Communications Blockset

For Use with Simulink®

- Modeling
- Simulation
- Implementation

Reference

Version 3

The MathWorks

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Communications Blockset Reference

Revision History:

May 2001	Online only	New for Version 2.0.1 (Release 12.1)
July 2002	Online only	Revised for Version 2.5 (Release 13)
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March 2005	Online only	Revised for Version 3.1 (Release 14SP2)

Contents

1 Blocks – Categorical List

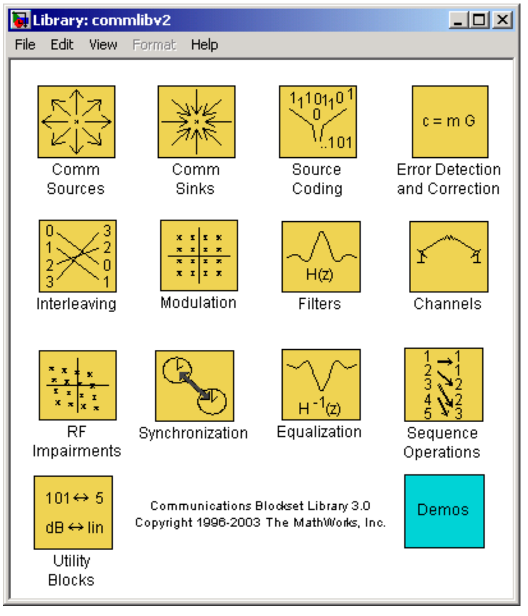
Blocks – Categorical List

Accessing the Libraries

You can access the main library of the Communications Blockset by entering

```
commlib
```

in the MATLAB® Command Window.



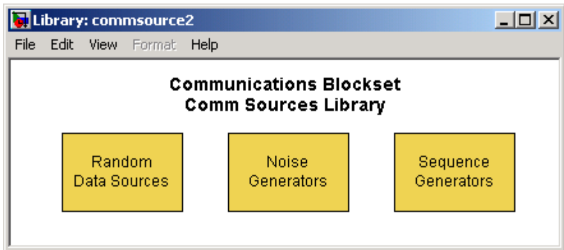
From the main library, you can access sublibraries by double-clicking their icons.

On Windows platforms, you can also use the Simulink® Library Browser to access libraries of the Communications Blockset. To open the Simulink Library Browser, enter `simulink` in the MATLAB Command Window.

Source code for the communications blocks can be found in `<MATLAB>\toolbox\commblks\sim\sfun`.

Communications Sources

Every communication system contains one or more sources. You can open the Comm Sources library by double-clicking its icon in the main Communications Blockset library.

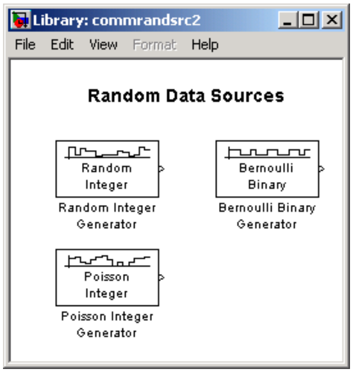


The Comms Sources library contains these sublibraries:

- Data Sources, which contains blocks that generate random data to simulate signal sources.
- Noise Generators, which contains blocks that generate random data to simulate channel noise.
- Sequence Generators, which contains blocks that generate sequences for spreading or synchronization in a communication system.

Data Sources

You can open the Data Sources sublibrary by double-clicking its icon in the Comm Sources library.

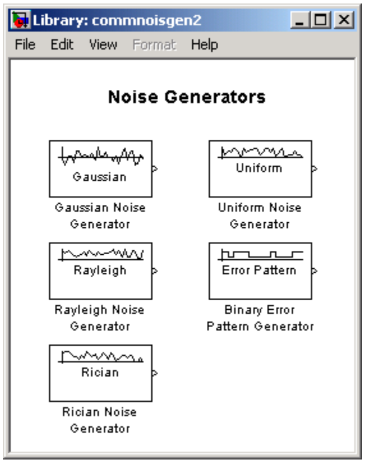


The table below lists and describes the blocks in the Data Sources sublibrary. For information about a specific block, see the reference pages that follow.

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

You can open the Noise Generators sublibrary by double-clicking its icon in the Comm Sources library.

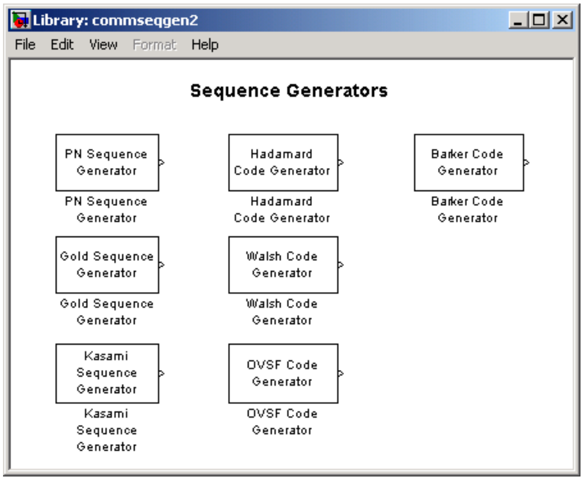


The table below lists and describes the blocks in the Noise Generators sublibrary. For information about a specific block, see the reference pages that follow.

Binary Error Pattern Generator	Generate binary vector while controlling number of 1s
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

You can open the Sequence Generators sublibrary by double-clicking its icon in Comm Sources library.

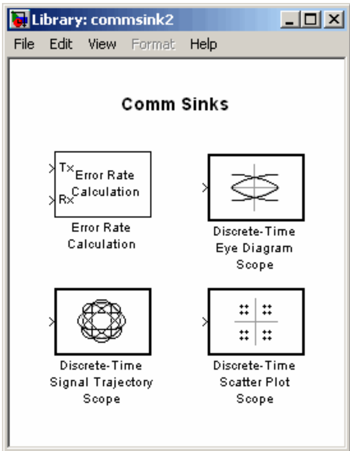


The table below lists and describes the blocks in the Sequence Generators sublibrary. For information about a specific block, see the reference pages that follow.

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

The Comm Sinks library provides sinks and display devices that facilitate analysis of communication system performance. You can open the Comm Sinks library by double-clicking its icon in the main Communications Blockset library.

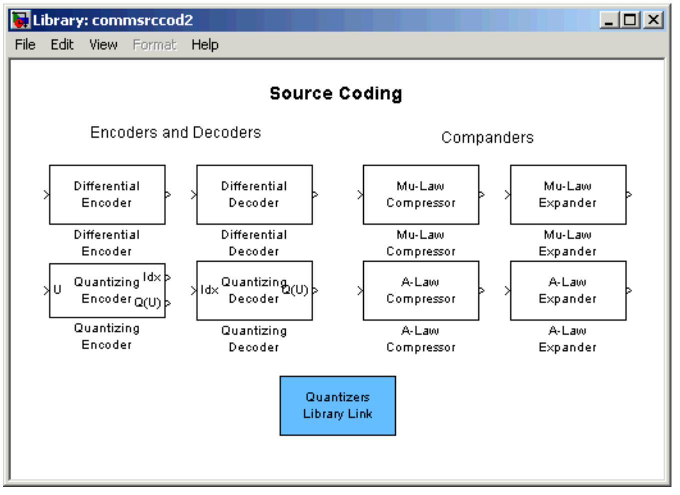


The table below lists and describes the blocks in the Comm Sinks library. For information about a specific block, see the reference pages that follow.

Discrete-Time Eye Diagram Scope	Display multiple traces of modulated signal
Discrete-Time Scatter Plot Scope	Display the 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

This blockset supports companders and scalar quantization. You can open the Source Coding library by double-clicking its icon in the main Communications Blockset library.



The table below lists and describes the blocks in the Source Coding library. For information about a specific block, see the reference pages that follow.

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

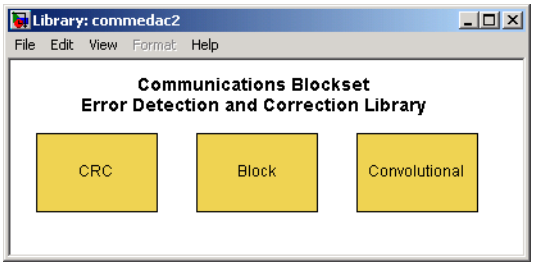
Quantizing Decoder	Decode quantization index according to codebook
Quantizing Encoder	Quantize signal using partition and codebook

Error Detection and Correction

The Error Detection and Correction library contains three sublibraries:

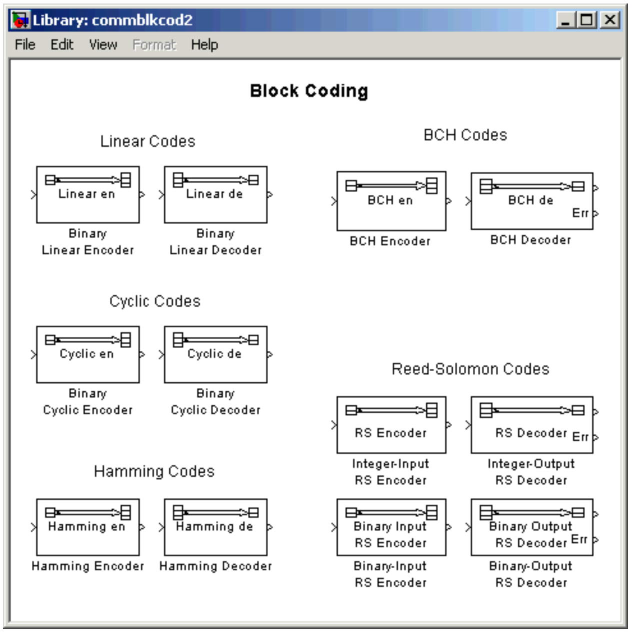
- Block, which contains blocks that implement the encoding and decoding of linear, cyclic, BCH, Hamming, and Reed-Solomon codes
- Convolutional, which contains blocks that implement convolutional encoding and decoding
- CRC, which contains blocks that append cyclic redundancy check (CRC) bits to data, and detect errors

The main Error Detection and Correction library appears below. You can open it by double-clicking its icon in the main Communications Blockset library. Each icon in the Error Detection and Correction window represents a sublibrary. In Simulink, double-clicking one of these icons opens the sublibrary.



Block Coding

You can open the Block sublibrary by double-clicking the Block icon in the main Error Detection and Correction library.



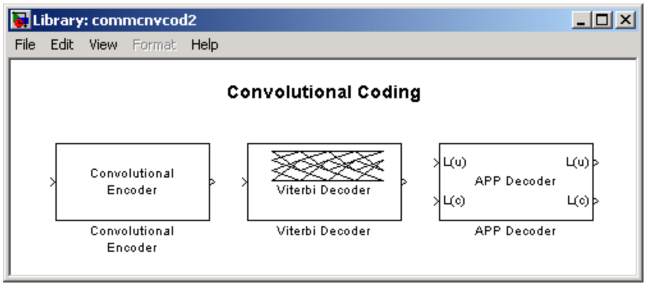
The table below lists and describes the blocks in the Block sublibrary of the Error Detection and Correction library. For information about a specific block, see the reference pages that follow.

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

Convolutional Coding

You can open the Convolutional sublibrary by double-clicking the Convolutional icon in the main Error Detection and Correction library.

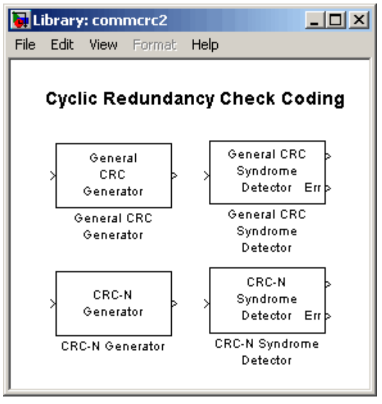


The table below lists and describes the blocks in the Convolutional sublibrary of the Error Detection and Correction library. For information about a specific block, see the reference pages that follow.

APP Decoder	Decode convolutional code using the 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

You can open the CRC sublibrary by double-clicking the CRC icon in the main Error Detection and Correction library.



The table below lists and describes the blocks in the CRC sublibrary of the Error Detection and Correction library. For information about a specific block, see the reference pages that follow.

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

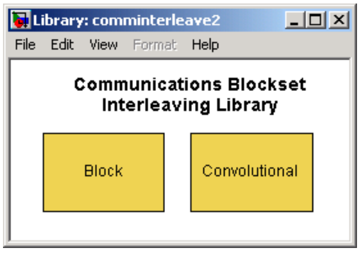
General CRC Generator	Generate CRC bits according to generator polynomial and append to input data frames
General CRC Syndrome Detector	Detect errors in input data frames according to generator polynomial

Interleaving

The Interleaving library contains two sublibraries:

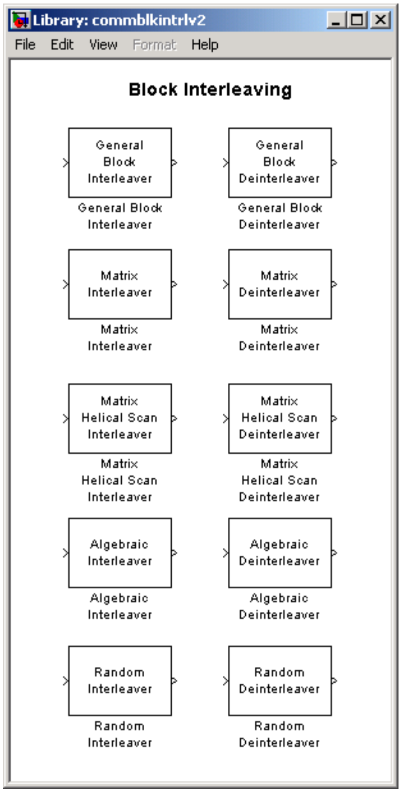
- Block
- Convolutional

The main Interleaving library appears below. You can open it by double-clicking its icon in the main Communications Blockset library. Each icon in the Interleaving window represents a sublibrary. In Simulink, double-clicking one of these icons opens the sublibrary.



Block Interleaving

You can open the Block sublibrary by double-clicking the Block icon in the main Interleaving library.



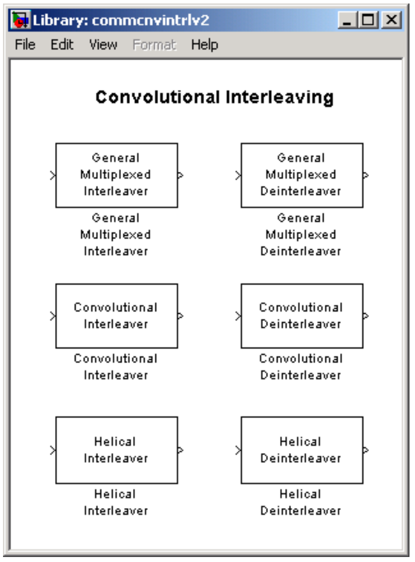
The table below lists and describes the blocks in the Block sublibrary of the Interleaving library. For information about a specific block, see the reference pages that follow.

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 a matrix by columns and emptying it by rows
Matrix Helical Scan Deinterleaver	Restore ordering of input symbols by filling a matrix along diagonals
Matrix Helical Scan Interleaver	Permute input symbols by selecting matrix elements along diagonals
Matrix Interleaver	Permute input symbols by filling a 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

You can open the Convolutional sublibrary by double-clicking the Convolutional icon in the main Interleaving library.



The table below lists and describes the blocks in the Convolutional sublibrary of the Interleaving library. For information about a specific block, see the reference pages that follow.

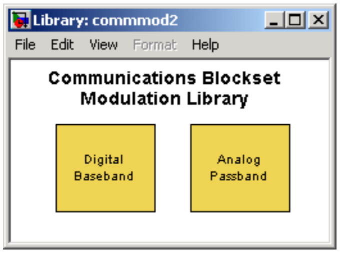
Convolutional Deinterleaver	Restore ordering of symbols that were permuted using shift registers
Convolutional Interleaver	Permute input symbols using set of shift registers
General Multiplexed Deinterleaver	Restore ordering of symbols using specified-delay shift registers
General Multiplexed Interleaver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

Modulation

The Modulation library contains these sublibraries, each of which addresses a category of modulation:

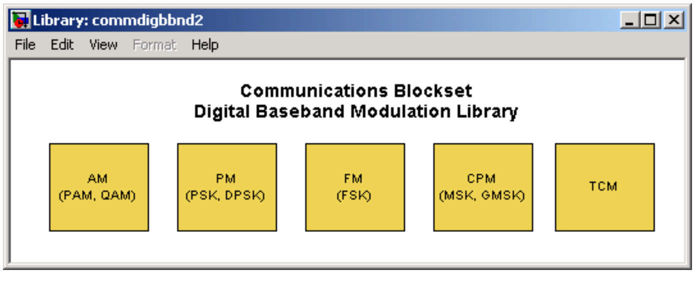
- Digital Baseband Modulation
- Analog Passband Modulation

The main Modulation library appears below. You can open it by double-clicking its icon in the main Communications Blockset library. Each icon in the Modulation window represents a sublibrary. In Simulink, double-clicking one of these icons opens the sublibrary.



Digital Baseband Modulation

You can open the Digital Baseband sublibrary of Modulation by double-clicking the Digital Baseband icon in the main Modulation library.

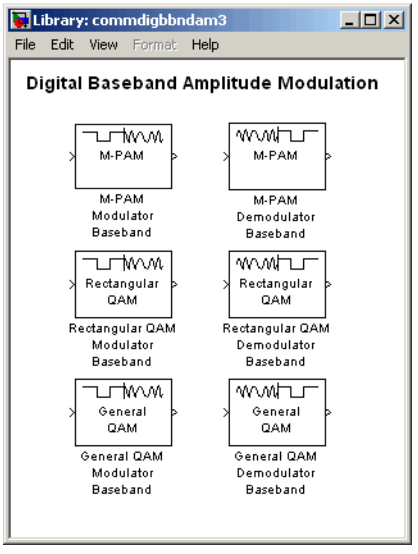


Digital Baseband is further divided into sublibraries according to specific modulation techniques:

- Amplitude modulation (PAM, QAM)
- Phase modulation (PSK, DPSK)
- Frequency modulation (FSK)
- Continuous phase modulation (MSK, GMSK)
- Trellis-coded modulation (TCM)

The figures and tables below show and list the blocks in the method-specific sublibraries. For information about a specific block, see the reference pages that follow.

