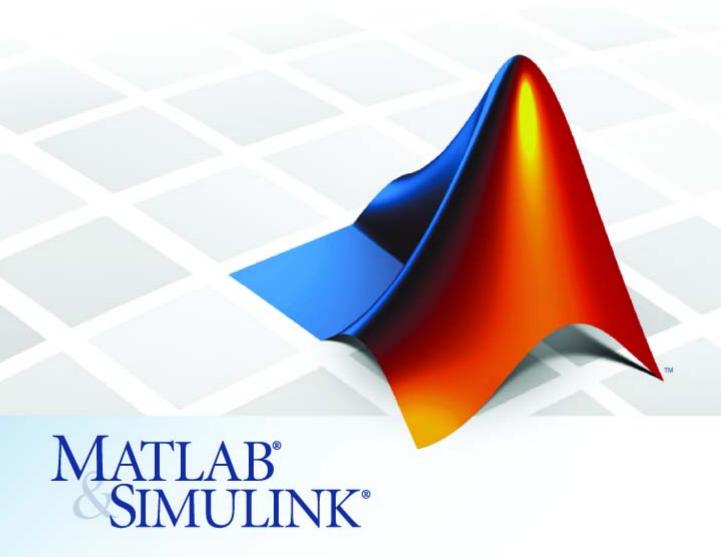
# Communications Blockset<sup>™</sup> 4 Reference





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For contact information about worldwide offices, see the MathWorks Web site.

Communications Blockset™ Reference

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#### **Revision History**

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July 2002	Online only	Revised for Version 2.5 (Release 13)
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October 2008	Online only	Revised for Version 4.1 (Release 2008b)
March 2009	Online only	Revised for Version 4.2 (Release 2009a)
September 2009	Online only	Revised for Version 4.3 (Release 2009b)
March 2010	Online only	Revised for Version 4.4 (Release 2010a)

## **Block Reference**

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Communications Sinks (p. 1-3) Error statistics and plotting

Source Coding (p. 1-3) Quantization, companding, and

differential coding

Error Detection and Correction Block, convolutional, and CRC

(p. 1-4) coding

Interleaving (p. 1-6) Block and convolutional interleaving

MIMO (p. 1-8) Multiple Input Multiple Output

blocks

Modulation (p. 1-8) Digital baseband and analog

passband modulation

Communications Filters (p. 1-12) Filtering and pulse shaping

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radio frequency components

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Utility Blocks (p. 1-17) 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 μ-law compressor for

source coding

Mu-Law Expander Implement μ-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

#### **MIMO**

OSTBC Combiner Combine inputs for received signals

and channel estimate according to orthogonal space-time block code

(OSTBC)

OSTBC Encoder Encode input message using

orthogonal space-time block code

(OSTBC)

#### **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. 1-10)

Digital Baseband CPM Sublibrary

(p. 1-10)

Digital Baseband TCM Sublibrary

(p. 1-11)

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

M-PAM Modulator Baseband

Modulate using M-ary pulse amplitude modulation

Rectangular QAM Demodulator
Baseband

Rectangular QAM Modulator
Baseband

Modulate using M-ary pulse amplitude modulation

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	
M-PSK Demodulator Baseband M-PSK Modulator Baseband	phase shift keying method
	phase shift keying method  Demodulate PSK-modulated data  Modulate using M-ary phase shift
M-PSK Modulator Baseband	phase shift keying method  Demodulate PSK-modulated data  Modulate using M-ary phase shift keying method  Demodulate OQPSK-modulated

QPSK Demodulator Baseband QPSK Modulator Baseband Demodulate QPSK-modulated data Modulate using quaternary phase shift keying method

### **Digital Baseband FM Sublibrary**

M-FSK Demodulator Baseband M-FSK Modulator Baseband Demodulate FSK-modulated data 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. 1-14)

Timing Phase Recovery (p. 1-14)

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

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

EVM Measurement Calculate vector magnitude

difference between ideal reference

signal and measured signal

Find Delay Find delay between two signals

Integer to Bit Converter Map vector of integers to vector of

bits

MER Measurement Measure signal-to-noise ratio (SNR)

in digital modulation applications

Unipolar to Bipolar Converter Map unipolar signal in range [0, M-1]

into bipolar signal

## Blocks — Alphabetical List

## **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

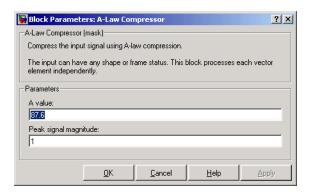
$$y = \begin{cases} \frac{A|x|}{1 + \log A} \operatorname{sgn}(x) & \text{for } 0 \le |x| \le \frac{V}{A} \\ \frac{V(1 + \log(A|x|/V))}{1 + \log A} \operatorname{sgn}(x) & \text{for } \frac{V}{A} < |x| \le V \end{cases}$$

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

The most commonly used A value is 87.6.

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

#### 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.

## Supported Data Type

Port	Supported Data Types
In	• double
Out	• double

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**

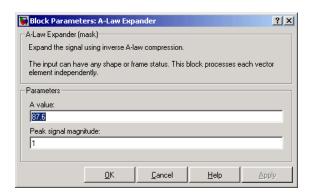


The A-Law Expander block recovers data that the A-Law Compressor block compressed. The formula for the A-law expander, shown below, is the inverse of the compressor function.

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

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.

## Supported Data Type

Port	Supported Data Types
In	• double
Out	• double

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**

**Purpose** 

Restore ordering of input symbols using algebraically derived

permutation

Library

Block sublibrary of Interleaving

**Description** 

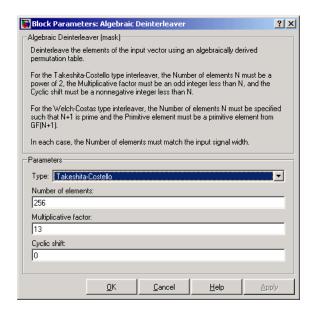
Algebraic Deinterleaver The Algebraic Deinterleaver block restores the original ordering of a sequence that was interleaved using the Algebraic Interleaver block. In typical usage, the parameters in the two blocks have the same values.

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 Takeshita-Costello and Welch-Costas. Each of these methods has parameters and restrictions that are specific to it; these are described on the reference page for the Algebraic Interleaver block.

#### 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**

#### 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.

## **Algebraic Interleaver**

#### **Purpose**

Reorder input symbols using algebraically derived permutation table

#### Library

Block sublibrary of Interleaving

#### **Description**

Algebraic Interleaver 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  $mod(A^k,N+1)$  - 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 nth element is

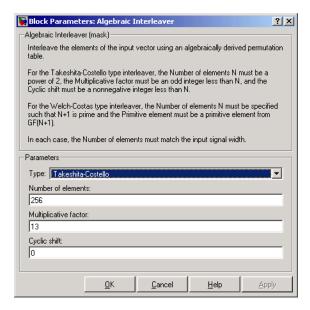
$$mod(k*(n-1)*n/2, N)$$

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.

# **Algebraic Interleaver**

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 \$1 receives the original signal, while the input port labeled \$2 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<sup>™</sup> 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.

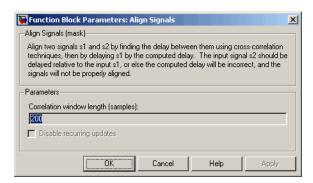
## **Examples**

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 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  $\mathbf{Q}=\mathbf{1}$
- 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<sup>TM</sup> 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

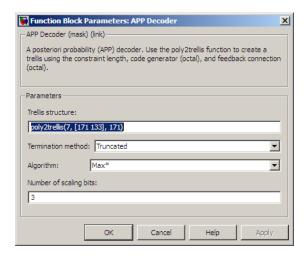
$$\log \sum_{i} \exp(a_i)$$

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

$$ln(1 + exp(-|a_{i-1} - a_i|))$$
 [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^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.

## **APP Decoder**

#### 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

#### **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** All values of power assume 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.

- 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 (Eb/No), where the block calculates the variance from these quantities that you specify in the dialog box:
  - **Eb/No**, the ratio of bit energy to noise power spectral density
  - Number of bits per symbol
  - Input signal power, the actual power of the symbols at the input of the block
  - Symbol period
- Signal to noise ratio (Es/No), where the block calculates the variance from these quantities that you specify in the dialog box:
  - **Es/No**, the ratio of signal energy to noise power spectral density
  - Input signal power, the actual power of the symbols at the input of the block
  - 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.

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

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.

**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 Eb/No, Es/No, 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:

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

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

where

- E<sub>s</sub> = Signal energy (Joules)
- $E_b = Bit energy (Joules)$
- $N_0$  = Noise power spectral density (Watts/Hz)
- $\bullet~T_{\rm sym}$  is the  ${\bf Symbol~period}$  parameter of the block in Es/No mode
- k is the number of information bits per input symbol
- $\bullet~T_{\mathrm{samp}}$  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:

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

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  $N_{\rm 0}/2$  Watts/Hz for real input signals, versus  $N_{\rm 0}$  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.

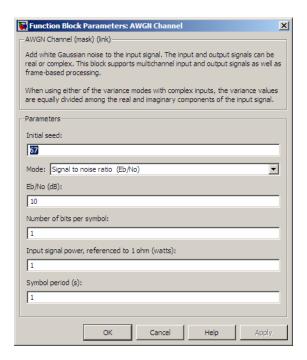
#### **Tunable Block Parameters**

The following table indicates which parameters are tunable, for different block modes.

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

You can tune parameters in normal mode, Accelerator mode and the Rapid Accelerator mode.

If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameters listed in the previous table without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.



## 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, referenced to 1 ohm (watts)

The mean square power of the input symbols (if **Mode** is Eb/No or Es/No) or input samples (if **Mode** is SNR), in watts. This field appears only if **Mode** is set to Eb/No, Es/No, or SNR.

#### 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 documentation)

## 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 Code Generator 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,  $\{X_i\}$ , is given by

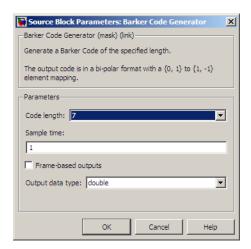
$$C_k = \sum_{j=1}^{N-k} X_j X_{j+k}$$

where  $X_j$  is an individual code symbol taking values +1 or -1 for j=1, 2, 3,..., 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]

## 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 block's output is frame-based or sample-based.

#### Samples per frame

The number of samples in a frame-based output signal. This field appears 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**

## **Purpose**

Implement baseband phase-locked loop

## Library

Components sublibrary of Synchronization

## **Description**



The Baseband PLL (phase-locked loop) block is a feedback control system that automatically adjusts the phase of a locally generated signal to match the phase of an input signal. Unlike the Phase-Locked Loop block, this block uses a baseband method and does not depend on a carrier frequency.

This PLL has these three components:

- An integrator used as a phase detector.
- A filter. You specify the filter's transfer function using the Lowpass filter numerator and Lowpass filter denominator parameters.
   Each is a vector that gives the respective polynomial's coefficients in order of descending powers of s.

To design a filter, you can use the Signal Processing Toolbox<sup>TM</sup> functions cheby1, and cheby2. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.

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

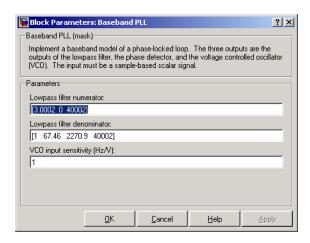
• A voltage-controlled oscillator (VCO). You specify the sensitivity of the VCO signal to its input using the **VCO input sensitivity** parameter. This parameter, measured in Hertz per volt, is a scale factor that determines how much the VCO shifts from its quiescent frequency.

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**

**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 - the number of punctures) elements. Each group of N input elements represents one codeword to be decoded. The values of (N + shortening length) and (K + shortening length) must produce a valid narrow-sense BCH code.

If the decoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

For a given codeword length N, only specific message lengths K are valid for a BCH code. For a full length BCH code, N must be of the form  $2^M$ -1, where  $3 \le M \le 16$ . If N is less than  $2^M$ -1, the block assumes that the code has been shortened by length  $2^{M-1}$  - N. However, if N is greater than or equal to  $2^{M-1}$ , **Primitive polynomial** must be specified to appropriately set the value of M.

No known analytic formula describes the relationship among the codeword length, message length, and error-correction capability. For a list of some valid values of K corresponding to values of N up to 511, see the behenc 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.

In the case of a decoder failure, the message portion of the decoder input is returned unchanged as the decoder output.

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

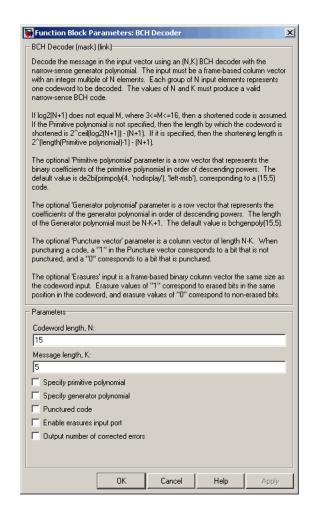
For information about the data types each block port supports, see the "Supported Data Type" on page 2-34 table on this page.

# Punctured Codes

This block supports puncturing when you select the **Punctured code** parameter. This selection enables the **Puncture vector** parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 means that the data symbol is passed unaltered, and 0 means that the data symbol is punctured (i.e., removed) from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

**Note** 1s and 0s have precisely opposite meanings for the puncture and erasure vectors. For an erasure vector, 1 means that the data symbol is to be replaced with an erasure symbol, and 0 means that the data symbol is passed unaltered. This convention is carried for both the encoder and the decoder.

## 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

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

This field defaults to de2bi(primpoly(4, 'nodisplay'), 'left-msb'), corresponding to a (15,5) code.

This field appears only when you select **Specify primitive polynomial**.

#### Specify generator polynomial

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

#### Generator polynomial

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

The length of the **Generator polynomial** must be *N-K*+1.

This field defaults to bchgenpoly(15,5).

This field appears only when you select **Specify generator** polynomial.

#### Disable generator polynomial checking

This check box appears only when you select **Specify generator** polynomial.

Each time a model initializes, the block performs a polynomial check. This check verifies that  $X^N+1$  is divisible by the user-defined generator polynomial, where N represents the full code word length. Selecting this check box disables the polynomial check. For larger codes, disabling the check speeds up the simulation process. You should always run the check at least once before disabling this feature.

#### Puncture code

Selecting this check box enables the field **Puncture vector**.

#### **Puncture vector**

This field is available only when **Puncture code** is selected.

A column vector of length *N-K*. A value of 1 in the **Puncture vector** corresponds to a bit that is not punctured, and a 0 corresponds to a bit that is punctured.

The default value is [ones(8,1); zeros(2,1)].

#### Enable erasures input port

Selecting this check box will open the Era port.

Through the Era port, you can input a frame-based binary column vector the same size as the codeword input.

Erasure values of 1 correspond to erased bits in the same position in the codeword, and values of 0 correspond to nonerased bits.

#### Output number of corrected errors

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

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

Port	Supported Data Types
Out	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
Era	Double-precision floating point
	Boolean
Err	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

#### **Pair Block**

BCH Encoder

## References

- [1] Wicker, Stephen B., Error Control Systems for Digital Communication and Storage, Upper Saddle River, N.J., Prentice Hall, 1995.
- [2] Berlekamp, Elwyn R., *Algebraic Coding Theory*, New York, McGraw-Hill, 1968.
- [3] Clark, George C., Jr., and J. Bibb Cain, *Error-Correction Coding for Digital Communications*, New York, Plenum Press, 1981.

# **BCH Decoder**

## See Also

bchdec (in Communications Toolbox documentation)

**Purpose** 

Create BCH code from binary vector data

Library

Block sublibrary of Channel Coding

## **Description**



This block supports punctures ("Shortening, Puncturing, and Erasures" provides a tutorial).

The BCH Encoder block creates a BCH code with message length K and codeword length (N - number of punctures). You specify both N and K directly in the dialog box.

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.

If the encoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

For a given codeword length N, only specific message lengths K are valid for a BCH code. For a full length BCH code, N must be of the form  $2^{M}$ -1, where  $3 \le M \le 16$ . If N is less than  $2^{M}$ -1, the block assumes that the code has been shortened by length  $2^{M-1}$  - N. However, if N is greater than or equal to  $2^{M-1}$ , **Primitive polynomial** must be specified to appropriately set the value of M.

No known analytic formula describes the relationship among the codeword length, message length, and error-correction capability. For a list of some valid values of K corresponding to values of N up to 511, see the behenc 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.

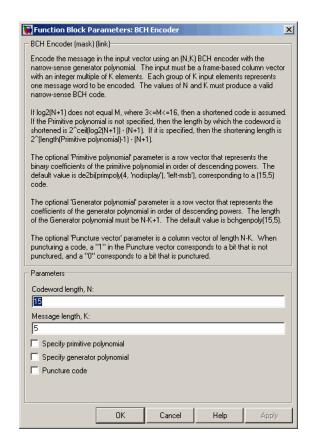
For information about the data types each block port supports, see the "Supported Data Type" on page 2-41 table on this page.

# Puncture Codes

This block supports puncturing when you select the **Puncture code** parameter. This selection enables the **Puncture vector** parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 means that the data symbol is passed unaltered, and 0 means that the data symbol is punctured (i.e., removed) from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

**Note** 1s and 0s have precisely opposite meanings for the puncture and erasure vectors. For an erasure vector, 1 means that the data symbol is to be replaced with an erasure symbol, and 0 means that the data symbol is passed unaltered. This convention is carried for both the encoder and the decoder.

## 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

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

This field defaults to de2bi(primpoly(4, 'nodisplay'), 'left-msb'), corresponding to a (15,5) code.

This field is available only when you select **Specify primitive polynomial**.

#### Specify generator polynomial

Selecting this check box enables the field Generator polynomial.

#### Generator polynomial

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

The length of the **Generator polynomial** must be *N-K*+1.

This field defaults to bchgenpoly (15,5).

This field appears only when you select **Specify generator** polynomial.

#### Disable generator polynomial checking

This check box appears only when you select **Specify generator polynomial**.

Each time a model initializes, the block performs a polynomial check. This check verifies that  $X^N+1$  is divisible by the user-defined generator polynomial, where N represents the full code word length. Selecting this check box disables the polynomial check. For larger codes, disabling the check speeds up the simulation process. You should always run the check at least once before disabling this feature.

#### Puncture code

Selecting this check box enables the field **Puncture vector**.

#### **Puncture vector**

A column vector of length *N-K*. A value of 1 in the **Puncture vector** corresponds to a bit that is not punctured, and a 0 corresponds to a bit that is punctured.

The field defaults to [ones(8,1); zeros(2,1)].

This field appears only when you select **Puncture code**.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
Out	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

**Pair Block** 

BCH Decoder

References

[1] Clark, George C., Jr., and J. Bibb Cain, *Error-Correction Coding for Digital Communications*, New York, Plenum Press, 1981.

**See Also** 

bchenc (in Communications Toolbox documentation)

## **Bernoulli Binary Generator**

**Purpose** 

Generate Bernoulli-distributed random binary numbers

Library

Random Data Sources sublibrary of Comm Sources

**Description** 



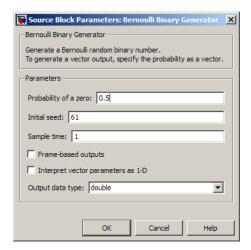
The Bernoulli Binary Generator block generates random binary numbers using a Bernoulli distribution. The Bernoulli distribution with parameter p produces zero with probability p and one with probability 1-p. The Bernoulli distribution has mean value 1-p and variance p(1-p). The **Probability of a zero** 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 (in Communications Toolboxdocumentation), rand (built-in MATLAB function)

# **Binary Cyclic Decoder**

#### **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 correspondingBinary 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.

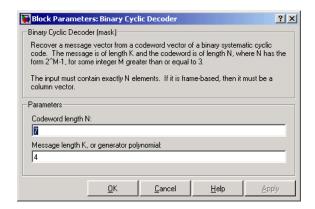
You can determine the systematic cyclic coding scheme in one of two ways:

- 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 Communications Toolbox cyclpoly function.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-46 table on this page.

# **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.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	Fixed-point
Out	Double-precision floating point
	Single-precision floating point

# **Binary Cyclic Decoder**

	Port	Supported Data Types
Pair Block See Also	Binary Cyclic Encoder  cyclpoly (Communications Toolbox	<ul> <li>Boolean</li> <li>8-, 16-, and 32-bit signed integers</li> <li>8-, 16-, and 32-bit unsigned integers</li> <li>Fixed-point</li> </ul>

### **Binary Cyclic Encoder**

#### **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.

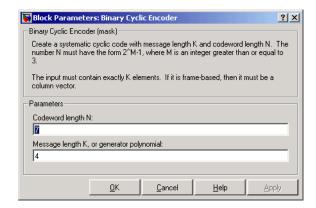
The input must contain exactly K elements. If it is frame-based, then it must be a column vector. The output is a vector of length N.

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 Communications Toolbox cyclpoly function.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-49 table on this page.

#### 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.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	Fixed-point
Out	Double-precision floating point
	Single-precision floating point

# **Binary Cyclic Encoder**

	Port	Supported Data Types
Pair Block See Also	Binary Cyclic Decoder  cyclpoly (in the Communications T	<ul> <li>Boolean</li> <li>8-, 16-, and 32-bit signed 'oolbote@mentation'</li> <li>8-, 16-, and 32-bit unsigned integers</li> <li>Fixed-point</li> </ul>

**Purpose** 

Create Reed-Solomon code from binary vector data

Library

Block sublibrary of Channel Coding

field for the code.

#### **Description**



The Binary-Input RS Encoder block creates a Reed-Solomon code with message length, K, and codeword length, (N - number of punctures). You specify both N and K directly in the dialog box. The symbols for the code are binary sequences of length M, corresponding to elements of the Galois field  $GF(2^M)$ , 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 2-52 below. The difference N-K must be an even integer.

This block can output shortened codewords when N and K are appropriately specified. To specify output codewords that are shortened by a length S, N and K must be specified in the dialog box as  $N_{\rm full} - S$  and  $K_{\rm full} - S$ , where  $N_{\rm full}$  and  $K_{\rm full}$  are the N and K of an unshortened code. If  $S < (N_{\rm full} + 1)/2$ , the encoder can automatically determine the value of  $N_{\rm full}$  and  $K_{\rm full}$ . However, if  $S \ge (N_{\rm full} + 1)/2$ , **Primitive polynomial** must be specified in order to properly define the extension

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 output is a frame-based column vector whose length is the same integer multiple of M\*(N - number of punctures), and whose data type is inherited from the input. For information about the data types each block port supports, see the "Supported Data Type" on page 2-56 table on this page.

For more information on representing data for Reed-Solomon codes, see the section "Integer Format (Reed-Solomon Only)" in *Communications Blockset User's Guide*.

If the encoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

The default value of M is the smallest integer that is greater than or equal to  $\log 2(N+1)$ , that is,  $\operatorname{ceil}(\log 2(N+1))$ . You can change the value of M from the default by specifying the primitive polynomial for  $\operatorname{GF}(2^M)$ , as described in "Specifying the Primitive Polynomial" on page 2-52 below. If N is less than  $2^M$ -1, the block assumes that the code has been shortened by length  $2^{M-1}$  - N.

Each M\*K input bits represent K integers between 0 and  $2^{M}$ -1. Similarly, each M\*(N - number of punctures) 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 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, 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  $3 \le N < 2^{16}-1$  and M must lie in the range  $3 \le M \le 16$ .

#### **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

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

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).

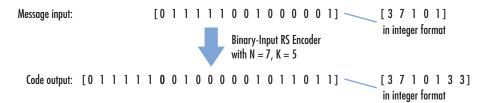
#### **Puncture Codes**

The block supports puncturing when you select the **Puncture code** parameter. This enables the **Puncture vector** parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, a 1 means that the data symbol is passed unaltered, and a 0 means that the data symbol is punctured (i.e., removed) from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

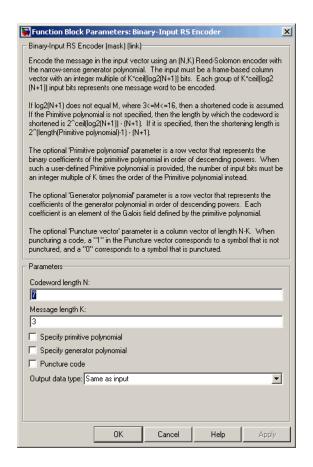
#### **Examples**

Suppose 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.



#### Dialog Box



#### Codeword length N

The codeword length. The output has vector length NC\*M\*(N - NP), where NC is the number of codewords being output, and NP is the number of punctures per codeword.

#### Message length K

The message length. The input has vector length NM\*M\*K, where NM is the number of messages per frame being input.

#### 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.

#### Puncture code

Selecting this check box enables the field **Puncture vector**.

#### **Puncture vector**

This field is available only when **Puncture code** is selected.

A column vector of length N-K. A value of 1 in the **Puncture vector** corresponds to an M-bit symbol that is not punctured, and a 0 corresponds to an M-bit symbol that is punctured.

The default value is [ones(2,1); zeros(2,1)].

#### 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.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	• 1-bit unsigned integer (ufix(1))
Out	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	• 1-bit unsigned integer (ufix(1))

**Pair Block** 

Binary-Output RS Decoder

**See Also** 

Integer-Input RS Encoder

### **Binary Linear Decoder**

**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.

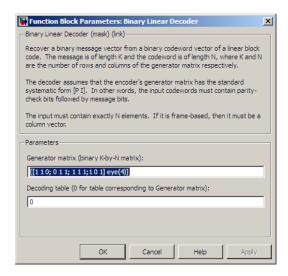
The input must contain exactly N elements. If it is frame-based, then it must be a column vector. The output is a vector of length K.

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 rth 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.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-58 table on this page.

### **Binary Linear Decoder**

#### 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.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers

## **Binary Linear Decoder**

Port	Supported Data Types
	• 8-, 16-, and 32-bit unsigned integers
	Fixed-point
Out	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	Fixed-point

### **Pair Block**

Binary Linear Encoder

### **Binary Linear Encoder**

**Purpose** 

Create linear block code from binary vector data

Library

Block sublibrary of Channel Coding

**Description** 

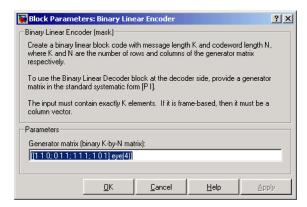


The Binary Linear Encoder block creates a binary linear block code using a generator matrix that you specify. If K is the message length of the code, then the **Generator matrix** parameter must have K rows. If N is the codeword length of the code, then **Generator matrix** must have N columns.

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.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-61 table on this page.

#### Dialog Box



#### Generator matrix

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

### Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	• Fixed-point
Out	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	Fixed-point

**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-number of punctures). You specify both N and K directly in the dialog box. The symbols for the code are binary sequences of length M, corresponding to elements of the Galois field  $GF(2^M)$ , 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 2-52. The difference N-K must be an even integer.

This block can output shortened codewords when N and K are appropriately specified. To specify output codewords that are shortened by a length S, N and K must be specified in the dialog box as  $N_{\rm full}-S$  and  $K_{\rm full}-S$ , where  $N_{\rm full}$  and  $K_{\rm full}$  are the N and K of an unshortened code. If  $S < (N_{\rm full}+1)/2$ , the encoder can automatically determine

the value of  $N_{\rm full}$  and  $K_{\rm full}$ . However, if  $S \ge (N_{\rm full} + 1)/2$ , **Primitive polynomial** must be specified in order to properly define the extension field for the code.

The input and output are binary-valued signals that represent codewords and messages, respectively. The input must be a frame-based column vector whose length is an integer multiple of M\*(N - number of punctures). The output is a frame-based column vector whose length is the same integer multiple of M\*K, and whose data type is inherited from the input. For information about the data types each block port supports, see the table on this page.

For more information on representing data for Reed-Solomon codes, see "Integer Format (Reed-Solomon Only)" in *Communications Blockset User's Guide*.

If the decoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

The default value of M is ceil(log2(N+1)), that is, the smallest integer greater than or equal to log2(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 2-52 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 2-53.

Each M\*K input bits represent K integers between 0 and  $2^{M}$ -1. Similarly, each M\*(N - number of punctures) output bits represent N integers between 0 and  $2^{M}$ -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 floor((N-K)/2) symbol errors (not bit errors) in each codeword.

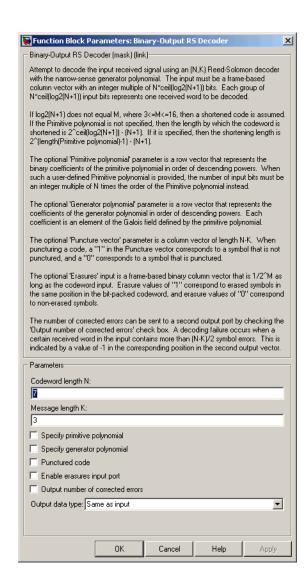
You can disable the second output by deselecting **Output port for number of corrected errors.** This removes the block's second output port.

# Punctured Codes

This block supports puncturing when you select the **Punctured code** parameter. This selection enables the **Puncture vector** parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 means that the data symbol is passed unaltered, and 0 means that the data symbol is punctured (i.e., removed) from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

**Note** 1s and 0s have precisely opposite meanings for the puncture and erasure vectors. For an erasure vector, 1 means that the data symbol is to be replaced with an erasure symbol, and 0 means that the data symbol is passed unaltered. This convention is carried for both the encoder and the decoder.

#### Dialog Box



#### Codeword length N

The codeword length. The input has vector length NC\*M\*(N - NP), where NC is the number of codewords being output, and NP is the number of punctures per codeword.

#### Message length K

The message length. The first output has vector length NM\*M\*K, where NM is the number of messages per frame being 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.

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.

#### Puncture code

Selecting this check box enables the field **Puncture vector**.

#### **Puncture vector**

This field is available only when **Puncture code** is selected.

A column vector of length N-K. A value of 1 in the **Puncture vector** corresponds to an M-bit symbol that is not punctured, and a 0 corresponds to an M-bit symbol that is punctured.

The default value is [ones(2,1); zeros(2,1)].

#### Enable erasures input port

Selecting this check box will open the port, Era.

Through the port, you can input a frame-based binary column vector that is 1/M times as long as the codeword input.

Erasure values of 1 correspond to erased symbols in the same position in the bit-packed codeword, and values of 0 correspond to nonerased symbols.

#### 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.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	• 1-bit unsigned integer (ufix(1))
Out	Double-precision floating point
	Single-precision floating point
	Boolean

Port	Supported Data Types
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	• 1-bit unsigned integer (ufix(1))
Era	Double-precision floating point
	Boolean
Err	Double-precision floating point

#### **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.

[3] Clark, George C., Jr., and J. Bibb Cain, *Error-Correction Coding for Digital Communications*, New York, Plenum Press, 1981.

#### **See Also**

Integer-Output RS Decoder

### **Binary Symmetric Channel**

**Purpose** 

Introduce binary errors

Library

Channels

#### **Description**



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

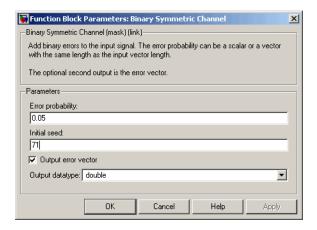
The input port is the transmitted binary signal. The input can be either a scalar, a sample-based vector, or a frame-based row vector. This block processes each vector element independently, and introduces an error in a given spot with probability **Error probability**.

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 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.

### **Binary Symmetric Channel**

#### Dialog Box



#### Error probability

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

#### 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.

#### Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	Fixed point (signed and unsigned)
	• Boolean

## **Binary Symmetric Channel**

Port	Supported Data Types
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
Output	Double-precision floating point
	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

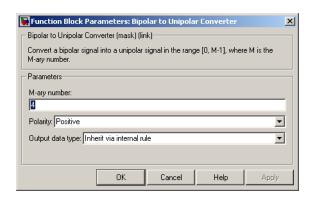
**Description** 

Bipolar to > Unipolar Converter 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,..., M-1}, where M is the **M-ary number** parameter, then the output consists of integers between 0 and M-1. This block is only designed to work when the input value is within the set {-M+1, -M+3, -M+5,..., M-1}, where M is the **M-ary number** parameter. If the input value is outside of this set of integers the output may not be valid.

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

Polarity Parameter Value	Output Corresponding to 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.

### **Bipolar to Unipolar Converter**

#### **Polarity**

A value of Positive causes the block to maintain the relative ordering of symbols in the alphabets. A value of Negative causes the block to reverse the relative ordering of symbols in the alphabets.

#### **Output Data Type**

The type of bipolar signal produced at the block's output.

The block supports the following output data types:

- Inherit via internal rule
- Same as input
- double
- int8
- uint8
- int16
- uint16
- int32
- uint32
- boolean

When the parameter is set to its default setting, Inherit via internal rule, the block determines the output data type based on the input data type.

- If the input signal is floating-point (either single or double), the output data type is the same as the input data type.
- If the input data type is not floating-point:
  - Based on the M-ary number parameter, the output data type is the ideal unsigned integer output word length required to contain the range [0 M-1] and is computed as follows:

### **Bipolar to Unipolar Converter**

ideal word length = ceil(log2(M))

- The block sets the output data type to be an unsigned integer, based on the smallest word length (in bits) that can fit best the computed ideal word length.

**Note** The selections in the **Hardware Implementation** pane pertaining to word length constraints do not affect how this block determines output data types.

#### **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].

If the value for the **M-ary number** is  $2^8$  the block gives an output of uint8.

If the value for the **M-ary number** is 2<sup>8</sup>+1 the block gives an output of uint16.

### **Pair Block**

Unipolar to Bipolar Converter

**Purpose** 

Map vector of bits to corresponding vector of integers

Library

**Utility Blocks** 

#### **Description**

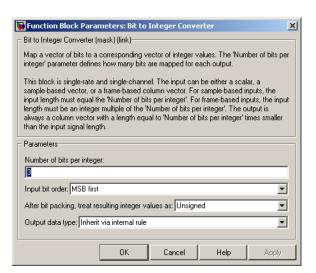
Bit to Integer Converter The Bit to Integer Converter block maps groups of bits in the input vector to integers in the output vector.

For unsigned integers, if M is the **Number of bits per integer**, then the block maps each group of M bits to an integer between 0 and  $2^{M}$ -1. As a result, the output vector length is 1/M times the input vector length. For signed integers, if M is the **Number of bits per integer**, then the block maps each group of M bits to an integer between  $-2^{M-1}$  and  $2^{M-1}$ -1.

If the input signal is a sample-based input, then it must be a vector whose length equals the value for 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 can accept input data types double, single, boolean, int8, uint8, int16, uint16, int32, uint32 and ufix1.

#### 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 32.

#### Input bit order

Defines whether the first bit of the input signal is the most significant bit (MSB) or the least significant bit (LSB). The default selection is MSB.

#### After bit packing, treat resulting integer value as

Indicates if the integer value input ranges should be treated as signed or unsigned. The default setting is Unsigned.

**Note** This parameter setting determines which **Output data type** selections are available.

#### Output data type

If the input values are unsigned integers, you can choose from the following **Output data type** options:

- Inherit via internal rule
- Smallest integer
- Same as input
- double
- single
- int8
- uint8
- int16
- uint16
- int32
- uint32

If the input values are signed integers, you can choose from the following **Output data type** options:

- Inherit via internal rule
- Smallest integer
- double
- single
- int8
- int16
- int32

The default selection for this parameter is Inherit via internal rule.

When the parameter is set to Inherit via internal rule, the block determines the output data type based on the input data type.

- If the input signal is floating-point (either double or single), the output data type is the same as the input data type.
- If the input data type is not floating-point, the output data type is determined as if the parameter is set to Smallest integer.

When the parameter is set to Smallest integer, the software selects the output data type based on the settings used in the **Hardware Implementation** pane of the Configuration Parameters dialog box.

- If ASIC/FPGA is selected, the output data type is the smallest ideal integer or fixed-point data type, based on the setting for the Number of bits per integer parameter.
- For all other selections, the output data type is the smallest available (signed or unsigned) integer word length that is large enough to fit the ideal minimum bit size.

**Examples** Refer to the example on the Integer to Bit Converter reference page:

Fixed-Point Integer To Bit and Bit To Integer Conversion (Audio

Scrambling and Descrambling Example)

See Also

bi2de, bin2dec

Pair Block Integer to Bit Converter

### **BPSK Demodulator Baseband**

**Purpose** 

Demodulate BPSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

#### **Description**

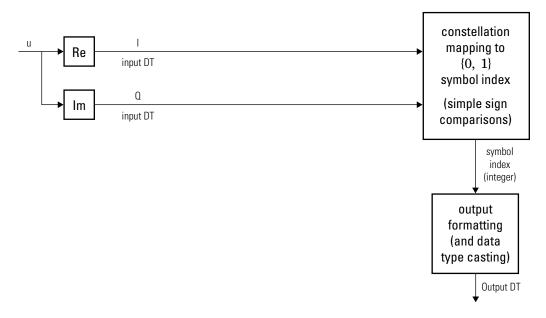


The BPSK Demodulator Baseband block demodulates a signal that was modulated using the binary phase shift keying method. The input is a baseband representation of the modulated signal. The input can be either a scalar or a frame-based column vector. The input must be a discrete-time complex signal. The block maps the points  $\exp(j\theta)$  and  $-\exp(j\theta)$  to 0 and 1, respectively, where  $\theta$  is the **Phase offset** parameter.

For information about the data types each block port supports, see the "Supported Data Types" on page 2-87 table on this page.

### **BPSK Demodulator Baseband**

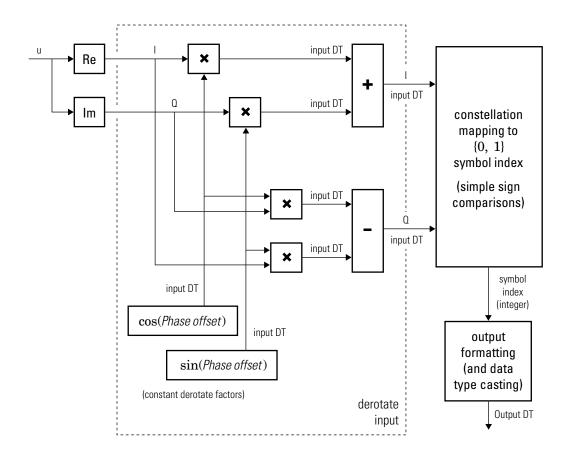
#### **Algorithm**



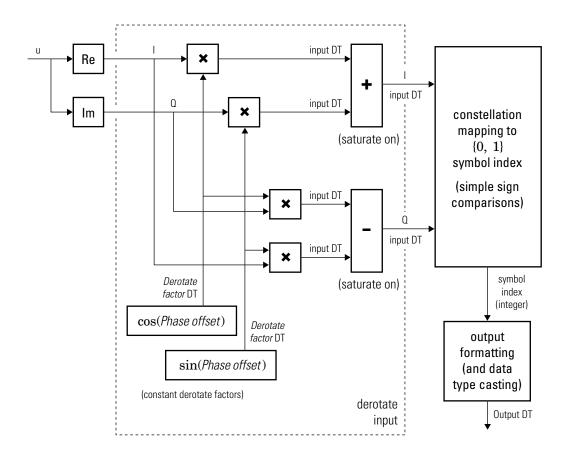
Hard-Decision BPSK Demodulator Signal Diagram for Trivial Phase

Offset (multiple of 
$$\frac{\pi}{2}$$
)

### **BPSK Demodulator Baseband**



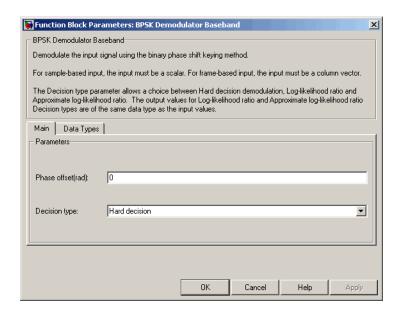
Hard-Decision BPSK Demodulator Floating-Point Signal Diagram for Nontrivial Phase Offset



#### Hard-Decision BPSK Demodulator Fixed-Point Signal Diagram for Nontrivial Phase Offset

The exact LLR and approximate LLR cases (soft-decision) are described in "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the *Communications ToolboxUser's Guide*.

### 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" in the *Communications ToolboxUser's Guide* 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

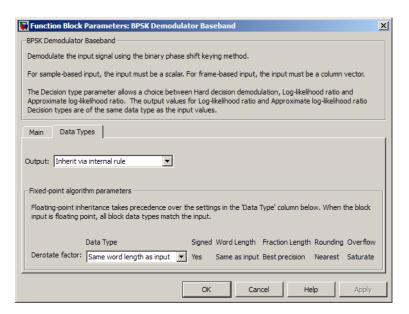
This parameter appears when the **Noise variance source** is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:

- Inf to -Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.



#### **Data Types Pane for Hard-Decision**

#### Output

When **Decision type** is set to Hard decision, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, uint32, or boolean.

When this parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is a floating-point type (single or double). If the input data type is fixed-point, the output data type will work as if this parameter is set to 'Smallest unsigned integer'.

When this parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in

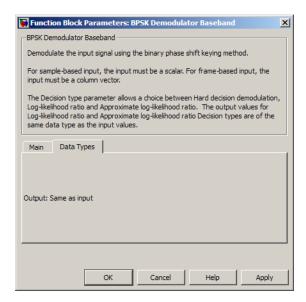
the **Hardware Implementation** pane, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

#### **Derotate factor**

This parameter only applies when the input is fixed-point and

**Phase offset** is not a multiple of  $\frac{\pi}{2}$ .

This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input.



#### **Data Types Pane for Soft-Decision**

When **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is inherited from the input (e.g., if the input is of data type double, the output is also of data type double).

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	Signed Fixed point (only for Hard decision mode)
Var	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	• ufix(1) in ASIC/FPGA and when <b>Decision type</b> is Hard decision modes

**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** 



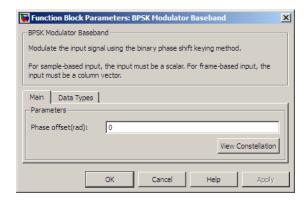
The BPSK Modulator Baseband block modulates using the binary phase shift keying method. The output is a baseband representation of the modulated signal. For information about the data types each block port supports, see the "Supported Data Types" on page 2-90 table on this page.

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.

#### **Constellation Visualization**

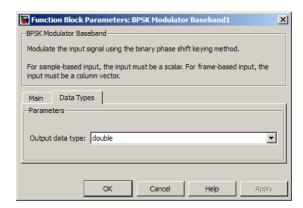
The BPSK Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the *Communications Blockset User's Guide*.

## 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.

### Output word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed-point for the **Output data type** parameter.

#### Set output fraction length to

Specify the scaling of the fixed-point output by either of the following 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.

#### 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 software. This parameter is only visible when you select User-defined for the **Output data type** parameter.

#### Output fraction length

For fixed-point output data types, specify the number of fractional bits, or bits to the right of the binary point. This parameter is only visible when you select Fixed-point or User-defined for the **Output data type** parameter and User-defined for the **Set output fraction length to** parameter.

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
Output	Double-precision floating point
	Single-precision floating point
	Fixed point (signed only)

### **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**

Charge Filt > Pump PD > PLL VCO > 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 software. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.

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

• A voltage-controlled oscillator (VCO). You specify characteristics of the VCO using the VCO input sensitivity, VCO quiescent frequency, VCO initial phase, and VCO output amplitude parameters.

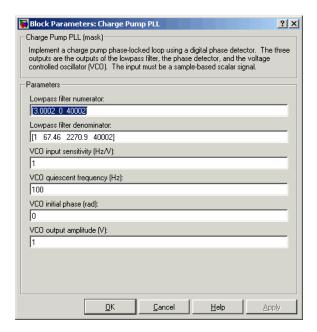
The input signal represents the received signal. The input must be a sample-based scalar signal. The three output ports produce:

- 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**

### **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.

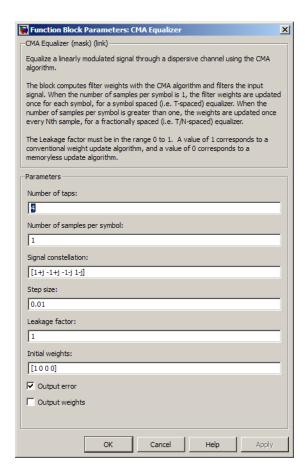
Port	Meaning	How to Enable
Err output	y(R -   y   <sup>2</sup> ), where y is the equalized signal and R is a constant related to the signal constellation	Check the Output error check box.
Wts output	A vector listing the weights after the block has processed either the current input frame or, in	Check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

Port	Meaning	How to Enable
	sample-based mode, the current input sample.	

# **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.

## 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**

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

**Library** Utility Blocks

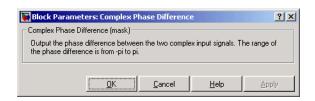
**Description** The Complex Pha



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

# **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

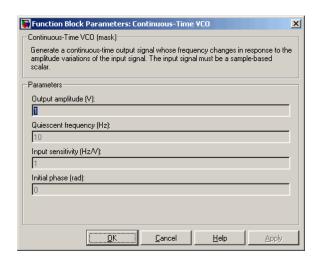
$$y(t) = A_c \cos \left( 2\pi f_c t + 2\pi k_c \int_0^t u(\tau) d\tau + \varphi \right)$$

where  $A_{\rm c}$  is the **Output amplitude** parameter,  $f_{\rm c}$  is the **Quiescent** frequency parameter,  $k_{\rm c}$  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

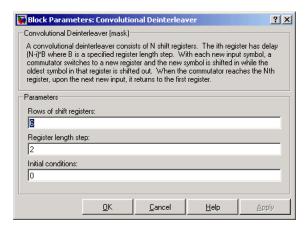
### **Description**

Convolutional Deinterleaver 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.

