise variance

This field appears when **Noise variance source** is set to Dialog.

If this value is very small (i.e., SNR is very high), LLR computations may yield Inf or -Inf because the LLR algorithm would involve computing exponentials of very large or very small numbers using finite precision arithmetic. In such cases, use of approximate LLR is recommended, as its algorithm does not involve computing exponentials.

## Pair Block

General QAM Modulator Baseband

## See Also

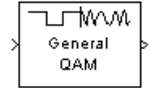
Rectangular QAM Demodulator Baseband

# General QAM Modulator Baseband

**Purpose** Modulate using quadrature amplitude modulation

**Library** AM, in Digital Baseband sublibrary of Modulation

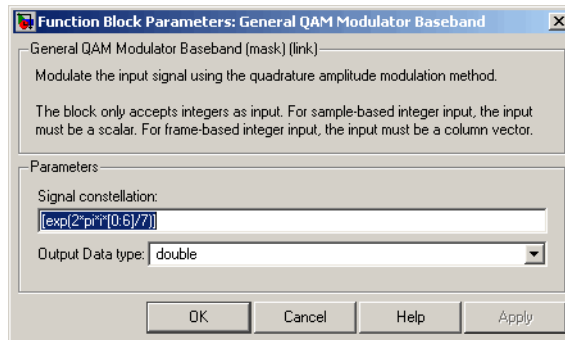
**Description** The General QAM Modulator Baseband block modulates using quadrature amplitude modulation. The output is a baseband representation of the modulated signal.



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 M-1. The block maps an input integer  $m$  to the  $(m+1)$ st value in the **Signal constellation** vector.

The input can be either a scalar or a frame-based column vector. 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`.

## Dialog Box



### Signal constellation

A real or complex vector that lists the constellation points.

### 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..

## **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 Simulink Fixed Point. 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 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.

## **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.

# General QAM Modulator Baseband

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**Pair Block**      General QAM Demodulator Baseband

**See Also**      Rectangular QAM Modulator Baseband

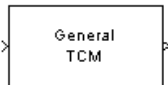
## Purpose

Decode trellis-coded modulation data, mapped using arbitrary constellation

## Library

Trellis-Coded Modulation

## Description



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.

### Input and Output Signals

The input signal must be a frame-based column vector containing complex numbers.

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 frame-based binary column vector whose length is  $k$  times the vector length of the input signal.

The input signal must be double or single. The reset port accepts double or boolean.

### 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.

# General TCM Decoder

---

If you select the **Enable the reset input port** check box, 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.

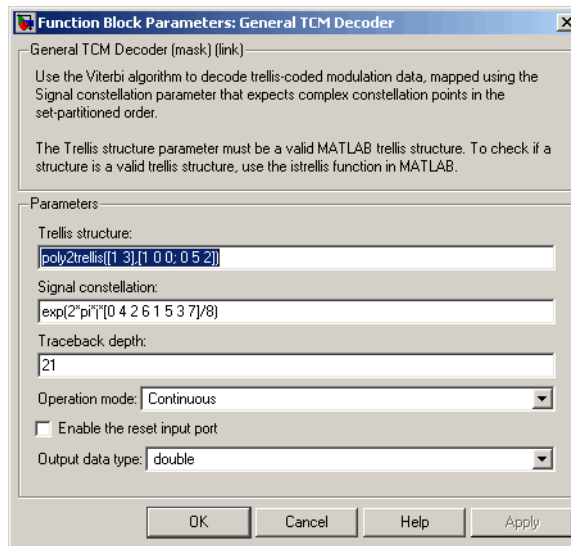
## Decoding Delay

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**.



## Dialog Box



### Trellis structure

MATLAB structure that contains the trellis description of the convolutional encoder.

### Signal constellation

A complex vector that lists the points in the signal constellation in set-partitioned order.

### 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. The 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 value to this port causes the block to set its internal memory to the initial state before processing the

# General TCM Decoder

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input data. This field 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.

## **Pair Block**

General TCM Encoder

## **See Also**

M-PSK TCM Decoder, Rectangular QAM TCM Decoder, poly2trellis

## **References**

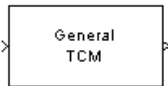
[1] Biglieri, E., D. Divsalar, P. J. McLane, and M. K. Simon, *Introduction to Trellis-Coded Modulation with Applications*, New York, Macmillan, 1991.

[2] Proakis, John G., *Digital Communications*, Fourth edition, New York, McGraw-Hill, 2001.

**Purpose** Convolutionally encode binary data and map using arbitrary constellation

**Library** Trellis-Coded Modulation

## Description



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 points in the signal constellation are listed in set-partitioned order in the **Signal constellation** parameter. This parameter is a complex vector whose length,  $M$ , equals 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.)

### Input and Output Signals

If the convolutional encoder represents a rate  $k/n$  code, then the General TCM Encoder block's input must be a frame-based binary column vector whose length is  $L*k$  for some positive integer  $L$ .

The output from the General TCM Encoder block is a frame-based complex column vector of length  $L$ .

The input signal must be boolean.

### 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 Encoder” in the Communications 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

```
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 to spend less time updating the diagram at the beginning of each simulation, compared to the usage in the previous bulleted item.

## 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.

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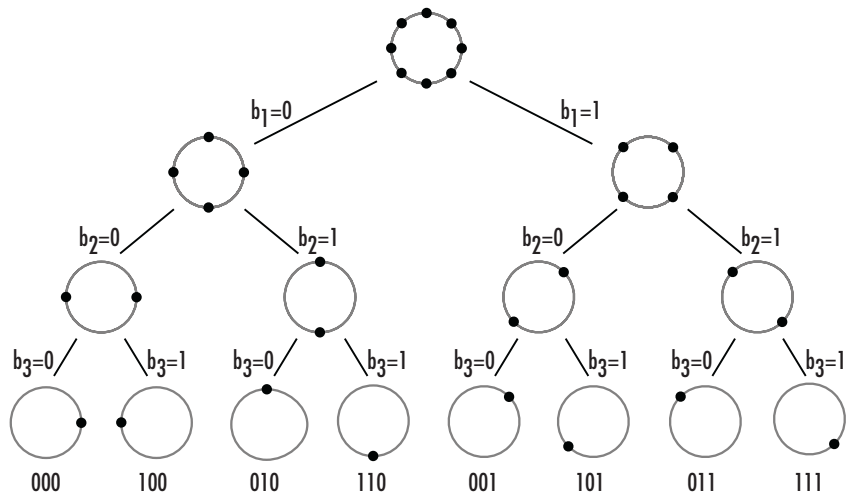
**Note** When you set the **Signal constellation** parameter, you must ensure that the constellation vector is already in set-partitioned order. Otherwise, the block might produce unexpected or suboptimal results.

---

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 ( $b_3, b_2, b_1$ ), 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)
```

which is a valid value for the **Signal constellation** parameter in this block.



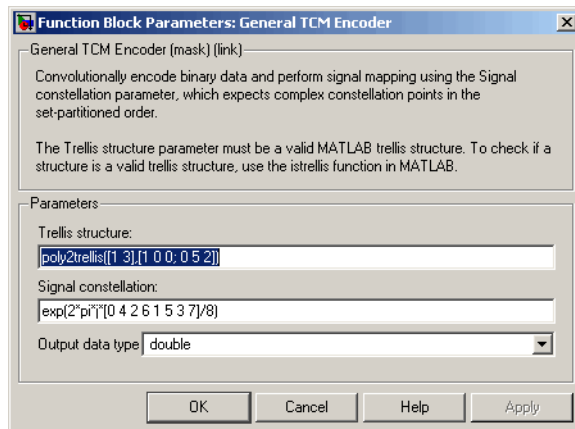
For other examples of signal constellations in set-partitioned order, see [1] or the reference pages for the M-PSK TCM Encoder and Rectangular QAM TCM Encoder blocks.

## 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].

# General TCM Encoder

## Dialog Box



### Trellis structure

MATLAB structure that contains the trellis description of the convolutional encoder.

### Signal constellation

A complex vector that lists the points in the signal constellation in set-partitioned order.

### 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

General TCM Decoder

## See Also

M-PSK TCM Encoder, Rectangular QAM 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.

# GMSK Demodulator Baseband

---

**Purpose** Demodulate GMSK-modulated data

**Library** CPM, in Digital Baseband sublibrary of Modulation

## Description



The GMSK Demodulator Baseband block demodulates a signal that was modulated using the Gaussian minimum shift keying method. The input is a baseband representation of the modulated signal.

The **BT product**, **Pulse length**, **Symbol prehistory**, and **Phase offset** parameters are as described on the reference page for the GMSK Modulator Baseband block.

## Traceback Length and Output Delays

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The **Traceback length** 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.

- If the input signal is sample-based, then the delay consists of  $D+1$  zero symbols.
- If the input signal is frame-based, then the delay consists of  $D$  zero symbols.

## Inputs and Outputs

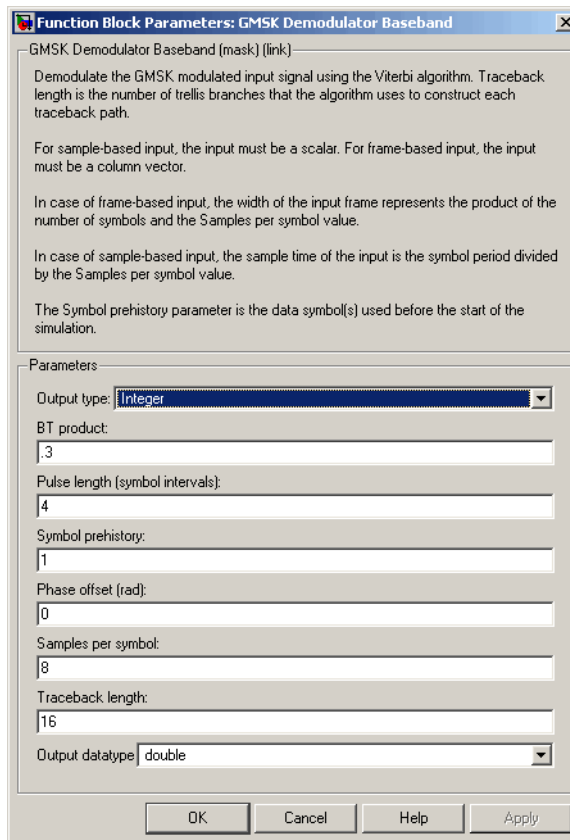
The input can be either a scalar or a frame-based column vector and must be of type `single` or `double`. If the **Output type** parameter is set to `Integer`, then the block produces values of 1 and -1. If the **Output type** parameter is set to `Bit`, then the block produces values of 0 and 1.

## Processing an Upsampled Modulated Signal

The input signal can be an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor. It must be a positive integer. For more information, see “Upsampled Signals and Rate Changes” in *Communications Blockset User’s Guide*.



## Dialog Box



### Output type

Determines whether the output consists of bipolar or binary values.

### BT product

The product of bandwidth and time.

### Pulse length (symbol intervals)

The length of the frequency pulse shape.

# GMSK Demodulator Baseband

---

**Symbol prehistory**

The data symbols used by the modulator before the start of the simulation.

**Phase offset (rad)**

The initial phase of the modulated waveform.

**Samples per symbol**

The number of input samples that represent each modulated symbol.

**Traceback length**

The number of trellis branches that the Viterbi Decoder block uses to construct each traceback path.

**Output data type**

The output data type can be boolean, int8, int16, int32, or double.

**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.

## Purpose

Modulate using Gaussian minimum shift keying method

## Library

CPM, in Digital Baseband sublibrary of Modulation

## Description



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.

$$g(t) = \frac{1}{2T} \left\{ Q \left[ 2\pi B_b \frac{t - \frac{T}{2}}{\sqrt{\ln(2)}} \right] - Q \left[ 2\pi B_b \frac{t + \frac{T}{2}}{\sqrt{\ln(2)}} \right] \right\}$$

$$Q(t) = \int_t^{\infty} \frac{1}{\sqrt{2\pi}} e^{-r^2/2} dt$$

The **Symbol prehistory** parameter is a scalar or vector that specifies the data symbols used 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.

In this block, a symbol of 1 causes a phase shift of  $\pi/2$  radians. The **Phase offset** parameter is the initial phase of the output waveform, measured in radians.

## Input Attributes

The input can be either a scalar or a frame-based column vector. If the **Input type** parameter is set to Integer, then the block accepts values

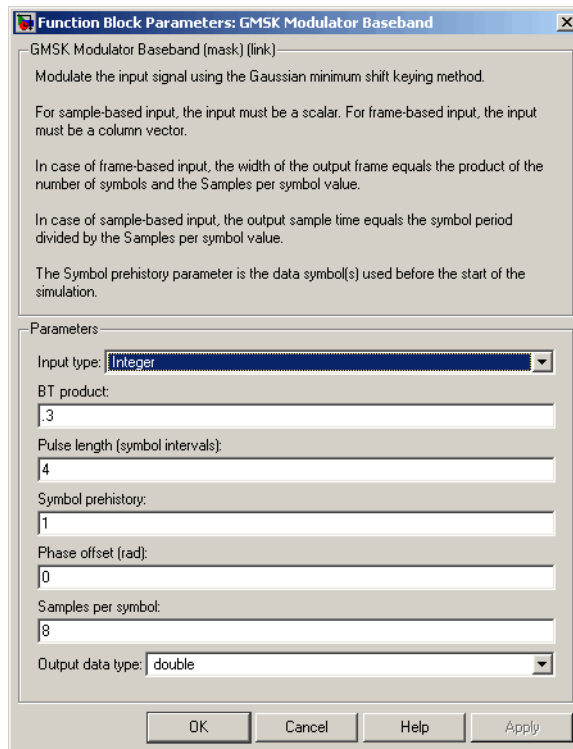
# GMSK Modulator Baseband

of 1 and -1. If the **Input type** parameter is set to Bit, then the block accepts values of 0 and 1.

## Upsampling the Modulated Signal

This block can output an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor. It must be a positive integer. For more information, see “Upsampled Signals and Rate Changes” in *Communications Blockset User’s Guide*.

## Dialog Box



### Input type

Indicates whether the input consists of bipolar or binary values.

**BT product**

The product of bandwidth and time.

**Pulse length (symbol intervals)**

The length of the frequency pulse shape.

**Symbol prehistory**

The data symbols used before the start of the simulation, in reverse chronological order.

**Phase offset (rad)**

The initial phase of the output waveform.

**Samples per symbol**

The number of output samples that the block produces for each integer or bit in the input.

**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**

GMSK Demodulator Baseband

**See Also**

CPM Modulator Baseband

**References**

[1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. *Digital Phase Modulation*. New York: Plenum Press, 1986.

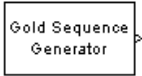
# Gold Sequence Generator

---

**Purpose** Generate Gold sequence from set of sequences

**Library** Sequence Generators sublibrary of Comm Sources

## Description



The Gold Sequence Generator block generates a Gold sequence. Gold sequences form a large class of sequences that have good periodic cross-correlation properties.

The Gold sequences are defined using a specified pair of sequences  $u$  and  $v$ , of period  $N = 2^n - 1$ , called a *preferred pair*, as defined in “Preferred Pairs of Sequences” on page 2-257 below. The set  $G(u, v)$  of Gold sequences is defined by

$$G(u, v) = \{u, v, u \oplus v, u \oplus Tv, u \oplus T^2v, \dots, u \oplus T^{N-1}v\}$$

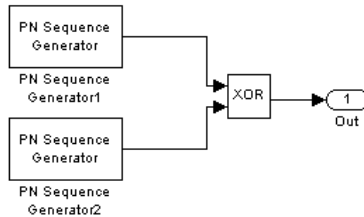
where  $T$  represents the operator that shifts vectors cyclically to the left by one place, and  $\oplus$  represents addition modulo 2. Note that  $G(u, v)$  contains  $N + 2$  sequences of period  $N$ . The Gold Sequence Generator block outputs one of these sequences according to the block’s parameters.

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 sequences to produce the output sequence, as shown in the following diagram.

# Gold Sequence Generator



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  $[1\ 0\ 0\ 1\ 0\ 1]$  both represent the polynomial  $z^5 + z^2 + 1$ .

The following table provides a short list of preferred pairs.

<b>n</b>	<b>N</b>	<b>Preferred Polynomial[1]</b>	<b>Preferred Polynomial[2]</b>
5	31	$[5\ 2\ 0]$	$[5\ 4\ 3\ 2\ 0]$
6	63	$[6\ 1\ 0]$	$[6\ 5\ 2\ 1\ 0]$
7	127	$[7\ 3\ 0]$	$[7\ 3\ 2\ 1\ 0]$

# Gold Sequence Generator

---

<b>n</b>	<b>N</b>	<b>Preferred Polynomial[1]</b>	<b>Preferred Polynomial[2]</b>
9	511	[9 4 0]	[9 6 4 3 0]
10	1023	[10 3 0]	[10 8 3 2 0]
11	2047	[11 2 0]	[11 8 5 2 0]

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  $[-2, -1, 0, 1, 2, \dots, 2^n-2]$ . The correspondence between **Sequence index** and the output sequence is given in the following table.

<b>Sequence Index</b>	<b>Output Sequence</b>
-2	$u$
-1	$v$
0	$u \oplus v$



Sequence Index	Output Sequence
1	$u \oplus Tv$
2	$u \oplus T^2v$
...	...
$2^n-2$	$u \oplus T^{2^n-2}v$

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 the **Reset on nonzero input** check box. 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 2-478 for an example.

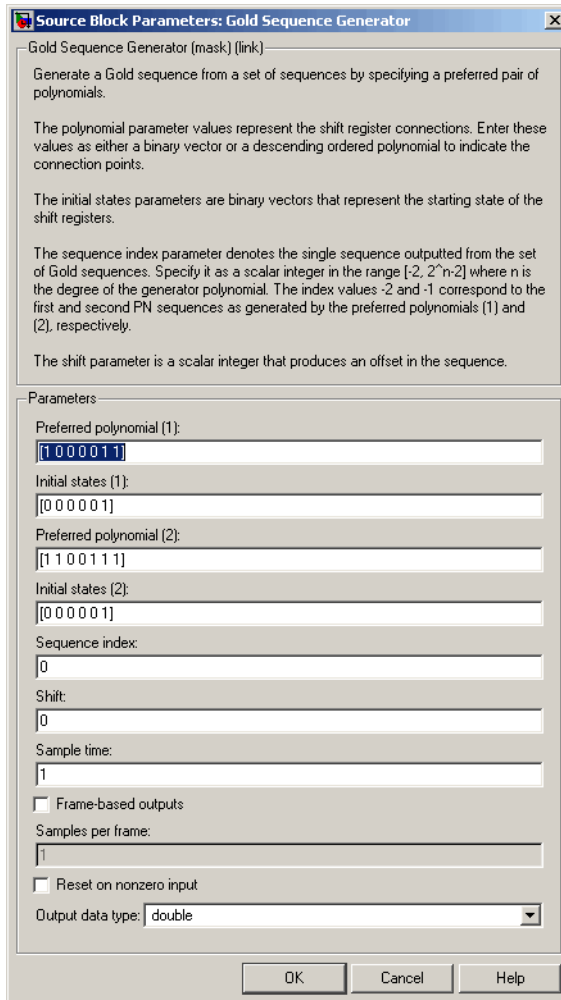
## Preferred Pairs of Sequences

The requirements for a pair of sequences  $u, v$  of period  $N = 2^n-1$  to be a preferred pair are as follows:

- $n$  is not divisible by 4
- $v = u[q]$ , where
  - $q$  is odd
  - $q = 2^k+1$  or  $q = 2^{2k}-2^k+1$
  - $v$  is obtained by sampling every  $q$ th symbol of  $u$
- $\text{gcd}(n, k) = \begin{cases} 1 & n \equiv 1 \pmod{2} \\ 2 & n \equiv 2 \pmod{4} \end{cases}$

# Gold Sequence Generator

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

**Preferred polynomial[1]**

Vector specifying the polynomial for the first sequence of the preferred pair.

**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.

**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 the **Frame-based outputs** check box.

**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 a boolean or double. By default, the block sets this to double.

# Gold Sequence Generator

---

## See Also

Kasami Sequence Generator, PN Sequence Generator

## References

- [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.

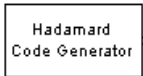
## Purpose

Generate Hadamard code from orthogonal set of codes

## Library

Sequence Generators sublibrary of Comm Sources

## Description



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  $H_N$ , is defined recursively as follows.

$$H_1 = [1]$$
$$H_{2N} = \begin{bmatrix} H_N & H_N \\ H_N & -H_N \end{bmatrix}$$

The  $N$ -by- $N$  Hadamard matrix has the property that

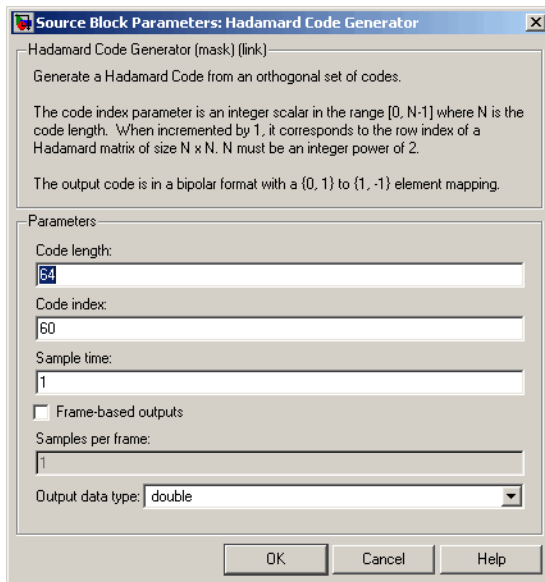
$$H_N H_N^T = N I_N$$

where  $I_N$  is the  $N$ -by- $N$  identity matrix.

The Hadamard Code Generator block outputs a row of  $H_N$ . 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  $[0, 1, \dots, N-1]$ , by the **Code index** parameter.

# Hadamard Code Generator

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

### 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 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 the **Frame-based outputs** check box.

**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**

OVSF Code Generator, Walsh Code Generator

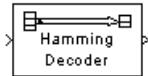
# Hamming Decoder

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**Purpose** Decode Hamming code to recover binary vector data

**Library** Block sublibrary of Channel Coding

## 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  $2^M - 1$  for some integer  $M$  greater than or equal to 3. Also,  $K$  must equal  $N - M$ .

The input must contain exactly  $N$  elements. If it is frame-based, then it must be a column vector. The output is a vector of length  $K$ .

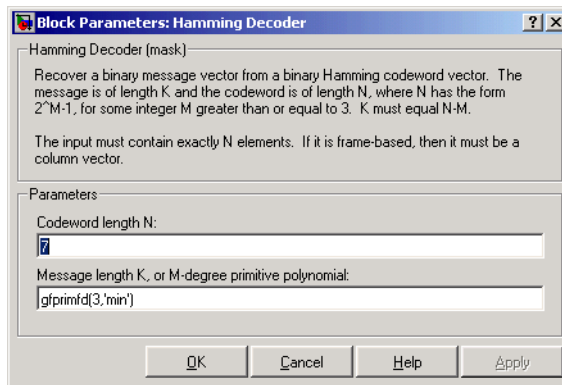
The coding scheme uses elements of the finite field  $GF(2^M)$ . You can either specify the primitive polynomial that the algorithm should use, or you can rely on the default setting:

- 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  $GF(2^M)$ .
- 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 `gfprimdf` function in Communications Toolbox.

This block supports double and boolean data types.



## Dialog Box



### Codeword length **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  $GF(2^M)$ .

## Pair Block

Hamming Encoder

## See Also

hamngen (Communications Toolbox)

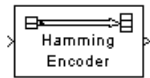
# Hamming Encoder

---

**Purpose** Create Hamming code from binary vector data

**Library** Block sublibrary of Channel Coding

## Description



The Hamming Encoder block creates a Hamming code with message length  $K$  and codeword length  $N$ . The number  $N$  must have the form  $2^M - 1$ , where  $M$  is an integer greater than or equal to 3. Then  $K$  equals  $N - M$ .

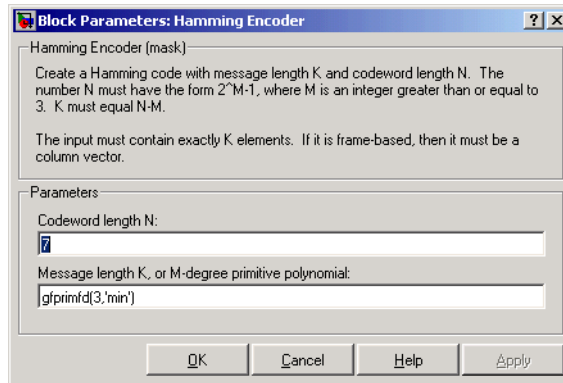
The input must contain exactly  $K$  elements. If it is frame-based, then it must be a column vector. The output is a vector of length  $N$ .

The coding scheme uses elements of the finite field  $GF(2^M)$ . 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  $GF(2^M)$ .
- 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 `gfprimdf` function in Communications Toolbox.

This block supports double and boolean data types.

## Dialog Box



### Codeword length 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  $GF(2^M)$ .

## Pair Block

Hamming Decoder

## See Also

hammgen (Communications Toolbox)

# Helical Deinterleaver

---

**Purpose** Restore ordering of symbols permuted by helical interleaver

**Library** Convolutional sublibrary of Interleaving

## Description



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  $k$ th output group in the array from column  $k \bmod C$ . The selection is helical because of the reduction modulo  $C$  and because the first symbol in the  $k$ th group is in row  $1 + (k-1) \cdot s$ , where  $s$  is the **Helical array step size** parameter.

The number of elements of the input vector must be  $C$  times  $N$ . If the input is frame-based, then it must be a column vector.

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

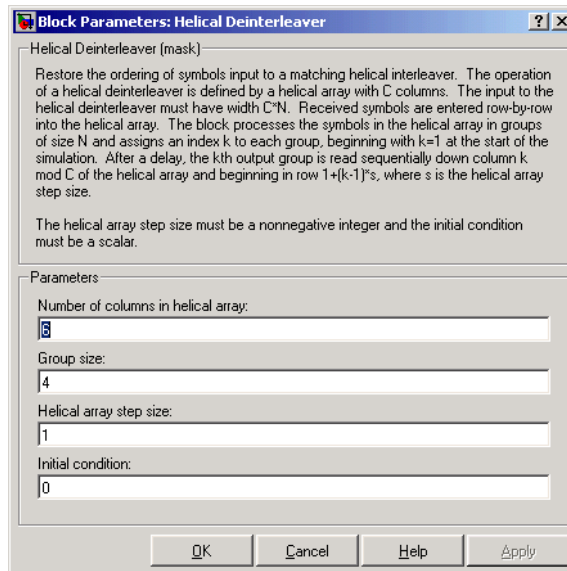
After processing a message with the Helical Interleaver block and the Helical Deinterleaver block, the deinterleaved data lags the original message by

$$CN \left\lceil \frac{s(C-1)}{N} \right\rceil$$

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  $m \cdot C \cdot N$  for some nonnegative integer  $m$ . You can use the Delay block in Signal Processing Blockset to adjust delays manually, if necessary.

## 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 Deinterleaver

---

## **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 condition**

A scalar that fills the array before the first input is placed.

## **Pair Block**

Helical Interleaver

## **See Also**

General Multiplexed Deinterleaver

## **References**

[1] Berlekamp, E. R. and P. Tong. "Improved Interleavers for Algebraic Block Codes." U. S. Patent 4559625, Dec. 17, 1985.

## Purpose

Permute input symbols using helical array

## Library

Convolutional sublibrary of Interleaving

## Description



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  $C*N$  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  $k$ 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  $k$ th group is in row  $1+(k-1)*s$ , 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.

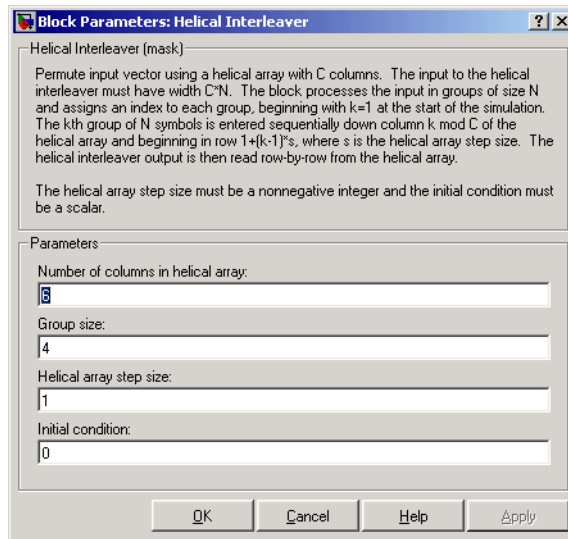
The block sends  $C*N$  symbols from the array to the output port by reading the next  $N$  rows sequentially. At a given time step, the output symbols might be the **Initial condition** parameter value, symbols from that time step's input vector, or symbols left in the array from a previous time step.

The number of elements of the input vector must be  $C$  times  $N$ . If the input is frame-based, then it must be a column vector.

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

## 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 array step size

The number of rows of separation between consecutive input groups in their respective columns of the helical array.

### Initial condition

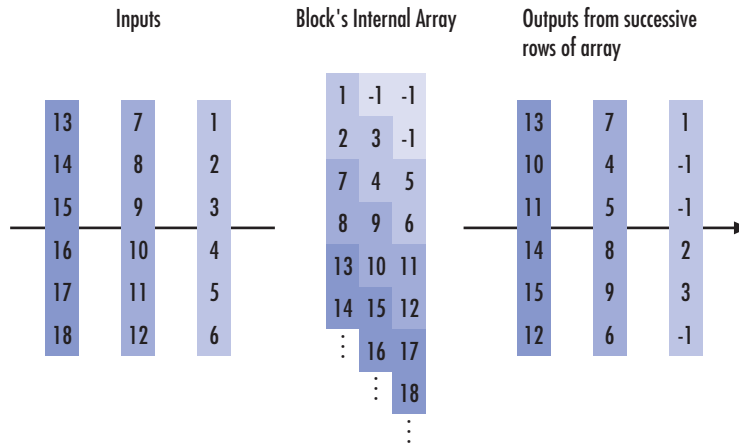
A scalar that fills the array before the first input is placed.

## Examples

Suppose that  $C = 3$ ,  $N = 2$ , 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

## See Also

General Multiplexed Interleaver

## References

[1] Berlekamp, E. R. and P. Tong. "Improved Interleavers for Algebraic Block Codes." U. S. Patent 4559625, Dec. 17, 1985.

# Ideal Rectangular Pulse Filter

---

**Purpose** Shape input signal using ideal rectangular pulses

**Library** Comm Filters

## Description



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.

## Inputs and Outputs

The input can be either a scalar or a frame-based column vector. double, single, and fixed-point data types are supported.

- If the input is sample-based, then the output sample time is  $1/N$  times the input sample time. The output dimensions match the input dimensions. You must set the **Input sampling mode** parameter to Sample-based.
- If the input is a frame-based  $k$ -by-1 matrix, then the output is a frame-based  $k*N$ -by-1 matrix. The output frame period matches the input frame period. You must set the **Input sampling mode** parameter to Frame-based.

The vector size (in frame-based mode), the pulse length, and the pulse delay are mutually independent. They do not need to satisfy any conditions with respect to each other.

## Normalization Methods

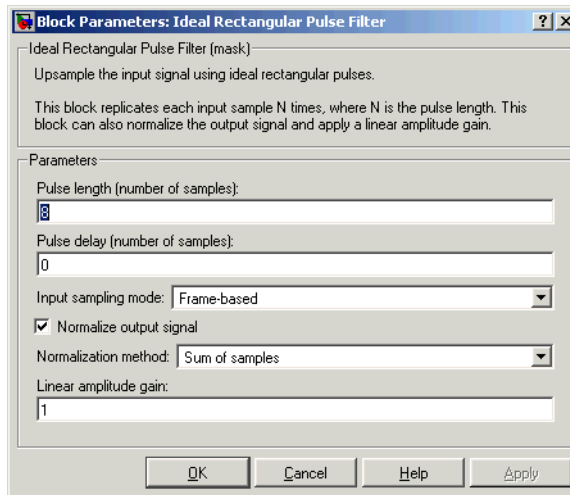
You determine the block's normalization behavior using the **Normalize output signal** and **Linear amplitude gain** parameters.

# Ideal Rectangular Pulse Filter

- If you clear the **Normalize output signal** check box, 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.

## Dialog Box



# Ideal Rectangular Pulse Filter

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## **Pulse length**

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**

The number of zeros that appear in the output at the beginning of the simulation, before the block replicates any input values.

## **Input sampling mode**

The type of input signal: Frame-based or Sample-based.

## **Normalize output signal**

If you select this, then the block scales the set of replicated values before applying the linear amplitude gain.

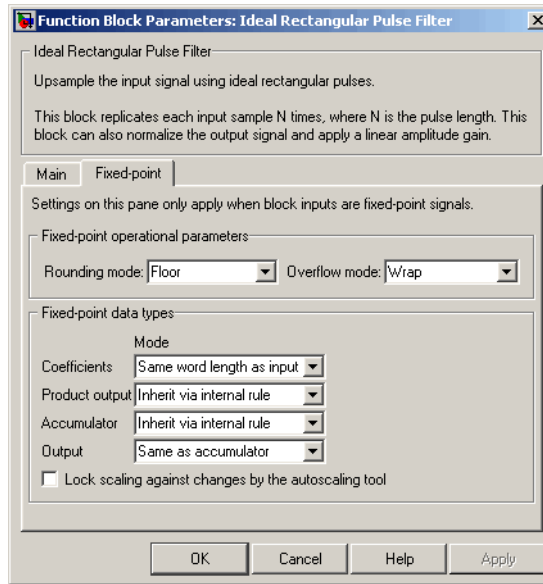
## **Normalization method**

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 output signal**.

## **Linear amplitude gain**

A positive scalar used to scale the output signal.

# Ideal Rectangular Pulse Filter



## Rounding mode

Select the rounding mode for fixed-point operations. The filter coefficients do not obey this parameter; they always round to Nearest.

## 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 *Signal Processing Blockset 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

# Ideal Rectangular Pulse Filter

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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 *Signal Processing Blockset 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.

## 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.

# Ideal Rectangular Pulse Filter

## 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.

## Examples

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 ( <b>Normalize output signal</b> cleared)	1	0, 0, 0, 1, 1, 1, 1, 2, 2, 2, 3, 3, 3, 3,...
None ( <b>Normalize output signal</b> cleared)	10	0, 0, 0, 10, 10, 10, 10, 20, 20, 20, 20, 30, 30, 30, 30,...
Sum of samples	1	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, ..., where $0.25 \times 4 = 1$
Sum of samples	10	0, 0, 0, 2.5, 2.5, 2.5, 2.5, 5, 5, 5, 5, 7.5, 7.5, 7.5, 7.5, ...
Energy per pulse	1	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, ..., where $(0.5)^2 \times 4 = 1^2$
Energy per pulse	10	0, 0, 0, 5, 5, 5, 5, 10, 10, 10, 10, 15, 15, 15, 15, ...



**See Also**      Upsample, Integrate and Dump

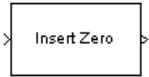
# Insert Zero

---

**Purpose** Distribute input elements in output vector

**Library** Sequence Operations

## Description



The Insert Zero block constructs an output vector by inserting zeros among the elements of the input vector. The input can be real or complex. The block determines where to place the zeros by using the **Insert zero vector** parameter. The **Insert zero vector** parameter is a binary vector whose elements are arranged so that:

- Each 1 indicates that the block should place the *next* element of the input in the output vector
- Each 0 indicates that the block should place a 0 in the output vector

If the input signal is sample-based, then the input vector length must equal the number of 1s in the **Insert zero vector** parameter.

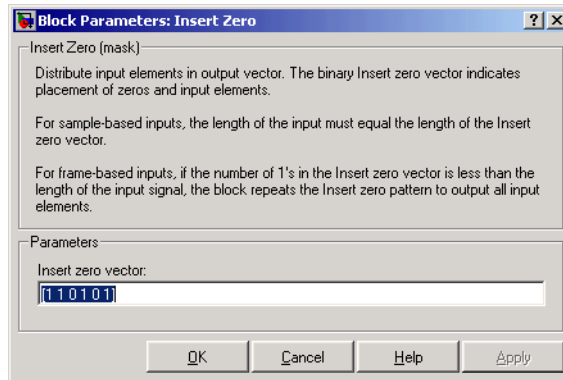
The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, `double`, and fixed-point. The data type of this output will be the same as that of the input signal.

To implement punctured coding using the Puncture and Insert Zero blocks, you should use the same vector for the **Insert zero vector** parameter in this block and for the **Puncture vector** parameter in the Puncture block.

## Frame-Based Processing

If the input signal is frame-based, then both it and the **Insert zero vector** parameter must be column vectors. The number of 1s in the **Insert zero vector** parameter must divide the input vector length. If the input vector length is greater than the number of 1s in the **Insert zero vector** parameter, then the block repeats the insertion pattern until it has placed all input elements in the output vector.

## Dialog Box



### Insert zero vector

A binary vector whose pattern of 0s and 1s indicates where the block should place either 0s or input vector elements, respectively, in the output vector.

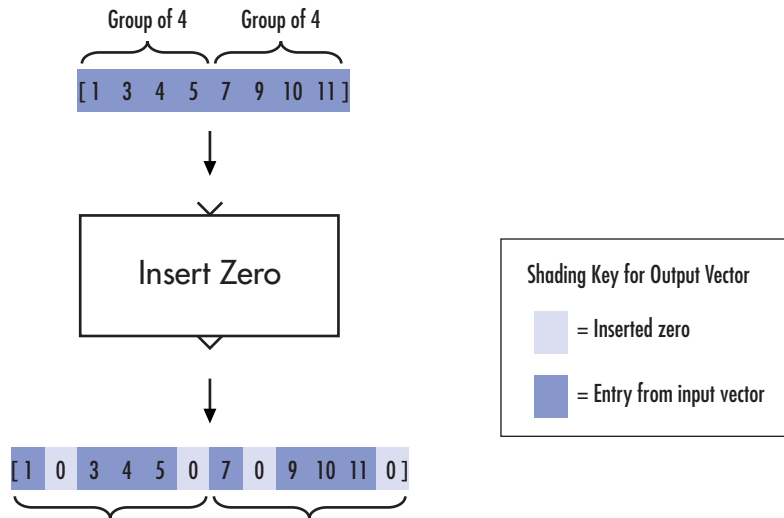
## Examples

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 1s.

The diagram below depicts the block's operation using this **Insert zero vector** parameter. Notice that the insertion pattern applies twice.

# Insert Zero

---



Compare this example with that on the reference page for the Puncture block.

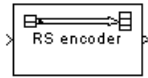
## See Also

Puncture

**Purpose** Create Reed-Solomon code from integer vector data

**Library** Block sublibrary of Channel Coding

## Description



The Integer-Input RS Encoder block creates a Reed-Solomon code with message length  $K$  and codeword length  $N$ . You specify both  $N$  and  $K$  directly in the block dialog. The symbols for the code are integers between 0 and  $2^M - 1$ , which represent elements of the finite field  $GF(2^M)$ . Restrictions on  $M$  and  $N$  are described in “Restrictions on  $M$  and the Codeword Length  $N$ ” on page 2-286 below. The difference  $N - K$  must be an even integer.

The input and output are integer-valued signals that represent messages and codewords, respectively. The input must be a frame-based column vector whose length is an integer multiple of  $K$ . The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. The output is a frame-based column vector whose length is the same integer multiple of  $N$ , and whose data type is inherited from the input. For more information on representing data for Reed-Solomon codes, see the section “Integer Format (Reed-Solomon Only)” in *Communications Blockset User’s Guide*.

The default value of  $M$  is the smallest integer that is greater than or equal to  $\log_2(N+1)$ , that is,  $\text{ceil}(\log_2(N+1))$ . You can change the value of  $M$  from the default by specifying the primitive polynomial for  $GF(2^M)$ , as described in “Specifying the Primitive Polynomial” on page 2-285 below. If  $N$  is less than  $2^M - 1$ , the block uses a shortened Reed-Solomon code.

An  $(N, K)$  Reed-Solomon code can correct up to  $\text{floor}((N-K)/2)$  symbol errors (*not* bit errors) in each codeword.

## Specifying the Primitive Polynomial

You can specify the primitive polynomial that defines the finite field  $GF(2^M)$ , 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  $GF(2)$  of degree  $M$ , in descending

# Integer-Input RS Encoder

---

order of powers. For example, to specify the polynomial  $x^3+x+1$ , enter the vector [1 0 1 1].

If you do not select **Specify primitive polynomial**, the block uses the default primitive polynomial of degree  $M = \text{ceil}(\log_2(N+1))$ . You can display the default polynomial by entering `primpoly(ceil(log2(N+1)))` at the MATLAB prompt.

## Restrictions on M and the Codeword Length 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  $3 < N < 2^{16}-1$ .
- If you do select **Specify primitive polynomial**,  $N$  must lie in the range  $3 \leq N < 2^M-1$  and  $M$  must lie in the range  $3 \leq M \leq 16$ .

## 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  $2^M-1$ . The vector represents a polynomial, in descending order of powers, whose coefficients are elements of  $\text{GF}(2^M)$  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

$$g(x) = (x+A^b)(x+A^{b+1})(x+A^{b+2})\dots(x+A^{b+N-K-1})$$

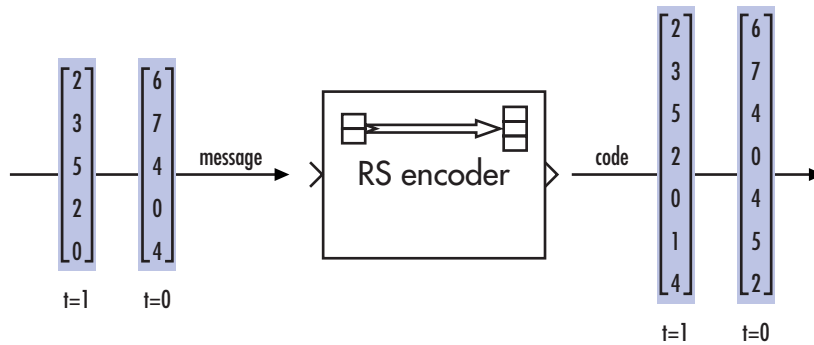
where  $A$  is the primitive element of the Galois field over which the input message is defined, and  $b$  is an integer.

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(N1,K1)`, where  $N1 = 2^M-1$  and  $K1 = K+(N1-N)$ , 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(N1,K1,poly)`.