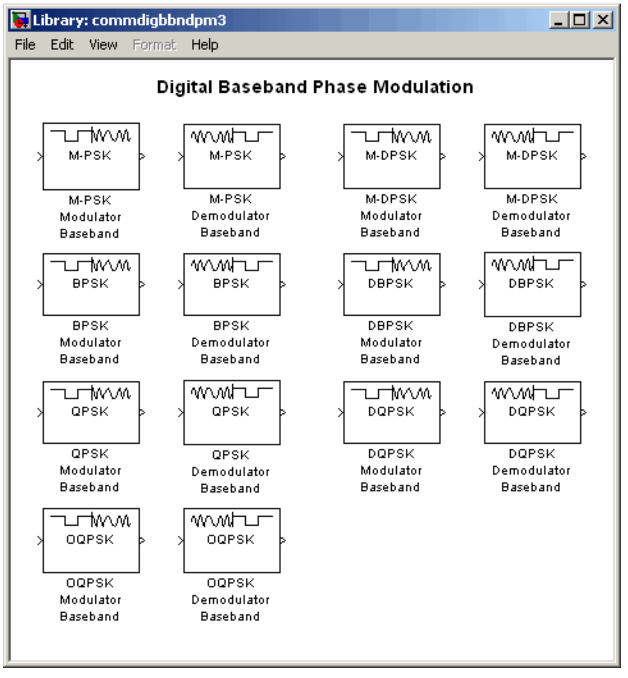
AM Sublibrary



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

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

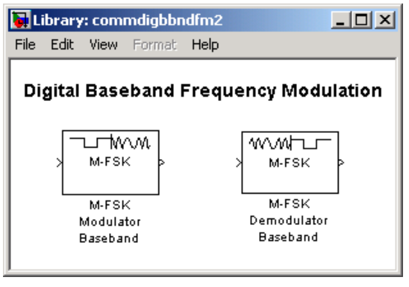
PM Sublibrary



BPSK Demodulator Baseband	Demodulate BPSK-modulated data
BPSK Modulator Baseband	Modulate using binary phase shift keying method
DBPSK Demodulator Baseband	Demodulate DBPSK-modulated data

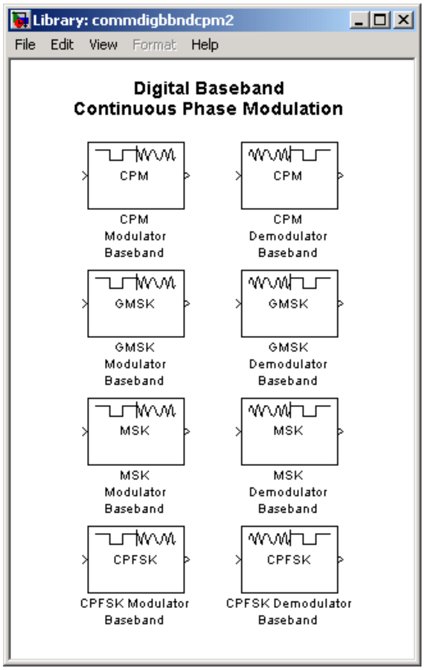
DBPSK Modulator Baseband	Modulate using differential binary phase shift keying method
DQPSK Demodulator Baseband	Demodulate DQPSK-modulated data
DQPSK Modulator Baseband	Modulate using differential quaternary phase shift keying method
M-DPSK Demodulator Baseband	Demodulate DPSK-modulated data
M-DPSK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

FM Sublibrary



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

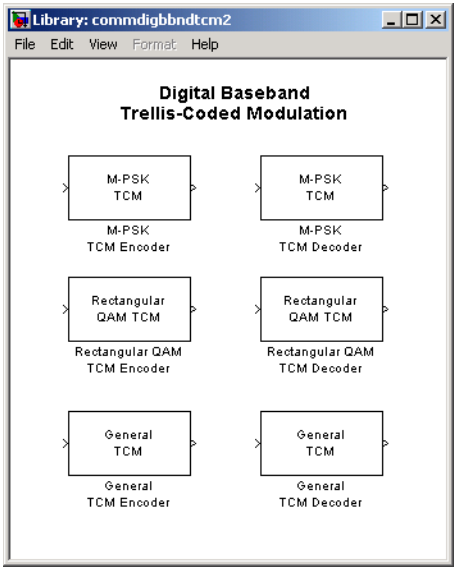
CPM Sublibrary



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

TCM Sublibrary

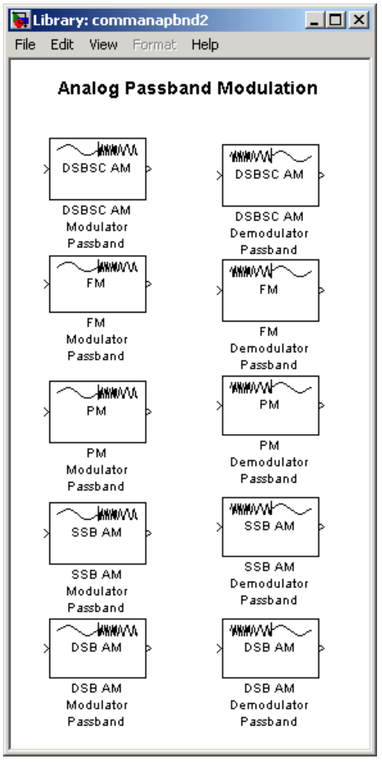


General TCM Decoder	Decode trellis-coded modulation data, mapped using arbitrary constellation
General TCM Encoder	Convolutionally encode binary data and map using arbitrary constellation
M-PSK TCM Decoder	Decode trellis-coded modulation data, modulated using PSK method

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

You can open the Analog Passband sublibrary of Modulation by double-clicking the Analog Passband icon in the main Modulation library.

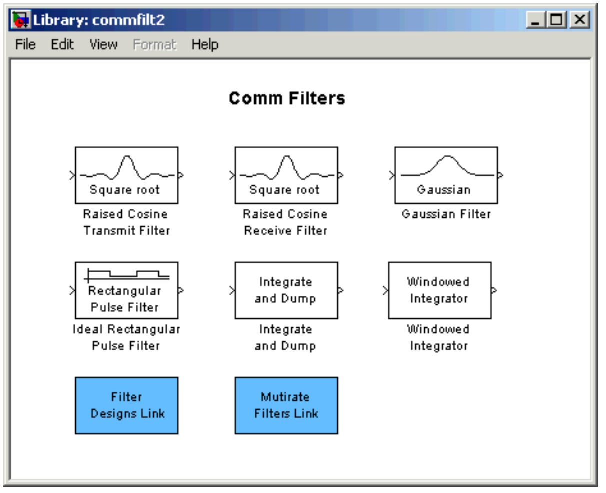


The table below lists and describes the blocks in the Analog Passband sublibrary of the Modulation library. For information about a specific block, see the reference pages that follow.

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

You can open the Comm Filters library by double-clicking its icon in the main Communications Blockset library.



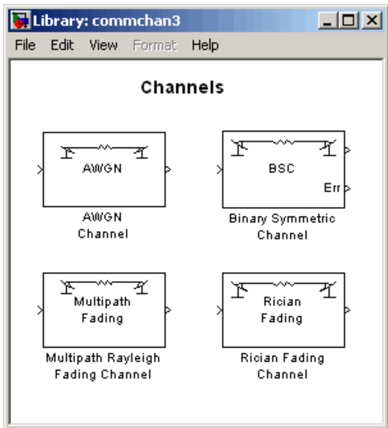
The table below lists and describes the blocks in the Comm Filters library. For information about a specific block, see the reference pages that follow.

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

The Channels library provides blocks for modeling channel impairments. You can open the Channels library by double-clicking its icon in the main Communications Blockset library.

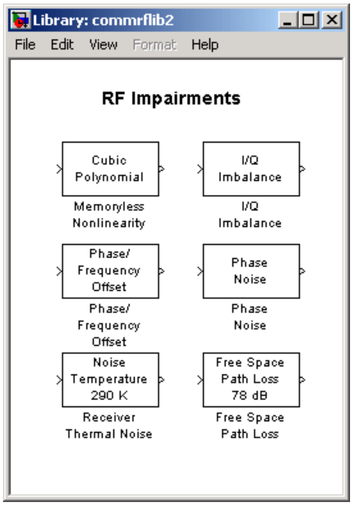


The table below lists and describes the blocks in the Channels library. For information about a specific block, see the reference pages that follow.

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
Rician Fading Channel	Simulate Rician fading propagation channel

RF Impairments

The RF Impairments library provides blocks that simulate radio frequency (RF) impairments at the receiver. You can open the RF Impairments library by double-clicking its icon in the main Communications Blockset library.



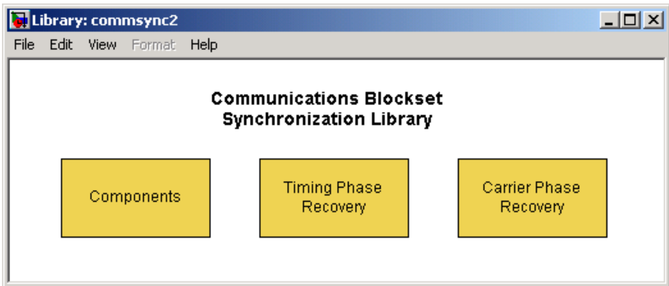
The table below lists and describes the blocks in the RF Impairments library. For information about a specific block, see the reference pages that follow.

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

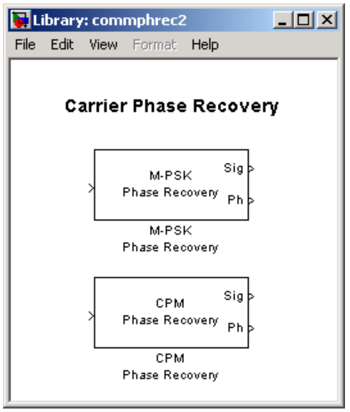
The Synchronization library provides blocks that help you perform synchronization at a receiver. You can open the Synchronization library by double-clicking its icon in the main Communications Blockset library.



The Synchronization library contains these sublibraries:

- Carrier Phase Recovery, which contains algorithms for recovering the carrier phase of a received signal
- Timing Phase Recovery, which contains algorithms for recovering the symbol timing phase of a received signal
- Synchronization Components, which contains blocks that you can use to build larger systems for synchronization

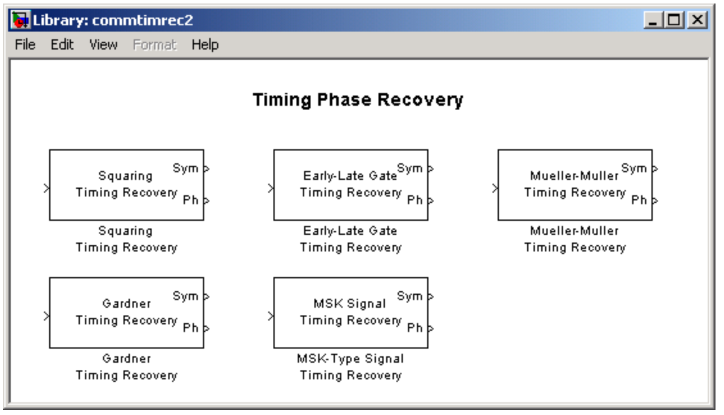
Carrier Phase Recovery



The table below lists and describes the blocks in the Carrier Phase Recovery library. For information about a specific block, see the reference pages that follow.

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

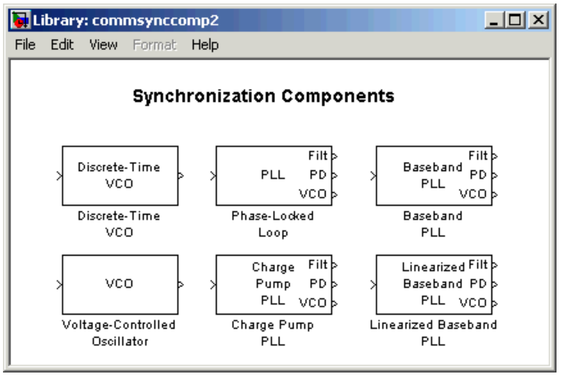
Timing Phase Recovery



The table below lists and describes the blocks in the Timing Phase Recovery library. For information about a specific block, see the reference pages that follow.

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

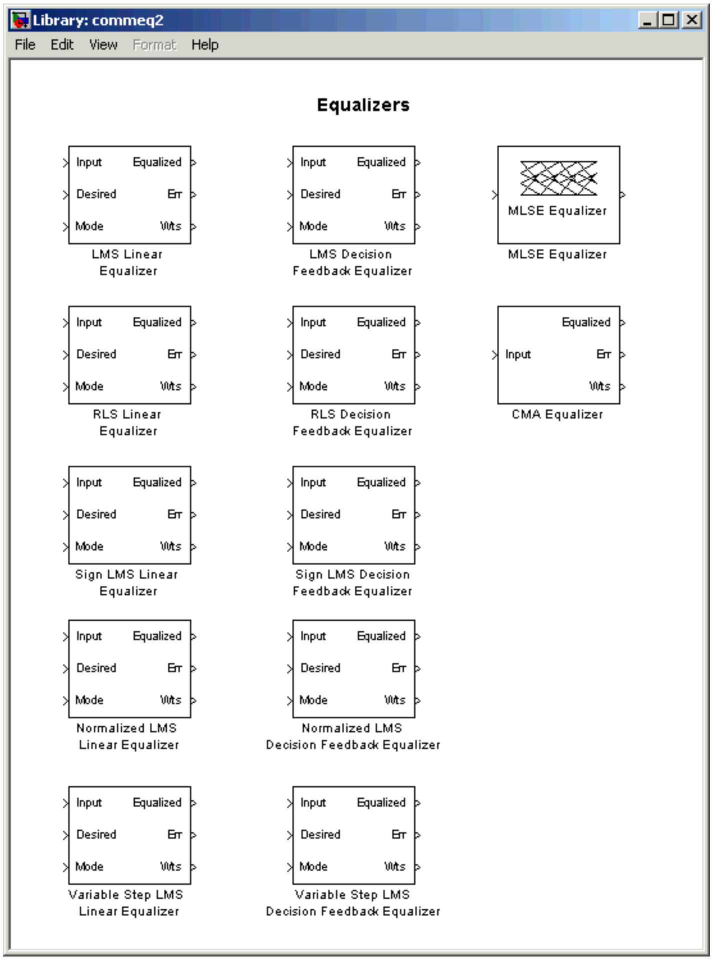


The table below lists and describes the blocks in the Synchronization Components library. For information about a specific block, see the reference pages that follow.

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 a baseband phase-locked loop
Phase-Locked Loop	Implement phase-locked loop to recover phase of input signal

Equalizers

You can open the Equalizers library by double-clicking its icon in the main Communications Blockset library.

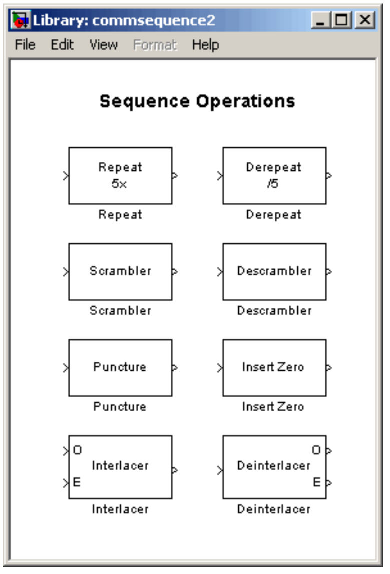


The table below lists and describes the blocks in the Equalizers library. For information about a specific block, see the reference pages that follow.

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

You can open the Sequence Operations library by double-clicking its icon in the main Communications Blockset library.



The table below lists and describes the Communications Blockset blocks in the Sequence Operations library. For information about a specific block, see the reference pages that follow.

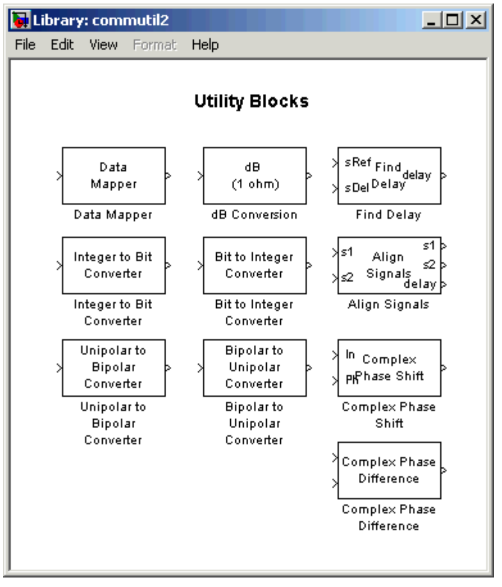
Deinterlacer	Distribute elements of input vector alternately between two output vectors
Derepeat	Reduce sampling rate by averaging consecutive samples
Descrambler	Descramble input signal
Insert Zero	Distribute input elements in output vector

Interlacer	Alternately select elements from two input vectors to generate output vector
Puncture	Output elements which correspond to 1s in binary Puncture vector
Scrambler	Scramble the input signal

The Repeat block, from the Signal Processing Blockset, is also included in this library for convenience.

Utility Blocks

You can open the Utility Blocks library by double-clicking its icon in the main Communications Blockset library.



The table below lists and describes the Communications Blockset blocks in the Utility Blocks library. For information about a specific block, see the reference pages that follow.

Align Signals	Align two signals by finding delay between them
Bipolar to Unipolar Converter	Map bipolar signal into unipolar signal in range [0, M-1]
Bit to Integer Converter	Map vector of bits to corresponding vector of integers
Complex Phase Difference	Output phase difference between two complex input signals

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

The dB Conversion block, from the Signal Processing Blockset, is also included in this library for convenience.

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

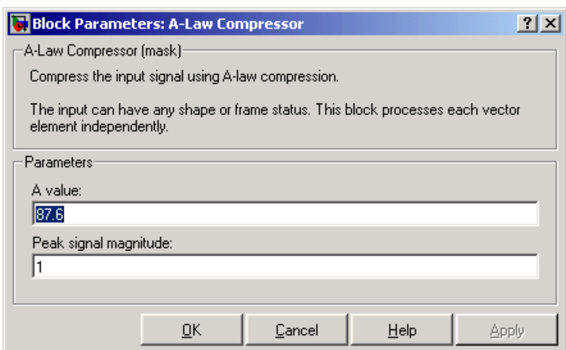
$$y = \begin{cases} \dfrac{A|x|}{1+\log A}\operatorname{sgn}(x) & \text{for } 0 \le |x| \le \dfrac{V}{A} \\ \dfrac{V\left(1+\log(A|x|/V)\right)}{1+\log A}\operatorname{sgn}(x) & \text{for } \dfrac{V}{A} < |x| \le V \end{cases}$$

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).

The most commonly used A value is 87.6.

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

Dialog Box



A value

The A-law parameter of the compressor.

Peak signal magnitude
The peak value of the input signal. This is also the peak value of the output signal.

Pair Block

A-Law Expander

See Also

Mu-Law Compressor

References

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

A-Law Expander

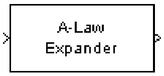
Purpose

Implement A-law expander for source coding

Library

Source Coding

Description

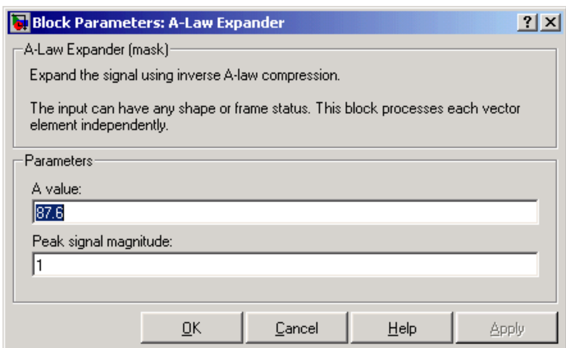


The A-Law Expander block recovers data that the A-Law Compressor block compressed. The formula for the A-law expander, shown below, is the inverse of the compressor function.

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

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.

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

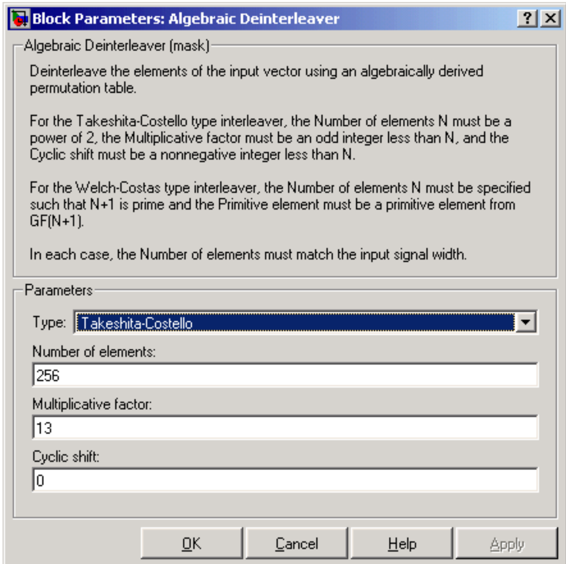
The Algebraic Deinterleaver block restores the original ordering of a sequence that was interleaved using theAlgebraic Interleaver block. In typical usage, the parameters in the two blocks have the same values.

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 theAlgebraic 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`.

Primitive element

An element of order N in the finite field GF(N+1). This field appears only if **Type** is set to `Welch-Costas`.

Pair Block

Algebraic Interleaver

See Also

General Block Deinterleaver

References

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

[2] Takeshita, O. Y. and D. J. Costello, Jr. "New Classes Of Algebraic Interleavers for Turbo-Codes." *Proc. 1998 IEEE International Symposium on Information Theory*, Boston, Aug. 16-21, 1998. 419.

Purpose

Reorder input symbols using algebraically derived permutation table

Library

Block sublibrary of Interleaving

Description

The Algebraic Interleaver block rearranges the elements of its input vector using a permutation that is algebraically derived. The **Number of elements** parameter, N, indicates how many numbers are in the input vector. If the input is frame-based, then it must be a column vector.

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

The **Type** parameter indicates the algebraic method that the block uses to generate the appropriate permutation table. Choices are `Takeshita-Costello` and `Welch-Costas`. Each of these methods has parameters and restrictions that are specific to it:

- If **Type** is set to `Welch-Costas`, then N+1 must be prime. The **Primitive element** parameter is an integer, A, between 1 and N that represents a primitive element of the finite field GF(N+1). This means that every nonzero element of GF(N+1) can be expressed as A raised to some integer power.