# **Convolutional Deinterleaver**

**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** 

Convolutional Encoder 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  $L^*k$  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  $L^*n$ .

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.

**Note** This block works for cases  $k \ge 1$ , where it has the same values for constraint lengths in each input stream (e.g., constraint lengths of [2 2] or [7 7] will work, but [5 4] will not).

### **Specifying the Encoder**

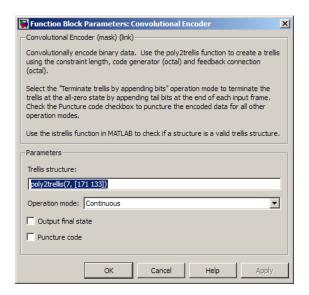
To define the convolutional encoder, use the **Trellis structure** parameter. This parameter is a MATLAB structure whose format is described in "Trellis Description of a Convolutional Encoder" in the Communications Toolbox documentation. You can use this parameter field in two ways:

# **Convolutional Encoder**

- 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.

### 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 state at the start of each frame.

In Terminate trellis by appending bits mode, the block treats each frame independently. For each input frame, extra bits are used to set the encoder states to all-zeros state at the end of the frame. The output length is given by

 $y = n \cdot (x+s)/k$ , where x is the number of input bits, and s = constraint length - 1 (or, in the case of multiple constraint

lengths, s = sum(ConstraintLength(i)-1)). The block supports this mode for 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.

#### Output final state

When you select **Output final state**, the second output port signal specifies the output state for the block. The output signal is a scalar, integer value. You can select **Output final state** for all operation modes except Terminate trellis by appending bits .

#### Specify initial state via input port

When you select **Specify initial state via input port** the second input port signal specifies the starting state for every frame in the block. The input signal must be a scalar, integer value. **Specify initial state via input port** appears only in Truncated operation mode.

#### 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 you select **Punctured code**.

# Puncture Pattern Examples

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

### **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.

# **Convolutional Encoder**

- [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**

**Purpose** 

Permute input symbols using set of shift registers

Library

Convolutional sublibrary of Interleaving

### **Description**

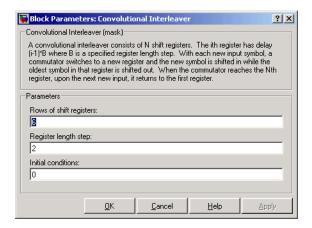
Convolutional Interleaver The Convolutional Interleaver block permutes the symbols in the input signal. Internally, it uses a set of shift registers. The delay value of the kth shift register is (k-1) times the **Register length step** parameter. The number of shift registers is the value of the **Rows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

The **Initial conditions** parameter indicates the values that fill each shift register at the beginning of the simulation (except for the first shift register, which has zero delay). If **Initial conditions** is a scalar, then its value fills all shift registers except the first; if **Initial conditions** is a column vector whose length is the **Rows of shift registers** parameter, then each entry fills the corresponding shift register. The value of the first element of the **Initial conditions** parameter is unimportant, since the first shift register has zero delay.

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**

[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, and the model uses a variable-step solver or a fixed-step solver with the **Tasking Mode** parameter set to SingleTasking, then the delay consists of D+1 zero symbols.
- If the input signal is frame-based, then the delay consists of D zero symbols.

The optimal **Traceback length** parameter value is dependent on minimum squared Euclidean distance calculations. Alternatively, a typical value, dependent on the number of states, can be chosen using the "five-times-the-constraint-length" rule, which corresponds

to  $5 \cdot \log 2(numStates)$ . The number of states is determined by the following equation:

$$numStates = egin{cases} p \cdot M^{(L-1)}, \textit{for even } m \ 2p \cdot M^{(L-1)}, \textit{for odd } m \end{cases}$$
 where:

- h = m/p is the modulation index proper rational form
  - = m = numerator of modulation index
  - = p = denominator of modulation index
- L is the Pulse length
- *M* is M-ary number

For a binary raised cosine pulse shape with a pulse length of 3, h=2/3,

this rule  $(5*\log 2(3*2^2) = 18)$  gives a result close to the optimum value of 20.

#### **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*.

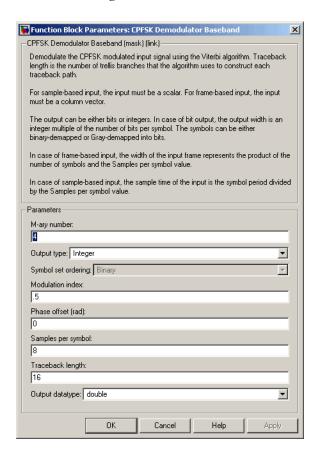
In binary output mode, the block first maps each input symbol to an intermediate value as in the integer output mode. The block then maps the odd integer k to the nonnegative integer (k+M-1)/2. Finally, the block maps each nonnegative integer to a binary word, using a mapping that depends on whether 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*.

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.

#### **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 CPFSK Demodulator Baseband block uses to construct each traceback path.

#### Output datatype

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

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers

### **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.

# **CPFSK Modulator Baseband**

**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*.

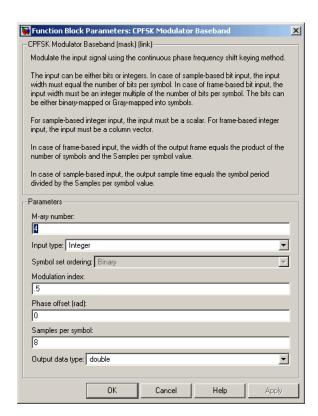
The input can be either a scalar or a frame-based column vector. If **Input type** is Bit, then the input can also be a vector of length K.

### **Upsampling the Modulated Signal**

This block can output an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor, which must be a positive integer. For non-binary schemes it must be greater than 1.

For more information, see "Upsampled Signals and Rate Changes" in Communications BlocksetUser's Guide.

### Dialog Box



#### M-ary number

The size of the alphabet.

## **CPFSK Modulator Baseband**

#### 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.

## Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Boolean (bit input mode only)
	• 8-, 16-, and 32-bit signed integers (integer input mode only)
Output	Double-precision floating point
	Single-precision floating point

### **Pair Block**

CPFSK Demodulator Baseband

## See Also

CPM Modulator Baseband, M-FSK Modulator Baseband

# **CPFSK Modulator Baseband**

### References

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

## **CPM Demodulator Baseband**

**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, and the model uses a variable-step solver or a fixed-step solver with the **Tasking Mode** parameter set to SingleTasking, then the delay consists of D+1 zero symbols.
- If the input signal is frame-based, then the delay consists of D zero symbols.

The optimal **Traceback length** parameter value is dependent on minimum squared Euclidean distance calculations. Alternatively, a typical value, dependent on the number of states, can be chosen using the "five-times-the-constraint-length" rule, which corresponds

to  $5 \cdot \log 2(numStates)$ . The number of states is determined by the following equation:

$$numStates = egin{cases} p \cdot M^{(L-1)}, \textit{for even } m \\ 2p \cdot M^{(L-1)}, \textit{for odd } m \end{cases}$$
 where:

- h = m/p is the modulation index proper rational form
  - $\mathbf{m} = \mathbf{m} = \mathbf{m}$
  - p = denominator of modulation index
- L is the Pulse length
- *M* is M-ary number

For a binary raised cosine pulse shape with a pulse length of 3, h=2/3,

this rule  $(5*\log 2(3*2^2) = 18)$  gives a result close to the optimum value of 20.

### **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*.

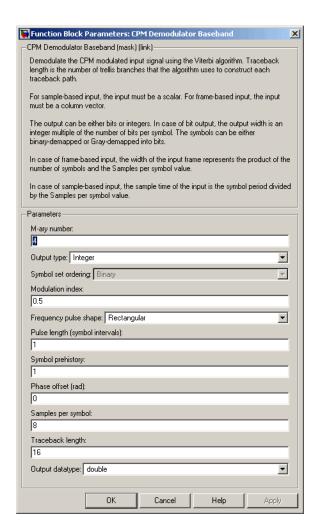
In binary output mode, the block first maps each input symbol to an intermediate value as in the integer output mode. The block then maps the odd integer k to the nonnegative integer (k+M-1)/2. Finally, the block maps each nonnegative integer to a binary word, using a mapping that depends on whether 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*.

# **CPM Demodulator Baseband**

## **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.

## **CPM Demodulator Baseband**

#### 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 CPM Demodulator block uses to construct each traceback path.

#### Output datatype

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

## Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers

Pair Block CPM Modulator Baseband

**See Also**CPFSK Demodulator Baseband, GMSK Demodulator Baseband, MSK Demodulator Baseband, Viterbi Decoder

Demodulator Daseband, vite

**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 correponding 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*.

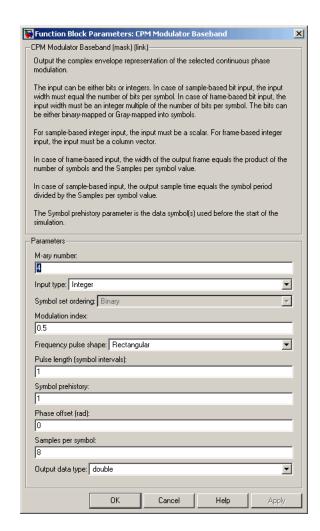
The input can be either a scalar or a frame-based column vector. If **Input type** is Bit, then the input can also be a vector of length K. For information about the data types each block port supports, see the "Supported Data Types" on page 2-131 table on this page.

## **Upsampling the Modulated Signal**

This block can output an upsampled version of the modulated signal. The **Samples per symbol** parameter is the upsampling factor, which must be a positive integer. For all non-binary schemes as defined by the pulse shapes, it must be greater than 1.

For more information, see "Upsampled Signals and Rate Changes" in Communications BlocksetUser's Guide.

## Dialog Box



#### 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.

#### 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.

## Supported Data Types

Port	Supported Data Types	
Input	Double-precision floating point	
	Boolean (bit input mode only)	
	• 8-, 16-, and 32-bit signed integers (integer input mode only)	
Output	Double-precision floating point	
	Single-precision floating point	

Pair Block CPM Demodulator Baseband

**See Also**CPFSK Modulator Baseband, GMSK Modulator Baseband, MSK Modulator Baseband

Modulator Baseban

**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:

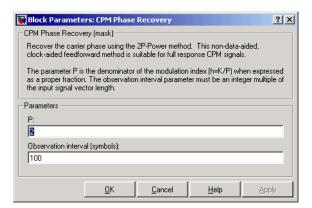
- The output port labeled Sig gives the result of rotating the input signal counterclockwise, where the amount of rotation equals the carrier phase estimate. The Sig output is thus a corrected version of the input signal, and has the same sample time and vector size as the input signal.
- The output port labeled Ph outputs the carrier phase estimate, in degrees, for all symbols in the observation interval. The Ph output is a scalar signal.

**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 <sup>16</sup> +x <sup>14</sup> +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

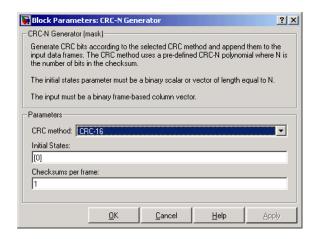
You specify the initial state of the internal shift register using the **Initial states** parameter. You specify the number of checksums that the block calculates for each input frame using the **Checksums per frame** parameter. For more detailed information, see the reference page for the General CRC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

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

### **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.

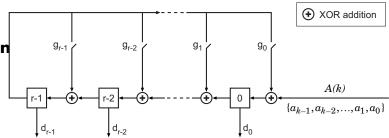
### Checksums per frame

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

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

$$g(x) = g_{r-1}x^{r-1} + g_{r-2}x^{r-2} + \dots + g_1x + g_0$$
, 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  $\{a_{k-1}, a_{k-2}, ..., a_2, a_1, a_0\}$  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

### **See Also**

General CRC Generator, General CRC Syndrome Detector

# **CRC-N Syndrome Detector**

### **Purpose**

Detect errors in input data frames according to selected CRC method

## Library

CRC sublibrary of Error Detection and Correction

## **Description**

CRC-N Syndrome Detector Err 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.

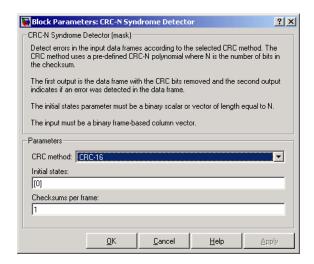
The CRC-N Syndrome Detector block is a simplified version of the General CRC Syndrome Detector block. You can select the generator polynomial for the CRC algorithm from a list of commonly used polynomials, given in the **CRC-N method** field in the block's dialog. N is the degree of the generator polynomial. 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.

### 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.

# **CRC-N Syndrome Detector**

[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

### **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 a scalar, a sample-based vector, or a frame-based signal. 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.

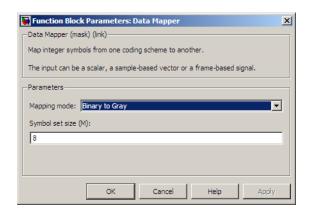
# **Data Mapper**

Binary to Gray Mode		Gray to Bir	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:

- $0 \rightarrow 1$
- $1 \rightarrow 5$
- $2 \rightarrow 0$
- $3 \rightarrow 4$
- $4 \rightarrow 2$
- $5 \rightarrow 3$

When you select the  $\mbox{\it Straight \ Through mode},$  the output equals the input.



### 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 appears if **Mapping mode** is set to User Defined.

## **DBPSK Demodulator Baseband**

**Purpose** 

Demodulate DBPSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

**Description** 

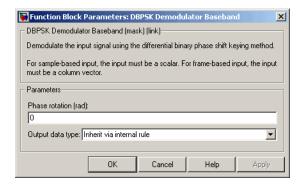


The DBPSK Demodulator Baseband block demodulates a signal that was modulated using the differential binary phase shift keying method. The input is a baseband representation of the modulated signal.

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**

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.

## Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

### **Pair Block**

DBPSK Modulator Baseband

### See Also

M-DPSK Demodulator Baseband, DQPSK Demodulator Baseband, BPSK Demodulator Baseband

## **DBPSK Modulator Baseband**

### **Purpose**

Modulate using differential binary phase shift keying method

## Library

PM, in Digital Baseband sublibrary of Modulation

### **Description**

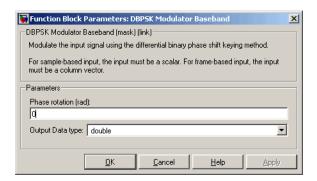


The DBPSK Modulator Baseband block modulates using the differential binary phase shift keying method. The output is a baseband representation of the modulated signal.

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.

## Supported Data Types

Port	Supported Data Types
Input • Double-precision floating point	
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
Output	Double-precision floating point
	Single-precision floating point

**Pair Block** 

DBPSK Demodulator Baseband

See Also

DQPSK Modulator Baseband, BPSK Modulator Baseband

## **Deinterlacer**

### **Purpose**

Distribute elements of input vector alternately between two output vectors

### Library

Sequence Operations

## **Description**

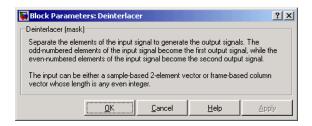


The Deinterlacer block accepts an input vector that has an even number of elements. The block alternately places the elements in each of two output vectors. As a result, each output vector size is half the input vector size. The output vectors have the same complexity and sample time of the input.

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

# See Also Demux (Simulink documentation)

## **Derepeat**

### **Purpose**

Reduce sampling rate by averaging consecutive samples

## Library

Sequence Operations

## **Description**

Derepeat /5

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 Signal Processing Blockset Repeat block. The positive integer N is the **Derepeat factor** parameter in the Derepeat dialog.

The **Initial condition** parameter prescribes elements of the output when it is still too early for the input data to show up in the output. If the dimensions of the **Initial condition** parameter match the output dimensions, then the parameter represents the initial output value. If **Initial condition** is a scalar, then it represents the initial value of each element in the output. The block does not support empty matrices for initial conditions.

The input can have any shape or frame status. The block 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, and **Framing** is set to Maintain input size, then the block assumes that the input is a vector or matrix whose elements represent samples from independent channels. The block averages samples from each channel independently over time. The output period is N times the input period, and the input and output sizes are identical. The output is delayed by one output period, and the first output value is the **Initial condition** value. If **Framing** is set to Maintain input rate, the block will generate an error message.

### **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.

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.

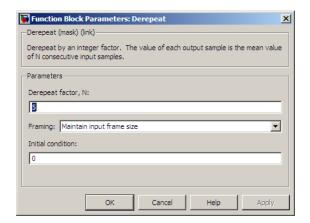
For example, if a single-channel input with a frame period of 1 second is derepeated by a factor of 4, then the output has a frame period of 4 seconds. The input and output frame sizes are equal.

#### Maintain input frame rate

The block reduces the sampling rate by using a proportionally smaller frame *size* than the input. **Derepeat factor** should be an integer factor of the number of rows in the input vector or matrix. For derepetition by a factor of N, the output frame size is 1/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.

#### **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.

#### **Initial condition**

The value with which to initialize the block.

### See Also

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

**Purpose** 

Descramble input signal

Library

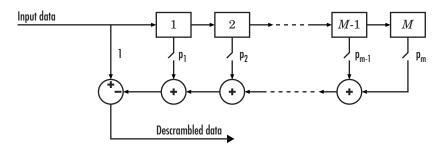
Sequence Operations

## **Description**

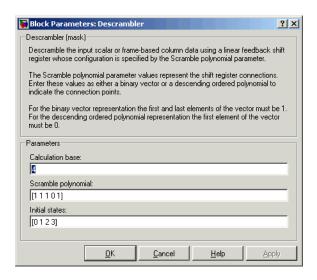


The Descrambler block descrambles the input signal, which must be a scalar or a frame-based column vector. The Descrambler block is the inverse of the Scrambler block. If you use the Scrambler block in the transmitter, then you should use the Descrambler block in the receiver.

Below is a schematic of the descrambler. All adders and the one subtracter 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

## **Purpose**

Decode binary signal using differential coding

# Library

Source Coding

# **Description**

Differential Decoder 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

 $m(t_0) = d(t_0)$  XOR **Initial condition** parameter value

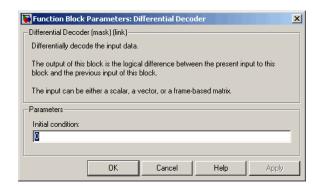
$$m(t_k) = d(t_k) XOR d(t_{k-1})$$

where

- d is the differentially encoded input.
- m is the output message.
- $t_k$  is the kth 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.

# **Differential Decoder**



# Dialog Box

#### Initial condition

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

# Supported Data Type

Port	Supported Data Types
In	• double
	• single
	• boolean
	• integer
	• fixed-point
Out	• double
	• single
	• boolean
	• integer
	• fixed-point

# References

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

# **Differential Decoder**

Pair Block Differential Encoder

# **Differential Encoder**

# **Purpose**

Encode binary signal using differential coding

# Library

Source Coding

# **Description**

Differential Encoder 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

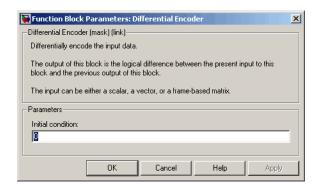
$$d(t_0) = m(t_0) \text{ XOR Initial condition}$$
 parameter value

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

where

- m is the input message
- d is the differentially encoded output.
- t<sub>k</sub> is the kth time step.
- XOR is the logical exclusive-or operator.

The input can be either a scalar or a vector.



# Dialog Box

#### Initial condition

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

# Supported Data Type

Port	Supported Data Types
In	• double
	• single
	• boolean
	• integer
	• fixed-point
Out	• double
	• single
	• boolean
	• integer
	• fixed-point

# References

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

# **Differential Encoder**

**Pair Block** 

Differential Decoder

**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 as double prior to display. In sample-based mode, the input signal must be a scalar value. In frame-based mode, the input signal must be a column vector or a scalar value.

#### Warning

When this block is set up in multi-rate mode, neither the Rapid Accelerator nor External mode allows for the uploading of data for display on this scope. To take full advantage of this scope block in multi-rate mode, run the model in Accelerator or Normal mode. Alternately, configure it to be in single-rate mode for use with Rapid Accelerator or External 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	0	0 0

Marker Style	Parameter Symbol	Appearance
Asterisk	*	* * *
Point		
Cross	х	× × ×

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

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.

Parameter	Recommended Setting
Samples per symbol	Same as the Samples per symbol setting in the modulator block, or the Interpolation factor setting in the interpolation block
Offset (samples)	0 to view the open part of the eye (Samples per symbol)/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.

Parameter	Recommended Setting
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 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 Open at start of simulation to view the signal at the start of simulation.
	Clear to view the signal after convergence to steady state and for greater initial speed.

Parameter	Recommended Setting
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

To use the scope in single-rate mode, the following conditions must be true:

$$sps * ( (td * (spt-1)) + ntpd) = Sf$$

where:

- sps = Samples per symbol
- td = Traces displayed
- spt = Symbols per trace
- ntpd = New traces per display
- Sf = Input frame size, in samples

## **Scope Options**

The scope title (in the window title bar) is the same as the block title. You can set the axis scaling by setting the y-axis minimum and y-axis maximum parameters on the **Axes Properties** panel.

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

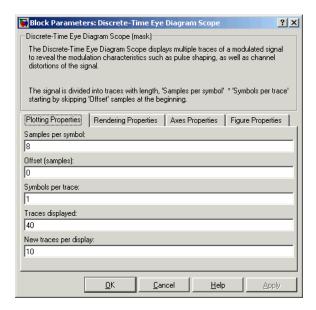
The properties listed in the **Axes** menu apply to all channels. Many of the parameters in this menu are also accessible through the block parameter dialog box. These are **Autoscale**, **Show grid**, **Frame** #, and **Save Position**. Below are descriptions of the other parameters listed in the **Axes**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

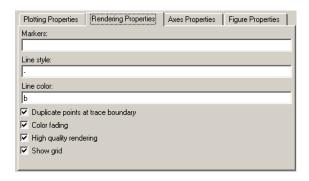
Positive integer specifying the number of symbols plotted per trace.

#### 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.



#### 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**

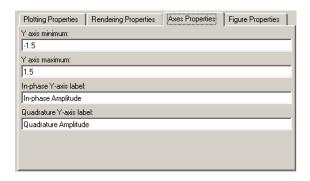
When selected, the points in the eye diagram fade as the interval of time after they are first plotted increases. Tunable.

#### 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.



#### Y-axis minimum

Minimum signal value the scope displays. Tunable.

#### Y-axis maximum

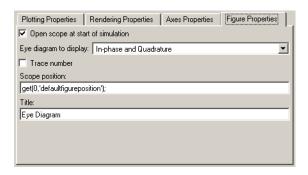
Maximum signal value the scope displays. Tunable.

#### 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.



### 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.

#### 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** 

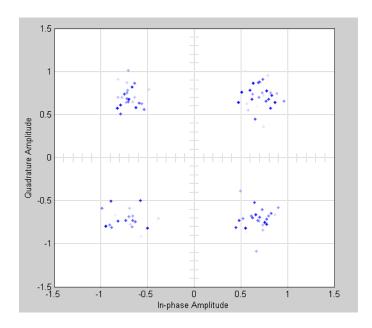


The Discrete-Time Scatter Plot Scope block displays scatter plots of a modulated signal, to reveal the modulation characteristics, such as pulse shaping or channel distortions of the signal.

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

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.



## Warning

When this block is set up in multi-rate mode, neither the Rapid Accelerator nor External mode allows for the uploading of data for display on this scope. To take full advantage of this scope block in multi-rate mode, run the model in Accelerator or Normal mode. Alternately, configure it to be in single-rate mode for use with Rapid Accelerator or External mode.

### **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.

Parameter	Recommended Setting
Samples per symbol	Same as the Samples per symbol setting in the modulator block, or the Interpolation factor setting in the interpolation block
Points displayed	10 times the alphabet size of the modulator
New points per display	Same as <b>Points 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.

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

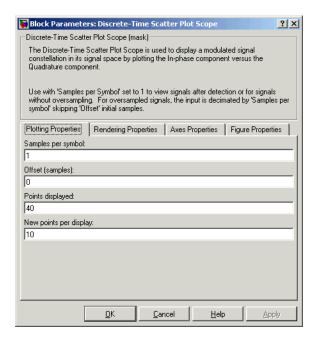
To use the scope in single-rate mode, the following conditions must be true:

$$sps * nppd = Sf$$

#### where:

- sps = Samples per symbol
- nppd = New points per display
- Sf = Input frame size, in samples

## Dialog Box



### Samples per symbol

Number of samples per symbol.

## Offset (samples)

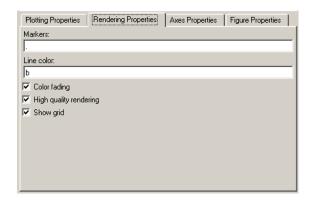
Nonnegative integer less than the number of samples per symbol, specifying the number of samples to skip before plotting points.

## 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

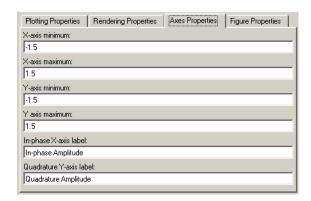
When selected, the points in the scatter plot fade as the interval of time after they are first plotted increases. Tunable.

## 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.



#### 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

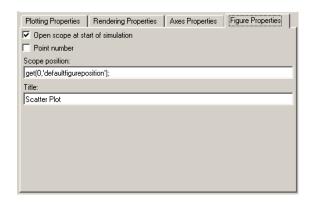
Maximum signal value the scope displays on the *y*-axis. Tunable.

#### 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 software illustrate how to use the Discrete-Time Scatter Plot Scope block:

- Digital Video Broadcasting Model Terrestrial
- HiperLAN/2

- Phase Noise Effects in 256 QAM
- Multipath Rayleigh Fading Channel

## See Also

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

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

component

**Library** Comm Sinks

**Description** T



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. In sample-based mode, the input signal must be a scalar value. In frame-based mode, the input signal must be a column vector or a scalar value.

#### Warning

When this block is set up in multi-rate mode, neither the Rapid Accelerator nor External mode allows for the uploading of data for display on this scope. To take full advantage of this scope block in multi-rate mode, run the model in Accelerator or Normal mode. Alternately, configure it to be in single-rate mode for use with Rapid Accelerator or External 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
Samples per symbol	Same as the Samples per symbol setting in the modulator block, or the Interpolation factor used in the interpolation block
Symbols displayed	10 times the alphabet size of the modulator, M
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.

Parameter	Recommended Setting
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

To use the scope in single-rate mode, the following conditions must be true:

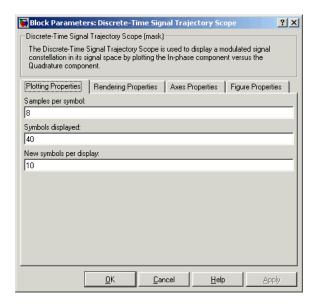
#### where:

$$sps * nspd = Sf$$

#### where:

- sps = Samples per symbol
- nspd = New symbols per display
- Sf = Input frame size, in samples

# Dialog Box



## Samples per symbol

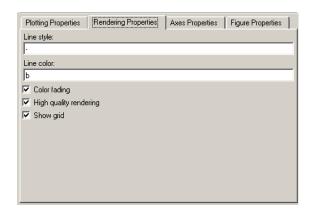
Number of samples per symbol.

## Symbols displayed

Total number of symbols plotted.

## New symbols per display

Number of new symbols that appear in each display.



#### Line markers

The line markers used in the signal trajectory. Tunable.

#### Line color

The line color used in the signal trajectory. Tunable.

### Color fading

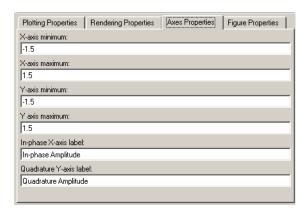
When selected, the points in the signal trajectory fade as the interval of time after they are first plotted increases. Tunable.

### 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.



#### 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

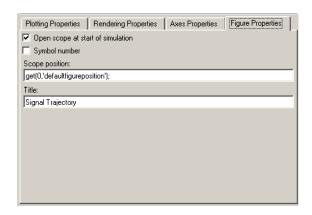
Maximum signal value the scope display on the *y*-axis. Tunable.

### In-phase X-axis label

Label for *x*-axis. Tunable.

## Quadrature Y-axis label

Label for y-axis. Tunable.



#### 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 software illustrate how to use the Discrete-Time Signal Trajectory Scope:

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

**See Also** 

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

# **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

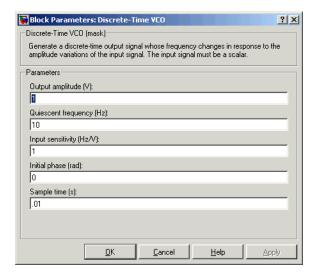
$$y(t) = A_c \cos \left( 2\pi f_c t + 2\pi k_c \int_0^t u(\tau) d\tau + \varphi \right)$$

where  $A_{\rm c}$  is the Output amplitude,  $f_{\rm c}$  is the Quiescent frequency,  $k_{\rm c}$  is the Input sensitivity, and  $\phi$  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. The block supports double precision only for code generation.

# 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** 

WWTLT DQPSK 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. For information about the data types each block port supports, see the "Supported Data Types" on page 2-192 table on this page.

## **Outputs and Constellation Types**

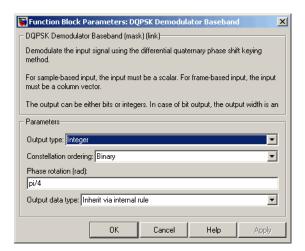
If the **Output type** parameter is set to Integer, then the block maps a phase difference of

 $\theta + \pi m/2$ 

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.

# **DQPSK Demodulator Baseband**

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Single-precision floating point
	Boolean when <b>Output type</b> isBit
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

Pair Block DQPSK Modulator Baseband

**See Also** M-DPSK Demodulator Baseband, DBPSK Demodulator Baseband,

QPSK Demodulator Baseband

# **DQPSK Modulator 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 information about the data types each block port supports, see the "Supported Data Types" on page 2-196 table on this page.

### **Inputs and Constellation Types**

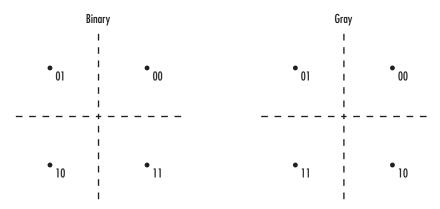
If the **Input type** parameter is set to **Integer**, then valid input values are 0, 1, 2, and 3. In this case, the input can be either a scalar or a frame-based column vector. If the first input is m, then the modulated symbol is

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

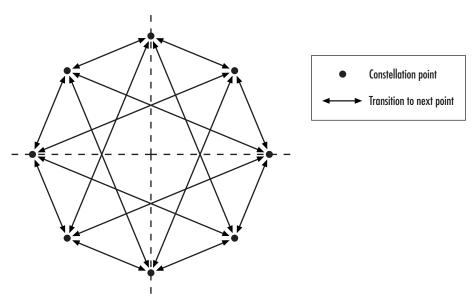
where  $\theta$  is the **Phase rotation** parameter. If a successive input is m, then the modulated symbol is the previous modulated symbol multiplied by  $\exp(j\theta + j\pi m/2)$ .

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 multiples 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**

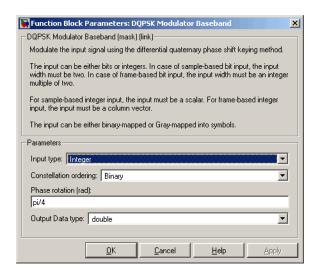


The figure below shows the signal constellation for the DQPSK modulation method when the **Phase rotation** parameter is  $\pi/4$ . The arrows indicate the four possible transitions from each symbol to the next symbol. The Binary and Gray options determine which transition is associated with each pair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

# **DQPSK Modulator Baseband**

# Supported Data Types

Port	Supported Data Types	
Input	Double-precision floating point	
	Single-precision floating point	
	Boolean when Input type is Bit	
	• 8-, 16-, and 32-bit signed integers	
	• 8-, 16-, and 32-bit unsigned integers	
Output	Double-precision floating point	
	Single-precision floating point	

**Pair Block** 

DQPSK Demodulator Baseband

See Also

 $\operatorname{M-DPSK}$  Modulator Baseband, DBPSK Modulator Baseband, QPSK Modulator Baseband

# **DSB AM Demodulator Passband**

**Purpose** 

Demodulate DSB-AM-modulated data

Library

Analog Passband Modulation, in Modulation

**Description** 



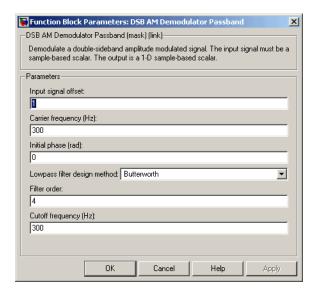
The DSB AM Demodulator Passband block demodulates a signal that was modulated using double-sideband amplitude modulation. The block uses the envelope detection method. The input is a passband representation of the modulated signal. Both the input and output signals are real sample-based scalar signals.

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.

# 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:

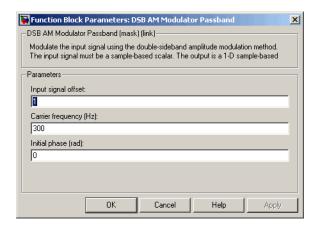
- *k* is the **Input signal offset** parameter.
- $f_c$  is the Carrier frequency parameter.
- $\theta$  is the **Initial phase** parameter.

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** 



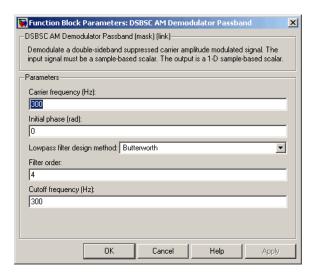
The DSBSC AM Demodulator Passband block demodulates a signal that was modulated using double-sideband suppressed-carrier amplitude modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real sample-based scalar signals.

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  ${\bf Lowpass}$  filter  ${\bf design}$   ${\bf method}$  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

# **DSBSC AM Modulator 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

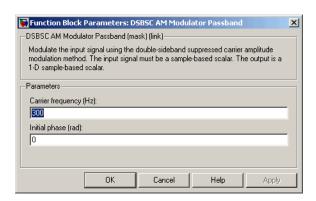
$$u(t)\cos(2\pi f_c t + \theta)$$

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**

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.

Early-Late Gate<sup>Sym</sup> > Timing Recovery Ph >

#### 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\*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**

- If the input signal is a frame-based column vector of length N\*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.

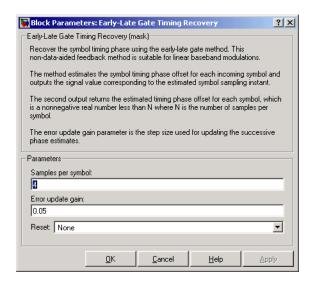
The Ph output contains nonnegative real numbers less than N. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time or frame period of the Ph output is the same as that of the Symoutput.

**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.

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink *User's Guide*.

#### Reset

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are None, Every

# **Early-Late Gate Timing Recovery**

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 kth symbol is e(k), given by

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

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
- $\bullet$  d<sub>k</sub> is the phase estimate for the kth symbol

For more information about the role that e(k) plays in this block's algorithm, see "Feedback Methods for Timing Phase Recovery" in *Communications 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**

T× Error Rate

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 Communications Blockset Repeat block using the Maintain input frame rate option.)

# **Error Rate Calculation**

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 Sel signal with zeros. The Error Rate Calculation block ignores zeros in the Sel 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.

If you plan to use this block along with the Real-Time Workshop® software, then you should not use the Workspace option; instead, use

# **Error Rate Calculation**

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

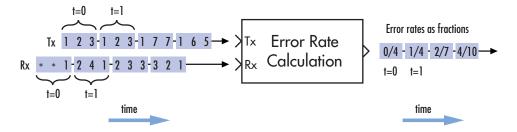
The input signals are both frame-based column vectors of length three. However, the schematic arranges each column vector horizontally and aligns pairs of vectors so as to reflect a receive delay of two samples. At

# **Error Rate Calculation**

each time step, the block compares elements of the Rx signal with those of the Tx signal that appear directly above them in the schematic. For instance, at time 1, the block compares 2, 4, and 1 from the Rx signal with 2, 3, and 1 from the Tx signal.

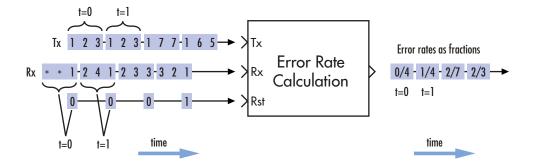
The values of the first two elements of Rx appear as asterisks because they do not influence the output. Similarly, the 6 and 5 in the Tx signal do not influence the output up to time 3, though they *would* influence the output at time 4.

In the error rates on the right side of the figure, each numerator at time t reflects the number of errors when considering the elements of Rx up through time t.



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

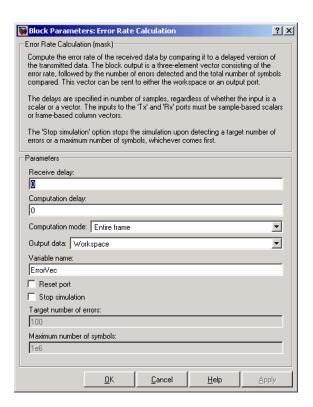
If the block's **Reset port** box had been checked and a reset had occurred at time = 3 seconds, then the last error rate would have been 2/3 instead of 4/10. This value 2/3 would reflect the comparison of 3, 2, and 1 from the Rx signal with 7, 7, and 1 from the Tx signal. The figure below illustrates this scenario.



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

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

If you use the Real-Time Workshop rapid simulation (RSim) target to build an RSim executable, then you can tune the **Target number of errors** and **Maximum number of symbols** parameters without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.



# 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

# **EVM Measurement**

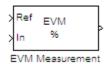
# **Purpose**

Calculate vector magnitude difference between ideal reference signal and measured signal

# Library

Utility Blocks

# **Description**



Error Vector Magnitude (EVM) is a measurement of modulator or demodulator performance in an impaired signal.

While certain mask selections can change EVM block behavior, the block always has two input signals: a reference signal (at the reference port, Ref) and a corrupted signal (at the input port, In). You must select which normalization method the block uses when performing EVM calculations and which calculations you want as outputs.

The block either normalizes to the average reference signal power, average constellation power, or peak constellation power. For RMS EVM, Max EVM, and X-percentile EVM, the output computations reflect the normalization method.

The default EVM output is RMS EVM in percent, with an option of Output maximum EVM or Output X-percentile EVM values. The maximum EVM represents the worst-case EVM value per burst. For the X-percentile option, you can select to output the number of symbols processed in the percentile computations.

The following table shows the output type, the activation (what selects the output computation), computation units, and the corresponding computation duration.

Output	Activation	Units	Computation Duration
RMS EVM	Default	Percentage	Per burst
Max EVM	Parameter setting	Percentage	Per burst

Output	Activation	Units	Computation Duration
Percentile EVM	Parameter setting	Percentage	Continuous
Number of symbols	Parameter setting if you select Output X-percentile EVM	None	Continuous

The computation duration in per burst mode spans the symbols in the present burst. The computation duration in continuous mode spans all the symbols across multiple bursts.

#### **Dimension**

The block computes measurements for bursts of data. The data must be of length N, where N is the size of the burst (sample-based or frame-based). When computing RMS EVM or Max EVM, the block computes a unique output for each incoming burst; therefore, the computation duration is per burst.

The block computes the X-percentile for all incoming symbols across multiple bursts. This computation duration is the continuous mode (in contrast to the per burst duration for RMS EVM or Max EVM).

### **Input Signals**

The input signals must be 1-D or 2-D sample-based column vectors or 2-D frame-based column vectors. The input and reference signals must have identical dimensions.

### **Output Signals**

The output is always a scalar value.

# **Data Type**

The block accepts double, single, and fixed-point data types. The output of the block is always double type.

# **Algorithms**

The EVM block provides three different normalization methods. You can normalize measurements according to the average power of the reference signal, average constellation power, or peak constellation power. Different industry standards follow one of these normalization methods.

The following table lists how the block calculates the RMS EVM value for different normalization methods.

EVM Normalization Method	Algorithm
Reference Signal	$EVM_{RMS} = \sqrt{rac{rac{1}{N}\sum\limits_{k=1}^{N}(e_k)}{rac{1}{N}\sum\limits_{k=1}^{N}(I_k^2 + Q_k^2)}}*100$
Average Power	$EVM_{RMS} = \sqrt{\frac{\frac{1}{N}\sum_{k=1}^{N}(e_k)}{P_{avg}}} *100$
Peak Power	$EVM_{RMS} = \sqrt{\frac{\frac{1}{N}\sum_{k=1}^{N}(e_k)}{P_{\text{max}}}} *100$

where,

$$e_k = (I_k - \tilde{I_k})^2 + (Q_k - \tilde{Q}_k)^2$$

 $I_{k} = \text{In-phase measurement of the } kth \text{ symbol in the burst}$ 

 $Q_k$  = Quadrature phase measurement of the kth symbol in the burst

N =Input vector length

 $P_{ans}$  = The value for **Average constellation power** 

 $P_{max}$  = The value for **Peak constellation power** 

 $I_k$  and  $Q_k$  represent ideal (reference) values.  $I_k$  and  $Q_k$  represent measured (received) symbols.

The max EVM is the maximum EVM value in a frame or

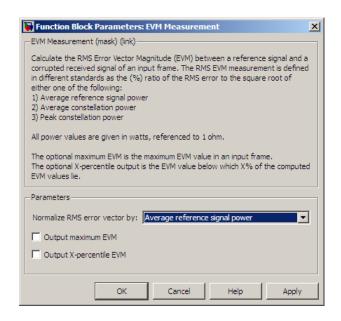
$$\begin{split} EVM_{\max} &= \max_{k \in [1,\dots,N]} \{EVM_k\} \\ \text{where } k \text{ is the } k\text{th symbol in a burst of length } N. \end{split}$$

The definition for EVM<sub>k</sub> varies depending upon which normalization method you select for computing measurements. The block supports the algorithms in the following table.

EVM Normalization	Algorithm
Reference Signal	$EVM_{k} = \sqrt{\frac{1}{N} \sum_{k=1}^{N} (I_{k}^{2} + Q_{k}^{2})} *100$

EVM Normalization	Algorithm
Average Power	$EVM_k = \sqrt{rac{e_k}{P_{avg}}}*100$
Peak Power	$EVM_k = \sqrt{\frac{e_k}{P_{\max}}} *100$

The block computes X-percentile EVM by creating a histogram of all the incoming  $EVM_k$  values. The output provides the EVM value below which X% of the EVM values lay.



# Dialog Box

#### Normalize RMS error vector by:

Selects the method by which the block normalizes measurements:

- Average reference signal power
- Average constellation power
- Peak constellation power

This parameter defaults to Average reference signal power.

#### Average constellation power:

Normalizes EVM measurement by the average constellation power. This parameter only appears if you set **Normalize RMS error vector** to Average constellation power.

#### Peak constellation power:

Normalizes EVM measurement by the peak constellation power. This parameter only appears if you set **Normalize RMS error vector** to Peak constellation power.

#### Output maximum EVM

Outputs the maximum EVM of an input vector or frame.

#### Output X-percentile EVM

Enables an output X-percentile EVM measurement. When you select this option, specify **X-percentile value** (%).

#### X-percentile value (%)

This parameter only appears when you select **Output X-percentile EVM**. The Xth percentile is the EVM value below which X% of all the computed EVM values lie. The parameter defaults to the 95th percentile. Therefore, 95% of all EVM values are below this output.

#### Output the number of symbols processed

Output X-percentile EVM. This parameter only appears when you select Output X-percentile EVM.

### **Examples**

To see an example using the EVM block, refer to Measuring Modulator Accuracy in the Communications Blockset User's Guide.

### References

- [1] IEEE Standard 802.16-2004: "Part 16: Air interface for fixed broadband wireless access systems," October 2004. http://ieee802.org/16/published.html
- [2] 3 GPP TS 45.005 V8.1.0 (2008-05): "Radio Access Network: Radio transmission and reception"
- [3] IEEE Standard 802.11a-1999: "Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications: High-speed Physical Layer in the 5 GHz Band," 1999.

# **See Also**

commmeasure.EVM, MER Measurement

**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 sDel 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 delay output port outputs signals of type double, and the chg output port outputs signals of type boolean.

The block's **Correlation window length** parameter specifies how many samples of the signals the block uses to calculate the cross-correlation. The delay output is a nonnegative integer less than the **Correlation window length**.

As the **Correlation window length** is increased, the reliability of the computed delay also increases. However, the processing time to compute the delay increases as well.

You can make the Find Delay block stop updating the delay after it computes the same delay value for a specified number of samples. To do so, select 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.

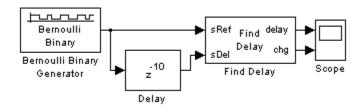
Another example of this usage is in "Computing Delays".