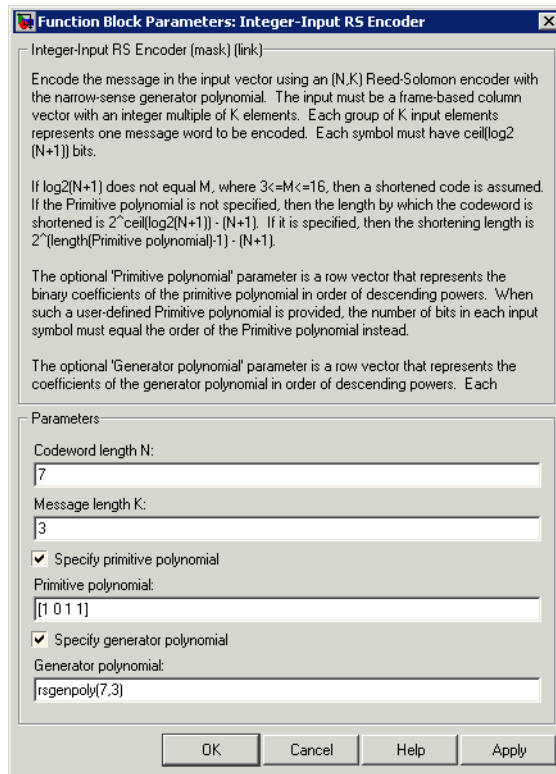
## Examples

Suppose  $M = 3$ ,  $N = 2^3 - 1 = 7$ , and  $K = 5$ . 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 N** is set to 7, **Message length K** is set to 5, and the default primitive and generator polynomials are used.



# Integer-Input RS Encoder

## Dialog Box



### Codeword length 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 field is available only when **Specify primitive polynomial** is selected.



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**

This field is available only when **Specify generator polynomial** is selected.

Integer row vector, whose entries are in the range from 0 to  $2^M-1$ , representing the generator polynomial in descending order of powers.

**Pair Block** Integer-Output RS Decoder

**See Also** Binary-Input RS Encoder

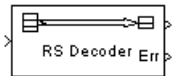
# Integer-Output RS Decoder

---

**Purpose** Decode Reed-Solomon code to recover integer vector data

**Library** Block sublibrary of Channel Coding

## Description



The Integer-Output RS Decoder block recovers a message vector from a Reed-Solomon codeword vector. For proper decoding, the parameter values in this block should match those in the corresponding Integer-Input RS Encoder block.

The Reed-Solomon code has message length  $K$  and codeword length  $N$ . You specify both  $N$  and  $K$  directly in the block dialog. The symbols for the code are integers between 0 and  $2^M-1$ , which represent elements of the finite field  $GF(2^M)$ . Restrictions on  $M$  and  $N$  are described in “Restrictions on  $M$  and the Codeword Length  $N$ ” on page 2-286 below. The difference  $N - K$  must be an even integer.

The input and output are integer-valued signals that represent messages and codewords, respectively. The input must be a frame-based column vector whose length is an integer multiple of  $K$ . The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. The output is a frame-based column vector whose length is the same integer multiple of  $N$ , and whose data type is inherited from the input. For more information on representing data for Reed-Solomon codes, see the section “Integer Format (Reed-Solomon Only)” in *Communications Blockset User’s Guide*.

The default value of  $M$  is  $\text{ceil}(\log_2(N+1))$ , 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  $GF(2^M)$ , as described in “Specifying the Primitive Polynomial” on page 2-285 below. If  $N$  is less than  $2^M-1$ , 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 2-286.

An  $(N, K)$  Reed-Solomon code can correct up to  $\text{floor}((N-K)/2)$  symbol errors (*not* bit errors) in each codeword.

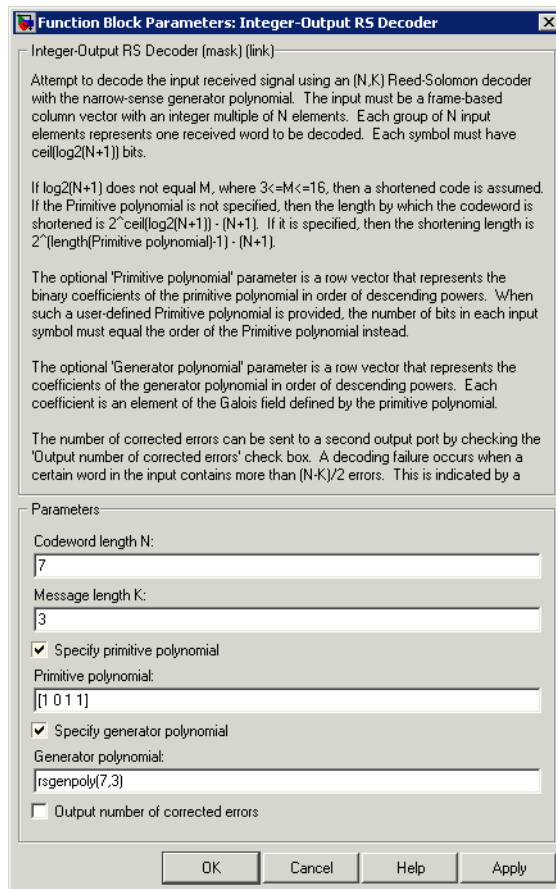
The second output is the number of errors detected during decoding of the codeword. A -1 indicates that the block detected more errors than it could correct using the coding scheme. An (N,K) Reed-Solomon code can correct up to  $\text{floor}((N-K)/2)$  symbol errors (*not* bit errors) in each codeword. 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.

The sample times of the input and output signals are equal.

# Integer-Output RS Decoder

## Dialog Box



### Codeword length 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 field is available only when **Specify primitive polynomial** is selected.

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

This field is available only when **Specify generator polynomial** is selected.

Integer row vector, whose entries are in the range from 0 to  $2^M-1$ , representing the generator polynomial in descending order of powers.

## 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.

## Algorithm

This block uses the Berlekamp-Massey decoding algorithm. For information about this algorithm, see the references listed below.

## Pair Block

Integer-Input RS 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.

## See Also

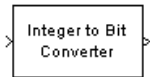
Binary-Output RS Decoder

# Integer to Bit Converter

**Purpose** Map vector of integers to vector of bits

**Library** Utility Blocks

## Description

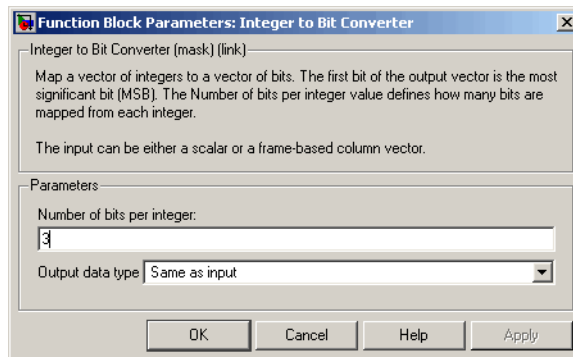


The Integer to Bit Converter block maps each integer in the input vector to a group of bits in the output vector. If  $M$  is the **Number of bits per integer** parameter, then the input integers must be between 0 and  $2^M-1$ . The block maps each integer to a group of  $M$  bits, using the first bit as the most significant bit. As a result, the output vector length is  $M$  times the input vector length.

The input can be either a scalar or a frame-based column vector.

The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`.

## 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 31.

### Output data type

The output data type can be set to `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, or `double`. If this field is set to `Same as input`, the output data type will be inherited from the input signal.

## Examples

If the input is [7; 13] and the **Number of bits per integer** parameter is 4, then the output is [0; 1; 1; 1; 1; 1; 0; 1]. The first group of four bits (0, 1, 1, 1) represents 7 and the second group of four bits (1, 1, 0, 1) represents 13. Notice that the output length is four times the input length.

## Pair Block

Bit to Integer Converter

# Integrate and Dump

---

**Purpose** Integrate discrete-time signal, resetting to zero periodically

**Library** Comm Filters

## Description



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.

This block supports inputs and outputs of type double, single, and fixed-point. The port data types are inherited from the signals that drive the block.

The integrate-and-dump operation is often used in a receiver model when the system's transmitter uses a simple square-pulse model. It can also be used in fiber optics and in spread-spectrum communication systems such as CDMA (code division multiple access) applications.

The input can be either a scalar or a frame-based matrix. If the input is frame-based, then it must have  $k \cdot N$  rows for some positive integer  $k$ , and the block processes each column independently.

The output contents, dimensions, and sample time are affected by the **Output intermediate values** check box, as follows:

- If you clear the check box, then the block outputs the cumulative sum at each reset time.
  - If the input is sample-based, 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 frame-based  $(k \cdot N)$ -by- $n$  matrix, then the output is  $k$ -by- $n$ . In this case, the block experiences no delay and the output frame period matches the input frame period.



- If you select the check box, then the block outputs the cumulative sum at each time step, including the reset times. The output has the same sample time and the same matrix dimensions as the input.

This block will work within a triggered subsystem, as long as it is used in the single-rate mode.

## 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 frame-based matrix with  $n$  columns and the **Offset** parameter is a length- $n$  vector, then the  $m$ th element of the **Offset** vector is the offset for the  $m$ 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.

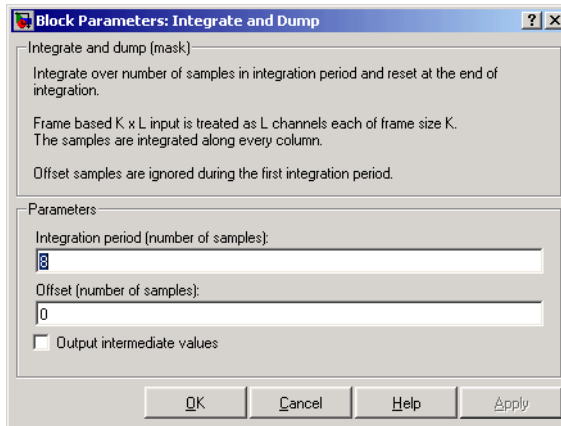
When the **Output intermediate values** check box is cleared, the block's output is delayed, relative to its input, throughout the simulation:

- If the input is sample-based, 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  $m$ th integration period in the output sample indexed by  $m+1$ .
- If the input is frame-based 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 2-299.

# Integrate and Dump

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## 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 suppresses the intermediate cumulative sums between successive resets.

## 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,...) 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	Sample-based 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 sample-based input.
Cleared	Frame-based 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 frame-based scalar.
Selected	Sample-based scalar	0, 0, 0, 4, 4+5, 4+5+6, 4+5+6+7, 8, 8+9, 8+9+10, 8+9+10+11, and 12, where the three 0s are initial transient values.
Selected	Frame-based column vector of length 4	0, 0, 0, 4, 4+5, 4+5+6, 4+5+6+7, 8, 8+9, 8+9+10, 8+9+10+11, and 12, where the three 0s are initial transient values. The output is a frame-based column vector of length 4.

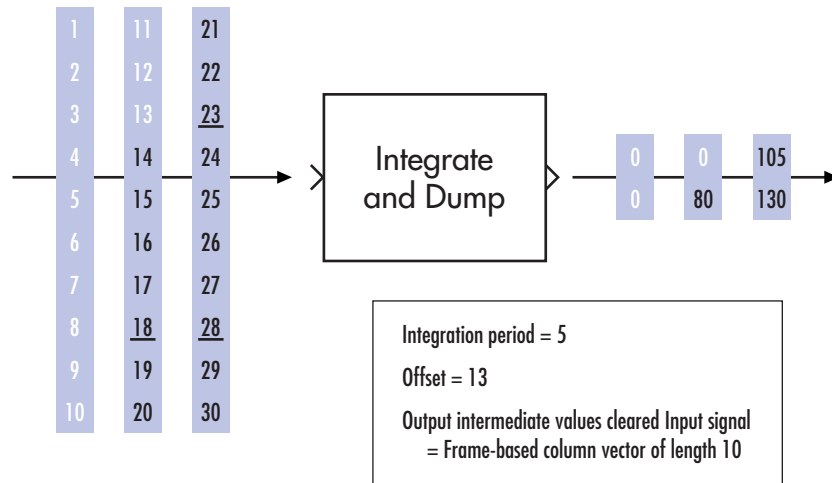
In all cases, the block discards the first three input samples (1, 2, and 3).

### 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.

# Integrate and Dump

The figure also indicates how the input and output values are organized as frame-based 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  $\text{ceil}(13/5)$  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), Ideal Rectangular Pulse Filter

**Purpose** Alternately select elements from two input vectors to generate output vector

**Library** Sequence Operations

## Description

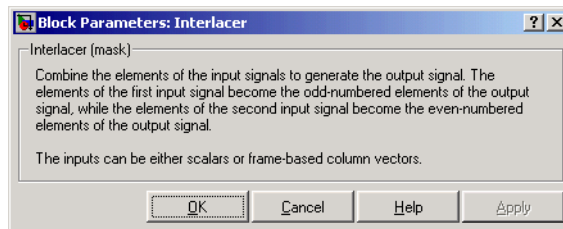


The Interlacer block accepts two inputs that have the same vector size, complexity, and sample time. It produces one output vector by alternating elements from the first input (labeled O for odd) and from the second input (labeled E for even) . As a result, the output vector size is twice that of either input. The output vector has the same complexity and sample time of the inputs.

The inputs can be either scalars or frame-based column vectors. 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 signals.

This block can be useful for combining in-phase and quadrature information from separate vectors into a single vector.

## Dialog Box



**Examples** If the two input vectors are frame-based with 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)

# I/Q Imbalance

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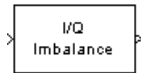
## Purpose

Create complex baseband model of signal impairments caused by imbalances between in-phase and quadrature receiver components

## Library

RF Impairments

## Description

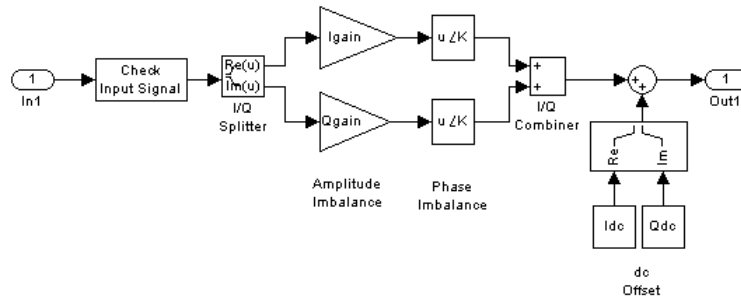


The I/Q Imbalance block creates a complex baseband model of the signal impairments caused by imbalances between in-phase and quadrature receiver components. Typically, these are caused by differences in the physical channels for the two components of the signal.

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 **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 **Look under mask**:



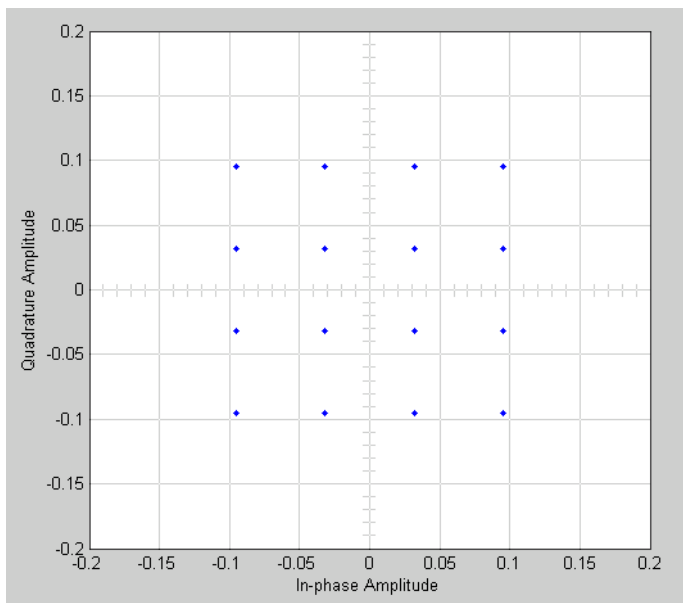
The value of the **I/Q amplitude imbalance (dB)** parameter is divided between the in-phase and quadrature components:

- If you enter a positive value  $X$  for the **I/Q amplitude imbalance (dB)**, the block applies a gain of  $+X/2$  dB to the in-phase component and a gain of  $-X/2$  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  $-X/2$  dB to the in-phase component and a gain of  $+X/2$  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:

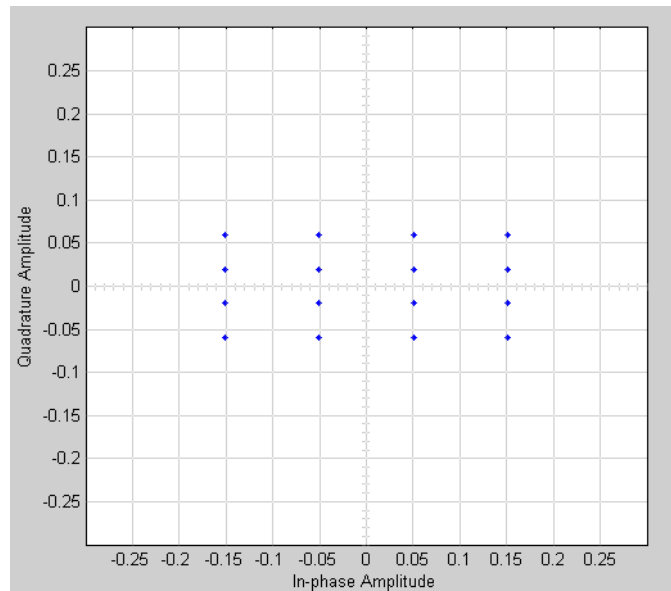
# I/Q Imbalance

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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:

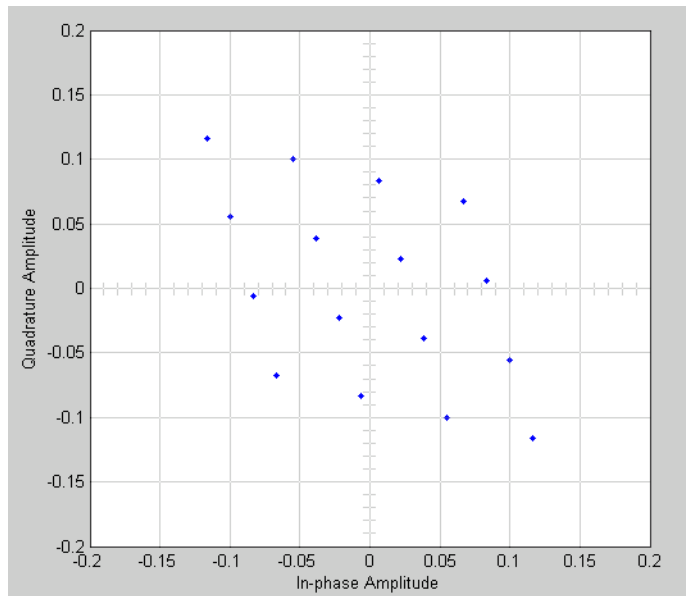




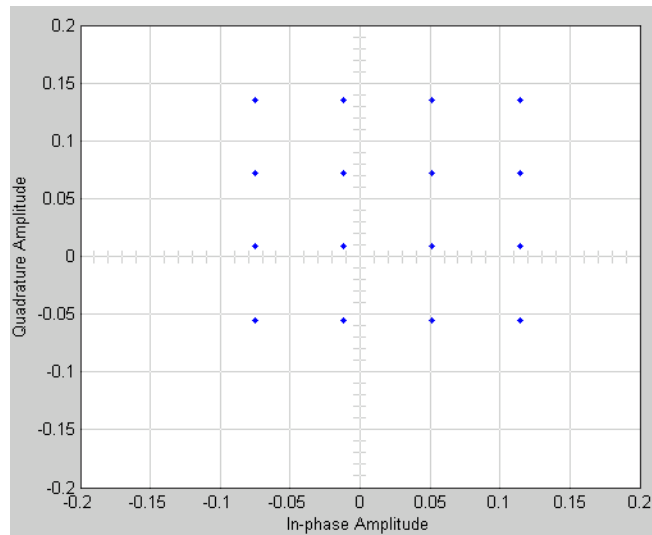
Observe that the scatter plot is stretched horizontally and compressed vertically compared to the undistorted constellation.

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

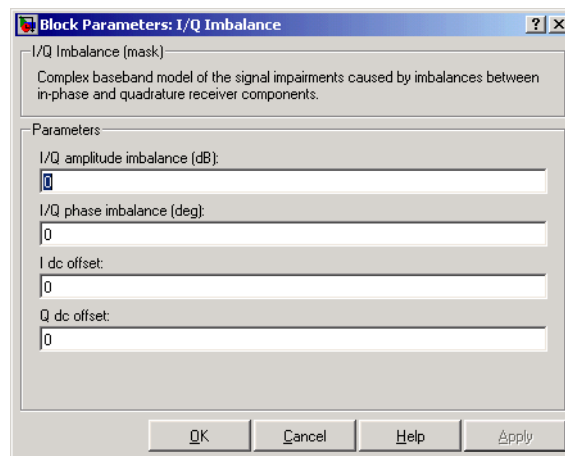


Setting the **I dc offset** to 0.02 and the **Q dc offset** to 0.04 shifts the constellation 0.02 to the right and 0.04 up, as shown below:



See “Scatter Plot Examples” for a description of the model that generates this plot.

## Dialog Box



# I/Q Imbalance

---

**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

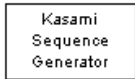
## Purpose

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 set of Kasami sequences. The Kasami sequences are a set of sequences that have good cross-correlation properties.

There are two classes 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  $N = 2^n - 1$ , 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.

$$K_s(u, n, m) = \begin{cases} u & m = -1 \\ u \oplus T^m w & m = 0, \dots, 2^{n/2} - 2 \end{cases}$$

### Small Set of Kasami Sequences for $n$ Even

Note that the small set contains  $2^{n/2}$  sequences.

For  $\text{mod}(n, 4) = 2$ , the large set of Kasami sequences is defined as follows. Let  $v$  be the sequence formed by decimating the sequence  $u$  by  $2^{n/2} + 1$ . 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.

# Kasami Sequence Generator

$$K_L(u, n, k, m) = \begin{cases} u & k = -2; m = -1 \\ v & k = -1; m = -1 \\ u \oplus T^k v & k = 0, \dots, 2^n - 2; m = -1 \\ u \oplus T^m w & k = -2; m = 0, \dots, 2^{n/2} - 2 \\ v \oplus T^m w & k = -1; m = 0, \dots, 2^{n/2} - 2 \\ u \oplus T^k v \oplus T^m w & k = 0, \dots, 2^n - 2; m = 0, \dots, 2^{n/2} - 2 \end{cases}$$

## Large Set of Kasami Sequences for mod(n, 4) = 2

The sequences described in the first three rows of the preceding figure correspond to the Gold sequences for mod(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

$$t(n) = 1 + 2^{(n+2)/2}, n \text{ even}$$

$$s(n) = \frac{1}{2}(t(n) + 1)$$

## 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,  $[1\ 0\ 0\ 0\ 0\ 0\ 1\ 0\ 1]$  and  $[8\ 2\ 0]$  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  $[-1, \dots, 2^{n/2} - 2]$ . The following table describes the output sequences corresponding to **Sequence index**  $m$ :

Sequence Index	Range of Indices	Output Sequence
-1	$m = -1$	$u$
$m$	$m = 0, \dots, 2^{n/2} - 2$	$u \oplus T^m w$

- To generate sequences from the large set, for  $\text{mod}(n, 4) = 2$ , where  $n$  is the degree of the **Generator polynomial**, you can specify **Sequence index** as an integer vector  $[k\ m]$ . In this case, the output sequence is from the large set. The range for  $k$  is  $[-2, \dots, 2^n - 2]$ , and the range for  $m$  is  $[-1, \dots, 2^{n/2} - 2]$ . The following table describes the output sequences corresponding to **Sequence index**  $[k\ m]$ :

# Kasami Sequence Generator

Sequence Index [k m]	Range of Indices	Output Sequence
[-2 -1]	$k = -2, m = -1$	u
[-1 -1]	$k = -1, m = -1$	v
[k -1]	$k = 0, 1, \dots, 2^n - 2$ $m = -1$	$u \oplus T^k v$
[-2 m]	$k = -2$ $m = 0, 1, \dots, 2^{n/2} - 2$	$u \oplus T^m w$
[-1 m]	$k = -1$ $m = 0, \dots, 2^{n/2} - 2$	$v \oplus T^m w$
[k m]	$k = 0, \dots, 2^n - 2$ $m = 0, \dots, 2^{n/2} - 2$	$u \oplus T^k v \oplus T^m w$

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 the **Reset on nonzero input** check box. 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 2-478 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.



# Kasami Sequence Generator

n	N	Polynomial	Set
4	15	[4 1 0]	Small
6	63	[6 1 0]	Large
8	255	[8 4 3 2 0]	Small
10	1023	[10 3 0]	Large
12	4095	[12 6 4 1 0]	Small

## Dialog Box

**Source Block Parameters: Kasami Sequence Generator**

Kasami Sequence Generator (mask) (link)

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 initial states parameter is a binary vector that represents the starting state of the shift register.

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.

The shift parameter is a scalar integer that produces an offset in the sequence.

Parameters

Generator polynomial:  
[1 0 0 0 1 1]

Initial states:  
[0 0 0 0 1]

Sequence index(es):  
0

Shift:  
0

Sample time:  
1

Frame-based outputs

Samples per frame:  
1

Reset on nonzero input

Output data type: double

OK Cancel Help

# Kasami Sequence Generator

---

Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

## **Generator polynomial**

Binary vector specifying the generator polynomial for the sequence  $u$ .

## **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.

## **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 the **Frame-based outputs** check box.

## **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

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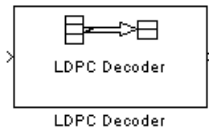
## Purpose

Decode binary low-density parity-check code specified by parity-check matrix

## Library

Block sublibrary of Error Detection and Correction

## Description



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 user-specified number of iterations or run until all parity-checks are satisfied.
- Output hard decisions or soft decisions (log-likelihood ratios) for decoded bits.

The LDPC Decoder block uses the same algorithm as the `fec.ldpcdec` Communications Toolbox object. See “Decoding Algorithm” for details.

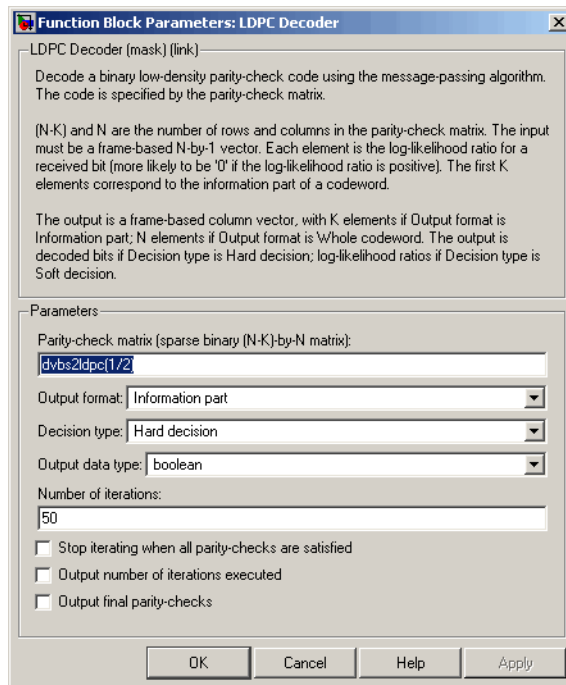
$(n - k)$  and  $n$  are the number of rows and columns, respectively, in the parity-check matrix.

The input must be a frame-based  $n \times 1$  vector and a real-valued 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.

Type `commdvbs2` at the command line to see a demo using this block.

## Dialog Box



### Parity-check matrix

This block can accept a sparse matrix with dimension  $n - k$  by  $n$  ( $n > k > 0$ ) of real numbers. All nonzero elements must be equal to 1.  $n$  and  $k$  are only limited by the amount of available memory.

### Output format

The options are Information part and Whole codeword.

The output is a real-valued signal in the form of a frame-based column vector. It has  $k$  elements if the output format is set to Information part, and  $n$  elements if set to whole codeword.

### Decision type

The options are Hard decision and Soft decision.

# LDPC Decoder

---

If this is set to `Hard decision`, the output is decoded bits (of type `double` or `boolean`). If set 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.

## **Pair Block**

LDPC Encoder

## **References**

[1] Gallager, Robert G., *Low-Density Parity-Check Codes*, Cambridge, MA, MIT Press, 1963.

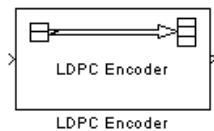
## Purpose

Encode binary low-density parity-check code specified by parity-check matrix

## Library

Block sublibrary of Error Detection and Correction

## Description



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.

The LDPC Encoder block uses the same algorithm as the `fec.ldpcenc` Communications Toolbox object. See “LDPC Encoding Method” for details.

$(n - k)$  and  $n$  are the number of rows and columns, respectively, in the parity-check matrix.

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  $k \times 1$  frame-based signal.

The output inherits the data type of the input, and the input must be binary-valued (0 or 1).

Type `commdvbs2` at the command line to see a demo using this block.

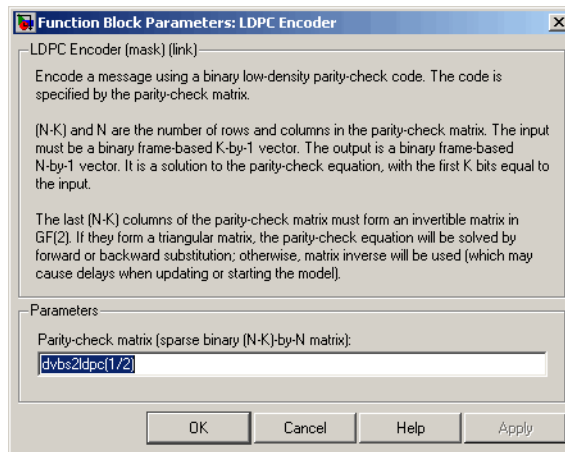
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**Note** Model initialization or update may take a long time, because a large matrix may need to be inverted (when the last  $(n - k)$  columns of the parity-check matrix is not triangular).

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# LDPC Encoder

## Dialog Box



### Parity-check matrix

This block can accept a sparse matrix with dimension  $n - k$  by  $n$  ( $n > k > 0$ ) of real numbers. All nonzero elements must be equal to 1.  $n$  and  $k$  are only limited by the amount of available memory.

The default value is the parity-check matrix of the half-rate LDPC code from the DVB-S.2 standard.

## Pair Block

LDPC Decoder



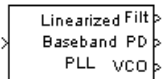
## Purpose

Implement linearized version of baseband phase-locked loop

## Library

Components sublibrary of Synchronization

## Description



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. 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.

The input signal represents the received signal. The input must be a sample-based scalar signal. The three output ports produce:

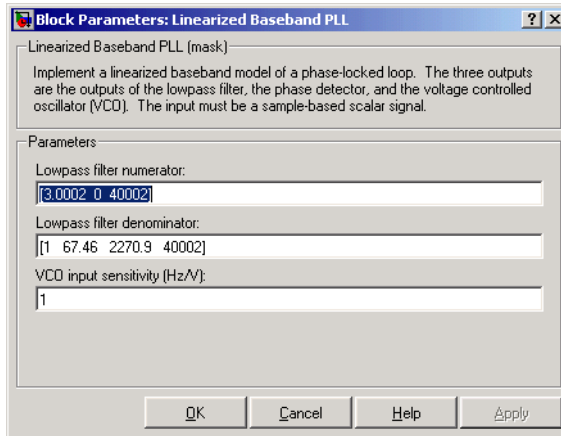
- The output of the filter

# Linearized Baseband PLL

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- The output of the phase detector
- The output of the VCO

## Dialog Box



### 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  $s$ .

### 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  $s$ .

### VCO input sensitivity (Hz/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 Blockset User's Guide*.

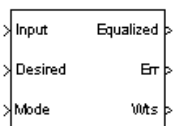
## Purpose

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 baseband signal through a dispersive channel. During the simulation, the block uses the 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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 “Retrieving the Weights and Error Signal” in *Communications Blockset User’s Guide*.

# LMS Decision Feedback Equalizer

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## 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 Blockset User’s Guide*.

## Equalizer Delay

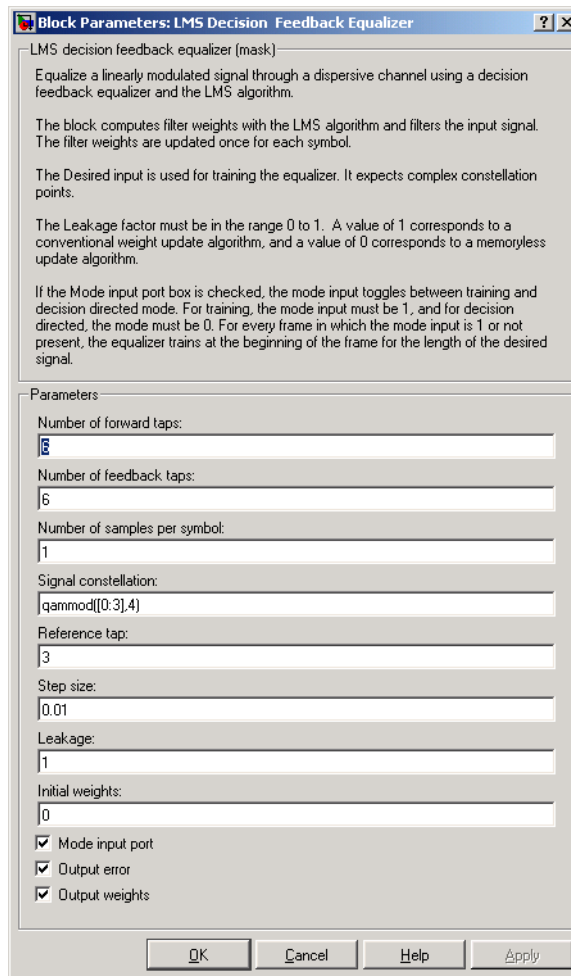
For proper equalization, you should set the **Reference tap** parameter so that it exceeds the delay, in symbols, between the transmitter’s modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer *output* is equal to

$$1 + (\text{Reference tap} - 1) / (\text{Number of samples per symbol})$$

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

## Dialog Box



### Number of forward taps

The number of taps in the forward filter of the decision feedback equalizer.

# LMS Decision Feedback Equalizer

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**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 check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

**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.

**Output weights**

If you check this 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.
- [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

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

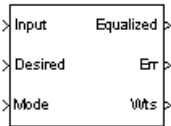
# LMS Linear Equalizer

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**Purpose** Equalize using linear equalizer that updates weights with LMS algorithm

**Library** Equalizers

**Description** 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. 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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 “Retrieving the Weights and Error Signal” in *Communications Blockset 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 Blockset User’s Guide*.

## Equalizer Delay

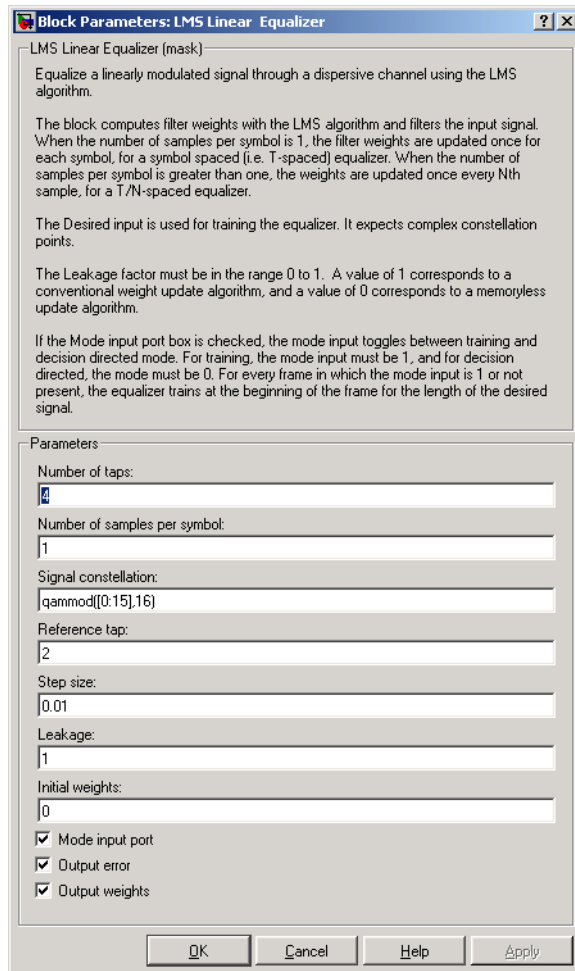
For proper equalization, you should set the **Reference tap** parameter so that it exceeds the delay, in symbols, between the transmitter’s modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer *output* is equal to

$$1 + (\text{Reference tap} - 1) / (\text{Number of samples per symbol})$$

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.

**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 check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

**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.

**Output weights**

If you check this box, the block outputs the current weights.

**Examples**

See “Example: LMS Linear Equalizer” and the Adaptive Equalization demo.

**References**

[1] Farhang-Boroujeny, B., *Adaptive Filters: Theory and Applications*, Chichester, England, Wiley, 1998.

# LMS Linear Equalizer

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[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

LMS Decision Feedback Equalizer, Normalized LMS Linear Equalizer, Sign LMS Linear Equalizer, Variable Step LMS Linear Equalizer, RLS Linear Equalizer, CMA Equalizer

## Purpose

Permute input symbols by filling matrix by columns and emptying it by rows

## Library

Block sublibrary of Interleaving

## Description

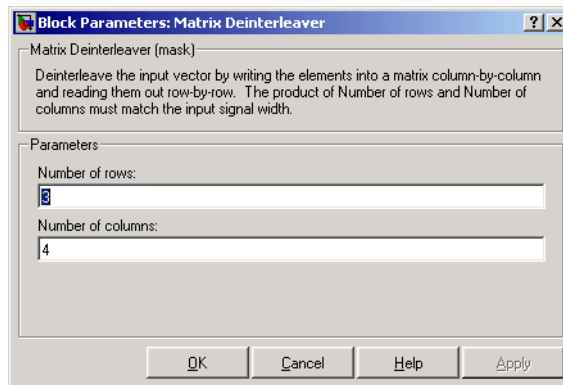


The Matrix Deinterleaver block performs block deinterleaving by filling a matrix with the input symbols column by column and then sending the matrix contents to the output port row by row. The **Number of rows** and **Number of columns** parameters are the dimensions of the matrix that the block uses internally for its computations.

The length of the input vector must be **Number of rows** times **Number of columns**. If the input is frame-based, then it must be a column vector.

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.

## Dialog Box



### Number of rows

The number of rows in the matrix that the block uses for its computations.

# Matrix Deinterleaver

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## **Number of columns**

The number of columns in the matrix that the block uses for its computations.

## **Examples**

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].

## **Pair Block**

Matrix Interleaver

## **See Also**

General Block Deinterleaver

# Matrix Helical Scan Deinterleaver

**Purpose** Restore ordering of input symbols by filling matrix along diagonals

**Library** Block sublibrary of Interleaving

## Description



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.

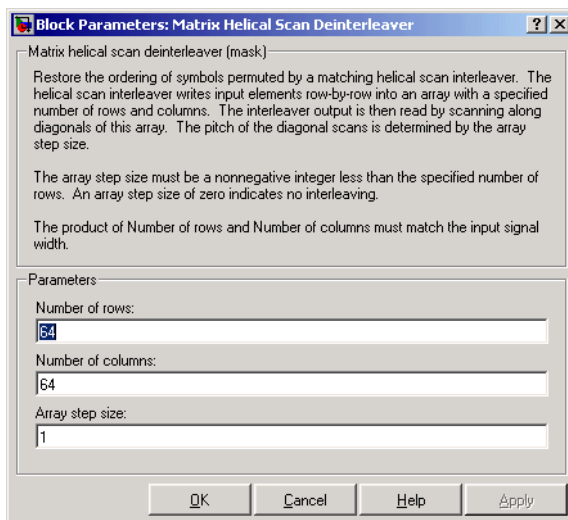
The **Array step size** parameter is the slope of each diagonal, that is, the amount by which the row index increases as the column index increases by one. This parameter must be an integer between zero and the **Number of rows** parameter. If the **Array step size** parameter is zero, then the block does not deinterleave and the output is the same as the input.

The number of elements of the input vector must be the product of **Number of rows** and **Number of columns**. If the input is frame-based, then it must be a column vector.

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.

# 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.

## Pair Block

Matrix Helical Scan Interleaver

## See Also

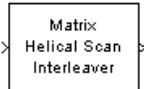
General Block Deinterleaver



**Purpose** Permute input symbols by selecting matrix elements along diagonals

**Library** Block sublibrary of Interleaving

## Description



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.

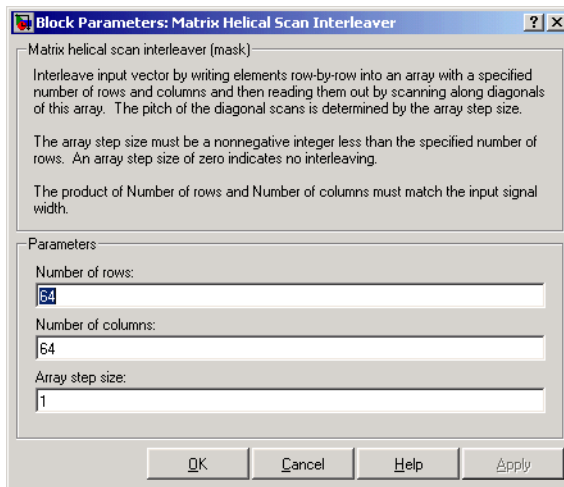
The **Array step size** parameter is the slope of each diagonal, that is, the amount by which the row index increases as the column index increases by one. This parameter must be an integer between zero and the **Number of rows** parameter. If the **Array step size** parameter is zero, then the block does not interleave and the output is the same as the input.

The number of elements of the input vector must be the product of **Number of rows** and **Number of columns**. If the input is frame-based, then it must be a column vector.

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.

# Matrix Helical Scan 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.

### Array step size

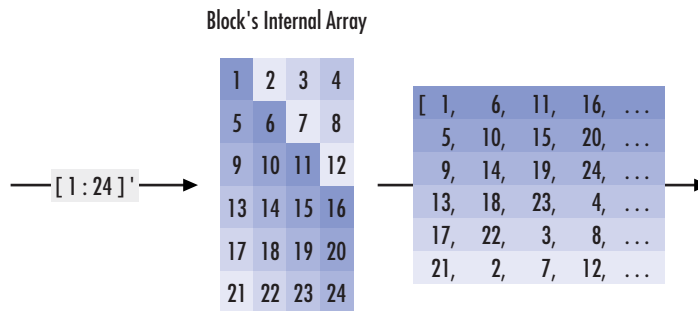
The slope of the diagonals that the block reads.