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

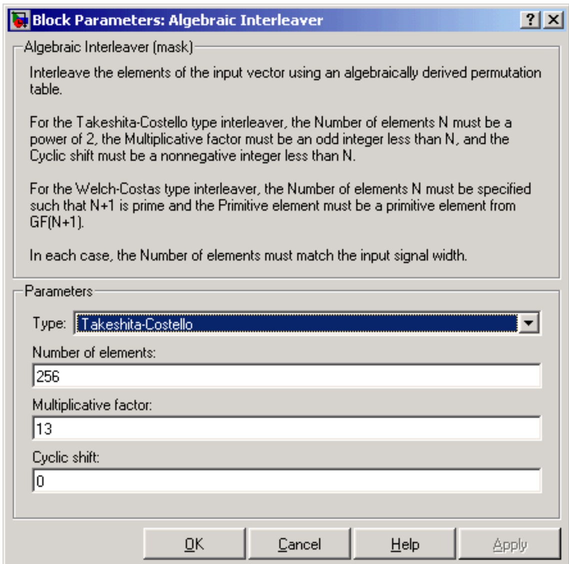
- If **Type** is set to `Takeshita-Costello`, then N must be 2^m for some integer m. The **Multiplicative factor** parameter, h, must be an odd integer less than N. The **Cyclic shift** parameter, k, must be a nonnegative integer less than N.

  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 order, one plus the element's successor. The interleaver's actual permutation table is the result of shifting the elements of the permutation vector left by the **Cyclic shift** parameter. (The block performs all computations on numbers and indices modulo N.)

Dialog Box



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

Number of elements
: The number of elements, N, in the input vector.

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

Cyclic shift
: The amount by which the block shifts indices when creating the permutation table. This field appears only if **Type** is set to `Takeshita-Costello`.

Primitive element

An element of order N in the finite field GF(N+1). This field appears only if **Type** is set to `Welch-Costas`.

Pair Block

Algebraic Deinterleaver

See Also

General Block Interleaver

References

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

[2] Takeshita, O. Y. and D. J. Costello, Jr. "New Classes Of Algebraic Interleavers for Turbo-Codes." *Proc. 1998 IEEE International Symposium on Information Theory*, Boston, Aug. 16-21, 1998. 419.

Align Signals

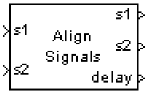
Purpose

Align two signals by finding delay between them

Library

Utility Blocks

Description



Align Signals

The Align Signals block aligns a signal with a delayed, and possibly distorted, version of itself. The block is particularly useful when you want to compare a transmitted and received signal to find the bit error rate, but do not know the delay in the received signal.

The input port labeled `s1` receives the original signal, while the input port labeled `s2` receives the delayed version of the signal. The two input signals must have the same sample times. The block calculates the delay between the two signal, and then

- Delays the first signal, `s1`, by the calculated value, and outputs it through the port labeled `s1`.
- Outputs the second signal `s2` without change through the port labeled `s2`.
- Outputs the delay value through the port labeled `delay`.

See "Computing Delays" in the Communications Blockset online documentation for more information about signal delays.

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

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

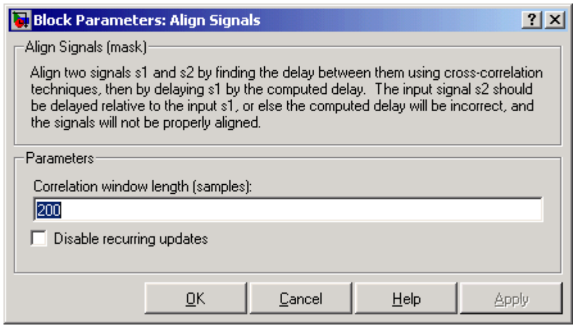
- Set the **Correlation window length** parameter sufficiently large so that the computed delay eventually stabilizes at a constant value. If the computed delay is not constant, you should increase **Correlation window length**. If the increased value of **Correlation window length** exceeds the duration 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 Using the Communications Blockset for an example that uses the Align Signals block in conjunction with the Error Rate Calculation block.

See "Setting the Correlation Window Length" on page 2-181, 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 the 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, then this block's L(c) vectors have length $Q*n$ for some positive integer Q. Similarly, if the decoded data uses an alphabet of 2^k possible output symbols, then this block's L(u) vectors have length $Q*k$. The integer Q is the number of frames that the block processes in each step.

The inputs can be either:

- Sample-based vectors having the same dimension and orientation, with Q = 1
- Frame-based column vectors with any positive integer for Q

If you only need the input L(c) and output L(u), then you can attach a Simulink Ground block to the input L(u) and a Simulink Terminator block to the output L(c).