# Finding the Delay to Help Align Words

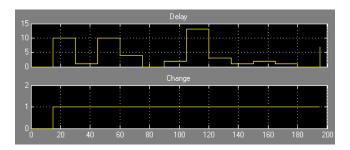
Another typical use of this block is to determine how to align the boundaries of frames with the boundaries of codewords or other types of data blocks. "Manipulating Delays" describes when such alignment is necessary and also illustrates, in the "Aligning Words of a Block Code" discussion, how to use the Find Delay block to solve the problem.

### **Setting the Correlation Window Length**

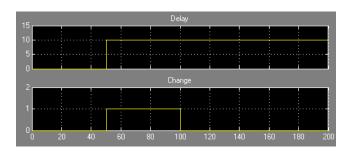
The next example illustrates how to tell when the **Correlation** window length is not sufficiently large.



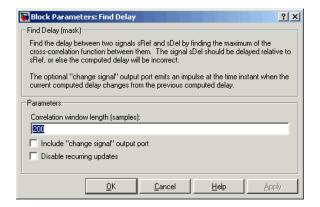
The model uses a Delay block to delay a signal by 10 samples, and uses the Find Delay block to compare the original signal with the delayed version. The model then displays the output of the Find Delay block in a scope. If the **Correlation window length** is 15, the scope shows that the calculated delay 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.

#### 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

# FM Demodulator Passband

Purpose Demodulate FM-modulated data

**Library** Analog Passband Modulation, in Modulation

**Description** 

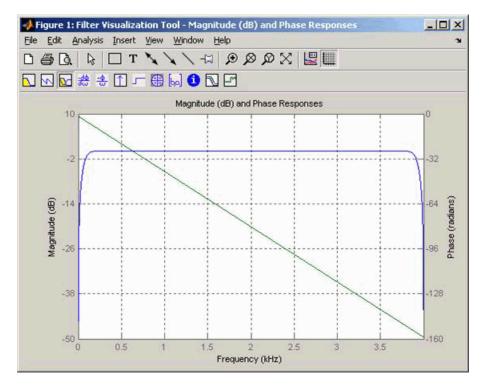


The FM Demodulator Passband block demodulates a signal that was modulated using frequency modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real sample-based scalar signals.

For best results, use a carrier frequency which is estimated to be larger than 10% of the reciprocal 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**

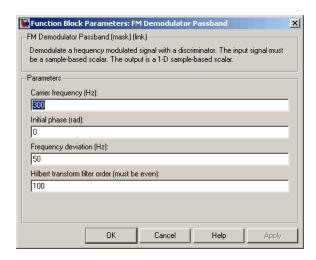


Note the bandwidth of the filter's magnitude response. By choosing a carrier frequency larger than 10% (but less than 90%) of the reciprocal of your input signal's sample time (8000 samples per second, in this example) or equivalently, a carrier frequency larger than 400Hz, we ensure that the Hilbert 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

**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:

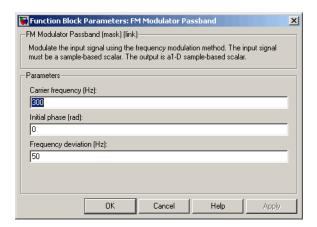
- $f_c$  is the Carrier frequency parameter.
- $\theta$  is the **Initial phase** parameter.
- ullet  $K_{\rm c}$  is the Modulation constant parameter.

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

### **Purpose**

Reduce amplitude of input signal by amount specified

# Library

RF Impairments

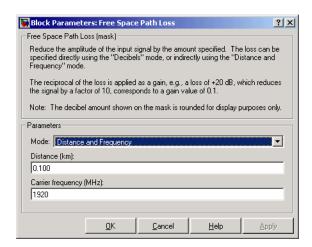
# **Description**

Free Space > Path Loss 10 dB 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.

# Free Space Path Loss

#### 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

### **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\*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:

# **Gardner Timing Recovery**

- If the input signal is a frame-based column vector of length N\*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.

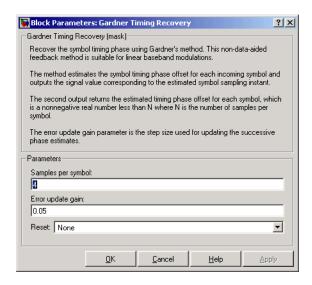
The Ph output contains nonnegative real numbers less than N. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time or frame period of the Ph output is the same as that of the Symoutput.

**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.

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink *User's Guide*.

#### Reset

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are None, Every

# **Gardner Timing Recovery**

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 kth symbol is e(k), given by

$$\begin{split} e(k) &= a_I(k) + a_Q(k) \\ a_I(k) &= \left\{ y_I((k-1)T + d_{k-1}) - y_I(kT + d_k) \right\} y_I(kT - T/2 + d_{k-1}) \\ a_Q(k) &= \left\{ y_Q((k-1)T + d_{k-1}) - y_Q(kT + d_k) \right\} y_Q(kT - T/2 + d_{k-1}) \end{split}$$

where

- $y_I$  and  $y_Q$  are the in-phase and quadrature components, respectively, of the block's input signal
- T is the symbol period
- $d_k$  is the phase estimate for the kth symbol

Notice from the expressions in curly braces above that the timing error detector approximates the derivative of y using finite differences.

For more information about the role that e(k) plays in this block's algorithm, see "Feedback Methods for Timing Phase Recovery" in *Communications Blockset User's Guide*.

# **Examples**

The commgardnerphrecov 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.

# **Gardner Timing Recovery**

[4] Oerder, M., "Derivation of Gardner's Timing-Error Detector from the ML principle", *IEEE Transactions on Communications*, Vol. COM-35, No. 6, June 1987, pp. 684-685.

# **See Also**

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

# **Gaussian Filter**

# **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 Signal Processing Toolbox gaussfir function 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 \* **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.

- Filter energy means that the sum of the squares of the coefficients equals 1.
- Peak amplitude means that the maximum coefficient equals 1.

After the block normalizes the set of filter coefficients as above, it multiplies all coefficients by the **Linear amplitude filter gain** parameter. As a result, the output is scaled by  $\sqrt{N}$ . If the output of this block feeds the input to the AWGN Channel block, specify the AWGN signal power parameter to be 1/N.

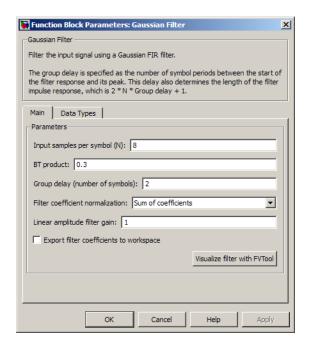
### **Input and Output Signals**

This block filters each channel of the input signal independently over time. The output frame status and dimensions are always the same as those of the input signal. When inputs are frame based, the block treats each column as an independent channel; the block filters each column. When inputs are sample based, the block treats each element of the input as an individual channel. 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.

#### 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.

#### 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

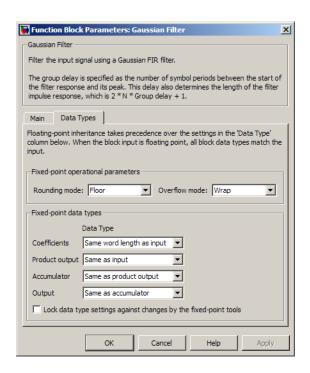
If you check this box, then the block creates a variable in the MATLAB workspace that contains the filter coefficients.

#### 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. The block uses the **Rounding** selection if a number cannot be represented exactly by the specified data type and scaling, it is rounded to a representable number. For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink® Fixed Point<sup>TM</sup> documentation.

#### Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

#### Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator). See "Filter Structure Diagrams" in the *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 the *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.

#### Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	• Single-precision floating point
	Signed fixed-point
Out	Double-precision floating point
	• Single-precision floating point
	Signed fixed-point

# **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)

# **Gaussian Noise Generator**

### **Purpose**

Generate Gaussian distributed noise with given mean and variance values

# Library

Noise Generators sublibrary of Comm Sources

# **Description**

Gaussian

The Gaussian Noise Generator block generates discrete-time white Gaussian noise. You must specify the **Initial seed** vector in the simulation.

The **Mean Value** and the **Variance** can be either scalars or vectors. If either of these is a scalar, then the block applies the same value to each element of a sample-based output or each column of a frame-based output. Individual elements or columns, respectively, are uncorrelated with each other.

When the **Variance** is a vector, its length must be the same as that of the **Initial seed** vector. In this case, the covariance matrix is a diagonal matrix whose diagonal elements come from the **Variance** vector. Since the off-diagonal elements are zero, the output Gaussian random variables are uncorrelated.

When the **Variance** is a square matrix, it represents the covariance matrix. Its off-diagonal elements are the correlations between pairs of output Gaussian random variables. In this case, the **Variance** matrix must be positive definite, and it must be N-by-N, where N is the length of the **Initial seed**.

The probability density function of *n*-dimensional Gaussian noise is

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

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**

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

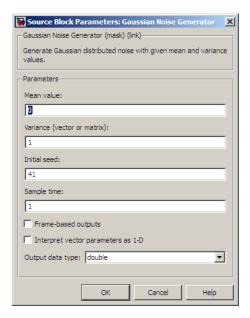
This returns a random prime number greater than 30. Entering randseed again produces a different prime number. If you supply an integer argument, randseed always returns the same prime for that integer. For example, randseed (5) always returns the same answer.

### **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 the *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**

#### 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 documentation), AWGN Channel, rand (built-in MATLAB function), randseed

# General Block Deinterleaver

### **Purpose**

Restore ordering of symbols in input vector

# Library

Block sublibrary of Interleaving

# **Description**

General > Block > Deinterleaver 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,

Output(Elements(k)) = 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

#### **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)

# General Block Interleaver

### **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,

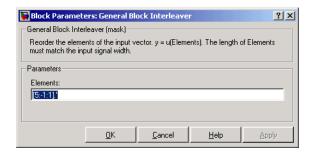
#### Output(k) = Input(**Elements**(k))

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**

**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)

# **General CRC Generator**

# **Purpose**

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

# Library

CRC sublibrary of Error Correction and Detection

# **Description**

General > CRC > Generator 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<sup>3</sup> + x<sup>2</sup> + 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<sup>3</sup> + x<sup>2</sup> + 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

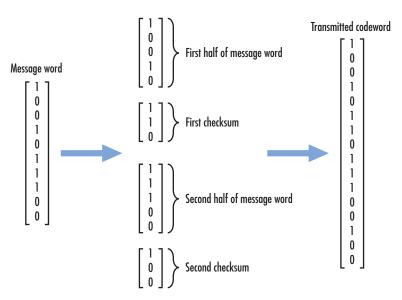
- **4** Appends the resulting checksums at the end of each subframe
- **5** Outputs concatenated subframes

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**

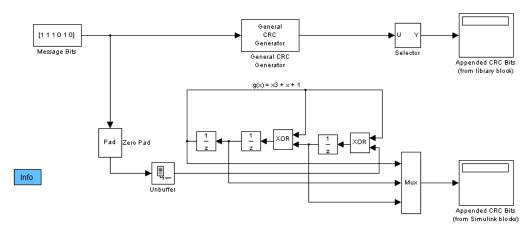
Suppose the size of the input frame is 10, the degree of the generator polynomial is 3, **Initial states** is [0], and **Checksums per frame** is 2. The block divides each input frame into two subframes of size 5 and appends a checksum of size 3 to each subframe, as shown below. The initial states are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

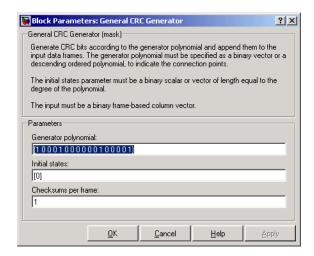
You can experiment with different initial states by changing the value of **Initial states** prior to running the simulation. For all values, the comparison (generated CRC bits from the library block with those generated from primitive Simulink blocks) holds true.

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.

### 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.

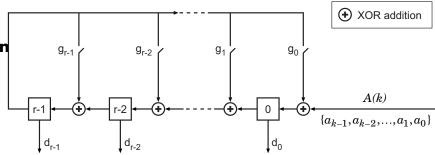
### Checksums per frame

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*.

Schematic of the CRC Implementation



The above circuit divides the polynomial

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

$$g(x) = g_{r-1}x^{r-1} + g_{r-2}x^{r-2} + \cdots + g_1x + g_0$$
 , 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  $\{a_{k-1}, a_{k-2}, \ldots, a_2, a_1, a_0\}$  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** 

General CRC Syndrome Detector

**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**

General CRC ;
Syndrome
Detector Err

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.

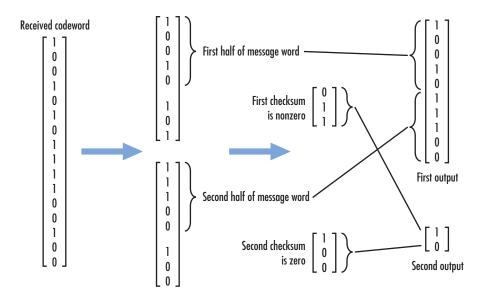
You specify the number of checksums the block calculates for each frame by the **Checksums per frame** parameter. If the **Checksums per frame** value is k, the size of the input frame is n, and the degree of the generator polynomial is r, then k must divide n - k\*r, which is the size of the message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

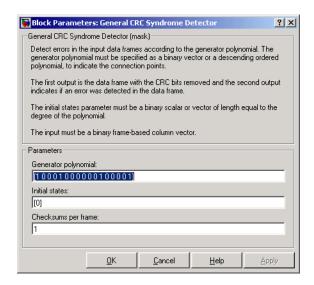
# **General CRC Syndrome Detector**



# **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.

### 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**

[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**

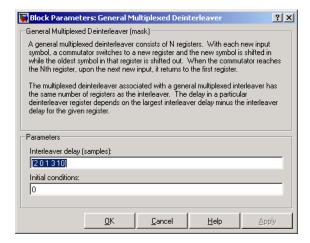
General > Multiplexed Deinterleaver The General Multiplexed Deinterleaver block restores the original ordering of a sequence that was interleaved using the General Multiplexed Interleaver block.

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**

General Multiplexed Interleaver 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.

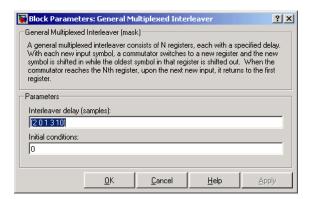
The block can accept the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The data type of this output will be the same as that of the input signal.

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:

**Purpose** 

Demodulate QAM-modulated data

Library

AM, in Digital Baseband sublibrary of Modulation

## **Description**

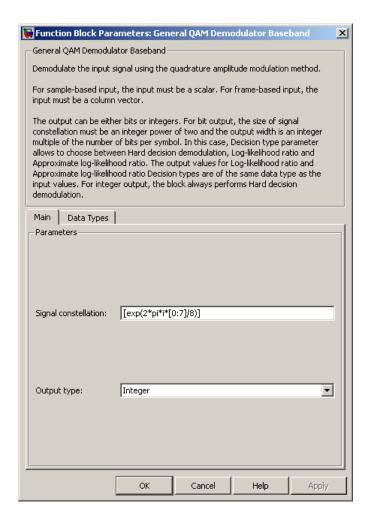


The General QAM Demodulator Baseband block demodulates a signal that was modulated using quadrature amplitude modulation. The input is a baseband representation of the modulated signal.

The input must be a discrete-time complex signal. The **Signal constellation** parameter defines the constellation by listing its points in a length-M 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. For information about the data types each block port supports, see the "Supported Data Types" on page 2-286 table on this page.

### 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.

If you set this parameter to Integer, the block produces integers.

If you set this parameter to Bit, the block produces a group of K bits, called a *binary word*, for each symbol, when **Decision type** is set to Hard decision. If **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, the block outputs bitwise LLR and approximate LLR, respectively.

#### **Decision type**

This field appears when Bit is selected in the pull-down list **Output type**.

Specifies the use of hard decision, LLR, or approximate LLR during demodulation. See "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the *Communications Toolbox User's Guide* for algorithm details.

#### Noise variance source

This field appears when you set Approximate log-likelihood ratio or Log-likelihood ratio for **Decision type**.

When you set this parameter to Dialog, you can then specify the noise variance in the **Noise variance** field. When you set this option to Port, a port appears on the block through which the noise variance can be input.

#### Noise variance

This parameter appears when the **Noise variance source** is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

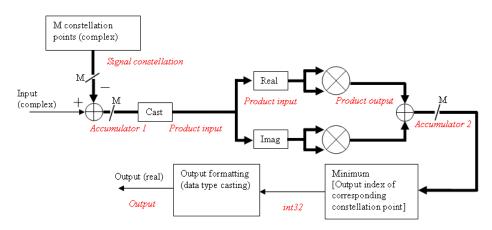
If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:

- Inf to -Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.

## Fixed-Point Signal Flow Diagrams

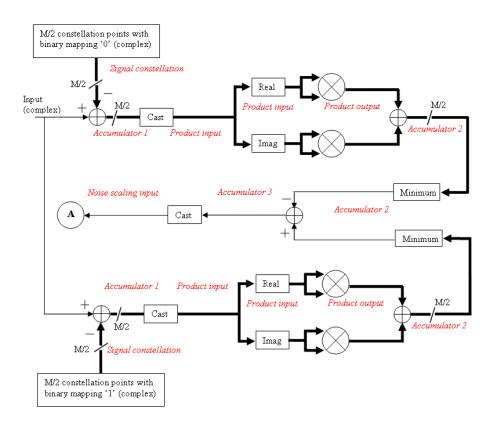


### Fixed-Point Signal Flow Diagram for Hard Decision Mode

**Note** In the figure above, M represents the size of the **Signal** constellation.

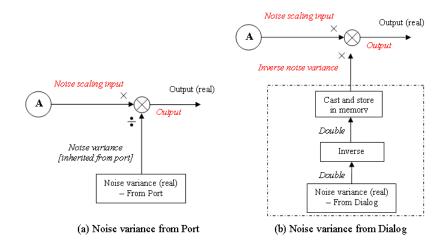
The general QAM Demodulator Baseband block supports fixed-point operations for computing Hard Decision (**Output** type set to Bit and **Decision type** is set to Hard decision) and Approximate LLR (**Output type** is set to Bit and **Decision type** is set to Approximate Log-Likelihood ratio) output values. The input values must have fixed-point data type for fixed-point operations.

**Note** Fixed-Point operations are NOT yet supported for Exact LLR output values.



Fixed-Point Signal Flow Diagram for Approximate LLR Mode

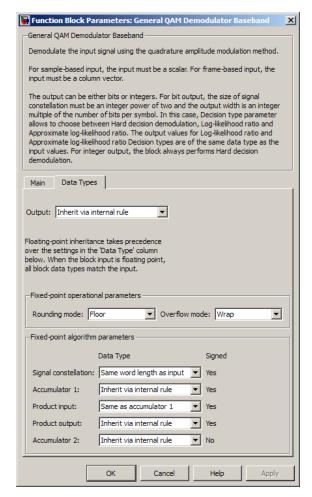
**Note** In the figure above, *M* represents the size of the **Signal** constellation.



Fixed-Point Signal Flow Diagram for Approximate LLR Mode: Noise Variance Operation Modes

**Note** If **Noise variance** is set to Dialog, the block performs the operations shown inside the dotted line once during initialization. The block also performs these operations if the **Noise variance** value changes during simulation.

### Data Types Attributes



**Fixed-Point Attributes for Hard Decision Mode** 

#### Output

The block supports the following Output options:

When you set the parameter to Inherit via internal rule (default setting), the block inherits the output data type from the

input port. The output data type is the same as the input data type if the input is of type single or double.

For integer outputs, you can set this block's output to Inherit via internal rule (default setting), Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, single, and double.

For bit outputs, when you set **Decision type** to Hard decision, you can set the output to Inherit via internal rule, Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, boolean, single, or double.

When you set **Decision type** to Log-likelihood ratio or Approximate log-likelihood ratio and the input is a floating point data type, then the output inherits its data type from the input. For example, if the input is of data type double, the output is also of data type double. When you set **Decision type** to Log-likelihood ratio or Approximate log-likelihood ratio, and the input is a fixed-point signal, the **Output** parameter, located in the Fixed-Point algorithm parameters region of the Data-Type tab, specifies the output data type.

When you set the parameter to Smallest unsigned integer, the output data type is selected based on the settings used in the **Hardware Implementation** pane of the Configuration Parameters dialog box. If you select ASIC/FPGA in the **Hardware Implementation** pane, the output data type is the ideal

minimum size, i.e., ufix(1) for bit outputs, and  $ufix(\lceil \log_2 M \rceil)$  for integer outputs. For all other choices, the **Output** data type 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).

### **Rounding Mode Parameter**

Use this parameter to specify the rounding method to be used when the result of a fixed-point calculation does not map exactly

to a number representable by the data type and scaling storing the result.

For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink Fixed Point documentation.

#### **Overflow Mode Parameter**

Use this parameter to specify the method to be used if the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result:

- Saturate represents positive overflows as the largest positive number in the range being used, and negative overflows as the largest negative number in the range being used.
- Wrap uses modulo arithmetic to cast an overflow back into the representable range of the data type. See Modulo Arithmetic for more information.

For more information, see the Rounding Mode Parameter subsection of Specifying Fixed-Point Attributes.

### Signal constellation

Use this parameter to define the data type of the **Signal** constellation parameter.

- When you select Same word length as input the word length
  of the Signal constellation parameter matches that of the
  input to the block. The fraction length is computed to provide
  the best precision for given signal constellation values.
- When you select Specify word length, the Word Length field appears, and you may enter a value for the word length. The fraction length is computed to provide the best precision for given signal constellation values.

#### Accumulator 1

Use this parameter to specify the data type for Accumulator 1:

- When you select Inherit via internal rule, the block automatically calculates the output word and fraction lengths. For more information, see the Inherit via Internal Rule subsection of the Signal Processing Blockset User's Guide.
- When you select Binary point scaling, you can enter the word length and the fraction length of **Accumulator 1**, in bits.

#### **Product Input**

Use this parameter to specify the data type for **Product input**.

- When you select Same as accumulator 1, the **Product Input** characteristics match those of **Accumulator 1**.
- When you select Binary point scaling you can enter the word length and the fraction length of **Product input**, in bits.

#### **Product Output**

Use this parameter to select the data type for Product output.

- When you select Inherit via internal rule, the block automatically calculates the output signal type. For more information, see the Inherit via Internal Rule subsection of the Signal Processing Blockset User's Guide.
- When you select Binary point scaling enter the word length and the fraction length for **Product output**, in bits.

#### Accumulator 2

Use this parameter to specify the data type for **Accumulator 2**:

 When you select Inherit via internal rule, the block automatically calculates the accumulator data type. The internal rule calculates the ideal, full-precision word length and fraction length as follows:

$$WL_{\text{ideal accumulator 2}} = WL_{\text{input to accumulator 2}}$$

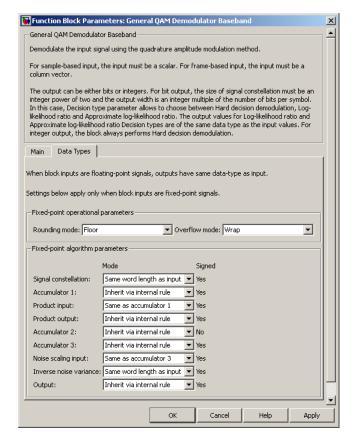
$$FL_{\text{ideal accumulator 2}} = FL_{\text{input to accumulator 2}}$$

After the full-precision result is calculated, your particular hardware may still affect the final word and fraction lengths set by the internal rule. For more information, see The Effect

of the Hardware Implementation Pane on the Internal Rule subsection of the Signal Processing Blockset User's Guide.

The internal rule always sets the sign of data-type to Unsigned.

 When you select Binary point scaling, you are able to enter the word length and the fraction length of Accumulator 2, in bits.



**Fixed-Point Attributes for Approximate LLR Mode** 

The settings for the following fixed-point parameters only apply when you set **Decision type** to Approximate log-likelihood ratio.

#### Accumulator 3

When you select Inherit via internal rule, the block automatically calculates the accumulator data type. The internal rule first calculates ideal, full-precision word length and fraction length as follows:

$$WL_{\text{ideal accumulator } 3} = WL_{\text{input to accumulator } 3} + 1$$

$$FL_{\text{ideal accumulator } 3} = FL_{\text{input to accumulator } 3}$$
.

After the full-precision result is calculated, your particular hardware may still affect the final word and fraction lengths set by the internal rule. For more information, see The Effect of the Hardware Implementation Pane on the Internal Rule subsection of the Signal Processing Blockset User's Guide.

The internal rule always sets the sign of data-type to Signed.

### Noise scaling input

- When you select Same as accumulator 3, the **Noise scaling** input characteristics match those of **Accumulator 3**.
- When you select Binary point scaling you are able to enter the word length and the fraction length of Noise scaling input, in bits.

#### Inverse noise variance

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

- When you select Same word length as input the word length
  of the Inverse noise variance parameter matches that of the
  input to the block. The fraction length is computed to provide
  the best precision for a given inverse noise variance value.
- When you select Specify word length, the **Word Length** field appears, and you may enter a value for the word length.

The fraction length is computed to provide the best precision for a given inverse noise variance value.

#### Output

When you select Inherit via internal rule, the **Output data type** is automatically set for you.

If you set the **Noise variance source** parameter to Dialog, the output is a result of product operation as shown in the Noise Variance Operation Modes Signal Flow DiagramFixed-Point Signal Flow Diagram for Approximate LLR Mode: Noise Variance Operation Modes on page 2-278. In this case, it follows the internal rule for Product data types specified in the Inherit via Internal Rule subsection of the *Signal Processing Blockset User's Guide*.

If the **Noise variance source** parameter is set to Port, the output is a result of division operation as shown in the signal flow diagram. In this case, the internal rule calculates the ideal, full-precision word length and fraction length as follows:

$$WL_{\text{output}} = \max(WL_{\text{Noise scaling input}}, WL_{\text{Noise variance}})$$

$$FL_{\rm output} = FL_{\rm Noise\ scaling\ input\ (dividend)} - FL_{\rm Noise\ variance\ (divisor)} \ .$$

After the full-precision result is calculated, your particular hardware may still affect the final word and fraction lengths set by the internal rule. For more information, see The Effect of the Hardware Implementation Pane on the Internal Rule subsection of the Signal Processing Blockset User's Guide.

The internal rule for **Output** always sets the sign of data-type to Signed.

For additional information about the parameters pertaining to fixed-point applications, see Specifying Fixed-Point Attributes.

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	• Signed fixed—point when <b>Output type</b> is Integer or <b>Output type</b> is Bit and <b>Decision type</b> is either Hard-decision or Approximate LLR)
Var	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Single-precision floating point
	• Boolean when Output type is Bit and Decision type is Hard-decision.
	• 8-, 16-, and 32-bit signed integers when <b>Output type</b> is <b>Integer</b> or <b>Output type</b> is <b>Bit</b> and <b>Decision type</b> is Hard-decision
	• 8-, 16-, and 32-bit unsigned integers when <b>Output type</b> is Integer or <b>Output type</b> is Bit and <b>Decision type</b> is Hard-decision
	• ufix(1) in ASIC/FPGA when <b>Output type</b> is <b>Bit</b> and <b>Decision type</b> is Hard-decision
	• $\mathit{ufix}(\lceil \log_2 M \rceil)$ in ASIC/FPGA when <b>Output type</b> is Integer
	• Signed fixed-point when <b>Output type</b> is Bit and <b>Decision type</b> is Approximate LLR

Pair Block General QAM Modulator Baseband

See Also Rectangular QAM Demodulator 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.

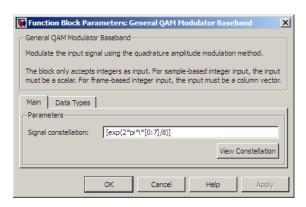
The **Signal constellation** parameter defines the constellation by listing its points in a length-M vector of complex numbers. The input signal values must be integers between 0 and M-1. The block maps an input integer m to the (m+1)st value in the **Signal constellation** vector.

The input can be either a scalar or a frame-based column vector. For information about the data types each block port supports, see the "Supported Data Types" on page 2-289 table on this page.

#### **Constellation Visualization**

The General QAM Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the *Communications Blockset User's Guide*.

### 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 software. This parameter is only visible when you select User-defined for the **Output data type** parameter.

#### Set output fraction length to

Specify the scaling of the fixed-point output by either of the following 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.

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	• 8-, 16-, 32-bit signed integers
	• 8-, 16-, 32-bit unsigned integers
	• $ufix(\lceil \log_2 M \rceil)$
Output	Double-precision floating point
	Single-precision floating point
	Signed fixed-point

Pair Block General QAM Demodulator Baseband

See Also Rectangular QAM Modulator Baseband

# **General TCM Decoder**

### **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.

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.

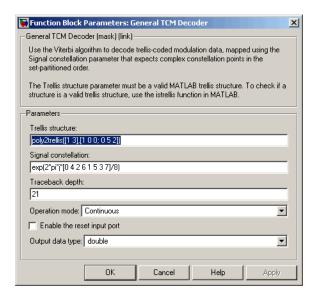
- In Truncated mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In Terminated mode, the block treats each frame independently. The traceback path always starts at the all-zeros 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

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,

 $Introduction\ to\ Trell is \hbox{-} Coded\ Modulation\ with\ Applications,\ New\ York,$ 

Macmillan, 1991.

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

# **General TCM Encoder**

### **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 a binary value.

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

The encoder registers begin in the all-zeros state. You can configure the encoder so that it resets its registers to the all-zeros state during the course of the simulation. To do this, set the Operation mode to Reset on nonzero input via port. The block then opens a second input port, labeled Rst. The signal at the Rst port is a scalar signal. When it is nonzero, the encoder resets before processing the data at the first input port.

### **Signal Constellations**

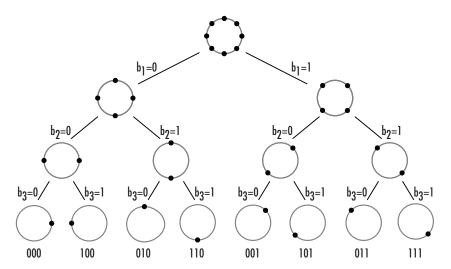
The trellis-coded modulation technique partitions the constellation into subsets called cosets so as to maximize the minimum distance between pairs of points in each coset.

**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.

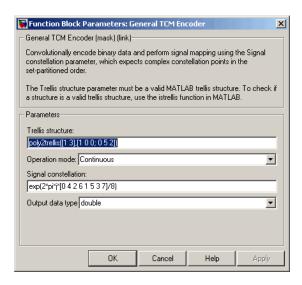


For other examples of signal constellations in set-partitioned order, see [1] or the reference pages for the M-PSK TCM Encoder and Rectangular QAM TCM Encoder blocks.

## **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].

### Dialog Box



#### Trellis structure

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

### Operation mode

In Continuous mode (default setting), the block retains the encoder states at the end of each frame, for use with the next frame.

In Truncated (reset every frame) mode, the block treats each frame independently. I.e., the encoder states are reset to all-zeros state at the start of each frame.

In Terminate trellis by appending bits mode, the block treats each frame independently. For each input frame, extra bits are used to set the encoder states to all-zeros state at the end of the frame. The output length is given by

 $y = n \cdot (x+s)/k$ , where x is the number of input bits, and s = constraint length - 1 (or, in the case of multiple constraint

# **General TCM Encoder**

lengths, s = sum(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.

#### 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, and the model uses a variable-step solver or a fixed-step solver with the **Tasking Mode** parameter set to SingleTasking, then the delay consists of D+1 zero symbols.
- If the input signal is frame-based, then the delay consists of D zero symbols.

The optimal **Traceback length** parameter value is dependent on minimum squared Euclidean distance calculations. Alternatively, a typical value, dependent on the number of states, can be chosen using the "five-times-the-constraint-length" rule, which corresponds

to  $5 \cdot \log 2(numStates)$  . The number of states is determined by the following equation:

$$numStates = \begin{cases} p \cdot M^{(L-1)}, for \, even \, m \\ 2p \cdot M^{(L-1)}, for \, odd \, m \end{cases}$$

# **GMSK Demodulator Baseband**

#### where:

- h = m/p is the modulation index proper rational form
  - = m = numerator of modulation index
  - = p =denominator of modulation index
- *L* is the Pulse length
- *M* is M-ary number

For a binary raised cosine pulse shape with a pulse length of 3, h=2/3,

this rule  $(5*\log 2(3*2^2) = 18)$  gives a result close to the optimum value of 20.

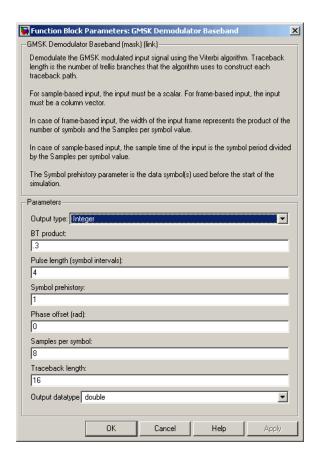
#### **Inputs and Outputs**

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 GMSK Demodulator Baseband block uses to construct each traceback path.

## Output data type

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

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers

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.

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

$$Q(t) = \int_{-1}^{\infty} \frac{1}{\sqrt{2\pi}} e^{-\tau^2/2} d\tau$$

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**

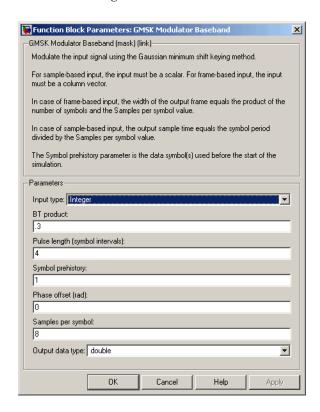
The input can be either a scalar or a frame-based column vector. If the **Input type** parameter is set to Integer, then the block accepts values

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.

# **GMSK Modulator Baseband**

#### 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.

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Boolean (bit input mode only)
	• 8-, 16-, and 32-bit signed integers (integer input mode only)
Output	Double-precision floating point
	Single-precision floating point

#### **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.

# **Purpose**

Generate Gold sequence from set of sequences

# Library

Sequence Generators sublibrary of Comm Sources

# **Description**

Gold Sequence Generator 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  $N=2^{\rm n}$  - 1, called a *preferred pair*, as defined in "Preferred Pairs of Sequences" on page 2-309 below. The set G(u, v) of Gold sequences is defined by

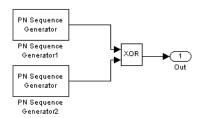
$$G(u,v) = \{u,v,u \oplus v,u \oplus Tv,u \oplus T^2v,...,u \oplus T^{N-1}v\}$$

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.



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.

n	N	Preferred Polynomial[1]	Preferred Polynomial[2]
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]

n	N	Preferred Polynomial[1]	Preferred Polynomial[2]
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, ...,  $2^{n}-2$ ]. The correspondence between **Sequence index** and the output sequence is given in the following table.

Sequence Index	Output Sequence
-2	u
-1	v
0	$u \oplus v$

Sequence Index	Output Sequence
1	$u \oplus Tv$
2	$u \oplus T^2v$
2 <sup>n</sup> -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 2-605 for an example.

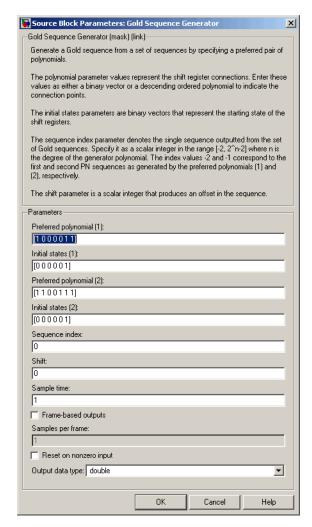
# **Preferred Pairs of Sequences**

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
  - $\blacksquare$  q is odd
  - $q = 2^k+1 \text{ or } q = 2^{2k}-2^k+1$
  - v is obtained by sampling every qth symbol of u

• 
$$\gcd(n,k) = \begin{cases} 1 & n \equiv 1 \mod 2 \\ 2 & n \equiv 2 \mod 4 \end{cases}$$

# 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 boolean, double or Smallest unsigned integer. By default, the block sets this to double.

When the parameter is set to Smallest unsigned integer, the output data type is selected based on the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in the Hardware Implementation pane, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

#### See Also

Kasami Sequence Generator, PN Sequence Generator

# References

- [1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.
- [2] Gold, R., "Maximal Recursive Sequences with 3-valued Recursive Cross-Correlation Functions," *IEEE Trans. Infor. Theory*, Jan., 1968, pp. 154-156.
- [3] Gold, R., "Optimal Binary Sequences for Spread Spectrum Multiplexing, *IEEE Trans. Infor. Theory*, Oct., 1967, pp. 619-621.
- [4] Sarwate, D.V., and M.B. Pursley, "Crosscorrelation Properties of Pseudorandom and Related Sequences," *Proc. IEEE*, Vol. 68, No. 5, May, 1980, pp. 583-619.
- [5] Dixon, Robert, Spread Spectrum Systems with Commercial Applications, Third Edition, Wiley–Interscience, 1994.

**Purpose** 

Generate Hadamard code from orthogonal set of codes

Library

Sequence Generators sublibrary of Comm Sources

**Description** 

Hadamard Code Generator 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  $H_{\rm N}$ , is defined recursively as follows.

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

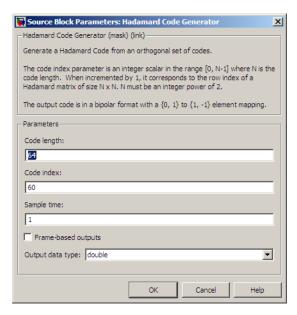
The N-by-N Hadamard matrix has the property that

$$H_N H_N^T = NI_N$$

where  $\boldsymbol{I}_{N}$  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, ..., N-1], by the **Code index** parameter.

# 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.

# **Hadamard Code Generator**

## 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**

## **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.

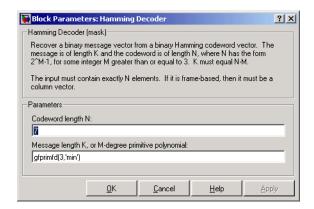
The input must contain exactly N elements. If it is frame-based, then it must be a column vector. The output is a vector of length K.

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 <code>qfprimdf(M)</code> 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 Communications Toolbox gfprimfd function.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-317 table on this page.

# 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  $GF(2^M)$ .

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	Fixed-point
Out	Double-precision floating point
	Single-precision floating point

# **Hamming Decoder**

	Port	Supported Data Types
Pair Block See Also	Hamming Encoder hammgen (Communications Toolbox)	<ul> <li>Boolean</li> <li>8-, 16-, and 32-bit signed integers</li> <li>8-, 16-, and 32-bit unsigned integers</li> <li>Fixed-point</li> </ul>

# **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.

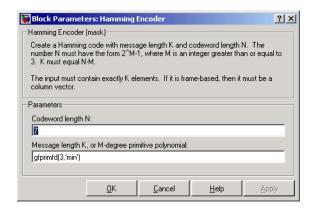
The coding scheme uses elements of the finite field  $GF(2^M)$ . You can either specify the primitive polynomial that the algorithm should use, or you can rely on the default setting:

- 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 Communications Toolbox gfprimfd function.

For information about the data types each block port supports, see the "Supported Data Type" on page 2-320 table on this page.

# **Hamming Encoder**

# 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  $GF(2^M)$ .

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	Fixed-point
Out	Double-precision floating point
	Single-precision floating point

# **Hamming Encoder**

	Port	Supported Data Types
Pair Block See Also	Hamming Decoder hammgen (Communications Toolbox)	<ul> <li>Boolean</li> <li>8-, 16-, and 32-bit signed integers</li> <li>8-, 16-, and 32-bit unsigned integers</li> <li>Fixed-point</li> </ul>

# **Helical Deinterleaver**

**Purpose** 

Restore ordering of symbols permuted by helical interleaver

Library

Convolutional sublibrary of Interleaving

**Description** 

Helical Deinterleaver 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\*N at each time step and inserts them into the next N rows of the array. The block also places the **Initial condition** parameter into certain positions in the top few rows of the array (not only to accommodate the helical pattern but also to preserve the vector indices of symbols that pass through the Helical Interleaver and Helical Deinterleaver blocks in turn).

The output consists of consecutive groups of N symbols. Counting from the beginning of the simulation, the block selects the kth output group in the array from column k mod C. The selection is helical because of the reduction modulo C and because the first symbol in the kth group is in row 1+(k-1)\*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**

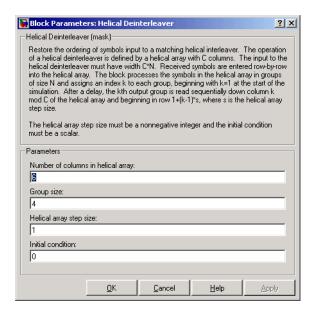
After processing a message with the Helical Interleaver block and the Helical Deinterleaver block, the deinterleaved data lags the original message by

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

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\*C\*N for some nonnegative integer m. You can use the Signal Processing Blockset Delay block 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**

Helical Interleaver 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.

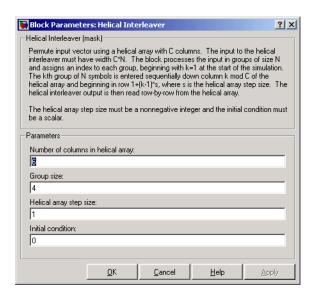
The block uses the array internally for its computations. If C is the **Number of columns in helical array** parameter, then the array has C columns and unlimited rows. If N is the **Group size** parameter, then the block accepts an input of length C\*N at each time step and partitions the input into consecutive groups of N symbols. Counting from the beginning of the simulation, the block places the kth group in the array along column k mod C. The placement is helical because of the reduction modulo C and because the first symbol in the kth 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.

# 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.

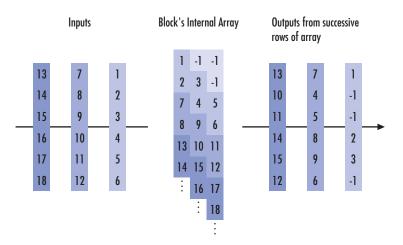
#### **Initial condition**

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.

**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. For information about the data types each block port supports, see the "Supported Data Type" on page 2-334 table on this page.

- 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

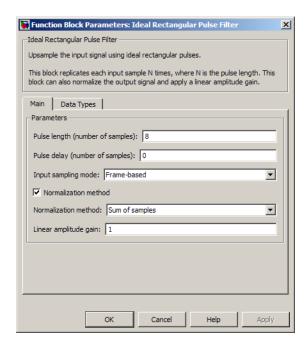
You determine the block's normalization behavior using the **Normalize output signal** and **Linear amplitude gain** parameters.

- 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.

The output is scaled by  $\sqrt{N}$ . If the output of this block feeds the input to the AWGN Channel block, specify the AWGN signal power parameter to be 1/N.

# Dialog Box



#### 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:  $\mbox{{\tt Frame-based}}$  or  $\mbox{{\tt Sample-based}}.$ 

# Normalize output signal

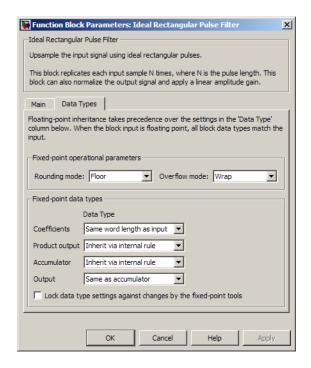
If you select this, then the block scales the set of replicated values before applying the linear amplitude gain.

#### 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.



#### Rounding mode

Use this parameter to specify the rounding method to be used when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. The filter coefficients do not obey this parameter; they always round to Nearest.

For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink Fixed Point documentation.

#### Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

#### Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator). See "Filter Structure Diagrams" in *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.

#### Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Fixed-point
Out	Double-precision floating point
	Single-precision floating point
	Signed fixed-point

# **Ideal Rectangular Pulse Filter**

## **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 (Normalize output signal cleared)	1	0, 0, 0, 1, 1, 1, 1, 2, 2, 2, 2, 3, 3, 3, 3,
None (Normalize output signal cleared)	10	0, 0, 0, 10, 10, 10, 10, 20, 20, 20, 20, 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*4=1
Sum of samples	10	0, 0, 0, 2.5, 2.5, 2.5, 2.5, 5, 5, 5, 5, 5, 7.5, 7.5, 7.5, 7.5, 7
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*4=1^2
Energy per pulse	10	0, 0, 0, 5, 5, 5, 5, 10, 10, 10, 10, 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.

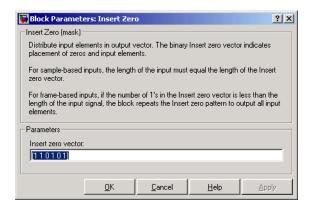
The block can accept the data types int8, uint8, int16, uint16, int32, uint32, boolean, single, double, and fixed-point. The data type of this output will be the same as that of the input signal.

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

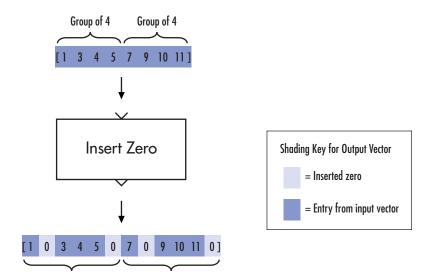
A binary vector whose pattern of 0s and 1s indicates where the block should place either 0s or input vector elements, respectively, in the output vector.

### **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

# Integer-Input RS Encoder

**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 - number of punctures). You specify both N and K directly in the block dialog. The symbols for the code are integers between 0 and  $2^{M}$ -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 2-340 below. The difference N - K must be an even integer.