## Examples

If the **Number of rows** and **Number of columns** parameters are 6 and 4, respectively, then the interleaver uses a 6-by-4 matrix for its internal computations. If the **Array step size** parameter is 1, then the diagonals are as shown in the figure below. Positions with the same color form part of the same diagonal, and diagonals with darker colors precede those with lighter colors in the output signal.

Given an input signal of  $[1:24]'$ , the block produces an output of

# Matrix Helical Scan Interleaver



[1; 6; 11; 16; 5; 10; 15; 20; 9; 14; 19; 24; 13; 18; 23; ...  
4; 17; 22; 3; 8; 21; 2; 7; 12]

## Pair Block

Matrix Helical Scan Deinterleaver

## See Also

General Block Interleaver

# Matrix Interleaver

## Purpose

Permute input symbols by filling matrix by rows and emptying it by columns

## Library

Block sublibrary of Interleaving

## Description



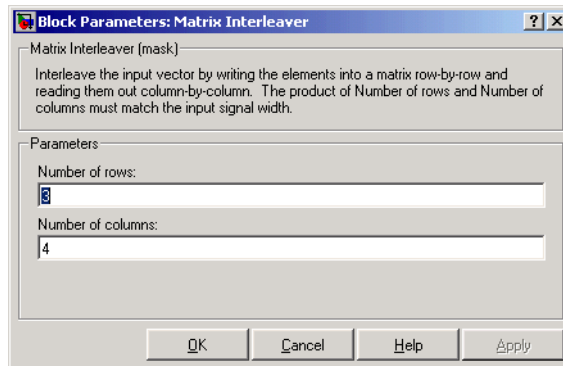
The Matrix Interleaver block performs block interleaving by filling a matrix with the input symbols row by row and then sending the matrix contents to the output port column by column.

The **Number of rows** and **Number of columns** parameters are the dimensions of the matrix that the block uses internally for its computations.

The number of elements of the input vector must be the product of **Number of rows** and **Number of columns**. If the input is frame-based, then it must be a column vector.

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.

## 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**

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**

Matrix Deinterleaver

**See Also**

General Block Interleaver

# M-DPSK Demodulator Baseband

---

**Purpose** Demodulate DPSK-modulated data

**Library** PM, in Digital Baseband sublibrary of Modulation

## Description



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. The input can be either a scalar or a frame-based column vector. The block accepts the input data types `single` and `double`.

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.

### Binary or Integer Outputs

If the **Output type** parameter is set to `Integer`, then the block demodulates a phase difference of

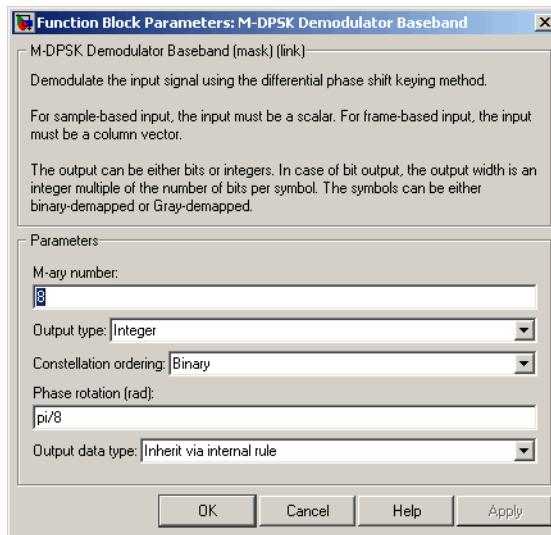
$$\theta + 2\pi k/M$$

to  $k$ , where  $\theta$  is the **Phase rotation** parameter and  $k$  is an integer between 0 and  $M-1$ .

If the **Output type** parameter is set to `Bit` and the **M-ary number** parameter has the form  $2^K$  for some positive integer  $K$ , then the block outputs binary representations of integers between 0 and  $M-1$ . It outputs a group of  $K$  bits, called a binary *word*, for each symbol.

In binary output mode, 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.

## 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.

### Phase rotation (rad)

The phase difference between the previous and current modulated symbols when the input is 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.

# M-DPSK Demodulator Baseband

---

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`.

**Pair Block** M-DPSK Modulator Baseband

**See Also** DBPSK Demodulator Baseband, DQPSK Demodulator Baseband, M-PSK Demodulator 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.



**Purpose** Modulate using M-ary differential phase shift keying method

**Library** PM, in Digital Baseband sublibrary of Modulation

## Description



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`.

## Inputs and Constellation Types

If the **Input type** parameter is set to `Integer`, then valid input values are integers between 0 and  $M-1$ . In this case, the input can be either a scalar or a frame-based column vector. If the first input is  $k_1$ , then the modulated symbol is

$$\exp\left(j\theta + j2\pi\frac{k_1}{m}\right)$$

where  $\theta$  is the **Phase rotation** parameter. If a successive input is  $k$ , then the modulated symbol is

$$\exp\left(j\theta + j2\pi\frac{k}{m}\right) \cdot (\text{previous modulated symbol})$$

If the **Input type** parameter is set to `Bit` and the **M-ary number** parameter has the form  $2^K$  for some positive integer  $K$ , then the block accepts binary representations of integers between 0 and  $M-1$ . It modulates each group of  $K$  bits, called a binary *word*. The input can be either a vector of length  $K$  or a frame-based column vector whose length is an integer multiple of  $K$ .

## M-DPSK Modulator Baseband

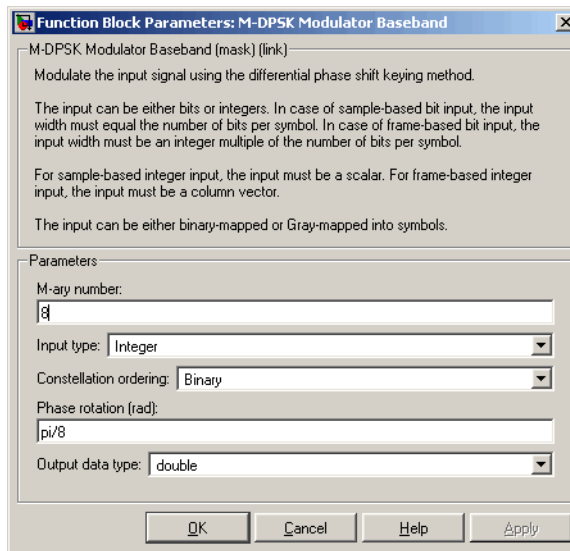
---

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 table below 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 Phase Difference	Gray-Coded Phase Difference
[0 0 0]	$j\theta$	$j\theta$
[0 0 1]	$j\theta + j\pi/4$	$j\theta + j\pi/4$
[0 1 0]	$j\theta + j\pi 2/4$	$j\theta + j\pi 3/4$
[0 1 1]	$j\theta + j\pi 3/4$	$j\theta + j\pi 2/4$
[1 0 0]	$j\theta + j\pi 4/4$	$j\theta + j\pi 7/4$
[1 0 1]	$j\theta + j\pi 5/4$	$j\theta + j\pi 6/4$
[1 1 0]	$j\theta + j\pi 6/4$	$j\theta + j\pi 4/4$
[1 1 1]	$j\theta + j\pi 7/4$	$j\theta + j\pi 5/4$

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.

### 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  $2^K$  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**.

# M-DPSK Modulator Baseband

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**Pair Block**      M-DPSK Demodulator Baseband

**See Also**      DBPSK Modulator Baseband, DQPSK Modulator Baseband, M-PSK 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.

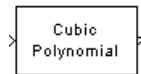
## Purpose

Apply memoryless nonlinearity to complex baseband signal

## Library

RF Impairments

## Description



The Memoryless Nonlinearity block applies a memoryless nonlinearity to a complex, baseband signal. You can use the block to model radio frequency (RF) impairments to a signal at the receiver.

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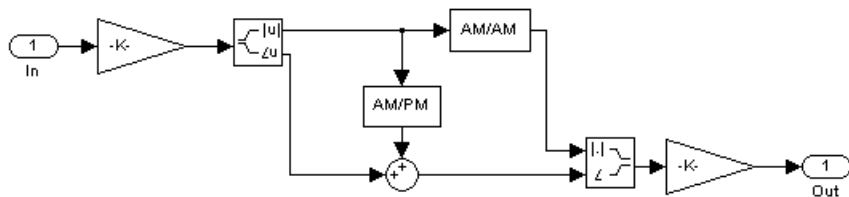
**Note** This block assumes a nominal impedance of 1 ohm.

---

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 five methods are implemented by subsystems underneath the block's mask. Each subsystem has the same basic structure, as shown in the following figure.



**Nonlinearity Subsystem**

# Memoryless Nonlinearity

---

All five subsystems apply 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.

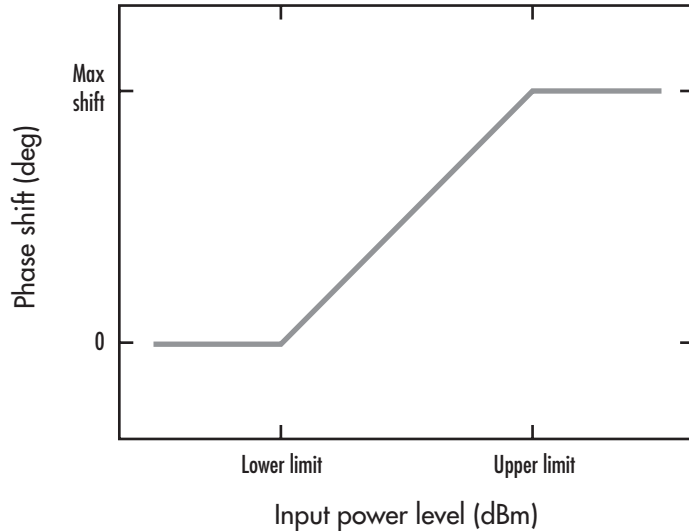
However, the subsystems implement the AM/AM and AM/PM conversions differently, according to the **Method** you specify.

If you want to see exactly how the Memoryless Nonlinearity block implements the conversions for a specific method, you can view the AM/AM and AM/PM subsystems that implement these conversions as follows:

- 1** Right-click on the Memoryless Nonlinearity block and select **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 2-349.
- 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 AM/PM conversion (dBm)**). Beyond these values, AM/PM conversion is constant at the values corresponding to the lower and upper input power limits, which are zero and  $(\text{AM/PM conversion}) \cdot (\text{upper input power limit} - \text{lower input power limit})$ , respectively.

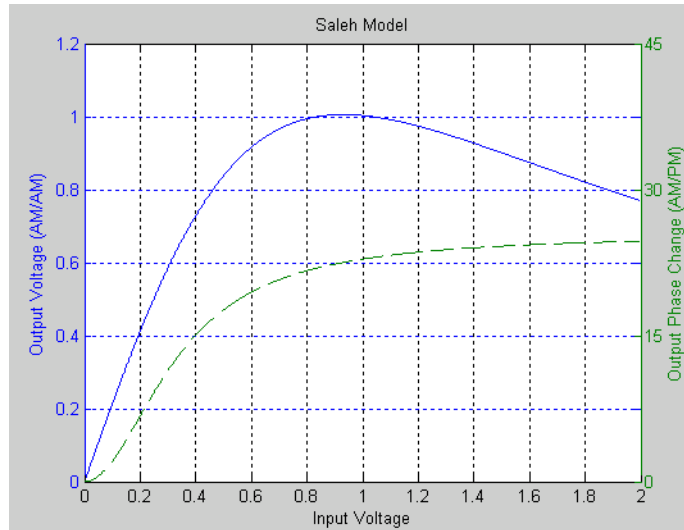
## AM/AM and AM/PM Characteristics of the Saleh Method

The following figure shows, for the Saleh method, plots of

- Output voltage against input voltage for the AM/AM conversion

# Memoryless Nonlinearity

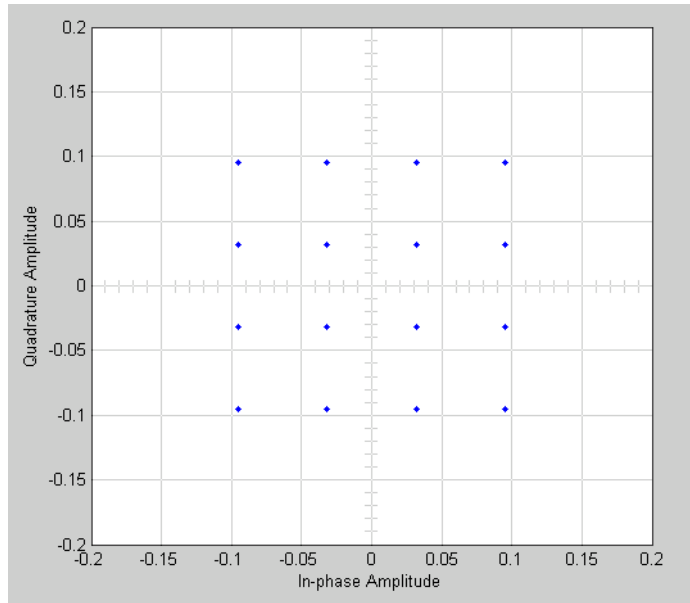
- Output phase against input voltage for the AM/PM conversion



## Example with 16-ary QAM

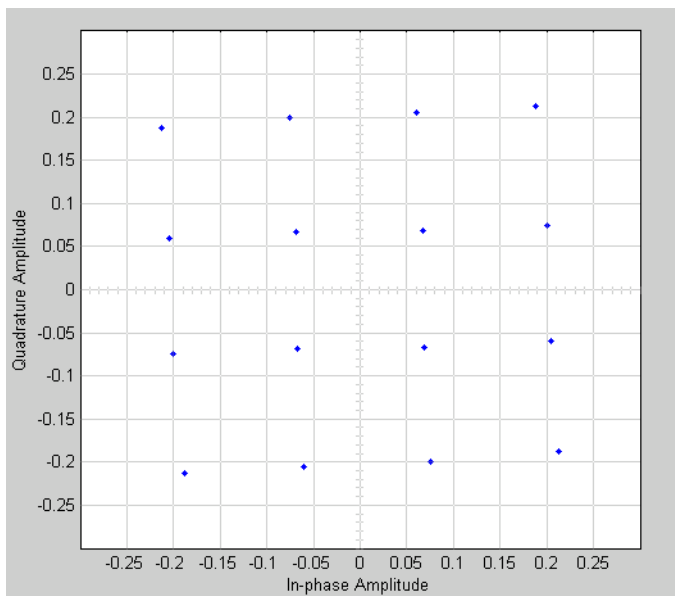
You can see the effect of the Memoryless Nonlinearity block on a signal modulated by 16-ary quadrature amplitude modulation (QAM) in a scatter plot. The constellation for 16-ary QAM without the effect of the Memoryless Nonlinearity block is shown in the following figure:





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 “Scatter Plot Examples” with the following parameter settings for the Rectangular QAM Modulator Baseband block:

- **Normalization method** set to Average Power
- **Average power (watts)** set to  $1e-2$

The following sections discuss parameters specific to the Saleh, Ghorbani, and Rapp models.

## Parameters for the Saleh 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, alpha and beta, are used to compute the amplitude gain for an input signal using the following function:

$$F_{AM/AM}(u) = \frac{\text{alpha} * u}{1 + \text{beta} * u^2}$$

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:

$$F_{AM/PM}(u) = \frac{\text{alpha} * u^2}{1 + \text{beta} * u^2}$$

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,  $[x_1 \ x_2 \ x_3 \ x_4]$ , are used to compute the amplitude gain for an input signal using the following function:

$$F_{AM/AM}(u) = \frac{x_1 u^{x_2}}{1 + x_3 u^{x_2}} + x_4 u$$

where  $u$  is the magnitude of the scaled signal.

The AM/PM parameters,  $[y_1 \ y_2 \ y_3 \ y_4]$ , are used to compute the phase change for an input signal using the following function:

# Memoryless Nonlinearity

---

$$F_{AM/PM}(u) = \frac{y_1 u^{y_2}}{1 + y_3 u^{y_2}} + y_4 u$$

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:

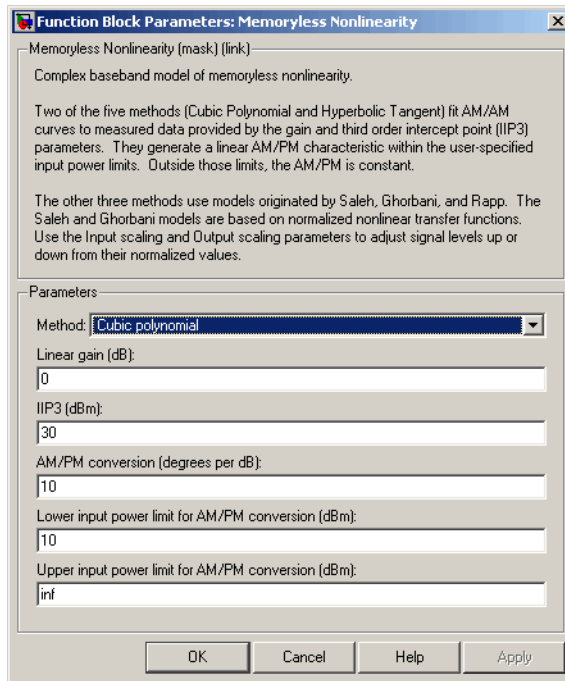
$$F_{AM/AM}(u) = \frac{u}{\left(1 + \left(\frac{u}{O_{sat}}\right)^{2S}\right)^{1/2S}}$$

where  $u$  is the magnitude of the scaled signal,  $S$  is the **Smoothness factor**, and  $O_{sat}$  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.

## Dialog Box



## Method

The nonlinearity method.

The following describes specific parameters for each method.

# Memoryless Nonlinearity

---

The image shows a software window titled "Parameters" with a light gray background. At the top, there is a dropdown menu labeled "Method:" with "Cubic polynomial" selected. Below this are several input fields with labels and values:

- Linear gain (dB): 0
- IIP3 (dBm): 30
- AM/PM conversion (degrees per dB): 10
- Lower input power limit for AM/PM conversion (dBm): 10
- Upper input power limit for AM/PM conversion (dBm): inf

## **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:

$$(\text{AM/PM conversion}) \cdot (\text{upper input power limit} - \text{lower input power limit})$$

Parameters

Method:

Linear gain (dB):

IIP3 (dBm):

AM/PM conversion (degrees per dB):

Lower input power limit for AM/PM conversion (dBm):

Upper input power limit for AM/PM conversion (dBm):

### **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:

$$(\text{AM/PM conversion}) \cdot (\text{upper input power limit} - \text{lower input power limit})$$

# Memoryless Nonlinearity

---

Parameters

Method: Saleh model

Input scaling (dB):  
0

AM/AM parameters [alpha beta]:  
[2.1587 1.1517]

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: Ghorbari model

Input scaling (dB):  
0

AM/AM parameters [x1 x2 x3 x4]:  
[8.1081 1.5413 6.5202 -0.0718]

AM/PM parameters [y1 y2 y3 y4]:  
[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 [x1 x2 x3 x4]

Vector specifying the AM/AM parameters.

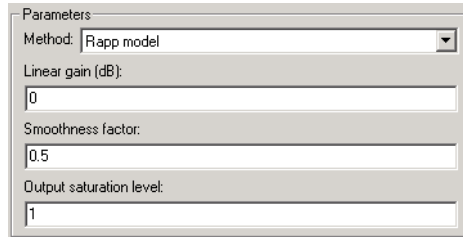


## AM/PM parameters [y1 y2 y3 y4]

Vector specifying the AM/PM parameters.

## Output scaling (dB)

Number that scales the output signal level.



The image shows a software parameter dialog box titled "Parameters". It contains four input fields:

- Method: Rapp model (dropdown menu)
- Linear gain (dB): 0 (text input)
- Smoothness factor: 0.5 (text input)
- Output saturation level: 1 (text input)

## 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

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 Digital Sound Broadcasting System", in Proceedings of the Second

# Memoryless Nonlinearity

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European Conference on Satellite Communications, Liege, Belgium,  
Oct. 22-24, 1991, pp. 179-184.

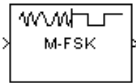
## Purpose

Demodulate FSK-modulated data

## Library

FM, in Digital Baseband sublibrary of Modulation

## Description



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. The input can be either a scalar or a frame-based column vector of type `single` or `double`.

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.

### Binary or Integer Outputs

If the **Output type** parameter is set to `Integer`, then the block outputs integers between 0 and  $M-1$ .

If the **Output type** parameter is set to `Bit` and the **M-ary number** parameter has the form  $2^K$  for some positive integer  $K$ , then the block outputs binary representations of integers between 0 and  $M-1$ . It outputs a group of  $K$  bits, called a binary *word*, for each symbol.

For boolean typed integer outputs, the **M-ary number** parameter must be 2. For `Bit` type outputs, the outputs must be of type `boolean` or `double`.

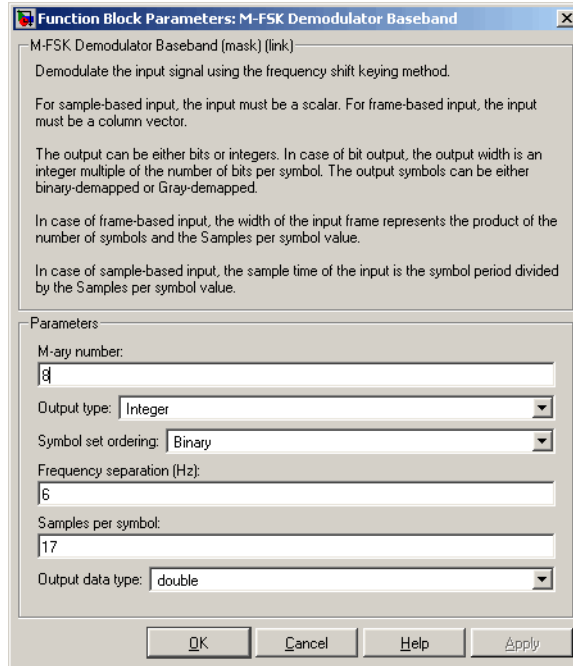
In binary output mode, the **Symbol set ordering** parameter indicates how the block maps an integer to a corresponding group of  $K$  output bits. See the reference pages for the M-FSK Modulator Baseband and M-PSK Modulator Baseband blocks for details.

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

# M-FSK Demodulator Baseband

lowest frequency is the negative frequency with the largest absolute value.

## 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  $2^K$  for some positive integer  $K$ .

### 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.

**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`.

**Pair Block** M-FSK Modulator Baseband

**See Also** CPFSK Demodulator Baseband

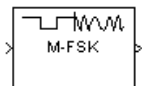
# M-FSK Modulator Baseband

---

**Purpose** Modulate using M-ary frequency shift keying method

**Library** FM, in Digital Baseband sublibrary of Modulation

## Description



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.

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 sampling frequency must be greater than M multiplied by the **Frequency separation** or the output signal will be aliased. Sampling frequency is **Samples per symbol** divided by the input symbol period (in seconds).

If the **Phase continuity** parameter is set to Continuous, then the modulated signal maintains its phase even when it changes its frequency. If the **Phase continuity** parameter is set to Discontinuous, then the modulated signal comprises portions of M sinusoids of different frequencies; thus, a change in the input value might cause a change in the phase of the modulated signal.

Note that a model with the M-FSK Modulator block will not run if it is used for sample-based signals unless **Tasking mode for periodic sample times** (in **Simulation > Configuration Parameters > Solver**) is set to SingleTasking.

## Input Signal Values

The input and output for this block are discrete-time signals. The **Input type** parameter determines whether the block accepts integers between 0 and M-1, or binary representations of integers:

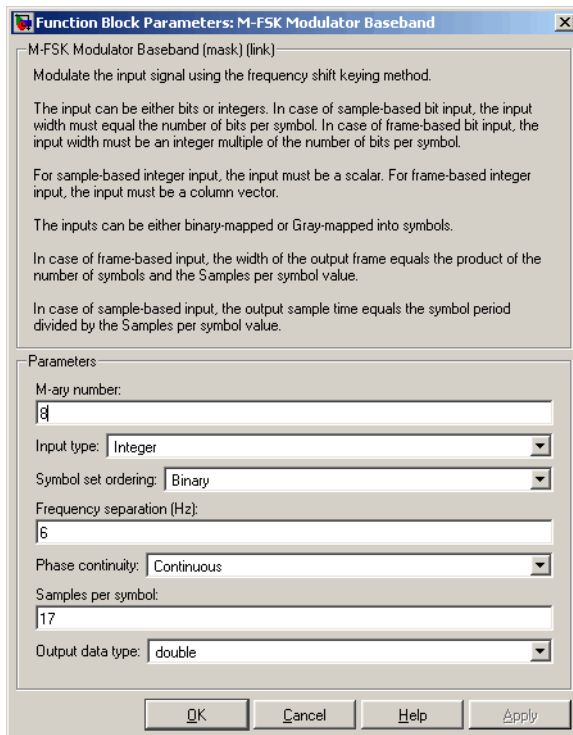
- If **Input type** is set to Integer, then the block accepts integers. The input can be either a scalar or a frame-based column vector of type int8, uint8, int16, uint16, int32, uint32, or a double with an integer value. They can also be boolean if the size of the alphabet is 2 (i.e. M = 2).

- If **Input type** is set to Bit, then the block accepts groups of K bits, called binary words. The input can be either a vector of length K or a frame-based column vector (whose length is an integer multiple of K), and must be boolean or double typed, valued from the set {0, 1}. The **Symbol set ordering** parameter indicates how the block assigns binary words to corresponding integers.
  - If **Symbol set ordering** is set to Binary, then the block uses a natural binary-coded ordering.
  - If **Symbol set ordering** is set to Gray, then the block uses a Gray-coded ordering. For details about the Gray coding, see the reference page for the M-PSK Modulator Baseband block.

Whether the input is an integer or a binary representation of an integer, the block maps the integer 0 to the highest frequency and maps the integer M-1 to the lowest frequency. In baseband simulation, the lowest frequency is the negative frequency with the largest absolute value.

# M-FSK Modulator Baseband

## Dialog Box



### 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 this parameter is set to Bit, then the **M-ary number** parameter must be  $2^K$  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.

**Samples per symbol**

The number of output samples that the block produces for each integer or binary word in the input.

**Output data type**

The output type of the block can be specified as either a double or a single. By default, the block sets this to double.

**Pair Block**

M-FSK Demodulator Baseband

**See Also**

CPFSK Modulator Baseband

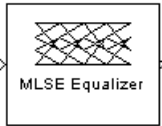
# MLSE Equalizer

---

**Purpose** Equalize using Viterbi algorithm

**Library** Equalizers

## Description



The MLSE Equalizer block uses the Viterbi algorithm to equalize a linearly modulated signal through a dispersive channel. The block receives a frame-based input signal and outputs the maximum likelihood sequence estimate (MLSE) of the signal, using an estimate of the channel modeled as a finite input response (FIR) filter.

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. The block displays an additional input port, labeled Ch, that receives a frame-based vector.

## 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  $k$ th 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

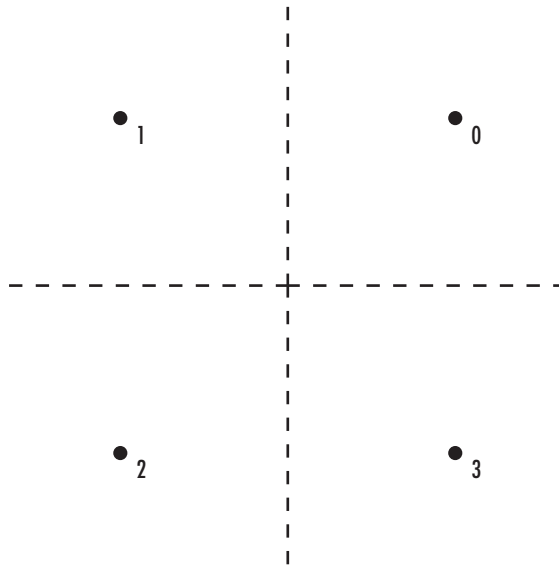
$$0 \rightarrow +1+i$$

$$1 \rightarrow -1+i$$

$$2 \rightarrow -1-i$$

$$3 \rightarrow +1-i$$

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.



## 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  $M-1$ , where  $M$  is the number of constellation points. An integer value of  $k-1$  in the vector corresponds to the  $k$ th entry in the **Constellation points** vector and, consequently, to a modulator input of  $k-1$ .

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 using the Matrix Concatenation block.

To learn how the block uses the preamble and postamble, see “Reset Every Frame” Operation Mode” on page 2-372 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.

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## 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.

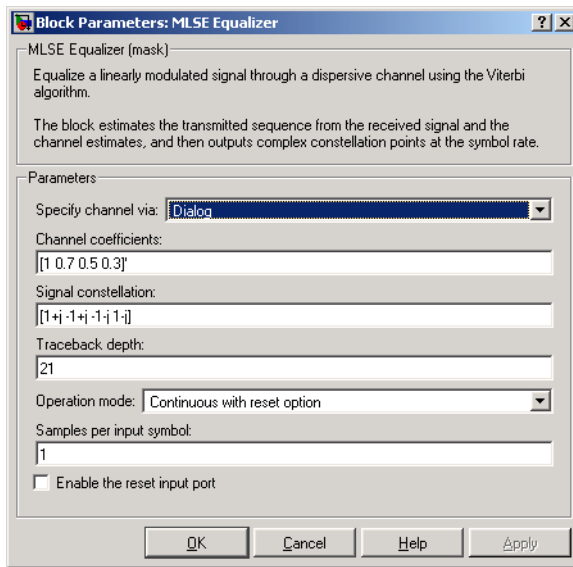
If you select the **Enable the reset input port** check box, the block displays another input port, labeled Rst. In this case, the block resets the state metrics whenever the scalar value at the Rst port is nonzero.

## 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 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.

**See Also**

LMS Linear Equalizer, LMS Decision Feedback Equalizer, RLS Linear Equalizer, RLS Decision Feedback Equalizer, CMA Equalizer

# MLSE Equalizer

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## 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

Demodulate PAM-modulated data

## Library

AM, in Digital Baseband sublibrary of Modulation

## Description



The M-PAM Demodulator Baseband block demodulates a signal that was modulated using the 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.

The input can be either a scalar or a frame-based column vector and must be of data type `single`, `double`, or signed fixed-point

---

**Note** This block assumes a nominal impedance of 1 ohm.

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## Output Signal Values

The **Output type** parameter determines whether the block produces integers or binary representations of integers. If **Output type** is set to `Integer`, then the block produces integers. If **Output type** is set to `Bit`, then the block produces a group of K bits, called a *binary word*, for each symbol. 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.

## Algorithm

The demodulator algorithm maps received input signal constellation values to M-ary integer symbol indices between 0 and 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**

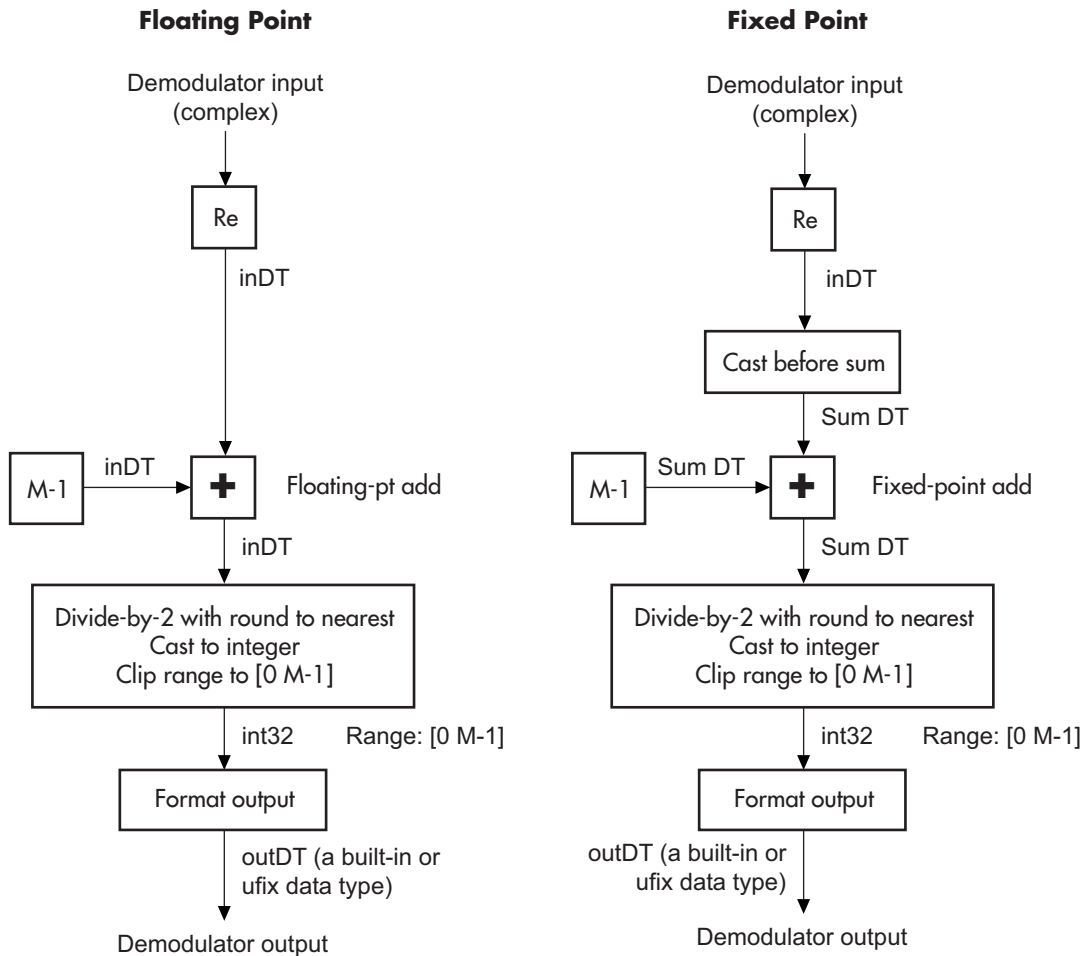
## M-PAM Demodulator Baseband

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and related parameters. This denormalized value is added to  $M-1$  to translate it into an approximate range between 0 and  $2 \times (\sqrt{M} - 1)$  (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  $M-1$  (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**.

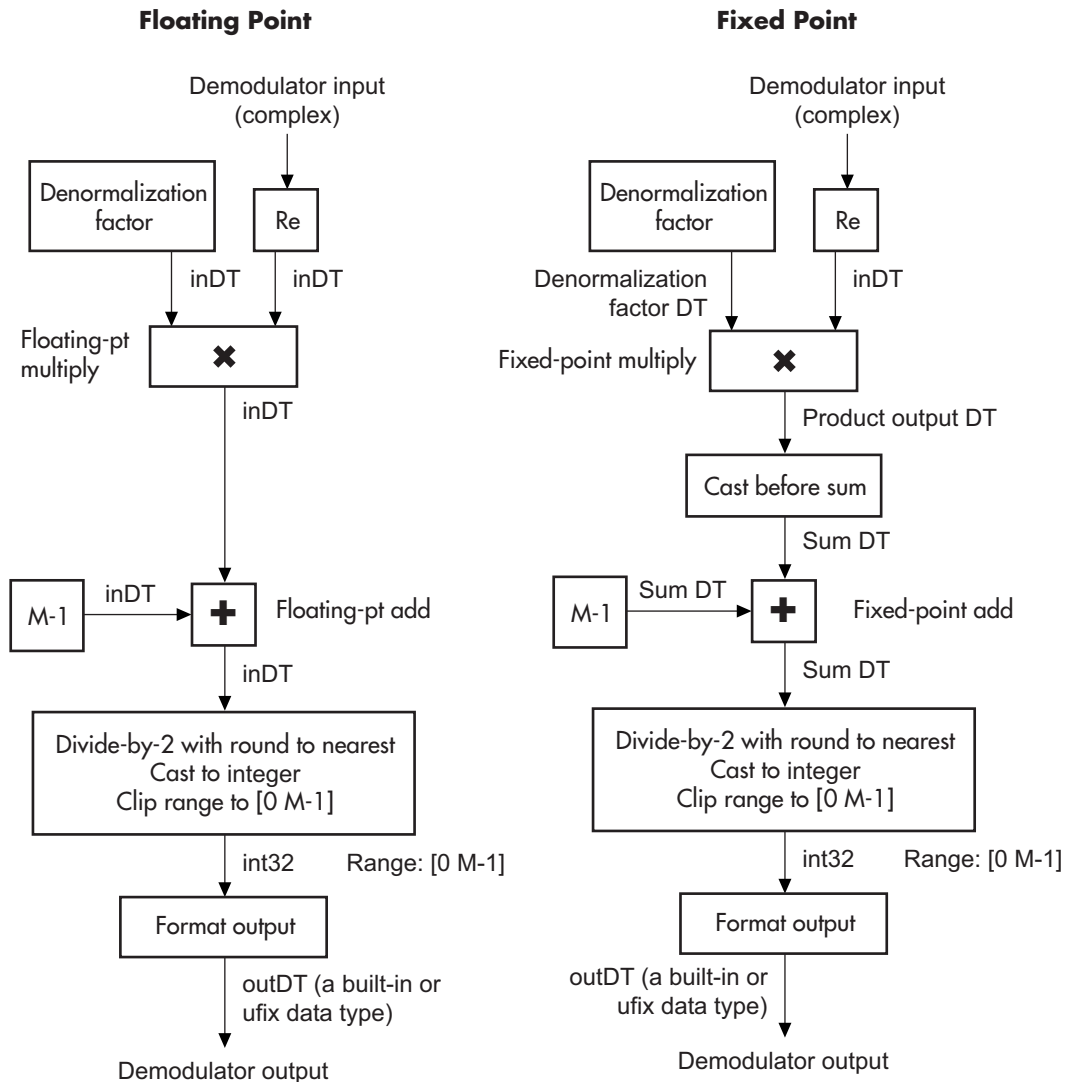
The following figures contains signal flow diagrams for floating-point and fixed-point algorithm operation. The floating-point diagrams apply when the input signal data type is double or single. The fixed-point diagrams apply when the input signal is a signed fixed-point data type. Note that the diagram is simplified when using normalized constellations (i.e., denormalization factor is 1).

# M-PAM Demodulator Baseband



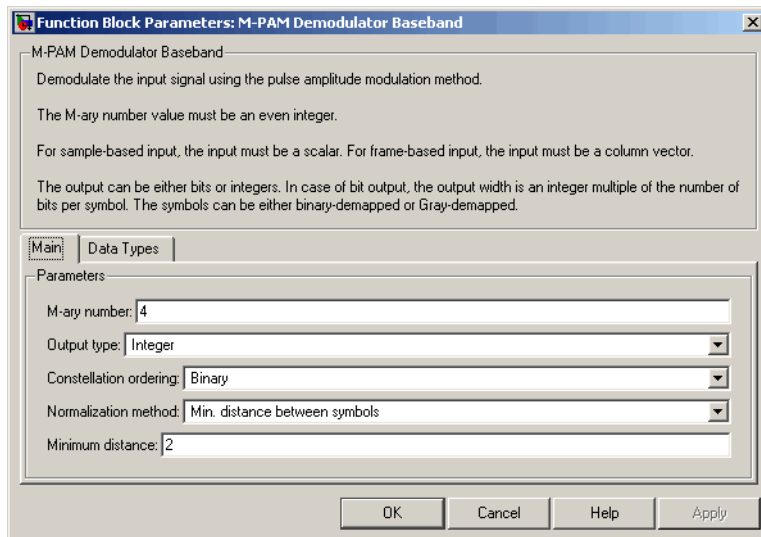
**Signal-Flow Diagrams with Denormalization Factor Equal to 1**

# M-PAM Demodulator Baseband



**Signal-Flow Diagrams with Nonunity Denormalization Factor**

## Dialog Box



### **M-ary number**

The number of points in the signal constellation. It must be an even integer.

### **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  $2^K$  for some positive integer  $K$ .

### **Constellation 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**.

### **Normalization method**

Determines how the block scales the signal constellation. Choices are **Min. distance between symbols**, **Average Power**, and **Peak Power**.

# M-PAM Demodulator Baseband

## Minimum distance

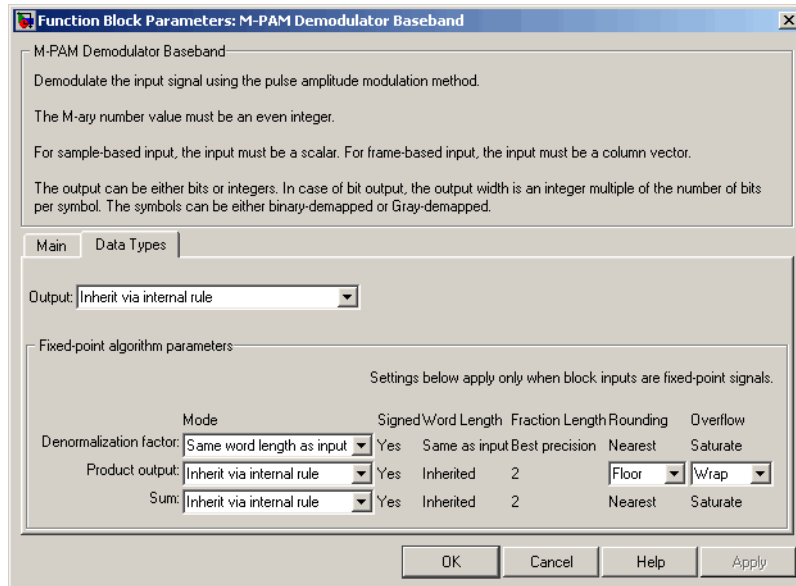
The distance between two nearest constellation points. This field appears only when **Normalization method** is set to Min. distance between symbols.

## Average power (watts)

The average power of the symbols in the constellation. This field appears only when **Normalization method** is set to Average Power.

## Peak power (watts)

The maximum power of the symbols in the constellation. This field appears only when **Normalization method** is set to Peak Power.



## 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 the **Hardware Implementation** pane of the Configuration Parameters dialog box of the block. 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**

Can be set to Same word length as input or Specify word length, in which case a field is enabled for user input.

### **Product output**

Can be set to Inherit via internal rule or Specify word length, in which case a field is enabled for user input.

Setting to Inherit via internal rule computes the product word length as follows. The ideal full-precision product word and fraction lengths are calculated as  $idealProdWL = inputWL + denormFactorWL$  and  $idealProdFL = inputFL + denormFactorFL$ .

After this computation, the product output word length is set to an  $adjustedProdWL$ , as shown in the following table. The product fraction length is set to 2, because additional fraction bits do not affect the fixed-point algorithm precision.