Specifying the Encoder

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

- If you have a variable in the MATLAB workspace that contains the trellis structure, then 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 in the next bulleted item.
- 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)
  ```

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, while the `Terminated` option indicates that the encoder forces the trellis to end each frame in the all-zeros state. If you use theConvolutional Encoder block with the **Reset** parameter set to `On each frame`, then use the `Truncated` option in this block.

Specifying Details of the Algorithm

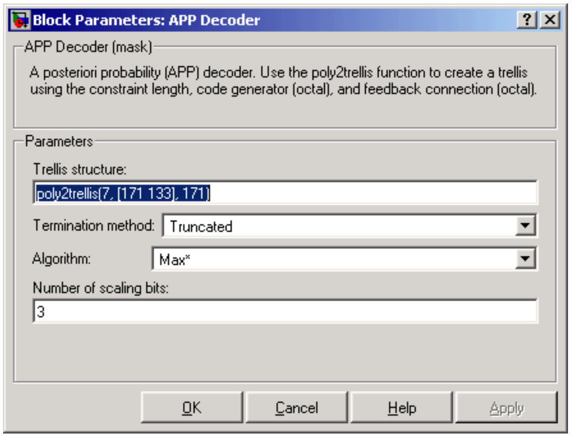
You can control part of the decoding algorithm using the **Algorithm** parameter. The `True APP` option implements a posteriori probability. To gain speed, both the `Max*` and `Max` options approximate expressions like

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

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.

The `Max*` option enables the **Scaling bits** parameter in the dialog. This parameter is the number of bits by which the block scales the data it processes internally. You can use this parameter to avoid losing precision during the computations. It is especially appropriate if your implementation uses fixed-point components. For more information about the `Max*` option, see the article by Viterbi among the references listed below.

Dialog Box



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

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

Algorithm
Either `True APP`, `Max*`, or `Max`.

Number of scaling bits
An integer between 0 and 8 that indicates by how many bits the decoder scales data in order to avoid losing precision. This field is active only when **Algorithm** is set to `Max*`.

See Also

Viterbi Decoder, Convolutional Encoder; `poly2trellis` (Communications Toolbox)

References

[1] Benedetto, Sergio and Guido Montorsi. "Performance of Continuous and Blockwise Decoded Turbo Codes." *IEEE Communications Letters*, vol. 1, May 1997. 77-79.

[2] 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. [This electronic journal is available at http://tmo.jpl.nasa.gov/tmo/progress_report/index.html.]

[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.

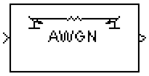
Purpose

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's Random Source block to generate the noise. 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.

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 power of the input symbols
  - **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 power of the input symbols
  - **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 power of the input samples
- `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.

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:

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

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

where

- E_s = Signal energy (Joules)
- E_b = Bit energy (Joules)
- N_0 = Noise power spectral density (Watts/Hz)
- T_{sym} is the **Symbol period** parameter of the block in `Es/No` mode
- k is the number of information bits per input symbol
- T_{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:

$$E_s/N_0 = 2\ \mathrm{SNR}\ (T_{sym}/T_{samp})$$

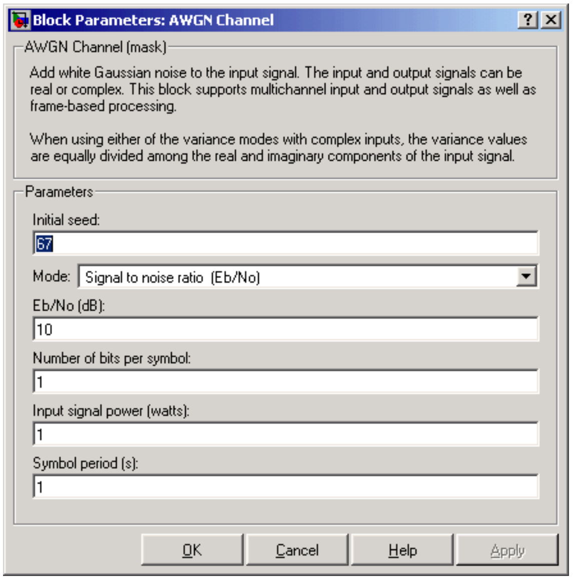
Note that the equation for the real case differs from the corresponding equation for the complex case by a factor of 2. This is so because the block uses a noise power spectral density of $N_0/2$ Watts/Hz for real input signals, versus N_0 Watts/Hz for complex signals.

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

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

If you use the Real-Time Workshop rapid simulation (RSim) target to build an RSim executable, then you can tune selected parameters without recompiling the model. This is useful for Monte Carlo simulations in which you run the simulation multiple times (perhaps on multiple computers) with different amounts of noise. The table below indicates, for different modes of the block, which parameters are tunable.

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**



Dialog Box

Initial seed
The seed for the Gaussian noise generator.

Mode
The mode by which you specify the noise variance: `Signal to noise ratio (Eb/No)`, `Signal to noise ratio (Es/No)`, `Signal to noise ratio (SNR)`, `Variance from mask`, or `Variance from port`.

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

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

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

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

Input signal power (watts)
The root 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" (`EbNo` mode)
- "Phase Noise Effects in 256-QAM - Demo" (`EsNo` mode)
- "Building a Frequency-Shift Keying Model" (`EsNo` mode)
- "Example: Using Raised Cosine Filters" (`SNR` mode)
- "Discrete Multitone Signaling Demo" (`Variance from mask` mode)

See Also

Random Source (Signal Processing Blockset)

Reference

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

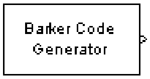
Purpose

Generate Barker Code

Library

Sequence Generators sublibrary of Comm Sources

Description



Barker codes, which are subsets of PN sequences, are commonly used for frame synchronization in digital communication systems. Barker codes have length at most 13 and have low correlation sidelobes. A correlation sidelobe is the correlation of a codeword with a time-shifted version of itself. The correlation sidelobe, C_k, for a k-symbol shift of an N-bit code sequence, $\{X_j\}$, is given by

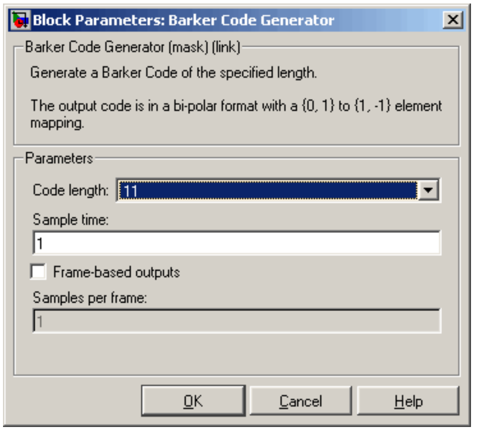
$$C_k = \sum_{j=1}^{N-k} X_j X_{j+k}$$

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

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

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

Dialog Box



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

Code length
: The length of the Barker code.

Sample time
: Period of each element of the output signal.

Frame-based outputs
: Determines whether the output is frame-based or sample-based.

Samples per frame
: The number of samples in a frame-based output signal. This field is active only if you select the **Frame-based outputs** check box.

See Also

PN Sequence Generator

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 thePhase-Locked Loop block, this block uses a baseband method and does not depend on a carrier frequency.

This PLL has these three components:

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

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

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

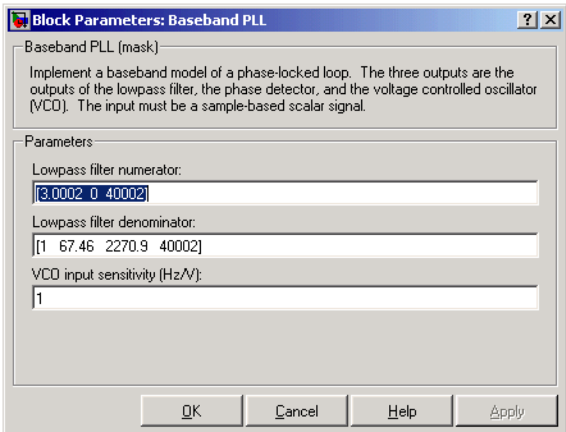
- A voltage-controlled oscillator (VCO). You specify the sensitivity of the VCO signal to its input using the **VCO input sensitivity** parameter. This parameter, measured in Hertz per volt, is a scale factor that determines how much the VCO shifts from its quiescent frequency.

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

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

This model is nonlinear; for a linearized version, use theLinearized 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 Using the Communications Blockset.

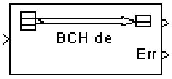
Purpose

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 correspondingBCH Encoder block.

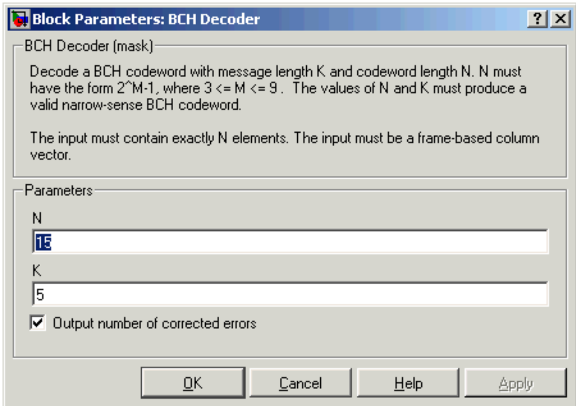
The input is the binary codeword vector and the first output is the corresponding binary message vector. If the BCH code has message length K and codeword length N, then the input has length N and the first output has length K. If the input is frame-based, then it must be a column vector.

N must have the form 2^M-1, where M is an integer greater than or equal to 3. For a given codeword length N, only specific message lengths K are valid for a BCH code. No known analytic formula describes the relationship among the codeword length, message length, and error-correction capability. For a list of some valid values of K corresponding to values of N up to 511, see the `bchenc` reference page in the Communications Toolbox documentation.

To have the block output error information, select **Output number of corrected errors**. This causes a second output port to appear. The second output is the number of errors detected during decoding of the codeword. A negative integer indicates that the block detected more errors than it could correct using the coding scheme.

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

Dialog Box



N

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

K

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

Output number of corrected errors

Checking this box causes the block to have an additional output port, which indicates the number of errors the block detected in the input codeword.

Pair Block

BCH Encoder

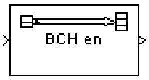
Purpose

Create BCH code from binary vector data

Library

Block sublibrary of Channel Coding

Description

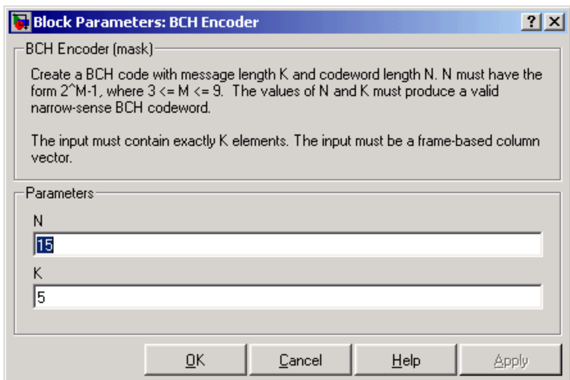


The BCH Encoder block creates a BCH code with message length K and codeword length N. You specify both N and K directly in the dialog box.

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

N must have the form 2^M-1, where M is an integer greater than or equal to 3. For a given codeword length N, only specific message lengths K are valid for a BCH code. No known analytic formula describes the relationship among the codeword length, message length, and error-correction capability. For a list of some valid values of K corresponding to values of N up to 511, see the `bchenc` reference page in the Communications Toolbox documentation.

Dialog Box



N

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

K

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

Pair Block

BCH Decoder

See Also

`bchenc` (Communications Toolbox)

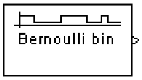
Purpose

Generate Bernoulli-distributed random binary numbers

Library

Data Sources sublibrary of Comm Sources

Description



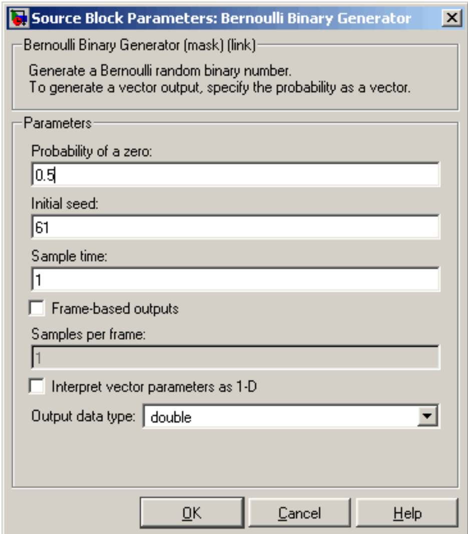
The Bernoulli Binary Generator block generates random binary numbers using a Bernoulli distribution. The Bernoulli distribution with parameter p produces zero with probability p and one with probability 1-p. The Bernoulli distribution has mean value 1-p and variance p(1-p). The **Probability of a zero** 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 Using the Communications Blockset 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.

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

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

Purpose

Decode systematic cyclic code to recover binary vector data

Library

Block sublibrary of Channel Coding

Description

The Binary Cyclic Decoder block recovers a message vector from a codeword vector of a binary systematic cyclic code. For proper decoding, the parameter values in this block should match those in the correspondingBinary Cyclic Encoder block.

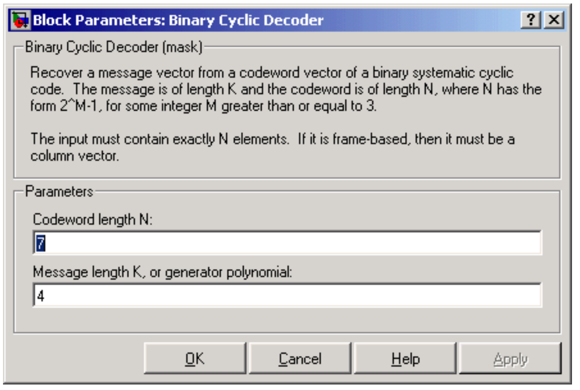
If the cyclic code has message length K and codeword length N, then N must have the form 2^M-1 for some integer M greater than or equal to 3.

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 `cyclpoly` function in the Communications Toolbox.

Dialog Box



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

Message length K, or generator polynomial
Either the message length, which is also the output vector length; or a binary vector that represents the generator polynomial for the code.

Pair Block

Binary Cyclic Encoder

See Also

`cyclpoly` (Communications Toolbox)

Binary Cyclic Encoder

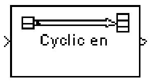
Purpose

Create systematic cyclic code from binary vector data

Library

Block sublibrary of Channel Coding

Description



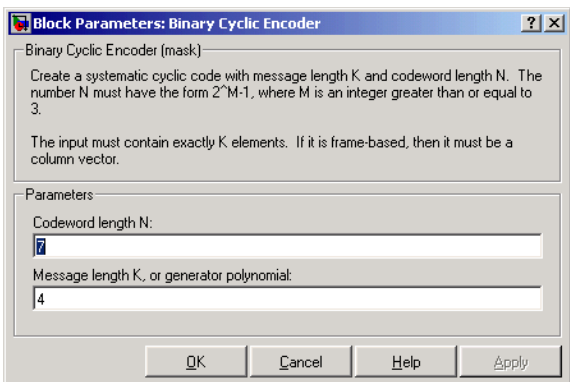
The Binary Cyclic Encoder block creates a systematic cyclic code with message length K and codeword length N. The number N must have the form 2^M-1, where M is an integer greater than or equal to 3.

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 `cyclpoly` function in the Communications Toolbox.

Dialog Box



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

Message length K, or generator polynomial
Either the message length, which is also the input vector length; or a binary vector that represents the generator polynomial for the code.

Pair Block

Binary Cyclic Decoder

See Also

`cyclpoly` (Communications Toolbox)

Binary Error Pattern Generator

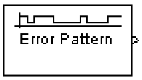
Purpose

Generate binary vector while controlling number of 1s

Library

Noise Generators sublibrary of Comm Sources

Description



The Binary Error Pattern Generator block outputs a random binary vector whose length is the **Block length** parameter. The **Probabilities** parameter helps determine how many 1s appear in each output vector. Once the number of 1s is determined, their placement is determined according to a uniform distribution.

If p_1, p_2,...p_m are the entries in the **Probabilities** parameter, then p_1 is the probability that the output vector will have a single 1, p_2 is the probability that the output vector will have exactly two 1s, and so on. Note that **Probabilities** must have sum less than or equal to one, and length less than or equal to **Block length**. Also, the probability of a zero vector is one minus the sum of **Probabilities**.

This block is useful in testing error-control coding algorithms.

Initial Seed

The scalar **Initial seed** parameter initializes the random number generator that the block uses to generate randiom errors. 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 this block using the Communications Blockset'srandseed function. At the MATLAB prompt, enter

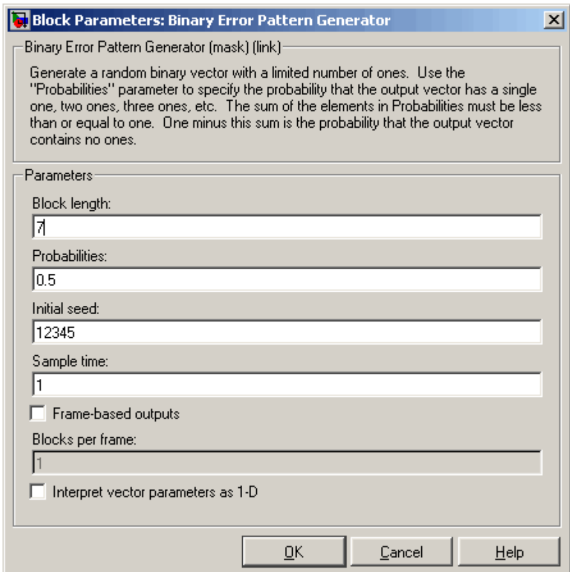
```
randseed
```

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 or column vector, a sample-based column vector, or a sample-based one-dimensional array. These attributes are controlled by the **Frame-based outputs**, **Blocks per frame**, and **Interpret vector parameters as 1-D** parameters. A frame-based output is a column vector whose size is the product of **Block length** and **Blocks per frame**. A sample-based output is a vector of length **Block length**.

Dialog Box



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

Block length

The length of each error pattern.

Probabilities
A vector whose kth entry indicates the probability that the error pattern has exactly k 1s.

Initial seed
The initial seed value for the random number generator. This must be 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.

Blocks per frame
The number of error patterns 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.

See Also

Bernoulli Binary Generator; `randerr` (Communications Toolbox)

Binary-Input RS Encoder

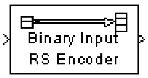
Purpose

Create Reed-Solomon code from binary vector data

Library

Block sublibrary of Channel Coding

Description



The Binary-Input RS Encoder block creates a Reed-Solomon code with message length K and codeword length N. You specify both N and K directly in the dialog box. The symbols for the code are binary sequences of length M, corresponding to elements of the Galois field GF(2^M), where the first bit in each sequence is the most significant bit. Restrictions on M and N are given in "Restrictions on the M and the Codeword Length N" on page 2-44 below. The difference N-K must be an even integer.

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

The default value of M is the smallest integer that is greater than or equal to log2(N+1), that is, `ceil(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 2-44 below. If N is less than 2^M-1, the block uses a shortened Reed-Solomon code.

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

An (N,K) Reed-Solomon code can correct up to `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 = ceil(log2(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