This block can output shortened codewords when N and K are appropriately specified. To specify output codewords that are shortened by a length S, N and K must be specified in the dialog box as  $N_{\rm full}-S$  and  $K_{\rm full}-S$ , where  $N_{\rm full}$  and  $K_{\rm full}$  are the N and K of an unshortened code. If  $S < (N_{\rm full}+1)/2$ , the encoder can automatically determine

the value of  $N_{\rm full}$  and  $K_{\rm full}$ . However, if  $S \ge (N_{\rm full} + 1)/2$ , **Primitive polynomial** must be specified in order to properly define the extension field for the code.

The input and output are integer-valued signals that represent messages and codewords, respectively. The input must be a frame-based column vector whose length is an integer multiple of K. 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 information about the data types each block port supports, see the "Supported Data Type" on page 2-345 table on this page.

For more information on representing data for Reed-Solomon codes, see the section "Integer Format (Reed-Solomon Only)" in *Communications Blockset User's Guide*.

If the encoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

The default value of M is the smallest integer that is greater than or equal to log 2(N+1), that is, ceil(log 2(N+1)). You can change the

# **Integer-Input RS Encoder**

value of M from the default by specifying the primitive polynomial for  $GF(2^M)$ , as described in "Specifying the Primitive Polynomial" on page 2-340 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 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 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  $3 \le N < 2^{M}-1$  and M must lie in the range  $3 \le M \le 16$ .

### **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<sup>M</sup>) 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

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

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).

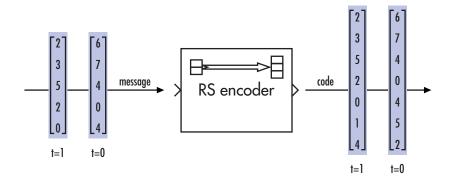
#### **Puncture Codes**

The block supports puncturing when you select the **Puncture code** parameter. This enables the **Puncture vector** parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, a 1 means that the data symbol is passed unaltered, and a 0 means that the data symbol is punctured (i.e., removed) from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

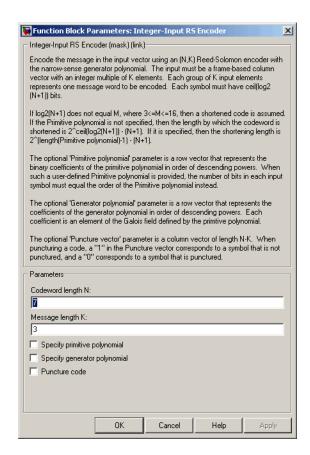
# **Examples**

Suppose 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.

# **Integer-Input RS Encoder**

#### 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.

#### Puncture code

Selecting this check box enables the field **Puncture vector**.

#### **Puncture vector**

This field is available only when **Puncture code** is selected.

A column vector of length *N-K*. A value of 1 in the **Puncture vector** corresponds to a symbol that is not punctured, and a 0 corresponds to a symbol that is punctured.

The default value is [ones(2,1); zeros(2,1)].

# **Integer-Input RS Encoder**

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
Out	Double-precision floating point
	Single-precision floating point
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

**Pair Block** 

Integer-Output RS Decoder

**See Also** 

Binary-Input RS Encoder

#### **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-number of punctures). You specify both N and K directly in the block dialog. The symbols for the code are integers between 0 and  $2^M$ -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 2-340 below. The difference N- K must be an even integer.

This block can output shortened codewords when N and K are appropriately specified. To specify output codewords that are shortened by a length S, N and K must be specified in the dialog box as  $N_{\rm full}-S$  and  $K_{\rm full}-S$ , where  $N_{\rm full}$  and  $K_{\rm full}$  are the N and K of an unshortened code. If  $S < (N_{\rm full}+1)/2$ , the encoder can automatically determine

the value of  $N_{\rm full}$  and  $K_{\rm full}$ . However, if  $S \ge (N_{\rm full} + 1)/2$ , **Primitive polynomial** must be specified in order to properly define the extension field for the code.

The input and output are integer-valued signals that represent codewords and messages, respectively. The input must be a frame-based column vector whose length is an integer multiple of (N - number of punctures). The output is a frame-based column vector whose length is the same integer multiple of K, and whose data type is inherited from the input. For information about the data types each block port supports, see the "Supported Data Type" on page 2-351 table on this page.

For more information on representing data for Reed-Solomon codes, see the section "Integer Format (Reed-Solomon Only)" in the *Communications Blockset User's Guide*.

If the decoder is processing multiple codewords per frame, then the same puncture pattern holds for all codewords.

The default value of M is ceil(log2(N+1)), that is, the smallest integer greater than or equal to log2(N+1). You can change the value of M from the default by specifying the primitive polynomial for GF(2<sup>M</sup>), as described in "Specifying the Primitive Polynomial" on page 2-340 below. If N is less than 2<sup>M</sup>-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 2-340.

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

The second output is the number of errors detected during decoding of the codeword. A -1 indicates that the block detected more errors than it could correct using the coding scheme. An (N,K) Reed-Solomon code can correct up to floor((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.

In the case of a decoder failure, the message portion of the decoder input is returned unchanged as the decoder output.

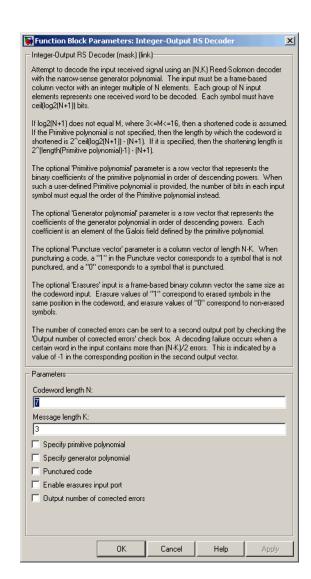
The sample times of the input and output signals are equal.

# Punctured Codes

This block supports puncturing when you select the **Punctured code** parameter. This selection enables the **Puncture vector** parameter, which takes in a binary vector to specify the puncturing pattern. For a puncture vector, 1 means that the data symbol is passed unaltered, and 0 means that the data symbol is punctured (i.e., removed) from the data stream. This convention is carried for both the encoder and the decoder. For more information, see "Shortening, Puncturing, and Erasures".

**Note** 1s and 0s have precisely opposite meanings for the puncture and erasure vectors. For an erasure vector, 1 means that the data symbol is to be replaced with an erasure symbol, and 0 means that the data symbol is passed unaltered. This convention is carried for both the encoder and the decoder.

### 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.

#### Puncture code

Selecting this check box enables the field **Puncture vector**.

#### Puncture vector

This field is available only when **Puncture code** is selected.

A column vector of length *N-K*. A value of 1 in the **Puncture vector** corresponds to a symbol that is not punctured, and a 0 corresponds to a symbol that is punctured.

The default value is [ones(2,1); zeros(2,1)].

#### **Enable erasures input port**

Selecting this check box will open the port, Era.

Through the port, you can input a frame-based binary column vector the same size as the codeword input.

Erasure values of 1 correspond to erased symbols in the same position in the codeword, and values of 0 correspond to nonerased symbols.

#### 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.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
Out	Double-precision floating point
	Single-precision floating point
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

Port	Supported Data Types
Era	Double-precision floating point
	Boolean
Err	Double-precision floating point
	Single-precision floating point
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

### **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.

[3] Clark, George C., Jr., and J. Bibb Cain, *Error-Correction Coding for Digital Communications*, New York, Plenum Press, 1981.

### **See Also**

Binary-Output RS Decoder

**Purpose** 

Map vector of integers to vector of bits

Library

Utility Blocks

### **Description**

Integer to Bit Converter The Integer to Bit Converter block maps each integer or fixed-point value in the input vector to a group of bits in the output vector.

The block maps each integer value (or stored integer when a fixed point input is used) to a group of M bits, using the selection for the **Output** bit order to determine the most significant bit. The resulting output vector length is M times the input vector length.

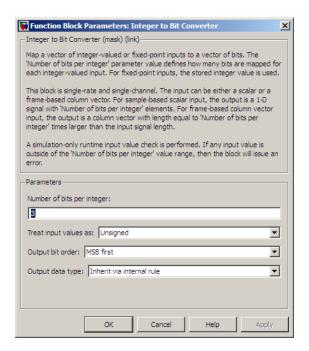
For integer-valued inputs, if M is the Number of bits per integer and Treat input values as is set to Unsigned, then the input values must be between 0 and  $2^{M}$ -1. If M is the Number of bits per integer and Treat input values as is set to Signed, then the block maps each group of M bits to an integer between  $-2^{M-1}$  and  $2^{M-1}$ -1.

This block is single-rate and single-channel. The input can be either a scalar or a frame-based column vector. For a sample-based scalar input, the output is a 1-D signal with M elements. For a frame-based column vector input, the output is a frame-based column vector with a length equal to M times the input signal length.

A simulation-only runtime input value check is performed. If any input value is outside of the **Number of bit per integer** value range, the block will issue an error.

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

### Dialog Box



#### Number of bits per integer

The number of bits the block uses to represent each integer of the input. This parameter must be an integer between 1 and 32.

#### Treat input values as

Indicate if the integer value input ranges should be treated as signed or unsigned. The default setting is Unsigned.

#### Output bit order

Define whether the first bit of the output signal is the most significant bit (MSB) or the least significant bit (LSB). The default selection is MSB first.

#### Output data type

You can choose the following **Output data type** options:

• Inherit via internal rule

- Smallest integer
- Same as input
- double
- single
- uint8
- uint16
- uint32

The default selection for this parameter is Inherit via internal rule.

When the parameter is set to Inherit via internal rule, the block determines the output data type based on the input data type.

- If the input signal is floating-point (either single or double), the output data type is the same as the input data type.
- If the input data type is not floating-point, the output data type is determined as if the parameter is set to Smallest integer.

When the parameter is set to Smallest integer, the block selects the output data type based on settings used in the **Hardware Implementation** pane of the Configuration Parameters dialog box.

- If ASIC/FPRA is selected, the output data type is the ideal one-bit size; ufix1.
- For all other selections, the output data type is an unsigned integer with the smallest available word length, as defined in the Hardware Implementation settings (e.g. uint8)

## **Examples**

# Fixed-Point Integer To Bit and Bit To Integer Conversion (Audio Scrambling and Descrambling Example)

#### **Overview**

This example illustrates how to use the Bit to Integer and Integer to Bit Converter blocks with fixed-point signals.

This example uses a simplified audio scrambler configuration and a 16-bit, fixed-point digital audio source, which is recorded speech. The left-side of the model represents the audio scrambler subsystem and the right-side represents the descrambler subsystem.

#### **Opening the Model**

You can open the model by typing doc\_audioscrambler at the MATLAB command line.

#### Structure

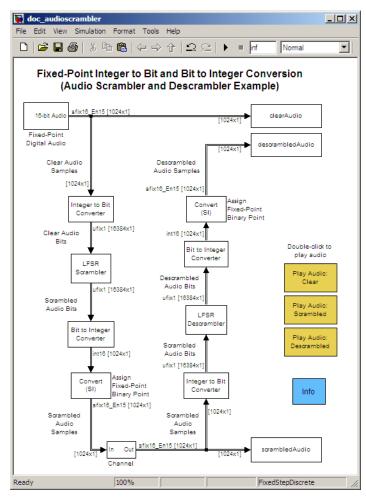
In the audio scrambler subsystem, the Integer to Bit Converter block unpacks each 16-bit audio sample into a binary, 1-bit signal. The binary signal passes to a linear feedback shift register (LFSR) scrambler, which randomizes the bits in a controllable way, thereby scrambling the signal. The Communications Blockset Scrambler block is used in the LFSR implementation. From the LFSR, the scrambled audio bits pass to the Bit to Integer Converter block. This block packs the scrambled 1-bit samples into 16-bit audio samples. The audio samples pass to the Data Type Conversion block, which converts the integer-based audio samples back into fixed-point samples.

The fixed-point samples pass from the scrambler subsystem to a channel. The channel sends the samples to the descrambler subsystem. For illustrative purposes, this example uses a noiseless channel. In an actual system, a channel may introduce noise. Removing such noise requires a more sophisticated design.

In the audio descrambler subsystem, the Integer to Bit Converter block unpacks each 16-bit audio sample into a binary, 1-bit signal. The binary signal passes to a linear feedback shift register (LFSR) descrambler, which randomizes the bits in a controllable way, reversing the scrambling process. This LFSR descrambler implementation uses the Communications Blockset Descrambler block. From the LFSR, the

descrambled audio bits pass to the Bit to Integer Converter block. This block packs the descrambled 1-bit samples into 16-bit audio samples. The audio samples pass to the Data Type Conversion block, which converts the integer-based audio samples back into fixed-point samples.

In Simulink, the sfix16\_En15 data type represents a signed (s) fixed-point (fix) signal with word length 16 and fraction length 15. Therefore, this model represents audio signals using the sfix16\_En15 data type, except when converting to and from 1-bit binary signals. All 1-bit signals are represented by ufix1, as seen at the output of the Integer to Bit Converter block. The frame-based audio source has a frame size (or number of samples per frame) of 1024. For more information on fixed-point signals, please refer to Fixed-Point Numbers in the Simulink documentation.



#### **Running the Model**

You must run the demo before you can listen to any of the audio signals.

You can run the demo by clicking Simulation > Start.

You can hear the audio signals by clicking the model's yellow, audio icons.

#### **Converter Block Settings**

In the audio scrambler and descrambler subsystems, the Integer to Bit Converter block settings are:

- Number of bits per integer: 16
- Treat input values as: Signed
- Output bit order: MSB first
- Output data type: Inherit via internal rule

In the audio scrambler and descrambler subsystems, the Bit to Integer Converter block settings are:

- Number of bits per integer: 16
- Input bit order: MSB first
- After bit packing, treat resulting integer values as: Signed
- Output data type: Inherit via internal rule

### Pair Block Bit to

Bit to Integer Converter

#### See Also

de2bi and dec2bin

# **Integrate and Dump**

#### **Purpose**

Integrate discrete-time signal, resetting to zero periodically

# Library

Comm Filters

## **Description**

Integrate and Dump 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.

The port data types are inherited from the signals that drive the block. For information about the data types each block port supports, see the "Supported Data Type" on page 2-365 table on this page.

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\*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\*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**

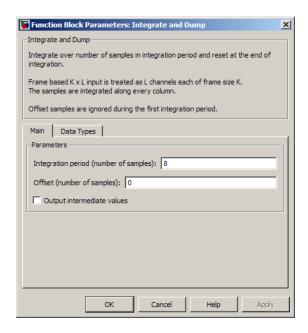
A nonzero value in the **Offset** parameter causes the block to output one or more zeros during the initial period while it discards input samples. If the input is a frame-based matrix with n columns and the **Offset** parameter is a length-n vector, then the mth element of the **Offset** vector is the offset for the mth 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 mth 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 2-366.

# **Integrate and Dump**

### Dialog Box



#### Integration period

The number of input samples between resets.

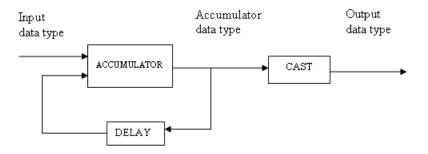
#### Offset

A nonnegative integer vector or scalar specifying the number of input samples to discard from each column of input data at the beginning of the simulation.

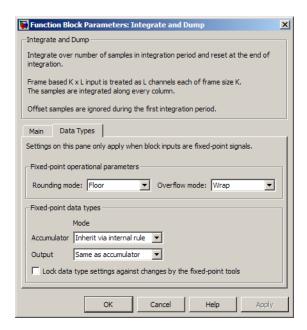
#### Output intermediate values

Determines whether the block suppresses the intermediate cumulative sums between successive resets.

### **Fixed-Point Signal Flow Diagram**



#### **Fixed-Point Attributes**



The settings for the following parameters only apply when block inputs are fixed-point signals.

# **Integrate and Dump**

#### Rounding mode

Use this parameter to specify the rounding method to be used when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result.

For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink Fixed Point documentation.

#### Overflow mode

Use this parameter to specify the method to be used if the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result:

- Saturate represents positive overflows as the largest positive number in the range being used, and negative overflows as the largest negative number in the range being used.
- Wrap uses modulo arithmetic to cast an overflow back into the representable range of the data type. See "Modulo Arithmetic" for more information.

#### Accumulator—Mode

Use the **Accumulator—Mode** parameter to specify how you would like to designate the accumulator word and fraction lengths:

- When you select Inherit via internal rule, the block automatically calculates the accumulator output word and fraction lengths.
- When you select Same as input, these characteristics match those of the input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator.

The bias of all signals in Signal Processing Blockset software is zero.

#### Output

Use the **Output** parameter to choose how you specify the word length and fraction length of the output of the block:

- When you select Same as accumulator, these characteristics match those of the accumulator.
- When you select Same as input, these characteristics match those of the input to the block.
- When you select Binary point scaling, enter the word length and the fraction length of the output, in bits.
- When you select Slope and bias scaling, enter the word length, in bits, and the slope of the output.

For additional information about the parameters pertaining to fixed-point applications, see Specifying Fixed-Point Attributes.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Fixed-point
Out	Double-precision floating point
	Single-precision floating point
	Fixed-point

### **Examples**

If **Integration period** is 4 and **Offset** is the scalar 3, then the table below shows how the block treats the beginning of a ramp (1, 2, 3, 4,...) in several situations. (The values shown in the table do not reflect vector sizes but merely indicate numerical values.)

# **Integrate and Dump**

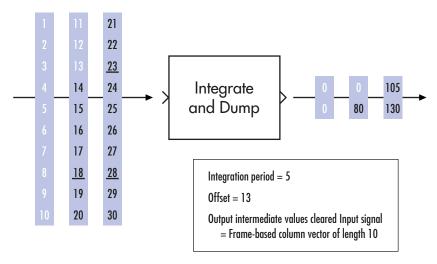
Output intermediate values Check Box	Input Signal Properties	First Several Output Values
Cleared	Sample-based scalar	0, 0, 4+5+6+7, and 8+9+10+11, where one 0 is an initial transient value and the other 0 is a delay value that results from the cleared check box and sample-based input.
Cleared	Frame-based 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.

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 ceil(13/5) output samples because the block discards 13 input samples and the integration period is 5. The first output sample after the transient effect is over, 80, corresponds to the sum 14+15+16+17+18 and appears at the time of the input sample 18. The next output sample, 105, corresponds to the sum 19+20+21+22+23 and appears at the time of the input sample 23. Notice that the input sample 23 is one frame later than the input sample 19; that is, this five-sample integration period spans two input frames. As a result, the output of 105 is delayed compared to the first input (19) that contributes to that sum.

#### **See Also**

Windowed Integrator, Discrete-Time Integrator (Simulink documentation), Ideal Rectangular Pulse Filter

# Interlacer

#### **Purpose**

Alternately select elements from two input vectors to generate output vector

### Library

Sequence Operations

# **Description**

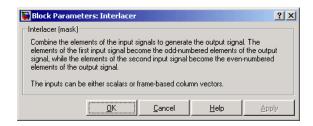


The Interlacer block accepts two inputs that have the same vector size, complexity, and sample time. It produces one output vector by alternating elements from the first input (labeled 0 for odd) and from the second input (labeled E for even). As a result, the output vector size is twice that of either input. The output vector has the same complexity and sample time of the inputs.

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 documentation)

#### **Purpose**

Create complex baseband model of signal impairments caused by imbalances between in-phase and quadrature receiver components

### Library

**RF** Impairments

# **Description**

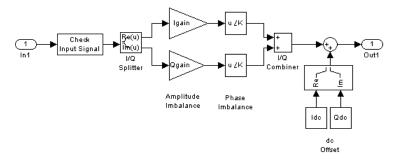


The I/Q Imbalance block creates a complex baseband model of the signal impairments caused by imbalances between in-phase and quadrature receiver components. Typically, these are caused by differences in the physical channels for the two components of the signal.

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**:



Let

 $I_a = I/Q$  amplitude imbalance

 $I_p = I/Q$  phase imbalance

 $I_{DC}$  = inphase DC offset

 $Q_{DC}$  = quadrature DC offset

Also let  $x = x_r + x_i$  be the complex input to the block, with  $x_r$  and  $x_i$  being the real and imaginary parts, respectively, of x. Let y be the complex output of the block.

Then, for an I/Q amplitude imbalance,  $I_a$ 

$$y_{\substack{Amplitude Imbalance \\ \mathbf{j} \star \mathbf{y}_{\substack{I, Amplitude Imbalance}}} = \begin{bmatrix} \mathbf{10}^{(0.5 * \frac{I_a}{20})} * x_r \end{bmatrix} + j [\mathbf{10}^{(-0.5 * \frac{I_a}{20})} * x_i] \triangleq y_{\substack{T, Amplitude Imbalance}} + j \mathbf{y}_{\substack{T, Amplitud Imbalance}} + j \mathbf{y}_{\substack{T, Amplitude Imbalance}} + j \mathbf{y}_{\substack{T,$$

For an I/Q phase imbalance  $I_p$ 

 $y_{PhaseImbalance} =$ 

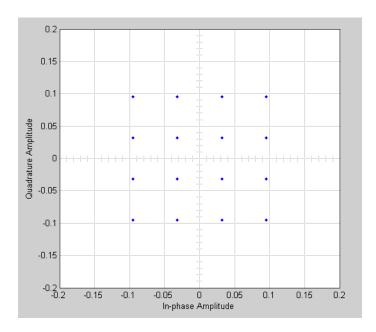
$$\begin{split} & [\exp(-0.5*j*\pi*\frac{I_p}{180})*y_{\mathit{r_{Amplitude\,Im\,balance}}}] + j\{\exp[j(\frac{\pi}{2}+0.5*\pi*\frac{I_p}{180})]*y_{i_{Amplitude\,Im\,balance}}\} \\ & \triangleq y_{\mathit{rPhaseImbalance}} + j*y_{\mathit{i_{PhaseImbalance}}} + j*y_{\mathit{i_{PhaseImbalance}}}\} \end{split}$$

For DC offsets 
$$I_{DC}$$
 and  $Q_{DC}$  
$$y_{PhaseImbalance} = y_{r \; PhaseImbalance} + j^*y_{i \; PhaseImbalance}$$
 
$$y = y_{r \; PhaseImbalance} + I_{DC} + j \; *y_{i \; PhaseImbalance} + Q_{DC}$$

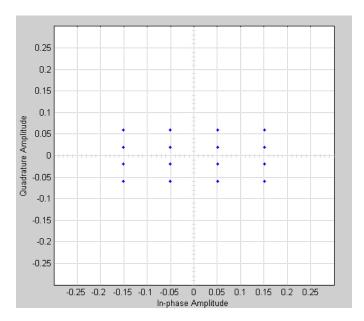
The value of the **I/Q** amplitude imbalance (dB) parameter is divided between the in-phase and quadrature components:

- 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:

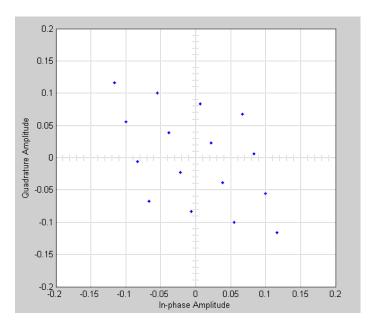


The following figure shows a scatter plot of an output signal, modulated by 16-ary QAM, from the I/Q block with I/Q amplitude imbalance (dB) set to 8 and all other parameters set to 0:

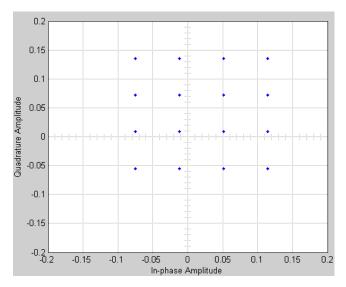


Observe that the scatter plot is stretched horizontally and compressed vertically compared to the undistorted constellation.

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:

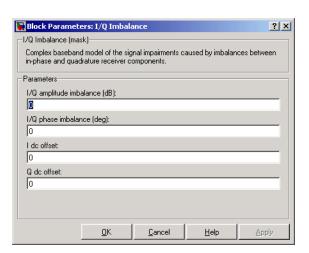


Setting the I dc offset to 0.02 and the Q dc offset to 0.04 shifts the constellation 0.02 to the right and 0.04 up, as shown below:



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** 

Kasami Sequence Generator 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  $N=2^{\rm n}$  - 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  $2^{\rm n/2}$  +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.

$$K_s(u,n,m) = \begin{cases} u & m = -1 \\ u \oplus T^m w & m = 0,...,2^{n/2} - 2 \end{cases}$$

## Small Set of Kasami Sequences for n Even

Note that the small set contains  $2^{n/2}$  sequences.

For 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  $2^{n/2+1}+1$ . The large set is defined by the following table, in which k and m are the shift parameters for the sequences v and w, respectively.

$$K_L(u,n,k,m) = \begin{cases} u & k = -2; \ m = -1 \\ v & k = -1; \ m = -1 \\ u \oplus T^k v & k = 0,...,2^n - 2; \ m = -1 \\ u \oplus T^m w & k = -2; \ m = 0,...,2^{n/2} - 2 \\ v \oplus T^m w & k = -1; \ m = 0,...,2^{n/2} - 2 \\ u \oplus T^k v \oplus T^m w & k = 0,...,2^n - 2; \ m = 0,...,2^{n/2} - 2 \end{cases}$$

### 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 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.

• 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  $[-1, ..., 2^{n/2} - 2]$ . 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 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  $[-2, ..., 2^n - 2]$ , and the range for m is  $[-1, ..., 2^{n/2} - 2]$ . The following table describes the output sequences corresponding to **Sequence index**  $[k \ m]$ :

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]	$k = -2$ $m = 0, 1,, 2^{n/2} - 2$	$u \oplus T^m w$
[-1 m]	$m = 0, 1,, 2^{m2} - 2$ $k = -1$ $m = 0,, 2^{n/2} - 2$	$v \oplus T^m w$
[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 2-605 for an example.

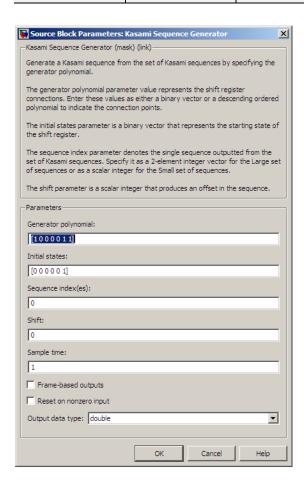
## **Polynomials for Generating Kasami Sequences**

The following table lists some of the polynomials that you can use to generate the Kasami set of sequences.

n	N	Polynomial	Set
4	15	[4 1 0]	Small
6	63	[6 1 0]	Large

n	N	Polynomial	Set
8	255	[8 4 3 2 0]	Small
10	1023	[10 3 0]	Large
12	4095	[12 6 4 1 0]	Small

## 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

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

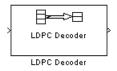
### **Purpose**

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.

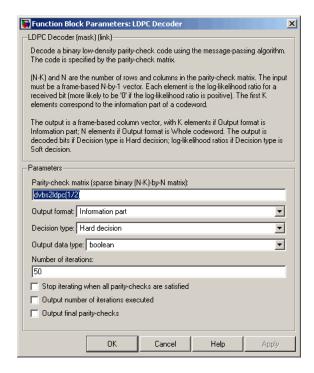
(n-k) and n are the number of rows and columns, respectively, in the parity-check matrix.

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 parameter accepts a sparse matrix with dimension  $n \cdot k$  by n (where n > k > 0) of real numbers. All nonzero elements must be equal to 1. The upper bound limit for the value of n is  $2^{31}$ -1

#### **Output** format

The 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.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
Out	Double-precision floating point
	Boolean when <b>Decision type</b> is Hard decision

### **Pair Block**

dvbs2ldpc and LDPC Encoder

## References

[1] Gallager, Robert G., Low-Density Parity-Check Codes, Cambridge, MA, MIT Press, 1963.

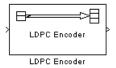
### **Purpose**

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.

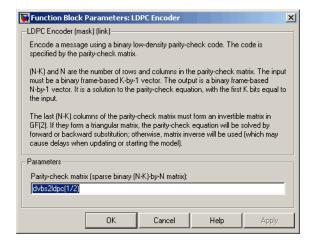
Both the input and the output are discrete-time signals. The ratio of the output sample time to the input sample time is k/n. The input must be a real  $k\times 1$  frame-based signal.

The output inherits the data type of the input, and the input must be binary-valued (0 or 1). For information about the data types each block port supports, see the "Supported Data Type" on page 2-388 table on this page.

Type commdvbs2 at the command line to see a demo using this block.

**Note** Model initialization or update may take a long time, because a large matrix may need to be inverted (when the last (n - k) columns of the parity-check matrix is not triangular).

## Dialog Box



### Parity-check matrix

This block can accept a sparse matrix with dimension  $n \cdot k$  by n (where n > k > 0) of real numbers. All nonzero elements must be equal to 1. The upper bound limit for the value of n is  $2^{31}$ -1

The default value is the parity-check matrix of the half-rate LDPC code from the DVB-S.2 standard.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	• Single-precision floating point
	• Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

Port	Supported Data Types
Out	Double-precision floating point
	Single-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

## **Pair Block**

dvbs21dpc and LDPC Decoder

## **Linearized Baseband PLL**

## **Purpose**

Implement linearized version of baseband phase-locked loop

## Library

Components sublibrary of Synchronization

## **Description**

Linearized Filt Baseband PD PLL VCO The Linearized Baseband PLL block is a feedback control system that automatically adjusts the phase of a locally generated signal to match the phase of an input signal. Unlike the Phase-Locked Loop block, this block uses a baseband model method. Unlike the Baseband PLL block, which uses a nonlinear model, this block simplifies the computations by using x to approximate  $\sin(x)$ . The baseband PLL model depends on the amplitude of the incoming signal but does not depend on a carrier frequency.

This PLL has these three components:

- An integrator used as a phase detector.
- A filter. You specify the filter's transfer function using the Lowpass filter numerator and Lowpass filter denominator parameters.
   Each is a vector that gives the respective polynomial's coefficients in order of descending powers of s.

To design a filter, you can use functions such as butter, cheby1, and cheby2 in Signal Processing Toolbox software. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.

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

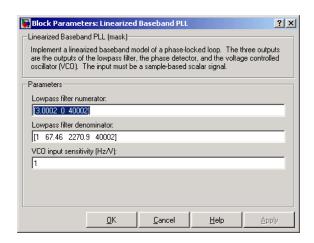
• A voltage-controlled oscillator (VCO). You specify the sensitivity of the VCO signal to its input using the **VCO input sensitivity** parameter. This parameter, measured in Hertz per volt, is a scale factor that determines how much the VCO shifts from its quiescent frequency.

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

## 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*.

### **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*.

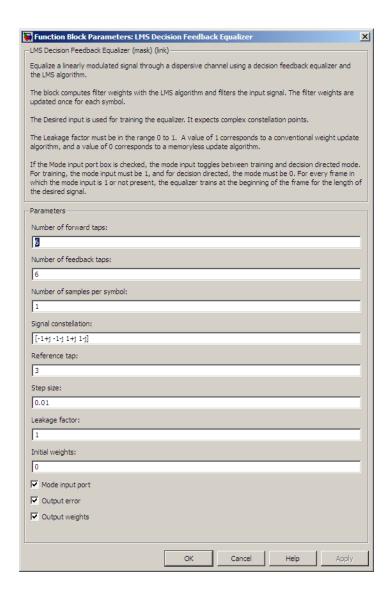
### **Equalizer Delay**

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+(Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

## 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.

### 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

### **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*.

## **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*.

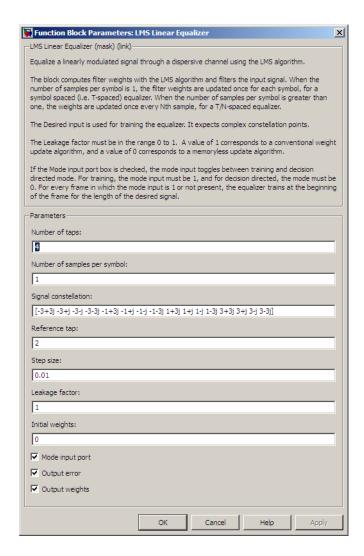
### **Equalizer Delay**

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+(Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## Dialog Box



### Number of taps

The number of taps in the filter of the linear equalizer.

## LMS Linear Equalizer

### Number of samples per symbol

The number of input samples for each symbol.

### Signal constellation

A vector of complex numbers that specifies the constellation for the 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

[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

## **Matrix Deinterleaver**

### **Purpose**

Permute input symbols by filling matrix by columns and emptying it by rows

## Library

Block sublibrary of Interleaving

## **Description**

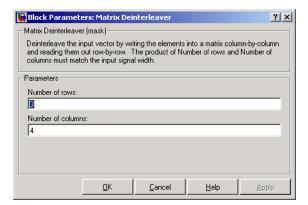


The Matrix Deinterleaver block performs block deinterleaving by filling a matrix with the input symbols column by column and then sending the matrix contents to the output port row by row. The **Number of rows** and **Number of columns** parameters are the dimensions of the matrix that the block uses 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**

#### 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** 

Matrix Helical Scan Deinterleaver 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.

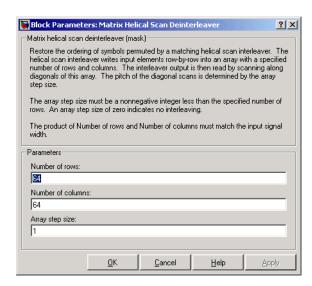
Helical fashion means that the block places input symbols along diagonals of the matrix. The number of elements in each diagonal matches the **Number of columns** parameter, after the block wraps past the edges of the matrix when necessary. The block traverses diagonals so that the row index and column 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.

## 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

## Matrix Helical Scan Interleaver

## **Purpose**

Permute input symbols by selecting matrix elements along diagonals

## Library

Block sublibrary of Interleaving

## **Description**

Matrix > Helical Scan Interleaver 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.

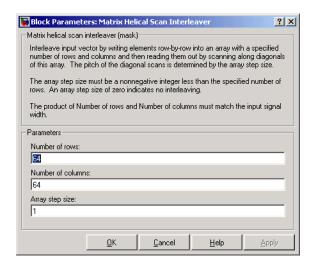
Helical fashion means that the block selects output symbols by selecting elements along diagonals of the matrix. The number of elements in each diagonal matches the **Number of columns** parameter, after the block wraps past the edges of the matrix when necessary. The block traverses diagonals so that 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.

## 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.

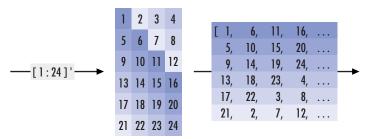
## **Examples**

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

## **Purpose**

Permute input symbols by filling matrix by rows and emptying it by columns

## Library

Block sublibrary of Interleaving

## **Description**



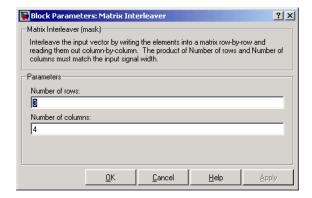
The Matrix Interleaver block performs block interleaving by filling a matrix with the input symbols row by row and then sending the matrix contents to the output port column by column.

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.

# **Matrix Interleaver**

#### 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. For information about the data types each block port supports, see the "Supported Data Types" on page 2-413 table on this page.

The **M-ary number** parameter, M, is the number of possible output symbols that can immediately follow a given output symbol. The block compares the current symbol to the previous symbol. The block's first output is the initial condition of zero (or a group of zeros, if the **Output type** parameter is set to Bit) because there is no previous symbol.

## **Binary or Integer Outputs**

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

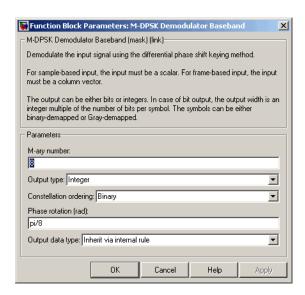
 $\theta + 2\pi k/M$ 

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.

## Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Single-precision floating point
	Boolean when <b>Output type</b> set to Bit
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

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.

# M-DPSK Modulator Baseband

**Purpose** 

Modulate using M-ary differential phase shift keying method

Library

PM, in Digital Baseband sublibrary of Modulation

**Description** 

> M-DPSK

The M-DPSK Modulator Baseband block modulates using the M-ary differential phase shift keying method. The output is a baseband representation of the modulated signal. The **M-ary number** parameter, M, is the number of possible output symbols that can immediately follow a given output symbol.

The input must be a discrete-time signal. For integer inputs, the block can accept the data types int8, uint8, int16, uint16, int32, uint32, single, and double. For bit inputs, the block can accept int8, uint8, int16, uint16, int32, uint32, boolean, single, and double.

## **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

$$\exp\left(j\theta + j2\pi\frac{k}{m}\right)$$
 (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.

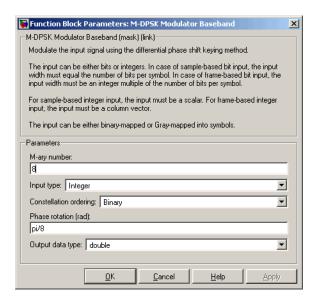
## M-DPSK Modulator Baseband

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]	jθ	jθ
[0 0 1]	$j\theta + j\pi/4$	jθ + jπ/4
[0 1 0]	$j\theta + j\pi 2/4$	jθ + jπ3/4
[0 1 1]	jθ + jπ3/4	$j\theta + j\pi 2/4$
[1 0 0]	$j\theta + j\pi 4/4$	$j\theta + j\pi 7/4$
[1 0 1]	$j\theta + j\pi 5/4$	jθ + jπ6/4
[1 1 0]	jθ + jπ6/4	$j\theta + j\pi 4/4$
[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.

## Supported Data Types

Port	Supported Data Types
Input • Double-precision floating point	
	Single-precision floating point
	Boolean (binary input mode only)
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
Output	Double-precision floating point
	Single-precision floating point

Pair Block M-DPSK Demodulator Baseband

See Also

DBPSK Modulator Baseband, DQPSK Modulator Baseband, M-PSK

Modulator Baseband

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**

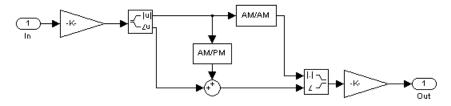
Cubic Polynomial The Memoryless Nonlinearity block applies a memoryless nonlinearity to a complex, baseband signal. You can use the block to model radio frequency (RF) impairments to a signal at the receiver.

**Note** All values of power assume a nominal impedance of 1 ohm.

The Memoryless Nonlinearity block provides five different methods for modeling the nonlinearity, which you specify by the **Method** parameter. The options for the **Method** parameter are

- Cubic polynomial
- Hyperbolic tangent
- Saleh model
- Ghorbani model
- Rapp model

The 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 Subsytem** 

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 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.

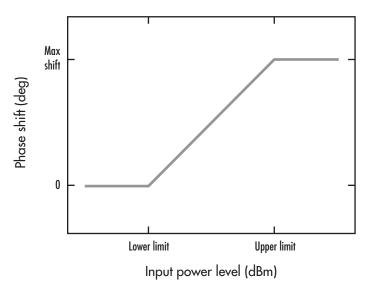
However, the subsystems implement the AM/AM and AM/PM conversions differently, according to the **Method** you specify.

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 2-418.
- **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:

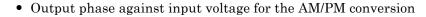


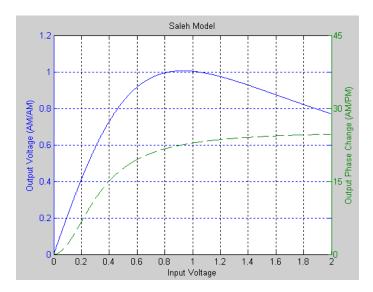
The AM/PM conversion scales linearly with input power value between the lower and upper limits of the input power level (specified by Lower input power limit for AM/PM conversion (dBm) and Upper input power limit for AM/PM conversion (dBm)). Beyond these values, AM/PM conversion is constant at the values corresponding to the lower and upper input power limits, which are zero and (AM/PM conversion) · (upper input power limit – lower input power limit), respectively.

## AM/AM and AM/PM Characteristics of the Saleh Method

The following figure shows, for the Saleh method, plots of

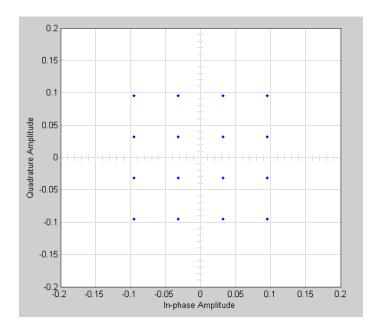
• Output voltage against input voltage for the AM/AM conversion



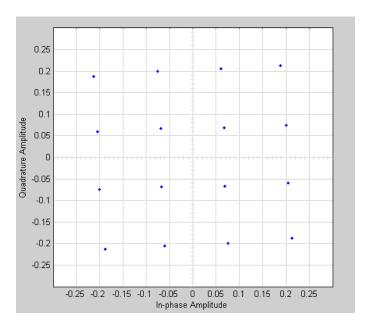


## **Example with 16-ary QAM**

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.



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:

$$F_{AM/PM}(u) = \frac{y_1 u^{y_2}}{1 + y_3 u^{y_2}} + y_4 u$$

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:

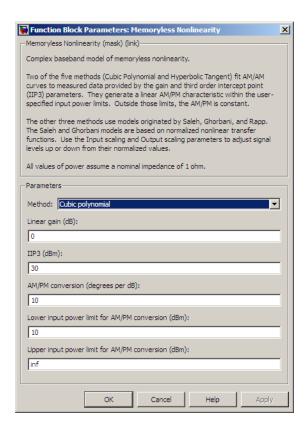
$$F_{AM/AM}(u) = \frac{u}{\left(1 + \left(\frac{u}{O_{sat}}\right)^{2S}\right)^{1/2S}}$$

where u is the magnitude of the scaled signal, S is the **Smoothness** factor, and  $O_{\rm sat}$  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.

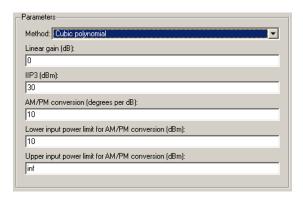
## Dialog Box



#### Method

The nonlinearity method.

The following describes specific parameters for each method.



#### Linear gain (db)

Scalar specifying the linear gain for the output function.

#### IIP3 (dBm)

Scalar specifying the third order intercept.

#### AM/PM conversion (degrees per dB)

Scaler specifying the AM/PM conversion in degrees per decibel.

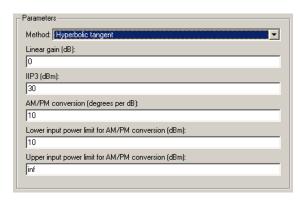
#### Lower input power limit (dBm)

Scalar specifying the minimum input power for which AM/PM conversion scales linearly with input power value. Below this value, the phase shift resulting from AM/PM conversion is zero.

### Upper input power limit (dBm)

Scalar specifying the maximum input power for which AM/PM conversion scales linearly with input power value. Above this value, the phase shift resulting from AM/PM conversion is constant. The value of this maximum shift is given by:

(AM/PM conversion) · (upper input power limit – lower input power limit)



#### Linear gain (db)

Scalar specifying the linear gain for the output function.

#### IIP3 (dBm)

Scalar specifying the third order intercept.

#### AM/PM conversion (degrees per dB)

Scalar specifying the AM/PM conversion in degrees per decibel.

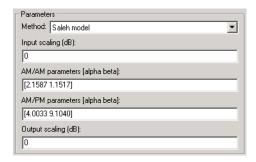
#### Lower input power limit (dBm)

Scalar specifying the minimum input power for which AM/PM conversion scales linearly with input power value. Below this value, the phase shift resulting from AM/PM conversion is zero.

## Upper input power limit (dBm)

Scalar specifying the maximum input power for which AM/PM conversion scales linearly with input power value. Above this value, the phase shift resulting from AM/PM conversion is constant. The value of this maximum shift is given by:

 $(AM\!/\!PM\ conversion) \cdot (upper\ input\ power\ limit-lower\ input\ power\ limit)$ 



#### Input scaling (dB)

Number that scales the input signal level.

#### AM/AM parameters [alpha beta]

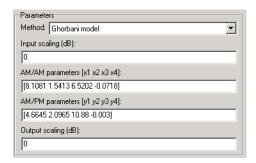
Vector specifying the AM/AM parameters.

#### AM/PM parameters [alpha beta]

Vector specifying the AM/PM parameters.

#### Output scaling (dB)

Number that scales the output signal level.



### Input scaling (dB)

Number that scales the input signal level.

## AM/AM parameters [x1 x2 x3 x4]

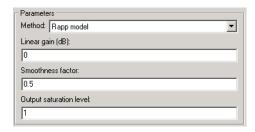
Vector specifying the AM/AM parameters.

#### AM/PM parameters [y1 y2 y3 y4]

Vector specifying the AM/PM parameters.

#### Output scaling (dB)

Number that scales the output signal level.



#### 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 Digitial Sound Broadcasting System", in Proceedings of the Second

European Conference on Satellite Communications, Liege, Belgium, Oct. 22-24, 1991, pp. 179-184.