# M-PAM Demodulator Baseband

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<b>If <i>idealProdWL</i> is...</b>	<b>Set <i>adjustedProdWL</i> to...</b>
Greater than or equal to 2	$idealProdWL - (idealProdFL - 2)$
Less than 2	Throw an error, because the input, denormalization factor, or both is not precise enough.

Finally, the adjusted product word length is set to be the smallest available word length (as specified in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the block) large enough to contain it.

## Sum

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 sets the sum word length to the smallest available word length (as specified in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the block) that fits the sum word length given by  $numBitsPerSymbol + 4$  (or, equivalently,  $ceil(\log_2(M)) + 4$ ). The sum fraction length is set to 2.

**Pair Block** M-PAM Modulator Baseband

**See Also** General QAM Demodulator Baseband



**Purpose** Modulate using M-ary pulse amplitude modulation

**Library** AM, in Digital Baseband sublibrary of Modulation

## Description



The M-PAM Modulator Baseband block modulates using M-ary pulse amplitude modulation. 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. It must be an even integer.

---

**Note** This block assumes 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

$$2m - M + 1$$

---

**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 Method Parameter	Scaling Condition
Min. distance between symbols	The nearest pair of points in the constellation is separated by the value of the <b>Minimum distance</b> parameter
Average Power	The average power of the symbols in the constellation is the <b>Average power</b> parameter
Peak Power	The maximum power of the symbols in the constellation is the <b>Peak power</b> parameter

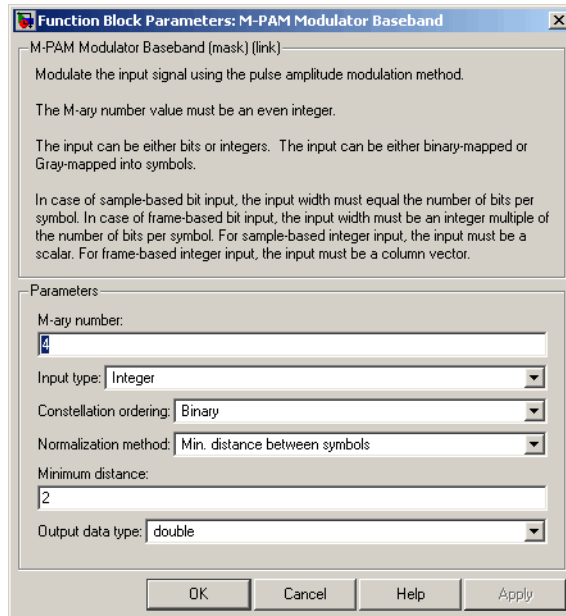
## Input Signal Values

The input and output for this block are discrete-time signals. The **Input type** parameter determines whether the block accepts integers between 0 and M-1, or binary representations of integers.

- If **Input type** is set to Integer, then the block accepts integers. The input can be either a scalar or a frame-based column vector of data type int8, uint8, int16, uint16, int32, uint32, single, or double.
- If **Input type** is set to Bit, then the block accepts groups of K bits, called binary words. The input can be either a vector of length K or a frame-based column vector whose length is an integer multiple of K. For bit inputs, the block can accept int8, uint8, int16, uint16, int32, uint32, boolean, single, and double. 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.
  - 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.

## 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  $2^K$  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

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## **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 (watts)**

The average power of the symbols in the constellation. This field appears only when **Normalization method** is set to Average Power.

## **Peak power (watts)**

The maximum power of the symbols in the constellation. 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 Simulink Fixed Point. 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 the **Output data type** parameter and **User-defined** for the **Set output fraction length to** parameter.

## Pair Block

M-PAM Demodulator Baseband

## See Also

General QAM Modulator Baseband

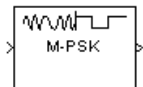
# M-PSK Demodulator Baseband

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**Purpose** Demodulate PSK-modulated data

**Library** PM, in Digital Baseband sublibrary of Modulation

## Description



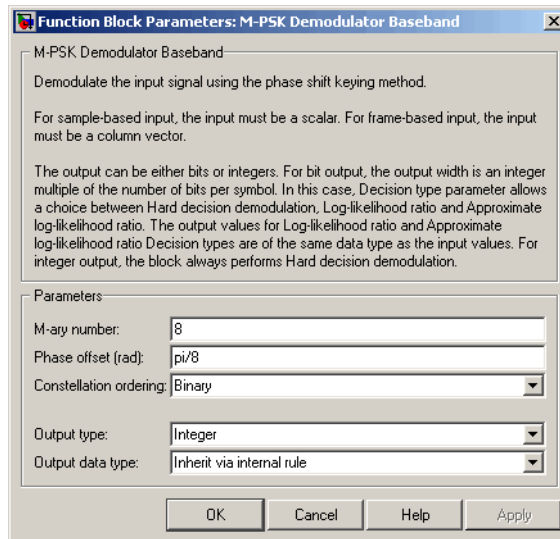
The M-PSK Demodulator Baseband block demodulates a signal that was modulated using the M-ary phase shift keying method. The input is a baseband representation of the modulated signal. The input and output for this block are discrete-time signals. The input can be either a scalar or a frame-based column vector of data types `single` or `double`. The **M-ary number** parameter,  $M$ , is the number of points in the signal constellation.

### Binary or Integer Outputs

If the **Output type** parameter is set to `Integer`, the block outputs integers between 0 and  $M-1$ , representing M-ary symbols.

If the **Output type** parameter is set to `Bit` and **Decision type** is set to `Hard decision`, the block outputs binary representations of integers between 0 and  $M-1$ . It outputs a group of  $K$  bits, called a *binary word*, for each symbol.

## Dialog Box



### M-ary number

The number of points in the signal constellation.

### Phase offset (rad)

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.

### 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.

# M-PSK Demodulator Baseband

---

The first element of this vector corresponds to the constellation point at 0 angle, with subsequent elements running counterclockwise. The last element corresponds to the  $-\pi/M$  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  $2^K$  for some positive integer K.

## Decision type

This field appears when Bit is selected in the drop-down list

### Output type.

Specifies the output to be bitwise hard decision, LLR, or approximate LLR. See “Exact LLR Algorithm” and “Approximate LLR Algorithm” for algorithm details.

## 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, when **Decision type** is set to Hard decision, output can be int8, uint8, int16, uint16, int32, uint32, boolean, single, or double. When **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is inherited from the input (i.e., if the input is of data type double, the output is also of data type double).

## 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.

## **Noise variance**

This field appears when **Noise variance source** is set to `Dialog`.

If this value is very small (i.e., SNR is very high), LLR computations may yield `Inf` or `-Inf` because the LLR algorithm would involve computing exponentials of very large or very small numbers using finite precision arithmetic. In such cases, use of approximate LLR is recommended, as its algorithm does not involve computing exponentials.

## **Pair Block**

M-PSK Modulator Baseband

## **See Also**

BPSK Demodulator Baseband, QPSK Demodulator Baseband, M-DPSK Demodulator Baseband

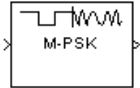
# M-PSK Modulator Baseband

---

**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 input and output for this block are discrete-time signals. 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`.

Alternative configurations of the block determine how the block interprets its input and arranges its output, as explained in the following sections.

### Binary Inputs

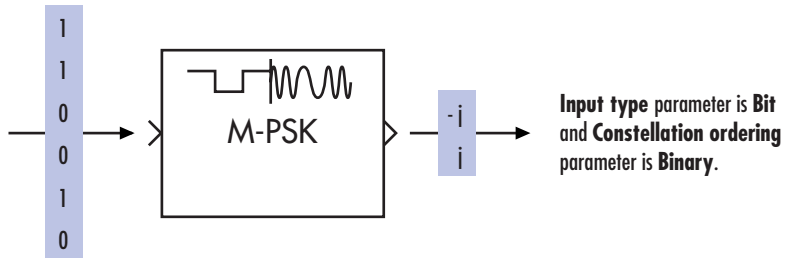
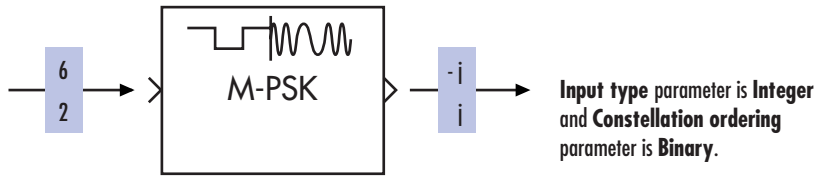
If the **Input type** parameter is set to `Bit` and the **M-ary number** parameter has the form  $2^K$  for some positive integer K, the block accepts binary representations of integers between 0 and M-1. It modulates each group of K bits, called a *binary word*. The input can be either a vector of length K or a frame-based column vector whose length is an integer multiple of K. For more information, see “Binary-Valued and Integer-Valued Signals” in *Communications Blockset User’s Guide*.

### Frame-Based Inputs

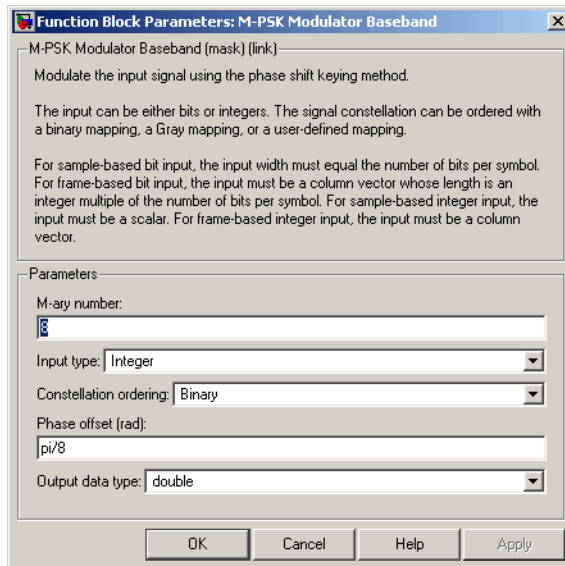
If the input is a frame-based column vector, the block processes several integers or several binary words in each time step. (If the **Input type** parameter is set to `Bit`, a binary word consists of  $\log_2(M)$  bits.)

For example, the following schematics illustrate how the block processes two 8-ary integers or binary words in one time step. The signals involved are all frame-based column vectors. In both cases, the **Phase offset** parameter is 0.

# M-PSK Modulator Baseband



## Dialog Box



# M-PSK Modulator Baseband

---

## **M-ary number**

The number of points in the signal constellation.

## **Input type**

Indicates whether the input consists of integers or groups of bits.

If this parameter is set to **Bit**, the **M-ary number** parameter must be  $2^K$  for some positive integer  $K$ .  $K$  consecutive elements in each channel or column represent a symbol, where  $K = \log_2(M)$ . The number of elements in each channel must be an integer multiple of  $K$ , and each element must be 0 or 1.

To use integers between 0 and  $M-1$  as input values, set this to **Integer**. In this case, the input can be either a scalar or a frame-based column vector.

## **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

$$\exp(j\theta + j2\pi m/M)$$

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  $M$ . The explicit mapping is described in “Algorithm” on page 2-398.

Selecting **User-defined** displays the field **Constellation mapping**, which allows for user-specified mapping.

## **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  $0 + \mathbf{Phase\ offset}$  angle, with subsequent elements running counterclockwise. The last element corresponds to the  $-\pi/M + \mathbf{Phase\ offset}$  constellation point.

### **Phase offset**

The phase of the zeroth point of the signal constellation.

### **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 enables 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.

### **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 Simulink Fixed Point. 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.

# M-PSK Modulator Baseband

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- 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.

## Algorithm

If the **Constellation ordering** parameter is set to Gray, the block internally assigns the binary inputs to points of a predefined Gray-coded signal constellation. The block's predefined M-ary Gray-coded signal constellation assigns the binary representation

```
de2bi(bitxor(M, floor(M/2)), log2(M), 'left-msb')
```

to the Mth 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

$$j\theta + j2\pi m/M$$

where  $\theta$  is the **Phase offset** parameter and  $m$  is an integer between 0 and  $M-1$  that satisfies

$$m \text{ XOR } \lfloor m/2 \rfloor = U$$

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')
```

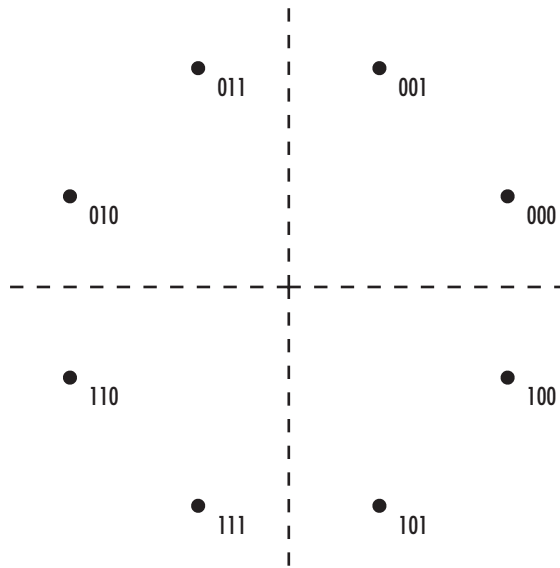
ans =

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  $\pi/8$ .

# M-PSK Modulator Baseband

---



**Pair Block**

M-PSK Demodulator Baseband

**See Also**

BPSK Modulator Baseband, QPSK Modulator Baseband, M-DPSK Modulator Baseband



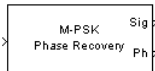
## Purpose

Recover carrier phase using M-Power method

## Library

Carrier Phase Recovery sublibrary of Synchronization

## Description



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 is 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 Outputs

The input signal must be a frame-based column vector or a sample-based scalar 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.

# M-PSK Phase Recovery

- 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.

---

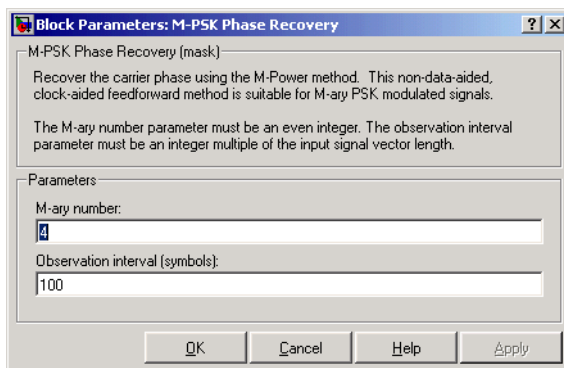
**Note** Because the block internally computes the argument of a complex number, the carrier phase estimate has an inherent ambiguity. The carrier phase estimate is between  $-180/M$  and  $180/M$  degrees and might differ from the actual carrier phase by an integer multiple of  $360/M$  degrees.

---

## 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, or 4 for a QAM signal. This must be an even integer.

**Observation interval**

The number of symbols for which the carrier phase is assumed constant.

**Examples**

See “Carrier Phase Recovery Example” in *Communications Blockset User’s Guide*.

**Algorithm**

If the symbols occurring during the observation interval are  $x(1)$ ,  $x(2)$ ,  $x(3)$ ,...,  $x(L)$ , then the resulting carrier phase estimate is

$$\frac{1}{M} \arg \left\{ \sum_{k=1}^L (x(k))^M \right\}$$

where the arg function returns values between -180 degrees and 180 degrees.

**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

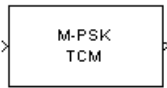
# M-PSK TCM Decoder

---

**Purpose** Decode trellis-coded modulation data, modulated using PSK method

**Library** Trellis-Coded Modulation

## Description



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 is the number of points in the signal constellation, which also equals the number of possible output symbols from the convolutional encoder. (That is,  $\log_2(\mathbf{M}\text{-ary number})$  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

The input signal must be a frame-based column vector containing complex numbers.

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 frame-based binary column vector whose length is  $k$  times the vector length of the input signal.

The input signal must be double or single. The reset port accepts double or boolean.

## 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.

If you select the **Enable the reset input** check box, 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.

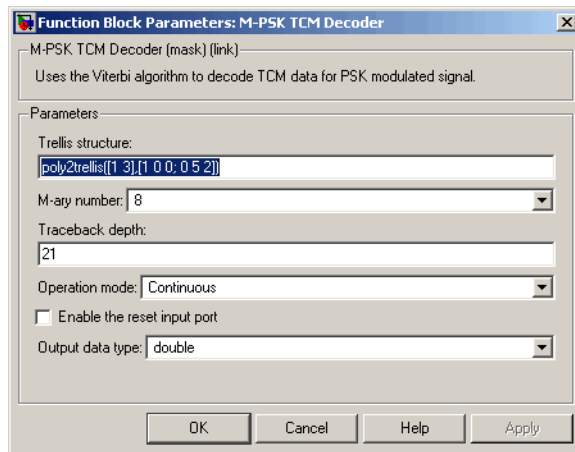
## Decoding Delay

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.

### 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.

## 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.

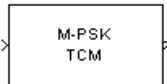
# M-PSK TCM Encoder

---

**Purpose** Convolutionally encode binary data and modulate using PSK method

**Library** Trellis-Coded Modulation

## Description



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,  $\log_2(\mathbf{M\text{-ary number}})$  is equal to  $n$  for a rate  $k/n$  convolutional code.)

## Input and Output Signals

If the convolutional encoder described by the trellis structure represents a rate  $k/n$  code, then the M-PSK TCM Encoder block's input must be a frame-based binary column vector whose length is  $L \cdot k$  for some positive integer  $L$ .

The output from the M-PSK TCM Encoder block is a frame-based complex column vector of length  $L$ .

The input signal must be boolean.

## 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 Encoder” in the Communications 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



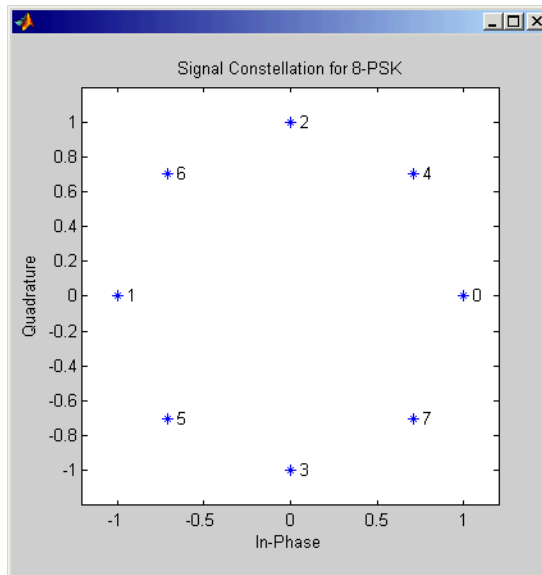
```
poly2trellis(7,[171 133],171)
```

- 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 to spend less time updating the diagram at the beginning of each simulation, compared to the usage in the previous bulleted item.

## 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 **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].

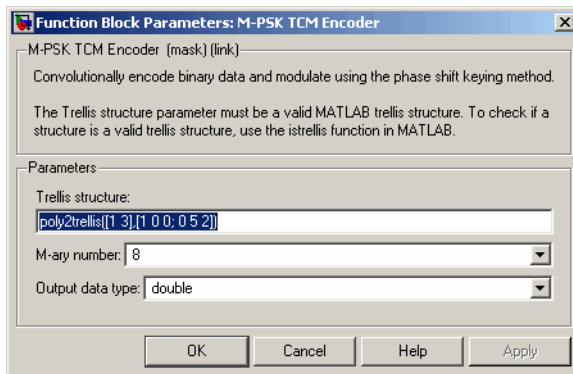


# M-PSK TCM Encoder

## 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].

## 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.

## 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

**Library** 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 is a baseband representation of the modulated signal. The **Phase offset** parameter is the initial phase of the modulated waveform.

## Traceback Length and Output Delays

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The **Traceback length** 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.

- If the input signal is sample-based, then the delay consists of  $D+1$  zero symbols.
- If the input signal is frame-based, then the delay consists of  $D$  zero symbols.

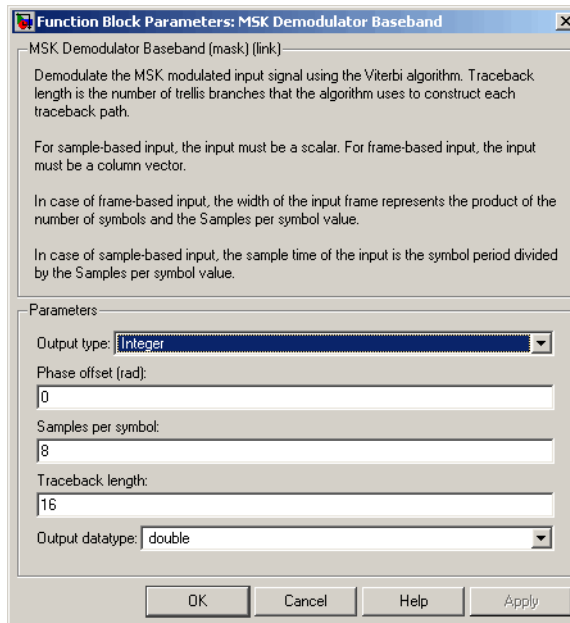
## Inputs and Outputs

The input can be either a scalar or a frame-based column vector and must be of type `single` or `double`. If the **Output type** parameter is set to `Integer`, then the block produces values of 1 and -1. If the **Output type** parameter is set to `Bit`, then the block produces values of 0 and 1.

## Processing an Upsampled Modulated Signal

The input signal can be an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor. It must be a positive integer. For more information, see “Upsampled Signals and Rate Changes” in *Communications Blockset User’s Guide*.

## Dialog 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.

### Traceback length

The number of trellis branches that the Viterbi Decoder 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

---

**Pair Block**      MSK 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.

## Purpose

Modulate using minimum shift keying method

## Library

CPM, in Digital Baseband sublibrary of Modulation

## Description



The MSK Modulator Baseband block modulates using the minimum shift keying method. The output is a baseband representation of the modulated signal.

The **Phase offset** parameter is the initial phase of the output waveform, measured in radians.

## Input Attributes

The input can be either a scalar or a frame-based column vector. If the **Input type** parameter is set to Integer, then the block accepts values of 1 and -1. If the **Input type** parameter is set to Bit, then the block accepts values of 0 and 1.

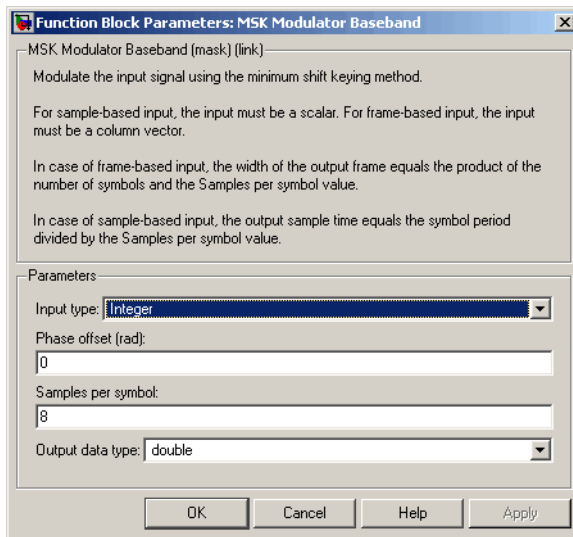
## Upsampling the Modulated Signal

This block can output an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor. It must be a positive integer. For more information, see “Upsampled Signals and Rate Changes” in *Communications Blockset User’s Guide*.

# MSK Modulator Baseband

---

## Dialog Box



### Input type

Indicates whether the input consists of bipolar or binary values.

### Phase offset (rad)

The initial phase of the output waveform.

### Samples per symbol

The number of output samples that the block produces for each integer or bit in the input.

### 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

MSK Demodulator Baseband

## See Also

CPM Modulator Baseband

## References

[1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg, *Digital Phase Modulation*, New York, Plenum Press, 1986.



# MSK-Type Signal Timing Recovery

## Purpose

Recover symbol timing phase using fourth-order nonlinearity method

## Library

Timing Phase Recovery sublibrary of Synchronization

## Description



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.

The input must be a scalar or a frame-based column vector. The input uses  $N$  samples to represent each symbol, where  $N > 1$  is the **Samples per symbol** parameter. If the input is frame-based, then its vector length is  $N \cdot R$ , where  $R$  is a positive integer that indicates the number of symbols per frame. If the input is sample-based, then its sample time is  $1/N$  times the underlying symbol period.

If the **Reset** parameter is set 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 signal. The sample time of the *Rst* input equals the symbol period if the input signal is sample-based, and the frame period if the input signal is frame-based.

## 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

# MSK-Type Signal Timing Recovery

---

symbol, which can be used for decision purposes. The values in the Sym output occur at the symbol rate:

- If the input signal is a frame-based column vector of length  $N \cdot R$ , then the Sym output is a frame-based column vector of length  $R$  having the same frame period.
- If the input signal is a sample-based scalar with sample time  $T/N$ , then the Sym output is a sample-based scalar with sample time  $T$ .
- The Ph output gives the phase estimate for each symbol in the input signal.

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 or frame period of the Ph output is the same as that of the Sym output.

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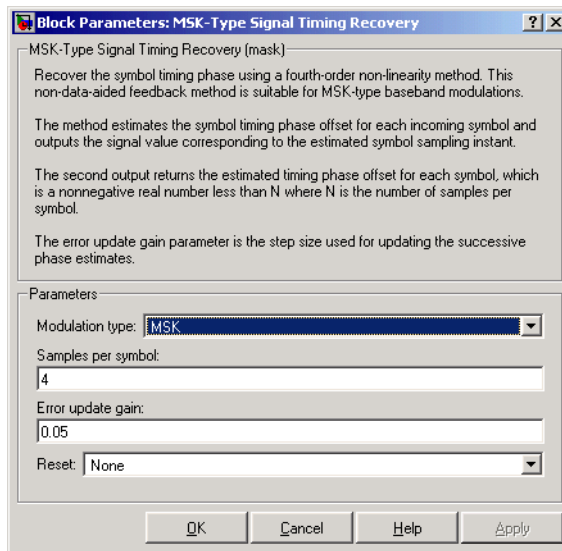
**Note** If the Ph output is very close to either zero or **Samples per symbol**, or if the actual timing phase offset in your input signal is very close to zero, then the block's accuracy might be compromised by small amounts of noise or jitter. The block works well when the timing phase offset is significant rather than very close to zero.

---

## Delays

This block incurs a delay of two symbols when the input signal is frame-based and three symbols when the input signal is sample-based.

## 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  $1/N$ , which corresponds to a slowly varying phase.

### 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.

# MSK-Type Signal Timing Recovery

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## 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  $k$ th symbol is  $e(k)$ , given in [2] by

$$e(k) = (-1)^{D+1} \operatorname{Re}\{r^2(kT - T_s + d_{k-1})r^{*2}((k-1)T - T_s + d_{k-2})\} \\ - (-1)^{D+1} \operatorname{Re}\{r^2(kT + T_s + d_{k-1})r^{*2}((k-1)T + T_s + d_{k-1})\}$$

where

- $r$  is the block's input signal
- $T$  is the symbol period
- $T_s$  is the sampling period
- $*$  means complex conjugate
- $d_k$  is the phase estimate for the  $k$ th 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 Blockset 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

# Mueller-Muller Timing Recovery

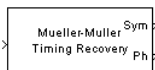
## Purpose

Recover symbol timing phase using Mueller-Muller method

## Library

Timing Phase Recovery sublibrary of Synchronization

## Description



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. The input must be a scalar or a frame-based column vector. The input uses  $N$  samples to represent each symbol, where  $N > 1$  is the **Samples per symbol** parameter. If the input is frame-based, then its vector length is  $N \cdot R$ , where  $R$  is a positive integer that indicates the number of symbols per frame. If the input is sample-based, then its sample time is  $1/N$  times the underlying symbol period.

If the **Reset** parameter is set 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. The sample time of the *Rst* input equals the symbol period if the input signal is sample-based, and the frame period if the input signal is frame-based.

## 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:
  - If the input signal is a frame-based column vector of length  $N \cdot R$ , then the *Sym* output is a frame-based column vector of length  $R$  having the same frame period.

# Mueller-Muller Timing Recovery

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- If the input signal is a sample-based scalar with sample time  $T/N$ , then the Sym output is a sample-based scalar with sample time  $T$ .
- The Ph output gives the phase estimate for each symbol in the input signal.

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 or frame period of the Ph output is the same as that of the Sym output.

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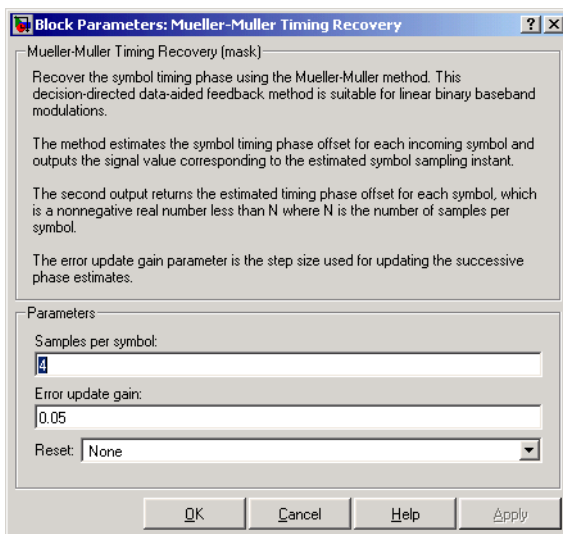
**Note** If the Ph output is very close to either zero or **Samples per symbol**, or if the actual timing phase offset in your input signal is very close to zero, then the block's accuracy might be compromised by small amounts of noise or jitter. The block works well when the timing phase offset is significant rather than very close to zero.

---

## Delays

This block incurs a delay of two symbols when the input signal is frame-based and three symbols when the input signal is sample-based.

## 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  $1/N$ , which corresponds to a slowly varying phase.

### 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 uses a timing error detector whose result for the  $k$ th symbol is  $e(k)$ , given by

# Mueller-Muller Timing Recovery

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$$e(k) = \text{Re}\{c_{k-1}^* y(kT + d_k) - c_k^* y((k-1)T + d_{k-1})\}$$

where

- $y$  is the block's input signal
- $c_k$  is the decision based on the sample value  $y(kT + d_k)$
- $T$  is the symbol period
- $d_k$  is the phase estimate for the  $k$ th 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 Blockset User's Guide*.

## References

- [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.
- [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



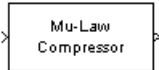
## Purpose

Implement  $\mu$ -law compressor for source coding

## Library

Source Coding

## Description



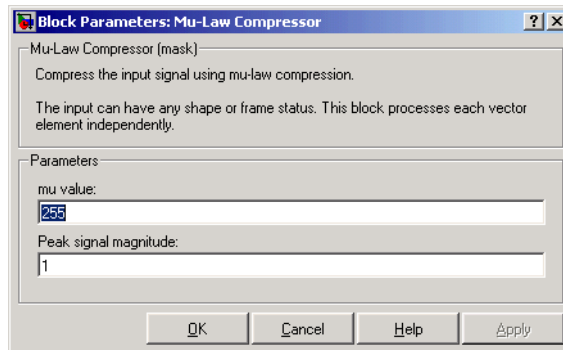
The Mu-Law Compressor block implements a  $\mu$ -law compressor for the input signal. The formula for the  $\mu$ -law compressor is

$$y = \frac{V \log(1 + \mu |x| / V)}{\log(1 + \mu)} \text{sgn}(x)$$

where  $\mu$  is the  $\mu$ -law parameter of the compressor,  $V$  is the peak magnitude of  $x$ ,  $\log$  is the natural logarithm, and  $\text{sgn}$  is the signum function (`sign` in MATLAB).

The input can have any shape or frame status. This block processes each vector element independently.

## Dialog Box



### mu value

The  $\mu$ -law parameter of the compressor.

### Peak signal magnitude

The peak value of the input signal. This is also the peak value of the output.

## Pair Block

Mu-Law Expander

# Mu-Law Compressor

---

## **See Also**

A-Law Compressor

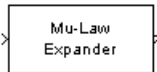
## **References**

[1] Sklar, Bernard. *Digital Communications: Fundamentals and Applications*. Englewood Cliffs, N.J.: Prentice-Hall, 1988.

**Purpose** Implement  $\mu$ -law expander for source coding

**Library** Source Coding

## Description

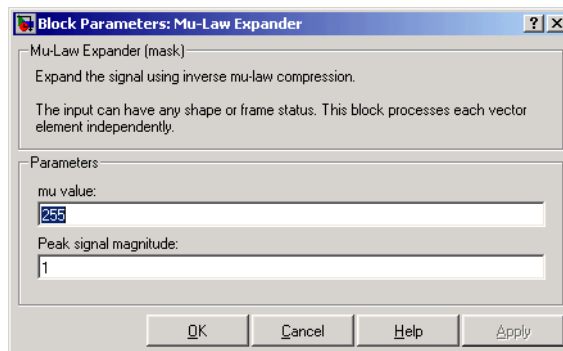


The Mu-Law Expander block recovers data that the Mu-Law Compressor block compressed. The formula for the  $\mu$ -law expander, shown below, is the inverse of the compressor function.

$$x = \frac{V}{\mu} \left( e^{|y| \log(1+\mu)/V} - 1 \right) \text{sgn}(y)$$

The input can have any shape or frame status. This block processes each vector element independently.

## Dialog Box



### mu value

The  $\mu$ -law parameter of the compressor.

### Peak signal magnitude

The peak value of the input signal. This is also the peak value of the output.

**Pair Block** Mu-Law Compressor

**See Also** A-Law Expander

# Mu-Law Expander

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## References

[1] Sklar, Bernard. *Digital Communications: Fundamentals and Applications*. Englewood Cliffs, N.J.: Prentice-Hall, 1988.

# Multipath Rayleigh Fading Channel

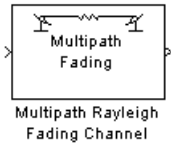
## Purpose

Simulate multipath Rayleigh fading propagation channel

## Library

Channels

## Description



The Multipath Rayleigh Fading Channel block implements a baseband simulation of a multipath Rayleigh fading propagation channel. This block is useful for modeling mobile wireless communication systems. For details about fading channels, see the references listed below.

This block accepts only frame-based complex signals at its input. To work with sample-based inputs, use the `Frame conversion` block of Signal Processing blockset to reformat the 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. The Jakes PSD (power spectral density) determines the spectrum of the Rayleigh process.

Since a multipath channel reflects signals at multiple places, a transmitted signal travels to the receiver along several paths that may have different lengths and hence different associated time delays. Fading occurs when signals traveling along different paths interfere with each other. In the block's parameter dialog, the **Delay vector** specifies the time delay for each path. If the **Normalize gain vector to 0 dB overall gain** box is unchecked, then the **Gain vector** specifies the gain for each path. If the box is checked, then the block uses a multiple of **Gain vector** instead of the **Gain vector** itself, choosing the scaling factor so that the channel's effective gain considering all paths is 0 dB.

The number of paths is the length of **Delay vector** or **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 is a scalar, then the block expands it into a vector whose size matches that of the other **vector** parameter.

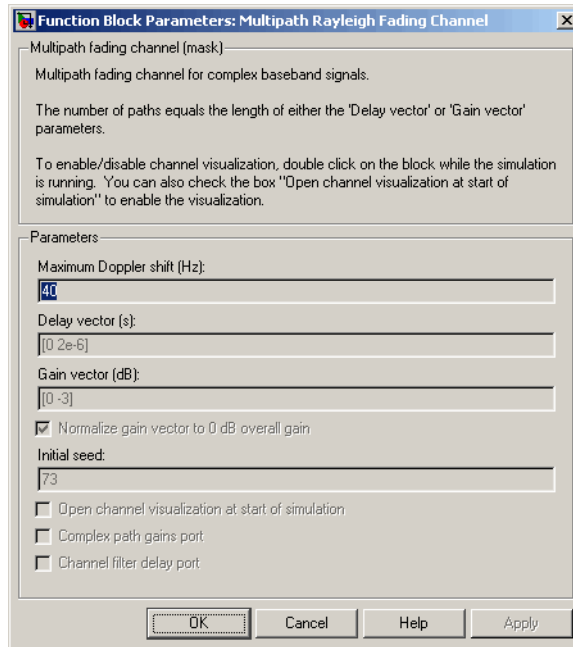
The block multiplies the input signal by samples of a Rayleigh-distributed complex random process. The scalar **Initial seed** parameter seeds the random number generator. Random numbers are

# Multipath Rayleigh Fading Channel

generated using the Ziggurat method, which is the same method used by the MATLAB `randn` function.

Double clicking this block during simulation or checking the block dialog's check-box labeled **Open channel visualization at start of simulation** will plot the channel characteristics using the channel visualization tool. See “Using the Channel Visualization Tool” in *Communications Toolbox User's Guide* for details.

## Dialog Box



### Maximum Doppler shift (Hz)

A positive scalar that indicates the maximum Doppler shift.

### Delay vector (s)

A vector that specifies the propagation delay for each path.

### Gain vector (dB)

A vector that specifies the gain for each path.

# Multipath Rayleigh Fading Channel

---

## 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 decibels.

## Initial seed

The scalar seed for the Gaussian noise generator.

## Algorithm

This implementation is based on the direct form simulator described in Reference [1] below.

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

$$f_d = (vf/c)\cos \theta$$

where  $f$  is the transmission carrier frequency and  $c$  is the speed of light. The Doppler frequency is the maximum Doppler shift arising from motion of the mobile.

## See Also

Rayleigh Noise Generator, Multipath Rician Fading Channel

## References

- [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.
- [3] Lee, William C. Y., *Mobile Communications Design Fundamentals*, 2nd Ed. New York, Wiley, 1993.

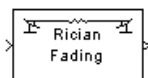
# Multipath Rician Fading Channel

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**Purpose** Simulate multipath Rician fading propagation channel

**Library** Channels

## Description



The Multipath Rician Fading Channel block implements a baseband simulation of a multipath Rician fading propagation channel. This block is useful for modeling mobile wireless communication systems when the transmitted signal can travel to the receiver along a dominant line-of-sight or direct path. For details about fading channels, see Channels in the Communication Blockset User's Guide.

This block accepts only frame-based complex signals at its input. To work with sample-based inputs, use the Frame Conversion block of Signal Processing blockset to reformat the signal. The sample time is inherited 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. The Jakes PSD (power spectral density) determines the spectrum of the Rician process.

Fading causes the signal to spread and become diffuse. The **K-factor** parameter, which is part of the statistical description of the Rician distribution, represents the ratio between direct-path (unspread) power and diffuse power. The ratio is expressed linearly, not in decibels. While the **Gain vector** parameter controls the overall gain through the channel, the **K-factor** parameter controls the gain's partition into direct and diffuse components.

The number of paths is the length of **Delay vector** or **Gain vector**, whichever is larger. If both of these parameters are vectors, they must have the same length; if exactly one of these parameters is a scalar, the block expands it into a vector whose size matches that of the other **vector** parameter.

The **Delay vector** parameter specifies the time delays in seconds, and the **Gain vector** parameter specifies the gains that apply to the input signal.

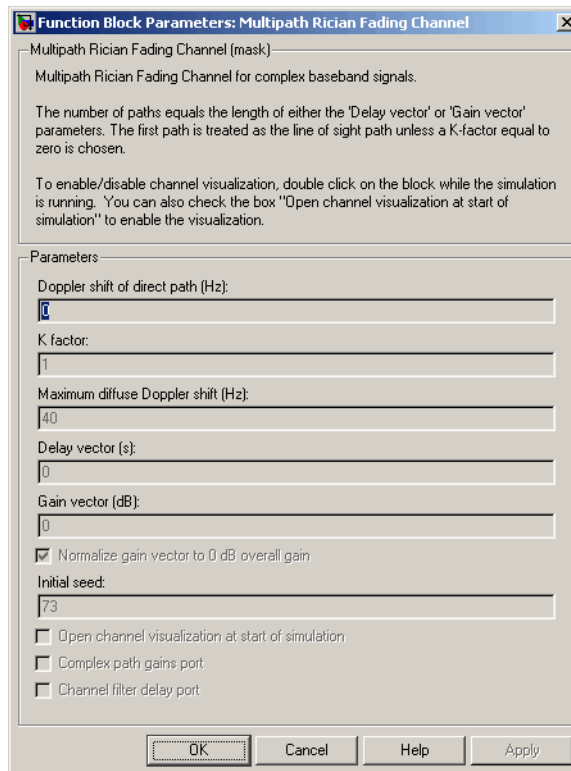


# Multipath Rician Fading Channel

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. Random numbers are generated using the Ziggurat method, which is the same method used by the MATLAB `randn` function.

Double-clicking this block during simulation or checking the block dialog's checkbox labeled **Open channel visualization at start of simulation** plots the channel characteristics using the channel visualization tool. See “Using the Channel Visualization Tool” in *Communications Toolbox User's Guide* for details.

## Dialog Box



# Multipath Rician Fading Channel

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**Doppler shift of direct path (Hz)**

A scalar that indicates the Doppler shift of the line-of-sight path.

**K-factor**

The ratio of power in the direct path to diffuse power. The ratio is expressed linearly, not in decibels.

**Maximum diffuse Doppler shift (Hz)**

A positive scalar that indicates the maximum diffuse Doppler shift.

**Delay vector(s)**

A vector that specifies the propagation delay for each 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**

Checking this box opens the channel visualization tool when a simulation is started.

**Complex path gains port**

Checking this box creates a port that outputs the values of the complex path gains for each path. This is an N-by-M multichannel frame, where N is the number of samples per frame and M is the number of discrete paths (number of delays).

**Channel filter delay port**

Checking this box creates a port that outputs the value of the delay (in samples) that is incurred by 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 “Simulation of Multipath Fading Channels: Methodology” in *Communications 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].

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

$$f_d = (vf/c)\cos \theta$$

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.

## See Also

Rician Noise Generator, Multipath Rayleigh Fading Channel

## References

- [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.

# Normalized LMS Decision Feedback Equalizer

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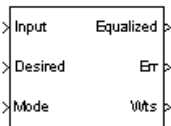
## Purpose

Equalize using decision feedback equalizer that updates weights with normalized LMS algorithm

## Library

Equalizers

## Description



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. 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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 Decision Feedback Equalizer

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- Weights output, as described in “Retrieving the Weights and Error Signal” in *Communications Blockset 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 Blockset 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

$$1 + (\text{Reference tap} - 1) / (\text{Number of samples per symbol})$$

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

Block Parameters: Normalized LMS Decision Feedback Equalizer

LMS decision feedback equalizer (mask)

Equalize a linearly modulated signal through a dispersive channel using a decision feedback equalizer and the normalized LMS algorithm.

The block computes filter weights with the normalized LMS algorithm and filters the input signal. When the number of samples per symbol is 1, the filter weights are updated once for each symbol, for a symbol spaced (i.e. T-spaced) equalizer. When the number of samples per symbol is greater than one, the weights are updated once every Nth sample, for a T/N-spaced equalizer.

The Desired input is used for training the equalizer. It expects complex constellation points.

The Leakage factor must be in the range 0 to 1. A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

If the Mode Input box is checked, the mode input toggles 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.

Parameters

Number of forward taps: 8

Number of feedback taps: 6

Number of samples per symbol: 1

Signal constellation: qammod([0:3],4)

Reference tap: 3

Step size: 0.01

Leakage: 1

Bias: 0

Initial weights: 0

Mode input port

Output error

Output weights

OK Cancel Help Apply

# Normalized 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.

**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 check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

# Normalized LMS Decision Feedback Equalizer

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## **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.

## **Output weights**

If you check this 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

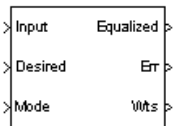
## Purpose

Equalize using linear equalizer that updates weights with normalized LMS algorithm

## Library

Equalizers

## Description



The Normalized LMS Linear Equalizer block uses a linear 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. 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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 “Retrieving the Weights and Error Signal” in *Communications Blockset User’s Guide*.

# Normalized 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 Blockset User’s Guide*.

## Equalizer Delay

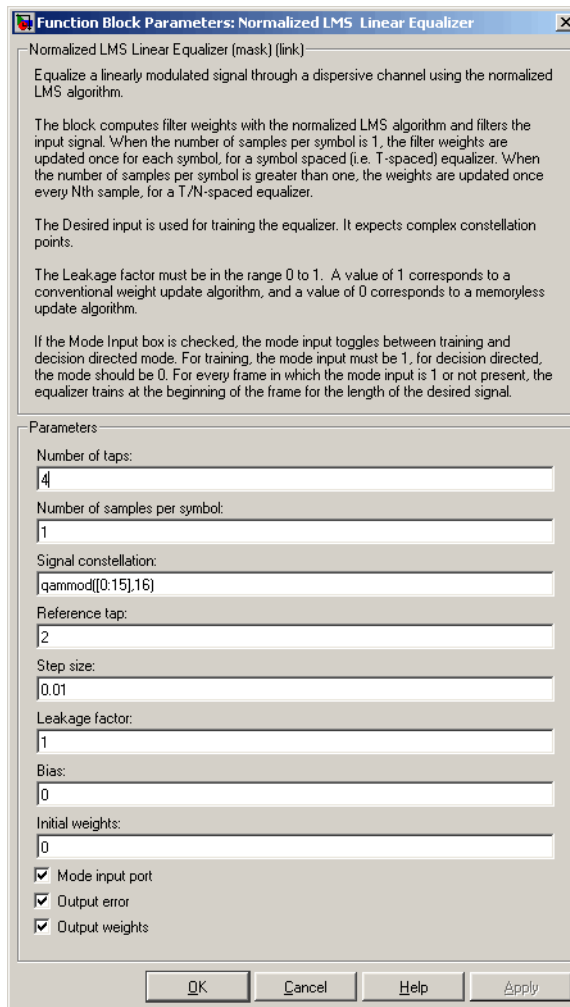
For proper equalization, you should set the **Reference tap** parameter so that it exceeds the delay, in symbols, between the transmitter’s modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer *output* is equal to

$$1 + (\text{Reference tap} - 1) / (\text{Number of samples per symbol})$$

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

# Normalized LMS Linear Equalizer

## Dialog Box



Function Block Parameters: Normalized LMS Linear Equalizer

Normalized LMS Linear Equalizer (mask) (link)

Equalize a linearly modulated signal through a dispersive channel using the normalized LMS algorithm.

The block computes filter weights with the normalized LMS algorithm and filters the input signal. When the number of samples per symbol is 1, the filter weights are updated once for each symbol, for a symbol spaced (i.e. T-spaced) equalizer. When the number of samples per symbol is greater than one, the weights are updated once every Nth sample, for a T/N-spaced equalizer.

The Desired input is used for training the equalizer. It expects complex constellation points.

The Leakage factor must be in the range 0 to 1. A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

If the Mode Input box is checked, the mode input toggles 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.

Parameters

Number of taps:  
4

Number of samples per symbol:  
1

Signal constellation:  
qammod[[0:15],16]

Reference tap:  
2

Step size:  
0.01

Leakage factor:  
1

Bias:  
0

Initial weights:  
0

Mode input port

Output error

Output weights

OK Cancel Help Apply

### 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.

# Normalized LMS Linear 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 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**

If you check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

**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.

**Output weights**

If you check this box, the block outputs the current weights.

**Examples**

See the Adaptive Equalization demo.

**References**

[1] Farhang-Boroujeny, B., *Adaptive Filters: Theory and Applications*, Chichester, England, Wiley, 1998.

# Normalized LMS Linear Equalizer

---

## **See Also**

Normalized LMS Decision Feedback Equalizer, LMS Linear Equalizer

# OQPSK Demodulator Baseband

---

**Purpose** Demodulate OQPSK-modulated data

**Library** PM, in Digital Baseband sublibrary of Modulation

## Description



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. The input can be either a scalar or a frame-based column vector. The block accepts the input data types `single` and `double`.

If the **Output type** parameter is set to `Integer`, then the block outputs integers between 0 and 3. If the **Output type** parameter is set to `Bit`, then the block outputs binary representations of such integers, in a binary-valued vector whose length is an even number.

The input symbol 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.

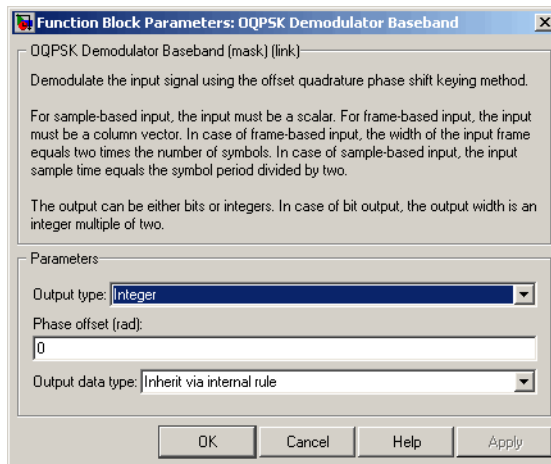
### Frame-Based Inputs

If the input is a frame-based column vector, then the block processes several integers or several pairs of bits, in each time step. In this case, the output sample time equals the input sample time, even though the symbol period is half the output period.

### Delays

The modulator-demodulator pair incurs a delay, as described in “Delays in Digital Modulation”.

## Dialog Box



### Output type

Determines whether the output consists of integers or pairs of bits.

### Phase offset (rad)

The amount by which the phase of the zeroth point of the signal constellation is shifted from  $\pi/4$ .

### 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.

## Pair Block

OQPSK Modulator Baseband

## See Also

QPSK Demodulator Baseband

# OQPSK Modulator Baseband

---

**Purpose** Modulate using offset quadrature phase shift keying method

**Library** PM, in Digital Baseband sublibrary of Modulation

## Description



The OQPSK Modulator Baseband block modulates using the offset quadrature phase shift keying method. The output is a baseband representation of the modulated signal.

If the **Input type** parameter is set to Integer, then valid input values are 0, 1, 2, and 3. In this case, the input can be either a scalar or a frame-based column vector.

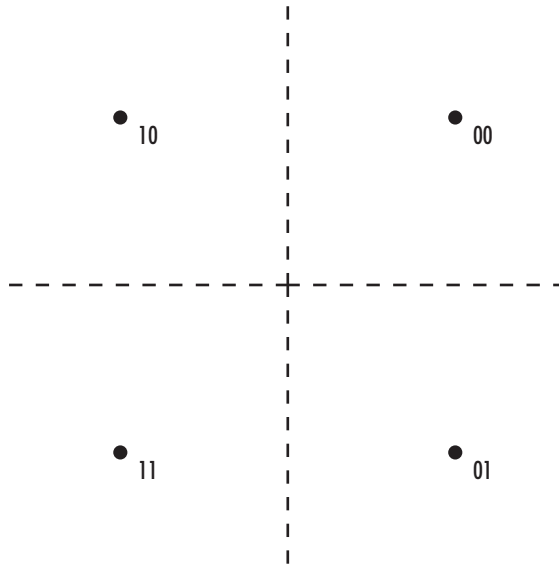
If the **Input type** parameter is set to Bit, then the input must be a binary-valued vector. In this case, the input can be either a vector of length two or a frame-based column vector whose length is an even integer.

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 symbol period is half the input period. The first output symbol is an initial condition of zero that is unrelated to the input values.

The constellation used to map bit pairs to symbols is in the following figure. If the block's **Phase offset** parameter is nonzero, then this constellation is rotated by that parameter value.





## Frame-Based Inputs

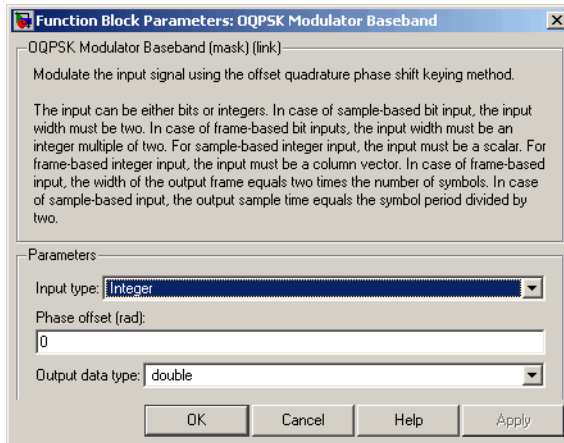
If the input is a frame-based column vector, then the block processes several integers or several pairs of bits in each time step. In this case, the output sample time equals the input sample time, even though the period of each output symbol is half the period of each integer or bit pair in the input.

## Delays

The modulator-demodulator pair incurs a delay, as described in “Delays in Digital Modulation”.

# OQPSK Modulator Baseband

## Dialog Box



### Input type

Indicates whether the input consists of integers or pairs of bits.

### Phase offset (rad)

The amount by which the phase of the zeroth point of the signal constellation is shifted from  $\pi/4$ .

### 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 Simulink Fixed Point. 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 the **Output data type** parameter and User-defined for the **Set output fraction length to** parameter.

**Pair Block**      OQPSK Demodulator Baseband

**See Also**      QPSK Modulator Baseband

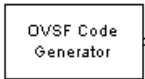
# OVSF Code Generator

---

**Purpose** Generate orthogonal variable spreading factor (OVSF) code from set of orthogonal codes

**Library** Spreading Codes

**Description**



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,  $C_N$ , which is defined 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  $C_N$ . Define  $C_{2N}$  by

$$C_{2N} = \begin{bmatrix} C_N(0) & C_N(0) \\ C_N(0) & -C_N(0) \\ C_N(1) & C_N(1) \\ C_N(1) & -C_N(1) \\ \dots & \dots \\ C_N(N-1) & C_N(N-1) \\ C_N(N-1) & -C_N(N-1) \end{bmatrix}$$

Note that  $C_N$  is only defined for  $N$  a power of 2. It follows by induction that the rows of  $C_N$  are orthogonal.

The OVSF codes can also be defined recursively by a tree structure, as shown in the following figure.



in the tree is assigned to a user, the codes  $C_{10}$ ,  $C_{20}$ ,  $C_{32}$ ,  $C_{83}$ , 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  $[0, 1, \dots, N - 1]$ , 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  $C_{N,k}$  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 0s to the left, if necessary, so that the resulting binary sequence  $x_1 x_2 \dots x_r$  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  $x_i = 0$ , and the lower branch if  $x_i = 1$ .

To reconstruct the code, recursively define a sequence of codes  $C_i$  for as follows. Let  $C_0$  be the root [1]. Assuming that  $C_i$  has been defined, for  $i < r$ , define  $C_{i+1}$  by

$$C_{i+1} = \begin{cases} C_i C_i & \text{if } x_i = 0 \\ C_i (-C_i) & \text{if } x_i = 1 \end{cases}$$

The code  $C_N$  has the specified **Spreading factor** and **Code index**.

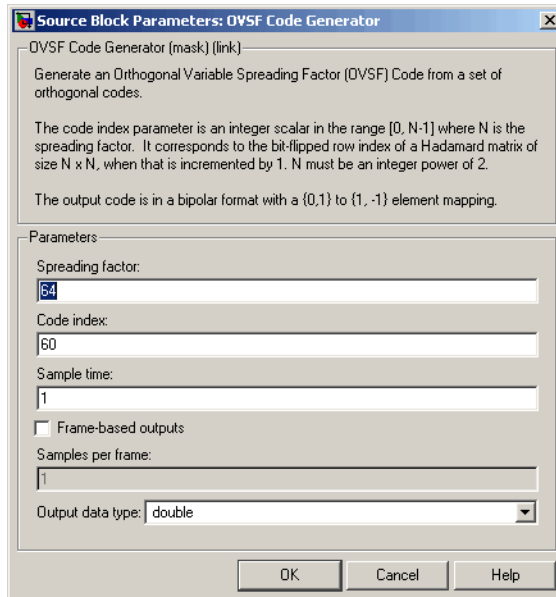
For example, to find the code with **Spreading factor** 16 and **Code index** 6, do the following:

- 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  $C_i$  according to the following table.

$i$	$x_i$	$C_i$
0		$C_0 = [1]$
1	0	$C_1 = C_0 \ C_0 = [1] \ [1]$
2	1	$C_2 = C_1 \ -C_1 = [1 \ 1] \ [-1 \ -1]$
3	1	$C_3 = C_2 \ -C_2 = [1 \ 1 \ -1 \ -1] \ [-1 \ -1 \ 1 \ 1]$
4	0	$C_4 = C_3 \ C_3 = [1 \ 1 \ -1 \ -1 \ -1 \ -1 \ 1 \ 1] \ [1 \ 1 \ -1 \ -1 \ -1 \ -1 \ 1 \ 1]$

The code  $C_4$  has **Spreading factor** 16 and **Code index** 6.

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

# OVSF Code Generator

---

**Spreading factor**

Positive integer that is a power of 2, specifying the length of the code.

**Code index**

Integer in the range [0, 1, ... , 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 the **Frame-based outputs** check box.

**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**

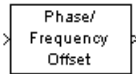
Hadamard Code Generator, Walsh Code Generator



**Purpose** Apply phase and frequency offsets to complex baseband signal

**Library** RF Impairments

**Description** The Phase/Frequency Offset block applies phase and frequency offsets to an incoming signal.



This block operates without any delay, and accepts real and complex inputs of data type `double` or `single`. Its output data type matches that of its input.

## Phase Offset

The block applies a phase offset to the input signal, specified by the **Phase offset** parameter.

## Frequency Offset

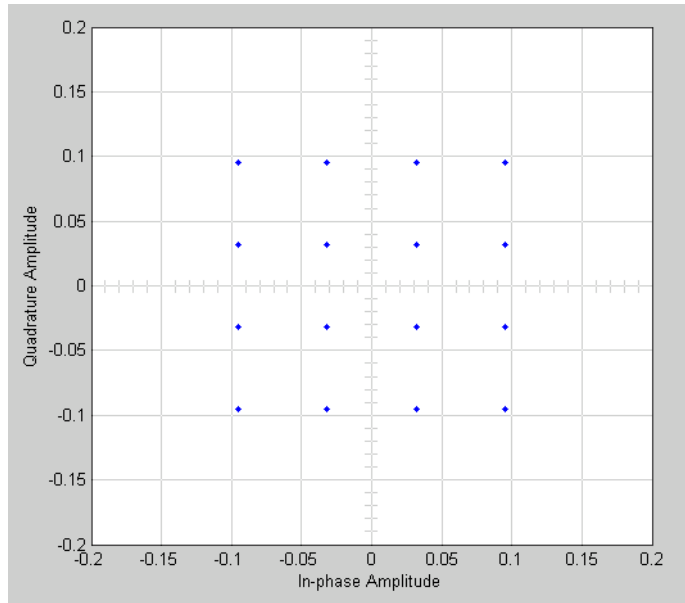
The block applies a frequency offset to the signal that is specified by the **Frequency offset** parameter. Alternatively, this information may be provided to the block through an input port that is enabled when the **Frequency offset from port** checkbox is selected.

The block supports signals of single-samples, multiple-sample single-channel frames, single-sample multi-channel frames, and multi-sample multi-channel frames. The two block parameters can be scalars, vectors, or matrices, as appropriate with the input signal's dimensions. If the input and frequency signals are both frame-based, they must both have the same frame size. If the frequency signal is multi-channel, it must have either one channel or the same number of channels as the input signal. If it is not a scalar, the number of samples in the channel must match the input signal.

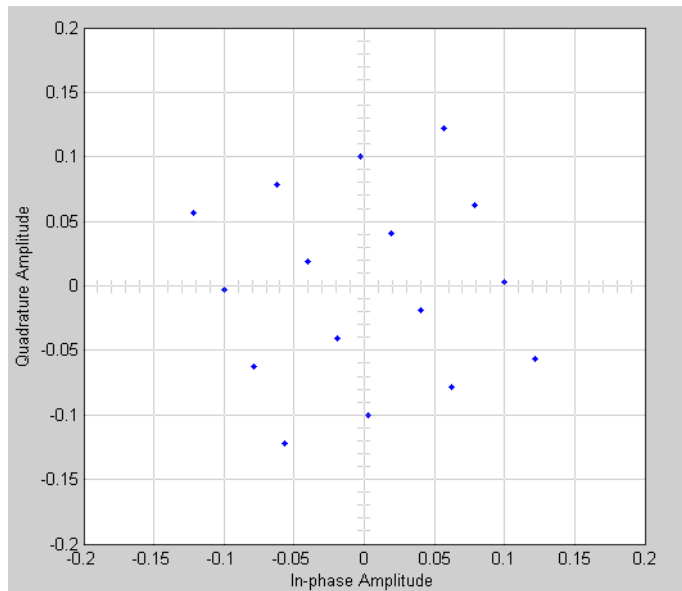
The effects of changing the block's parameters are illustrated by the following scatter plots of a signal modulated by 16-ary quadrature amplitude modulation (QAM). The usual 16-ary QAM constellation without the effect of the Phase/Frequency Offset block is shown in the first scatter plot:

# Phase/Frequency Offset

---



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:

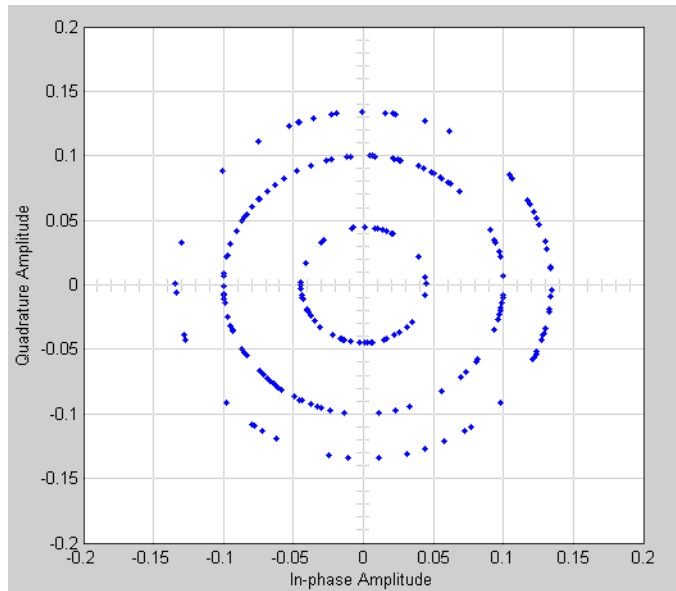


Observe that each point in the constellation is rotated by a 20 degree angle counterclockwise.

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 “Scatter Plot Examples” for a description of the model that generates this plot.

## **I/O Ports**

Only 1-D (unoriented) and 2-D ports are supported, as well as 2-D frame-based (single or multichannel) ports.

Scalar (length-one) sample-based data ports are also supported, but nonscalar sample-based ports are not.

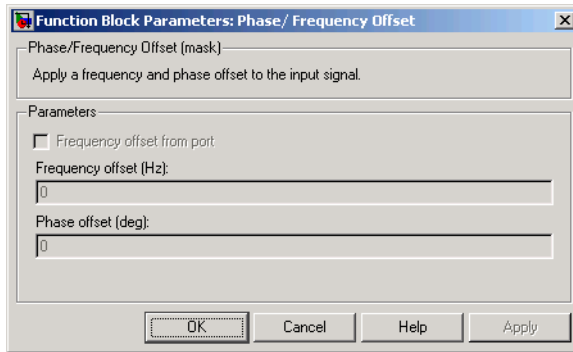
**Interdependent Parameter-Port Dimensions Table**

<b>Data I/O Frmstat</b>	<b>No. of Dim.</b>	<b>Data I/O Dim.</b>	<b>Frame Size</b>	<b>No. of Channels</b>	<b>Freq. or Phase Offset Parameter Dim.</b>
Any	Any	Scalar	1	1	Scalar
Frame-based	2	$M \times 1$	$M$	1	$M, M \times 1, 1 \times M, 1, 1 \times 1$
Frame-based	2	$1 \times N$	1	$N$	$N, N \times 1, 1 \times N, 1, 1 \times 1$
Frame-based	2	$M \times N$	$M$	$N$	$M \times N, N, N \times 1, 1 \times N, 1, 1 \times 1$
Sample-based	2	$M \times 1$ (Errors — not currently supported)			
Sample-based	2	$1 \times N$ (Errors — not currently supported)			
Sample-based	2	$M \times N$ (Errors — not currently supported)			
Sample-based	1	$L$ (Errors — not currently supported)			

# 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.

### Phase offset

Specifies the phase offset in degrees.

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

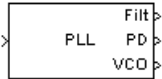
## Purpose

Implement phase-locked loop to recover phase of input signal

## Library

Components sublibrary of Synchronization

## Description



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'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. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.