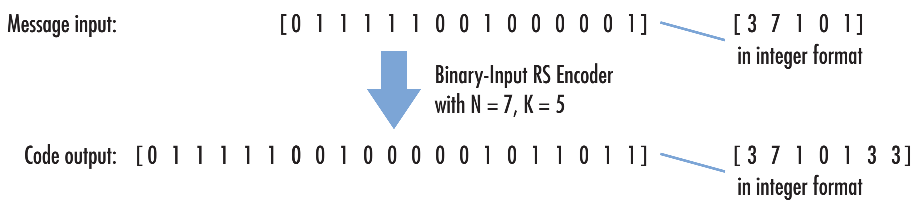
$$g(x) = (x+A^b)(x+A^{b+1})(x+A^{b+2})...(x+A^{b+N-K-1})$$

where A is the primitive element of the Galois field over which the input message is defined, and *b* is a non-negative integer.

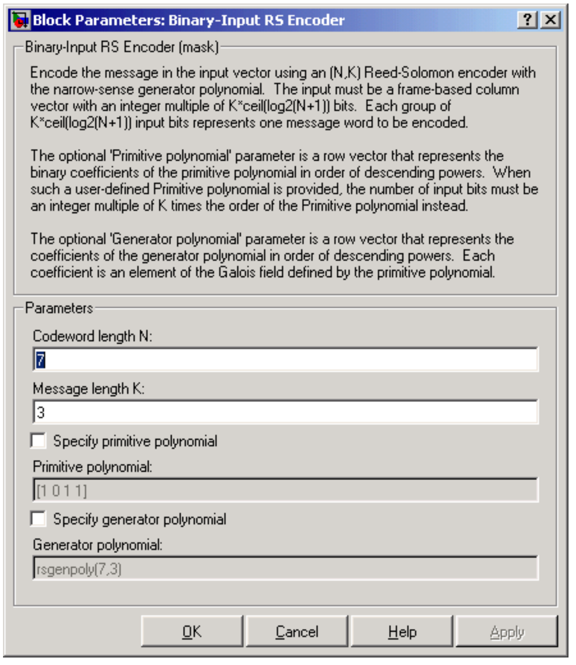
If you do not select **Specify generator polynomial**, the block uses the default generator polynomial, corresponding to *b*=1, for Reed-Solomon encoding. You can display the default generator polynomial by entering `rsgenpoly(N1,K1)`, where `N1=2^M-1` and `K1=K+(N1-N)`, at the MATLAB prompt, if you 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)`.

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 M*N.

Message length K
: The message length. The input has vector length M*K.

Specify primitive polynomial
: When you select this box, you can specify the primitive polynomial as a binary row vector.

Primitive polynomial
: Binary row vector representing the primitive polynomial in descending order of powers.

Specify generator polynomial
: When you select this box, you can specify the generator polynomial as an integer row vector.

Generator polynomial

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

Pair Block

Binary-Output RS Decoder

See Also

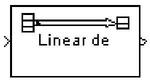
Integer-Input RS Encoder

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.

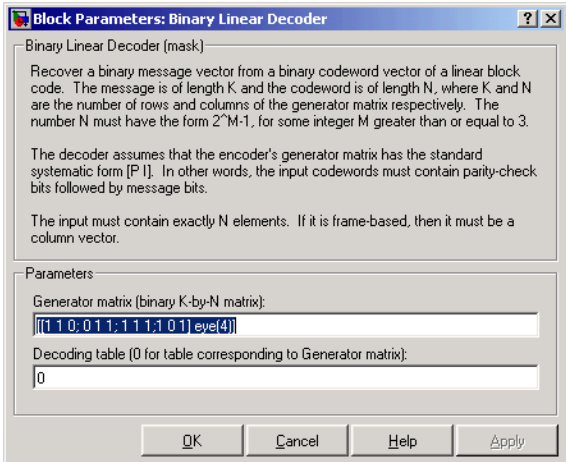


The **Generator matrix** parameter is the generator matrix for the block code. For proper decoding, this should match the **Generator matrix** parameter in the correspondingBinary Linear Encoder block. If N is the codeword length of the code, then **Generator matrix** must have N columns. If K is the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

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

The decoder tries to correct errors, using the **Decoding table** parameter. If **Decoding table** is the scalar 0, then the block defaults to the table produced by the Communications Toolbox function `syndtable`. Otherwise, **Decoding table** must be a 2^{N-K}-by-N binary matrix. The *r*th row of this matrix is the correction vector for a received binary codeword whose syndrome has decimal integer value *r*-1. The syndrome of a received codeword is its product with the transpose of the parity-check matrix.

Dialog Box



Generator matrix

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

Decoding table

Either a 2^{N-K}-by-N matrix that lists correction vectors for each codeword's syndrome; or the scalar 0, in which case the block defaults to the table corresponding to the **Generator matrix** parameter.

Pair Block

Binary Linear Encoder

Binary Linear Encoder

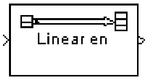
Purpose

Create linear block code from binary vector data

Library

Block sublibrary of Channel Coding

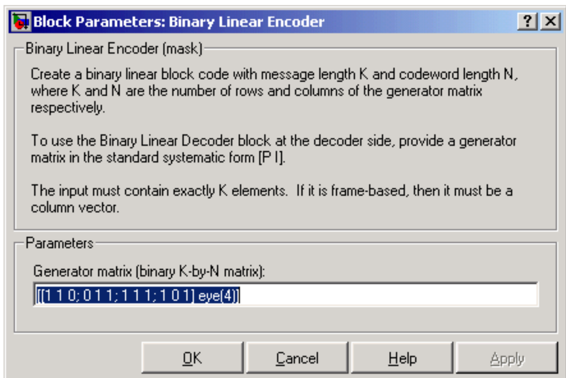
Description



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

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

Dialog Box



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

Pair Block

Binary Linear Decoder

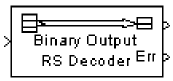
Purpose

Decode Reed-Solomon code to recover binary vector data

Library

Block sublibrary of Channel Coding

Description



The Binary-Output RS Decoder block recovers a binary message vector from a binary Reed-Solomon codeword vector. For proper decoding, the parameter values in this block should match those in the correspondingBinary-Input RS Encoder block.

The Reed-Solomon code has message length K and codeword length N. You specify both N and K directly in the dialog box. The symbols for the code are binary sequences of length M, corresponding to elements of the Galois field GF(2^M), where the first bit in each sequence is the most significant bit. Restrictions on M and N are described in “Restrictions on the M and the Codeword Length N” on page 2-44. The difference N-K must be an even integer.

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

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 2-44 below. If N is less than 2^M-1, the block uses a shortened Reed-Solomon code.

You can also specify the generator polynomial for the Reed-Solomon code, as described in “Specifying the Generator Polynomial” on page 2-44.

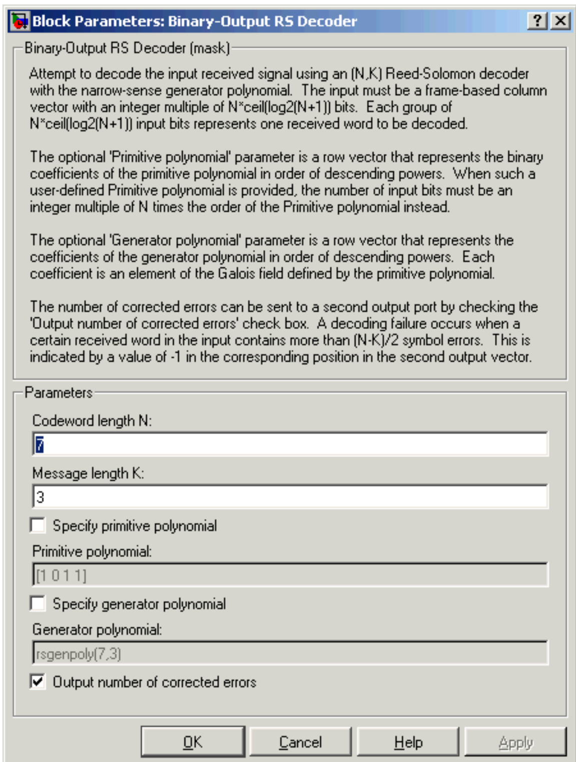
Each M*K input bits represent K integers between 0 and 2^M-1. Similarly, each M*N output bits represent N integers between 0 and

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

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. The data type of this output is also inherited from the input signal.

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

Dialog Box



Codeword length N
The codeword length. The input has vector length M*N.

Message length K
The message length. The first output has vector length M*K.

Specify primitive polynomial
When you select this box, you can specify the primitive polynomial as a binary row vector.

Primitive polynomial
Binary row vector representing the primitive polynomial in descending order of powers.

Specify generator polynomial
When you select this box, you can specify the generator polynomial as an integer row vector.

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

Output number of corrected errors
When you select this box, the block outputs the number of corrected errors in each word through a second output port.

Algorithm

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

Pair Block

Binary-Input RS Encoder

References

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

[2] Berlekamp, Elwyn R., *Algebraic Coding Theory*, New York, McGraw-Hill, 1968.

See Also

Integer-Output RS Decoder

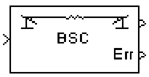
Purpose

Introduce binary errors

Library

Channels

Description

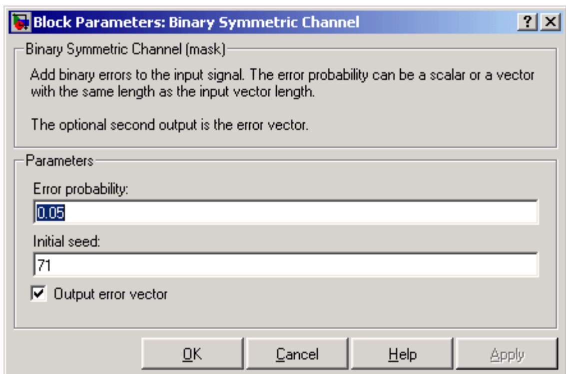


The Binary Symmetric Channel block introduces binary errors to the signal transmitted through this channel.

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

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

Dialog Box



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

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.

See Also

Bernoulli Binary Generator

Purpose

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

Library

Utility Blocks

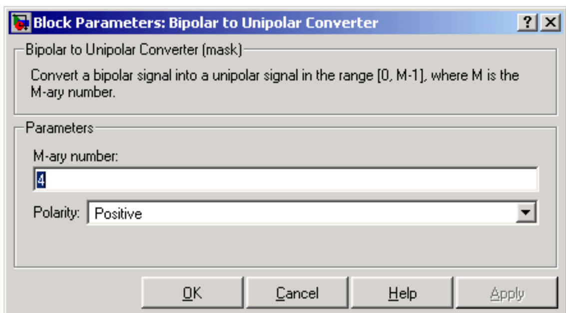
Description

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

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

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

Dialog Box



M-ary number
: The number of symbols in the bipolar or unipolar alphabet.

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

Examples

If the input is [-3; -1; 1; 3], the **M-ary number** parameter is `4`, and the **Polarity** parameter is `Positive`, then the output is [0; 1; 2; 3]. Changing the **Polarity** parameter to `Negative` changes the output to [3; 2; 1; 0].

Pair Block

Unipolar to Bipolar Converter

Purpose

Map vector of bits to corresponding vector of integers

Library

Utility Blocks

Description

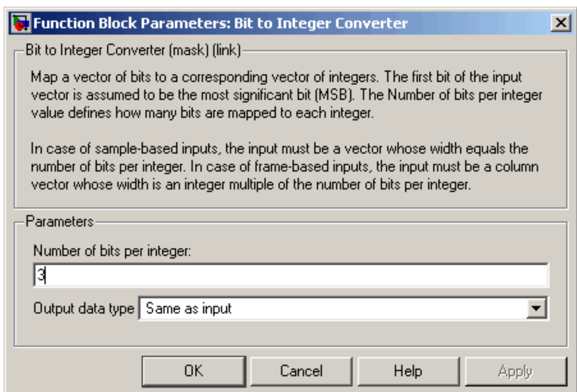
The Bit to Integer Converter block maps groups of bits in the input vector to integers in the output vector. If M is the **Number of bits per integer** parameter, then the block maps each group of M bits to an integer between 0 and 2^M-1. As a result, the output vector length is 1/M times the input vector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

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

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

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

Dialog Box



Number of bits per integer

The number of input bits that the block maps to each integer of the output. This parameter must be an integer between 1 and 31.

Output data type

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

Examples

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

Pair Block

Integer to Bit Converter

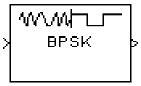
Purpose

Demodulate BPSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

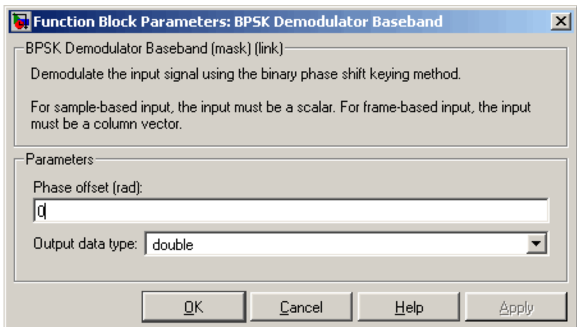
Description



The BPSK Demodulator Baseband block demodulates a signal that was modulated using the binary phase shift keying method. The input is a baseband representation of the modulated signal. The input can be either a scalar or a frame-based column vector. The block can accept the data types `single` and `double`.

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

Dialog Box



Phase offset (rad)
: The phase of the zeroth point of the signal constellation.

Output data type
: The output data type can be `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, or `double`.

Pair Block

BPSK Modulator Baseband

See Also

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

BPSK Modulator Baseband

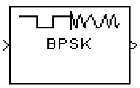
Purpose

Modulate using binary phase shift keying method

Library

PM, in Digital Baseband sublibrary of Modulation

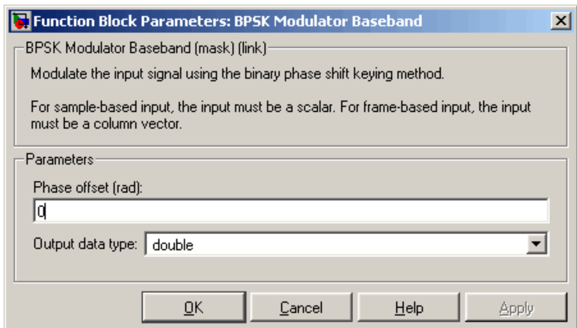
Description



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

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

Dialog Box



Phase offset (rad)
: The phase of the zeroth point of the signal constellation.

Output data type
: The block can output the data types `single` and `double`.

Pair Block

BPSK Demodulator Baseband

See Also

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

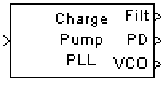
Purpose

Implement charge pump phase-locked loop using digital phase detector

Library

Components sublibrary of Synchronization

Description



The Charge Pump PLL (phase-locked loop) block automatically adjusts the phase of a locally generated signal to match the phase of an input signal. It is suitable for use with digital signals.

This PLL has these three components:

- A sequential logic phase detector, also called a digital phase detector or a phase/frequency detector.
- A filter. You specify the filter's transfer function using the **Lowpass filter numerator** and **Lowpass filter denominator** parameters. Each is a vector that gives the respective polynomial's coefficients in order of descending powers of s.

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

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

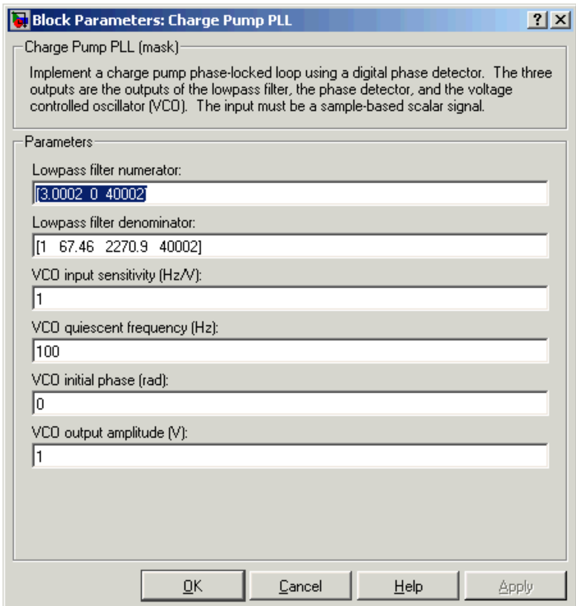
- A voltage-controlled oscillator (VCO). You specify characteristics of the VCO using the **VCO input sensitivity**, **VCO quiescent frequency**, **VCO initial phase**, and **VCO output amplitude** parameters.

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

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

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 π. 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 Using the Communications Blockset.

CMA Equalizer

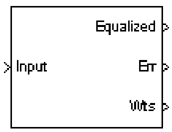
Purpose

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.

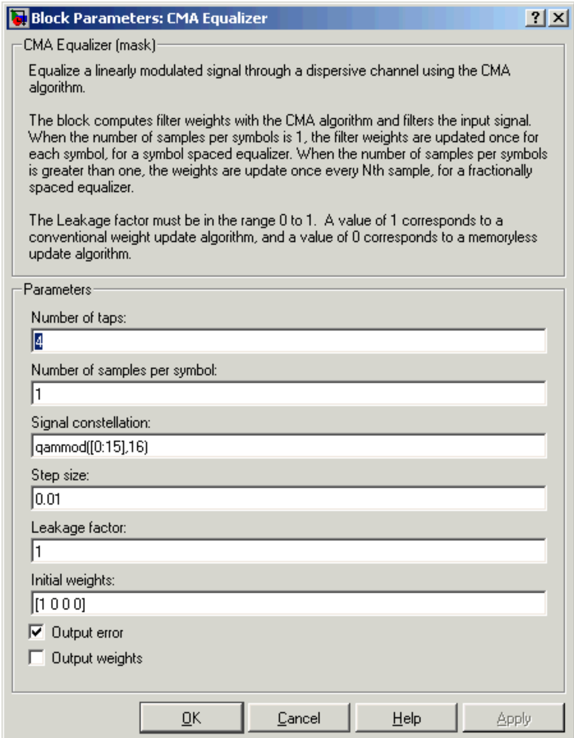
You can configure the block to have one or more of the extra ports listed in the table below.

Port	Meaning	How to Enable
`Err` output	$y(R - \|y\|^2)$, 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 sample-based mode, the current input sample.	Check the **Output weights** check box.

Equalizer Delay

The delay between the transmitter's modulator output and the CMA equalizer output is typically unknown (unlike the delay for other adaptive equalizers in this blockset). If you need to determine the delay, you can use the Find Delay block.

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

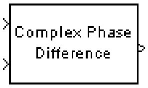
Purpose

Output phase difference between two complex input signals

Library

Utility Blocks

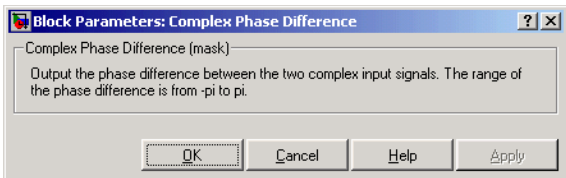
Description



The Complex 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 -π and π.

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

Dialog Box



See Also

Complex Phase Shift

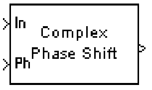
Purpose

Shift phase of complex input signal by second input value

Library

Utility Blocks

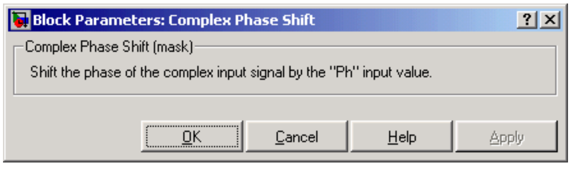
Description



The Complex Phase Shift block accepts a complex signal at the port labeled `In`. The output is the result of shifting this signal's phase by an amount specified by the real signal at the input port labeled `Ph`. The `Ph` input is measured in radians, and must have the same size and frame status as the `In` input.

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

Dialog Box



See Also

Complex Phase Difference

Continuous-Time VCO

Purpose

Implement voltage-controlled oscillator

Library

Components sublibrary of Synchronization

Description

The Continuous-Time VCO (voltage-controlled oscillator) block generates a signal whose frequency shift from the **Quiescent frequency** parameter is proportional to the input signal. The input signal is interpreted as a voltage. If the input signal is $u(t)$, then the output signal is

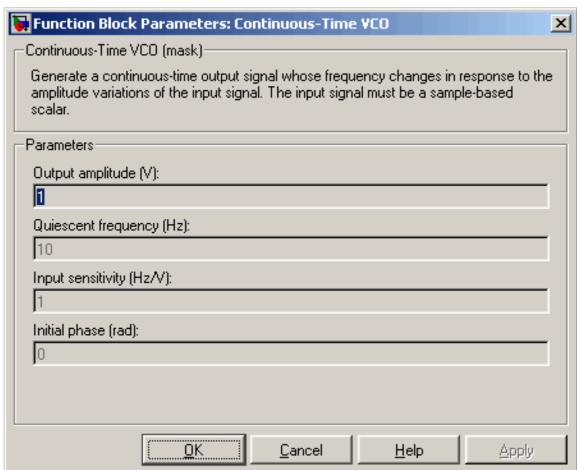
$$y(t) = A_c \cos\left(2\pi f_c t + 2\pi k_c \int_0^t u(\tau)d\tau + \varphi\right)$$

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

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

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

Dialog Box



Output amplitude
The amplitude of the output.

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

Input sensitivity
This value scales the input voltage and, consequently, the shift from the **Quiescent frequency** value. The units of **Input sensitivity** are Hertz per volt.

Initial phase
The initial phase of the oscillator in radians.

See Also

Discrete-Time VCO

Convolutional Deinterleaver

Purpose

Restore ordering of symbols that were permuted using shift registers

Library

Convolutional sublibrary of Interleaving

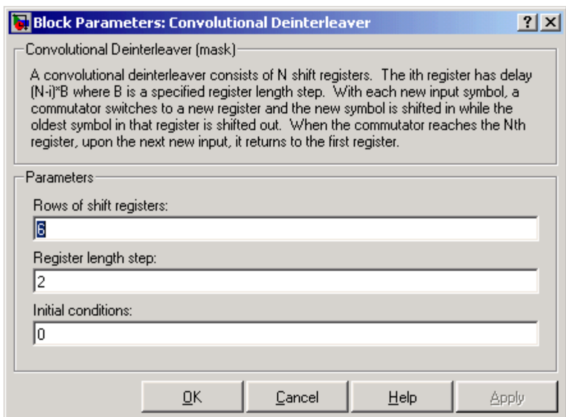
Description

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

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

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

Dialog Box



Rows of shift registers
: The number of shift registers that the block uses internally.

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

Initial conditions

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

Examples

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

Pair Block

Convolutional Interleaver

See Also

General Multiplexed Deinterleaver, Helical Deinterleaver

References

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

[2] Forney, G., D., Jr. "Burst-Correcting Codes for the Classic Bursty Channel." *IEEE Transactions on Communications*, vol. COM-19, October 1971. 772-781.

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

Convolutional Encoder

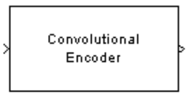
Purpose

Create convolutional code from binary data

Library

Convolutional sublibrary of Channel Coding

Description



The Convolutional Encoder block encodes a sequence of binary input vectors to produce a sequence of binary output vectors. This block can process multiple symbols at a time.

Input and Output Sizes

If the encoder takes k input bit streams (that is, can receive 2^k possible input symbols), then 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), then 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`, and `uint32`. The port data types are inherited from the signals that drive the block. The input reset port supports `double` and `boolean` typed signals.

Specifying the Encoder

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

- If you have a variable in the MATLAB workspace that contains the trellis structure, then 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 in the next bulleted item.
- If you want to specify the encoder using its constraint length, generator polynomials, and possibly feedback connection polynomials,

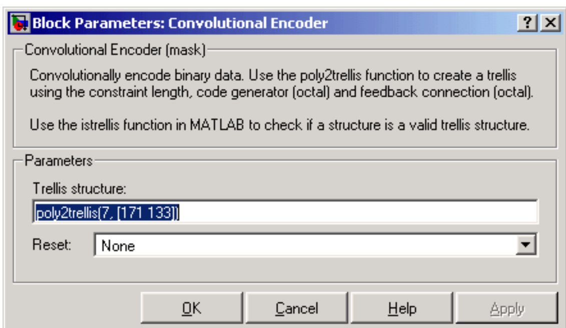
then use a `poly2trellis` command within the **Trellis structure** field. For example, to use an encoder with a constraint length of 7, code generator polynomials of 171 and 133 (in octal numbers), and a feedback connection of 171 (in octal), set the **Trellis structure** parameter to