## M-FSK Demodulator Baseband

**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.

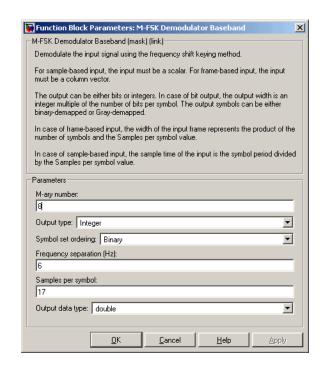
For boolean typed integer outputs, the M-ary number parameter must be 2. For Bit type outputs, the outputs must be of type boolean or double.

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

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.

# M-FSK Demodulator Baseband

#### 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.

## Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Boolean
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers

**Pair Block** 

M-FSK Modulator Baseband

**See Also** 

CPFSK Demodulator 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:

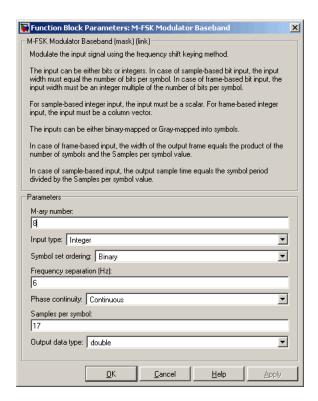
• If **Input type** is set to **Integer**, then the block accepts integers. The input can be either a scalar or a frame-based column vector of type int8, uint8, int16, uint16, int32, uint32, or a double with an integer value. They can also be boolean if the size of the alphabet is 2 (i.e. M = 2).

# M-FSK Modulator Baseband

- 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.

## 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.

# M-FSK Modulator Baseband

#### 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.

## Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Boolean (bit input mode only)
	• 8-, 16-, and 32-bit signed integers (integer input mode only)
	• 8-, 16-, and 32-bit unsigned integers (integer input mode only)
Output	Double-precision floating point
	Single-precision floating point

## **Pair Block**

M-FSK Demodulator Baseband

## See Also

CPFSK Modulator Baseband

**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 kth complex number in the vector is the constellation point to which the modulator maps the integer k-1.

**Note** The sequence of constellation points must be consistent between the modulator in your model and the **Signal constellation** parameter in this block.

# **MLSE Equalizer**

For example, to specify the constellation given by the mapping

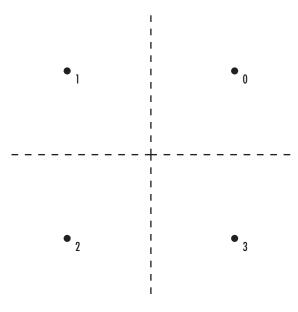
$$0 \rightarrow +1 + i$$

$$1 \rightarrow -1 + i$$

$$2 \rightarrow -1 - i$$

$$3 \rightarrow +1-i$$

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 kth 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 software using the Matrix Concatenation block.

To learn how the block uses the preamble and postamble, see ""Reset Every Frame" Operation Mode" on page 2-441 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

# **MLSE Equalizer**

block decodes the preamble and assigns a metric of 0 to the decoded state. If the preamble does not decode to a unique state -- that is, if the length of the preamble is less than the channel memory -- the decoder assigns a metric of 0 to all states that can be represented by the preamble. Whenever you specify a preamble, the traceback path ends at one of the states represented by the preamble.

- If you do not specify a postamble, the traceback path starts at the state with the smallest metric.
- If you specify a postamble, the traceback path begins at the state represented by the postamble. If the postamble does not decode to a unique state, the decoder identifies the smallest of all possible decoded states that are represented by the postamble and begins traceback decoding at that state.

**Note** In Reset every frame mode, the input to the MLSE Equalizer block must contain at least T symbols, not including an optional preamble, where T is the **Traceback depth** parameter.

## **Continuous Operation Mode**

An alternative way that the Viterbi algorithm can transition between successive frames is called Continuous with reset option mode. You can choose this mode using the **Operation mode** parameter.

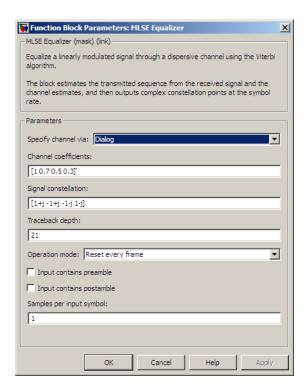
In Continuous with reset option mode, the block initializes the metrics of all states to 0 at the beginning of the simulation. At the end of each frame, the block saves the internal state metric for use in computing the traceback paths in the next frame.

If you select 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.

## 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.

# **MLSE Equalizer**

#### 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

# **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.

# **MER Measurement**

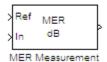
### **Purpose**

Measure signal-to-noise ratio (SNR) in digital modulation applications

# Library

Utility Blocks

# **Description**



The Modulation Error Ratio (MER) is a measure of the signal-to-noise ratio (SNR) in digital modulation applications. You can use these types of measurements to determine system performance in communications applications. For example, determining if an EDGE system conforms to 3GPP radio transmission standards requires accurate MER, Minimum MER, and 95th percentile for the MER measurements. The block measures all outputs in decibels (dB).

The MER block receives an ideal input signal (at reference port, Ref) and an AWGN corrupted signal (at input port, In). The MER block then outputs a measure of the modulation accuracy by comparing these inputs. The Modulation Error Ratio is the ratio of the average reference signal power to the mean square error. This ratio corresponds to the SNR of the AWGN channel.

The block output defaults to MER in decibels (dB), with an option of Output minimum MER or Output X-percentile MER values. The minimum MER represents the best-case MER value per burst. For the X-percentile option, you can select to output the number of symbols processed in the percentile computations.

The following table shows the output type, the activation (what selects the output computation), computation units, and the corresponding computation duration.

Output	Activation	Units	Computation Duration
MER	Default	Decibels	Per burst
Min MER	Parameter setting	Decibels	Per burst

Output	Activation	Units	Computation Duration
Percentile MER	Parameter setting	Decibels	Continuous
Number of symbols	Parameter setting if you select Output X-percentile MER	None	Continuous

#### **Dimension**

The block computes measurements for bursts of data. The data must be of length N symbols, where N is the size of the burst (sample-based or frame-based). The block computes a unique output for each incoming burst; therefore, the computation duration is per burst.

#### **Input Signals**

The input signals must be 1-D or 2-D sample-based column vectors or 2-D frame-based column vectors. The input and reference signals must have identical dimensions.

### **Output Signals**

The output is always a scalar value.

### **Data Type**

The block accepts double, single, and fixed-point data types. The output of the block is always double type.

### **Algorithms**

MER is a measure of the SNR in a modulated signal calculated in dB. MER over N symbols is

$$MER = 10 * log_{10} \left( \frac{\sum\limits_{n=1}^{N} \left(I_{k}^{2} + Q_{k}^{2}\right)}{\sum\limits_{n=1}^{N} \left(e_{k}\right)} \right) dB$$

The MER for the  $k_{th}$  symbol is

$$MER_{k} = 10 * log_{10} \left( \frac{\frac{1}{N} \sum_{n=1}^{N} \left( I_{k}^{2} + Q_{k}^{2} \right)}{e_{k}} \right) db$$

The minimum MER represents the minimum MER value in a burst or

$$MER_{\min} = \min_{k \in [1,...,N]} \{MER_k\}$$
 where

$$e_k = (I_k - \tilde{I_k})^2 + (Q_k - \tilde{Q}_k)^2$$

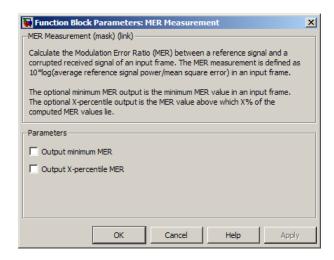
 $I_k$  = In-phase measurement of the kth symbol in the burst

 $Q_k$  = Quadrature phase measurement of the kth symbol in the burst

 $I_k$  and  $Q_k$  represent ideal (reference) values.  $\tilde{I}_k$  and  $\tilde{Q}_k$  represent measured (received) symbols.

The block computes X-percentile MER by creating a histogram of all the incoming  $MER_k$  values. The output provides the MER value above which X% of the MER values lay.

# Dialog Box



#### **Output Minimum MER**

Outputs the minimum MER of an input vector or frame.

#### Output X-percentile MER

Enables an output X-percentile MER measurement. When you select this option, specify **X-percentile value** (%).

#### X-Percentile value (%)

This parameter only appears when you select **Output X-percentile MER**. The Xth percentile is the MER value above which X% of all the computed MER values lie. The parameter defaults to the 95th percentile. Therefore, 95% of all MER values are above this output.

### Output the number of symbols processed

Outputs the number of symbols that the block uses to compute the **Output X-percentile MER**. This parameter only appears when you select **Output X-percentile MER**.

#### **See Also** EVM Measurement

# **MER Measurement**

### References

[1]  $Digital\ Video\ Broadcasting\ (DVB)$ : Measurement guidelines for DVB systems, DVB (ETSI) Standard ETR290, May 1997.

**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 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. For information about the data types each block port supports, see the "Supported Data Types" on page 2-459 table on this page.

**Note** All values of power assume a nominal impedance of 1 ohm.

### **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.

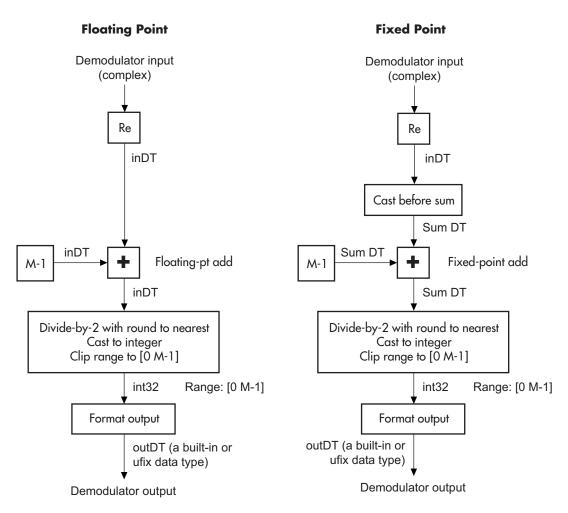
# **Algorithm**

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.

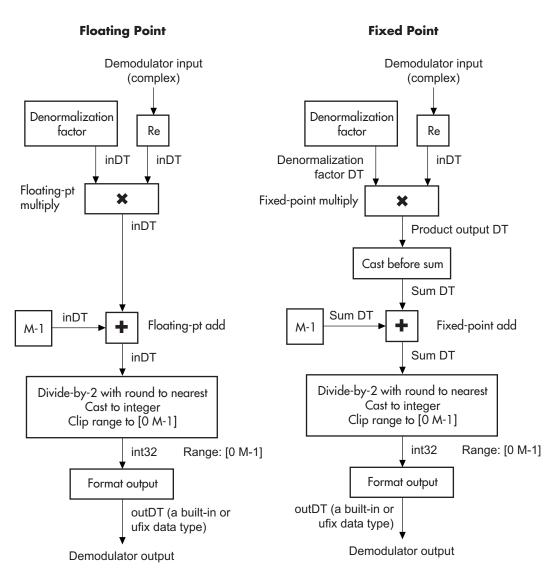
The integer symbol index computation is performed by first scaling the real part of the input signal constellation (possibly with noise) by

a denormalization factor derived from the **Normalization method** and related parameters. This denormalized value is added to M-1 to translate it into an approximate range between 0 and 2 x (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).

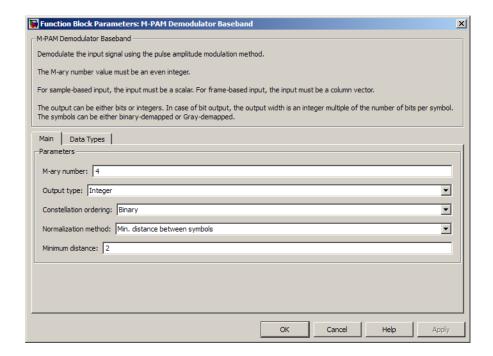


Signal-Flow Diagrams with Denormalization Factor Equal to 1



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.

#### Minimum distance

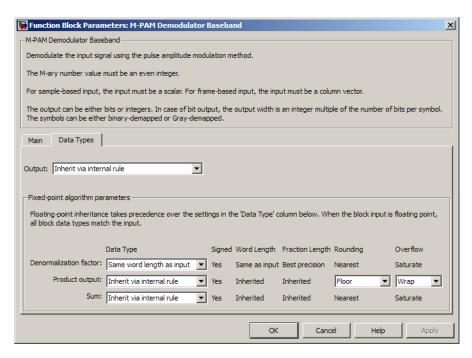
The distance between two nearest constellation points. This field appears only when **Normalization method** is set to Min. distance between symbols.

#### Average power, referenced to 1 ohm (watts)

The average power of the symbols in the constellation, referenced to 1 ohm. This field appears only when **Normalization method** is set to Average Power.

#### Peak power, referenced to 1 ohm (watts)

The maximum power of the symbols in the constellation, referenced to 1 ohm. This field appears only when **Normalization** method is set to Peak Power.



#### Output

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 model. If ASIC/FPGA is selected in the **Hardware Implementation** pane, the output data type is the ideal minimum size, i.e., ufix(1) for bit outputs, and ufix(ceil(log2(M))) for integer outputs. For all other selections, it is an unsigned integer with the smallest available word length large enough to fit the ideal minimum size, usually corresponding to the size of a char (e.g., uint8).

For integer outputs, this parameter can be set to Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, single, and double. For bit outputs, the options are Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, boolean, single, or double.

#### **Denormalization factor**

This parameter applies when a fixed-point input is not normalized. It can be set to Same word length as input or Specify word length, in which case a field is enabled for user input. A best-precision fraction length is always used.

#### **Product output**

This parameter only applies when the input is a fixed-point signal and there is a nonunity (not equal to 1) denormalized factor. It can be set to Inherit via internal rule or Specify word length, which enables a field for user input.

Setting to Inherit via internal rule computes the full-precision product word length and fraction length. Inherit Rule for Product Data Types in *Signal Processing Blockset User's Guide* describes the full-precision Product output internal rule.

Setting to Specify word length allows you to define the word length. The block computes a best-precision fraction length based on the word length specified and the pre-computed worst-case (min/max) real world value **Product output** result. The worst-case **Product output** result is precomputed by multiplying the denormalized factor with the worst-case (min/max) input signal range, purely based on the input signal data type.

The block uses the **Rounding** method when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink Fixed Point documentation.

#### Sum

This parameter only applies when the input is a fixed-point signal. It can be set to Inherit via internal rule, Same as product output, or Specify word length, in which case a field is enabled for user input

Setting Inherit via internal rule computes the full-precision sum word length and fraction length, based on the two inputs to the Sum in the fixed-point Hard Decision Algorithm signal flow diagram. The rule is the same as the fixed-point inherit rule of the internal **Accumulator data type** parameter in the Simulink Sum block.

Setting Specify word length allows you to define the word length. A best precision fraction length is computed based on the word length specified in the pre-computed maximum range necessary for the demodulated algorithm to produce accurate

results. The signed fixed-point data type that has the best precision fully containing the values in the range 2 \* (M-1) for the specified word length.

Setting to Same as product output allows the Sum data type to be the same as the **Product output** data type (when **Product output** is used). If the **Product output** is not used, then this setting will be ignored and the Inherit via internal rule Sum setting will be used.

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	Signed fixed-point
Output	Double-precision floating point
	Single-precision floating point
	• Boolean when Output type is Bit
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	• ufix(1) in ASIC/FPGA when <b>Output type</b> is Bit
	• $\mathit{ufix}ig(\lceil\log_2 M ceilig)$ in ASIC/FPGA when $oldsymbol{Output}$ type is Integer

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** 

M-PAM

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** All values of power assume a nominal impedance of 1 ohm.

### **Constellation Size and Scaling**

Baseband M-ary pulse amplitude modulation using the block's default signal constellation maps an integer m between 0 and M-1 to the complex value

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.

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.
- 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 information about the data types each block port supports, see the "Supported Data Types" on page 2-465 table on this page.

The **Constellation ordering** parameter indicates how the block assigns binary words to points of the signal constellation.

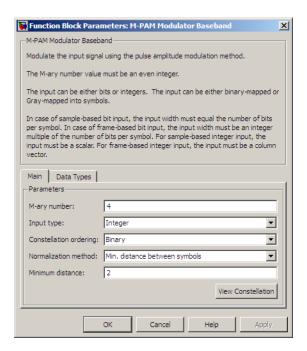
- If Constellation ordering is set to Binary, then the block uses a natural binary-coded constellation.
- If Constellation ordering is set to Gray, then the block uses a Gray-coded constellation.

For details about the Gray coding, see the reference page for the M-PSK Modulator Baseband block.

#### **Constellation Visualization**

The M-PAM Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the *Communications Blockset User's Guide*.

### 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.

#### Normalization method

Determines how the block scales the signal constellation. Choices are Min. distance between symbols, Average Power, and Peak Power.

#### Minimum distance

The distance between two nearest constellation points. This field appears only when **Normalization method** is set to Min. distance between symbols.

#### Average power, referenced to 1 ohm (watts)

The average power of the symbols in the constellation, referenced to 1 ohm. This field appears only when **Normalization method** is set to Average Power.

#### Peak power, referenced to 1 ohm (watts)

The maximum power of the symbols in the constellation, referenced to 1 ohm. This field appears only when **Normalization** method is set to Peak Power.

#### Output data type

The output data type can be set to double, single, Fixed-point, User-defined, or Inherit via back propagation.

Setting this parameter to Fixed-point or User-defined enables fields in which you can further specify details. Setting this parameter to Inherit via back propagation, sets the output data type and scaling to match the following block.

#### Output word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed-point for the **Output data type** parameter.

#### User-defined data type

Specify any signed built-in or signed fixed-point data type. You can specify fixed-point data types using the sfix, sint, sfrac, and fixdt functions from Simulink Fixed Point software. This parameter is only visible when you select User-defined for the **Output data type** parameter.

#### Set output fraction length to

Specify the scaling of the fixed-point output by either of the following methods:

- Choose Best precision to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose User-defined to specify the output scaling in the **Output fraction length** parameter.

This parameter is only visible when you select Fixed-point for the **Output data type** parameter or when you select User-defined and the specified output data type is a fixed-point data type.

#### Output fraction length

For fixed-point output data types, specify the number of fractional bits, or bits to the right of the binary point. This parameter is only visible when you select Fixed-point or User-defined for the **Output data type** parameter and User-defined for the **Set output fraction length to** parameter.

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	Boolean when M-ary number is 2 or Input type is Bit
	• 8-, 16-, 32-bit signed integers
	• 8-, 16-, 32-bit unsigned integers
	• $\mathit{ufix}(\lceil \log_2 M \rceil)$ when $\mathbf{Input}\ \mathbf{type}\ \mathrm{is}\ Integer$
Output	Double-precision floating point
	Single-precision floating point
	Signed fixed-point

Pair Block M-PAM Demodulator Baseband

See Also General QAM Modulator Baseband

**Purpose** 

Demodulate PSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

**Description** 

WWTLT M-PSK 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 **M-ary number** parameter, M, is the number of points in the signal constellation. The input can be either a scalar or a frame-based column vector. For information about the data types each block port supports, see the "Supported Data Types" on page 2-476 table on this page.

### **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.

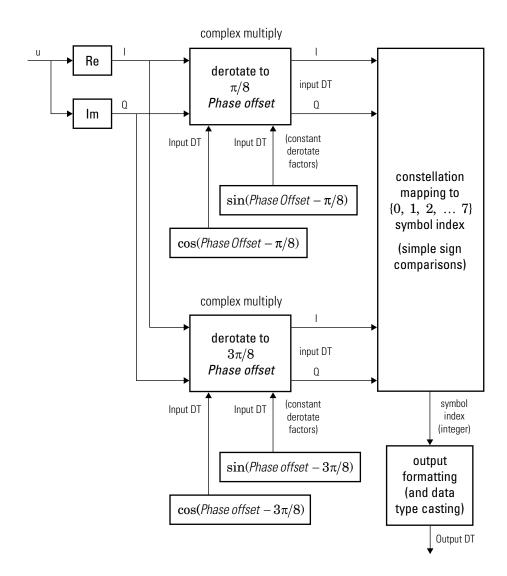
If the **Output type** parameter is set to Bit and **Decision type** is set to Hard decision, the block outputs binary representations of integers between 0 and M-1. It outputs a group of K bits, called a *binary word*, for each symbol.

**Algorithm** 

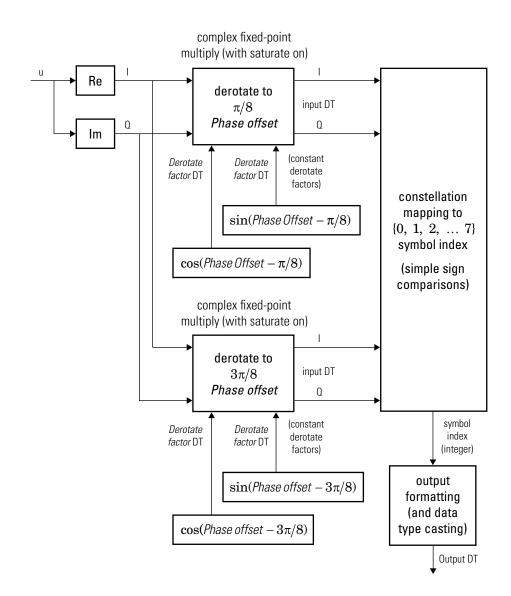
For M=2, refer to the BPSK Demodulator Baseband block reference page.

For M=4, refer to the QPSK Demodulator Baseband block reference page.

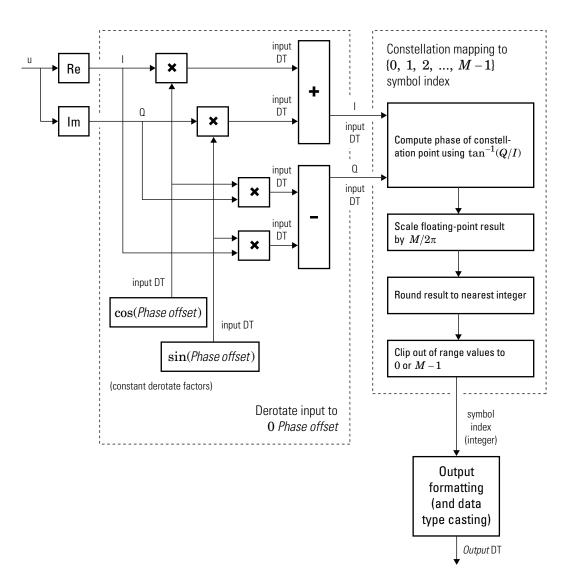
For M=8 and greater, see the following signal diagrams.



Hard-Decision 8-PSK Demodulator Floating-Point Signal Diagram



Hard-Decision 8-PSK Demodulator Fixed-Point Signal Diagram



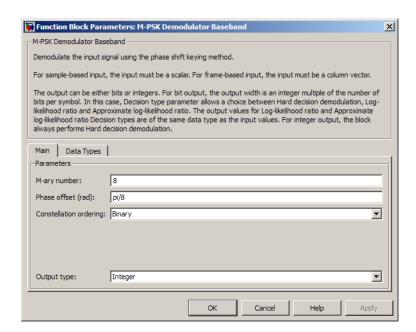
Hard-Decision M-PSK Demodulator (M > 8) Floating-Point Signal Diagram for Nontrivial Phase Offset

For M > 8, in order to improve speed and implementation costs, no derotation arithmetic is performed when **Phase offset** is 0,  $\pi/2$ ,  $\pi$ , or  $3\pi/2$  (i.e., when it is trivial).

Also, for M > 8, this block will only support inputs of type double and single.

The exact LLR and approximate LLR cases (soft-decision) are described in "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the *Communications Toolbox User's Guide*.

### Dialog Box



### M-ary number

The number of points in the signal constellation.

#### Phase offset

The phase of the zeroth point of the signal constellation.

#### Constellation ordering

Determines how the block maps 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.

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" in the *Communications Toolbox User's Guide* 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

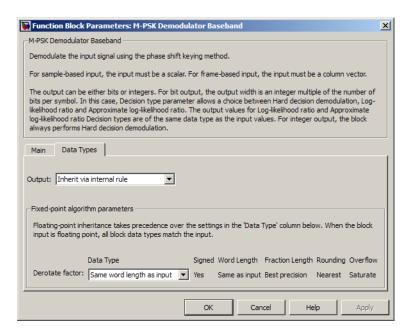
This parameter appears when the **Noise variance source** is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:

- Inf to -Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.



#### **Data Types Pane for Hard-Decision**

#### Output

For bit outputs, when **Decision type** is set to Hard decision, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, uint32, or boolean.

For integer outputs, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, or uint32.

When this parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is a floating-point type (single or double). If the input data type is fixed-point (supported only when **M-ary** 

**number** is 2, 4, or 8), the output data type will work as if this parameter is set to 'Smallest unsigned integer'.

When this parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the model.

If ASIC/FPGA is selected in the **Hardware Implementation** pane, and **Output type** is Bit, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

If ASIC/FPGA is selected in the **Hardware Implementation** pane, and **Output type** is Integer, the output data type is the ideal minimum integer size, i.e., ufix(ceil(log2(M))). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit the ideal minimum size, usually corresponding to the size of a char (e.g., uint8).

#### **Derotate factor**

This parameter only applies when **M-ary number** is 2, 4, or 8, the input is fixed-point, and **Phase offset** is nontrivial (multiple

of 
$$\frac{\pi}{2}$$
 for M=2, odd multiple of  $\frac{\pi}{4}$  for M=4, or any value for M=8).

This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input.



#### **Data Types Pane for Soft-Decision**

For bit outputs, when **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is inherited from the input (e.g., if the input is of data type double, the output is also of data type double).

# Supported Data Types

Port	Supported Data Types	
Input	Double-precision floating point	
	Single-precision floating point	
	• Signed fixed point only when $M \le 8$ and <b>Decision type</b> is Hard-decision	
Var	Double-precision floating point	
	Single-precision floating point	
Output	Double-precision floating point	
	Single-precision floating point	
	• Boolean when <b>M-ary number</b> is 2	
	• 8-, 16-, and 32-bit signed integers	
	• 8-, 16-, and 32-bit unsigned integers	
	• $\mathit{ufix}(\lceil \log_2 M \rceil)$ when $\mathbf{Output}\ \mathbf{type}\ \mathrm{is}\ Integer$	

# **Pair Block**

M-PSK Modulator Baseband

### **See Also**

BPSK Demodulator Baseband, QPSK Demodulator Baseband, M-DPSK Demodulator 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 information about the data types each block port supports, see the "Supported Data Types" on page 2-484 table on this page.

Alternative configurations of the block determine how the block interprets its input and arranges its output, as explained in the following sections.

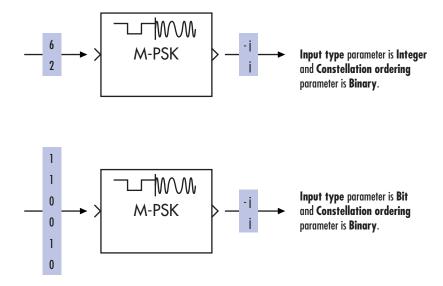
### **Binary Inputs**

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 BlocksetUser'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<sub>2</sub>(M) bits.)

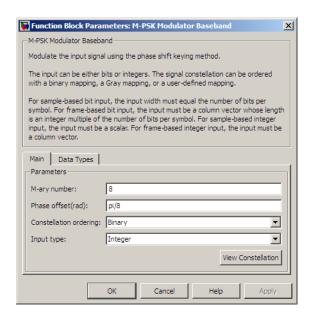
For example, the following schematics illustrate how the block processes two 8-ary integers or binary words in one time step. The signals involved are all frame-based column vectors. In both cases, the **Phase offset** parameter is 0.



#### **Constellation Visualization**

The M-PSK Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the *Communications Blockset User's Guide*.

## Dialog Box



#### M-ary number

The number of points in the signal constellation.

#### Phase offset

The phase of the zeroth point of the signal constellation.

#### Constellation ordering

Determines how the block maps an integer or group of K input bits to the corresponding symbol.

If set to Binary, baseband M-ary phase shift keying modulation with a phase offset of  $\theta$  maps an integer m between 0 and M-1 to the complex value

$$\exp(j\theta + j2\pi m/M)$$

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 2-481.

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 + **Phase offset** angle, with subsequent elements running counterclockwise. The last element corresponds to the -pi/M + **Phase offset** constellation point.

#### 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.

#### 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 software. This parameter is only visible when you select User-defined for the **Output data type** parameter.

#### Output fraction length

Specify the scaling of the fixed-point output by either of the following methods:

- 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.