```
[num, den] = cheby2(3,40,100,'s')
```

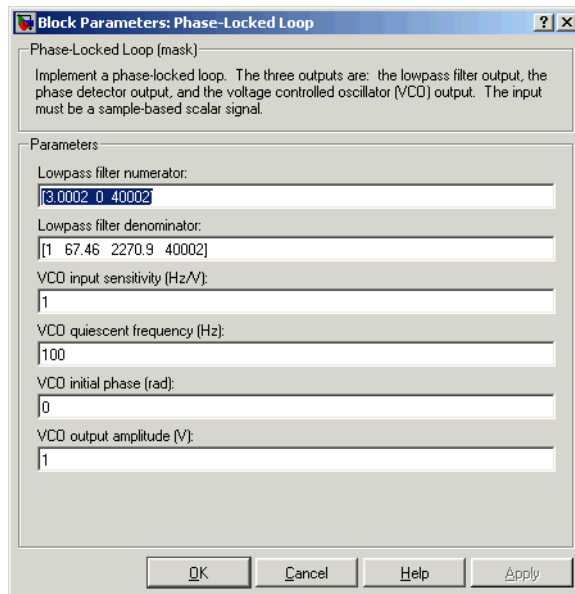
- A voltage-controlled oscillator (VCO). You specify characteristics of the VCO using the **VCO quiescent frequency**, **VCO initial phase**, and **VCO output amplitude** parameters.

The input signal represents the received signal. The input must be a sample-based scalar 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's 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's transfer function, represented as a vector that lists the coefficients in order of descending powers of  $s$ .

### VCO input sensitivity (Hz/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 (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**

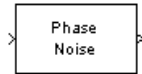
For more information about phase-locked loops, see the works listed in “Selected Bibliography for Synchronization” in *Communications Blockset User’s Guide*.

# Phase Noise

**Purpose** Apply receiver phase noise to complex baseband signal

**Library** RF Impairments

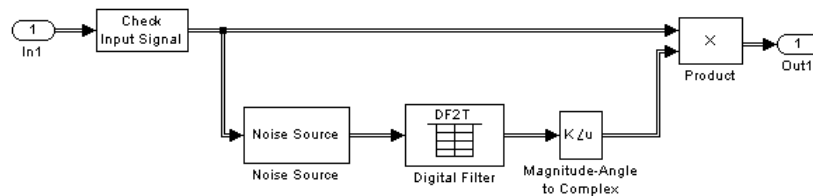
**Description** The Phase Noise block applies phase noise to a complex, baseband signal. The block applies the phase noise as follows:



**1** Generates additive white Gaussian noise (AWGN) and filters it with a digital filter.

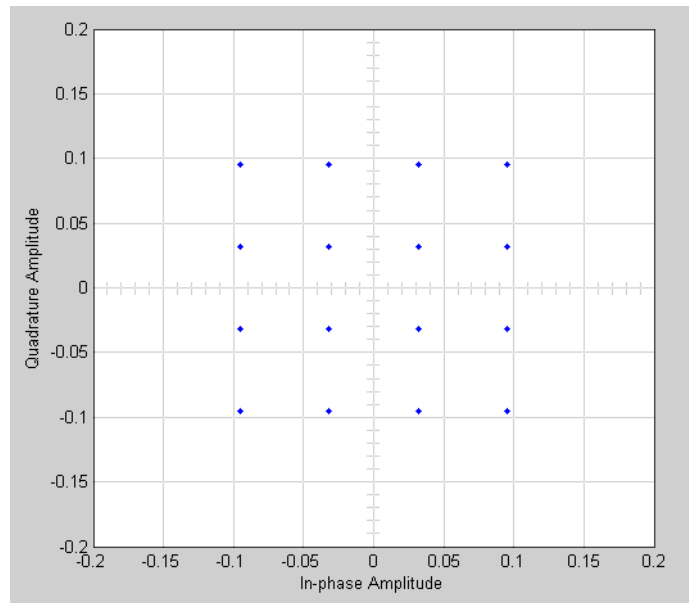
**2** Adds the resulting noise to the angle component of the input signal.

You can view the block's implementation of phase noise by right-clicking on the block and selecting **Look under mask**. This displays the following figure:



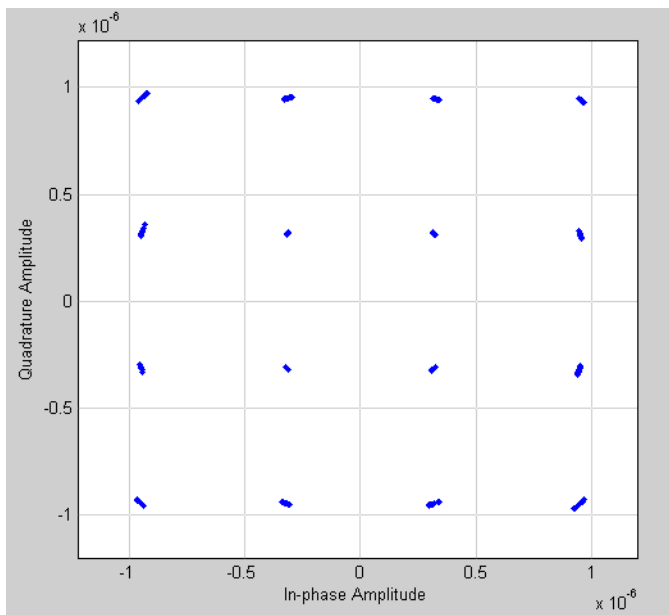
You can view the construction of the Noise Source subsystem by double-clicking it.

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:



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 (dBc/Hz)** set to -70 and **Frequency offset (Hz)** set to 100:

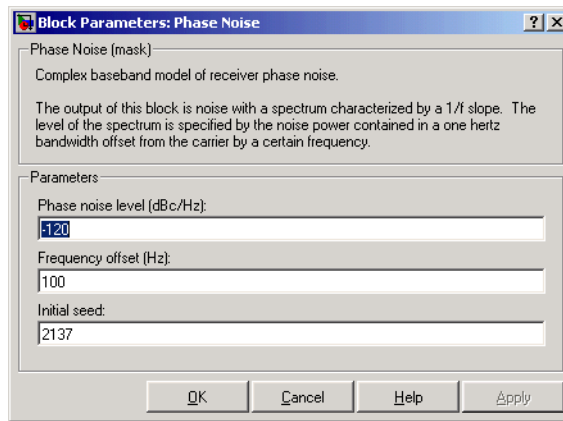
# Phase Noise



This plot is generated by the model described in “Scatter Plot Examples” with the following parameter settings for the Rectangular QAM Modulator Baseband block:

- **Normalization method** set to Average Power
- **Average power (watts)** set to  $1e-12$

## Dialog Box



### Phase noise level (dBc/Hz)

Scalar specifying the phase noise level.

### Frequency offset (Hz)

Scalar specifying the frequency offset in Hertz.

### Initial seed

Nonnegative integer specifying the initial seed for the random number generator the block uses to generate noise.

## See Also

Phase/Frequency Offset

## References

- [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

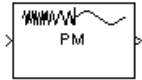
# PM Demodulator Passband

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**Purpose** Demodulate PM-modulated data

**Library** Analog Passband Modulation, in Modulation

## Description

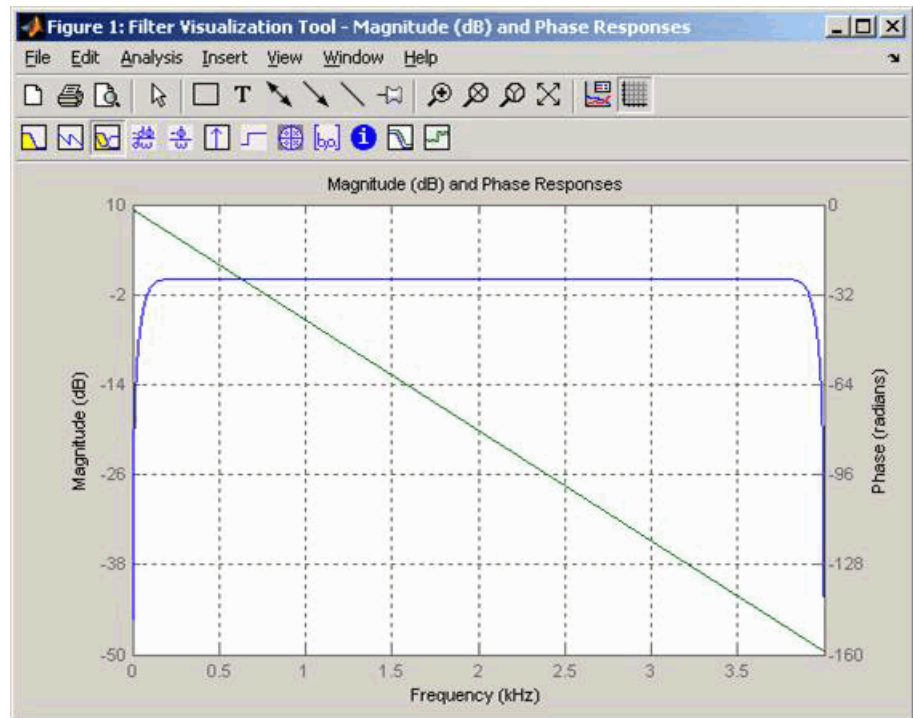


The PM Demodulator Passband block demodulates a signal that was modulated using phase modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real sample-based scalar signals.

For best results, use a carrier frequency which is estimated to be larger than 10% of your input signal's sample time. This is due to the implementation of the Hilbert transform by means of a filter.

In the following example, we sample a 10Hz 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 time (8000 samples per second, in this example) or equivalently, a carrier frequency larger than 400Hz, 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.

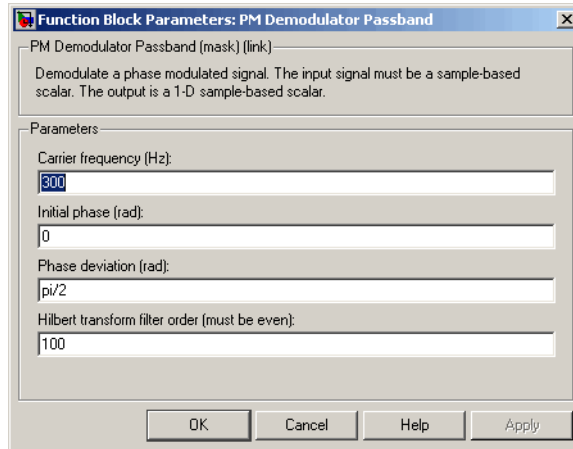
Typically, an appropriate **Carrier frequency** value is much higher than the highest frequency of the input signal. By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the **Carrier frequency** parameter.

# PM Demodulator Passband

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This block works only with real inputs of type double. This block is not suited to be placed inside a triggered subsystem.

## Dialog Box



### Carrier frequency (Hz)

The frequency of the carrier.

### Initial phase (rad)

The initial phase of the carrier in radians.

### Frequency deviation (Hz)

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



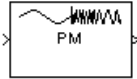
## Purpose

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 sample-based scalar signals.

If the input is  $u(t)$  as a function of time  $t$ , then the output is

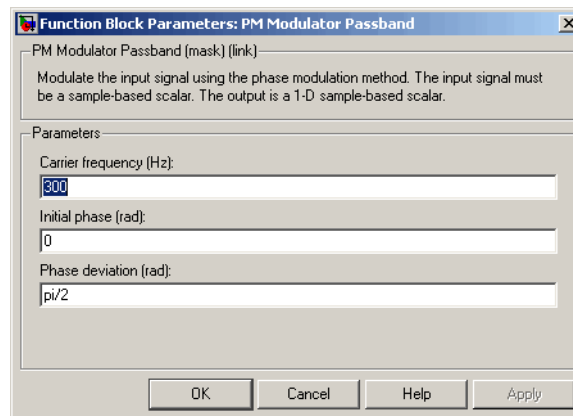
$$\cos(2\pi f_c t + K_c u(t) + \theta)$$

where  $f_c$  is the **Carrier frequency** parameter,  $\theta$  is the **Initial phase** parameter, and  $K_c$  is the **Modulation constant** 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 is not suited to be placed inside a triggered subsystem.

## Dialog Box



# PM Modulator Passband

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**Carrier frequency (Hz)**

The frequency of the carrier.

**Initial phase (rad)**

The initial phase of the carrier in radians.

**Phase deviation**

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

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polynomial in  $z$ ,  $g_r z^r + g_{r-1} z^{r-1} + g_{r-2} z^{r-2} + \dots + g_0$ . The coefficient  $g_k$  is 1 if there is a connection from the  $k$ th register, as labeled in the preceding diagram, to the adder. The leading term  $g_r$  and the constant term  $g_0$  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, [1 0 0 0 0 0 1 0 1] and [8 2 0] 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.

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**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.

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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
<b>Generator polynomial</b>	$g1 = [1\ 0\ 0\ 0\ 0\ 0\ 1\ 0\ 1]$	$g2 = [8\ 2\ 0]$
Degree of generator polynomial	8, which is $\text{length}(g1) - 1$	8
<b>Initial states</b>	$[1\ 0\ 0\ 0\ 0\ 0\ 1\ 0]$	$[1\ 0\ 0\ 0\ 0\ 0\ 1\ 0]$

**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.

	<b>T = 0</b>	<b>T = 1</b>	<b>T = 2</b>	<b>...</b>	<b>T = d</b>	<b>T = d+1</b>
<b>Shift = 0</b>	$x_0$	$x_1$	$x_2$	...	$x_d$	$x_{d+1}$
<b>Shift = d</b>	$x_d$	$x_{d+1}$	$x_{d+2}$	...	$x_{2d}$	$x_{2d+1}$

Alternatively, you can set **Output mask vector (or scalar shift value)** to a binary vector, corresponding to a polynomial in  $z$ ,  $m_{r-1}z^{r-1} + m_{r-2}z^{r-2} + \dots + m_1z + m_0$ , of degree at most  $r-1$ . The mask vector

# PN Sequence Generator

corresponding to a shift of  $d$  is the vector that represents  $m(z) = z^d$  modulo  $g(z)$ , where  $g(z)$  is the generator polynomial. For example, if the degree of the generator polynomial is 4, then the mask vector corresponding to  $d = 2$  is  $[0 \ 1 \ 0 \ 0]$ , 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 Toolbox function `shift2mask`.

You can use an external signal to reset the values of the internal shift register to the initial state by selecting the **Reset on nonzero input** check box. 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  $[1 \ 0 \ 0 \ 1 \ 1 \ 0 \ 1 \ 1]$  when there is no reset. You then select the **Reset on nonzero input** check box and input a reset signal  $[0 \ 0 \ 0 \ 1]$ . The following table shows three possibilities for the properties of the reset signal and the PN Sequence Generator block.

Reset Signal Properties	PN Sequence Generator block	Reset Signal, Output Signal																				
Sample-based Sample time = 1	Sample-based Sample time = 1	<div style="text-align: center;">Reset</div> <table style="border-collapse: collapse; margin: auto;"> <tr> <td style="padding: 0 10px;">0</td> <td style="padding: 0 10px;">0</td> <td style="padding: 0 10px;">0</td> <td style="border: 1px solid black; padding: 0 5px;">1</td> <td style="padding: 0 10px;"></td> <td style="padding: 0 10px;"></td> <td style="padding: 0 10px;"></td> <td style="padding: 0 10px;"></td> <td style="padding: 0 10px;"></td> <td style="padding: 0 10px;"></td> </tr> <tr> <td style="padding: 0 10px;">1</td> <td style="padding: 0 10px;">0</td> <td style="padding: 0 10px;">0</td> <td style="border: 1px solid black; padding: 0 5px;">1</td> <td style="padding: 0 10px;">0</td> <td style="padding: 0 10px;">0</td> <td style="padding: 0 10px;">1</td> <td style="padding: 0 10px;">1</td> <td style="padding: 0 10px;">0</td> <td style="padding: 0 10px;">1</td> </tr> </table>	0	0	0	1							1	0	0	1	0	0	1	1	0	1
0	0	0	1																			
1	0	0	1	0	0	1	1	0	1													

Reset Signal Properties	PN Sequence Generator block	Reset Signal, Output Signal
Frame-based <b>Sample time = 1</b> <b>Samples per frame = 2</b>	Frame-based <b>Sample time = 1</b> <b>Samples per frame = 2</b>	Reset 0 0 0 <span style="border: 1px solid black; padding: 2px;">1</span> 1 0 0 <span style="border: 1px solid black; padding: 2px;">1</span> 0 0 1 1 0 1 1
Sample-based <b>Sample time = 2</b> <b>Samples per frame = 1</b>	Frame-based <b>Sample time = 1</b> <b>Samples per frame = 2</b>	Reset 0 0 0 <span style="border: 1px solid black; padding: 2px;">1</span> 1 0 0 1 0 0 <span style="border: 1px solid black; padding: 2px;">1</span> 0 0 1 1 0 1 1

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 **Generator polynomial** to a value from the following table. See [1] for more information about the shift-register configurations that these polynomials represent.

# PN Sequence Generator

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<b>r</b>	<b>Generator Polynomial</b>	<b>r</b>	<b>Generator Polynomial</b>
2	[2 1 0]	21	[21 19 0]
3	[3 2 0]	22	[22 21 0]
4	[4 3 0]	23	[23 18 0]
5	[5 3 0]	24	[24 23 22 17 0]
6	[6 5 0]	25	[25 22 0]
7	[7 6 0]	26	[26 25 24 20 0]
8	[8 6 5 4 0]	27	[27 26 25 22 0]
9	[9 5 0]	28	[28 25 0]
10	[10 7 0]	29	[29 27 0]
11	[11 9 0]	30	[30 29 28 7 0]
12	[12 11 8 6 0]	31	[31 28 0]
13	[13 12 10 9 0]	32	[32 31 30 10 0]
14	[14 13 8 4 0]	33	[33 20 0]
15	[15 14 0]	34	[34 15 14 1 0]
16	[16 15 13 4 0]	35	[35 2 0]
17	[17 14 0]	36	[36 11 0]
18	[18 11 0]	37	[37 12 10 2 0]
19	[19 18 17 14 0]	38	[38 6 5 1 0]
20	[20 17 0]	39	[39 8 0]
40	[40 5 4 3 0]	47	[47 14 0]
41	[41 3 0]	48	[48 28 27 1 0]
42	[42 23 22 1 0]	49	[49 9 0]
43	[43 6 4 3 0]	50	[50 4 3 2 0]
44	[44 6 5 2 0]	51	[51 6 3 1 0]



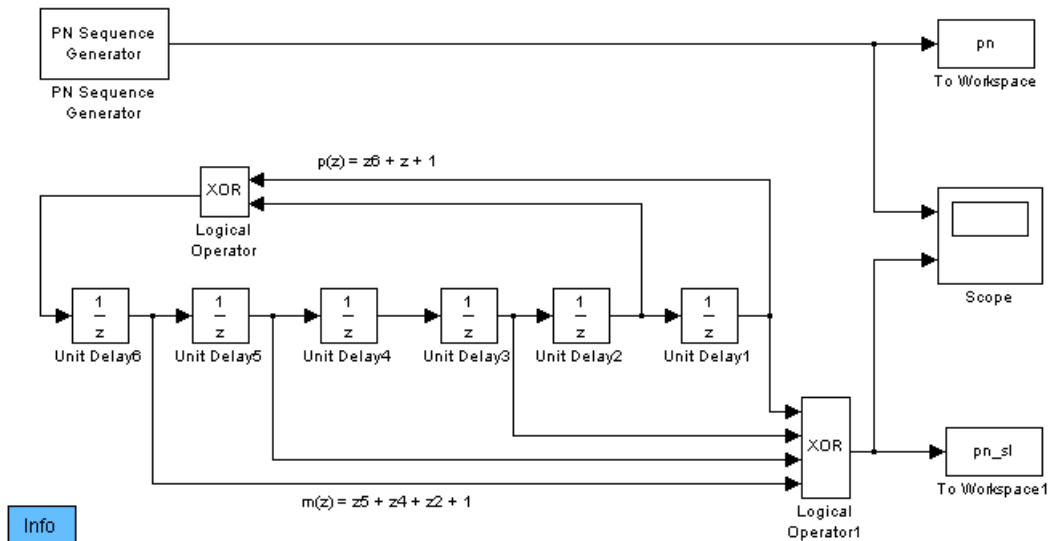
# PN Sequence Generator

r	Generator Polynomial	r	Generator Polynomial
45	[45 4 3 1 0]	52	[52 3 0]
46	[46 21 10 1 0]	53	[53 6 2 1 0]

## 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. The model is located at: *matlabroot/help/toolbox/commblocks/commblocks\_examples/doc\_pnseq2*.

### PN Sequence Generation



Info

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,

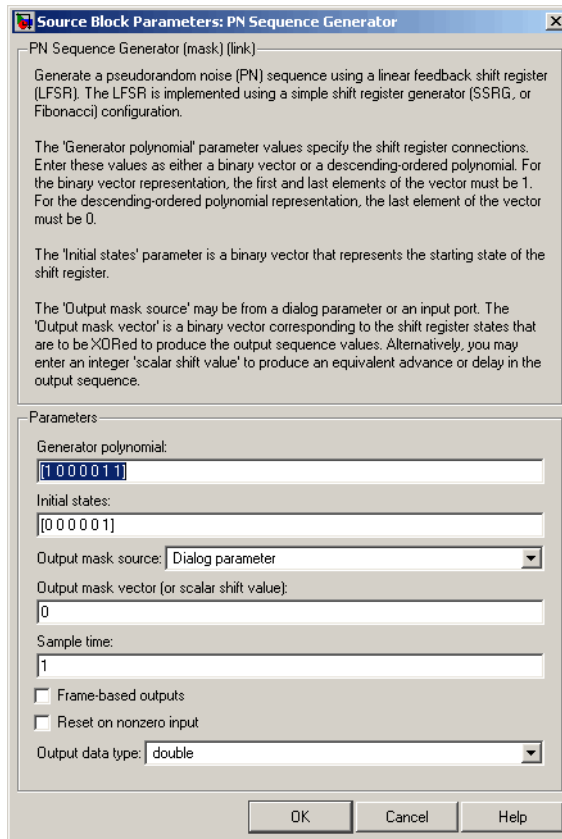
# PN Sequence Generator

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**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.



## Dialog Box

Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

### Generator polynomial

Polynomial that determines the shift register’s feedback connections.

### Initial states

Vector of initial states of the shift registers.

# PN Sequence Generator

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## **Output mask source**

Specifies how output mask information is given to the block.

When set to `Dialog` parameter, the field **Output mask vector (or scalar shift value)** is enabled for user input. When set to `Input port`, a port appears on the block icon for output mask vector (or scalar shift value) input.

## **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.

## **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 the **Frame-based outputs** check box.

## **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 a boolean or double. By default, the block sets this to double.

## **See Also**

Kasami Sequence Generator, Scrambler

## **References**

[1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.

[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.

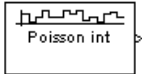
# Poisson Integer Generator

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**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.

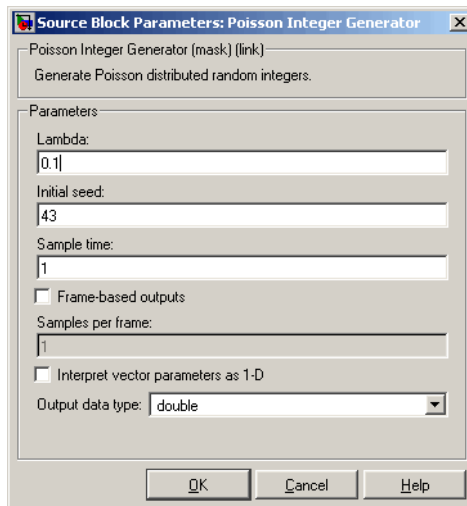
You can use the Poisson Integer Generator to generate noise in a binary transmission channel. In this case, the Poisson parameter **Lambda** should be less than 1, usually much less.

## 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 “Signal Attribute Parameters for Random Sources” in *Communications Blockset 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.

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

### 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

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## **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**

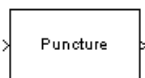
Random Integer Generator; `poissrnd` (Statistics Toolbox)



**Purpose** Output elements which correspond to 1s in binary Puncture vector

**Library** Sequence Operations

## Description



The Puncture block creates an output vector by removing selected elements of the input vector and preserving others. The input can be a real or complex vector of length  $K$ . The block determines which elements to remove or preserve by using the binary **Puncture vector** parameter:

- If **Puncture vector**( $k$ ) = 0, then the  $k$ th element of the input vector does not become part of the output vector.
- If **Puncture vector**( $k$ ) = 1, then the  $k$ th element of the input vector is preserved in the output vector.

Here,  $k$  is between 1 and  $K$ . The preserved elements appear in the output vector in the same order in which they appear in the input vector.

The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, `double`, and `fixed-point`. The data type of the output will be the same as that of the input signal.

### Frame-Based Processing

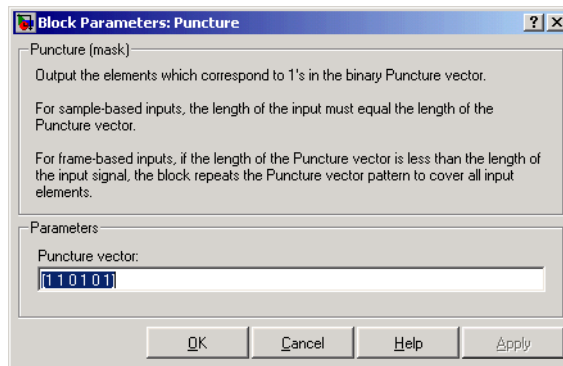
If the input is frame-based, then both it and the **Puncture vector** parameter must be column vectors. The length of the **Puncture vector** parameter must divide  $K$ . The block repeats the puncturing pattern, if necessary, to cover all input elements. That is, in the bulleted items above you can replace **Puncture vector**( $k$ ) by **Puncture vector**( $n$ ), where

$$n = \text{mod}(k, \text{length}(\mathbf{Puncture\ vector}))$$

and `mod` is the modulus function (`mod` in MATLAB).

# Puncture

## Dialog Box



## Puncture vector

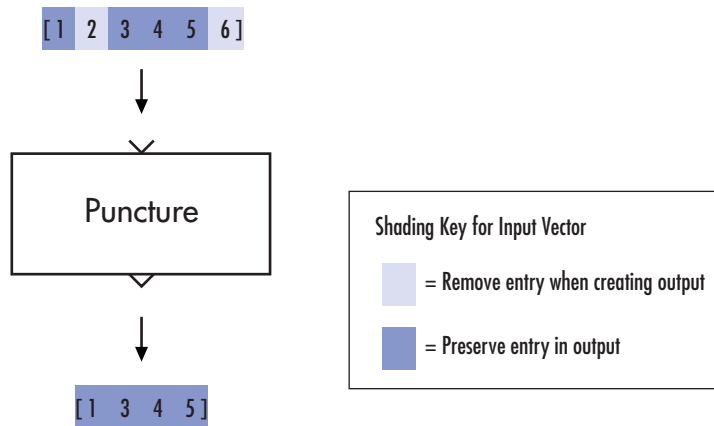
A binary vector whose pattern of 0s (1s) indicates which elements of the input the block should remove (preserve).

## 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  $[1:6]$ , using this **Puncture vector** parameter.



## See Also

Insert Zero

# QPSK Demodulator Baseband

**Purpose** Demodulate QPSK-modulated data

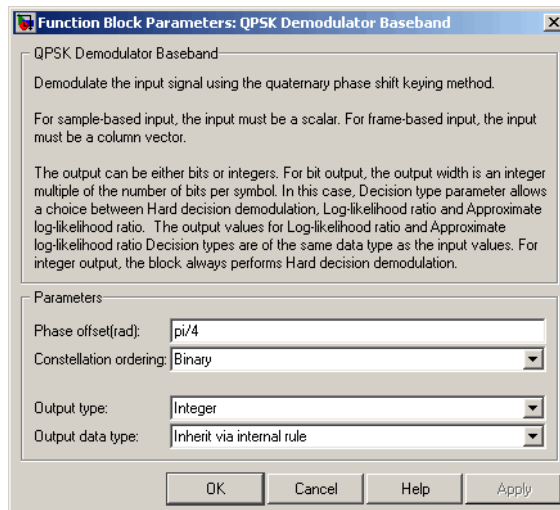
**Library** PM, in Digital Baseband sublibrary of Modulation

**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 discrete-time complex signal. The input can be either a scalar or a frame-based column vector. The block accepts the input data types `single` and `double`.

## 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 pairs of 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.

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.

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

This field appears when Bit is selected in the drop-down list **Output type**.

Specifies the use of hard decision, LLR, or approximate LLR during demodulation. See “Exact LLR Algorithm” and “Approximate LLR Algorithm” for algorithm details.

## 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, when **Decision type** is set to Hard decision, the output can be int8, uint8, int16, uint16, int32, uint32,

# QPSK Demodulator Baseband

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boolean, single, or double. When **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is inherited from the input (i.e., if the input is of data type double, the output is also of data type double).

## 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.

## Noise variance

This field appears when **Noise variance source** is set to Dialog.

If this value is very small (i.e., SNR is very high), LLR computations may yield Inf or -Inf because the LLR algorithm would involve computing exponentials of very large or very small numbers using finite precision arithmetic. In such cases, use of approximate LLR is recommended, as its algorithm does not involve computing exponentials.

## Pair Block

QPSK Modulator Baseband

## See Also

M-PSK Demodulator Baseband, BPSK Demodulator Baseband, DQPSK Demodulator Baseband

**Purpose** Modulate using quaternary phase shift keying method

**Library** PM in Digital Baseband sublibrary of Modulation

## Description



The QPSK Modulator Baseband block modulates using the quaternary phase shift keying method. The output is a baseband representation of the modulated signal.

## Inputs and Constellation Types

If the **Input type** parameter is set to Integer, then valid input values are 0, 1, 2, and 3. If **Constellation ordering** is set to Binary, for input  $m$  the output symbol is

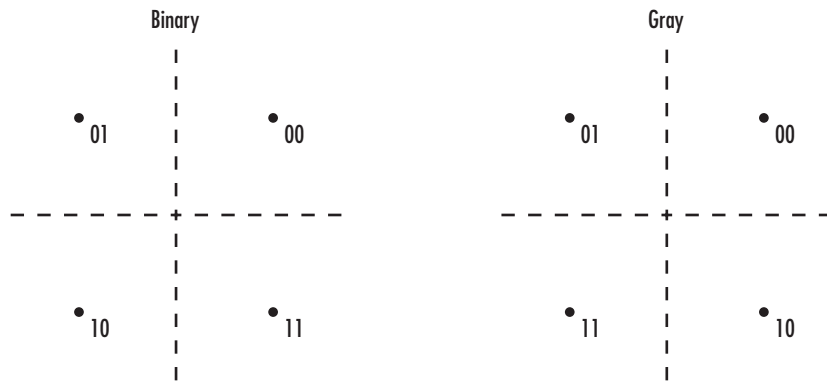
$$\exp(j\theta + j\pi m/2)$$

where  $\theta$  is the **Phase offset** parameter (see the following figure for Gray constellation ordering). In this case, the input can be either a scalar or a frame-based column vector.

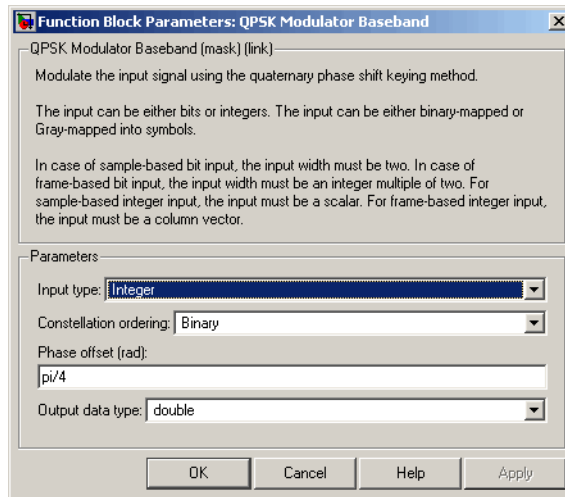
For integer inputs, the block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. For bit inputs, the block can accept `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, and `double`.

If the **Input type** parameter is set to Bit, then the input contains pairs of binary values. The input can be either a vector of length 2 or a frame-based column vector whose length is an even integer. If the **Phase offset** parameter is set to  $\pi/4$ , then the block uses one of the signal constellations in the following figure, depending on whether the **Constellation ordering** parameter is set to Binary or Gray.

# QPSK Modulator Baseband



## Dialog Box



### Input type

Indicates whether the input consists of integers or pairs of bits.

### Constellation ordering

Determines how the block maps each pair of input bits or input integers to constellation symbols.



**Phase offset (rad)**

The phase of the zeroth point of the signal constellation.

**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 Simulink Fixed Point. 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.

# QPSK Modulator Baseband

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## **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.

## **Pair Block**

QPSK Demodulator Baseband

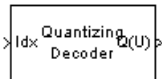
## **See Also**

M-PSK Modulator Baseband, BPSK Modulator Baseband, DQPSK Modulator Baseband

**Purpose** Decode quantization index according to codebook

**Library** Source Coding

## Description



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  $N-1$ , then the output is the  $(k+1)$ st element of **Quantization codebook**.

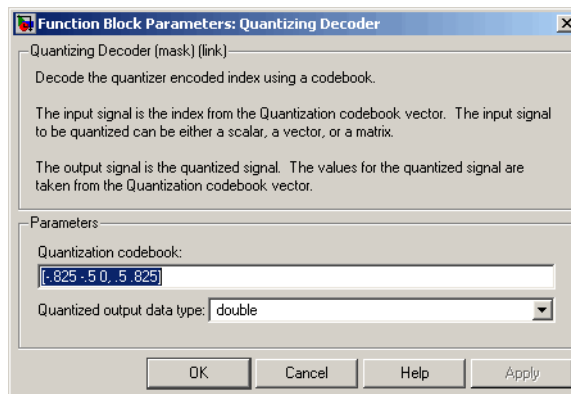
The input can be either a scalar, a vector, or a matrix. The input must be a discrete-time signal. This block processes each vector element independently.

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**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.

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## Dialog Box



# Quantizing Decoder

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## **Quantization codebook**

A real vector that prescribes the output value corresponding to each nonnegative integer of the input.

## **Pair Block**

Quantizing Encoder

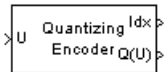
## **See Also**

Scalar Quantizer (Signal Processing Blockset)

**Purpose** Quantize signal using partition and codebook

**Library** Source Coding

## Description



The Quantizing Encoder block quantizes the input signal according to the **Partition** vector and encodes the input signal according to the **Codebook** vector. The input signal can be either a scalar, a vector, or a matrix. This block processes each vector element independently.

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  $x < P(1)$
- $m$  if  $P(m) < x < P(m+1)$
- $n$  if  $P(n) < x$

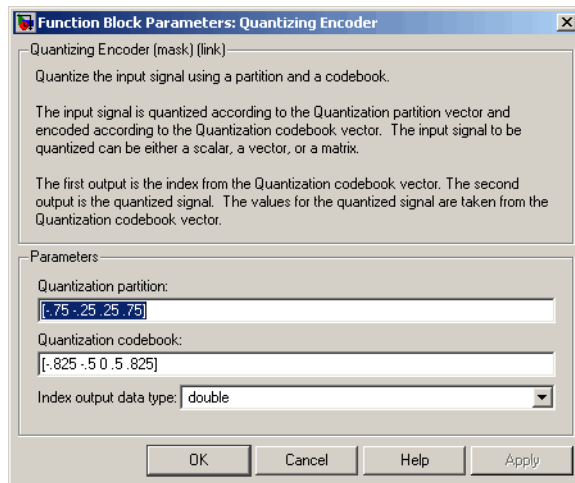
The **Quantization codebook** parameter, whose length is  $n+1$ , prescribes a value for each partition in the quantization. The first element of **Quantization codebook** is the value for the interval between negative infinity and the first element of  $P$ . The second output signal from this block contains the quantization of the input signal based on the quantization indices and prescribed values.

You can use the function `lloyd`s in Communications Toolbox with a representative sample of your data as training data, to obtain appropriate partition and codebook parameters.

# Quantizing Encoder

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## Dialog Box



### Quantization partition

The vector of endpoints of the partition intervals.

### Quantization codebook

The vector of output values assigned to each partition.

## Pair Block

Quantizing Decoder

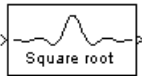
## See Also

Scalar Quantizer (Signal Processing Blockset), 11oyds (Communications Toolbox)

**Purpose** Filter input signal, possibly downsampling, using raised cosine FIR filter

**Library** Comm Filters

## Description



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 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:  $2 * N * \text{Group delay} + 1$ , 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:

- $20 \log_{10}(\text{Input samples per symbol}(N) \times \text{Linear amplitude filter gain})$  for a normal filter.
- $20 \log_{10}(\text{Input samples per symbol}(N) \times \text{Linear amplitude filter gain})$  for a square root filter.

### Downsampling the Filtered Signal

To have the block downsample the filtered signal, set the **Output mode** parameter to Downsampling. If  $L$  is the **Downsampling factor** parameter value, then the block retains  $1/L$  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  $1, L+1, 2*L+1, 3*L+1$ , etc.

# Raised Cosine Receive Filter

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- 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 case above.

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 and Output Signals

The input signal must be a scalar or a frame-based column vector. double, single, and fixed-point data types are supported. Set the **Input sampling mode** parameter according to whether the input is sample-based or frame-based.

If **Output mode** is set to None, then the input and output signals share the same sampling mode, sample time, and vector length.

If **Output mode** is set to Downsampling and **Downsampling factor** is  $L$ , then  $L$  and the input sampling mode determine characteristics of the output signal:

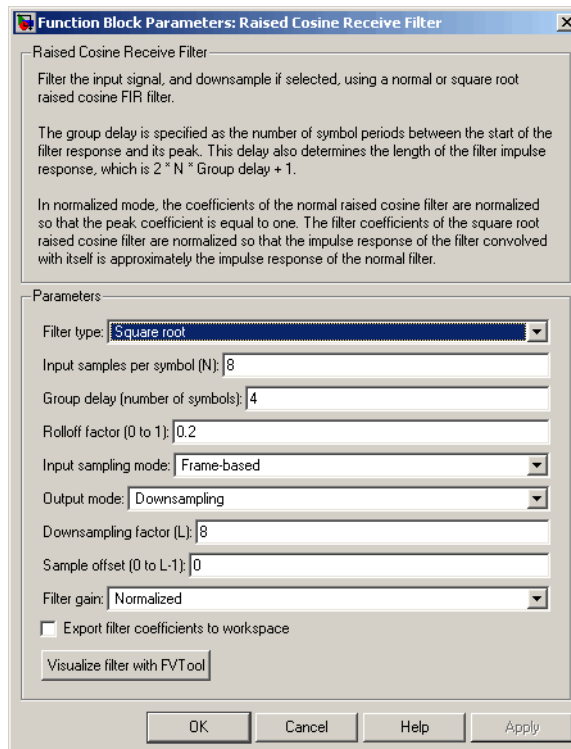
- If the input is sample-based, then the output is sample-based and the output sample time is  $1/L$  times the input sample time.
- If the input is frame-based, then the output is a frame-based vector whose length is  $1/L$  times the length of the input vector. The output frame period equals the input frame period.

## 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.



## 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.

### Rolloff factor

The rolloff factor for the filter, a real number between 0 and 1.

# Raised Cosine Receive Filter

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## **Input sampling mode**

The type of input signal: Frame-based or Sample-based.

## **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**

If you check this box, then the block creates a variable in the MATLAB workspace that contains the filter coefficients.

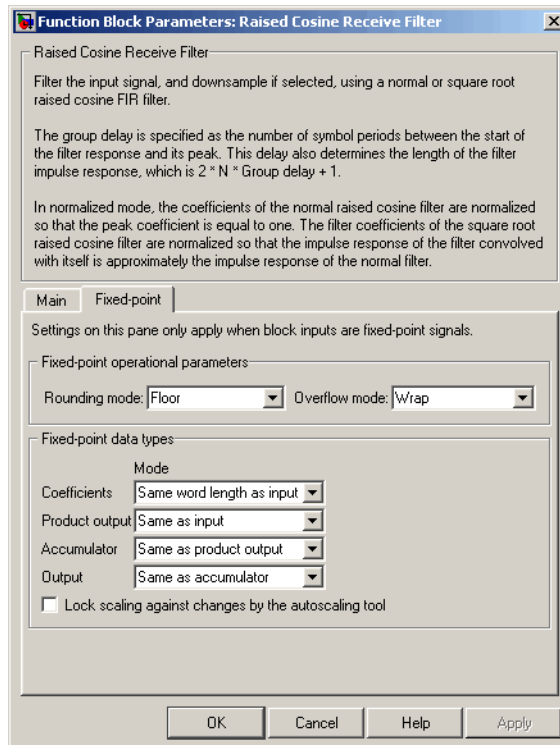
## **Coefficient variable name**

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.



## Rounding mode

Select the rounding mode for fixed-point operations. The filter coefficients do not obey this parameter; they always round to Nearest.

## Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

# Raised Cosine Receive Filter

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## 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 *Signal Processing Blockset 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 *Signal Processing*

*Blockset 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.

## **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.

# Raised Cosine Receive Filter

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- 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.

**Pair Block**      Raised Cosine Transmit Filter

**See Also**      Gaussian Filter, `rcosine`, `rcosflt`

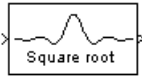
## Purpose

Upsample and filter input signal using raised cosine FIR filter

## Library

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

$$h(t) = \frac{\sin(\pi t / T)}{(\pi t / T)} \cdot \frac{\cos(\pi R t / T)}{(1 - 4R^2 t^2 / T^2)}$$

The impulse response of a square root raised cosine filter with rolloff factor  $R$  is

$$h(t) = 4R \frac{\cos((1 + R)\pi t / T) + \frac{\sin((1 - R)\pi t / T)}{(4Rt / T)}}{\pi\sqrt{T} \left(1 - (4Rt / T)^2\right)}$$

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  $2 * N * \mathbf{Group\ delay} + 1$ .

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

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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.

If the **Filter gain** parameter is chosen to be User-specified, then the passband gain of the filter is:

- $20 \log_{10}(\text{Upsampling factor}(N) \times \text{Linear amplitude filter gain})$  for a normal filter.
- $20 \log_{10}(\sqrt{\text{Upsampling factor}(N)} \times \text{Linear amplitude filter gain})$  for a square root filter.

## Input and Output Signals

The input signal must be a scalar or a frame-based column vector. double, single, and fixed-point data types are supported. Set the **Input sampling mode** parameter according to whether the input is sample-based or frame-based.

The input sampling mode and N, the value of the **Upsampling factor** parameter, determine characteristics of the output signal:

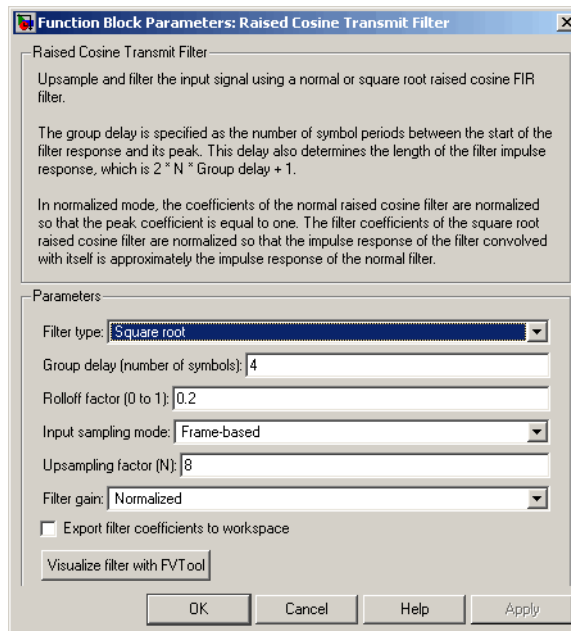
- If the input is a sample-based scalar, then the output is a sample-based scalar and the output sample time is N times the input sample time.
- If the input is frame-based, then the output is a frame-based vector whose length is N times the length of the input vector. The output frame period equals the input frame period.



## 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.

## 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.

# Raised Cosine Transmit Filter

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**Rolloff factor**

The rolloff factor for the filter, a real number between 0 and 1.

**Input sampling mode**

The type of input signal: Frame-based or Sample-based.

**Upsampling factor**

An integer greater than 1 representing the number of samples per symbol in the filtered output signal.

**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**

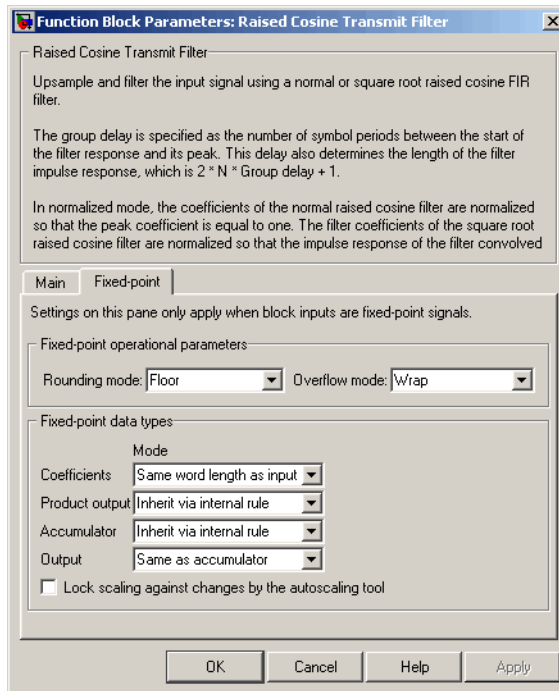
If you check this box, then the block creates a variable in the MATLAB workspace that contains the filter coefficients.

**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.



## Rounding mode

Select the rounding mode for fixed-point operations. The filter coefficients do not obey this parameter; they always round to Nearest.

## 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 *Signal Processing Blockset Reference Guide* for illustrations depicting the use of the coefficient data types in this block:

# Raised Cosine Transmit Filter

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- 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 *Signal Processing Blockset 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.

## 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.

# Raised Cosine Transmit Filter

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- 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.

**Pair Block**      Raised Cosine Receive Filter

**See Also**      Gaussian Filter, rcosine, rcosflt

**Purpose** Restore ordering of input symbols using random permutation

**Library** Block sublibrary of Interleaving

## Description

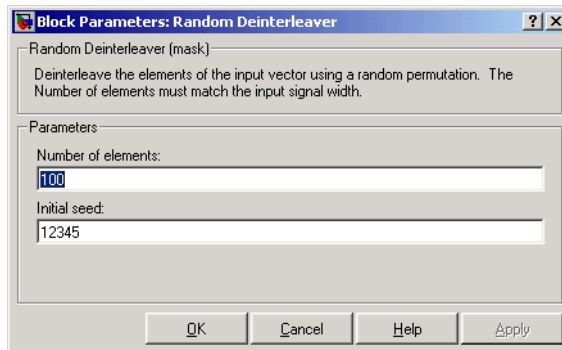


The Random Deinterleaver block rearranges the elements of its input vector using a random permutation. The **Initial seed** parameter initializes the random number generator that the block uses to determine the permutation. If this block and the Random Interleaver block have the same value for **Initial seed**, then the two blocks are inverses of each other.

The **Number of elements** parameter indicates how many numbers are in the input vector. If the input is frame-based, then it must be a column vector.

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.

## Dialog Box



### Number of elements

The number of elements in the input vector.

### Initial seed

The initial seed value for the random number generator.

**Pair Block** Random Interleaver

# Random Deinterleaver

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## **See Also**

General Block Deinterleaver



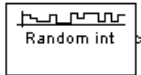
## Purpose

Generate integers randomly distributed in range  $[0, M-1]$

## Library

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 **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

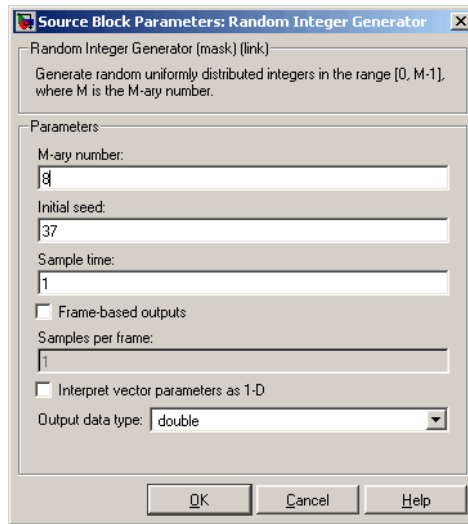
The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the **Frame-based outputs**, **Samples per frame**, and **Interpret vector parameters as 1-D** parameters. See “Signal Attribute Parameters for Random Sources” in *Communications Blockset 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.

# Random Integer Generator

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## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

### **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.

## **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.

## **See Also**

randint (Communications Toolbox)

# Random Interleaver

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**Purpose** Reorder input symbols using random permutation

**Library** Block sublibrary of Interleaving

## Description

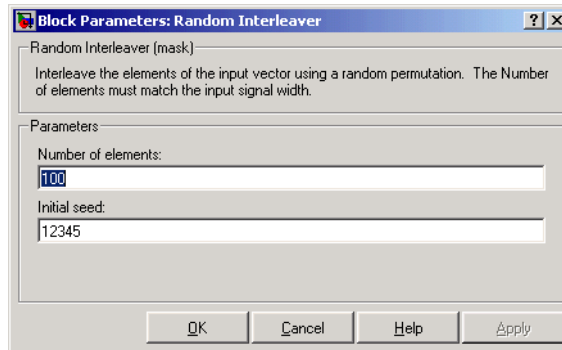


The Random Interleaver block rearranges the elements of its input vector using a random permutation. The **Number of elements** parameter indicates how many numbers are in the input vector. If the input is frame-based, then it must be a column vector.

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.

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.

## Dialog Box



### Number of elements

The number of elements in the input vector.

### Initial seed

The initial seed value for the random number generator.

**Pair Block** Random Deinterleaver

**See Also**      [General Block Interleaver](#)

# Rayleigh Noise Generator

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**Purpose** Generate Rayleigh distributed noise

**Library** Noise Generators sublibrary of Comm Sources

**Description** 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 Blockset `randseed` function. At the MATLAB prompt, enter