```
poly2trellis(7,[171 133],171)
```

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, use one of these values of the **Reset** parameter:

- The value `None` indicates that the encoder never resets.
- The value `On each frame` indicates that the encoder resets at the beginning of each frame, before processing the next frame of input data
- The value `On nonzero Rst input` causes the block to have 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.

Reset

Determines whether and under what circumstances the encoder resets to the all-zeros state before processing the input data. Choices are `None`, `On each frame`, and `On nonzero Rst input`. The last option causes the block to have a second input port, labeled `Rst`.

See Also

Viterbi Decoder, APP Decoder

References

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

[2] Gitlin, Richard D., Jeremiah F. Hayes, and Stephen B. Weinstein. *Data Communications Principles*. New York: Plenum, 1992.

Purpose Permute input symbols using set of shift registers

Library Convolutional sublibrary of Interleaving

Description

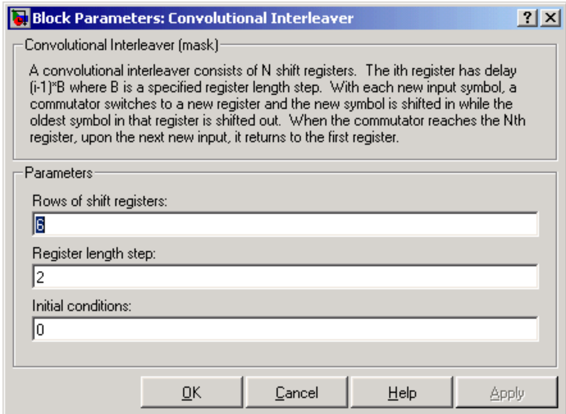
The Convolutional Interleaver block permutes the symbols in the input signal. Internally, it uses a set of shift registers. The delay value of the kth shift register is (k-1) times the **Register length step** parameter. The number of shift registers is the value of the **Rows of shift registers** parameter.

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.

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

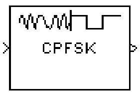
Purpose

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 π radians is the phase shift in the modulated signal due to the latest symbol, when that symbol is the integer 1. The **Phase offset** parameter is the initial phase of the modulated waveform.

Traceback Length and Output Delays

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The **Traceback length** parameter, D, in this block is the number of trellis branches used to construct each traceback path. D influences the output delay, which is the number of zero symbols that precede the first meaningful demodulated value in the output.

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

Outputs and Symbol Sets

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

If the **Output type** parameter is set to `Bit`, then the block produces groupings of K bits. Each grouping is called a binary *word*.

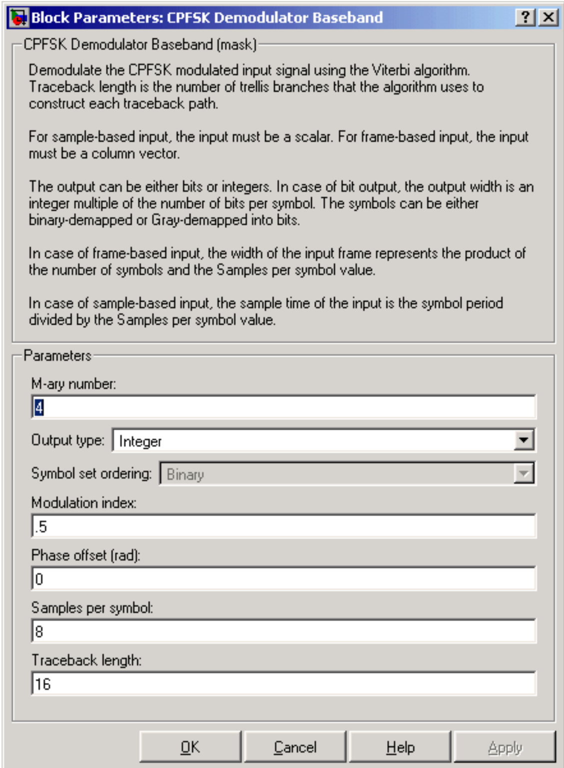
In binary output mode, the block first maps each input symbol to an intermediate value as in the integer output mode. The block then maps the odd integer k to the nonnegative integer (k+M-1)/2. Finally, the block maps each nonnegative integer to a binary word, using a mapping that depends on whether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

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

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 Using the Communications Blockset.

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 Viterbi Decoder block uses to construct each traceback path.

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

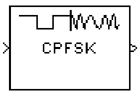
Purpose

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 π 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.

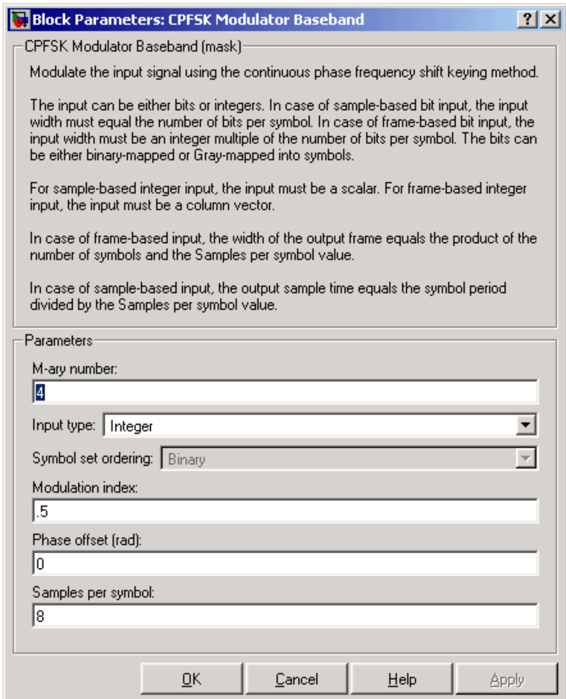
In binary input mode, the block maps each binary word to an integer between 0 and M-1, using a mapping that depends on whether the **Symbol set ordering** parameter is set to `Binary` or `Gray`. The block then maps the integer k to the intermediate value 2k-(M-1) and proceeds as in the integer input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

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

Upsampling the Modulated Signal

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

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.

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.

Pair Block

CPFSK Demodulator Baseband

See Also

CPM Modulator Baseband, M-FSK Modulator Baseband

References

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

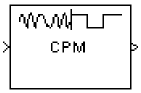
Purpose

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.

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 theCPM Modulator Baseband block.

Traceback Length and Output Delays

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The **Traceback length** parameter, D, in this block is the number of trellis branches used to construct each traceback path. D influences the output delay, which is the number of zero symbols that precede the first meaningful demodulated value in the output.

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

Outputs and Symbol Sets

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

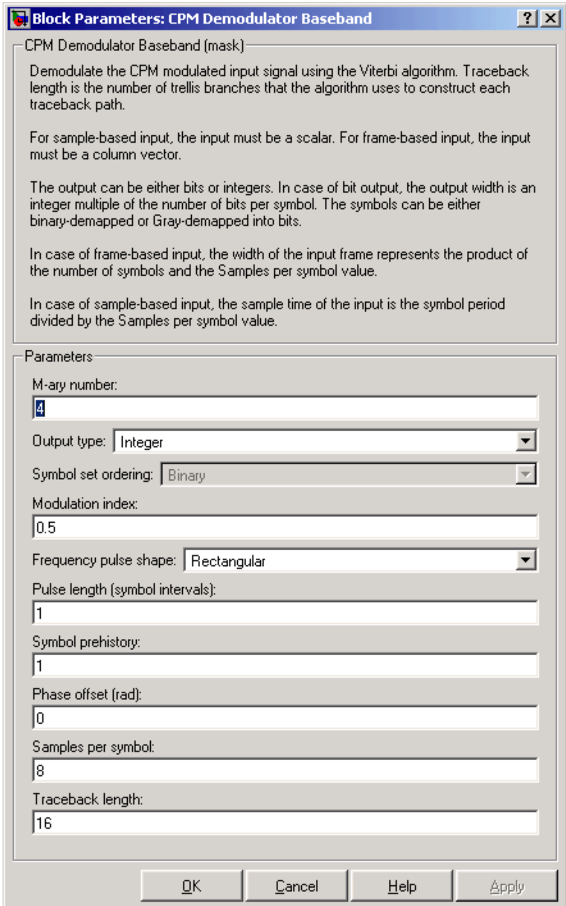
If the **Output type** parameter is set to `Bit`, then the block produces groupings of K bits. Each grouping is called a binary *word*.

In binary output mode, the block first maps each input symbol to an intermediate value as in the integer output mode. The block then maps the odd integer k to the nonnegative integer (k+M-1)/2. Finally, the block maps each nonnegative integer to a binary word, using a mapping that depends on whether the **Symbol set ordering** parameter is set to `Binary` or `Gray`. For more information about Gray and binary coding, see "Binary-Valued and Integer-Valued Signals" in Using the Communications Blockset.

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 Using the Communications Blockset.

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.

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.

Traceback length

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

Pair Block

CPM Modulator Baseband

See Also

CPFSK Demodulator Baseband, GMSK Demodulator Baseband, MSK Demodulator Baseband, Viterbi Decoder

References

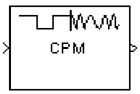
[1] Anderson, John B., Tor Aulin, and Carl-Erik Sundberg. *Digital Phase Modulation*. New York: Plenum Press, 1986.

CPM Modulator Baseband

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 π 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.

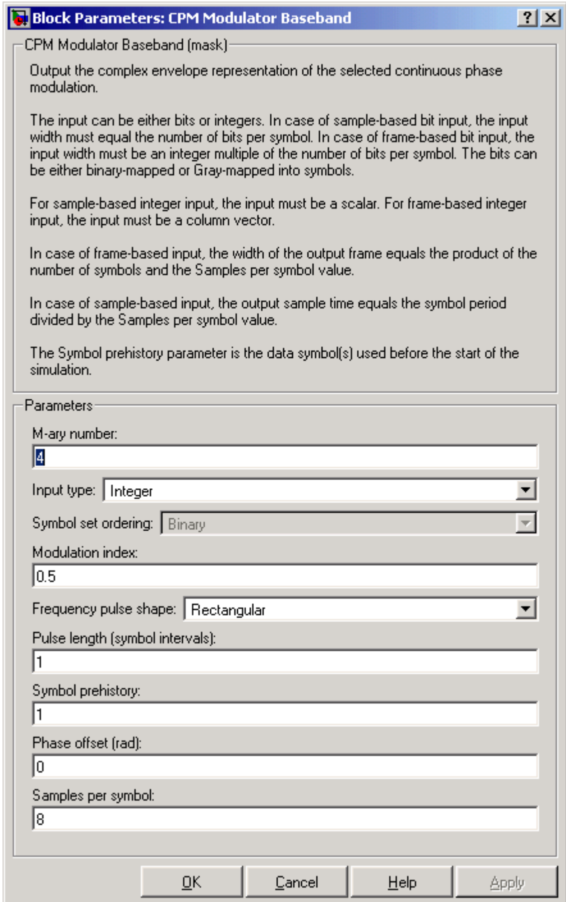
In binary input mode, the block maps each binary word to an integer between 0 and M-1, using a mapping that depends on whether the **Symbol set ordering** parameter is set to `Binary` or `Gray`. The block then maps the integer k to the intermediate value 2k-(M-1) and proceeds as in the integer input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

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

Upsampling the Modulated Signal

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

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.

Pair Block

CPM Demodulator Baseband

See Also

CPFSK Modulator Baseband, GMSK Modulator Baseband, MSK Modulator Baseband

References

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

CPM Phase Recovery

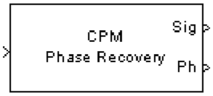
Purpose

Recover carrier phase using 2P-Power method

Library

Carrier Phase Recovery sublibrary of Synchronization

Description



The CPM Phase Recovery block recovers the carrier phase of the input signal using the 2P-Power method. This feedforward, non-data-aided, clock-aided method is suitable for systems that use these types of baseband modulation: continuous phase modulation (CPM), minimum shift keying (MSK), continuous phase frequency shift keying (CPFSK), and Gaussian minimum shift keying (GMSK). This block is suitable for use with blocks in the Baseband Continuous Phase Modulation library.

If you express the modulation index for CPM as a proper fraction, $h = K / P$, then P is the number to which the name "2P-Power" refers.

The 2P-Power method assumes that the carrier phase is constant over a series of consecutive symbols, and returns an estimate of the carrier phase for the series. The **Observation interval** parameter is the number of symbols for which the carrier phase is assumed constant. This number must be an integer multiple of the input signal's vector length.

Input and Outputs

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

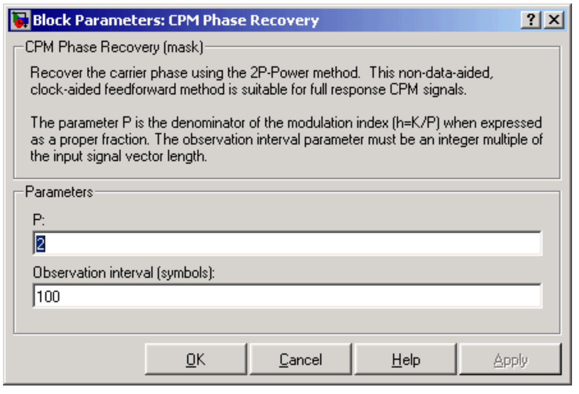
- The output port labeled `Sig` gives the result of rotating the input signal counterclockwise, where the amount of rotation equals the carrier phase estimate. The `Sig` output is thus a corrected version of the input signal, and has the same sample time and vector size as the input signal.
- The output 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.

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

CRC-N Generator

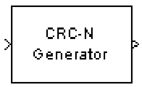
Purpose

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

Library

CRC sublibrary of Error Detection and Correction

Description



The CRC-N Generator block generates cyclic redundancy code (CRC) bits for each input data frame and appends them to the frame. The CRC-N Generator block is a simplified version of the General CRC Generator block. With the CRC-N Generator block, you can select the generator polynomial for the CRC algorithm from a list of commonly used polynomials, given in the **CRC-N method** field in the block's dialog. N is degree of the generator polynomial. The table below lists the options for the generator polynomial.

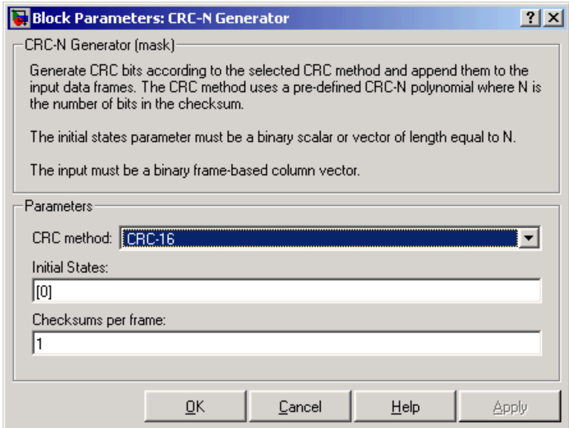
CRC Method	Generator Polynomial	Number of Bits
CRC-32	$x^{32}+x^{26}+x^{23}+x^{22}+x^{16}+x^{12}+x^{11}+x^{10}+x^{8}+x^{7}+x^{5}+x^{4}+x^{2}+x+1$	32
CRC-24	$x^{24}+x^{23}+x^{14}+x^{12}+x^{8}+1$	24
CRC-16	$x^{16}+x^{15}+x^{2}+1$	16
Reversed CRC-16	$x^{16}+x^{14}+x+1$	16
CRC-8	$x^{8}+x^{7}+x^{6}+x^{4}+x^{2}+1$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

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

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 Using the Communications Blockset.

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.

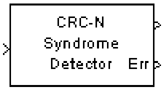
Pair Block CRC-N Syndrome Detector

See Also General CRC Generator, General CRC Syndrome Detector

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

Library CRC sublibrary of Error Detection and Correction

Description



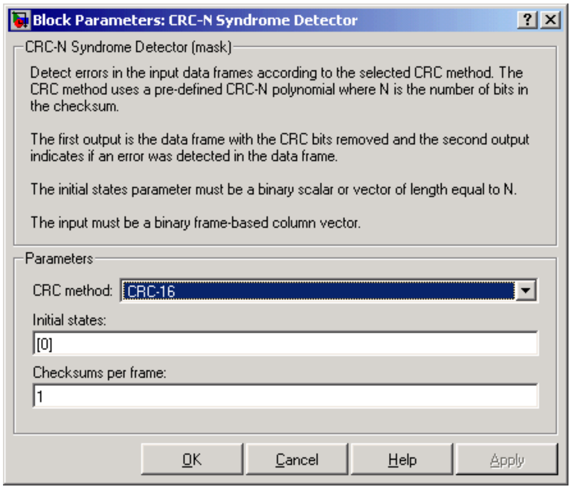
The CRC-N Syndrome Detector block computes checksums for its entire input frame. The block's second output is a vector whose size is the number of checksums, and whose entries are 0 if the checksum computation yields a zero value, and 1 otherwise. The block's first output is the set of message words with the checksums removed.

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

The parameter settings for the CRC-N Syndrome Detector block should match those of the CRC-N Generator block.

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

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 Using the Communications Blockset.

References

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

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

Pair Block

CRC-N Generator

See Also

General CRC Generator, General CRC Syndrome Detector

Data Mapper

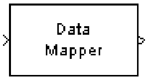
Purpose

Map integer symbols from one coding scheme to another

Library

Utility Blocks

Description



The Data Mapper block accepts integer inputs and produces integer outputs. You can select one of four mapping modes: `Binary to Gray`, `Gray to Binary`, `User Defined`, or `Straight Through`.

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

Gray coding is an ordering of binary numbers such that all adjacent numbers differ by only one bit. However, the inputs and outputs of this block are integers, not binary vectors. As a result, the first two mapping modes perform code conversions as follows:

- In the `Binary to Gray` mode, the output from this block is the integer equivalent of the Gray code bit representation for the input integer.
- In the `Gray to Binary` mode, the output from this block is the integer position of the binary equivalent of the input integer in a Gray code ordering.

As an example, the table below shows both the `Binary to Gray` and `Gray to Binary` mappings for integers in the range 0 to 7. In the Binary to Gray Mode Output column, notice that binary representations in successive rows differ by exactly one bit. In the Gray to Binary Mode columns, notice that sorting the rows by Output value creates a Gray code ordering of Input binary representations.

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

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

$0 \rightarrow 1$

$1 \rightarrow 5$

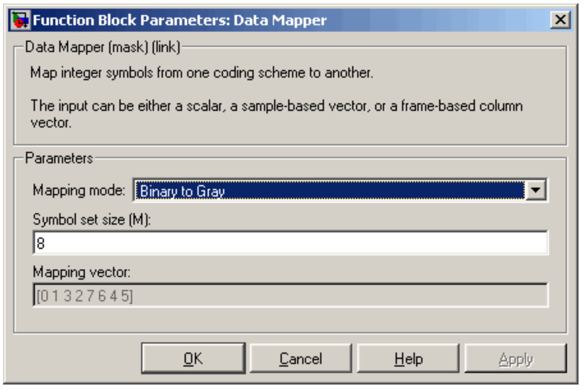
$2 \rightarrow 0$

$3 \rightarrow 4$

$4 \rightarrow 2$

$5 \rightarrow 3$

When you select the `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 is active only when **Mapping mode** is set to `User Defined`.

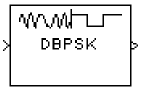
Purpose

Demodulate DBPSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

Description

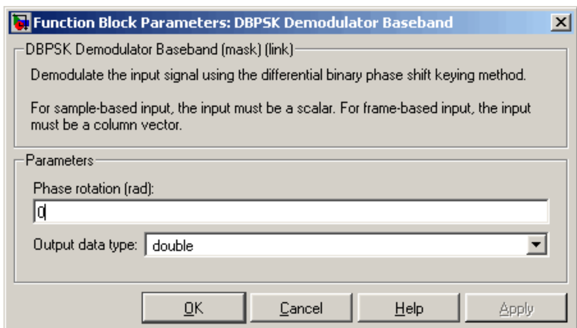


The DBPSK Demodulator Baseband block demodulates a signal that was modulated using the differential binary phase shift keying method. The input is a baseband representation of the modulated signal.

The input must be a discrete-time complex signal. The block compares the current symbol to the previous symbol. It maps phase differences of θ and π+θ, respectively, to outputs of 0 and 1, respectively, where θ 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
: For both integer and bit inputs, this block can output the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, and `double`.

Pair Block DBPSK Modulator Baseband

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

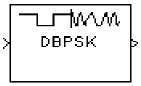
Purpose

Modulate using differential binary phase shift keying method

Library

PM, in Digital Baseband sublibrary of Modulation

Description

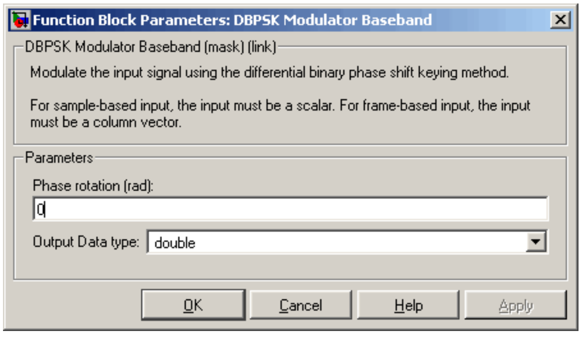


The DBPSK Modulator Baseband block modulates using the differential binary phase shift keying method. The output is a baseband representation of the modulated signal.

The input must be a discrete-time binary-valued signal. The input can be either a scalar or a frame-based column vector. For both integer and bit inputs, the block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, and `double`. These rules govern this modulation method when the **Phase rotation** parameter is θ:

- If the first input bit is 0 or 1, respectively, then the first modulated symbol is exp(jθ) or -exp(jθ), respectively.
- If a successive input bit is 0 or 1, respectively, then the modulated symbol is the previous modulated symbol multiplied by exp(jθ) or -exp(jθ), respectively.

Dialog Box



Phase rotation (rad)

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

Output Data type

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

Pair Block

DBPSK Demodulator Baseband

See Also

DQPSK Modulator Baseband, BPSK Modulator Baseband

Purpose

Distribute elements of input vector alternately between two output vectors

Library

Sequence Operations

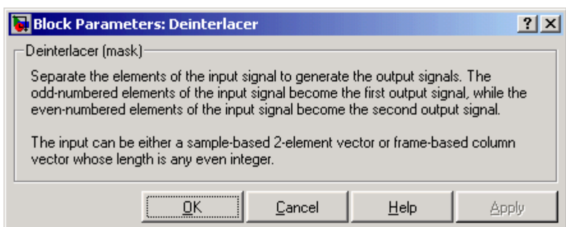
Description

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

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

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

Dialog Box



Examples

If the input vector is frame-based with value [1; 5; 2; 6; 3; 7; 4; 8], then the two output vectors are [1; 2; 3; 4] and [5; 6; 7; 8]. Notice that this is the inverse of the example on the reference page for the Interlacer block.

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

Pair Block

Interlacer

See Also

Demux (Simulink)

Purpose

Reduce sampling rate by averaging consecutive samples

Library

Sequence Operations

Description

The Derepeat block resamples the discrete input at a rate 1/N times the input sample rate by averaging N consecutive samples. This is one possible inverse of the Repeat block (Signal Processing Blockset). The positive integer N is the **Derepeat factor** parameter in the Derepeat dialog.