# **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
```

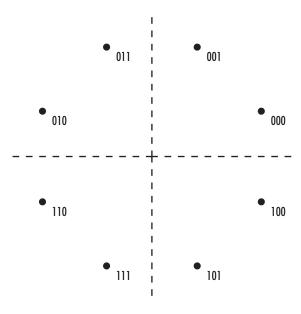
where  $\theta$  is the **Phase offset** parameter and m is an integer between 0 and M-1 that satisfies

$$m \text{ XOR } \lfloor m/2 \rfloor = U$$

For example, if M = 8, the binary representations that correspond to the zeroth through seventh phases are as follows.

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 /8.



# Supported Data Types

Port	Supported Data Types	
Input	Double-precision floating point	
	Single-precision floating point	
	Boolean when Input type is Bit	
	• 8-, 16-, and 32-bit signed integers	
	• 8-, 16-, and 32-bit unsigned integers	
	• $\mathit{ufix}(\lceil \log_2 M \rceil)$ when $\mathbf{Input\ type}$ is $Integer$	
Output	Double-precision floating point	
	Single-precision floating point	
	Signed Fixed point	

**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**

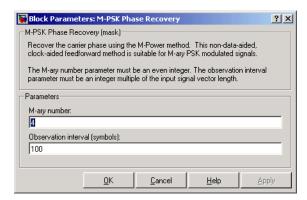
• The output port labeled Ph outputs the carrier phase estimate, in degrees, for all symbols in the observation interval. The Ph output is a scalar signal.

**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,  $\log_2(\mathbf{M-ary\ number})$  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.

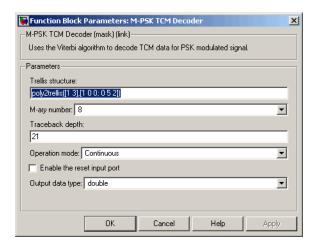
- In Truncated mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In Terminated mode, the block treats each frame independently. The traceback path always starts at the all-zeros 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**.

## 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**

M-PSK TCM The M-PSK TCM Encoder block implements trellis-coded modulation (TCM) by convolutionally encoding the binary input signal and mapping the result to a PSK signal constellation.

The **M-ary number** parameter is the number of points in the signal constellation, which also equals the number of possible output symbols from the convolutional encoder. (That is,  $\log_2(\mathbf{M-ary\ number})$  is equal to n for a rate 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\*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 a binary value.

### **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 apoly2trellis 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)

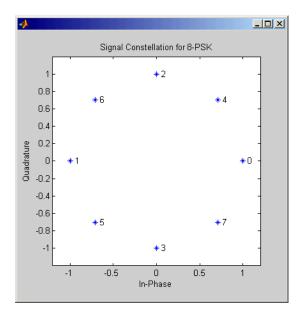
• If you have a variable in the MATLAB workspace that contains the trellis structure, then enter its name as the **Trellis structure** parameter. This way is faster because it causes Simulink software to spend less time updating the diagram at the beginning of each simulation, compared to the usage in the previous bulleted item.

The encoder registers begin in the all-zeros state. You can configure the encoder so that it resets its registers to the all-zeros state during the course of the simulation. To do this, set the Operation mode to Reset on nonzero input via port. The block then opens a second input port, labeled Rst. The signal at the Rst port is a scalar signal. When it is nonzero, the encoder resets before processing the data at the first input port.

### **Signal Constellations**

The trellis-coded modulation technique partitions the constellation into subsets called cosets, so as to maximize the minimum distance between pairs of points in each coset. This block internally forms a valid partition based on the value you choose for the **M-ary number** parameter.

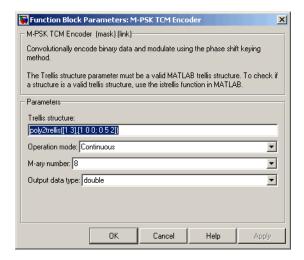
The figure below shows the labeled set-partitioned signal constellation that the block uses when **M-ary number** is 8. For constellations of other sizes, see [1].



# **Coding Gains**

Coding gains of 3 to 6 decibels, relative to the uncoded case can be achieved in the presence of AWGN with multiphase trellis codes [3].

## Dialog Box



#### Trellis structure

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

## Operation mode

In Continuous mode (default setting), the block retains the encoder states at the end of each frame, for use with the next frame.

In Truncated (reset every frame) mode, the block treats each frame independently. I.e., the encoder states are reset to all-zeros state at the start of each frame.

In Terminate trellis by appending bits mode, the block treats each frame independently. For each input frame, extra bits are used to set the encoder states to all-zeros state at the end of the frame. The output length is given by

 $y = n \cdot (x+s)/k$ , where x is the number of input bits, and s = constraint length - 1 (or, in the case of multiple constraint

# M-PSK TCM Encoder

lengths, s = sum(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.

#### 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, and the model uses a variable-step solver or a fixed-step solver with the **Tasking Mode** parameter set to SingleTasking, then the delay consists of D+1 zero symbols.
- If the input signal is frame-based, then the delay consists of D zero symbols.

The optimal **Traceback length** parameter value is dependent on minimum squared Euclidean distance calculations. Alternatively, a typical value, dependent on the number of states, can be chosen using the "five-times-the-constraint-length" rule, which corresponds

to  $5 \cdot \log 2(numStates)$ . The number of states is determined by the following equation:

$$numStates = egin{cases} p \cdot M^{(L-1)}, for \, even \, m \\ 2 \, p \cdot M^{(L-1)}, for \, odd \, m \end{cases}$$
 where:

# **MSK Demodulator Baseband**

- h = m/p is the modulation index proper rational form
  - $\blacksquare$  *m* = numerator of modulation index
  - = p =denominator of modulation index
- *L* is the Pulse length
- *M* is M-ary number

For a binary raised cosine pulse shape with a pulse length of 3, h=2/3,

this rule  $(5*\log 2(3*2^2)=18)$  gives a result close to the optimum value of 20.

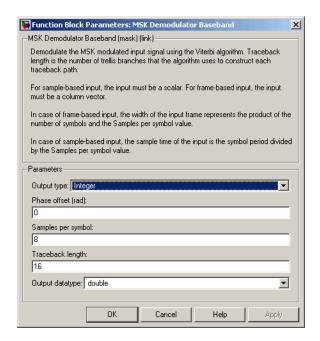
## **Inputs and Outputs**

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 MSK Demodulator Baseband block uses to construct each traceback path.

### Output data type

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

# **MSK Demodulator Baseband**

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	• Boolean
	• 8-, 16-, and 32-bit signed integers

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.

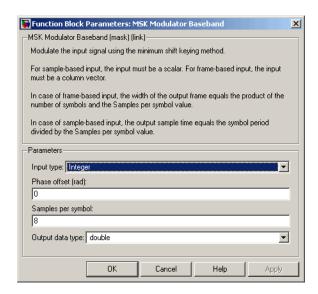
### **Input Attributes**

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. For information about the data types each block port supports, see the "Supported Data Types" on page 2-503 table on this page.

### **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.

#### 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.

# Supported Data Types

Port	Supported Data Types	
Input	Double-precision floating point	
	Boolean (bit input mode only)	
	• 8-, 16-, and 32-bit signed integers (integer input mode only)	
Output	Double-precision floating point	
	Single-precision floating point	

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.

### **Purpose**

Recover symbol timing phase using fourth-order nonlinearity method

## Library

Timing Phase Recovery sublibrary of Synchronization

## **Description**

MSK Signal Sym > Timing Recovery Ph 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\*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

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\*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.

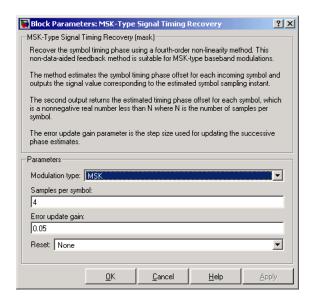
The Ph output contains nonnegative real numbers less than N. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time or frame period of the Ph output is the same as that of the Symoutput.

**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.

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink *User's Guide*.

#### Reset

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are None, Every frame, and On nonzero input via port. The last option causes the block to have a second input port, labeled Rst.

# **Algorithm**

This block's algorithm extracts timing information by passing the sampled baseband signal through a fourth-order nonlinearity followed by a digital differentiator whose output is smoothed to yield an error signal. The algorithm then uses the error signal to make the sampling adjustments.

More specifically, this block uses a timing error detector whose result for the kth symbol is e(k), given in [2] by

$$\begin{split} e(k) = & \; (-1)^{D+1} \operatorname{Re}\{r^2(kT - T_s + d_{k-1})r^{*2}((k-1)T - T_s + d_{k-2})\} \\ - & \; (-1)^{D+1} \operatorname{Re}\{r^2(kT + T_s + d_{k-1})r^{*2}((k-1)T + T_s + d_{k-1})\} \end{split}$$

where

- r is the block's input signal
- T is the symbol period
- T<sub>s</sub> is the sampling period
- \* means complex conjugate
- $\boldsymbol{d}_k$  is the phase estimate for the kth symbol
- D is 1 for MSK and 2 for Gaussian MSK modulation

For more information about the role that e(k) plays in this block's algorithm, see "Feedback Methods for Timing Phase Recovery" in *Communications 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**

Mueller-Muller Sym >
Timing Recovery Ph

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\*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\*R, then the Sym output is a frame-based column vector of length R having the same frame period.

# **Mueller-Muller Timing Recovery**

- 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.

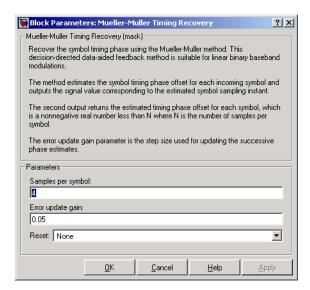
The Ph output contains nonnegative real numbers less than N. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time or frame period of the Ph output is the same as that of the Symoutput.

**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.

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink *User's Guide*.

#### Reset

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are None, Every

# **Mueller-Muller Timing Recovery**

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 kth symbol is e(k), given by

$$e(k) = \operatorname{Re}\{c_{k-1}^*y(kT+d_k) - c_k^*y((k-1)T+d_{k-1})\}$$

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
- $\bullet$  d<sub>k</sub> is the phase estimate for the kth symbol

For more information about the role that e(k) plays in this block's algorithm, see "Feedback Methods for Timing Phase Recovery" in *Communications BlocksetUser'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 µ-law compressor for source coding

## Library

Source Coding

# **Description**



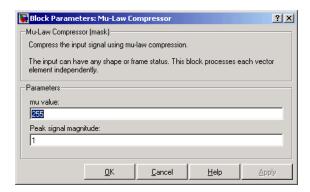
The Mu-Law Compressor block implements a  $\mu\text{-law}$  compressor for the input signal. The formula for the  $\mu\text{-law}$  compressor is

$$y = \frac{V \log(1 + \mu |x|/V)}{\log(1 + \mu)} \operatorname{sgn}(x)$$

where  $\mu$  is the  $\mu$ -law parameter of the compressor, V is the peak magnitude of x, log is the natural logarithm, and sgn is the signum function (sign in MATLAB).

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

### Dialog Box



#### mu value

The µ-law parameter of the compressor.

#### Peak signal magnitude

The peak value of the input signal. This is also the peak value of the output.

# **Mu-Law Compressor**

Supported Data Type

Port	Supported Data Types
In	• double
Out	• double

Pair Block Mu-Law Expander

See Also A-Law Compressor

**References** [1] Sklar, Bernard. Digital Communications: Fundamentals and

Applications. Englewood Cliffs, N.J.: Prentice-Hall, 1988.

## **Purpose**

Implement µ-law expander for source coding

# Library

Source Coding

# **Description**

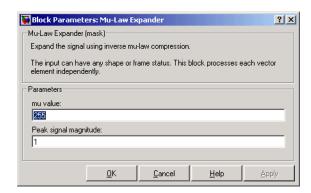


The Mu-Law Expander block recovers data that the Mu-Law Compressor block compressed. The formula for the  $\mu$ -law expander, shown below, is the inverse of the compressor function.

$$x = \frac{V}{\mu} \left( e^{|y|\log(1+\mu)/V} - 1 \right) \operatorname{sgn}(y)$$

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

## Dialog Box



#### mu value

The µ-law parameter of the compressor.

### Peak signal magnitude

The peak value of the input signal. This is also the peak value of the output.

# **Mu-Law Expander**

Supported Data Type

Port	Supported Data Types
In	• double
Out	• double

Pair Block Mu-Law Compressor

See Also A-Law Expander

**References** [1] Sklar, Bernard. Digital Communications: Fundamentals and

Applications. Englewood Cliffs, N.J.: Prentice-Hall, 1988.

### **Purpose**

Simulate multipath Rayleigh fading propagation channel

# Library

Channels

## **Description**



Multipath Rayleigh Fading Channel The Multipath Rayleigh Fading Channel block implements a baseband simulation of a multipath Rayleigh fading propagation channel. You can use this block to model mobile wireless communication systems. For details about fading channels, see the references listed below.

This block accepts 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 block inherits sample time from the input signal. The input signal must have a discrete sample time greater than 0.

Relative motion between the transmitter and receiver causes Doppler shifts in the signal frequency. You can specify the Doppler spectrum of the Rayleigh process using the **Doppler spectrum type** parameter. For channels with multiple paths, you can assign each path a different Doppler spectrum, by entering a vector of doppler objects in the **Doppler spectrum** field.

Because a multipath channel reflects signals at multiple places, a transmitted signal travels to the receiver along several paths, each of which may have differing lengths and associated time delays. In the block's parameter dialog box, the **Discrete path delay vector** specifies the time delay for each path. If you do not check **Normalize gain vector to 0 dB overall gain**, then the **Average path gain vector** specifies the gain for each path. When you check the box, the block uses a multiple of **Average path gain vector** instead of the **Average path gain vector** itself, choosing the scaling factor so that the channel's effective gain, considering all paths, is 0 dB.

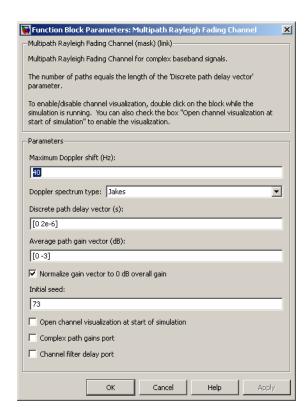
The number of paths indicates the length of **Discrete path delay vector** or **Average path gain vector**, whichever is larger. If both of these parameters are vectors, then they must have the same length; if exactly one of these parameters contains a scalar value, then the block

expands it into a vector whose size matches that of the other vector parameter.

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. Then, the block generates random numbers using the Ziggurat method. This method is the same one used by the MATLAB random function.

Double-clicking this block during simulation or checking the block dialog's check-box 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



### Maximum Doppler shift (Hz)

A positive scalar value that indicates the maximum Doppler shift.

### Doppler spectrum type

Specifies the Doppler spectrum of the Rayleigh process.

This parameter defaults to Jakes Doppler spectrum. Alternatively, you can also choose any of the following types:

- Flat
- Gaussian
- Rounded

- Restricted Jakes
- Asymmetrical Jakes
- Bi-Gaussian
- Bell

For all Doppler spectrum types except Jakes and Flat, you can choose one or more parameters to control the shape of the spectrum.

You can also select Specify as dialog parameter for the **Doppler spectrum type**. Specify the Doppler spectrum by entering an object in the **Doppler spectrum** field. See the doppler function reference in *Communications Toolbox User's Guide* for details on how to construct Doppler objects, and also for the meaning of the parameters associated with the various Doppler spectrum types.

### Discrete path delay vector (s)

A vector that specifies the propagation delay for each path.

### Average path gain vector (dB)

A vector that specifies the gain for each path.

### Normalize gain vector to 0 dB overall gain

Checking this box causes the block to scale the **Gain vector** parameter so that the channel's effective gain (considering all paths) is 0 dB.

#### **Initial** seed

The scalar seed for the Gaussian noise generator.

### Open channel visualization at start of simulation

Checking this box opens the channel visualization tool when a simulation begins.

### Complex path gains port

Checking this box creates a port that outputs the values of the complex path gains for each path. In this N-by-M multichannel

frame, N represents the number of samples per frame and M represents 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 results from the filtering operation of this block. This delay is zero if only one path is simulated, but can be greater than zero if more than one path is present. See "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]. A detailed explanation of the implementation, including a review of the different Doppler spectra, can be found in [4].

Some wireless applications, such as standard GSM (Global System for Mobile Communication) systems, prefer to specify Doppler shifts in terms of the speed of the mobile. If the mobile moves at speed v making an angle of  $\theta$  with the direction of wave motion, then the Doppler shift is

$$f_{\rm d} = (vf/c)\cos\theta$$

where f is the transmission carrier frequency and c is the speed of light. The Doppler frequency represents the maximum Doppler shift arising from motion of the mobile.

# Example

# Generating Ideal Theoretical BER Results for a Rayleigh Fading Channel

This example illustrates how to generate ideal theoretical BER results for a flat Rayleigh fading channel. The model uses reproduces known theoretical results and shows the correct BER performance for a flat Rayleigh fading channel. In this example, you will run the model and compare the simulation results to the BERTool theoretical results for verification purposes. Note that the EbNo value for the model's AWGN block is 5 dB. You can change the noise power by double-clicking

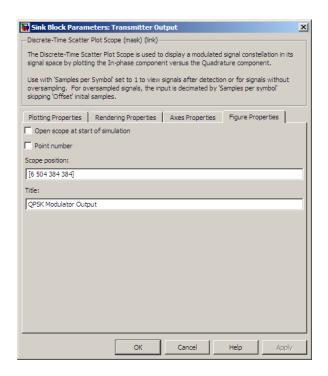
the AWGN block and entering another numeric value in the EbNo parameter.

### **Opening the Model**

You can open the model by clicking here in the MATLAB Help browser. Alternatively, you can type doc\_qpsk\_rayleigh\_derotated at the MATLAB command line.

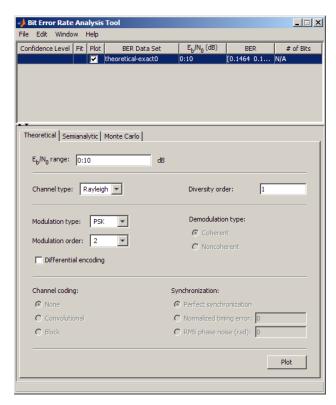
### **Running the Model and Comparing Results**

- 1 You can run the demo by clicking **Simulation > Start**.
- **2** After the model collects more than 5000 errors, click the stop button.
- **3** Close the three scopes.
- 4 In the Simulink model window, double-click the Transmitter Output block. In the mask window, click the Figure Properties tab, uncheck Open scope at start of Simulation, then click OK.



- 5 In the Simulink model window, double-click the Rayleigh Channel Output block. In the mask window, click the Figure Properties tab, uncheck Open scope at start of Simulation, then click OK.
- 6 In the Simulink model window, double-click the Noisy Rayleigh Channel Output block. In the mask window, click the Figure Properties tab, uncheck Open scope at start of Simulation, then clock OK.
- 7 In the Simulink model window, double-click the Error Rate Calculation block, check **Stop simulation**, enter 5000 for **Target number of error**, then click **OK**.
- **8** Click the play button to rerun the demo.

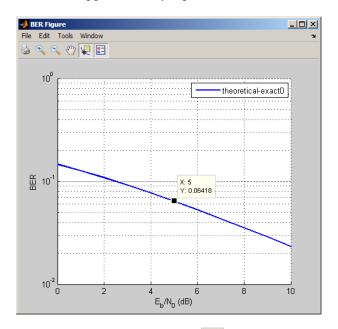
- **9** Open BERTool by typing bertool at the MATLAB command line.
- 10 In BERTool, click the Theoretical tab and make the following selections:



- For Eb/No range enter 0:10
- For Channel type, select Rayleigh
- For **Diversity Order** enter 1
- For Modulation order, select 4

#### 11 Click Plot.

**12** Since the Simulink model uses an EbNo value of 5 dB, verify the probability of error on the BERTool curve at 5 dB. The two values should be approximately equal.



Click the Data Cursor button (second from right) and click on the BERTool curve at 5dB.

# See Also

Rayleigh Noise Generator, Multipath Rician Fading Channel, doppler

# 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.
- [4] Iskander, Cyril-Daniel, A MATLAB-based Object-Oriented Approach to Multipath Fading Channel Simulation, a MATLAB Central submission available from www.mathworks.com.

**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. You can use this block to model mobile wireless communication systems when the transmitted signal can travel to the receiver along a dominant line-of-sight or direct path. For details about fading channels, see Channels in the *Communications 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. You can specify the Doppler spectrum of the Rician process using the **Doppler spectrum type** pop-up menu. For channels with multiple paths, you can assign each path a different Doppler spectrum, by entering a vector of doppler objects in the **Doppler spectrum** field.

Because a multipath channel reflects signals at multiple places, a transmitted signal travels to the receiver along several paths, each of which may have differing lengths and associated time delays. In the block's parameter dialog box, the **Discrete path delay vector** specifies the time delay for each path. If you do not check the **Normalize gain vector to 0 dB overall gain** box, then the **Average path gain vector** specifies the gain for each path. When you check the box, the block uses a multiple of **Average path gain vector** instead of the **Average path gain vector** itself, choosing the scaling factor so that the channel's effective gain considering all paths is 0 dB.

The number of paths indicates the length of **Discrete path delay vector** or **Average path gain vector**, whichever is larger. If both of these parameters are vectors, they must have the same length; if exactly

one of these parameters contains a scalar value, the block expands it into a vector whose size matches that of the other vector parameter.

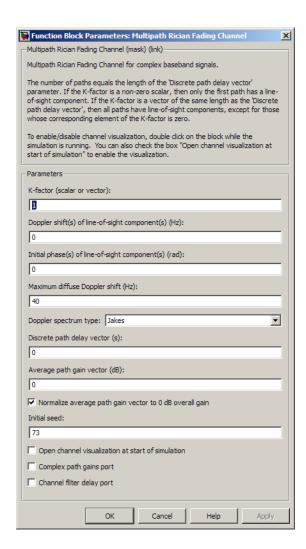
Fading causes the signal to become diffuse. The **K-factor** parameter, which is part of the statistical description of the Rician distribution, represents the ratio between the power in the line-of-sight component and the power in the diffuse component. The ratio is expressed linearly, not in decibels. While the Average path gain vector parameter controls the overall gain through the channel, the **K-factor** parameter controls the gain's partition into line-of-sight and diffuse components.

You can specify the **K-factor** parameter as a scalar or a vector. If the **K-factor** parameter is a scalar, then the first discrete path of the channel is a Rician fading process (it contains a line-of-sight component) with the specified **K-factor**, while the remaining discrete paths indicate independent Rayleigh fading processes (with no line-of-sight component). If the **K-factor** parameter is a vector of the same size as **Discrete path delay vector**, then each discrete path is a Rician fading process with a **K-factor** given by the corresponding element of the vector. You can attribute the line-of-sight component a Doppler shift, through the **Doppler shift(s)** of line-of-sight component(s) parameter, and an initial phase, through the **Initial phase(s)** of line-of-sight component(s) and **Initial phase(s)** of line-of-sight component(s) parameters must be of the same size as the K-factor parameter.

The block multiplies the input signal by samples of a Rician-distributed complex random process. The scalar **Initial seed** parameter seeds the random number generator. The block then generates random numbers using the Ziggurat method, which is the same method used by the MATLAB random function.

Double-clicking this block during simulation or selecting the block dialog's check box labeled **Open channel visualization at start of simulation** plots the channel characteristics using the channel visualization tool. See "Using the Channel Visualization Tool" in *Communications Toolbox User's Guide* for details.

## Dialog Box



#### K-factor

The ratio of power in the line-of-sight component to the power in the diffuse component. The ratio is expressed linearly, not in decibels. If **K-factor** is a scalar value, then the first discrete path

is a Rician fading process (it contains a line-of-sight component) with the specified K-factor, while the remaining discrete paths are independent Rayleigh fading processes (with no line-of-sight component). If **K-factor** is a vector of the same size as **Discrete path delay vector**, then each discrete path is a Rician fading process with a **K-factor** given by the corresponding element of the vector.

### Doppler shift(s) of line-of-sight components(s) (Hz)

The Doppler shift of the line-of-sight component. It must be a scalar (if **K-factor** is a scalar) or a vector of the same size as **K-factor**. If this parameter contains a scalar value, then the line-of-sight component of the first discrete path has the specified Doppler shift, while the remaining discrete paths become independent Rayleigh fading processes. If the parameter contains a vector, then the line-of-sight component of each discrete path has a Doppler shift given by the corresponding element of the vector.

### Initial phase(s) of line-of-sight component(s) (rad)

The initial phase of the line-of-sight component. It must be either a scalar (if **K-factor** is a scalar value) or a vector of the same size as **K-factor**.

### Maximum diffuse Doppler shift (Hz)

A positive scalar value that indicates the maximum diffuse Doppler shift.

## Doppler spectrum type

Specifies the Doppler spectrum of the Rician process.

This parameter defaults to Jakes Doppler spectrum. Alternately, you can choose any of the following types:

- Flat
- Gaussian
- Rounded
- Restricted Jakes

- Asymmetrical Jakes
- Bi-Gaussian
- Bell

For all Doppler spectrum types except Jakes and Flat, You can use one or more parameters to control the shape of the spectrum.

You can also select Specify as dialog parameter for the **Doppler spectrum type**. Specify the Doppler spectrum by entering an object in the **Doppler spectrum** field. See the doppler function reference in *Communications Toolbox User's Guide* for details on how to construct doppler objects, and for the meaning of the parameters associated with the various Doppler spectrum types.

### Discrete delay vector(s)

A vector that specifies the propagation delay for each path.

### Average path gain vector (dB)

A vector that specifies the gain for each path.

#### Initial seed

The scalar seed for the Gaussian noise generator.

### Open channel visualization at start of simulation

Checking this box opens the channel visualization tool when a simulation begins.

## Complex path gains port

Checking this box creates a port that outputs the values of the complex path gains for each path. In this N-by-M multichannel frame, N represents the number of samples per frame and M represents 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 results from the filtering operation of this block. This delay is zero if only one path is simulated, but can be greater than zero if more than one path is present. See

"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]. A detailed explanation of the implementation, including a review of the different Doppler spectra, can be found in [4].

Some wireless applications, such as standard GSM (Global System for Mobile Communication) systems, prefer to specify Doppler shifts in terms of the speed of the mobile. If the mobile moves at speed v making an angle of  $\theta$  with the direction of wave motion, the Doppler shift is

$$f_{\rm d} = (vf/c)\cos\theta$$

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, doppler

## 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.

[4] Iskander, Cyril-Daniel, A MATLAB-based Object-Oriented Approach to Multipath Fading Channel Simulation, a MATLAB Central submission available from www.mathworks.com.

### **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.

• 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*.

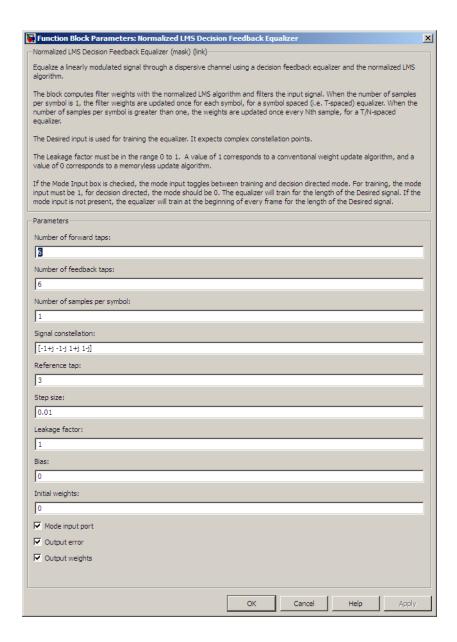
### **Equalizer Delay**

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+(Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

### 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.

### 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.

### **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

### **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*.

### **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*.

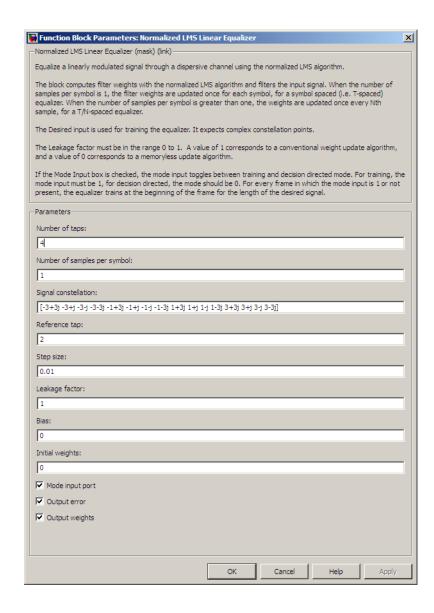
## **Equalizer Delay**

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+(Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## 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 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.

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.

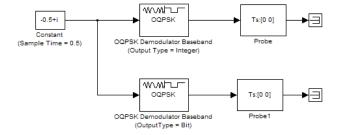
The input must be a discrete-time complex signal. The input can be either a scalar or a frame-based column vector. For information about the data types each block port supports, see the "Supported Data Types" on page 2-551 table on this page.

If the **Output type** parameter is set to Integer, then the block outputs integer symbol values between 0 and 3. If the **Output type** parameter is set to Bit, then the block outputs 2—bit binary representations of such integers, in a binary-valued vector whose length is an even number.

The block produces one output symbol for each pair of input symbols. 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.

## **Sample-Based Inputs**

If the input is a sample-based scalar, then the block outputs a single integer symbol (or one pair of bits) for each output time step. The input sample period is half the output symbol period, in this case.



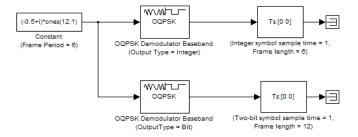
To open the completed model, click here in the MATLAB Help browser

### Frame-Based Inputs

If the input is a frame-based column vector, then the block processes several integer symbols (or several pairs of bits) for each time step. In this case, the frame-based output port frame time equals that of the input.

The output frame length is adjusted based on the input frame length and the setting for the **Output type** parameter. When you set the **Output type** parameter to Integer, the output frame length is half that of the input frame length. When you set the **Output type** parameter to Bit, the output frame length equals the input frame length (because each symbol is two bits).

**Note** The input frame length must be even or else the block will error out.



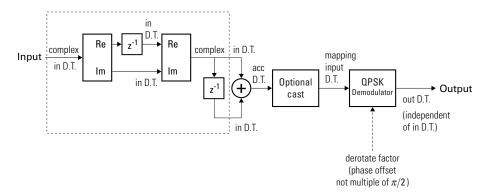
To open the completed model, click here in the MATLAB Help browser

## **Delays**

The modulator-demodulator pair incurs a delay, as described in "Delays in Digital Modulation".

# **OQPSK Demodulator Baseband**

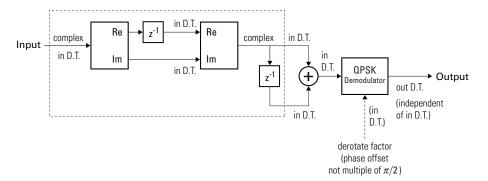
# Signal Flow Diagram



### **OQPSK Fixed-Point Signal Flow Diagram**

**Note** Every two input samples produce one output symbol. In the preceding figure, the dotted line represents the region comprised of input sample processing.

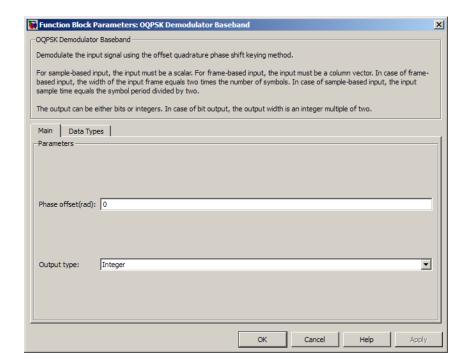
# **OQPSK Demodulator Baseband**



### **OQPSK Floating Point Signal Flow Diagram**

**Note** Every two input samples produce one output symbol. In the preceding figure, the dotted line represents the region comprised of input sample processing.

## Dialog Box

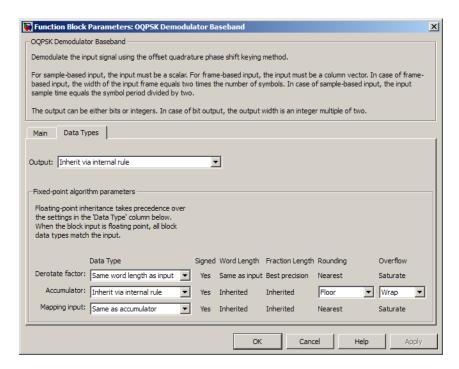


#### Phase offset (rad)

The amount by which the phase of the zeroth point of the signal constellation is shifted from  $\pi/4$ .

### Output type

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



#### **Data Type Pane**

#### Output

For bit outputs, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, uint32, or boolean.

For integer outputs, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, or uint32.

When this parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is a floating-point type (single or double).

# **OQPSK Demodulator Baseband**

If the input data type is fixed-point, the output data type will work as if this parameter is set to 'Smallest unsigned integer'.

When this parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the model.

If ASIC/FPGA is selected in the **Hardware Implementation** pane, and **Output type** is Bit, the output data type is the ideal minimum one-bit size, i.e., ufix1. For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

If ASIC/FPGA is selected in the **Hardware Implementation** pane, and **Output type** is Integer, the output data type is the ideal minimum two-bit size, i.e., ufix2. For all other selections, it is an unsigned integer with the smallest available word length large enough to fit two bits, usually corresponding to the size of a char (e.g., uint8).

#### **Derotate factor**

This parameter only applies when the input is fixed-point and

**Phase offset** is not a multiple of 
$$\frac{\Pi}{2}$$
.

This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input.

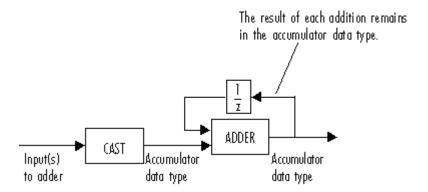
#### Accumulator

Specify the data type for the Accumulator. You can set this parameter to Inherit via internal rule, Same as input or Binary point scaling.

The **Accumulator** parameter only applies for fixed-point inputs. The selections you make for the **Rounding** and **Overflow** parameters affect the **Accumulator**.

# **OQPSK Demodulator Baseband**

Fixed-point Communications Blockset blocks that must hold summation results for further calculation usually allow you to specify the data type and scaling of the accumulator. Most such blocks cast to the accumulator data type prior to summation:



See the reference page of a specific block in the Block Reference for details on the accumulator data type of a specific block.

Use the **Accumulator—Mode** parameter to specify how you would like to designate the accumulator word and fraction lengths:

- When you select Inherit via internal rule, the accumulator output word and fraction lengths are automatically calculated for you. Refer to Inherit via Internal Rule for more information.
- When you select Same as product output, these characteristics match those of the product output.
- When you select Same as input, these characteristics match those of the first input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.

# **OQPSK Demodulator Baseband**

 When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator. The bias of all signals in Signal Processing Blockset software is zero.

### Rounding

The block uses the **Rounding** method when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink Fixed Point documentation.

### Overflow

Specify the method of storing the result when the magnitude of a fixed-point calculation result that does not does not fit within the range of the data type selected. You can select either Wrap or Saturate for this parameter.

For more information refer to Overflow in the Precision and Range subsection of the Signal Processing Blockset Getting Started Guide.

# Mapping input

This can be set to Same as accumulator or Binary point scaling. This parameter only applies for fixed-point inputs.

# Supported Data Types

Port	Supported Data Types		
Input	Double-precision floating point		
	• Single-precision floating point		
	Signed fixed point		

# **OQPSK Demodulator Baseband**

Port	Supported Data Types		
Output	Double-precision floating point		
	Single-precision floating point		
	Boolean when <b>Output type</b> is Bit		
	• 8-, 16-, and 32-bit signed integers		
	• 8-, 16-, and 32-bit unsigned integers		
	• ufix(1) in ASIC/FPGA when <b>Output type</b> is Bit		
	• ufix(2) in ASIC/FPGA when <b>Output type</b> is Integer		

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** 

> OQPSK

The OQPSK Modulator Baseband block modulates using the offset quadrature phase shift keying method. The output is a baseband representation of the modulated signal.

**Note** The OQPSK modulator block upsamples by a factor of 2.

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.

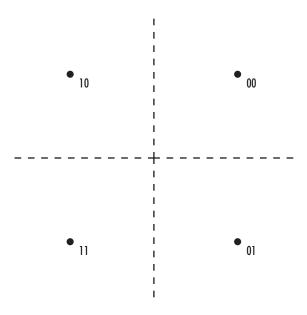
If the **Input type** parameter is set to Bit, then the input must be a binary-valued vector. In this case, the input can be either a vector of length two or a frame-based column vector whose length is an even integer.

For information about the data types each block port supports, see the "Supported Data Types" on page 2-558 table on this page.

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.

# **OQPSK Modulator Baseband**

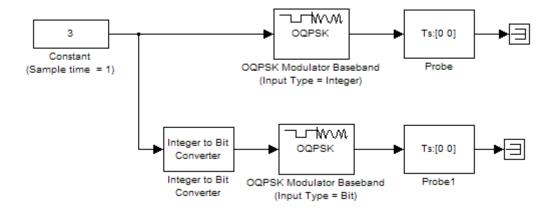


# **Sample-Based Inputs**

For sample-based inputs, the output sample time equals half of the input symbol period.

- In bit input mode, the input must be of length two.
- In integer input mode, the input must be a scalar value.

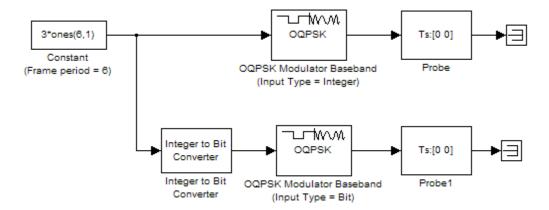
If the input is a sample-based scalar, then the block outputs a single integer symbol for each time step. In this case, the sample-based input symbol period is half the output symbol period.



To open the completed model, click here in the MATLAB Help browser

# 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 sample is half the period of each integer or bit pair in the input.

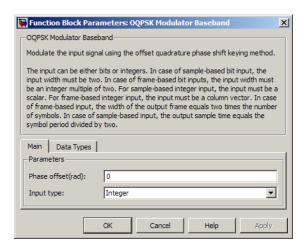


To open the completed model, click here in the MATLAB Help browser

### **Delays**

The modulator-demodulator pair incurs a delay, as described in "Delays in Digital Modulation".

# Dialog Box



### Phase offset (rad)

The amount by which the phase of the zeroth point of the signal constellation is shifted from  $\pi/4$ .

### Input type

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

# Output data type

The output data type can be set to double, single, Fixed-point, User-defined, or Inherit via back propagation.

Setting this parameter to Fixed-point or User-defined enables fields in which you can further specify details. Setting this parameter to Inherit via back propagation, sets the output data type and scaling to match the following block.

# **OQPSK Modulator Baseband**

### Output word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed-point for the **Output data type** parameter.

### 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.

### Output fraction length

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.

# **OQPSK Modulator Baseband**

# Supported Data Types

Port	Supported Data Types			
Input	Double-precision floating point			
	Single-precision floating point			
	Boolean when Input type is Bit			
	• 8-, 16-, and 32-bit signed integers			
	• 8-, 16-, and 32-bit unsigned integers			
	• ufix(1) when <b>Input type</b> is Bit			
	• ufix(2) when <b>Input type</b> is Integer			
Output	Double-precision floating point			
	Single-precision floating point			
	Signed fixed point			

**Pair Block** 

OQPSK Demodulator Baseband

See Also

QPSK Modulator Baseband

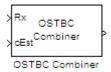
## **Purpose**

Combine inputs for received signals and channel estimate according to orthogonal space-time block code (OSTBC)

# Library

### MIMO

# **Description**

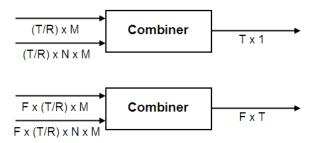


The OSTBC Combiner block combines the input signal (from all of the receive antennas) and the channel estimate signal to extract the soft information of the symbols encoded by an OSTBC. The input channel estimate may not be constant during each codeword block transmission and the combining algorithm uses only the estimate for the first symbol period per codeword block. A symbol demodulator or decoder would follow the Combiner block in a MIMO communications system.

The block conducts the combining operation for each symbol independently. The combining algorithm depends on the structure of the OSTBC. For more information, see the OSTBC Combining Algorithms section of this help page.

### **Dimension**

Along with the time and spatial domains for OSTBC transmission, the block supports an optional dimension, over which the combining calculation is independent. This dimension can be thought of as the frequency domain for OFDM-based applications. The following illustration indicates the supported dimensions for inputs and output of the OSTBC Combiner block.



The following table describes each variable for the block.

Variable	Description
F	The additional dimension; typically the frequency dimension. The combining calculation is independent of this dimension.
N	Number of transmit antennas.
M	Number of receive antennas.
Т	Output symbol sequence length in time domain.
R	Symbol rate of the code.

**Note** On the two inputs, T/R is the symbol sequence length in the time domain.

F can be any positive integers. M can be 1 through 8, indicated by the **Number of receive antennas** parameter. N can be 2, 3 or 4, indicated by the **Number of transmit antennas** parameter. The time domain length T/R must be a multiple of the codeword block length (2 for Alamouti; 4 for all other OSTBC). For N = 2, T/R must be a multiple of 2. When N > 2, T/R must be a multiple of 4. R defaults to 1 for 2

antennas. R can be either  $\frac{3}{4}$  or  $\frac{1}{2}$  for more than 2 antennas.

The supported dimensions for the block depend upon the values of F and M. For one receive antenna (M=1), the received signal input must be a column vector or a full 2–D matrix, depending on the value for F. The corresponding channel estimate input must be a full 2-D or 3-D matrix.

For more than one receive antenna (M > 1), the received signal input must be a full 2-D or 3-D matrix, depending on the value for F. Correspondingly, the channel estimate input must be a 3-D or 4-D matrix, depending on the value for F.

To understand the block's dimension propagation, refer to the following table.

	Input 1 (Received Signal)	Input 2 (Channel Estimate)	Output
F = 1 and $M = 1$	Column vector	2-D	Column vector
F = 1 and $M > 1$	2-D	3-D	Column vector
F > 1 and $M = 1$	2-D	3-D	2-D
F > 1 and M > 1	3-D	4-D	2-D

# **Data Type**

For information about the data types each block port supports, see the "Supported Data Type" on page 2-569 table on this page. The output signal inherits the data type from the inputs. The block supports different fixed-point properties for the two inputs. For fixed-point signals, the output word length and fractional length depend on the block's mask parameter settings. See Fixed-Point Signals for more information about fixed-point data propagation of this block.

#### **Frames**

The output inherits the frameness of the received signal input. For either column vector or full 2-D matrix input signal, the input can be either frame-based or sample-based. A 3–D or 4–D matrix input signal must have sample-based input.

# **OSTBC Combining Algorithms**

The OSTBC Combiner block supports five different OSTBC combining computation algorithms. Depending on the selection for **Rate** and **Number of transmit antennas**, you can select one of the algorithms shown in the following table.

Transmit Antenna	Rate	Computational Algorithm per Codeword Block Length
2	1	
		$ \begin{pmatrix} \hat{s}_1 \\ \hat{s}_2 \end{pmatrix} = \frac{1}{\ H\ ^2} \sum_{j=1}^{M} \begin{pmatrix} h_{1,j}^* \ r_{1,j} + h_{2,j} \ r_{2,j}^* \\ h_{2,j}^* \ r_{1,j} - h_{1,j} \ r_{2,j}^* \end{pmatrix}. $
3	1/2	
		$ \begin{pmatrix} \hat{s}_1 \\ \hat{s}_2 \end{pmatrix} = \frac{1}{\ H\ ^2} \sum_{j=1}^{M} \begin{pmatrix} h_{1,j}^* \ r_{1,j} + h_{2,j} \ r_{2,j}^* + h_{3,j} \ r_{3,j}^* \\ h_{2,j}^* \ r_{1,j} - h_{1,j} \ r_{2,j}^* - h_{3,j} \ r_{4,j}^* \end{pmatrix} . $
3	3/4	
		$\begin{pmatrix} \hat{s}_{1} \\ \hat{s}_{2} \\ \hat{s}_{3} \end{pmatrix} = \frac{1}{\ H\ ^{2}} \sum_{j=1}^{M} \begin{pmatrix} h_{1,j}^{*} r_{1,j} + h_{2,j} r_{2,j}^{*} - h_{3,j} r_{3,j}^{*} \\ h_{2,j}^{*} r_{1,j} - h_{1,j} r_{2,j}^{*} - h_{3,j} r_{4,j}^{*} \\ h_{3,j}^{*} r_{1,j} + h_{1,j} r_{3,j}^{*} + h_{2,j} r_{4,j}^{*} \end{pmatrix}.$
4	1/2	
		$ \begin{pmatrix} \hat{s}_1 \\ \hat{s}_2 \end{pmatrix} = \frac{1}{\ H\ ^2} \sum_{j=1}^M \begin{pmatrix} h_{1,j}^* \ r_{1,j} + h_{2,j} \ r_{2,j}^* + h_{3,j3,j}^* r_{3,j} + h_{4,j} \ r_{4,j}^* \\ h_{2,j}^* \ r_{1,j} - h_{1,j} \ r_{2,j}^* + h_{4,j}^* \ r_{3,j} - h_{3,j} \ r_{4,j}^* \end{pmatrix} . $
4	3/4	
		$ \begin{pmatrix} \hat{s}_1 \\ \hat{s}_2 \\ \hat{s}_3 \end{pmatrix} = \frac{1}{\ H\ ^2} \sum_{j=1}^{M} \begin{pmatrix} h_{1,j}^* \ r_{1,j} + h_{2,j} \ r_{2,j}^* - h_{3,j} \ r_{3,j}^* - h_{4,j}^* \ r_{4,j} \\ h_{2,j}^* \ r_{1j} - h_{1,j} \ r_{2,j}^* + h_{4,j}^* \ r_{3,j} - h_{3,j} \ r_{4,j}^* \\ h_{3,j}^* \ r_{1j} + h_{4,j}^* \ r_{2,j} + h_{1,j} \ r_{3,j}^* + h_{2,j} \ r_{4,j}^* \end{pmatrix} . $

 $\hat{s}_k$  represents the estimated kth symbol in the OSTBC codeword matrix.  $h_{ij}$  represents the estimate for the channel from the ith transmit antenna and the jth receive antenna. The values of i and j can range from 1 to N (the number of transmit antennas) and to M (the number of receive antennas) respectively.  $r_{ij}$  represents the lth symbol at the jth receive antenna per codeword block. The value of l can range from 1 to

the codeword block length.  $\|H\|^2$  represents the summation of channel

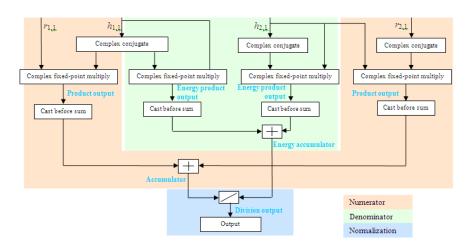
power per link, i.e., 
$$||H||^2 = \sum_{i=1}^{N} \sum_{j=1}^{M} ||h_{ij}||^2$$

## **Fixed-Point Signals**

Use the following formula for  $\hat{s}_1$  for Alamouti code with 1 receive antenna to highlight the data types used for fixed-point signals.

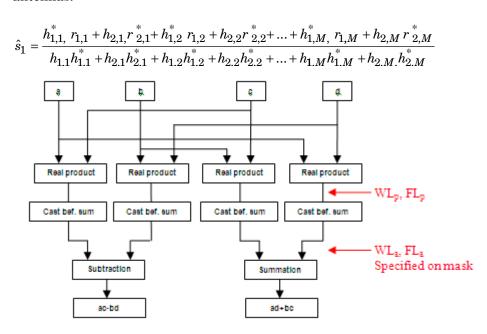
$$\hat{s}_1 = \frac{h_{1,1,}^* \ r_{1,1} + h_{2,1,} r_{2,1}^*}{\|H\|^2} = \frac{h_{1,1,}^* \ r_{1,1} + h_{2,1,} r_{2,1}^*}{h_{1,1} h_{1,1}^* + h_{2,1,} h_{2,1}^*}$$

In this equation, the data types for **Product output** and **Accumulator** correspond to the product and summation in the numerator. Similarly, the types for **Energy product output** and **Energy accumulator** correspond to the product and summation in the denominator.



Signal Flow Diagram for s<sub>1</sub> Combining Calculation of Alamouti Code with One Receive Antenna

The following formula shows the data types used within the OSTBC Combiner block for fixed-point signals for more than one receive antenna for Alamouti code, where M represents the number of receive antennas.



### Signal Flow Diagram for Complex Multiply of a + ib and c + id

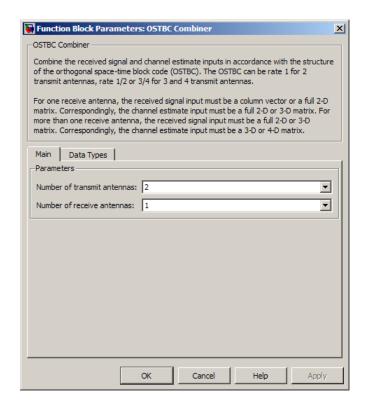
For Binary point scaling, you can not specify  $WL_p$  and  $FL_p$ . Instead, the blocks determine these values implicitly from  $WL_a$  and  $FL_a$ 

The Internal Rule for **Product output** and **Energy product output** are:

- When you select Inherit via internal rule, the internal rule determines  $WL_p$  and  $FL_p$ . Therefore,  $WL_a = WL_p + 1$  and  $FL_a = FL_p$
- For Binary point scaling, you specify  $WL_a$  and  $FL_a$ . Therefore,  $WL_p=WL_a-1$  and  $FL_a=FL_p$ .

For information on how the Internal Rule applies to the **Accumulator** and **Energy Accumulator**, see Inherit via Internal Rule in the *Signal Processing Blockset User's Guide*.

# Dialog Box



### **Block Parameters**

### Number of transmit antennas

Sets the number of transmit antennas. The block supports 2, 3, or 4 transmit antennas. This value defaults to 2.

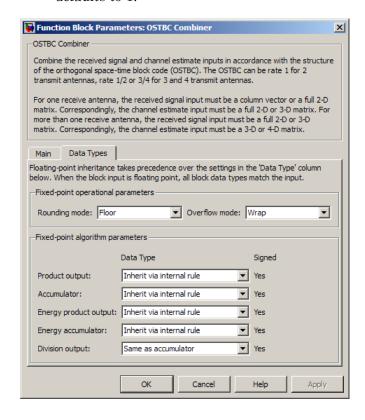
### Rate

Sets the symbol rate of the code. You can specify either 3/4 or 1/2. This field only appears when you use more than 2 transmit

antennas. This field defaults to  $\frac{3}{4}$  for more than 2 transmit antennas. For 2 transmit antennas, there is no rate option and the implicit (default) rate defaults to 1.

### Number of receive antennas

The number of antennas the block uses to receive signal streams. The block supports from 1 to 8 receive antennas. This value defaults to 1.



### Rounding mode

Sets the rounding mode for fixed-point calculations. The block uses the rounding mode if a value cannot be represented exactly by the specified data type and scaling. When this occurs, the value is rounded to a representable number. For more information refer to Rounding in the *Simulink Fixed-Point User's Guide*.

### Overflow mode

Sets the overflow mode for fixed-point calculations. Use this parameter to specify the method to be used if the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result. For more information refer to Precision and Range in the Precision and Range section of the *Signal Processing Blockset User's Guide*.

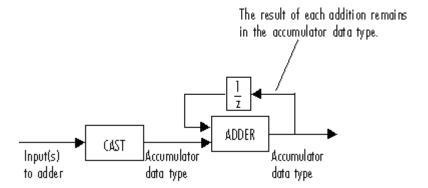
### **Product Output**

Complex product in the numerator for the diversity combining. For more information refer to the Fixed-Point Signals section of this help page.

### Accumulator

Summation in the numerator for the diversity combining.

Fixed-point Communications Blockset blocks that must hold summation results for further calculation usually allow you to specify the data type and scaling of the accumulator. Most such blocks cast to the accumulator data type prior to summation:



See the reference page of a specific block in the Block Reference for details on the accumulator data type of a specific block.

Use the **Accumulator—Mode** parameter to specify how you would like to designate the accumulator word and fraction lengths:

- When you select Inherit via internal rule, the accumulator output word and fraction lengths are automatically calculated for you. Refer to Inherit via Internal Rule for more information.
- When you select Same as product output, these characteristics match those of the product output.
- When you select Same as input, these characteristics match those of the first input to the block.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the accumulator, in bits.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the accumulator. The bias of all signals in Signal Processing Blockset software is zero.

### **Energy product output**

Complex product in the denominator for calculating total energy in the MIMO channel .

### **Energy accumulator**

Summation in the denominator for calculating total energy in the MIMO channel.

### Division output

Normalized diversity combining by total energy in the MIMO channel.

# Supported Data Type

Port	Supported Data Types	
Rx	Double-precision floating point	
	Single-precision floating point	
	Signed Fixed-point	
cEst	Double-precision floating point	
	Single-precision floating point	
	Signed Fixed-point	
Out	Double-precision floating point	
	Single-precision floating point	
	Signed Fixed-point	

# **Examples**

For an example of this block in use, see OSTBC Over 3x2 Rayleigh Fading Channel in the Communications blockset *User's Guide*. The model shows the use of a rate ¾ OSTBC for 3 transmit and 2 receive antennas with BPSK modulation using independent fading links and AWGN.

You can also see the block in the Concatenated OSTBC with TCM demo by typing commtcmostbc or the IEEE 802.16-2004 OFDM

# **OSTBC Combiner**

PHY Link, Including Space-Time Block Coding demo by typing commwman80216dstbc at the MATLAB command line.

See Also OSTBC Encoder

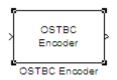
# **Purpose**

Encode input message using orthogonal space-time block code (OSTBC)

# Library

MIMO

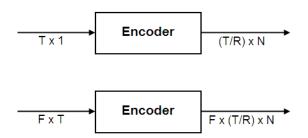
# **Description**



The OSTBC Encoder block encodes an input symbol sequence using orthogonal space-time block code (OSTBC). The block maps the input symbols block-wise and concatenates the output codeword matrices in the time domain. For more information, see the OSTBC Encoding Algorithms section of this help page.

### **Dimension**

The block supports time and spatial domains for OSTBC transmission. It also supports an optional dimension, over which the encoding calculation is independent. This dimension can be thought of as the frequency domain. The following illustration indicates the supported dimensions for the inputs and output of the OSTBC Encoder block.



The following table describes the variables.

Variable	Description
F	The additional dimension; typically the frequency domain. The encoding does not depend on this dimension.
Т	Input symbol sequence length for the time domain.

Variable	Description	
R	Symbol rate of the code.	
N	Number of transmit antennas.	

**Note** On the output, T/R is the symbol sequence length in time domain.

F can be any positive integer. N can be 2, 3 or 4, indicated by **Number of transmit antennas**. For N=2, R must be 1. For N=3 or 4, R can be 3/4 or 1/2, indicated by **Rate**. The time domain length T must be a multiple of the number of symbols in each codeword matrix. Specifically, for N=2 or R=1/2, T must be a multiple of 2 and when R=3/4, T must be a multiple of 3.

To understand the block's dimension propagation, refer to the following table.

Dimension	Input	Output
F = 1	Column vector	2-D
F > 1	2-D	3-D

## **Data Type**

For information about the data types each block port supports, see the "Supported Data Type" on page 2-576 table on this page. The output signal inherits the data type from the input signal. For fixed-point signals, the complex conjugation may cause overflows which the fixed-point parameter **Overflow mode** must handle.

### **Frames**

The output signal inherits frame type from the input signal. A column vector input requires either frame-based or sample-based input; otherwise, the input must be sample-based.

# **OSTBC Encoding Algorithms**

The OSTBC Encoder block supports five different OSTBC encoding algorithms. Depending on the selection for **Rate** and **Number of transmit antennas**, the block implements one of the algorithms in the following table:

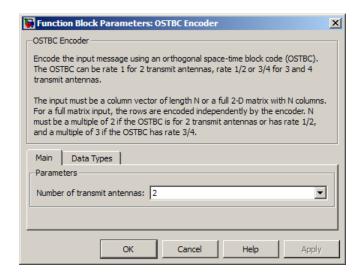
Transmit Antenna	Rate	OSTBC Codeword Matrix
2	1	$\begin{pmatrix}s_1&s_2\\-s_2^*&s_1^*\end{pmatrix}$
3	1/2	$egin{pmatrix} s_1 & s_2 & 0 \ -s_2^* & s_1^* & 0 \ 0 & 0 & s_1 \ 0 & 0 & -s_2^* \end{pmatrix}$
3	3/4	$egin{pmatrix} s_1 & s_2 & s_3 \ -s_2^* & s_1^* & 0 \ s_3^* & 0 & -s_1^* \ 0 & s_3^* & -s_2^* \end{pmatrix}$

# **OSTBC Encoder**

Transmit Antenna	Rate	OSTBC Codeword Matrix
4	1/2	$egin{pmatrix} s_1 & s_2 & 0 & 0 \ -s_2^* & s_1^* & 0 & 0 \ 0 & 0 & s_1 & s_2 \ 0 & 0 & -s_2^* & s_1^* \end{pmatrix}$
4	3/4	$egin{pmatrix} s_1 & s_2 & s_3 & 0 \ -s_2^* & s_1^* & 0 & s_3 \ s_3^* & 0 & -s_1^* & s_2 \ 0 & s_3^* & -s_2^* & -s_1 \ \end{pmatrix}$

In each matrix, its (l,i) entry indicates the symbol transmitted from the ith antenna in the lth time slot of the block. The value of i can range from 1 to N (the number of transmit antennas). The value of l can range from 1 to the codeword block length.

# Dialog Box



### **Block Parameters**

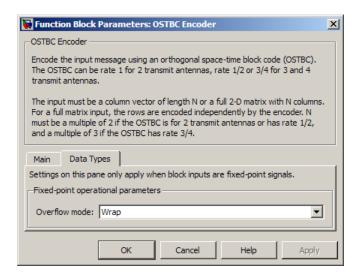
### Number of transmit antennas

Sets the number of antennas at the transmitter side. The block supports 2, 3, or 4 transmit antennas. The value defaults to 2.

#### Rate

Sets the symbol rate of the code. You can specify either 3/4 or 1/2. This field only appears when using more than 2 transmit

antennas. This field defaults to  $\frac{3}{4}$  for more than 2 transmit antennas. For 2 transmit antennas, there is no rate option and the rate defaults to 1.



### Overflow mode

Sets the overflow mode for fixed-point calculations. Use this parameter to specify the method to be used if the magnitude of a fixed-point calculation result does not fit into the range of the data type and scaling that stores the result. For more information refer to "Precision and Range" in the *Signal Processing Blockset User's Guide*.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Signed Fixed-point
Out	Double-precision floating point
	Single-precision floating point
	Signed Fixed-point

# **Examples**

For an example of this block in use, see OSTBC Over 3x2 Rayleigh Fading Channel in the Communications blockset *User's Guide*. The model shows the use of a rate ¾ OSTBC for 3 transmit and 2 receive antennas with BPSK modulation using independent fading links and AWGN

You can also see the block in the Concatenated OSTBC with TCM demo by typing commtcmostbc. View the IEEE 802.16–2004 OFDM PHY Link, Including Space-Time Block Coding demo by typing commwman80216dstbc at the MATLAB command line.

# See Also OSTBC Combiner

# **OVSF Code Generator**

### **Purpose**

Generate orthogonal variable spreading factor (OVSF) code from set of orthogonal codes

# Library

Spreading Codes

# **Description**

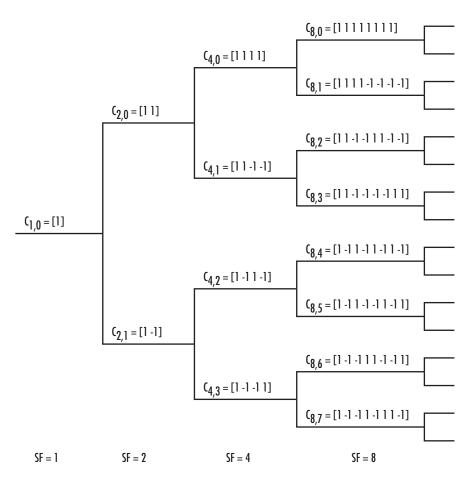
OVSF Code Generator The OVSF Code Generator block generates an OVSF code from a set of orthogonal codes. OVSF codes were first introduced for 3G communication systems. OVSF codes are primarily used to preserve orthogonality between different channels in a communication system.

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 kth 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  $C_{\rm N}$  is only defined for N a power of 2. It follows by induction that the rows of  $C_{\rm N}$  are orthogonal.

The OVSF codes can also be defined recursively by a tree structure, as shown in the following figure.



If [C] is a code length  $2^r$  at depth r in the tree, where the root has depth 0, the two branches leading out of C are labeled by the sequences  $[C\ C]$  and  $[C\ -C]$ , which have length  $2^{r+1}$ . The codes at depth r in the tree are the rows of the matrix  $C_N$ , where  $N=2^r$ .

Note that two OVSF codes are orthogonal if and only if neither code lies on the path from the other code to the root. Since codes assigned to different users in the same cell must be orthogonal, this restricts the number of available codes for a given cell. For example, if the code  $C_{41}$ 

in the tree is assigned to a user, the codes  $C_{10}$ ,  $C_{20}$ ,  $C_{82}$ ,  $C_{83}$ , 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, ..., N-1], where N is the spreading factor. If the code appears at depth r in the preceding tree, the **Spreading factor** is  $2^{\rm r}$ . 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  $C_{\rm N,\,k}$  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.

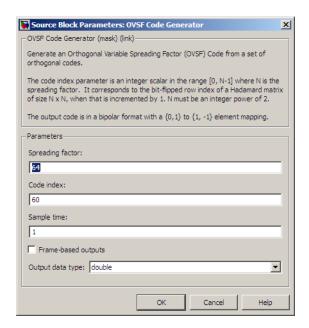
For example, to find the code with **Spreading factor 16** and **Code index 6**, do the following:

- **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	<b>x</b> <sub>i</sub>	C <sub>i</sub>
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	$C_3 = C_2 - C_2 = [1 \ 1 \ -1 \ -1] [-1 \ -1 \ 1]$
4	0	$C_4 = C_3 C_3 = [1 \ 1 \ -1 \ -1 \ -1 \ 1 \ 1] [1 \ 1 \ -1 \ -1 \ -1 \ 1 \ 1]$

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**

Phase/ > Frequency > Offset 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.

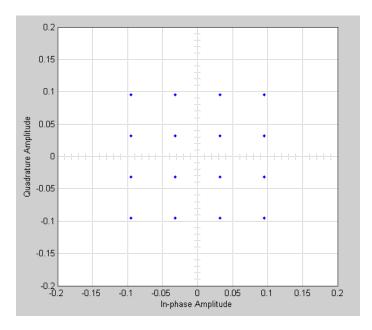
## **Frequency Offset**

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.

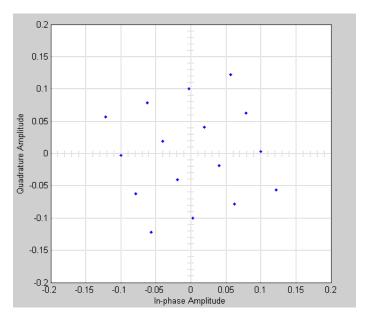
The effects of changing the block's parameters are illustrated by the following scatter plots of a signal modulated by 16-ary quadrature amplitude modulation (QAM). The usual 16-ary QAM constellation without the effect of the Phase/Frequency Offset block is shown in the first scatter plot:

# **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:

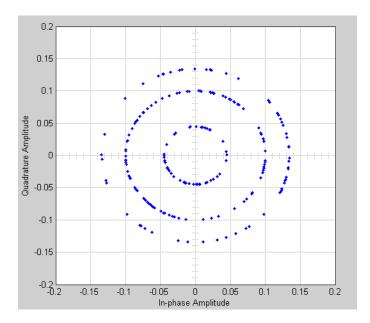
# **Phase/Frequency Offset**



Observe that each point in the constellation is rotated by a 20 degree angle counterclockwise.

If you set **Frequency offset** to 2 and **Phase offset** to 0, the angles of points in the constellation change linearly over time. This causes points in the scatter plot to shift radially, as shown in the following figure:

# **Phase/Frequency Offset**



Note that every point in the scatter plot has magnitude equal to a point in the original constellation.

See "Scatter Plot Examples" for a description of the model that generates this plot.

## I/O Ports

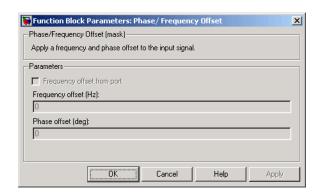
Only 1-D (unoriented) and 2-D ports are supported, as well as 2-D frame-based (single or multichannel) ports.

Scalar (length-one) sample-based data ports are also supported, but nonscalar sample-based ports are not.

# **Phase/Frequency Offset**

# **Interdependent Parameter-Port Dimensions Table**

Data I/O Frmstat	No. of Dim.	Data I/O Dim.	Frame Size	No. of Channels	Freq. or Phase Offset Parameter Dim.
Any	Any	Scalar	1	1	Scalar
Frame-based	2	M x 1	M	1	M, M x 1, 1 x M, 1, 1 x 1
Frame-based	2	1 x N	1	N	N, N x 1, 1 x N, 1, 1 x 1
Frame-based	2	M x N	M	N	M x N, N, N x 1, 1 x N, 1, 1 x 1
Sample-based	2	M x 1 (Errors — not currently supported)			
Sample-based	2	1 x N (Errors — not currently supported)			
Sample-based	2	M x N (Errors — not currently supported)			
Sample-based	1	L (Errors — not currently supported)			



### Dialog Box

#### Frequency offset from port

Selecting this option opens a port on the block through which you can input the frequency offset information.

#### Frequency offset

Specifies the frequency offset in hertz.

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink *User's Guide*.

#### Phase offset

Specifies the phase offset in degrees.

This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode. If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. For more information, see Tunable Parameters in the Simulink *User's Guide*.

# **Phase/Frequency Offset**

If **Frequency offset** and **Phase offset** are both vectors or both matrices, their dimensions (vector lengths, or number of rows and columns) must be the same.

# **See Also** Phase Noise

# **Phase-Locked Loop**

## **Purpose**

Implement phase-locked loop to recover phase of input signal

# Library

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 software. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.

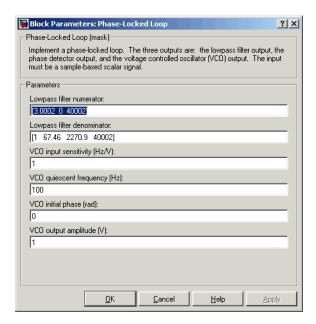
```
[num, den] = cheby2(3,40,100, 's')
```

• A voltage-controlled oscillator (VCO). You specify characteristics of the VCO using the VCO quiescent frequency, VCO initial phase, and VCO output amplitude parameters.

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

## 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.

# **Phase-Locked Loop**

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.

### **Purpose**

Apply receiver phase noise to complex baseband signal

# Library

RF Impairments

## **Description**

> Phase > Noise

The Phase Noise block applies phase noise to a complex, baseband signal. The block applies the phase noise as follows:

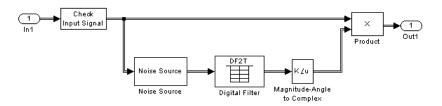
- 1 Generates additive white Gaussian noise (AWGN) and filters it with a digital filter.
- **2** Adds the resulting noise to the angle component of the input signal.

The Phase Noise block generates phase noise over the entire spectral

observation window, from 0 Hz (or as close as possible to 0 Hz) to  $\pm \frac{F_S}{2}$ , where  $F_s$  represents the sampling frequency. The noise is scaled so that it is at the block-specific phase noise level at the specified frequency

offset. The block generates a phase noise with  $\frac{1}{f}$  characteristic over the entire frequency range.

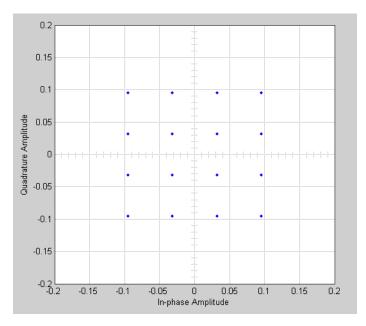
You can view the block's implementation of phase noise by right-clicking on the block and selecting **Look under mask**. This displays the following figure:



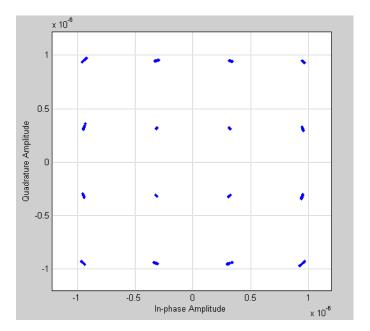
You can view the construction of the Noise Source subsystem by double-clicking it.

# **Phase Noise**

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:



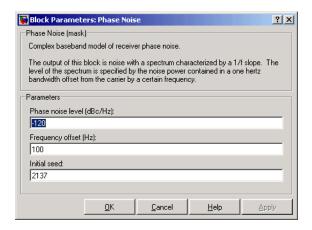
The following figure shows a scatter plot of an output signal, modulated by 16-ary QAM, from the Phase Noise block with **Phase noise level** (dBc/Hz) set to -70 and Frequency offset (Hz) set to 100:



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**

**Purpose** 

Demodulate PM-modulated data

Library

Analog Passband Modulation, in Modulation

**Description** 

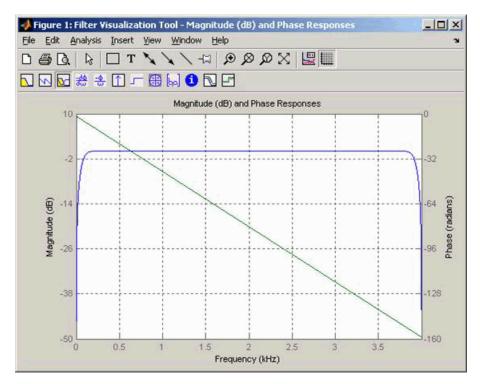


The PM Demodulator Passband block demodulates a signal that was modulated using phase modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real sample-based scalar signals.

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**

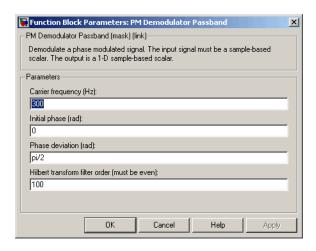


Note the bandwidth of the filter's magnitude response. By choosing a carrier frequency larger than 10% (but less than 90%) of the input signal's sample time (8000 samples per second, in this example) or equivalently, a carrier frequency larger than 400Hz, we ensure that the Hilbert Transform Filter will 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.

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

# **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

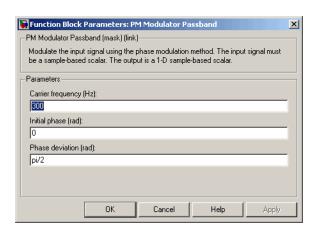
$$\cos(2\pi f_c t + K_c u(t) + \theta)$$

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**

#### 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**

**Purpose** 

Generate pseudonoise sequence

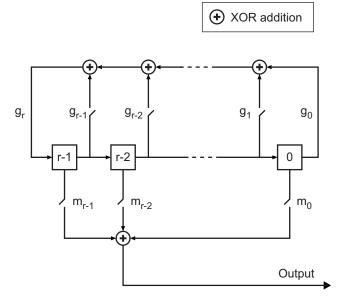
Library

Sequence Generators sublibrary of Comm Sources

**Description** 

PN Sequence Generator The PN Sequence Generator block generates a sequence of pseudorandom binary numbers using a linear-feedback shift register (LFSR). The LFSR is implemented using a simple shift register generator (SSRG, or Fibonacci) configuration. A pseudonoise sequence can be used in a pseudorandom scrambler and descrambler. It can also be used in a direct-sequence spread-spectrum system.

The PN Sequence Generator block uses a shift register to generate sequences, as shown below.



All r registers in the generator update their values at each time step according to the value of the incoming arrow to the shift register. The adders perform addition modulo 2. The shift register is described by the **Generator Polynomial** parameter, which is a primitive binary

polynomial in z,  $g_r z^r + g_{r\cdot 1} z^{r\cdot 1} + g_{r\cdot 2} z^{r\cdot 2} + ... + g_0$ . The coefficient  $g_k$  is 1 if there is a connection from the kth 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.

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

For example, the following table indicates two sets of parameter values that correspond to a generator polynomial of  $p(z) = z^8 + z^2 + 1$ .

# **PN Sequence Generator**

Quantity	Example 1	Example 2
Generator polynomial	g1 = [1 0 0 0 0 0 1 0 1]	g2 = [8 2 0]
Degree of generator polynomial	8, which is length(g1)-1	8
Initial states	[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.

Alternatively, you can set **Output mask vector (or scalar shift value)** to a binary vector, corresponding to a polynomial in z,  $m_{r-1}z^{r-1} + m_{r-2}z^{r-2} + ... + m_1z + m_0$ , of degree at most r-1. The mask vector

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-based	Reset
Sample time = $1$	Sample time = 1	0 0 0 1
		10010011011
Frame-based	Frame-based	Reset
Sample time =1	Sample time = 1	0 0 0 1

# **PN Sequence Generator**

Reset Signal Properties	PN Sequence Generator block	Reset Signal, Output Signal
Samples per frame = 2	Samples per frame = 2	
Sample-based Sample time = 2	Frame-based Sample time = 1	Reset 0 0 0 1
Samples per frame = 1	Samples per frame = 2	1 0 0 1 0 0 1 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**

r	Generator Polynomial	r	Generator Polynomial
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]

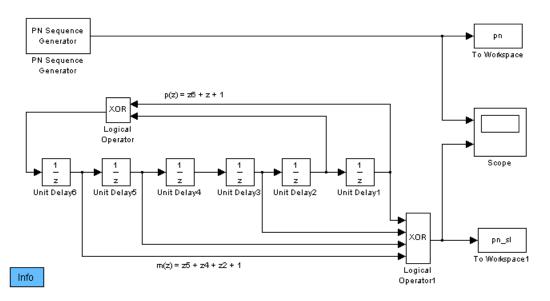
r	Generator Polynomial	r	Generator Polynomial
45	[45 4 3 1 0]	52	[52 3 0]
46	[46 21 10 1 0]	53	[53 6 2 1 0]

# **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.

This model is located at: matlabroot/help/toolbox/commblks/examples/doc\_pnseq2. If you are reading this reference page in the MATLAB Help Browser, type doc pnseq2 to open the model.

### PN Sequence Generation

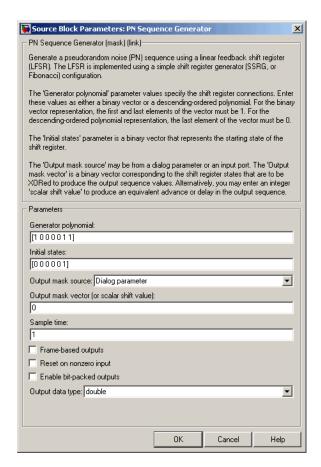


# **PN Sequence Generator**

For the chosen generator polynomial,  $p(z) = z^6 + z + 1$ , the model generates a PN sequence of period 63, using both the library block and corresponding Simulink blocks. It shows how the two parameters, **Initial states** and **Output mask vector** (or scalar shift value), are interpreted in the latter schematic.

You can experiment with different initial states, by changing the value of **Initial states** prior to running the simulation. For all values, the two generated sequences are the same.

Using the PN Sequence Generator block allows you to easily generate PN sequences of large periods.



## 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.

#### 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.

#### **Enable bit-packed outputs**

When selected, the field **Number of packed bits** and the option **Interpret bit-packed values as signed** is enabled.

### Number of packed bits

Indicates how many bits to pack into each output data word (allowable range is 1 to 32).

# **PN Sequence Generator**

#### Interpret bit-packed values as signed

Indicates whether packed bits are treated as signed or unsigned integer data values. When selected, a 1 in the most significant bit (sign bit) indicates a negative value.

#### Output data type

By default, this is set to double.

When **Enable bit-packed outputs** is not selected, the output data type can be specified as a double, boolean, or Smallest unsigned integer. When the parameter is set to Smallest unsigned integer, the output data type is selected based on the settings used in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in the **Hardware Implementation** pane, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

When Enable bit-packed outputs is selected, the output data type can be specified as double or Smallest integer. When the parameter is set to Smallest integer, the output data type is selected based on Interpret bit-packed values as signed, Number of packed bits, and the settings used in the Hardware Implementation pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in the Hardware Implementation pane, the output data type is the ideal minimum n-bit size, i.e., sfix(n) or ufix(n), based on Interpret bit-packed values as signed. For all other selections, it is a signed or unsigned integer with the smallest available word length large enough to fit n bits.

## **See Also**

Kasami Sequence Generator, Scrambler

## **References**

[1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.

# **PN Sequence Generator**

[2] Lee, J. S., and L. E. Miller, *CDMA Systems Engineering Handbook*, Artech House, 1998.

[3] Golomb, S.W., Shift Register Sequences, Aegean Park Press, 1967.

# **Poisson Integer Generator**

**Purpose** 

Generate Poisson-distributed random integers

Library

Random Data Sources sublibrary of Comm Sources

**Description** 

Poisson int

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.

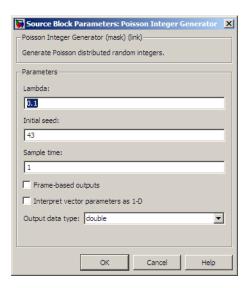
You can use the Poisson Integer Generator to generate noise in a binary transmission channel. In this case, the Poisson parameter **Lambda** should be less than 1, usually much less.

#### **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**

#### 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 kth element of the input vector does not become part of the output vector.
- If **Puncture vector**(k) = 1, then the kth 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.

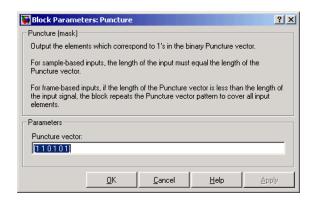
# **Frame-Based Processing**

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 = mod(k,length(**Puncture vector**))

and mod is the modulus function (mod in MATLAB).

## Dialog Box



#### **Puncture vector**

A binary vector whose pattern of 0s (1s) indicates which elements of the input the block should remove (preserve).

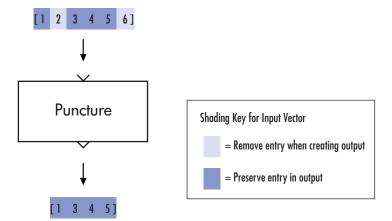
# **Examples**

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.

# **Puncture**



See Also Insert Zero

# **QPSK Demodulator Baseband**

**Purpose** 

Demodulate QPSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

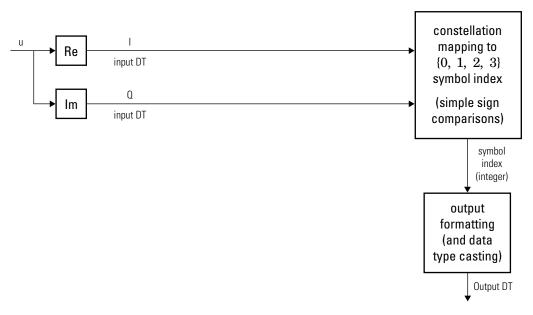
**Description** 



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

The input must be a discrete-time complex signal. The input can be either a scalar or a frame-based column vector. For information about the data types each block port supports, see the "Supported Data Types" on page 2-628 table on this page.

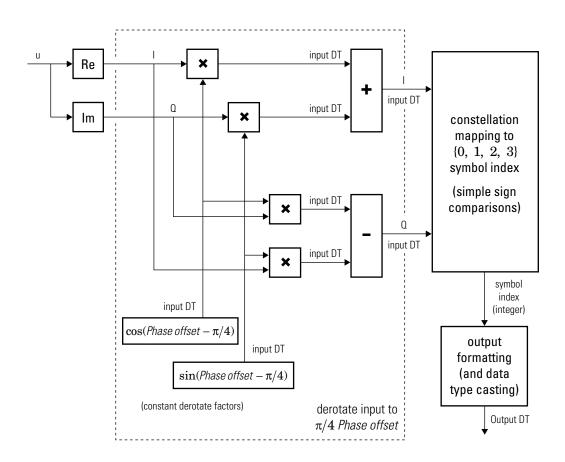
#### **Algorithm**



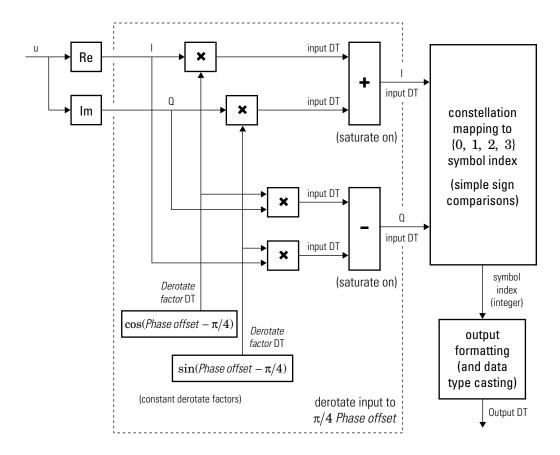
Hard-Decision QPSK Demodulator Signal Diagram for Trivial Phase

Offset (odd multiple of  $\frac{\pi}{4}$ )

# **QPSK Demodulator Baseband**



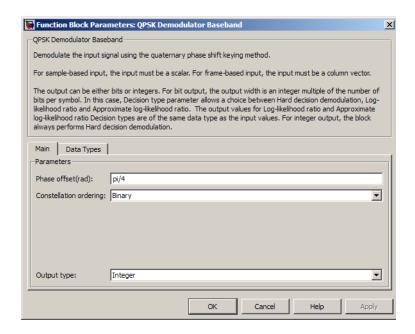
Hard-Decision QPSK Demodulator Floating-Point Signal Diagram for Nontrivial Phase Offset



#### Hard-Decision QPSK Demodulator Fixed-Point Signal Diagram for Nontrivial Phase Offset

The exact LLR and approximate LLR cases (soft-decision) are described in "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the *Communications Toolbox User's Guide*.

# Dialog Box



#### Phase offset (rad)

The phase of the zeroth point of the signal constellation.

#### Constellation ordering

Determines how the block maps each integer to a pair of output bits.

#### Output type

Determines whether the output consists of integers or bits.

If the **Output type** parameter is set to Integer and **Constellation ordering** is set to Binary, then the block maps the point

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

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

The reference page for the QPSK Modulator Baseband block shows the signal constellations for the cases when **Constellation ordering** is set to either Binary or Gray.

If the **Output type** is set to Bit, then the output contains pairs of binary values if **Decision type** is set to Hard decision. The most significant bit (i.e. the left-most bit in the vector), is the first bit the block outputs.

If the **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, then the output contains bitwise LLR or approximate LLR values, respectively.

#### **Decision type**

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" in the *Communications Toolbox User's Guide* 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 parameter appears when the **Noise variance source** is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

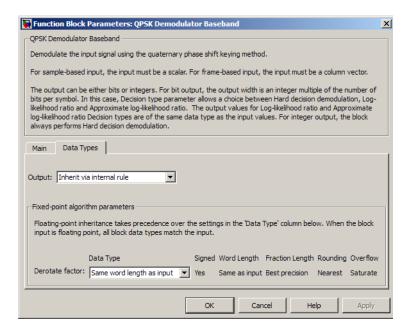
If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the

parameter without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:

- Inf to -Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.



**Data Types Pane for Hard-Decision** 

#### Output

For bit outputs, when **Decision type** is set to Hard decision, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, uint32, or boolean.

For integer outputs, the output data type can be set to 'Inherit via internal rule', 'Smallest unsigned integer', double, single, int8, uint8, int16, uint16, int32, or uint32.

When this parameter is set to 'Inherit via internal rule' (default setting), the block will inherit the output data type from the input port. The output data type will be the same as the input data type if the input is a floating-point type (single or double). If the input data type is fixed-point, the output data type will work as if this parameter is set to 'Smallest unsigned integer'.

When this parameter is set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the model.

If ASIC/FPGA is selected in the **Hardware Implementation** pane, and **Output type** is Bit, the output data type is the ideal minimum one-bit size, i.e., ufix(1). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit one bit, usually corresponding to the size of a char (e.g., uint8).

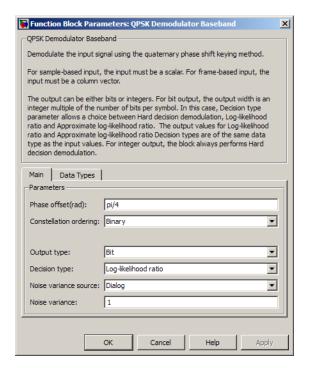
If ASIC/FPGA is selected in the **Hardware Implementation** pane, and **Output type** is Integer, the output data type is the ideal minimum two-bit size, i.e., ufix(2). For all other selections, it is an unsigned integer with the smallest available word length large enough to fit two bits, usually corresponding to the size of a char (e.g., uint8).

#### **Derotate factor**

This parameter only applies when the input is fixed-point and

**Phase offset** is not an odd multiple of  $\frac{\pi}{4}$ .

This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input.



#### **Data Types Pane for Soft-Decision**

For bit outputs, when **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, the output data type is inherited from the input (e.g., if the input is of data type double, the output is also of data type double).

# **Supported** Data **Types**

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	Signed fixed-point when <b>Decision type</b> is Hard-decision
Var	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Single-precision floating point
	Boolean when Output type is Bit and Decision type is Hard-decision
	• 8-, 16-, 32- bit signed integers
	• 8-, 16-, 32- bit unsigned integers
	• ufix(1) in ASIC/FPGA when <b>Output type</b> is Bit
	• ufix(2) in ASIC/FPGA when <b>Output type</b> is Integer

**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**

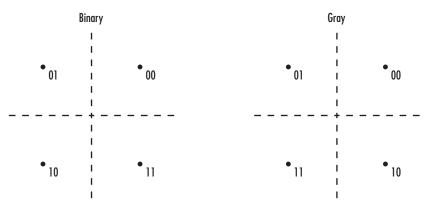
If the **Input type** parameter is set to **Integer**, then valid input values are 0, 1, 2, and 3. If **Constellation ordering** is set to Binary, for input m the output symbol is

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

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 information about the data types each block port supports, see the "Supported Data Types" on page 2-633 table on this page.

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.

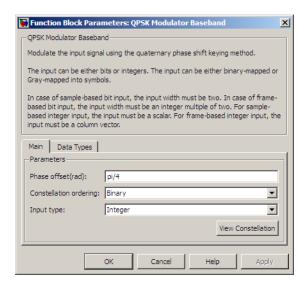


In the previous figure, the most significant bit (i.e. the left-most bit), is the first bit input to the block. For additional information about Gray mapping, see the Algorithm section of the M-PSK Modulator Baseband help page.

#### **Constellation Visualization**

The QPSK Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the *Communications Blockset User's Guide*.

# Dialog Box



#### Phase offset (rad)

The phase of the zeroth point of the signal constellation.

#### Constellation ordering

Determines how the block maps each pair of input bits or input integers to constellation symbols.

#### Input type

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

#### Output data type

The output data type can be set to double, single, Fixed-point, User-defined, or Inherit via back propagation.

Setting this parameter to Fixed-point or User-defined enables fields in which you can further specify details. Setting this parameter to Inherit via back propagation, sets the output data type and scaling to match the following block.

#### Output word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible when you select Fixed-point for the **Output data type** parameter.

#### Set output fraction length to

Specify the scaling of the fixed-point output by either of the following methods:

- Choose Best precision to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose User-defined to specify the output scaling in the Output fraction length parameter.

This parameter is only visible when you select Fixed-point for the **Output data type** parameter or when you select User-defined and the specified output data type is a fixed-point data type.

#### User-defined data type

Specify any signed built-in or signed fixed-point data type. You can specify fixed-point data types using the sfix, sint, sfrac, and fixdt functions from Simulink Fixed Point. This parameter is only visible when you select User-defined for the **Output** data type parameter.

#### Output fraction length

For fixed-point output data types, specify the number of fractional bits or bits to the right of the binary point. This parameter is only visible when you select Fixed-point or User-defined for the **Output data type** parameter and User-defined for the **Set output fraction length to** parameter.

# Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	Boolean when Input type is Bit
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	• ufix(2) when <b>Input type</b> is Integer
Output	Double-precision floating point
	Single-precision floating point
	Signed fixed point

Pair Block QPSK Demodulator Baseband

See Also M-PSK Modulator Baseband, BPSK Modulator Baseband, DQPSK

Modulator Baseband

# **Quantizing Decoder**

**Purpose** 

Decode quantization index according to codebook

Library

Source Coding

**Description** 

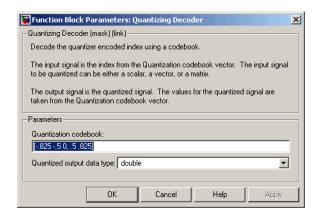


The Quantizing Decoder block converts quantization indices to the corresponding codebook values. The **Quantization codebook** parameter, a vector of length N, prescribes the possible output values. If the input is an integer k between 0 and N-1, then the output is the (k+1)st element of **Quantization codebook**.

The input must be a discrete-time signal. This block processes each vector element independently. For information about the data types each block port supports, see the "Supported Data Type" on page 2-635 table on this page.

**Note** The Quantizing Encoder block also uses a **Quantization codebook** parameter. The first output of that block corresponds to the input of Quantizing Decoder, while the second output of that block corresponds to the output of Quantizing Decoder.

# Dialog Box



# **Quantizing Decoder**

#### Quantization codebook

A real vector that prescribes the output value corresponding to each nonnegative integer of the input.

#### Quantized output data type

Select the output data type.

# Supported Data Type

Port	Supported Data Types
Idx	Double-precision floating point
	Single-precision floating point
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
Q(U)	Double-precision floating point
	Single-precision floating point

# **Pair Block**

Quantizing Encoder

# **See Also**

Scalar Quantizer (Obsolete) (Signal Processing Blockset documentation)

# **Quantizing Encoder**

# **Purpose**

Quantize signal using partition and codebook

# Library

Source Coding

# **Description**

U Quantizing Idx > Encoder Q(U) >

The Quantizing Encoder block quantizes the input signal according to the **Partition** vector and encodes the input signal according to the **Codebook** vector. This block processes each vector element independently. The input must be a discrete-time signal. This block processes each vector element independently. For information about the data types each block port supports, see the "Supported Data Type" on page 2-637 table on this page.

The first output is the quantization index. The second output is the quantized signal. The values for the quantized signal are taken from the **Codebook** vector.

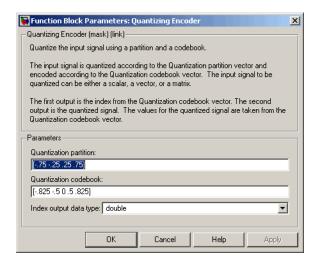
The **Quantization partition** parameter, P, is a real vector of length n whose entries are in strictly ascending order. The quantization index (second output signal value) corresponding to an input value of x is

- 0 if 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 lloyds in Communications Toolbox with a representative sample of your data as training data, to obtain appropriate partition and codebook parameters.

# Dialog Box



#### Quantization partition

The vector of endpoints of the partition intervals.

#### Quantization codebook

The vector of output values assigned to each partition.

# Index output data type

Select the output data type.

# Supported Data Type

Port	Supported Data Types
U	Double-precision floating point
	Single-precision floating point
	Signed fixed-point
Idx	Double-precision floating point
	• 8-, 16-, and 32-bit signed integers

# **Quantizing Encoder**

Port	Supported Data Types
	• 8-, 16-, and 32-bit unsigned integers
Q(U)	Double-precision floating point
	Single-precision floating point
	Signed fixed-point

Pair Block

Quantizing Decoder

**See Also** 

Scalar Quantizer (Obsolete) (Signal Processing Blockset documentation), 11oyds (Communications Toolbox documentation)

#### **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 FIR Decimation block implements this functionality. The Raised Cosine Receive Filter block's icon shows the filter's impulse response.

#### Characteristics of the Filter

Characteristics of the raised cosine filter are the same as in the Raised Cosine Transmit Filter block, except that the length of the filter's input response has a slightly different expression: 2 \* N \* Group delay + 1, where N is the value of the Input samples per symbol parameter (not the Upsampling factor parameter, as in the case of the Raised Cosine Transmit Filter block).

If the **Filter gain** parameter is chosen to be **User-specified**, then the passband gain of the filter is:

- 20 log<sub>10</sub>(Input samples per symbol(N)×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. By default, downsampling is on. 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.

• If the **Sample offset** parameter is a positive integer less than L, then the block initially discards that number of samples from the filtered signal and downsamples the remaining data as in the previous case.

To preserve the entire filtered signal and avoid downsampling, set **Output mode** to None. This setting is appropriate, for example, when the output from the filter block forms the input to a timing phase recovery block such as Squaring Timing Recovery. The timing phase recovery block performs the downsampling in that case.

## **Input and Output Signals**

The input signal must be a scalar value or a frame-based column vector. For information about the data types each block port supports, see the "Supported Data Type" on page 2-647 table on this page. You can select either Maintain input frame size or Maintain input frame rate for the **Framing** parameter.

If you set **Output mode** to None, then the input and output signals share the same sampling mode, sample time, and vector length.

If you set **Output mode** to Downsampling and **Downsampling factor** is L, then L and the input sampling mode determine characteristics of the output signal:

- 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.

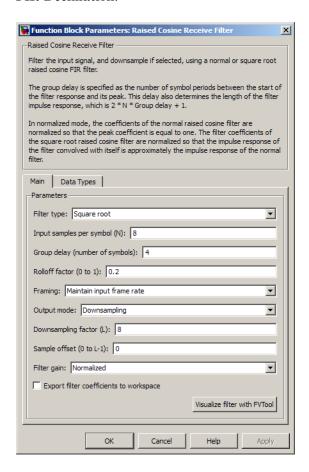
## **Exporting Filter Coefficients to the MATLAB Workspace**

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.

# Latency

For information pertaining to the latency of the block, see details in FIR Decimation.

# 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.

#### **Framing**

Selects the framing method. The framing method choices are: Maintain input frame size or Maintain input frame rate.

#### Output mode

Determines whether or not the block downsamples the signal after filtering. Choices are Downsampling and None.

#### **Downsampling factor**

The factor by which the block downsamples the signal after filtering. This field appears only if **Output mode** is set to Downsampling.

## Sample offset

The number of filtered samples the block discards before downsampling. This field appears only if **Output mode** is set to Downsampling.

#### Filter gain

Determines how the block scales the filter coefficients. Choices are Normalized and User-specified.

#### Linear amplitude filter gain

A positive scalar used to scale the filter coefficients. This field appears only if **Filter gain** is set to User-specified.

## Export filter coefficients to workspace

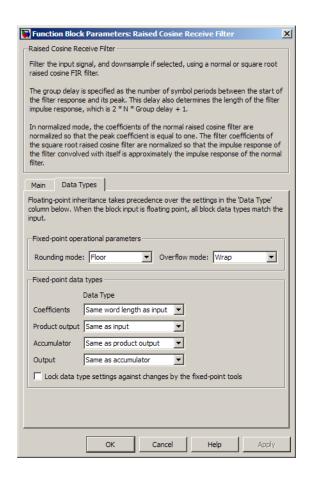
If you check this box, then the block creates a variable in the MATLAB workspace that contains the filter coefficients.

#### 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 block uses the **Rounding mode** when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. The filter coefficients do not obey this parameter; they always round to Nearest. For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink Fixed Point documentation.

#### Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

#### Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator).

See the Coefficients section of the FIR Decimation help page and "Filter Structure Diagrams" in *Signal Processing Blockset Reference Guide* for illustrations depicting the use of the coefficient data types in this block:

See the Coefficients subsection of the Digital Filter help page for descriptions of parameter settings.

- When you select Same word length as input, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select Specify word length, you are able to enter
  the word length of the coefficients, in bits. In this mode, the
  fraction length of the coefficients is automatically set to the
  binary-point only scaling that provides you with the best
  precision possible given the value and word length of the
  coefficients.
- When you select Binary point scaling, you are able to enter the word length and the fraction length of the coefficients, in bits. If applicable, you are able to enter separate fraction lengths for the numerator and denominator coefficients.
- When you select Slope and bias scaling, you are able to enter the word length, in bits, and the slope of the coefficients.
   If applicable, you are able to enter separate slopes for the

numerator and denominator coefficients. This block requires power-of-two slope and a bias of zero.

 The filter coefficients do not obey the Rounding mode and the Overflow mode parameters; they are always saturated and rounded to Nearest.

#### **Product output**

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Filter Structure Diagrams" and "Multiplication Data Types" in *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.

#### Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Signed fixed-point
Out	Double-precision floating point
	Single-precision floating point
	Signed fixed-point

## **Pair Block**

Raised Cosine Transmit Filter

# 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

$$h(t) = \frac{\sin(\pi t/T)}{(\pi t/T)} \cdot \frac{\cos(\pi R t/T)}{(1 - 4R^2t^2/T^2)}$$

The impulse response of a square root raised cosine filter with rolloff factor R is

$$h(t) = 4R \frac{\cos\left((1+R)\pi t/T\right) + \frac{\sin\left((1-R)\pi t/T\right)}{(4Rt/T)}}{\pi\sqrt{T}\left(1-(4Rt/T)^2\right)}$$

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 \* 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

bandwidth of the filter. For example, a rolloff factor of .5 means that the bandwidth of the filter is 1.5 times the input sampling frequency.

The **Filter gain** parameter indicates how the block normalizes the filter coefficients. If you choose Normalized, then the block uses an automatic scaling:

- If **Filter type** is **Normal**, then the block normalizes the filter coefficients so that the peak coefficient equals 1.
- If **Filter type** is Square root, then the block normalizes the filter coefficients so that the convolution of the filter with itself produces a normal raised cosine filter whose peak coefficient equals 1.

As a result, the output is scaled by  $\sqrt{N}$ . If the output of this block feeds the input to the AWGN Channel block, specify the AWGN signal power parameter to be 1/N.

If the **Filter gain** parameter is chosen to be User-specified, then the passband gain of the filter is:

- 20 log<sub>10</sub>(Upsampling factor(N)×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 must be a discrete-time signal. This block processes each vector element independently. For information about the data types each block port supports, see the "Supported Data Type" on page 2-657 table on this page. You can select either Maintain input frame size or Maintain input frame rate for the **Framing** parameter.

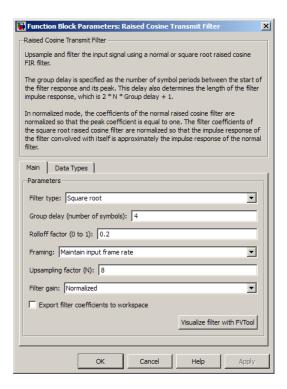
The Framing method and the value of the **Upsampling factor (N)** parameter determine the characteristics of the output signal:

- 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.

#### Rolloff factor

The rolloff factor for the filter, a real number between 0 and 1.

#### Framing

Selects the framing method. The framing method choices are: Maintain input frame size or Maintain input frame rate.

#### **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

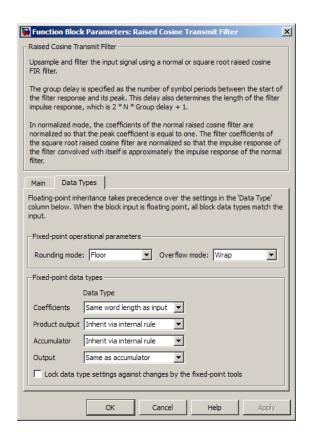
If you check this box, then the block creates a variable in the MATLAB workspace that contains the filter coefficients.

#### 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 block uses the **Rounding mode** when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. The filter coefficients do not obey this parameter; they always round to Nearest. For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink Fixed Point documentation.

#### Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

#### Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator). See "Filter Structure Diagrams" in *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.

#### Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Signed fixed-point
Out	Double-precision floating point
	Single-precision floating point
	Signed fixed-point

## **Pair Block**

Raised Cosine Receive Filter

# **See Also**

Gaussian Filter, rcosine, rcosflt

# **Random Deinterleaver**

#### **Purpose**

Restore ordering of input symbols using random permutation

# Library

Block sublibrary of Interleaving

# **Description**

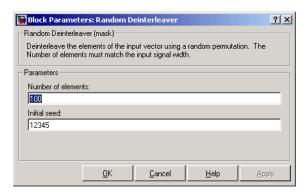


The Random Deinterleaver block rearranges the elements of its input vector using a random permutation. The **Initial seed** parameter initializes the random number generator that the block uses to determine the permutation. If this block and the Random Interleaver block have the same value for **Initial 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**