```
randseed
```

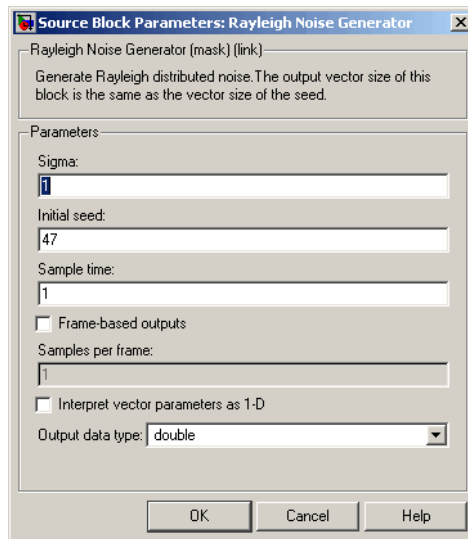
This returns a random prime number greater than 30. Entering `randseed` again produces a different prime number. If you supply an integer argument, `randseed` always returns the same prime for that integer. For example, `randseed(5)` always returns the same answer.

## 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 “Signal Attribute Parameters for Random Sources” in *Communications Blockset 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.

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

# Rayleigh Noise Generator

---

## **Sigma**

Specify  $\sigma$  as defined in the Rayleigh 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 Rayleigh Fading Channel; ray1rnd (Statistics Toolbox)

## **References**

[1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.



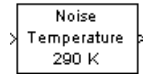
## Purpose

Apply receiver thermal noise to complex baseband signal

## Library

RF Impairments

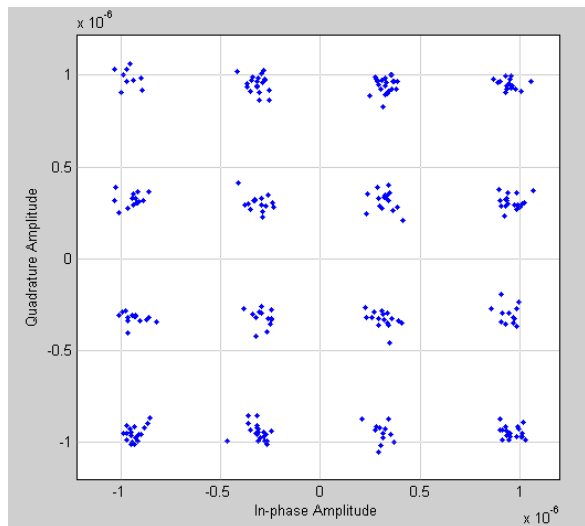
## Description



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  $1 + (\text{Noise temperature} / 290)$ .
- `Noise figure` specifies the noise as  $10 \cdot \log_{10}(1 + (\text{Noise temperature} / 290))$ . 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.

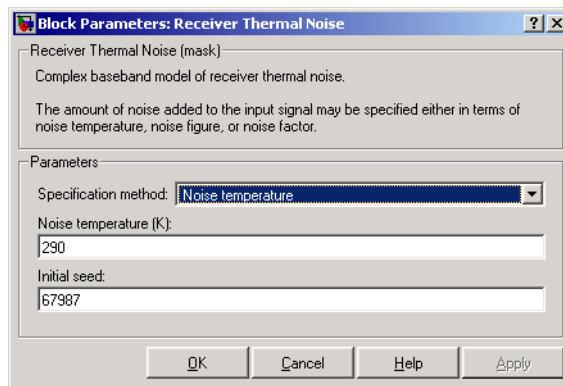


# Receiver Thermal Noise

This plot is generated by the model described in “Scatter Plot Examples” with the following parameter settings:

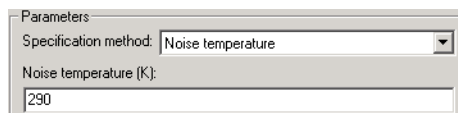
- Rectangular QAM Modulator Baseband
  - **Normalization method** set to Average Power
  - **Average power (watts)** set to  $1e-12$
- Receiver Thermal Noise
  - **Specification method** set to Noise figure
  - **Noise figure (dB)** set to 3.01

## Dialog Box



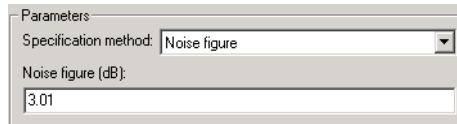
## Specification method

The method by which you specify the amount of noise. The choices are Noise temperature, Noise figure, and Noise factor.



## Noise temperature (K)

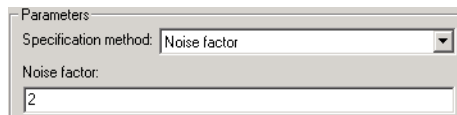
Scalar specifying the amount of noise in degrees kelvin.



A screenshot of a software parameter dialog box. The title bar reads "Parameters". Inside the dialog, there is a label "Specification method:" followed by a dropdown menu showing "Noise figure". Below that is a label "Noise figure (dB):" followed by a text input field containing the value "3.01".

## 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.



A screenshot of a software parameter dialog box. The title bar reads "Parameters". Inside the dialog, there is a label "Specification method:" followed by a dropdown menu showing "Noise factor". Below that is a label "Noise factor:" followed by a text input field containing the value "2".

## 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

Free Space Path Loss

# Rectangular QAM Demodulator Baseband

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**Purpose** Demodulate rectangular-QAM-modulated data

**Library** AM, in Digital Baseband sublibrary of Modulation

## Description



The Rectangular QAM Demodulator Baseband block demodulates a signal that was modulated using quadrature amplitude modulation with a constellation on a rectangular lattice.

The signal constellation has  $M$  points, where  $M$  is the **M-ary number** parameter.  $M$  must have the form  $2^K$  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.

The input can be either a scalar or a frame-based column vector of data types `single` or `double` in all modes, as well as signed fixed-point (binary-point only scaling) in hard-decision modes.

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**Note** This block assumes a nominal impedance of 1 ohm.

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## Hard Decision Algorithm

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  $\sqrt{M} - 1$  to translate them into an approximate range between 0 and  $2 \times (\sqrt{M} - 1)$  (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 and

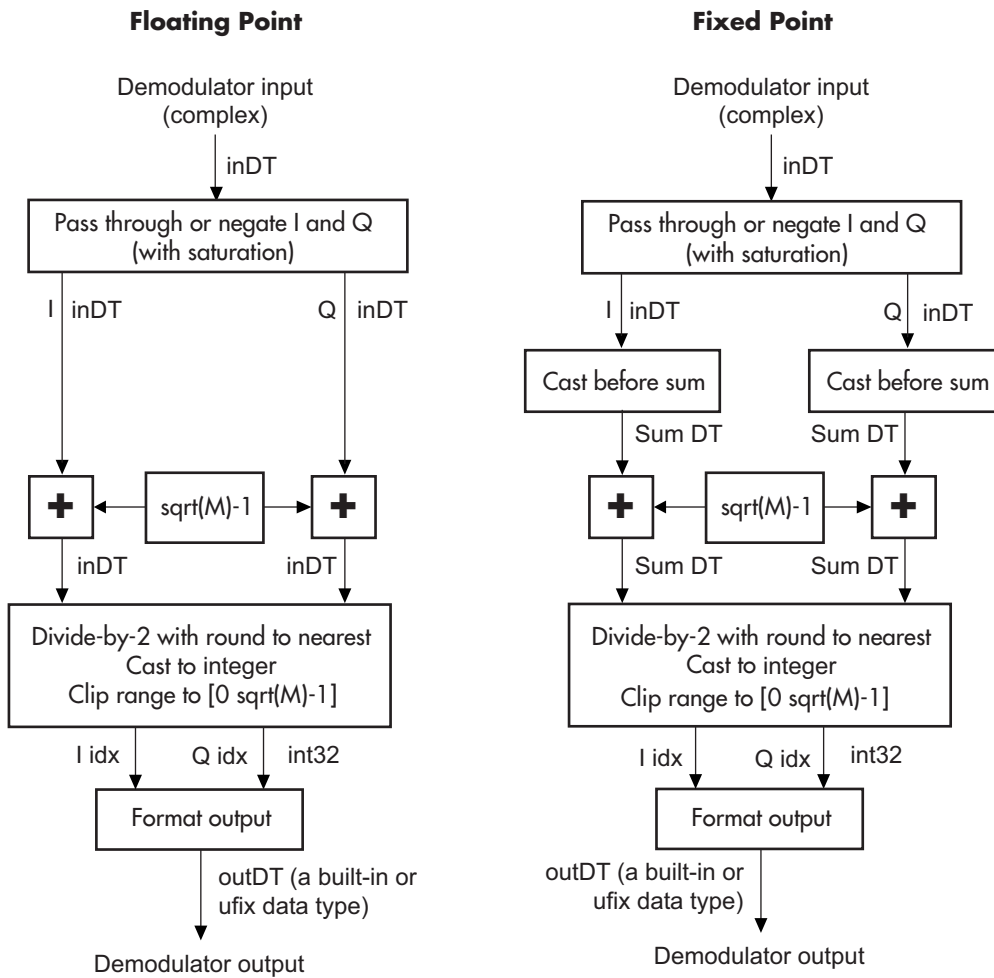
# Rectangular QAM Demodulator Baseband

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$\sqrt{M} - 1$  (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 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**.

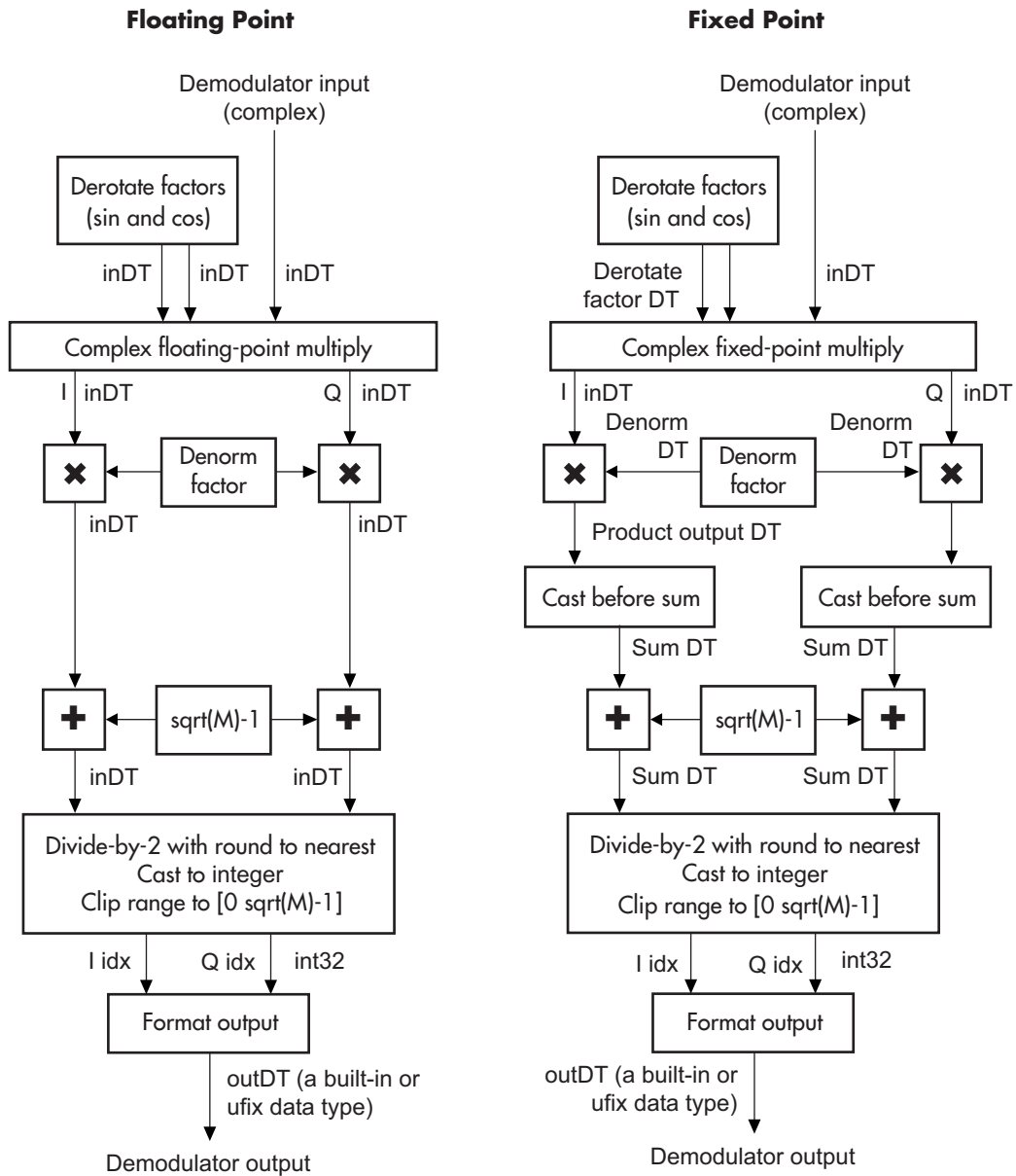
The following figures contains signal flow diagrams for floating-point and fixed-point algorithm operation. The floating-point diagrams apply when the input signal data type is double or single. The fixed-point diagrams apply when the input signal is a signed fixed-point data type. Note that the diagram is simplified when **Phase offset** is a multiple of  $\pi/2$ , or the derived denormalization factor is 1.

# Rectangular QAM Demodulator Baseband



**Signal-Flow Diagrams with Trivial Phase Offset and Denormalization Factor Equal to 1**

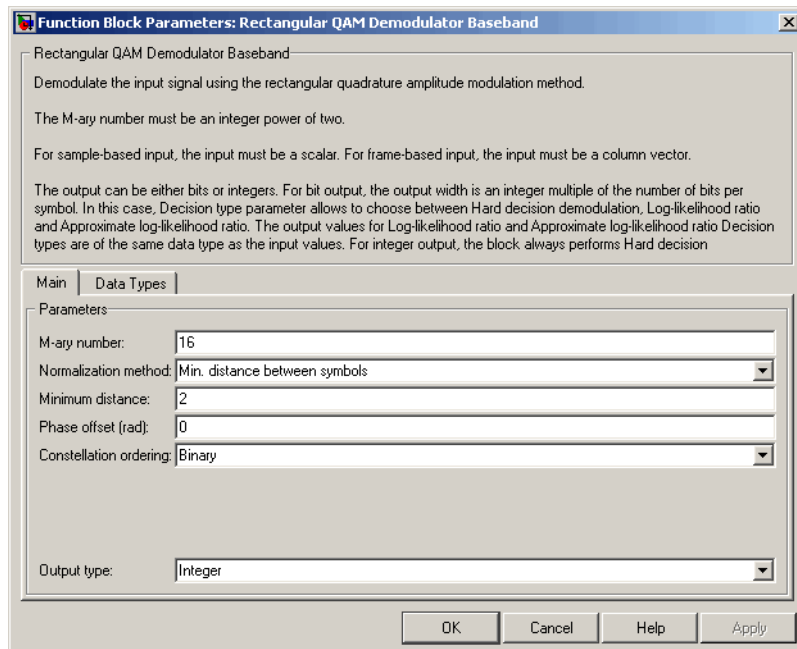
# Rectangular QAM Demodulator Baseband



**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  $2^K$  for some positive integer  $K$ .

### Normalization method

Determines how the block scales the signal constellation. Choices are Min. distance between symbols, Average Power, and Peak Power.

### Minimum distance

This field appears when **Normalization method** is set to Min. distance between symbols.

The distance between two nearest constellation points.



# Rectangular QAM Demodulator Baseband

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## **Average power (watts)**

This field appears when **Normalization method** is set to Average Power.

The average power of the symbols in the constellation.

## **Peak power (watts)**

This field appears when **Normalization method** is set to Peak Power.

The maximum power among the symbols in the constellation.

## **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 field 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  $[0, M-1]$ . 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.

## **Output type**

Determines whether the block produces integers or binary representations of integers.

# Rectangular QAM Demodulator Baseband

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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 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” for algorithm details.

## **Noise variance source**

This field appears when Approximate log-likelihood ratio or Log-likelihood ratio is selected for **Decision type**.

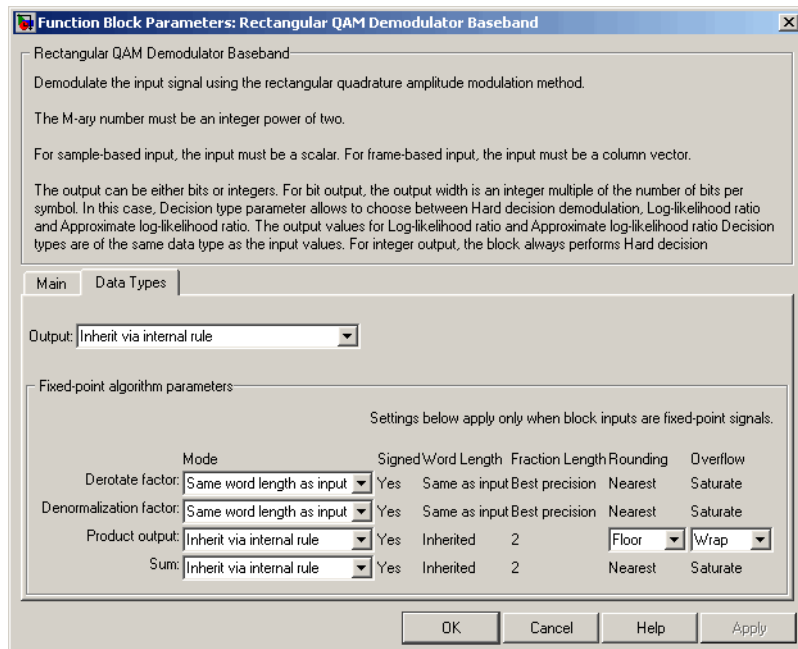
When set to Dialog, the noise variance can be specified in the **Noise variance** field. When set to Port, a port appears on the block through which the noise variance can be input.

## **Noise variance**

This field appears when **Noise variance source** is set to Dialog.

If this value is very small (i.e., SNR is very high), LLR computations may yield Inf or -Inf because the LLR algorithm would involve computing exponentials of very large or very small numbers using finite precision arithmetic. In such cases, use of approximate LLR is recommended, 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'.

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 block. 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  $ufix(\lceil \log_2(M) \rceil)$  for integer outputs. For all other

# Rectangular QAM Demodulator Baseband

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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  $\pi/2$ .

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.

## **Product output**

This parameter only applies when the input is fixed-point and there is a nonunity (not equal to 1) denormalization factor.

This can be set to Inherit via internal rule or Specify word length, in which case a field is enabled for user input.

Setting to Inherit via internal rule computes the product word length as follows. The ideal full-precision product word and fraction lengths are calculated as  $idealProdWL = inputWL + denormFactorWL$  and  $idealProdFL = inputFL + denormFactorFL$ .

# Rectangular QAM Demodulator Baseband

After this computation, the product output word length is set to an *adjustedProdWL*, as shown in the following table. The product fraction length is set to 2, because additional fraction bits do not affect the fixed-point algorithm precision.

If <i>idealProdWL</i> is...	Set <i>adjustedProdWL</i> to...
Greater than or equal to 2	$idealProdWL - (idealProdFL - 2)$
Less than 2	Throw an error, because the input, denormalization factor, or both is not precise enough.

Finally, the adjusted product word length is set to be the smallest available word length (as specified in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the block) large enough to contain it.

## Sum

This parameter only applies when the input is fixed-point.

This 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 sets the sum word length to the smallest available word length (as specified in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the block) that fits the sum word length given by  $numBitsPerSymbol + 4$  (or, equivalently,  $\text{ceil}(\log_2(M)) + 4$ ). The sum fraction length is set to 2.

**Pair Block**      Rectangular QAM Modulator Baseband

**See Also**        General QAM Demodulator Baseband

# Rectangular QAM Demodulator Baseband

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## References

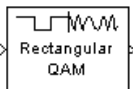
- [1] Smith, Joel G., "Odd-Bit Quadrature Amplitude-Shift Keying," *IEEE Transactions on Communications*, Vol. COM-23, March 1975, 385–389.

# Rectangular QAM Modulator Baseband

**Purpose** Modulate using rectangular quadrature amplitude modulation

**Library** AM, in Digital Baseband sublibrary of Modulation

**Description** 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.



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**Note** This block assumes a nominal impedance of 1 ohm.

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## Constellation Size and Scaling

The signal constellation has  $M$  points, where  $M$  is the **M-ary number** parameter.  $M$  must have the form  $2^K$  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.

Value of Normalization Method Parameter	Scaling Condition
Min. distance between symbols	The nearest pair of points in the constellation is separated by the value of the <b>Minimum distance</b> parameter
Average Power	The average power of the symbols in the constellation is the <b>Average power</b> parameter
Peak Power	The maximum power of the symbols in the constellation is the <b>Peak power</b> parameter

# Rectangular QAM Modulator Baseband

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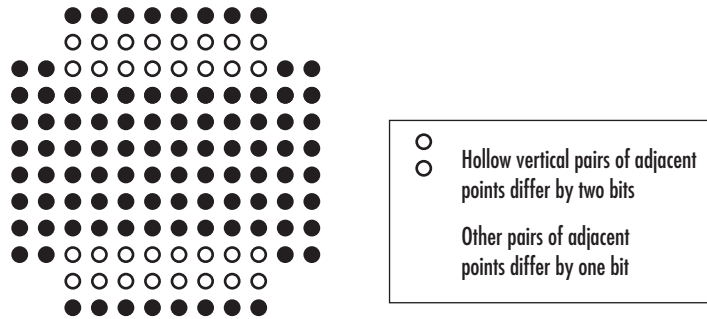
## Input Signal Values

The input and output for this block are discrete-time signals. The **Input type** parameter determines whether the block accepts integers between 0 and  $M-1$ , or binary representations of integers:

- If **Input type** is set to Integer, the block accepts integers. The input can be either a scalar or a frame-based column vector, and the block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`.
- If **Input type** is set to Bit, the block accepts groups of  $K$  bits, called *binary words*. The input can be either a vector of length  $K$  or a frame-based column vector whose length is an integer multiple of  $K$ . For bit inputs, the block can accept `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, and `double`. 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:
  - 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.



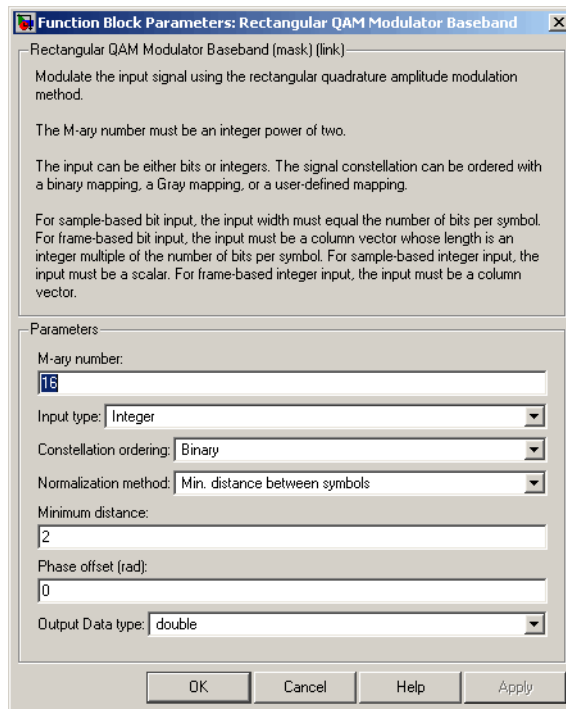
# Rectangular QAM Modulator Baseband



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  $M = 4$ .

# Rectangular QAM Modulator Baseband

## Dialog Box



### M-ary number

The number of points in the signal constellation. It must have the form  $2^K$  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.

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 right to left. 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 (watts)**

The average power of the symbols in the constellation. This field appears only when **Normalization method** is set to Average Power.

## **Peak power (watts)**

The maximum power of the symbols in the constellation. This field appears only when **Normalization method** is set to Peak Power.

## **Phase offset (rad)**

The rotation of the signal constellation, in radians.

## **Output data type**

The output data type can be set to double, single, Fixed-point, User-defined, or Inherit via back propagation.

# Rectangular QAM Modulator Baseband

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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 Simulink Fixed Point. 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 the **Output data type** parameter and `User-defined` for the **Set output fraction length to** parameter.

## **Pair Block**

Rectangular QAM Demodulator Baseband

# Rectangular QAM Modulator Baseband

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## **See Also**

General QAM Modulator Baseband

## **References**

[1] Smith, Joel G., "Odd-Bit Quadrature Amplitude-Shift Keying," *IEEE Transactions on Communications*, Vol. COM-23, March 1975, 385–389.

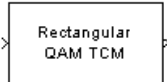
# Rectangular QAM TCM Decoder

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**Purpose** Decode trellis-coded modulation data, modulated using QAM method

**Library** Trellis-Coded Modulation

**Description** 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 is the number of points in the signal constellation, which also equals the number of possible output symbols from the convolutional encoder. (That is,  $\log_2(\mathbf{M}\text{-ary number})$  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

The input signal must be a frame-based column vector containing complex numbers.

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 frame-based 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 each frame, the block saves its internal state metric for use with the next frame.

If you select the **Enable the reset input** check box, 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.

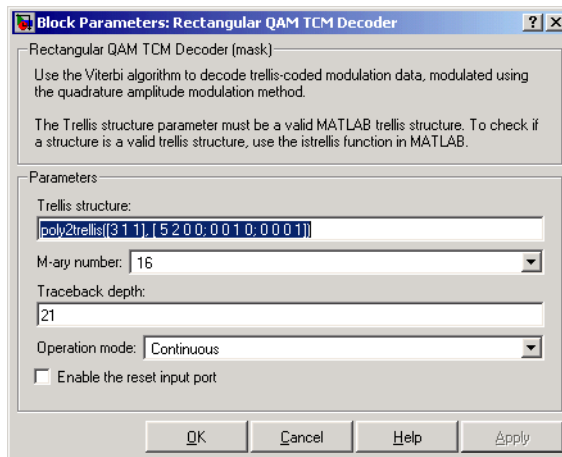
- In Truncated mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In Terminated mode, the block treats each frame independently. The traceback path always starts at the all-zeros 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**.

## Dialog Box



# Rectangular QAM TCM Decoder

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## **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.

## **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.

## **Pair Block**

Rectangular QAM 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.

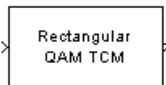
[2] Proakis, John G., *Digital Communications*, Fourth edition, New York, McGraw-Hill, 2001.



**Purpose** Convolutionally encode binary data and modulate using QAM method

**Library** Trellis-Coded Modulation

## Description



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,  $\log_2(\mathbf{M}\text{-ary number})$  is equal to  $n$  for a rate  $k/n$  convolutional code.)

### Input 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 frame-based binary column vector whose length is  $L \cdot k$  for some positive integer  $L$ .

The output from the Rectangular QAM TCM Encoder block is a frame-based 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 Encoder” in the Communications 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