The **Initial condition** parameter prescribes elements of the output when it is still too early for the input data to show up in the output. If the dimensions of the **Initial condition** parameter match the output dimensions, then the parameter represents the initial output value. If **Initial condition** is a scalar, then it represents the initial value of each element in the output.

The input can have any shape or frame status. The block can accept the data types `single` and `double`. The data type of the output will be the same as that of the input signal.

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

Sample-Based Operation

If the input is sample-based, then the block assumes that the input is a vector or matrix whose elements represent samples from independent channels. The block averages samples from each channel independently over time. The output period is N times the input period, and the input and output sizes are identical. The output is delayed by one output period, and the first output value is the **Initial condition** value.

Frame-Based Operation

If the input is frame-based, then the block derepeats each frame, treating distinct channels independently. Each element of the output is the average of N consecutive elements along a *column* of the input matrix. The **Derepeat factor** must be less than the frame size.

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

- `Maintain input frame size`

  The block reduces the sampling rate by using a proportionally longer frame *period* at the output port than at the input port. For derepetition by a factor of N, the output frame period is N times the input frame period, but the input and output frame sizes are equal. The output is delayed by one output frame, and the first output frame is determined only by the **Initial condition** value.

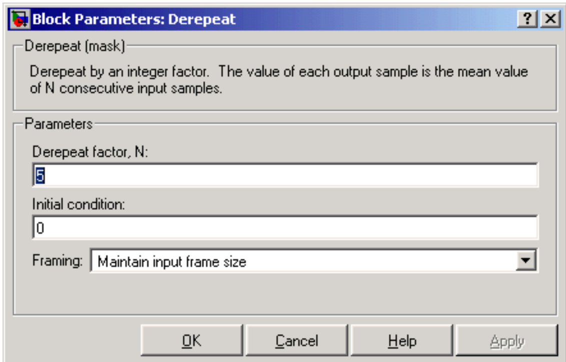
  For example, if a single-channel input with a frame period of 1 second is derepeated by a factor of 4, then the output has a frame period of 4 seconds. The input and output frame sizes are equal.

- `Maintain input frame rate`

  The block reduces the sampling rate by using a proportionally smaller frame *size* than the input. For derepetition by a factor of N, the output frame size is 1/N times the input frame size, but the input and output frame rates are equal. When you use this option, the **Initial condition** parameter does not apply and the block incurs no delay, because the input data immediately shows up in the output.

  For example, if a single-channel input with 64 elements is derepeated by a factor of 4, then the output contains 16 elements. The input and output frame periods are equal.

Dialog Box



Derepeat factor, N
: The number of consecutive input samples to average in order to produce each output sample.

Initial condition
: The value with which to initialize the block.

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

See Also

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

Descrambler

Purpose

Descramble input signal

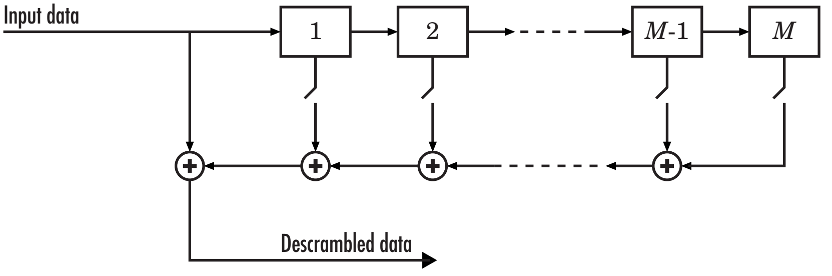
Library

Sequence Operations

Description

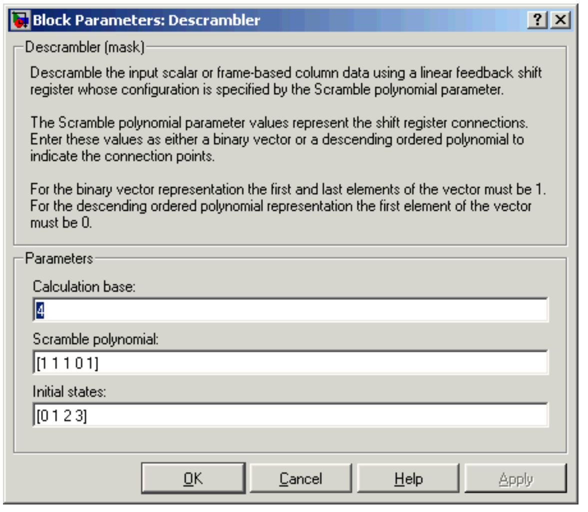
The Descrambler block descrambles the input signal, which must be a scalar or a frame-based column vector. The Descrambler block is the inverse of theScrambler block. If you use the Scrambler block in the transmitter, then you should use the Descrambler block in the receiver.

Below is a schematic of the descrambler. All adders perform addition 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. The **Initial states** can be different in the two blocks, considering the transmitting and receiving filter delay. See the reference page for theScrambler block for more information about these parameters.

Dialog Box



Calculation base

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

Scramble polynomial

A polynomial that defines the connections in the scrambler.

Initial states

The states of the scrambler's registers when the simulation starts.

Pair Block

Scrambler

Differential Decoder

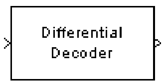
Purpose

Decode binary signal using differential coding

Library

Source Coding

Description



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

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

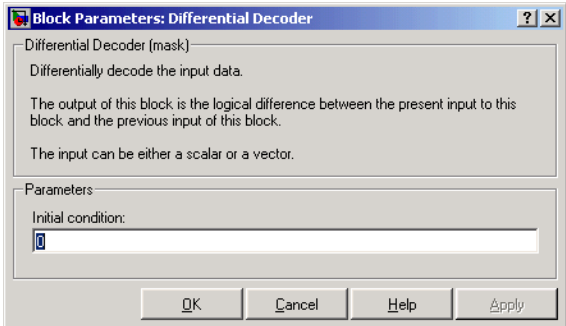
$m(t_k) = d(t_k)$ XOR $d(t_{k-1})$

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 or a vector. This block processes each vector element independently.

Dialog Box



Initial condition

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

References

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

Pair Block

Differential Encoder

Differential Encoder

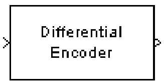
Purpose

Encode binary signal using differential coding

Library

Source Coding

Description



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

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

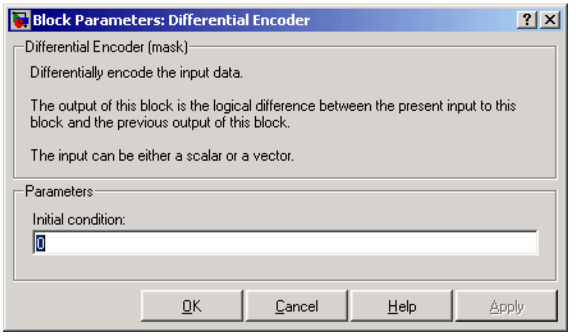
$d(t_k) = d(t_{k-1})$ XOR $m(t_k)$

where

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

The input can be either a scalar or a vector. This block processes each vector element independently.

Dialog Box



Initial condition

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

References

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

Pair Block

Differential Decoder

Discrete-Time Eye Diagram Scope

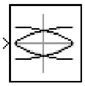
Purpose

Display multiple traces of modulated signal

Library

Comm Sinks

Description

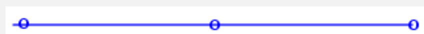


The Discrete Eye Diagram Scope block displays multiple traces of a modulated signal to produce an eye diagram. You can use the block to reveal the modulation characteristics of the signal, such as pulse shaping or channel distortions.

The Discrete-Time Eye Diagram Scope block has one input port. The input signal can be either real or complex. The input signal must be a sample-based scalar in sample-based mode. The input must be a frame-based column vector or a scalar in frame-based mode.

Marker and Line Styles

The **Marker**, **Line style**, and **Line color** parameters, on the **Rendering Properties** panel, control the appearance of the signal trajectory. The **Marker** parameter specifies the marker style for points in the eye diagram. The following table lists some of the available line markers.

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

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

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 **Traces displayed** 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 **Duplicate points at trace boundary** for modulations such as PSK and QAM. Clear to display the phase trees for MSK, CPFSK, GFSK, GMSK, and other continuous phase modulations.
Color fading	Check **Color fading** for animation that resembles an oscilloscope. Clear for greater speed and animation that resembles a plot.

Parameter	Recommended Setting
High quality rendering	Check **High quality rendering** 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.
Y-axis minimum	Approximately 10% less than the expected minimum value of the signal
Y-axis maximum	Approximately 10% greater than the expected maximum value of the signal

Scope Options

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

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

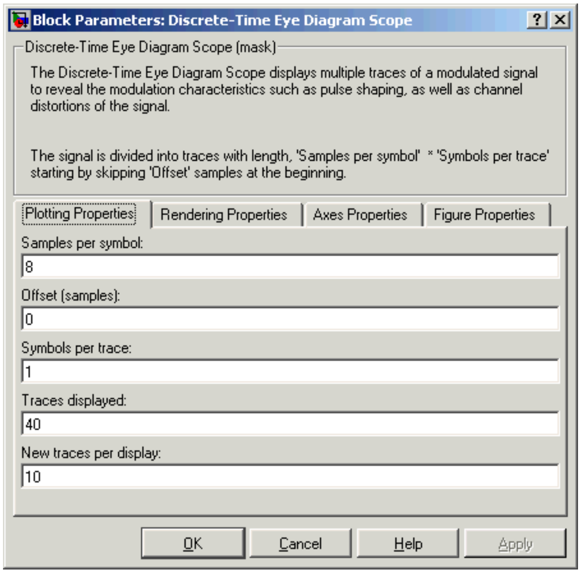
The properties listed in the **Axes** menu apply to all channels. Many of the parameters in this menu are also accessible through the block parameter dialog box. These are **Autoscale**, **Show grid**, **Frame #**, and **Save Position**. Below are descriptions of the other parameters listed in the **Axes**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



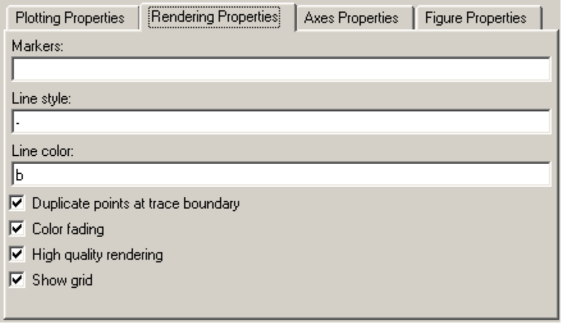
Samples per symbol
: Number of samples per symbol. Use with **Symbols per trace** to determine the number of samples per trace.

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

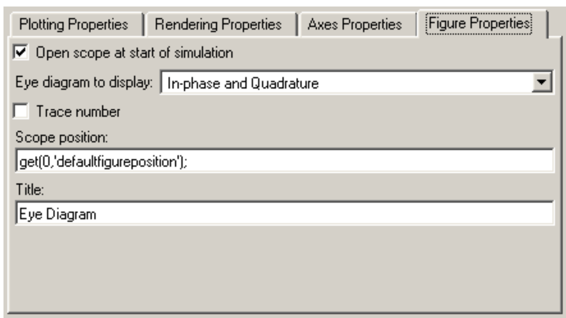
Maximum signal value the scope displays. Tunable.

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

Discrete-Time Scatter Plot Scope

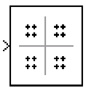
Purpose

Display the in-phase and quadrature components of modulated signal constellation

Library

Comm Sinks

Description

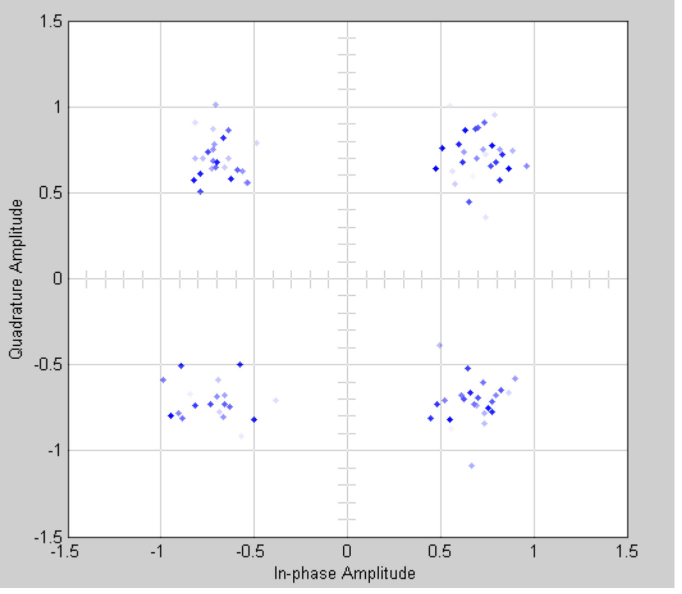


The Discrete-Time Scatter Plot Scope block displays scatter plots of a modulated signal, to reveal the modulation characteristics, such as pulse shaping or channel distortions of the signal.

The Discrete-Time Scatter Plot Scope block has one input port. The input signal must be complex. The input signal must be a sample-based scalar in sample-based mode. The input must be a frame-based column vector or a scalar in frame-based mode.

See the reference page for the Discrete-Time Signal Trajectory Scope block to compare the preceding scatter plot with the trajectory of the same signal. The Discrete-Time Signal Trajectory Scope block connects the points displayed by the Discrete-Time Scatter Plot Scope block to display the signal trajectory.

Setting **Samples per symbol** to `8`, increasing **Points displayed** to `100`, and running the model for `100` seconds produces the following scatter plot.



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.

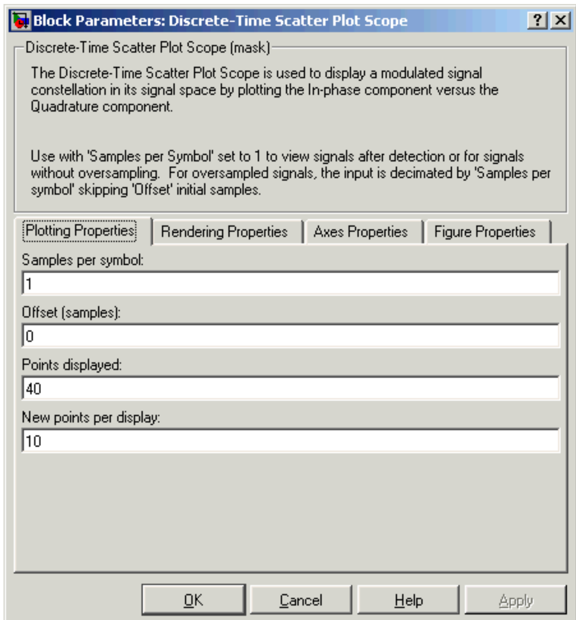
Recommended Settings

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 **Points displayed** for greater speed A small positive integer for best animation
Line style	Solid dash (-)
Line color	Blue (b)
Color fading	Check **Color fading** for animation that resembles an oscilloscope. Clear for greater speed and animation that resembles a plot.
High quality rendering	Check **High quality rendering** for higher quality rendering. Clear for greater speed.
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
X-axis minimum	Approximately 10% less than the expected minimum value of the signal
X-axis maximum	Approximately 10% greater than the expected maximum value of the signal

Dialog Box

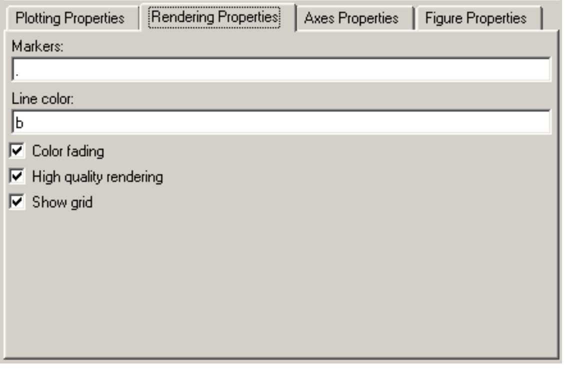


Samples per symbol
: Number of samples per symbol.

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

Points displayed
Total number of points plotted.

New points per display
Number of new points that appear in each display.



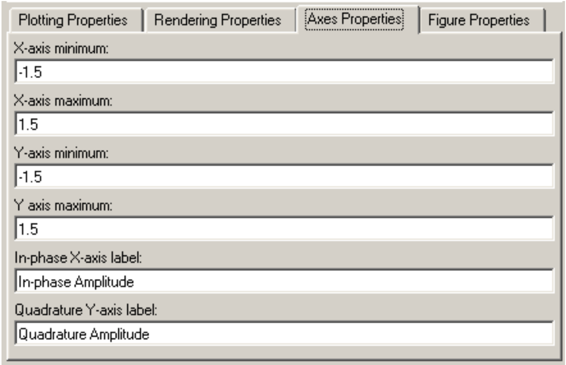
Markers
Line markers used in the scatter plot. Tunable.

Line color
The line color used in the scatter plot. Tunable.

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

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

Show grid
Toggles the scope grid on and off. Tunable.



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 the Communications Blockset illustrate how to use the Discrete-Time Scatter Plot Scope block:

- Digital Video Broadcasting Model
- DS Spread Spectrum Example

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

See Also

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

Discrete-Time Signal Trajectory Scope

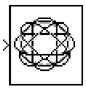
Purpose

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

Library

Comm Sinks

Description



The Discrete-Time Signal Trajectory Scope displays the trajectory of a modulated signal in its signal space by plotting its in-phase component versus its quadrature component.

The Discrete-Time Signal Trajectory Scope block has one input port. The input signal must be complex. The input signal must be a sample-based scalar in sample-based mode. The input must be a frame-based column vector or a scalar in frame-based mode.

Line Style and Color

The **Line style** and **Line color** parameters on the **Rendering Properties** panel control the appearance of the signal trajectory. The **Line style** parameter specifies the style for lines in the signal trajectory. For details on the options for these parameters, see the reference page for the Discrete-Time Eye Diagram Scope block.

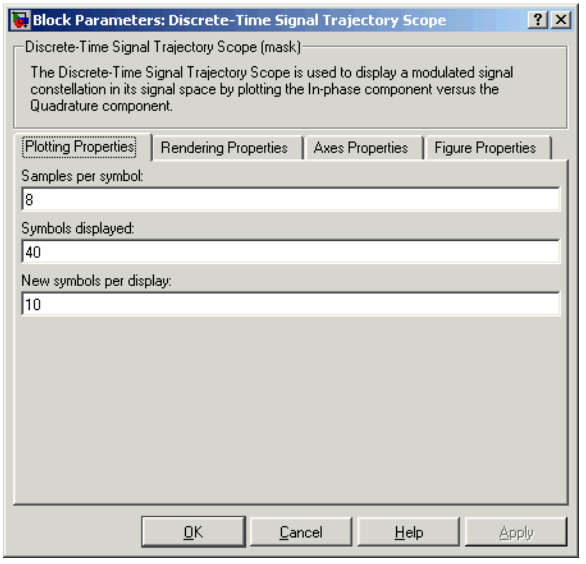
Recommended Settings

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

Parameter	Recommended Setting
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

Parameter	Recommended Setting
New symbols per display	Same as **Symbols displayed** for greater speed A small positive integer for best animation
Line style	Solid dash (`-`)
Line color	Blue (`b`)
Color fading	Check **Color fading** for animation that resembles an oscilloscope. Clear for greater speed and animation that resembles a plot.
High quality rendering	Check **High quality rendering** for higher quality rendering. Clear for greater speed.
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.
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

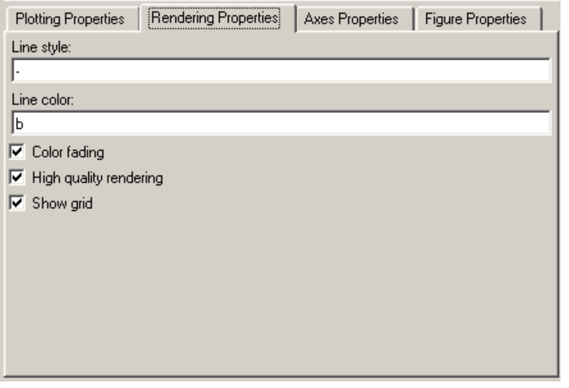
Dialog Box



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.



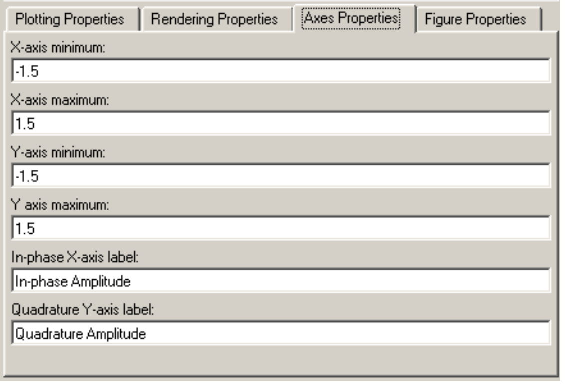
Line markers
The line markers used in the signal trajectory. Tunable.

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.

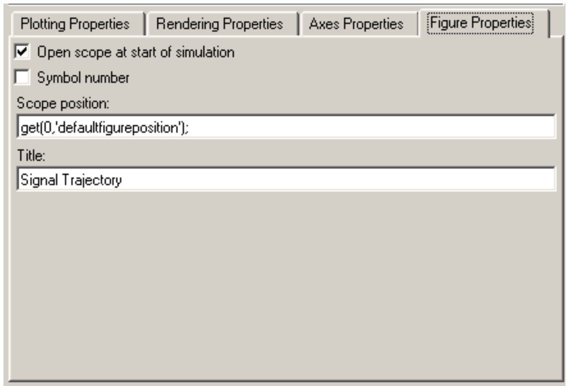
X-axis maximum
Maximum value the scope displays on the *x*-axis. Tunable.

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 the Communications Blockset 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

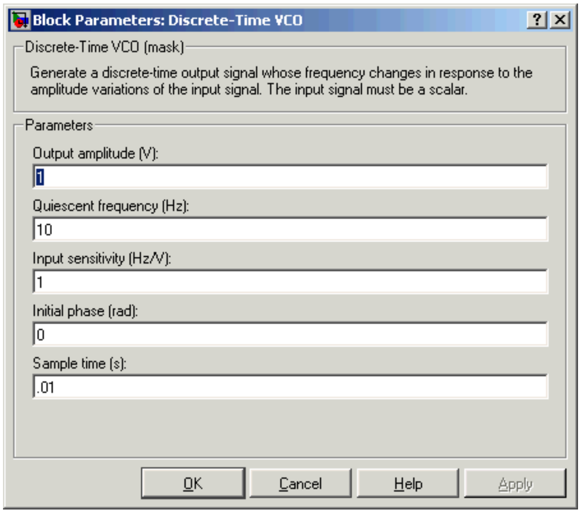
$$y(t) = A_c \cos\left(2\pi f_c t + 2\pi k_c \int_0^t u(\tau)d\tau + \varphi\right)$$

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

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

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

Dialog Box



Output amplitude
The amplitude of the output.

Quiescent frequency (Hz)
The frequency of the oscillator output when the input signal is zero.

Input sensitivity
This value scales the input voltage and, consequently, the shift from the **Quiescent frequency** value. The units of **Input sensitivity** are Hertz per volt.

Initial phase (rad)
The initial phase of the oscillator in radians.

Sample time
The calculation sample time.

See Also

Continuous-Time VCO

DQPSK Demodulator Baseband

Purpose Demodulate DQPSK-modulated data

Library PM, in Digital Baseband sublibrary of Modulation

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

The input must be a discrete-time complex signal. The output depends on the phase difference between the current symbol and the previous symbol. The first integer (or binary pair, if the **Output type** parameter is set to `Bit`) in the block's output is the initial condition of zero because there is no previous symbol.

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

Outputs and Constellation Types

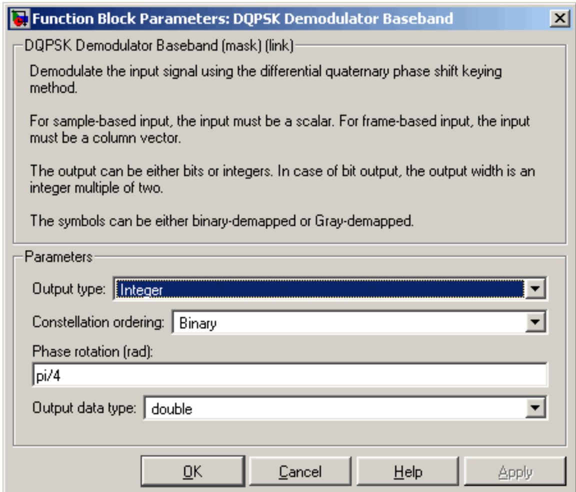
If the **Output type** parameter is set to `Integer`, then the block maps a phase difference of

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

to m, where θ 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 theDQPSK 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

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

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

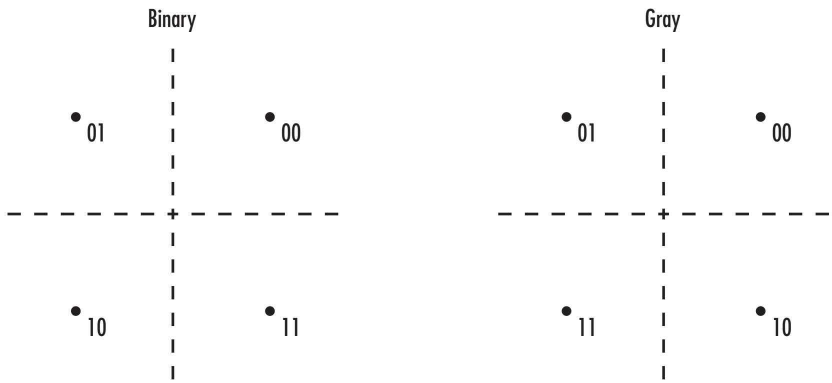
Inputs and Constellation Types

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

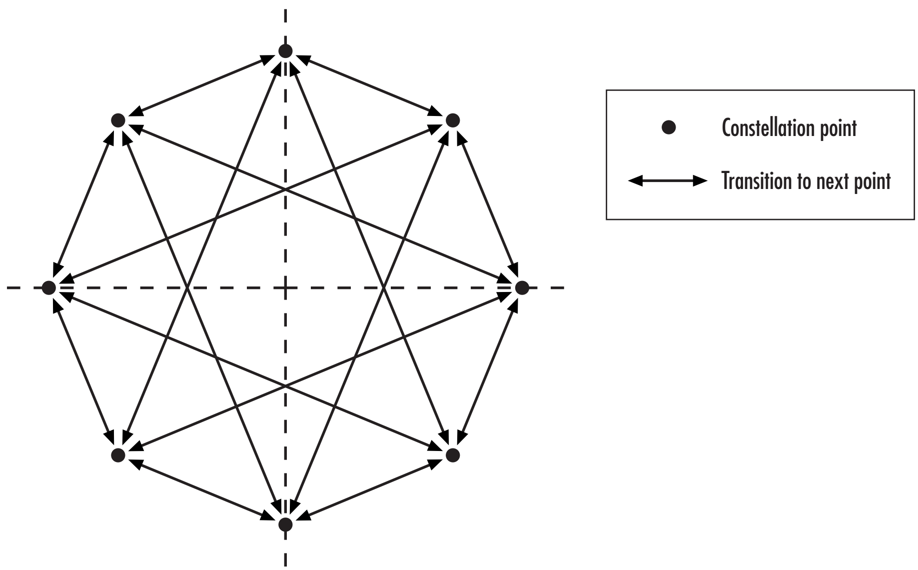
$$\exp(j\theta + j\pi m/2)$$

where θ 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.

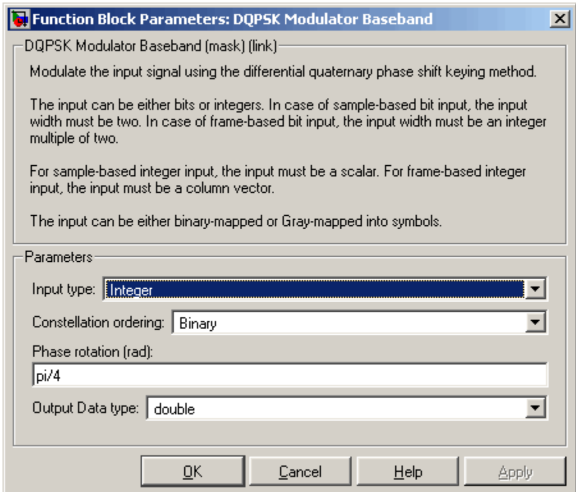


The figure below shows the signal constellation for the DQPSK modulation method when the **Phase rotation** parameter is $\pi/4$. The arrows indicate the four possible transitions from each symbol to the next symbol. The `Binary` and `Gray` options determine which transition is associated with each pair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



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

Dialog Box



Input type
: Indicates whether the input consists of integers or pairs of bits.

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

Phase rotation (rad)
: The phase difference between the previous and current modulated symbols when the input is zero.

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

Pair Block

DQPSK Demodulator Baseband

See Also

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

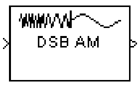
Purpose

Demodulate DSB-AM-modulated data

Library

Analog Passband Modulation, in Modulation

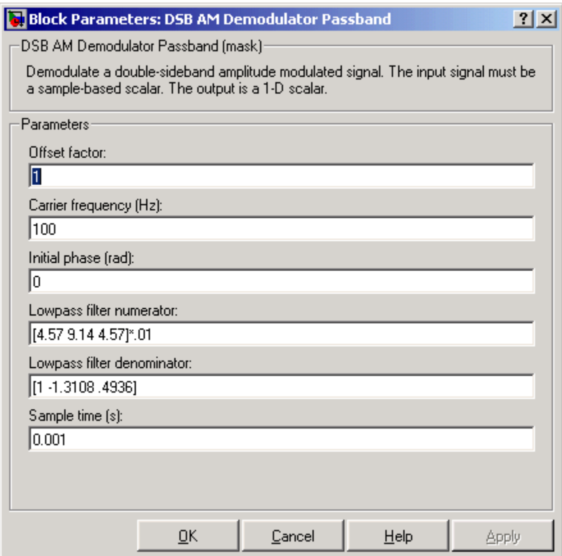
Description