## See Also General Block Deinterleaver

# **Random Integer Generator**

**Purpose** 

Generate integers randomly distributed in range [0, M-1]

Library

Random Data Sources sublibrary of Comm Sources

**Description** 

Pandom int

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.

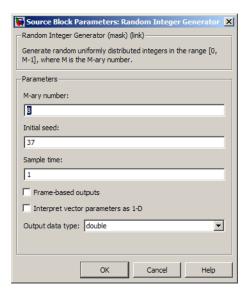
If the **Initial seed** parameter is a constant, then the resulting noise is repeatable.

## **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 BlocksetUser'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.

### M-ary number

The positive integer, or vector of positive integers, that indicates the range of output values.

#### Initial seed

The initial seed value for the random number generator. The vector length of the seed determines the length of the output vector.

### Sample time

The period of each sample-based vector or each row of a frame-based matrix.

#### Frame-based outputs

Determines whether the output is frame-based or sample-based. This box is active only if **Interpret vector parameters as 1-D** is unchecked.

# **Random Integer Generator**

### Samples per frame

The number of samples in each column of a frame-based output signal. This field is active only if **Frame-based outputs** is checked.

### Interpret vector parameters as 1-D

If this box is checked, then the output is a one-dimensional signal. Otherwise, the output is a two-dimensional signal. This box is active only if **Frame-based outputs** is unchecked.

### Output data type

The output type of the block can be specified as a boolean, int8, uint8, int16, uint16, int32, uint32, single, or double. By default, the block sets this to double. Single outputs may lead to different results when compared with double outputs for the same set of parameters. For Boolean typed outputs, the M-ary number must be 2.

## See Also randint (Communications Toolbox)

**Purpose** 

Reorder input symbols using random permutation

Library

Block sublibrary of Interleaving

## **Description**

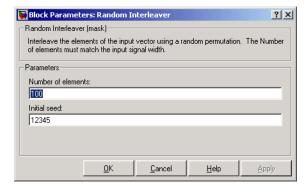


The Random Interleaver block rearranges the elements of its input vector using a random permutation. The **Number of elements** parameter indicates how many numbers are in the input vector. If the input is frame-based, then it must be a column vector.

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

# **Random Interleaver**

See Also

General Block Interleaver

# **Rayleigh Noise Generator**

**Purpose** 

Generate Rayleigh distributed noise

Library

Noise Generators sublibrary of Comm Sources

Description

Rayleigh

The Rayleigh Noise Generator block generates Rayleigh distributed noise. The Rayleigh probability density function is given by

$$f(x) = \begin{cases} \frac{x}{\sigma^2} \exp\left(-\frac{x^2}{2\sigma^2}\right) & x \ge 0\\ 0 & x < 0 \end{cases}$$

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.

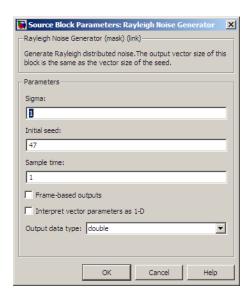
# **Rayleigh Noise Generator**

## **Attributes of Output Signal**

The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the **Frame-based outputs**, **Samples per frame**, and **Interpret vector parameters as 1-D** parameters. See "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; raylrnd (Statistics Toolbox)

## References

[1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.

## **Receiver Thermal Noise**

## **Purpose**

Apply receiver thermal noise to complex baseband signal

## Library

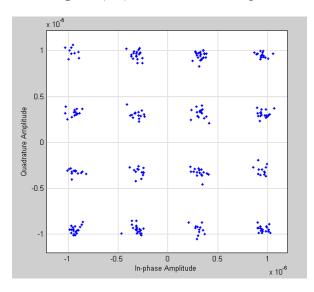
RF Impairments

## **Description**

Noise > Temperature 290 K The Receiver Thermal Noise block simulates the effects of thermal noise on a complex, baseband signal. You can specify the amount of thermal noise in three ways, according to which **Specification method** you select:

- Noise temperature specifies the noise in degrees kelvin.
- Noise factor specifies the noise as 1+(Noise temperature / 290).
- Noise figure specifies the noise as 10\*log10(1+(Noise temperature / 290)). This is the decibel equivalent of Noise factor.

The following scatter plot shows the effect of the Receiver Thermal Noise block, with **Specification method** set to Noise figure and **Noise figure (dB)** set to 3.01, on a signal modulated by 16-QAM.

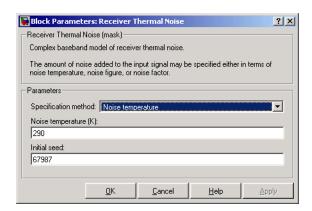


## **Receiver Thermal Noise**

This plot is generated by the model described in "Scatter Plot Examples" with the following parameter settings:

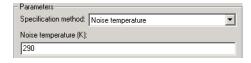
- Rectangular QAM Modulator Baseband
  - Normalization method set to Average Power
  - Average power (watts) set to 1e-12
- Receiver Thermal Noise
  - Specification method set to Noise figure
  - Noise figure (dB) set to 3.01

## 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.

## **Receiver Thermal Noise**



### Noise figure

Scalar specifying the amount of noise in decibels relative to a noise temperature of 290 degrees kelvin. A **Noise figure** setting of 0 dB indicates a noiseless system.



#### Noise factor

Scalar specifying the amount of noise relative to a noise temperature of 290 degrees kelvin.

#### Initial seed

The initial seed value for the random number generator that generates the noise.

# **See Also** Free Space Path Loss

**Purpose** 

Demodulate rectangular-QAM-modulated data

Library

AM, in Digital Baseband sublibrary of Modulation

**Description** 

WW Rectangular

The Rectangular QAM Demodulator Baseband block demodulates a signal that was modulated using quadrature amplitude modulation with a constellation on a rectangular lattice.

**Note** All values of power assume a nominal impedance of 1 ohm.

The signal constellation has M points, where M is the **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. For information about the data types each block port supports, see the "Supported Data Types" on page 2-681 table on this page.

## 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

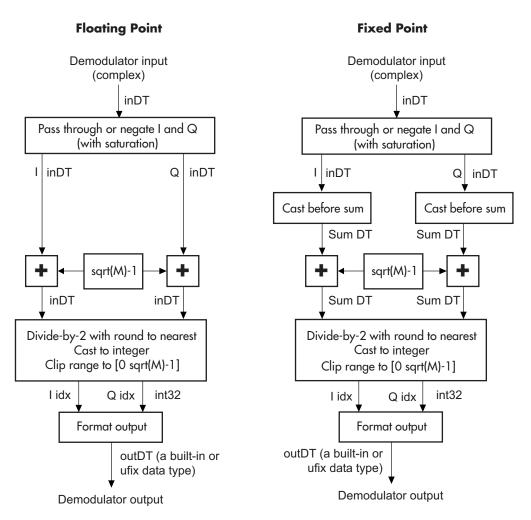
between 0 and  $2\times(\sqrt{M}-1)$  (plus noise). The resulting values are then rescaled via a divide-by-two (or, equivalently, a right-shift by one bit for fixed-point operation) to obtain a range approximately between 0

and  $\sqrt{M}-1$  (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  $\boldsymbol{Output\ data\ type}.$ 

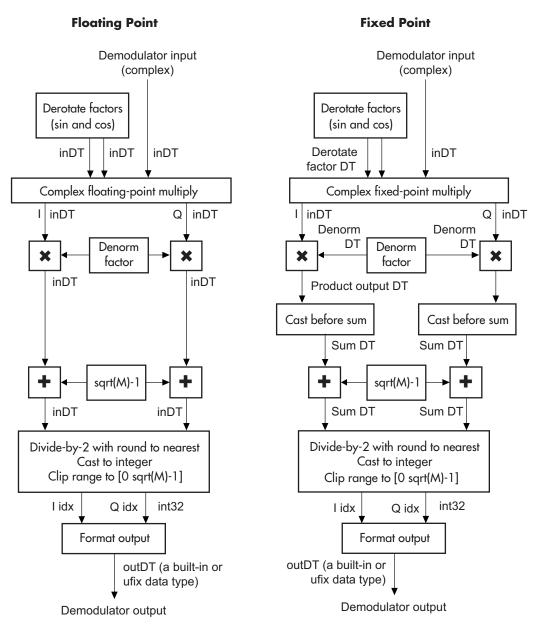
The following figures contains signal flow diagrams for floating-point and fixed-point algorithm operation. The floating-point diagrams apply when the input signal data type is double or single. The fixed-point diagrams apply when the input signal is a signed fixed-point data type. Note that the diagram is simplified when **Phase offset** is a multiple of

 $\frac{\pi}{}$ 

 $\overline{2}$ , and/or the derived denormalization factor is 1.

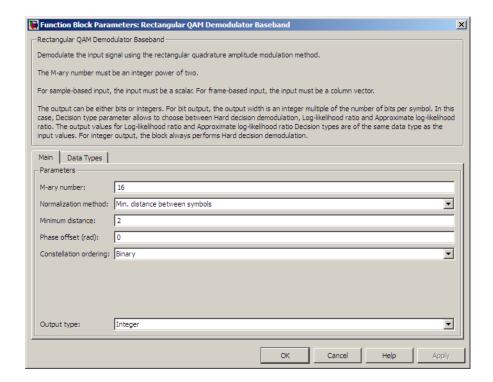


Signal-Flow Diagrams with Trivial Phase Offset and Denormalization Factor Equal to 1



Signal-Flow Diagrams with Nontrivial Phase Offset and Nonunity Denormalization Factor

## 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 parameter appears when **Normalization method** is set to Min. distance between symbols.

The distance between two nearest constellation points.

### Average power, referenced to 1 ohm (watts)

The average power of the symbols in the constellation, referenced to 1 ohm. This field appears only when **Normalization method** is set to Average Power.

### Peak power, referenced to 1 ohm (watts)

The maximum power of the symbols in the constellation, referenced to 1 ohm. This field appears only when **Normalization** method is set to Peak Power.

#### Phase offset (rad)

The rotation of the signal constellation, in radians.

#### Constellation ordering

Determines how the block assigns binary words to points of the signal constellation. More details are on the reference page for the Rectangular QAM Modulator Baseband block.

Selecting User-defined displays the field Constellation mapping, allowing for user-specified mapping.

### Constellation mapping

This parameter appears when User-defined is selected in the pull-down list Constellation ordering.

This is a row or column vector of size M and must have unique integer values in the range [0, 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.

If set to Integer, the block produces integers.

If set to Bit, the block produces a group of K bits, called a *binary word*, for each symbol, when **Decision type** is set to Hard decision. If **Decision type** is set to Log-likelihood ratio or Approximate log-likelihood ratio, the block outputs bitwise LLR and approximate LLR, respectively.

### **Decision type**

This parameter appears when Bit is selected in the pull-down list **Output type**.

Specifies the use of hard decision, LLR, or approximate LLR during demodulation. See "Exact LLR Algorithm" and "Approximate LLR Algorithm" in the *Communications Toolbox User's Guide* for algorithm details.

#### Noise variance source

This parameter appears when Approximate log-likelihood ratio or Log-likelihood ratio is selected for **Decision type**.

When set to Dialog, the noise variance can be specified in the **Noise variance** field. When set to Port, a port appears on the block through which the noise variance can be input.

#### Noise variance

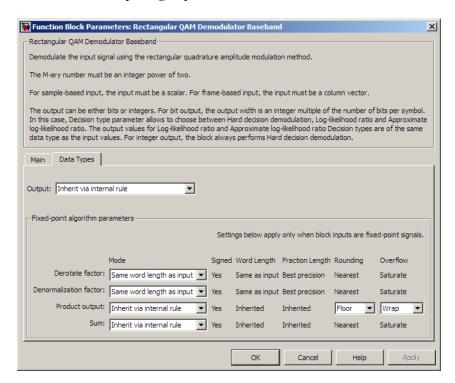
This parameter appears when the **Noise variance source** is set to Dialog and specifies the noise variance in the input signal. This parameter is tunable in normal mode, Accelerator mode and Rapid Accelerator mode.

If you use the Real-Time Workshop® rapid simulation (RSIM) target to build an RSIM executable, then you can tune the parameter without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise.

The LLR algorithm involves computing exponentials of very large or very small numbers using finite precision arithmetic and would yield:

- Inf to -Inf if Noise variance is very high
- NaN if Noise variance and signal power are both very small

In such cases, use approximate LLR, as its algorithm does not involve computing exponentials.



#### 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 model. If ASIC/FPGA is selected in the **Hardware Implementation** pane, the output data type is the ideal minimum size, i.e., ufix(1) for bit outputs, and ufix(ceil(log2(M))) for integer outputs. For all other selections, it is an unsigned integer with the smallest available word length large enough to fit the ideal minimum size, usually corresponding to the size of a char (e.g., uint8).

For integer outputs, this parameter can be set to Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, single, and double. For bit outputs, the options are Smallest unsigned integer, int8, uint8, int16, uint16, int32, uint32, boolean, single, or double.

#### **Derotate factor**

This parameter only applies when the input is fixed-point and

 $\frac{\pi}{2}$ 

**Phase offset** is not a multiple of  $\frac{1}{2}$ .

This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input.

#### **Denormalization factor**

This parameter only applies when the input is fixed-point and the derived denormalization factor is nonunity (not equal to 1). This scaling factor is derived from **Normalization method** and other parameter values in the block dialog.

This can be set to Same word length as input or Specify word length, in which case a field is enabled for user input. A best-precision fraction length is always used.

#### **Product output**

This parameter only applies when the input is a fixed-point signal and there is a nonunity (not equal to 1) denormalized factor. It

can be set to Inherit via internal rule or Specify word length, which enables a field for user input.

Setting to Inherit via internal rule computes the full-precision product word length and fraction length. Inherit Rule for Product Data Types in *Signal Processing Blockset User's Guide* describes the full-precision **Product output** internal rule.

Setting to Specify word length allows you to define the word length. The block computes a best-precision fraction length based on the word length specified and the pre-computed worst-case (min/max) real world value **Product output** result. The worst-case **Product output** result is precomputed by multiplying the denormalized factor with the worst-case (min/max) input signal range, purely based on the input signal data type.

The block uses the **Rounding mode** when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink Fixed Point documentation.

#### Sum

This parameter only applies when the input is a fixed-point signal. It can be set to Inherit via internal rule, Same as product output, or Specify word length, in which case a field is enabled for user input

Setting to Inherit via internal rule computes the full-precision sum word length and fraction length, based on the two inputs to the Sum in the fixed-point Hard Decision Algorithm signal flow diagram. The rule is the same as the fixed-point inherit rule of the internal **Accumulator data type** parameter in the SimulinkSimulinkSimulink Sum block.

Setting to Specify word length allows you to define the word length. A best precision fraction length is computed based on the word length specified in the pre-computed maximum range necessary for the demodulated algorithm to produce accurate results. The signed fixed-point data type that has the best

precision fully containing the values in the range  $2*(\sqrt{M}-1)$  for the specified word length.

Setting to Same as product output allows the Sum data type to be the same as the **Product output** data type (when **Product output** is used). If the **Product output** is not used, then this setting will be ignored and the Inherit via internal rule Sum setting will be used.

## Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	Signed fixed-point
Var	Double-precision floating point
	Single-precision floating point
Output	Double-precision floating point
	Single-precision floating point
	Boolean when Output type is Bit
	• 8-, 16-, and 32-bit signed integers
	• 8-, 16-, and 32-bit unsigned integers
	• ufix(1) in ASIC/FPGA when <b>Output type</b> is Bit
	• $ufix(\lceil \log_2 M \rceil)$ in ASIC/FPGA when <b>Output type</b> is Integer

Pair Block Rectangular QAM Modulator Baseband

See Also General QAM Demodulator Baseband

**References** [1] Smith, Joel G., "Odd-Bit Quadrature Amplitude-Shift Keying," *IEEE* 

Transactions on Communications, Vol. COM-23, March 1975, 385–389.

**Purpose** 

Modulate using rectangular quadrature amplitude modulation

Library

AM, in Digital Baseband sublibrary of Modulation

**Description** 

Rectangular > QAM

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.

**Note** All values of power assume a nominal impedance of 1 ohm.

## **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

## **Input Signal Values**

The input and output for this block are discrete-time signals.

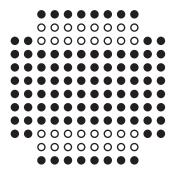
The **Input type** parameter determines whether the block accepts integers between 0 and M-1, or binary representations of integers:

- If **Input type** is set to **Integer**, the block accepts integers. The input can be either a scalar or a frame-based column vector.
- 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 information about the data types each block port supports, see the "Supported Data Types" on page 2-689 table on this page.

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.



Hollow vertical pairs of adjacent points differ by two bits

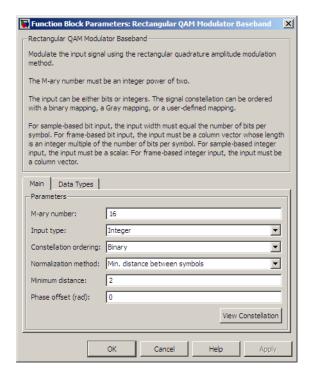
Other pairs of adjacent points differ by one bit

For details about the Gray coding, see the reference page for the M-PSK Modulator Baseband block and the paper listed in References. Because the in-phase and quadrature components are assigned independently, the Gray and binary orderings coincide when M = 4.

#### **Constellation Visualization**

The Rectangular QAM Modulator Baseband block provides the capability to visualize a signal constellation from the block mask. This Constellation Visualization feature allows you to visualize a signal constellation for specific block parameters. For more information, see the Constellation Visualization section of the *Communications Blockset User's Guide*.

## 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 left to right. The last element corresponds to the bottom-rightmost point.

This field appears when User-defined is selected in the drop-down list **Constellation ordering**.

#### Normalization method

Determines how the block scales the signal constellation. Choices are Min. distance between symbols, Average Power, and Peak Power.

#### Minimum distance

The distance between two nearest constellation points. This field appears only when **Normalization method** is set to Min. distance between symbols.

### Average power, referenced to 1 ohm (watts)

The average power of the symbols in the constellation, referenced to 1 ohm. This field appears only when **Normalization method** is set to Average Power.

#### Peak power, referenced to 1 ohm (watts)

The maximum power of the symbols in the constellation, referenced to 1 ohm. This field appears only when **Normalization** method is set to Peak Power.

### Phase offset (rad)

The rotation of the signal constellation, in radians.

#### 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 software. This parameter is only visible when you select User-defined for the **Output data type** parameter.

### Set output fraction length to

Specify the scaling of the fixed-point output by either of the following methods:

- Choose Best precision to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose User-defined to specify the output scaling in the Output fraction length parameter.

This parameter is only visible when you select Fixed-point for the **Output data type** parameter or when you select User-defined and the specified output data type is a fixed-point data type.

#### Output fraction length

For fixed-point output data types, specify the number of fractional bits, or bits to the right of the binary point. This parameter is only visible when you select Fixed-point or User-defined for the **Output data type** parameter and User-defined for the **Set output fraction length to** parameter.

## Supported Data Types

Port	Supported Data Types
Input	Double-precision floating point
	Single-precision floating point
	Boolean when Input type is Bit
	• 8-, 16-, 32-bit signed integers
	• 8-, 16-, 32-bit unsigned integers
	• $\mathit{ufix}(\lceil \log_2 M \rceil)$ when $\mathbf{Input}\ \mathbf{type}\ \mathrm{is}\ Integer$
Output	Double-precision floating point
	Single-precision floating point
	Signed fixed-point

Pair Block Rectangular QAM Demodulator Baseband

See Also General QAM Modulator Baseband

**References** [1] Smith, Joel G., "Odd-Bit Quadrature Amplitude-Shift Keying," *IEEE Transactions on Communications*, Vol. COM-23, March 1975, 385–389.

# **Rectangular QAM TCM Decoder**

### **Purpose**

Decode trellis-coded modulation data, modulated using QAM method

## Library

Trellis-Coded Modulation

## **Description**

Rectangular QAM TCM 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,  $\log_2(\mathbf{M-ary\ number})$  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 theRectangular 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.

# **Rectangular QAM TCM Decoder**

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.

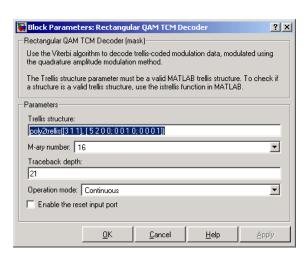
- In Truncated mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In Terminated mode, the block treats each frame independently. The traceback path always starts at the all-zeros 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

#### 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.

# **Rectangular QAM TCM Encoder**

### **Purpose**

Convolutionally encode binary data and modulate using QAM method

## Library

Trellis-Coded Modulation

## **Description**

Rectangular QAM TCM The Rectangular QAM TCM Encoder block implements trellis-coded modulation (TCM) by convolutionally encoding the binary input signal and mapping the result to a QAM signal constellation.

The **M-ary number** parameter is the number of points in the signal constellation, which also equals the number of possible output symbols from the convolutional encoder. (That is,  $\log_2(\mathbf{M-ary\ number})$  is equal to n for a rate 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\*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.

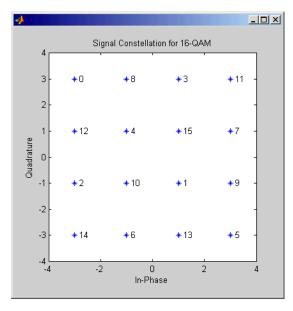
The encoder registers begin in the all-zeros state. You can configure the encoder so that it resets its registers to the all-zeros state during the course of the simulation. To do this, set the Operation mode to Reset on nonzero input via port. The block then opens a second input port, labeled Rst. The signal at the Rst port is a scalar signal. When it is nonzero, the encoder resets before processing the data at the first input port.

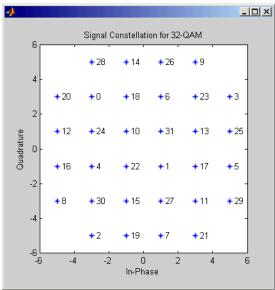
## **Signal Constellations**

The trellis-coded modulation technique partitions the constellation into subsets called cosets, so as to maximize the minimum distance between pairs of points in each coset. This block internally forms a valid partition based on the value you choose for the **M-ary number** parameter.

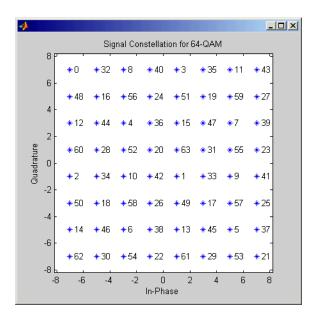
The figures below show the labeled set-partitioned signal constellations that the block uses when **M-ary number** is 16, 32, and 64. For constellations of other sizes, see Biglieri, E., D. Divsalar, P. J. McLane and M. K. Simon, *Introduction to Trellis-Coded Modulation with Applications*, New York, Macmillan, 1991.

# **Rectangular QAM TCM Encoder**





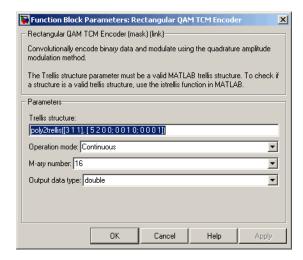
# **Rectangular QAM TCM Encoder**



## **Coding Gains**

Coding gains of 3 to 6 decibels, relative to the uncoded case can be achieved in the presence of AWGN with multiphase trellis codes. For more information, see Biglieri, E., D. Divsalar, P. J. McLane and M. K. Simon, *Introduction to Trellis-Coded Modulation with Applications*, New York, Macmillan, 1991.

## Dialog Box



#### Trellis structure

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

## Operation mode

In Continuous mode (default setting), the block retains the encoder states at the end of each frame, for use with the next frame.

In Truncated (reset every frame) mode, the block treats each frame independently. I.e., the encoder states are reset to all-zeros state at the start of each frame.

In Terminate trellis by appending bits mode, the block treats each frame independently. For each input frame, extra bits are used to set the encoder states to all-zeros state at the end of the frame. The output length is given by

 $y = n \cdot (x+s)/k$ , where x is the number of input bits, and

s = constraint length - 1 (or, in the case of multiple constraint

# Rectangular QAM TCM Encoder

lengths, s = sum(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.

### 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.

**Purpose** Resample input at higher rate by repeating values

**Library** Signal Operations

**Description** The Filter block is a Signal Processing Blockset block. For more

information, see the Repeat block reference page in the Signal

Processing Blockset documentation.

# **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



$$f(x) = \begin{cases} \frac{x}{\sigma^2} I_0 \left( \frac{mx}{\sigma^2} \right) \exp \left( -\frac{x^2 + m^2}{2\sigma^2} \right) & x \ge 0 \\ 0 & x < 0 \end{cases}$$

where:

- σ 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
- ullet  $I_0$  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

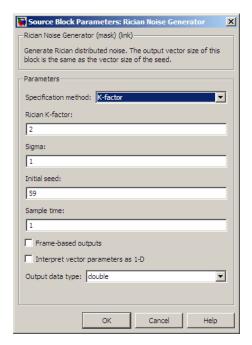
This returns a random prime number greater than 30. Entering randseed again produces a different prime number. If you supply an integer argument, randseed always returns the same prime for that integer. For example, randseed (5) always returns the same answer.

## **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.

## 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  $m_{\rm I}$  and  $m_{\rm Q}$ , respectively, of the Gaussian components. These fields appear only if **Specification method** is Quadrature components.

## **Rician Noise Generator**

#### Sigma

The variable  $\sigma$  in the Rician probability density function.

#### Initial seed

The initial seed value for the random number generator.

#### Sample time

The period of each sample-based vector or each row of a frame-based matrix.

### Frame-based outputs

Determines whether the output is frame-based or sample-based. This box is active only if **Interpret vector parameters as 1-D** is unchecked.

### Samples per frame

The number of samples in each column of a frame-based output signal. This field is active only if **Frame-based outputs** is checked.

### Interpret vector parameters as 1-D

If this box is checked, then the output is a one-dimensional signal. Otherwise, the output is a two-dimensional signal. This box is active only if **Frame-based outputs** is unchecked.

### Output data type

The output can be set to double or single data types.

## **See Also** Multipath Rician Fading Channel

## References

[1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.

## **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 BlocksetUser'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 BlocksetUser'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*.

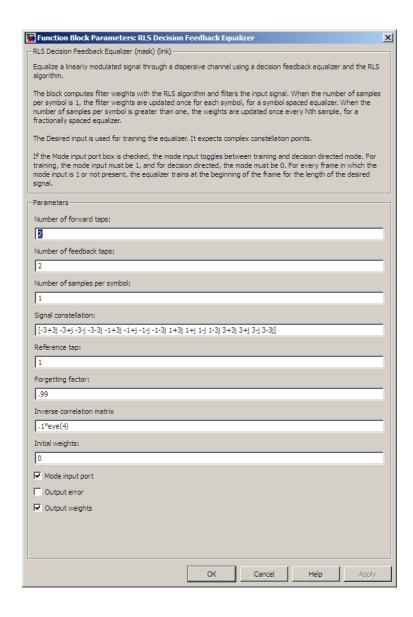
## **Equalizer Delay**

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+(Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

## 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.

## 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**

## **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*.

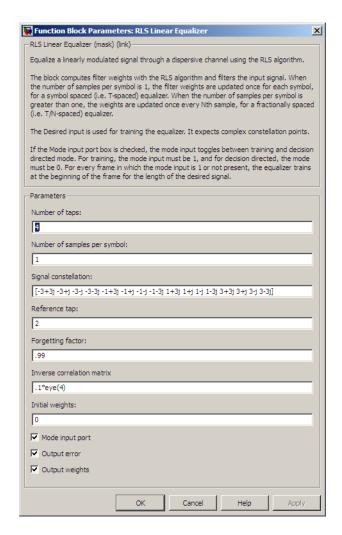
### **Equalizer Delay**

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+(Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## 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.

# **RLS Linear Equalizer**

[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

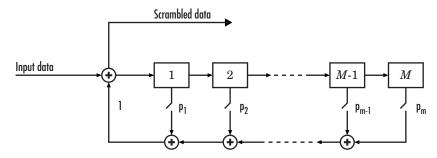
Sequence Operations

**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  $p(z^1) = 1 + p_1 z^1 + p_2 z^2 + ...$ , 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

## **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 BlocksetUser's Guide.

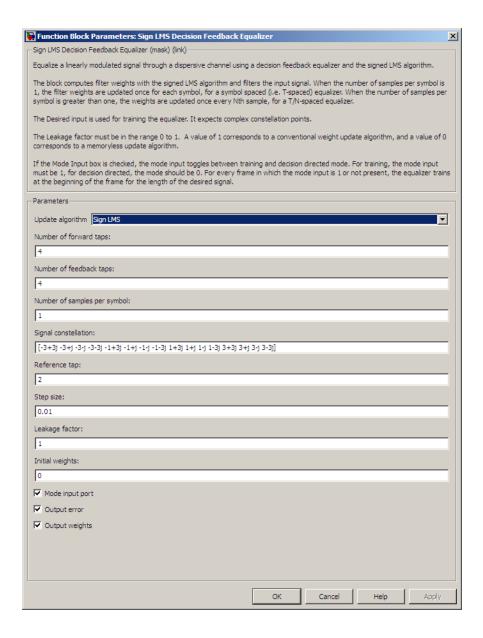
## **Equalizer Delay**

For proper equalization, you should set the **Reference tap** parameter so that it exceeds the delay, in symbols, between the transmitter's modulator output and the equalizer input. When this condition is satisfied, the total delay, in symbols, between the modulator output and the equalizer *output* is equal to

### 1+(Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

## Dialog Box



### Update algorithm

The specific type of signed LMS algorithm that the block uses to update the equalizer weights.

#### 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.

### 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

# Sign LMS Linear 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

- Mode input, as described in "Controlling the Use of Training or Decision-Directed Mode" in *Communications BlocksetUser'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*.

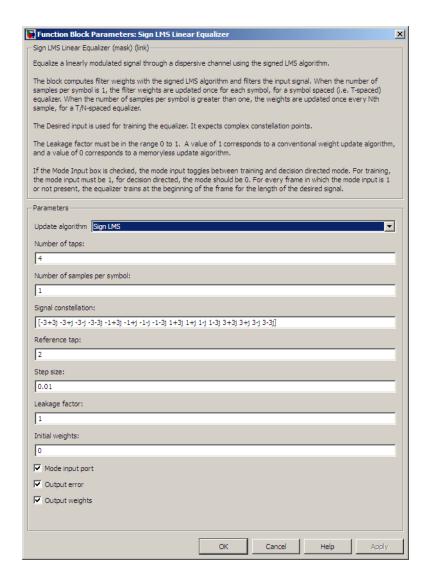
## **Equalizer Delay**

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+(Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## Dialog Box



# Sign LMS Linear Equalizer

### Update algorithm

The specific type of signed LMS algorithm that the block uses to update the equalizer weights.

#### 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.

# Sign LMS Linear Equalizer

**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**

Squaring Sym > Timing Recovery Ph 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.

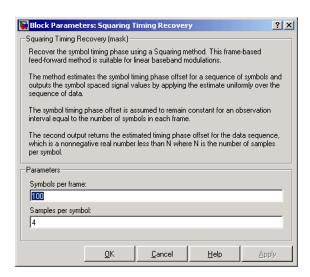
**Note** The block assumes that the phase offset is constant for all symbols in the entire input frame. If necessary, use the Buffer block to reorganize your data into frames over which the phase offset can be assumed constant. If the assumption of constant phase offset is valid, then a larger frame length yields a more accurate phase offset estimate.

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

$$-\frac{1}{2\pi} \mathrm{arg} \Biggl( \sum_{m=0}^{\mathrm{LN-1}} \bigl| x_{m+1} \bigr|^2 \exp(-\mathrm{j} 2\pi m/\mathrm{N}) \Biggr)$$

# **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

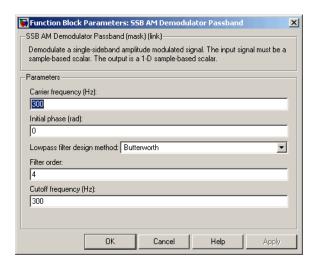
**Description** 



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 Tranform 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

$$y = u + jH\{u\}$$

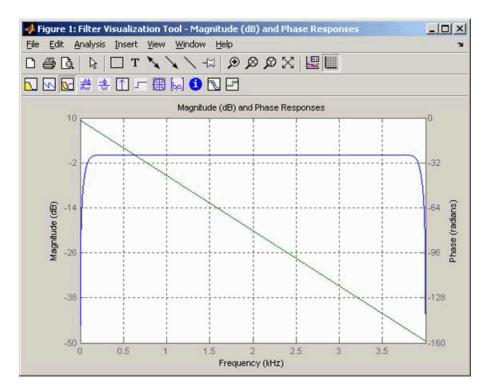
where  $j = \sqrt{-1}$  and H{} denotes the Hilbert transform. The real part of the output in each channel is a replica of the real input in that

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.

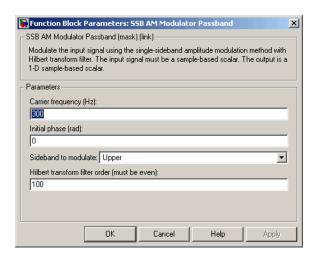


Note the bandwidth of the filter's magnitude response. By choosing a carrier frequency larger than 10% (but less than 90%) of the input signal's sample time (8000 samples per second, in this example) or equivalently, a carrier frequency larger than 400Hz, we ensure that the Hilbert Transform Filter will 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 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** 

Uniform :

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.

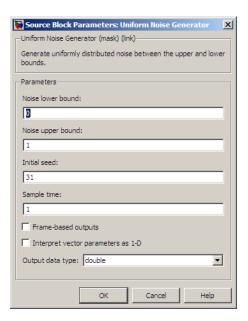
If all the elements of the output vector are to be independent and identically distributed (i.i.d.), then you can use a scalar for the **Noise lower bound** and **Noise upper bound** parameters. Alternatively, you can specify the range for each element of the output vector individually, by using vectors 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 BlocksetUser'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 documentation); rand (built-in MATLAB function)

# **Unipolar to Bipolar Converter**

**Purpose** 

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

Library

**Utility Blocks** 

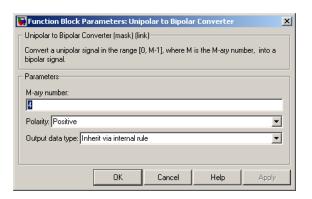
## **Description**

Unipolar to > Bipolar Converter 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. If M is odd, then the output is even. This block is only designed to work when the input value is within the set  $\{0,1,2...(M-1)\}$ , where M is the **M-ary number** parameter. If the input value is outside of this set of integers the output may not be valid.

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

Polarity Parameter Value	Output Corresponding to 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.

# **Unipolar to Bipolar Converter**

#### **Polarity**

A value of Positive causes the block to maintain the relative ordering of symbols in the alphabets. A value of Negative causes the block to reverse the relative ordering of symbols in the alphabets.

#### **Output Data Type**

The type of bipolar signal produced at the block's output.

The block supports the following output data types:

- Inherit via internal rule
- Same as input
- double
- int8
- int16
- int32

When the parameter is set to its default setting, Inherit via internal rule, the block determines the output data type based on the input data type.

- If the input signal is floating-point (either single or double), the output data type is the same as the input data type.
- If the input data type is not floating-point:
  - Based on the M-ary number parameter, an ideal signed integer output word length required to contain the range [-(M-1)M-1] is computed as follows:

ideal word length = ceil(log2(M))+1

**Note** The +1 is associated with the need for the sign bit.

# **Unipolar to Bipolar Converter**

- The block sets the output data type to be a signed integer, based on the smallest word length (in bits) that can fit best the computed ideal word length.

**Note** The selections in the **Hardware Implementation** pane pertaining to word length constraints do not affect how this block determines output data types.

## **Examples**

If the input is [0; 1; 2; 3], the **M-ary number** parameter is 4, and the **Polarity** parameter is Positive, then the output is [-3; -1; 1; 3]. Changing the **Polarity** parameter to Negative changes the output to [3; 1; -1; -3].

If the value for the **M-ary number** is  $2^7$  the block gives an output of int8.

If the value for the **M-ary number** is  $2^{7}+1$  the block gives an output of int16.

## **Pair Block**

Bipolar to Unipolar Converter

## **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.

• 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*.

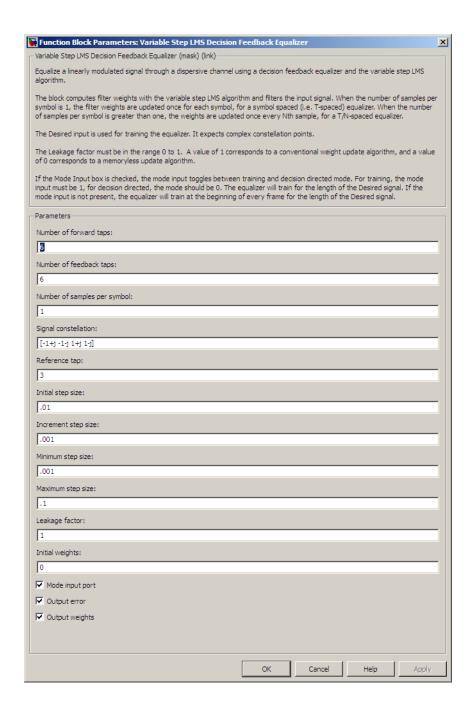
## **Equalizer Delay**

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+(Reference tap-1)/(Number of samples per symbol)

Because the channel delay is typically unknown, a common practice is to set the reference tap to the center tap of the forward filter.

## 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.

#### Initial step size

The step size that the variable-step-size LMS algorithm uses at the beginning of the simulation.

#### Increment step size

The increment by which the step size changes from iteration to iteration

## Minimum step size

The smallest value that the step size can assume.

## Maximum step size

The largest value that the step size can assume.

## Leakage factor

The leakage factor of the variable-step-size LMS algorithm, a number between 0 and 1. A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.

### Initial weights

A vector that concatenates the initial weights for the forward and feedback taps.

#### Mode input port

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

## **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*.

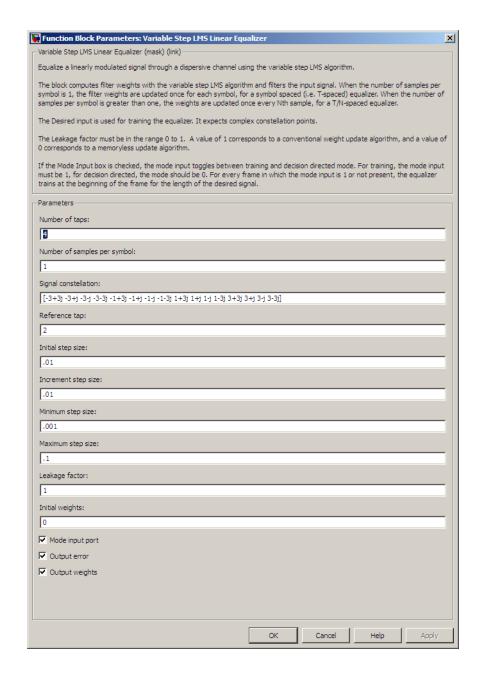
### **Equalizer Delay**

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+(Reference tap-1)/(Number of samples per symbol)

Since the channel delay is typically unknown, a common practice is to set the reference tap to the center tap.

## 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 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.

#### **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

## Viterbi Decoder

**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, boolean, 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.

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 <sup>b</sup> -1, where <i>b</i> is the <b>Number of</b> soft decision bits	0: most confident decision for logical zero
parameter.		2 <sup>b</sup> -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	
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 constr (using the polynomial description), "enough" means k\*max(constr-1). For feedback encoders, this mode is appropriate if the corresponding Convolutional Encoder block has Operation mode set to Terminate trellis by appending bits.

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.

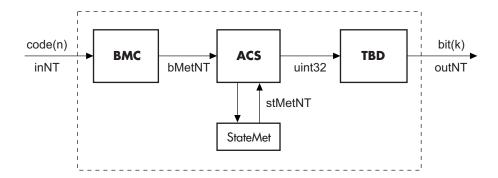
Using a reset port on this block is analogous to setting **Operation** mode in the Convolutional Encoder block to Reset on nonzero input via port.

The reset port supports double or boolean typed signals.

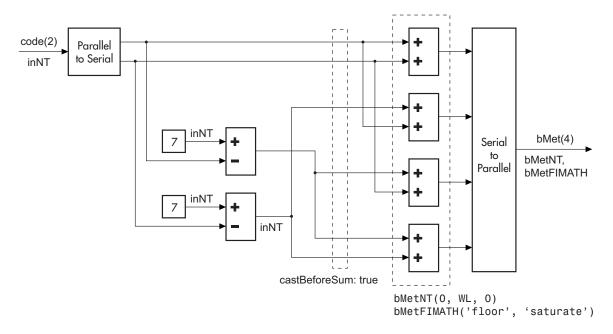
## **Fixed-Point Signal Flow Diagram**

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.

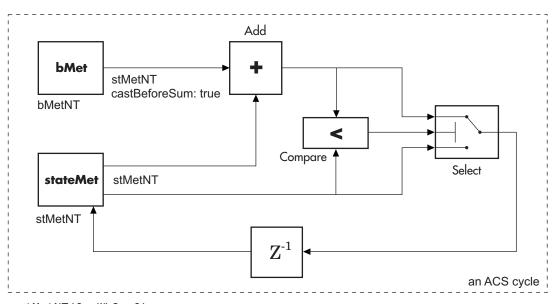
# Viterbi Decoder



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, in NT, bMetNT, stMetNT, and outNT are numerictype objects, and bMetFIMATH and stMetFIMATH, are fimath objects.

## Puncture Pattern Examples

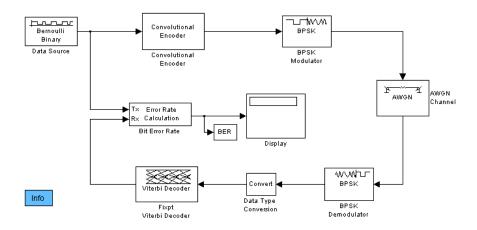
For some commonly used puncture patterns for specific rates and polynomials, see the last three references.

## Fixed-Point Viterbi Decoding Examples

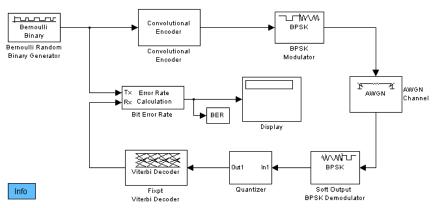
The following two example models showcase the fixed-point Viterbi decoder block used for both hard- and soft-decision convolutional decoding.

If you are reading this reference page in the MATLAB Help Browser, click 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\commblks\examples.

#### Fixed-point Hard-Decision Viterbi Decoding



#### 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\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 values of the data, you are also able to determine whether the selected container is of the appropriate size.

## Viterbi Decoder

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.

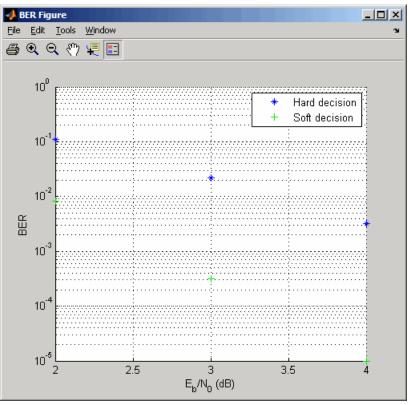
Upon simulating the model, note that the double-precision and fixed-point BER results are the same. They are the same because the fixed-point parameters for the model have been selected to avoid any loss of precision while still being most efficient.

### **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:

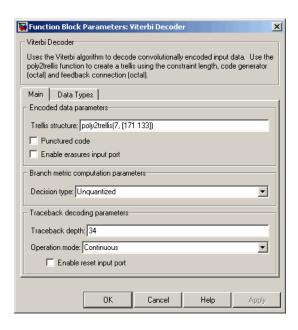
- 1 Type bertool at the MATLAB command prompt.
- 2 Go to the Monte Carlo pane.
- **3** Set the **Eb/No range** to 2:5.
- **4** Set the **Simulation model** to doc\_fixpt\_vitharddec.mdl. Make sure that the model is on path.
- **5** Set the **BER** variable name to BER.
- 6 Set the Number of errors to 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.



## 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

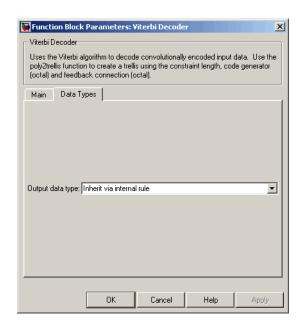
The number of trellis branches used to construct each traceback path.

#### 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.



#### Output data type

The output signal's data type can be double, single, boolean, int8, uint8, int16, uint16, int32, uint32, or set to 'Inherit via internal rule' or 'Smallest unsigned integer'.

When set to 'Smallest unsigned integer', the output data type is selected based on the settings used in the **Hardware Implementation** pane of the Configuration Parameters dialog box of the model. If ASIC/FPGA is selected in the **Hardware Implementation** pane, the output data type is ufix(1). For all other selections, it is an unsigned integer with the smallest specified wordlength corresponding to the char value (e.g., uint8).

When set to 'Inherit via internal rule' (the default setting), the block selects double-typed outputs for double inputs, single-typed outputs for single inputs, and behaves similarly to

the 'Smallest unsigned integer' option for all other typed inputs.

#### 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.

# **Walsh Code Generator**

## **Purpose**

Generate Walsh code from orthogonal set of codes

# Library

Sequence Generators sublibrary of Comm Sources

# **Description**

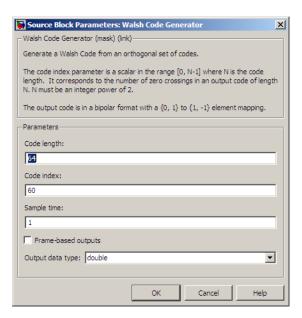
Walsh Code Generator Walsh codes are defined as a set of N codes, denoted  $W_{\rm j}$ , for j = 0, 1, ... , N - 1, which have the following properties:

- $W_i$  takes on the values +1 and -1.
- $W_i[0] = 1$  for all j.
- $W_i$  has exactly j zero crossings, for j = 0, 1, ..., N-1.
- $\bullet \quad W_j W_k^T = \begin{cases} 0 & j \neq k \\ N & j = k \end{cases}$
- Each code Wj 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, ..., 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, ..., 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.

## 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.

# **Walsh Code Generator**

#### 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

**Purpose** 

Integrate over time window of fixed length

Library

Comm Filters

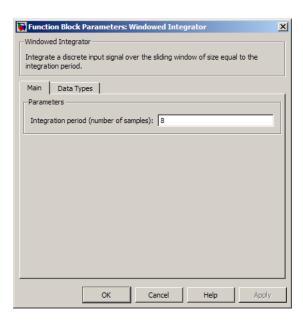
## **Description**



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),..., then the nth 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.

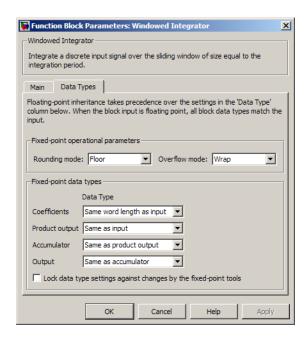
The input can be either a scalar or a frame-based matrix. If the input is frame-based, then the block processes each column independently. The output has the same sample time and matrix size as the input. For information about the data types each block port supports, see the "Supported Data Type" on page 2-775 table on this page.

## 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 block uses the **Rounding mode** when the result of a fixed-point calculation does not map exactly to a number representable by the data type and scaling storing the result. The filter coefficients do not obey this parameter; they always round to Nearest. For more information, see Rounding Modes in the Signal Processing Blockset User's Guide or "Rounding Mode: Simplest" in the Simulink Fixed Point documentation.

#### Overflow mode

Select the overflow mode for fixed-point operations. The filter coefficients do not obey this parameter; they are always saturated.

#### Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator). See "Filter Structure Diagrams" in *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.

#### Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling tool in the Fixed-Point Tool.

# Supported Data Type

Port	Supported Data Types
In	Double-precision floating point
	Single-precision floating point
	Signed Fixed-point
Out	Double-precision floating point
	Single-precision floating point
	Signed fixed-point

## **Examples**

If **Integration period** is 3 and the input signal is a ramp (1, 2, 3, 4,...), then some of the sums that form the output of this block are as follows:

- 0+0+1=1
- $\bullet$  0+1+2 = 3
- 1+2+3=6
- $\bullet$  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 documentation)

# Function Reference

# 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 documentation), commliblist

# commlib

Purpose Open main Communications Blockset library

Syntax commlib

**Description** commlib opens the latest version of the Communications Blockset

library.

See Also simulink (Simulink documentation), dsplib (Signal Processing Blockset

documentation)

# commstartup

#### **Purpose**

Default Simulink model settings for Communications Blockset software

# **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 application includes a similar dspstartup script, which assigns different model settings. For modeling communication systems, you should use commstartup alone.

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

To be more specific, the settings in commstartup cause models to:

- $\bullet~$  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
- $\bullet~$  Set the Simulink Boolean logic signals parameter to Off
- Avoid saving output or time information to the workspace
- Produce an error upon detecting an algebraic loop
- Inline parameters if you use the Model Reference feature of Simulink

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

## See Also

startup

#### **Purpose**

Generate prime numbers for use as random number seeds

## **Syntax**

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)

# **Description**

The randseed function produces random prime numbers that work well as seeds for random source blocks or noisy channel blocks in Communications Blockset software. It is recommended you use the randseed function when specifying the initial seed parameters of the following blocks: Gaussian, Rayleigh, and Rician Noise Generator.

**Note** The randseed function uses a local stream of numbers that is independent from the global stream of numbers in the MATLAB software. Use of this function does not affect the state of the global random number stream.

out = randseed generates a random prime number between 31 and  $2^{17}$ -1, using the MATLAB function rand.

out = randseed(state) generates a random prime number after setting the state of rand to the positive integer state. This syntax produces the same output for a particular value of state.

out = randseed(state,m) generates a column vector of m random
primes.

out = randseed(state, m, n) generates an m-by-n matrix of random primes.

out = randseed(state,m,n,rmin) generates an m-by-n matrix of random primes between rmin and  $2^{17}$ -1.

# randseed

out = randseed(state,m,n,rmin,rmax) generates an m-by-n matrix
of random primes between rmin and rmax.

#### **Examples**

To generate a two-element sample-based row vector of random bits using the Bernoulli Random Binary Generator block, you can set **Probability** of a zero to [0.1 0.5] and set **Initial seed** to randseed(391,1,2).

To generate three streams of random data from three different blocks in a single model, you can define out = randseed(93,3) in the MATLAB workspace and then set the three blocks' **Initial seed** parameters to out(1), out(2), and out(3), respectively.

## **See Also**

rand, primes

#### **Purpose**

Save Communications Blockset model using blocks of earlier version

## **Syntax**

flag = saveas commblks(sys,new sys,version)

# **Description**

**Note** This function is obsolete.

Simulink software provides a similar functionality through its **Save As...** option in the **File** menu of its models.

saveas\_commblks 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\_commblks(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\_commblks 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.



# Algorithm Used to Decode BCH and Reed-Solomon Codes

- "Errors-only Decoding" on page A-2
- "Errors and Erasures Decoding" on page A-3
- "References" on page A-4

# **Errors-only Decoding**

See Errors-only Decoding in the Communications Toolbox User's Guide.

# **Errors and Erasures Decoding**

The process for decoding errors and erasures is very similar to errors-only decoding. Because the decoder knows the positions of the erasures, it only needs to determine the error magnitudes in the erased positions.

The errors-and-erasures decoder calculates an *erasure locator polynomial*,  $\Gamma(z)$ , according to

$$\Gamma(z) = \prod_{k=1}^{e} \left(1 - z\alpha^{j_k}\right)$$

where e is the number of erasures in a codeword, and  $j_k$  is the kth erasure in a codeword

The Berlekamp algorithm is then used to calculate a new errors and erasures locator polynomial,  $\Psi(z)$ , which is the convolution of  $\Gamma(z)$  and  $\Lambda(z)$ . The error evaluator polynomial,  $\Omega(z)$ , is then calculated as the convolution of  $\Psi(z)$  and S(z). Finally, the error magnitudes are then calculated as

$$e_{i_m} = \frac{\Omega(\alpha^{-i_m})}{\Psi'(\alpha^{-i_m})}$$



# References

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