```
poly2trellis(7,[171 133],171)
```

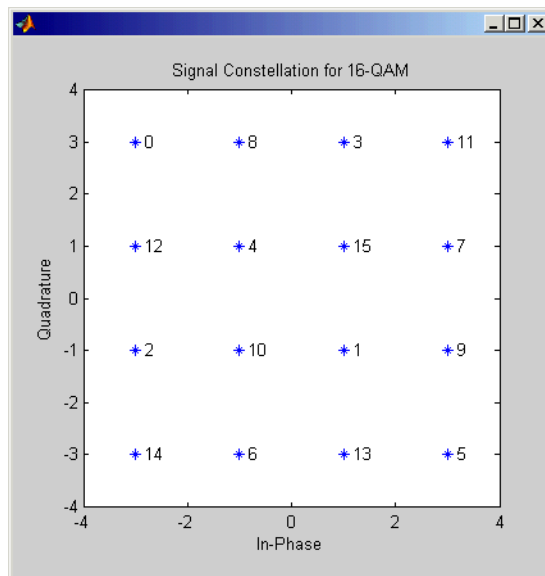
# Rectangular QAM TCM Encoder

- 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.

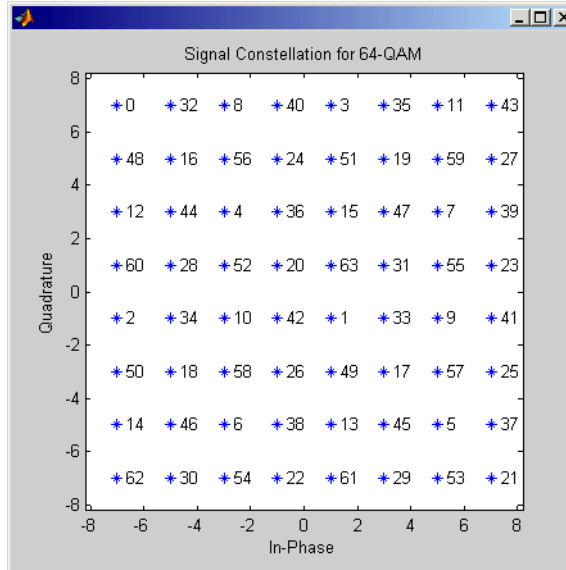
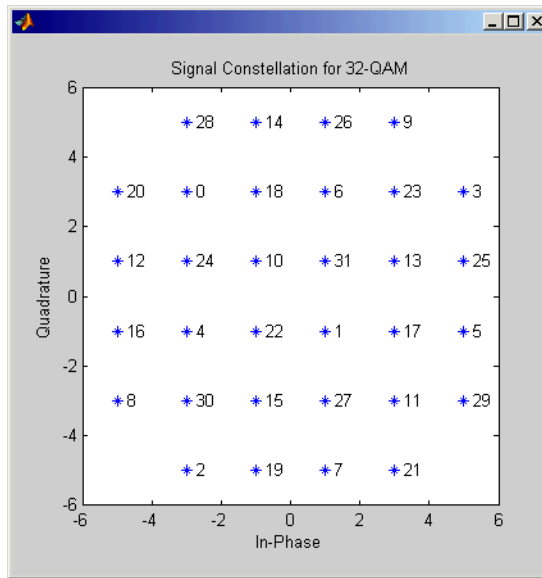
## Signal Constellations

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 **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 [1].



# Rectangular QAM TCM Encoder

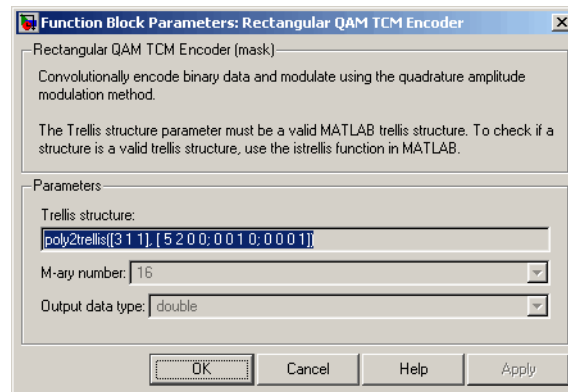


# Rectangular QAM TCM Encoder

## 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].

## 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.

## 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.

# Rician Noise Generator

## Purpose

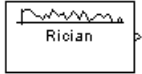
Generate Rician distributed noise

## Library

Noise Generators sublibrary of Comm Sources

## Description

The Rician Noise Generator block generates Rician distributed noise. The Rician probability density function is given by



$$f(x) = \begin{cases} \frac{x}{\sigma^2} I_0\left(\frac{mx}{\sigma^2}\right) \exp\left(-\frac{x^2 + m^2}{2\sigma^2}\right) & x \geq 0 \\ 0 & x < 0 \end{cases}$$

where:

- $\sigma$  is the standard deviation of the Gaussian distribution that underlies the Rician distribution noise
- $m^2 = m_I^2 + m_Q^2$ , where  $m_I$  and  $m_Q$  are the mean values of two independent Gaussian components
- $I_0$  is the modified 0th-order Bessel function of the first kind given by

$$I_0(y) = \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{y \cos t} dt$$

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.

## 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 Blockset `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.

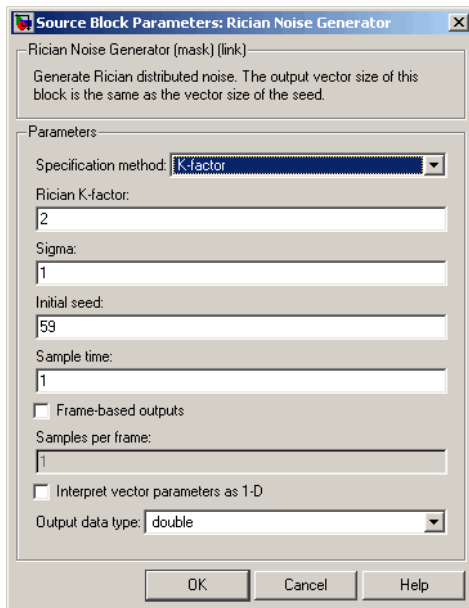
## Attributes of Output Signal

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 “Signal Attribute Parameters for Random Sources” in *Communications Blockset 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

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

### Specification method

Either K-factor or Quadrature components.

### Rician K-factor

$K = m^2/(2\sigma^2)$ , where  $m$  is as in the Rician probability density function. This field appears only if **Specification method** is K-factor.

### In-phase component (mean), Quadrature component (mean)

The mean values  $m_I$  and  $m_Q$ , respectively, of the Gaussian components. These fields appear only if **Specification method** is Quadrature components.

### 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

**References**

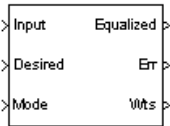
[1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.

# RLS Decision Feedback Equalizer

**Purpose** Equalize using decision feedback equalizer that updates weights with RLS algorithm

**Library** Equalizers

**Description** 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. 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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 “Retrieving the Weights and Error Signal” in *Communications Blockset 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 Blockset User’s Guide*.

## Equalizer Delay

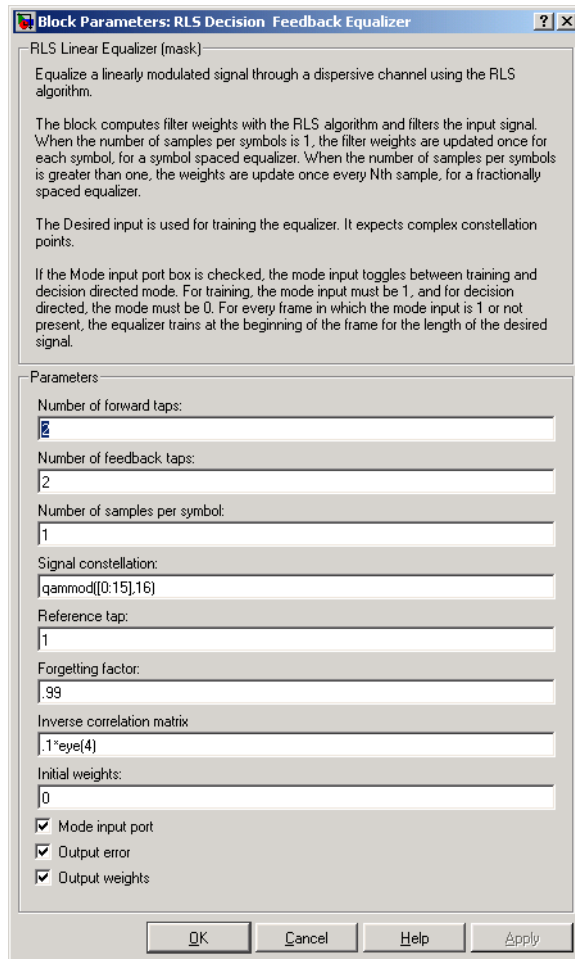
For proper equalization, you should set the **Reference tap** parameter so that it exceeds the delay, in symbols, between the transmitter’s modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer *output* is equal to

$$1 + (\text{Reference tap} - 1) / (\text{Number of samples per symbol})$$

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.

**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**

If you check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

**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.

**Output weights**

If you check this box, the block outputs the current forward and feedback weights, concatenated into one vector.

# RLS Decision Feedback Equalizer

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## 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.

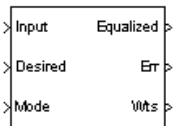
## See Also

RLS Linear Equalizer, LMS Decision Feedback Equalizer, CMA Equalizer

**Purpose** Equalize using linear equalizer that updates weights using RLS algorithm

**Library** Equalizers

## Description



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. 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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 “Retrieving the Weights and Error Signal” in *Communications Blockset User’s Guide*.

# RLS Linear Equalizer

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## 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 Blockset User’s Guide*.

## Equalizer Delay

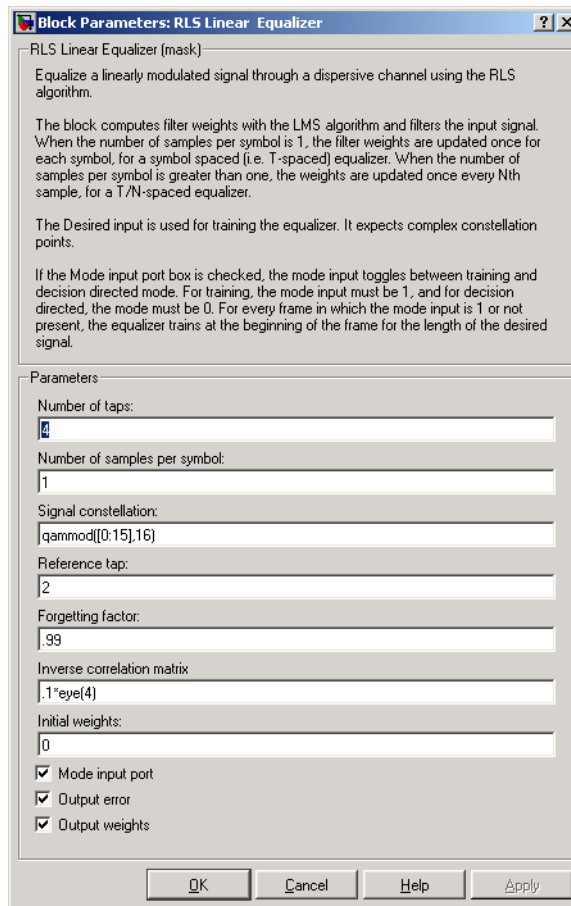
For proper equalization, you should set the **Reference tap** parameter so that it exceeds the delay, in symbols, between the transmitter’s modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer *output* is equal to

$$1 + (\text{Reference tap} - 1) / (\text{Number of samples per symbol})$$

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.



## 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.

# RLS Linear 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 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**

If you check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

**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.

**Output weights**

If you check this box, the block outputs the current weights.

**Examples**

See the Adaptive Equalization demo.

**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**

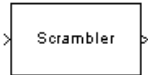
RLS Decision Feedback Equalizer, LMS Linear Equalizer, CMA Equalizer

# Scrambler

**Purpose** Scramble input signal

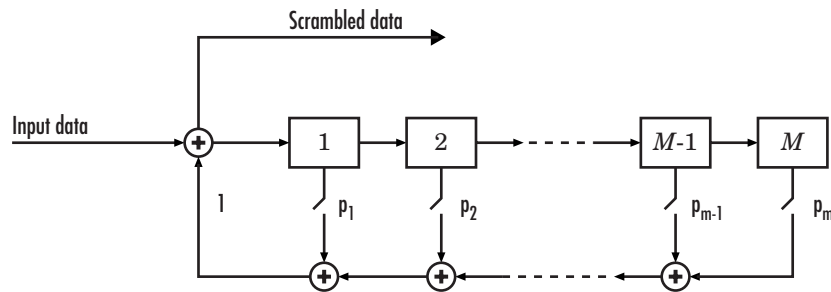
**Library** Sequence Operations

## Description



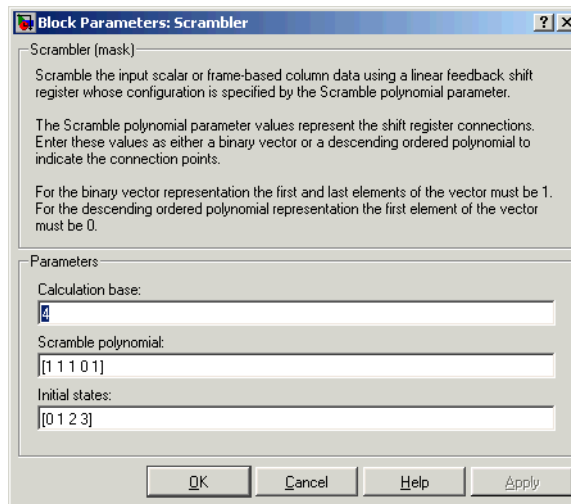
The Scrambler block scrambles the input signal, which must be a scalar or a frame-based column vector. If the **Calculation base** parameter is  $N$ , then the input values must be integers between 0 and  $N-1$ .

One purpose of scrambling is to reduce the length of strings of 0s or 1s in a transmitted signal, since a long string of 0s or 1s 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. Each switch in the scrambler is on or off as defined by the **Scramble polynomial** parameter. You can specify the polynomial by listing its coefficients in order of ascending powers of  $z^{-1}$ , where  $p(z^{-1}) = 1 + p_1z^{-1} + p_2z^{-2} + \dots$ , or by listing the powers of  $z$  that appear in the polynomial with a coefficient of 1. For example  $p = [1 \ 0 \ 0 \ 0 \ 0 \ 1 \ 0 \ 1]$  and  $p = [0 \ -6 \ -8]$  both represent the polynomial  $p(z^{-1}) = 1 + z^{-6} + z^{-8}$ .

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.)



## Dialog Box

### Calculation base

The calculation base  $N$ . The input and output of this block are integers in the range  $[0, N-1]$ .

### 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

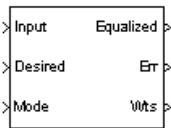
# Sign LMS Decision Feedback Equalizer

---

**Purpose** Equalize using decision feedback equalizer that updates weights with signed LMS algorithm

**Library** Equalizers

## Description



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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 “Retrieving the Weights and Error Signal” in *Communications Blockset 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 Blockset User’s Guide*.

## Equalizer Delay

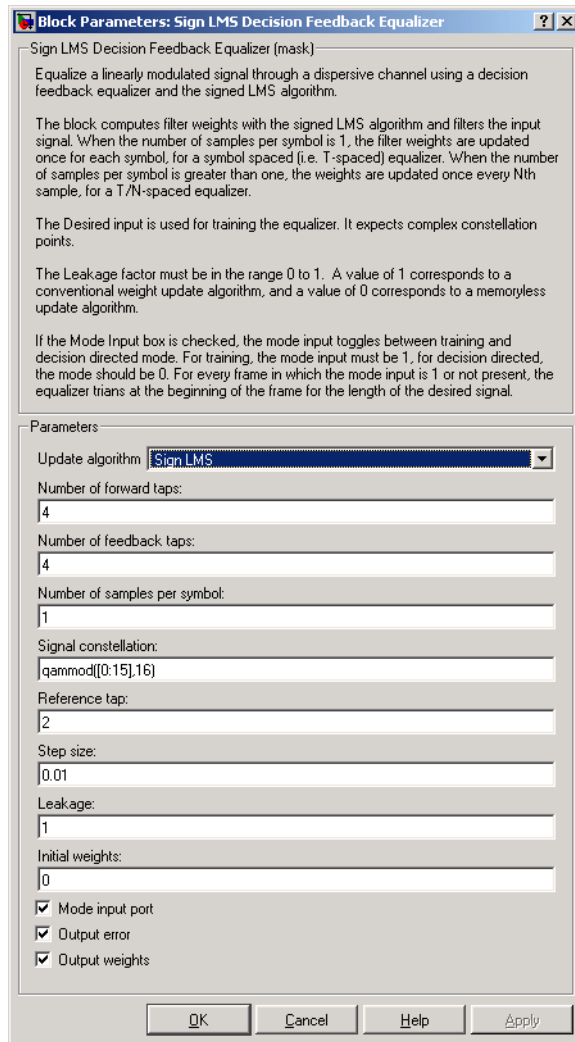
For proper equalization, you should set the **Reference tap** parameter so that it exceeds the delay, in symbols, between the transmitter’s modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer *output* is equal to

$$1 + (\text{Reference tap} - 1) / (\text{Number of samples per symbol})$$

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

## 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.

**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.

**Mode input port**

If you check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

**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.

# Sign LMS Decision Feedback Equalizer

---

## Output weights

If you check this 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.

[2] Kurzweil, Jack, *An Introduction to Digital Communications*, New York, Wiley, 2000.

## See Also

Sign LMS Linear Equalizer, LMS Decision Feedback Equalizer

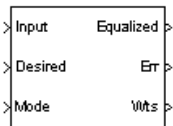
## Purpose

Equalize using linear equalizer that updates weights with signed LMS algorithm

## Library

Equalizers

## Description



The Sign LMS Linear Equalizer block uses a linear 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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 Linear Equalizer

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- Mode input, as described in “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 “Retrieving the Weights and Error Signal” in *Communications Blockset 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 Blockset User’s Guide*.

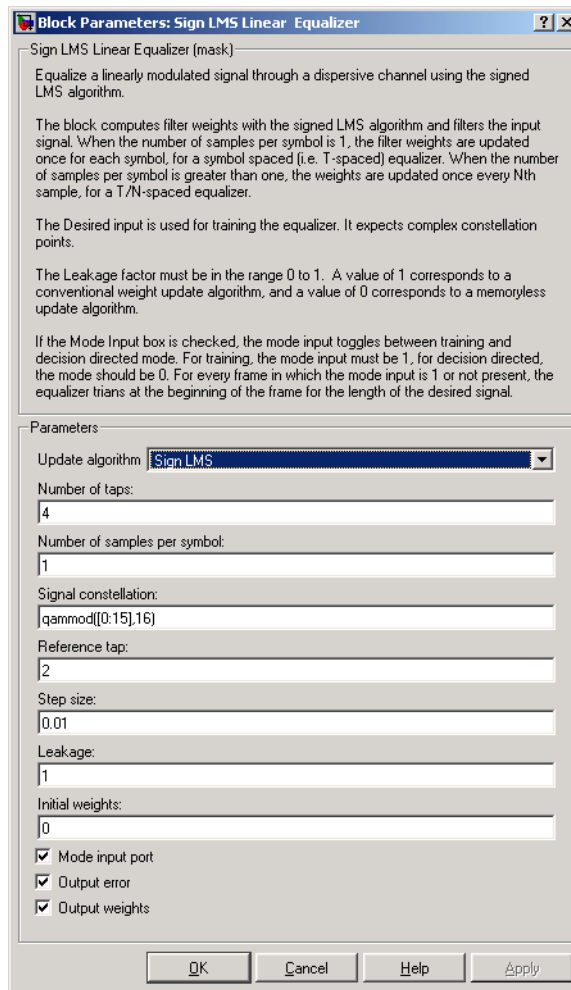
## Equalizer Delay

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

$$1+(\text{Reference tap}-1)/(\text{Number of samples per symbol})$$

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## 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**

If you check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

**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.

**Output weights**

If you check this box, the block outputs the current weights.

**Examples**

See the Adaptive Equalization demo.

## 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

Sign LMS Decision Feedback Equalizer, LMS Linear Equalizer

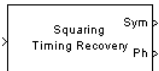
# Squaring Timing Recovery

---

**Purpose** Recover symbol timing phase using squaring method

**Library** Timing Phase Recovery sublibrary of Synchronization

## Description



The Squaring Timing Recovery block recovers the symbol timing phase of the input signal using a squaring method. This frame-based, 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. The input to this block must be a frame-based column vector 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.

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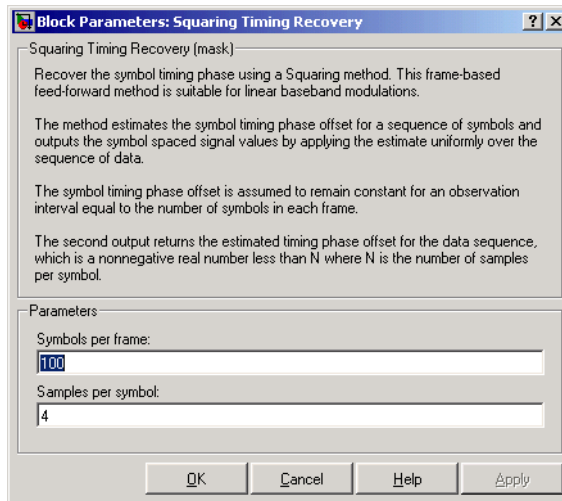
The block estimates the phase offset for the symbols in each input frame and applies the estimate uniformly over the input frame. The block outputs frame-based signals, each containing one sample per symbol. The frame size of each output therefore equals the **Symbols per frame** parameter value. The 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.



- The output port labeled Ph gives the phase estimate for each symbol in the input frame. All elements in this output frame 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.

## Dialog Box



### 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.

## Algorithm

This block uses a timing estimator that returns

$$-\frac{1}{2\pi} \arg \left( \sum_{m=0}^{LN-1} |x_{m+1}|^2 \exp(-j2\pi m/N) \right)$$

# Squaring Timing Recovery

---

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 Blockset User's Guide*.

## Examples

See "Squaring Timing Phase Recovery Example" in *Communications Blockset User's Guide*.

## References

[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.

## See Also

Gardner Timing Recovery, Early-Late Gate Timing Recovery

# SSB AM Demodulator Passband

**Purpose** Demodulate SSB-AM-modulated data

**Library** Analog Passband Modulation, in Modulation

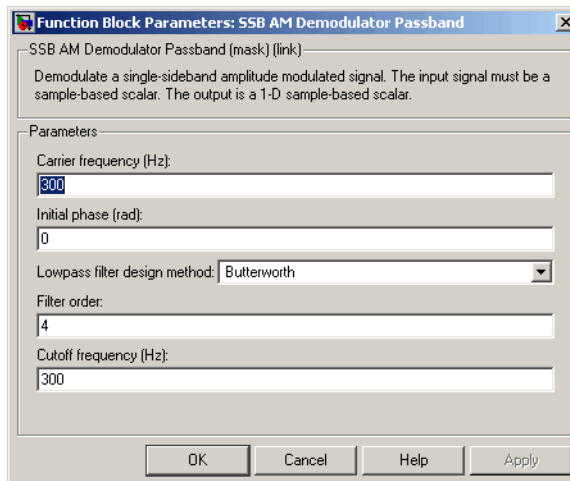
## Description



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 sample-based scalar signals.

This block works only with real inputs of type `double`. This block is not suited to be placed inside a triggered subsystem.

## Dialog Box



### 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.

# SSB AM Demodulator Passband

---

## **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**

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**

SSB AM Modulator Passband

## **See Also**

DSB AM Demodulator Passband, DSBSC AM Demodulator Passband

## Purpose

Modulate using single-sideband amplitude modulation

## Library

Analog Passband Modulation, in Modulation

## Description



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 sample-based 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

$$u(t)\cos(f_c t + \theta) \mp u(t)\sin(f_c t + \theta)$$

where:

- $f_c$  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 Transform Filter

This block uses the Analytic Signal block from the Signal Processing Blockset Transforms library.

The Analytic Signal block computes the complex analytic signal corresponding to each channel of the real M-by-N input,  $u$

$$y = u + jH\{u\}$$

where  $j = \sqrt{-1}$  and  $H\{\}$  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

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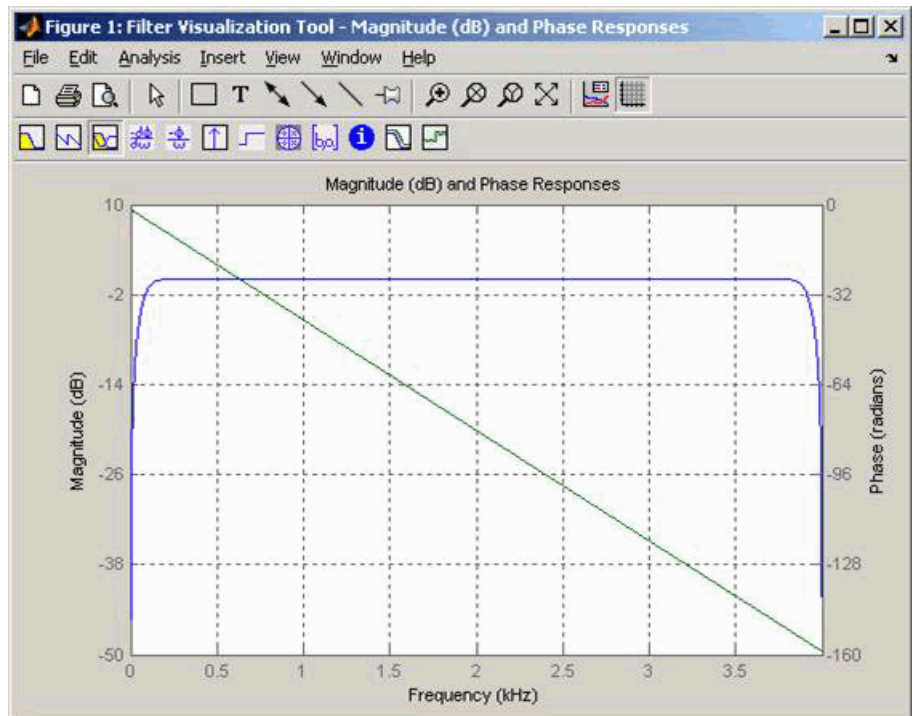
channel; the imaginary part is the Hilbert transform of the input. In the frequency domain, the analytic signal retains the positive frequency content of the original signal while zeroing-out negative frequencies and doubling the DC component.

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  $n/2$  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 time. This is due to the implementation of the Hilbert transform by means of a filter.

In the following example, we sample a 10Hz 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 400Hz, 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.

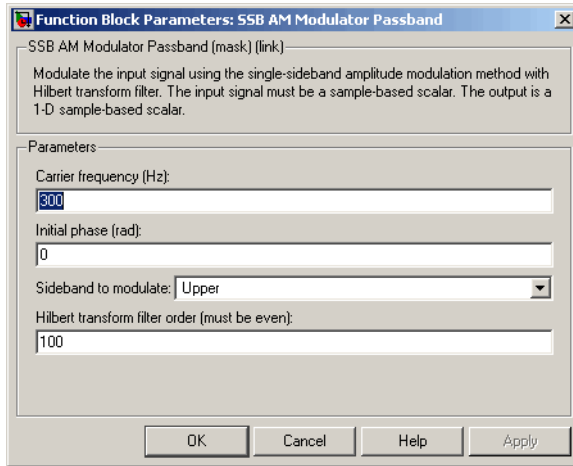
Typically, an appropriate **Carrier frequency** value is much higher than the highest frequency of the input signal. By the Nyquist sampling theorem, the reciprocal of the model's sample time (defined by the model's signal source) must exceed twice the **Carrier frequency** parameter.

# SSB AM Modulator Passband

---

This block works only with real inputs of type double. This block is not suited to be placed 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

SSB AM Demodulator Passband

## See Also

DSB AM Modulator Passband, DSBSC AM Modulator Passband; hilbiir (Communications Toolbox)



## References

- [1] Peebles, Peyton Z, Jr. *Communication System Principles*. Reading, Mass.: Addison-Wesley, 1976.

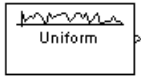
# Uniform Noise Generator

---

**Purpose** Generate uniformly distributed noise between upper and lower bounds

**Library** Noise Generators sublibrary of Comm Sources

## Description



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.

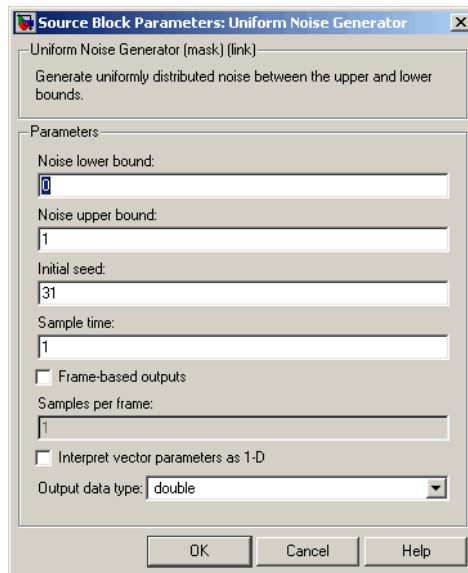
If all the elements of the output vector are to be independent and identically distributed (i.i.d.), then you can use a scalar for the **Noise lower bound** and **Noise upper bound** parameters. Alternatively, you can specify the range for each element of the output vector individually, by using vectors 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. See “Signal Attribute Parameters for Random Sources” in *Communications Blockset 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.

## Dialog Box



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

### 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.

### 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.

# Uniform Noise 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 can be set to double or single data types.

## **See Also**

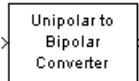
Random Source (Signal Processing Blockset); rand (built-in MATLAB function)

# Unipolar to Bipolar Converter

**Purpose** Map unipolar signal in range  $[0, M-1]$  into bipolar signal

**Library** Utility Blocks

## Description

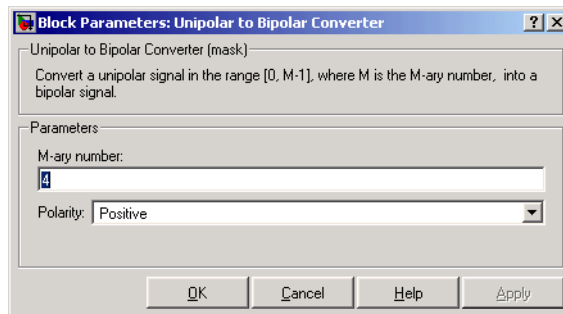


The Unipolar to Bipolar Converter block maps the unipolar input signal to a bipolar output signal. If the input consists of integers between 0 and  $M-1$ , where  $M$  is the **M-ary number** parameter, then the output consists of integers between  $-(M-1)$  and  $M-1$ . If  $M$  is even, then the output is odd, and vice-versa.

The table below shows how the block's mapping depends on the **Polarity** parameter.

Polarity Parameter Value	Output Corresponding to Input Value of $k$
Positive	$2k-(M-1)$
Negative	$-2k+(M-1)$

## Dialog Box



### M-ary number

The number of symbols in the bipolar or unipolar alphabet.

### Polarity

A value of Positive (respectively, Negative) causes the block to maintain (respectively, reverse) the relative ordering of symbols in the alphabets.

# Unipolar to Bipolar Converter

---

## Examples

If the input is [0; 1; 2; 3], the **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].

## Pair Block

Bipolar to Unipolar Converter

# Variable Step LMS Decision Feedback Equalizer

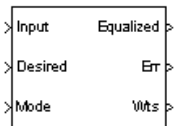
## Purpose

Equalize using decision feedback equalizer that updates weights with variable-step-size LMS algorithm

## Library

Equalizers

## Description



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. 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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 “Retrieving the Weights and Error Signal” in *Communications Blockset 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 Blockset 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

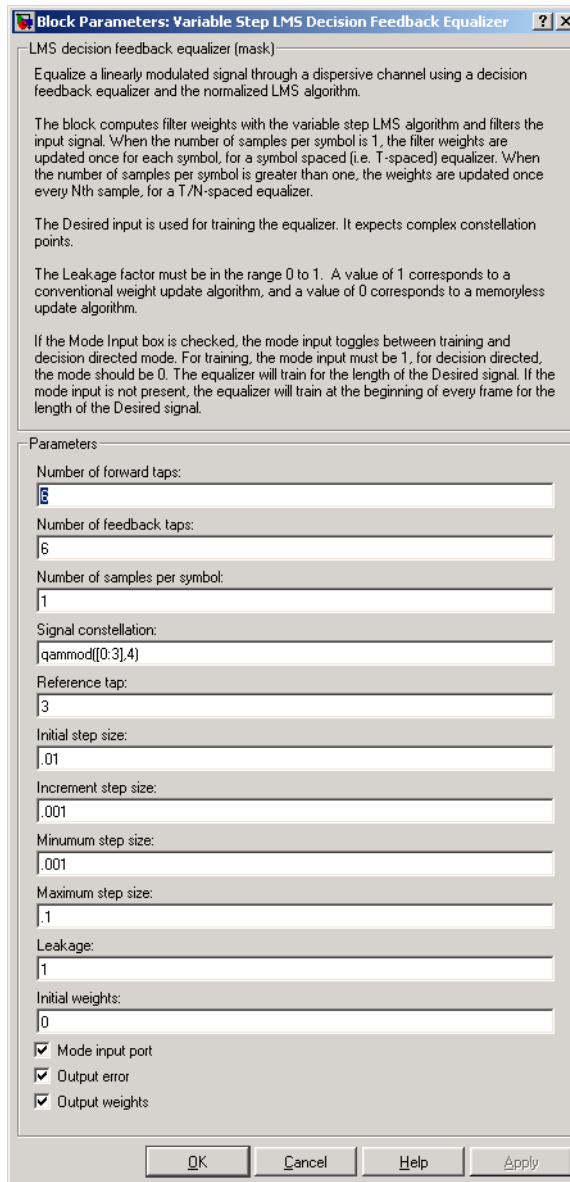
$$1 + (\text{Reference tap} - 1) / (\text{Number of samples per symbol})$$

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



Block Parameters: Variable Step LMS Decision Feedback Equalizer

LMS decision feedback equalizer (mask)

Equalize a linearly modulated signal through a dispersive channel using a decision feedback equalizer and the normalized LMS algorithm.

The block computes filter weights with the variable step LMS algorithm and filters the input signal. When the number of samples per symbol is 1, the filter weights are updated once for each symbol, for a symbol spaced (i.e. T-spaced) equalizer. When the number of samples per symbol is greater than one, the weights are updated once every Nth sample, for a T/N-spaced equalizer.

The Desired input is used for training the equalizer. It expects complex constellation points.

The Leakage factor must be in the range 0 to 1. A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

If the Mode Input box is checked, the mode input toggles 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.

Parameters

Number of forward taps:  
8

Number of feedback taps:  
6

Number of samples per symbol:  
1

Signal constellation:  
qammod([0:3],4)

Reference tap:  
3

Initial step size:  
.01

Increment step size:  
.001

Minimum step size:  
.001

Maximum step size:  
.1

Leakage:  
1

Initial weights:  
0

Mode input port

Output error

Output weights

OK Cancel Help Apply

# Variable Step 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.

**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.

# Variable Step LMS Decision Feedback Equalizer

---

## **Mode input port**

If you check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

## **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.

## **Output weights**

If you check this 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**

Variable Step LMS Linear Equalizer, LMS Decision Feedback Equalizer

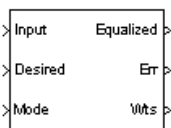
# Variable Step LMS Linear Equalizer

---

**Purpose** Equalize using linear equalizer that updates weights with variable-step-size LMS algorithm

**Library** Equalizers

**Description** 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. 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 port labeled **Input** receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled **Desired** receives a training sequence whose length is less than or equal to the number of symbols in the **Input** signal. Valid training symbols are those listed in the **Signal constellation** vector.

This block accepts only frame-based signals. If the value of **Reference tap** is equal to or greater than the frame size, the block will not work properly.

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 “Controlling the Use of Training or Decision-Directed Mode” in *Communications Blockset 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 “Retrieving the Weights and Error Signal” in *Communications Blockset 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 Blockset 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

$$1 + (\text{Reference tap} - 1) / (\text{Number of samples per symbol})$$

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

# Variable Step LMS Linear Equalizer

## Dialog Box

**Block Parameters: Variable Step LMS Linear Equalizer** [?] [X]

Normalized LMS Linear Equalizer (mask)

Equalize a linearly modulated signal through a dispersive channel using the normalized LMS algorithm.

The block computes filter weights with the LMS algorithm and filters the input signal. When the number of samples per symbol is 1, the filter weights are updated once for each symbol, for a symbol spaced (i.e. T-spaced) equalizer. When the number of samples per symbol is greater than one, the weights are updated once every Nth sample, for a T/N-spaced equalizer.

The Desired input is used for training the equalizer. It expects complex constellation points.

The Leakage factor must be in the range 0 to 1. A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

If the Mode Input box is checked, the mode input toggles 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.

Parameters

Number of taps:

Number of samples per symbol:

Signal constellation:

Reference tap:

Initial step size:

Increment step size:

Minimum step size:

Maximum step size:

Leakage:

Initial weights:

Mode input port

Output error

Output weights

OK Cancel Help Apply

# Variable Step 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.

**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**

If you check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

# Variable Step LMS Linear Equalizer

---

## **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.

## **Output weights**

If you check this box, the block outputs the current weights.

## **Examples**

See the Adaptive Equalization demo.

## **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



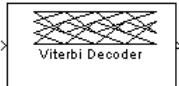
## Purpose

Decode convolutionally encoded data using Viterbi algorithm

## Library

Convolutional sublibrary of Channel Coding

## 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.

### 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  $L*k$ . The integer  $L$  is the number of frames that the block processes in each step.

The input can be either a sample-based vector with  $L = 1$ , or a frame-based column vector with any positive integer for  $L$ .

The block supports nondouble data typed input and output signals based on the **Decision type** selected from the mask. For unquantized decisions, the block accepts double or single typed inputs.

For hard decisions, the block accepts the input data types `ufix(1)`, `double`, `single`, `int8`, `uint8`, `int16`, `uint16`, `int32`, and `uint32`.

For soft decisions, the block accepts fixed-point signals (e.g., `ufix(3)`), `double`, `single`, `int8`, `uint8`, `int16`, `uint16`, `int32`, and `uint32`. The fixed-point input signal must have a fraction length of 0 and be unsigned.

### Input Values and Decision Types

The entries of the input vector are either bipolar, binary, or integer data, depending on the **Decision type** parameter.

# Viterbi Decoder

Decision type Parameter	Possible Entries in Decoder Input	Interpretation of Values
Unquantized	Real numbers	Positive real: logical zero Negative real: logical one
Hard Decision	0, 1	0: logical zero 1: logical one
Soft Decision	Integers between 0 and $2^b-1$ , where $b$ is the <b>Number of soft decision bits</b> parameter.	0: most confident decision for logical zero $2^b-1$ : most confident decision for logical one Other values represent less confident decisions.

To illustrate the soft decision situation more explicitly, the table below 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
5	Third most confident one

Input Value	Interpretation
6	Second most confident one
7	Most confident one

## Operation Modes for Frame-Based Inputs

If the input signal is frame-based, the block has three possible methods for transitioning between successive 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 frame, for use with the next frame. Each traceback path is treated independently.
- In Truncated mode, the block treats each frame 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 frame 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 frame to fill all memory registers of the feed-forward encoder. If the encoder has  $k$  input streams and constraint length vector  $\text{constr}$  (using the polynomial description), “enough” means  $k \cdot \max(\text{constr} - 1)$ . For feedback encoders, this mode is appropriate if the corresponding Convolutional Encoder block has **Operation mode** set to Terminate trellis by appending bits.

In the special case when the frame-based input signal contains only one symbol, the Continuous mode is most appropriate.

## 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 the input signal is sample-based, the decoding delay consists of  $D$  zero symbols
- If the input signal is frame-based and the **Operation mode** parameter is set 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 frame.

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 the **Enable reset input port** check box 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:

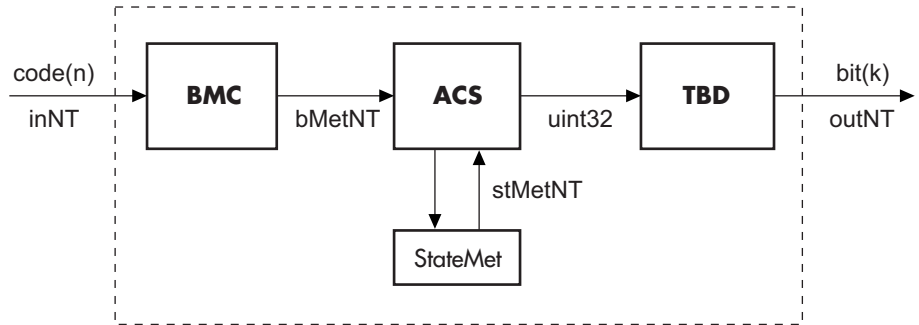
- Sets the all-zeros state metric to zero.
- Sets all other state metrics to the maximum value.
- Sets the traceback memory to zero.

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.

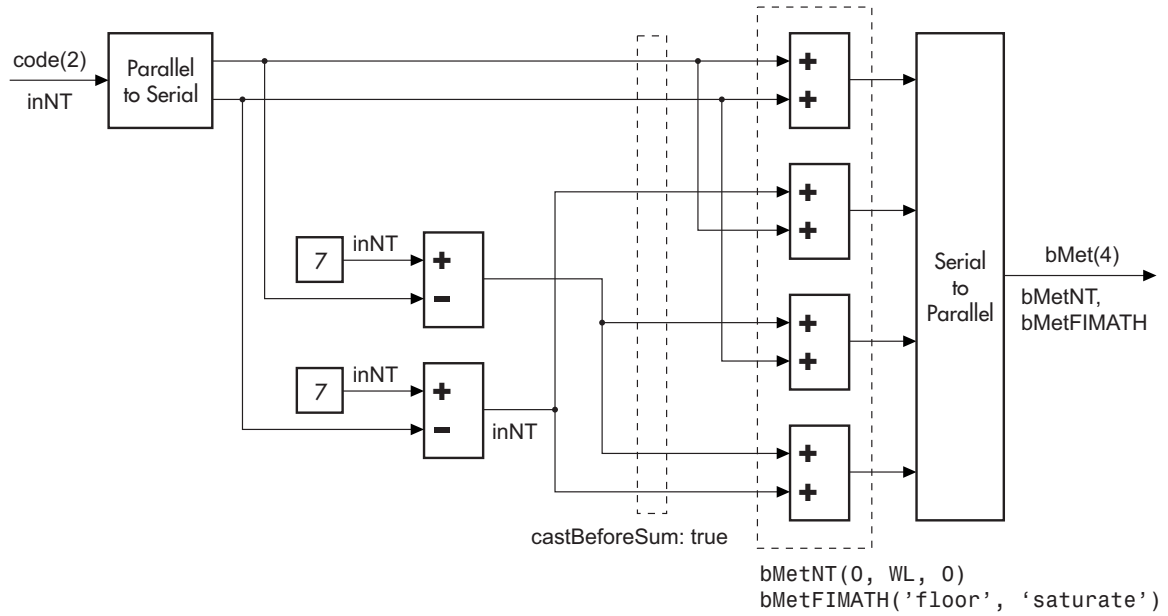
## Fixed-Point Signal Flow Diagram

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

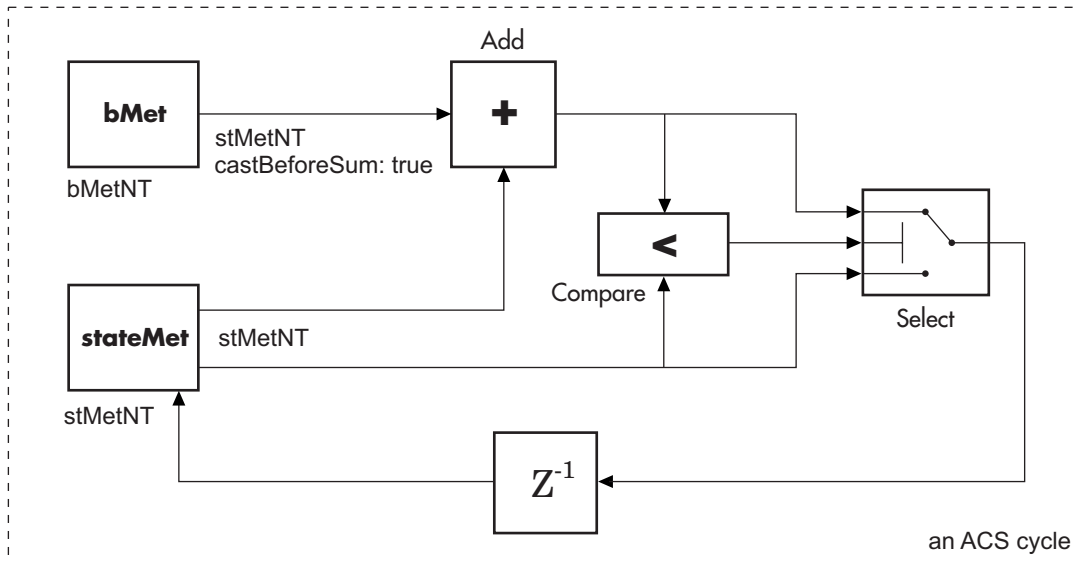
As an example of a BMC diagram, a 1/2 rate,  $nsdec = 3$  signal flow would be as follows.



$$WL = nsdec + n - 1$$

$$n = 2 \Rightarrow WL = 4$$

The ACS component is generally illustrated as shown in the following diagram.

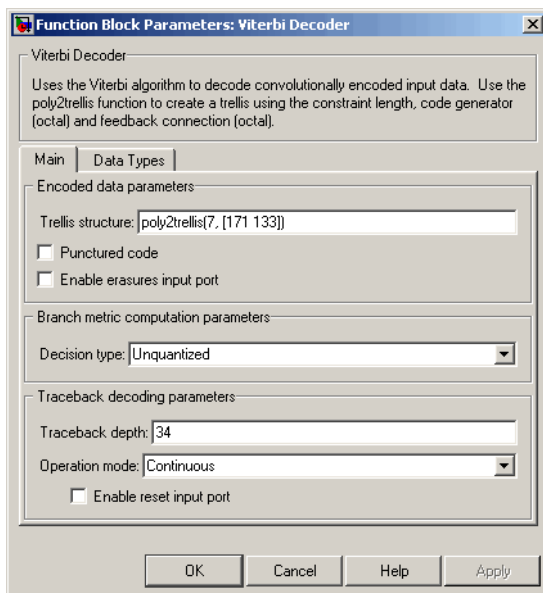


```
stMetNT(0, WL2, 0)
stMetFIMATH('floor', 'saturate')
```

Where WL2 is specified on the mask by the user.

In the flow diagrams above, `inNT`, `bMetNT`, `stMetNT`, and `outNT` are numeric type objects, and `bMetFIMATH` and `stMetFIMATH`, are fimath objects.

# Viterbi Decoder



## 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.

### Puncture vector

Constant puncture pattern vector used at the transmitter (encoder). The puncture vector is a pattern of 1s and 0s, where the 0s indicate the punctured bits. This field appears when the check box **Punctured code** is selected.

### 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 1s and 0s, where the 1s 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**

Unquantized, Hard Decision, or Soft Decision.

**Number of soft decision bits**

The number of soft decision bits used 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**

Check this 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 used to construct each traceback path.

**Operation mode**

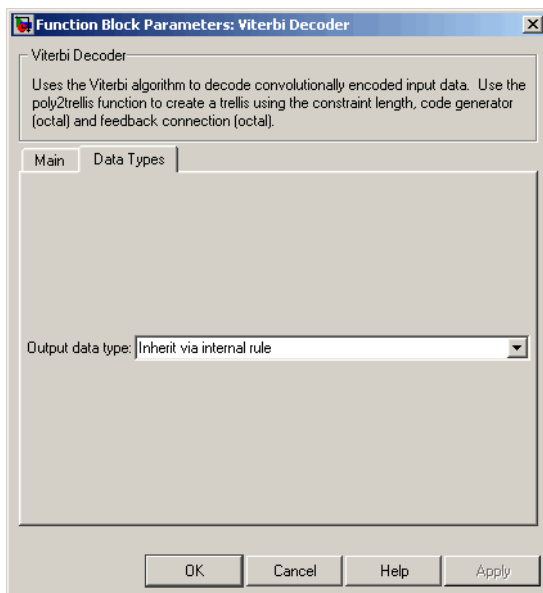
Method for transitioning between successive input frames. For frame-based input, the choices are Continuous, Terminated, and Truncated. Sample-based input must use the Continuous mode.

**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.

# 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 block. 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 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.

## Puncture Pattern Examples

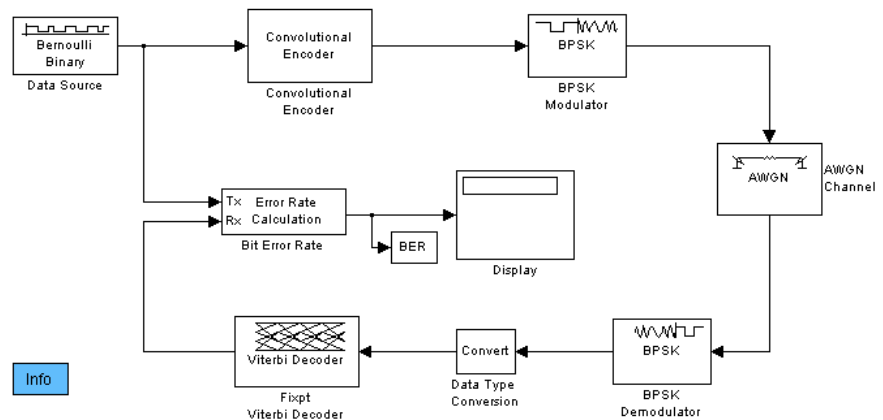
For some commonly used puncture patterns for specific rates and polynomials, see the last three references.

## 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.

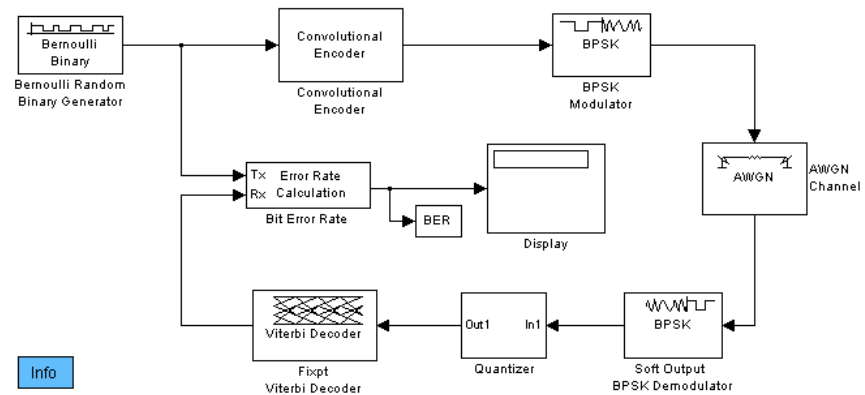
If you are reading this reference page in the MATLAB® Help Browser, click [here](#) (hard-decision) and [here](#) (soft-decision) to open the models. These can also be found as `doc_fixpt_vitharddec.mdl` and `doc_fixpt_vitsoftdec.mdl` under `help\toolbox\commblocks\commblocks_examples`.

### 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 (click here to open the model, which can be found at `help\toolbox\commblks\commblks_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.

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 1e6 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  $2^{N-1}$ . The internal branch metrics are calculated using an unsigned integer of word length = (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 **Tools > Fixed-Point Settings**. In the Fixed-Point Setting GUI, select the **Logging mode** to `Min`, `max` and `overflow`, 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

# Viterbi Decoder

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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 **Tools > Fixed-Point Settings**. In the Fixed-point Settings GUI, select the **Data type override** to be `True doubles`. 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

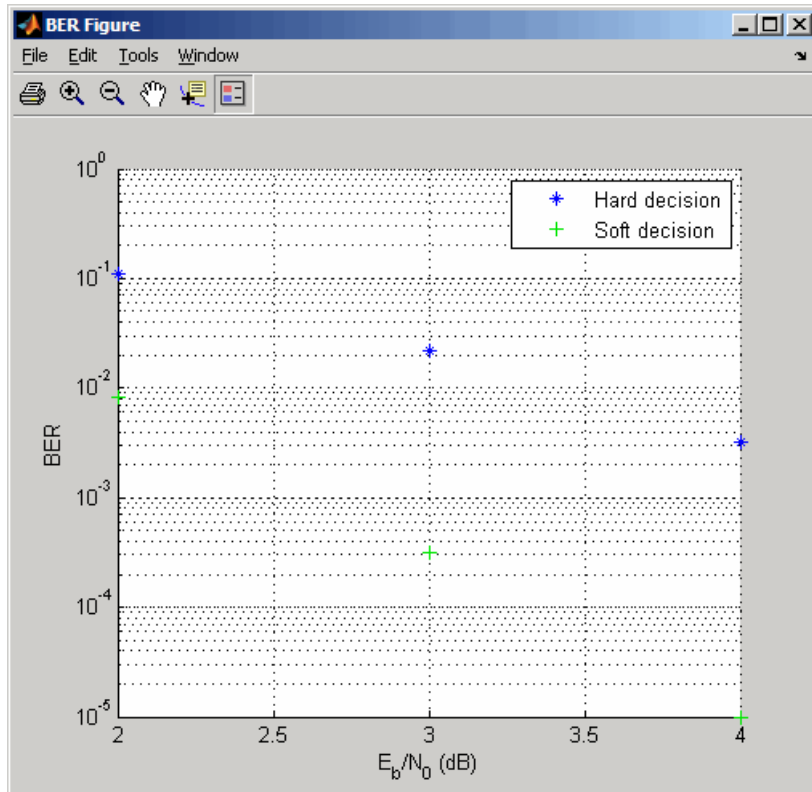
The two models are set up to run from within BERTool to generate a simulation curve that compares the BER performance for hard-decision versus soft-decision decoding.

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 1e6.

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**.

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.

# Viterbi Decoder

---

## See Also

Convolutional Encoder, APP 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.



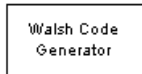
## Purpose

Generate Walsh code from orthogonal set of codes

## Library

Sequence Generators sublibrary of Comm Sources

## Description



Walsh codes are defined as a set of  $N$  codes, denoted  $W_j$ , for  $j = 0, 1, \dots, N - 1$ , which have the following properties:

- $W_j$  takes on the values +1 and -1.
- $W_j[0] = 1$  for all  $j$ .
- $W_j$  has exactly  $j$  zero crossings, for  $j = 0, 1, \dots, N - 1$ .

$$W_j W_k^T = \begin{cases} 0 & j \neq k \\ N & j = k \end{cases}$$

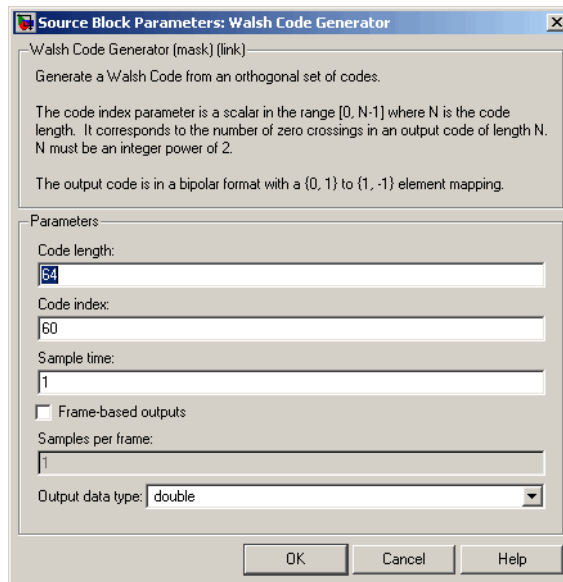
- Each code  $W_j$  is either even or odd with respect to its midpoint.

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  $[0, \dots, N - 1]$ . If you set **Walsh code index** equal to an integer  $j$ , the output code has exactly  $j$  zero crossings, for  $j = 0, 1, \dots, N - 1$ .

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



Opening this dialog box causes a running simulation to pause. See “Changing Source Block Parameters” in the online Simulink documentation for details.

### 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, ... , 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 the **Frame-based outputs** check box. 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

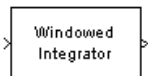
# Windowed Integrator

---

**Purpose** Integrate over time window of fixed length

**Library** Comm Filters

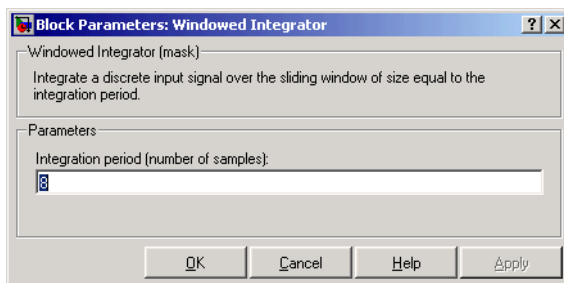
## Description



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  $x(1), x(2), x(3), \dots$ , then the  $n$ th output sample is the sum of the  $x(k)$  values for  $k$  between  $n-N+1$  and  $n$ . In cases where  $n-N+1$  is less than 1, the block uses an initial condition of 0 to represent those samples.

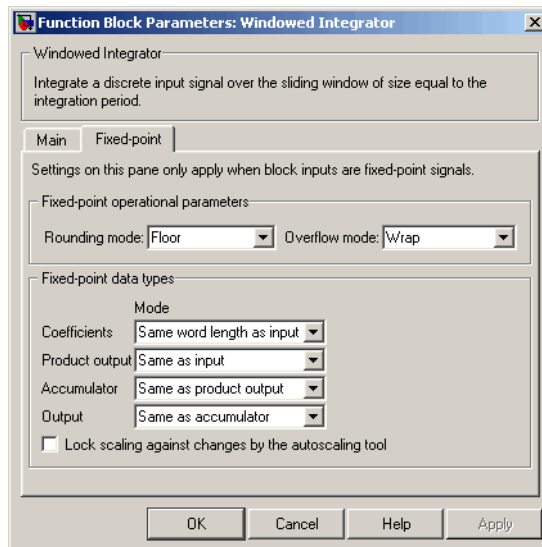
The input can be either a scalar or a frame-based matrix. If the input is frame-based, then the block processes each column independently. The output has the same sample time and matrix size as the input. `double`, `single`, and fixed-point data types are supported.

## Dialog Box



### Integration period

The length of the interval of integration, measured in samples.



## Rounding mode

Select the rounding mode for fixed-point operations. The filter coefficients do not obey this parameter; they always round to Nearest.

## 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 *Signal Processing Blockset 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 *Signal Processing Blockset 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.

## 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.

# Windowed Integrator

---

## 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.

## Examples

If **Integration period** is 3 and the input signal is a ramp (1, 2, 3, 4,...), 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. If the input signal is a sample-based scalar, then the values 1, 3, 6,... are successive values of the scalar output signal. If the input signal is a frame-based column vector, then the values 1, 3, 6,... are organized into output frames that have the same vector length as the input frames.

## See Also

Integrate and Dump, Discrete-Time Integrator (Simulink)



# Functions — Alphabetical List

---

# comm\_links

---

**Purpose** Library link information for Communications Blockset blocks

**Syntax**

```
comm_links
comm_links(sys)
comm_links(sys,color)
```

**Description** `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.

**See Also** `liblinks` (Signal Processing Blockset), `commliblist`

<b>Purpose</b>	Open main Communications Blockset library
<b>Syntax</b>	<code>commlib</code>
<b>Description</b>	<code>commlib</code> opens the latest version of the Communications Blockset library.
<b>See Also</b>	<code>simulink</code> (Simulink), <code>dsplib</code> (Signal Processing Blockset)

# commstartup

---

**Purpose** Default Simulink model settings for Communications Blockset

**Syntax** commstartup

**Description** 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 session, but not to previously created models.

---

**Note** The Signal Processing Blockset 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

**See Also** startup

---

<b>Purpose</b>	Generate prime numbers for use as random number seeds
<b>Syntax</b>	<pre>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)</pre>
<b>Description</b>	<p>The randseed function is designed for producing random prime numbers that work well as seeds for random source blocks or noisy channel blocks in Communications Blockset.</p> <p><code>out = randseed</code> generates a random prime number between 31 and <math>2^{17}-1</math>, using the MATLAB function <code>rand</code>.</p> <p><code>out = randseed(state)</code> generates a random prime number after setting the state of <code>rand</code> to the positive integer state. This syntax produces the same output for a particular value of state.</p> <p><code>out = randseed(state,m)</code> generates a column vector of <code>m</code> random primes.</p> <p><code>out = randseed(state,m,n)</code> generates an <code>m</code>-by-<code>n</code> matrix of random primes.</p> <p><code>out = randseed(state,m,n,rmin)</code> generates an <code>m</code>-by-<code>n</code> matrix of random primes between <code>rmin</code> and <math>2^{17}-1</math>.</p> <p><code>out = randseed(state,m,n,rmin,rmax)</code> generates an <code>m</code>-by-<code>n</code> matrix of random primes between <code>rmin</code> and <code>rmax</code>.</p>
<b>Examples</b>	<p>To generate a two-element sample-based row vector of random bits using the Bernoulli Random Binary Generator block, you can set <b>Probability of a zero</b> to [0.1 0.5] and set <b>Initial seed</b> to <code>randseed(391,1,2)</code>.</p> <p>To generate three streams of random data from three different blocks in a single model, you can define <code>out = randseed(93,3)</code> in the MATLAB workspace and then set the three blocks' <b>Initial seed</b> parameters to <code>out(1)</code>, <code>out(2)</code>, and <code>out(3)</code>, respectively.</p>

# randseed

---

## See Also

rand, primes

**Purpose** Save Communications Blockset model using blocks of earlier version

**Syntax** `flag = saveas_commblocks(sys,new_sys,version)`

**Description**

---

**Note** This function is obsolete.

Simulink provides a similar functionality through its **Save As...** option in the **File** menu of its models.

---

saveas\_commblocks is designed to replace Communications Blockset R14SP2 (version 3.1) blocks with R14 (version 3.0) or R14SP1 (version 3.0.1) blocks.

`flag = saveas_commblocks(sys,new_sys,version)` saves the model `sys` to a previous version specified by `version`, with the name `new_sys`. `version` must be either R14 or R14SP1. `flag` is true if the operation is successful, else it is false. It displays the list of blocks that are updated in the command window.

saveas\_commblocks updates the following blocks in the model:

- AWGN Channel
- Binary Symmetric Channel
- Multipath Rician Fading Channel
- Voltage-Controlled Oscillator or Continuous-Time VCO
- All blocks from the AM sub-library of the Digital Baseband Modulation library (commdigbbndam3). These blocks are replaced by the corresponding blocks from the old AM library (commdigbbndam2).
- All blocks from the PM sub-library of the Digital Baseband Modulation library (commdigbbndpm3). These blocks are replaced by the corresponding blocks from the old PM library (commdigbbndpm2).
- Multipath Rayleigh Fading Channel block from the Channels library (commchan3). These blocks are replaced by the Multipath Rayleigh

## saveas\_commblks

---

Fading Channel block from the old Channels library (commchan2). Note that the default parameters of this block are used after the model is saved.

### Example

```
success =  
saveas_commblks('myCommsSystem', 'myCommsSystemR14sp1', 'R14SP1')
```

saves the system myCommsSystem to one using the blocks in the R14SP1 version of Communications Blockset, and returns true on successful operation.



## A

A-Law Compressor block 2-2  
A-Law Expander block 2-4  
Algebraic Deinterleaver block 2-6  
Algebraic Interleaver block 2-9  
Align Signals block 2-12  
APP Decoder block 2-15  
AWGN Channel block 2-19

## B

Barker Code Generator block 2-26  
Baseband PLL block 2-28  
BCH Decoder block 2-30  
BCH Encoder block 2-33  
Bernoulli Binary Generator block 2-36  
Binary Cyclic Decoder block 2-39  
Binary Cyclic Encoder block 2-41  
Binary Linear Decoder block 2-48  
Binary Linear Encoder block 2-50  
Binary Symmetric Channel block 2-55  
Binary-Input RS Encoder block 2-43  
Binary-Output RS Decoder block 2-51  
Bipolar to Unipolar Converter block 2-57  
Bit to Integer Converter block 2-59  
block coding library  
    reference for 1-4  
block interleaving library  
    reference for 1-6  
BPSK Demodulator Baseband block 2-61  
BPSK Modulator Baseband block 2-63

## C

carrier phase recovery library 1-13  
Channels library  
    reference for 1-12  
Charge Pump PLL block 2-65  
CMA Equalizer block 2-68  
Comm Sinks library

    reference for 1-3  
Comm Sources library  
    reference for 1-2  
comm\_links function 3-2  
commlib function 3-3  
commstartup function 3-4  
Communication Filters library  
    reference for 1-11  
Complex Phase Difference block 2-72  
Complex Phase Shift block 2-73  
convolutional coding library  
    reference for 1-5  
Convolutional Deinterleaver block 2-76  
Convolutional Encoder block 2-78  
Convolutional Interleaver block 2-82  
convolutional interleaving library  
    reference for 1-7  
CPFSK Demodulator Baseband block 2-85  
CPFSK Modulator Baseband block 2-89  
CPM Demodulator Baseband block 2-92  
CPM Modulator Baseband block 2-97  
CPM Phase Recovery block 2-102  
CRC library  
    reference for 1-6  
CRC-N Generator block 2-105  
CRC-N Syndrome Detector block 2-109

## D

Data Mapper block 2-112  
DBPSK Demodulator Baseband block 2-115  
DBPSK Modulator Baseband block 2-117  
Deinterlacer block 2-119  
Derepeat block 2-121  
Descrambler block 2-124  
Differential Decoder block 2-126  
Differential Encoder block 2-128  
digital modulation libraries  
    reference for 1-8  
Discrete-Time Eye Diagram Scope block 2-130

Discrete-Time Scatter Plot Scope block 2-139  
Discrete-Time Signal Trajectory Scope  
  block 2-147  
Discrete-Time VCO block 2-154  
DQPSK Demodulator Baseband block 2-156  
DQPSK Modulator Baseband block 2-159  
DSB AM Demodulator Passband block 2-163  
DSB AM Modulator Passband block 2-166  
DSBSC AM Demodulator Passband block 2-168  
DSBSC AM Modulator Passband block 2-171

## E

Early-Late Gate Timing Recovery block 2-173  
Equalizers library  
  reference for 1-14  
Error Detection and Correction library  
  reference for 1-4  
Error Rate Calculation block 2-177

## F

Find Delay block 2-186  
FM Demodulator Passband block 2-191  
FM Modulator Passband block 2-194  
Free Space Path Loss block 2-196

## G

Gardner Timing Recovery block 2-198  
Gaussian Filter block 2-203  
Gaussian Noise Generator block 2-211  
General Block Deinterleaver block 2-215  
General Block Interleaver block 2-217  
General CRC Generator block 2-219  
General CRC Syndrome Detector block 2-225  
General Multiplexed Deinterleaver block 2-229  
General Multiplexed Interleaver block 2-231  
General QAM Demodulator Baseband  
  block 2-233

General QAM Modulator Baseband block 2-236  
General TCM Decoder block 2-239  
General TCM Encoder block 2-243  
GMSK Demodulator Baseband block 2-248  
GMSK Modulator Baseband block 2-251  
Gold Sequence Generator block 2-254

## H

Hadamard Code Generator block 2-261  
Hamming Decoder block 2-264  
Hamming Encoder block 2-266  
Helical Deinterleaver block 2-268  
Helical Interleaver block 2-271

## I

I/Q Imbalance block 2-302  
Ideal Rectangular Pulse Filter block 2-274  
Insert Zero block 2-282  
Integer to Bit Converter block 2-294  
Integer-Input RS Encoder block 2-285  
Integer-Output RS Decoder block 2-290  
Integrate and Dump block 2-296  
Interlacer block 2-301  
Interleaving library  
  reference for 1-6

## K

Kasami Sequence Generator block 2-309

## L

LDPC Decoder block 2-316  
LDPC Encoder block 2-319  
Linearized Baseband PLL block 2-321  
LMS Decision Feedback Equalizer block 2-323  
LMS Linear Equalizer block 2-328

**M**

M-DPSK Demodulator Baseband block 2-342  
 M-DPSK Modulator Baseband block 2-345  
 M-FSK Demodulator Baseband block 2-363  
 M-FSK Modulator Baseband block 2-366  
 M-PAM Demodulator Baseband block 2-377  
 M-PAM Modulator Baseband block 2-385  
 M-PSK Demodulator Baseband block 2-390  
 M-PSK Modulator Baseband block 2-394  
 M-PSK Phase Recovery block 2-401  
 M-PSK TCM Decoder block 2-404  
 M-PSK TCM Encoder block 2-408  
 Matrix Deinterleaver block 2-333  
 Matrix Helical Scan Deinterleaver block 2-335  
 Matrix Helical Scan Interleaver block 2-337  
 Matrix Interleaver block 2-340  
 Memoryless Nonlinearity block 2-349  
 MLSE Equalizer block 2-370  
 Modulation library  
     reference for 1-8  
 MSK Demodulator Baseband block 2-412  
 MSK Modulator Baseband block 2-415  
 MSK-Type Signal Timing Recovery block 2-417  
 Mu-Law Compressor block 2-425  
 Mu-Law Expander block 2-427  
 Mueller-Muller Timing Recovery block 2-421  
 Multipath Rayleigh Fading Channel block 2-429  
 Multipath Rician Fading Channel block 2-432

**N**

Normalized LMS Decision Feedback Equalizer  
     block 2-436  
 Normalized LMS Linear Equalizer block 2-441

**O**

OQPSK Demodulator Baseband block 2-446  
 OQPSK Modulator Baseband block 2-448  
 OVFSF Code Generator block 2-452

**P**

Phase Noise block 2-466  
 Phase/Frequency Offset block 2-457  
 Phase-Locked Loop block 2-463  
 PM Demodulator Passband block 2-470  
 PM Modulator Passband block 2-473  
 PN Sequence Generator block 2-475  
 Poisson Integer Generator block 2-486  
 Puncture block 2-489

**Q**

QPSK Demodulator Baseband block 2-492  
 QPSK Modulator Baseband block 2-495  
 Quantizing Decoder block 2-499  
 Quantizing Encoder block 2-501

**R**

Raised Cosine Receive Filter block 2-503  
 Raised Cosine Transmit Filter block 2-511  
 Random Deinterleaver block 2-519  
 Random Integer Generator block 2-521  
 Random Interleaver block 2-524  
 randseed function 3-5  
 Rayleigh Noise Generator block 2-526  
 Receiver Thermal Noise block 2-529  
 Rectangular QAM Demodulator Baseband  
     block 2-532  
 Rectangular QAM Modulator Baseband  
     block 2-543  
 Rectangular QAM TCM Decoder block 2-550  
 Rectangular QAM TCM Encoder block 2-553  
 Rician Noise Generator block 2-558  
 RLS Decision Feedback Equalizer block 2-562  
 RLS Linear Equalizer block 2-567

**S**

saveas\_commblocks function 3-7

- Scrambler block 2-572
- Sequence Operations library 1-16
- Sign LMS Decision Feedback Equalizer block 2-574
- Sign LMS Linear Equalizer block 2-579
- sinks library
  - reference for 1-3
- Source Coding library
  - reference for 1-3
- sources library
  - reference for 1-2
- Squaring Timing Recovery block 2-584
- SSB AM Demodulator Passband block 2-587
- SSB AM Modulator Passband block 2-589
- synchronization components library 1-14
- Synchronization library
  - reference for 1-13

## **T**

- timing phase recovery library 1-13

## **U**

- Uniform Noise Generator block 2-594
- Unipolar to Bipolar Converter block 2-597
- Utility Blocks library 1-16

## **V**

- Variable Step LMS Decision Feedback Equalizer block 2-599
- Variable Step LMS Linear Equalizer block 2-604
- Viterbi Decoder block 2-609
- Voltage-Controlled Oscillator block 2-74

## **W**

- Walsh Code Generator block 2-625
- Windowed Integrator block 2-628