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

In the course of demodulating, this block uses a filter whose transfer function is described by the **Lowpass filter numerator** and **Lowpass filter denominator** parameters.

Dialog Block



Offset factor

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 numerator
The numerator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s.

Lowpass filter denominator
The denominator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s. For an FIR filter, set this parameter to `1`.

Sample time
The sample time of the output signal.

Pair Block

DSB AM Modulator Passband

DSB AM Modulator Passband

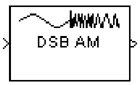
Purpose

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

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

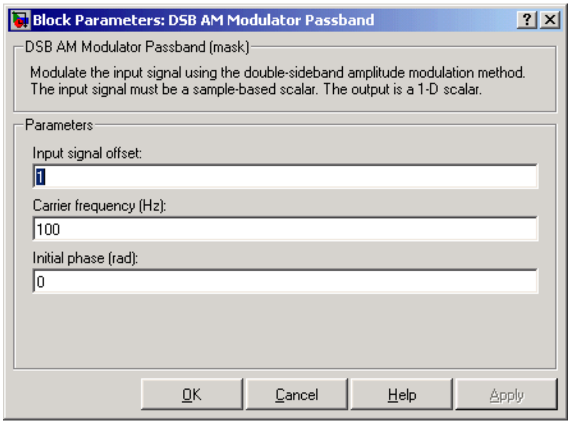
where:

- k is the **Input signal offset** parameter.
- f_c is the **Carrier frequency** parameter.
- θ 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.

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

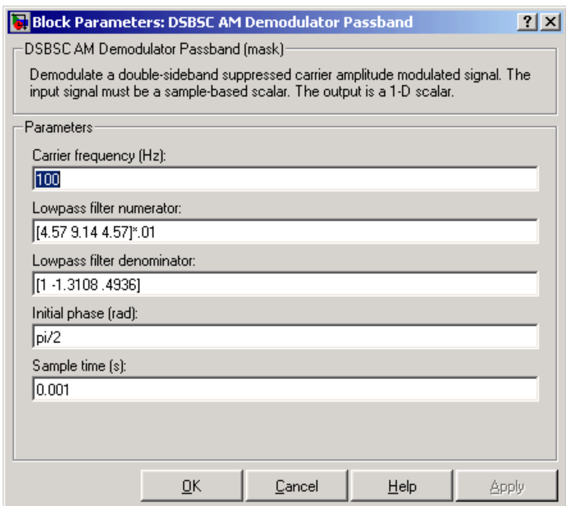
Analog Passband Modulation, in Modulation

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 transfer function is described by the **Lowpass filter numerator** and **Lowpass filter denominator** parameters.

Dialog Box



Carrier frequency (Hz)
: The carrier frequency in the corresponding DSBSC AM Modulator Passband block.

Lowpass filter numerator
The numerator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s.

Lowpass filter denominator
The denominator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s. For an FIR filter, set this parameter to `1`.

Initial phase (rad)
The initial phase of the carrier in radians.

Sample time
The sample time of the output signal.

Pair Block

DSBSC AM Modulator Passband

See Also

DSB AM Demodulator Passband, SSB AM Demodulator Passband

Purpose

Modulate using double-sideband suppressed-carrier amplitude modulation

Library

Analog Passband Modulation, in Modulation

Description

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

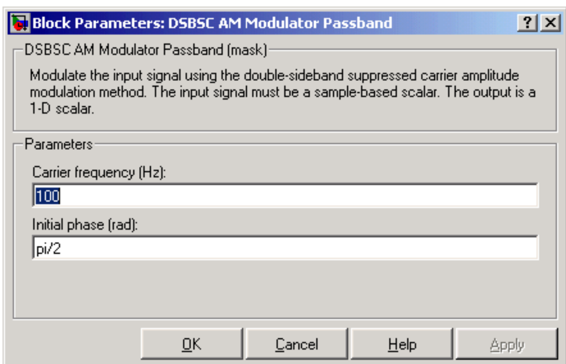
If the input is $u(t)$ as a function of time t, then the output is

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

where f_c is the **Carrier frequency** parameter and θ is the **Initial phase** parameter.

Typically, an appropriate **Carrier frequency** value is much higher than the highest frequency of the input signal.

Dialog Box



Carrier frequency (Hz)
The frequency of the carrier.

Initial phase (rad)
The initial phase of the carrier in radians.

Pair Block DSBSC AM Demodulator Passband

See Also DSB AM Modulator Passband, SSB AM Modulator Passband

Early-Late Gate Timing Recovery

Purpose

Recover symbol timing phase using early-late gate method

Library

Timing Phase Recovery sublibrary of Synchronization

Description

The Early-Late Gate Timing Recovery block recovers the symbol timing phase of the input signal using the early-late gate method. This block implements a non-data-aided feedback method.

Inputs

By default, the block has one input port. Typically, the input signal is the output of a receive filter that is matched to the transmitting pulse shape. For best results, the input signal power should be normalized. The input must be a scalar or a frame-based column vector. The input uses N samples to represent each symbol, where N > 1 is the **Samples per symbol** parameter. If the input is frame-based, then its vector length is N*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:

- 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.

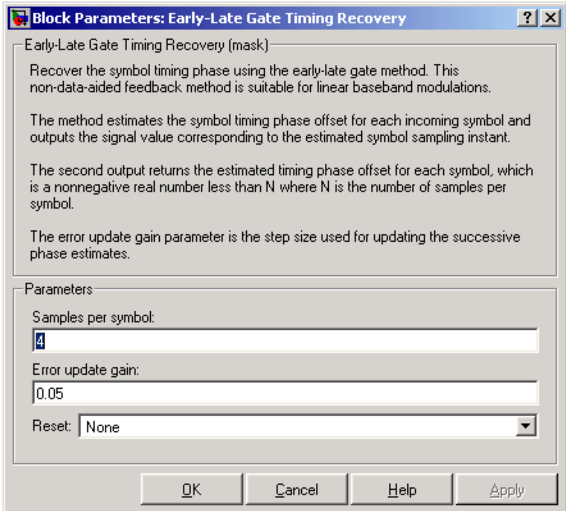
  The `Ph` output contains nonnegative real numbers less than N. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time or frame period of the `Ph` output is the same as that of the `Sym` output.

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

Delays

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

Dialog Box



Samples per symbol
: The number of samples, N, that represent each symbol in the input signal. This must be greater than 1.

Error update gain
: A positive real number representing the step size that the block uses for updating successive phase estimates. Typically, this number is less than 1/N, which corresponds to a slowly varying phase.

Reset
: Determines whether and under what circumstances the block restarts the phase estimation process. Choices are `None`, `Every frame`, and `On nonzero input via port`. The last option causes the block to have a second input port, labeled Rst.

Algorithm

This block uses a timing error detector whose result for the kth symbol is e(k), given by

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

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

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

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

Error Rate Calculation

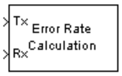
Purpose

Compute bit error rate or symbol error rate of input data

Library

Comm Sinks

Description



The Error Rate Calculation block compares input data from a transmitter with input data from a receiver. It calculates the error rate as a running statistic, by dividing the total number of unequal pairs of data elements by the total number of input data elements from one source.

You can use this block to compute either symbol or bit error rate, because it does not consider the magnitude of the difference between input data elements. If the inputs are bits, then the block computes the bit error rate. If the inputs are symbols, then it computes the symbol error rate.

This block inherits the sample time of its inputs.

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 data type `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, or `double`. If `Tx` is a scalar and `Rx` is a vector, or vice-versa, then the block compares the scalar with each element of the vector. (Overall, the block behaves as if you had preprocessed the scalar signal with the Signal Processing Blockset's Repeat block using the `Maintain input frame rate` option.)

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. (See the Zero Pad block in the Signal Processing Blockset.) 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, then you should not use the `Workspace` option; instead, use the `Port` option below and connect the output port to a Simulink To Workspace block.

- If you set the **Output data** parameter to `Port`, then an output port appears. This output port contains the *running* error statistics.

Delays

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

- The **Receive delay** parameter is the number of samples by which the received data lags behind the transmitted data. This parameter tells the block which samples "correspond" to each other and should be compared. The receive delay persists throughout the simulation.
- The **Computation delay** parameter tells the block to ignore the specified number of samples at the beginning of the comparison.

If you do not know the receive delay in your model, you can use the Align Signals block, which automatically compensates for the delay. If you use the Align Signals block, you should set the **Receive delay** in the Error Rate Calculation block to `0`.

Alternatively, you can use the Find Delay block to find the value of the delay, and then set the **Receive delay** parameter in the Error Rate Calculation block to that value.

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

If you use the `Select samples from mask` or `Select samples from port` option, then each delay parameter refers to the number of samples that the block receives, whether the block ultimately ignores some of them or not.

Stopping the Simulation Based on Error Statistics

You can configure this block so that its error statistics control the duration of simulation. This is useful for computing reliable steady-state error statistics without knowing in advance how long transient effects might last. To use this mode, check the **Stop simulation** check box. The block attempts to run the simulation until it detects **Target number of errors** errors. However, the simulation stops before detecting enough errors if the time reaches the model's **Stop time** setting (in the **Configuration Parameters** dialog box), if the Error Rate Calculation block makes **Maximum number of symbols** comparisons, or if another block in the model directs the simulation to stop.

To ignore either of the two stopping criteria in this block, set the corresponding parameter (**Target number of errors** or **Maximum number of symbols**) to `Inf`. For example, to reach a target number of errors without stopping the simulation early, set **Maximum number of symbols** to `Inf` and set the model's **Stop time** to `Inf`.

Examples

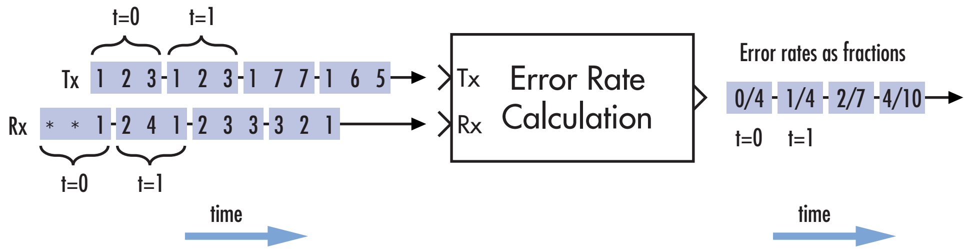
The figure below shows how the block compares pairs of elements and counts the number of error events. This example assumes that the sample time of each input signal is 1 second and that the block's parameters are as follows:

- **Receive delay** = `2`
- **Computation delay** = `0`
- **Computation mode** = `Entire frame`

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

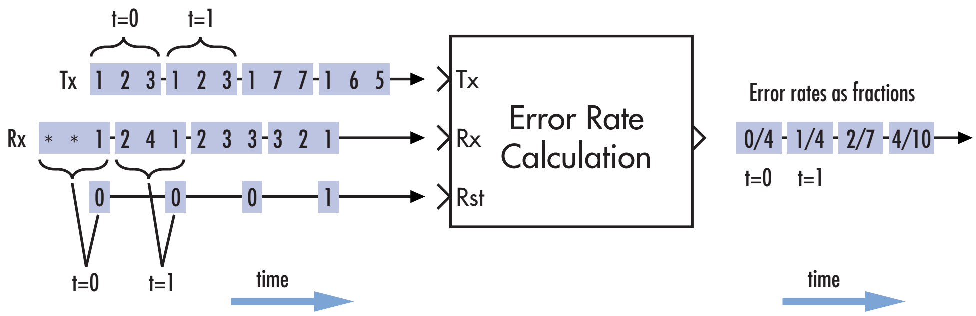
The values of the first two elements of `Rx` appear as asterisks because they do not influence the output. Similarly, the 6 and 5 in the `Tx` signal do not influence the output up to time 3, though they *would* influence the output at time 4.

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*.

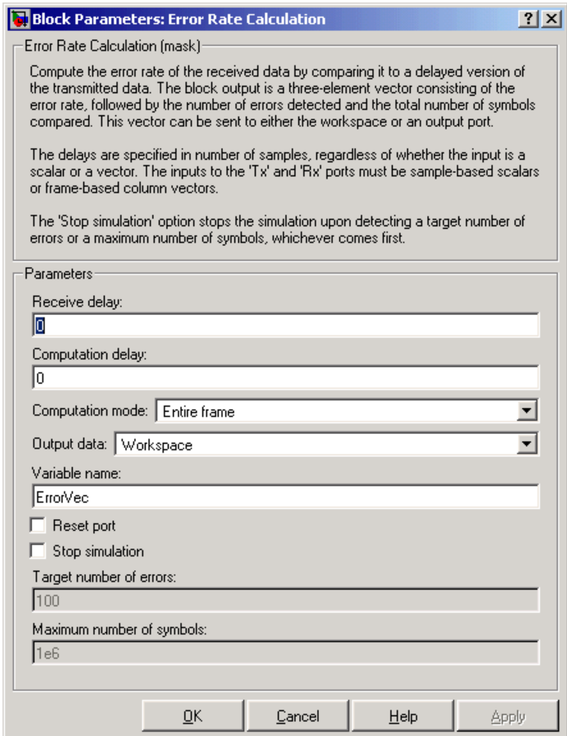


Note: Tx and Rx inputs are frame-based column vectors.

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.



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

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 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**.

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.

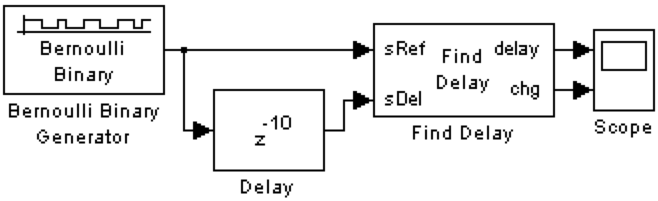
Another example of this usage is in "Computing Delays".

Finding the Delay to Help Align Words

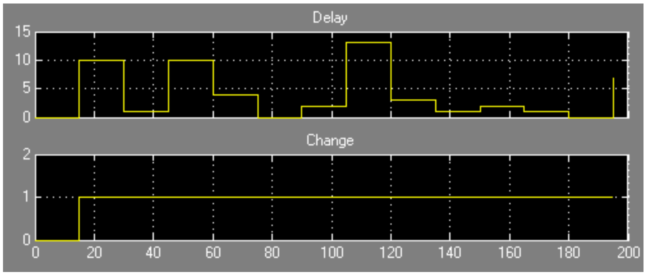
Another typical use of this block is to determine how to align the boundaries of frames with the boundaries of codewords or other types of data blocks. "Manipulating Delays" describes when such alignment is necessary and also illustrates, in the "Aligning Words of a Block Code" discussion, how to use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

Setting the Correlation Window Length

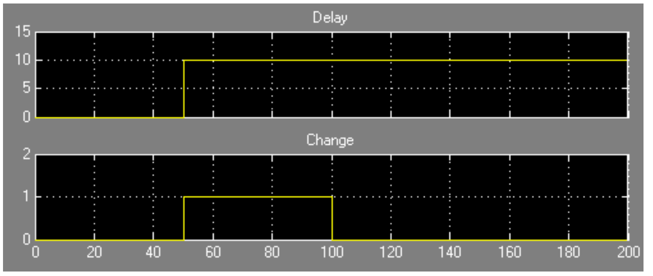
The next example illustrates how to tell when the **Correlation window length** is not sufficiently large.



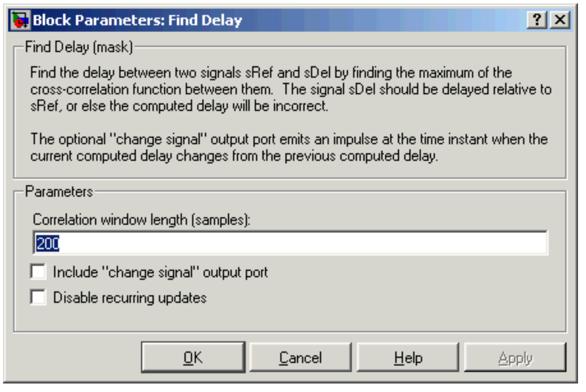
The model uses a Delay block to delay a signal by 10 samples, and uses the Find Delay block to compare the original signal with the delayed version. The model then displays the output of the Find Delay block in a scope. If the **Correlation window length** is 15, the scope shows that the calculated delay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

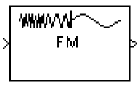
Purpose

Demodulate FM-modulated data

Library

Analog Passband Modulation, in Modulation

Description

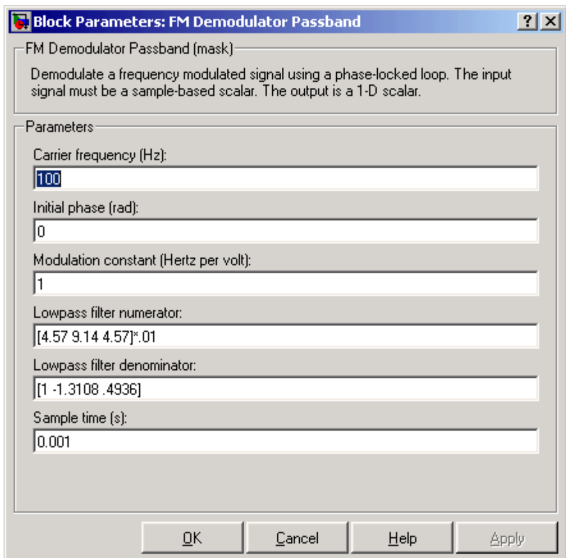


The FM Demodulator Passband block demodulates a signal that was modulated using frequency modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real sample-based scalar signals.

In the course of demodulating, the block uses a filter whose transfer function is described by the **Lowpass filter numerator** and **Lowpass filter denominator** parameters.

The block uses a voltage-controlled oscillator (VCO) in the demodulation. The **Initial phase** parameter gives the initial phase of the VCO.

Dialog Box



Carrier frequency (Hz)
: The carrier frequency in the corresponding FM Modulator Passband block.

Initial phase (rad)
The initial phase of the VCO in radians.

Modulation constant (Hertz per volt)
The modulation constant in the corresponding FM Modulator Passband block.

Lowpass filter numerator
The numerator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s.

Lowpass filter denominator
The denominator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s. For an FIR filter, set this parameter to `1`.

Sample time
The sample time in the corresponding FM Modulator Passband block.

Pair Block

FM Modulator Passband

FM Modulator Passband

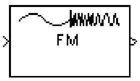
Purpose

Modulate using frequency modulation

Library

Analog Passband Modulation, in Modulation

Description



The FM Modulator Passband block modulates using frequency modulation. The output is a passband representation of the modulated signal. The output signal's frequency varies with the input signal's amplitude. Both the input and output signals are real sample-based scalar signals.

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

$$\cos\left(2\pi f_c t + 2\pi K_c \int_0^t u(\tau)d\tau + \theta\right)$$

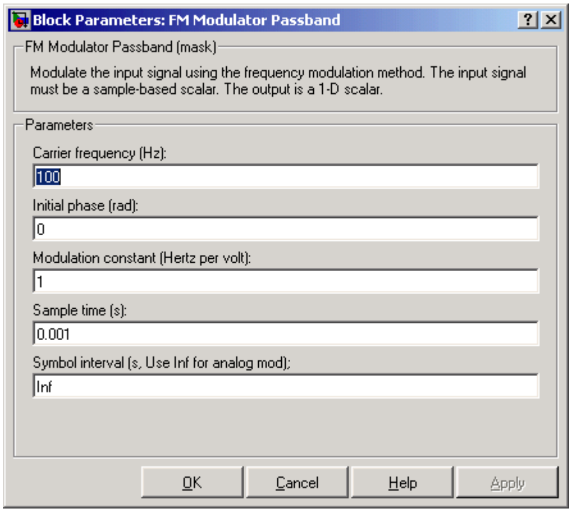
where:

- f_c is the **Carrier frequency** parameter.
- θ is the **Initial phase** parameter.
- K_c is the **Modulation constant** parameter.

Typically, an appropriate **Carrier frequency** value is much higher than the highest frequency of the input signal.

By the Nyquist sampling theorem, the reciprocal of the **Sample time** parameter must exceed twice the **Carrier frequency** parameter.

Dialog Box



Carrier frequency (Hz)
The frequency of the carrier.

Initial phase (rad)
The initial phase of the carrier in radians.

Modulation constant (Hertz per volt)
The modulation constant K_c.

Sample time
The sample time of the output signal. It must be a positive number.

Symbol interval
Typically set to `Inf`.

Pair Block

FM Demodulator Passband

Free Space Path Loss

Purpose

Reduce amplitude of input signal by amount specified

Library

RF Impairments

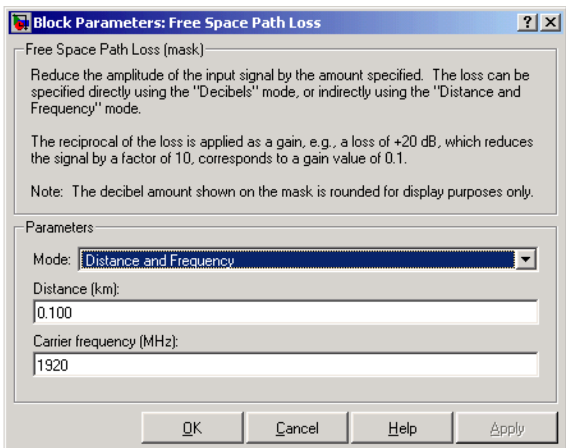
Description

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

- By the **Distance (km)** and **Carrier frequency (MHz)** parameters, if you specify `Distance and Frequency` in the **Mode** field
- By the **Loss (dB)** parameter, if you specify `Decibels` in the **Mode** field

The input to this block must be a complex signal.

Dialog Box



Mode

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

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

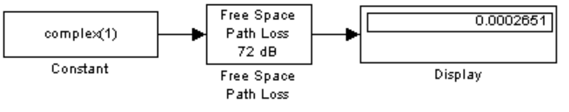
Distance
Distance between transmitter and receiver in kilometers. This parameter appears when you set **Mode** to `Distance and Frequency`.

Carrier frequency (MHz)
The carrier frequency in megahertz. This parameter appears when you set **Mode** to `Distance and Frequency`.

Examples

The model below illustrates the effect of the Free Space Path Loss block with the following parameter settings:

- **Mode** is set to `Distance and Frequency`.
- **Distance (km)** is set to `0.5`
- **Carrier frequency (MHz)** is set to `180`



See Also

Memoryless Nonlinearity

Gardner Timing Recovery

Purpose

Recover symbol timing phase using Gardner's method

Library

Timing Phase Recovery sublibrary of Synchronization

Description

The Gardner Timing Recovery block recovers the symbol timing phase of the input signal using Gardner's method. This block implements a non-data-aided feedback method that is independent of carrier phase recovery. The timing error detector that forms part of this block's algorithm requires at least two samples per symbol, one of which is the point at which the decision can be made.

Inputs

By default, the block has one input port. Typically, the input signal is the output of a receive filter that is matched to the transmitting pulse shape. For best results, the input signal power should be less than 1. The input must be a scalar or a frame-based column vector. The input uses N samples to represent each symbol, where N > 1 is the **Samples per symbol** parameter. If the input is frame-based, then its vector length is N*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.
  - If the input signal is a sample-based scalar with sample time T/N, then the `Sym` output is a sample-based scalar with sample time T.
- The `Ph` output gives the phase estimate for each symbol in the input.

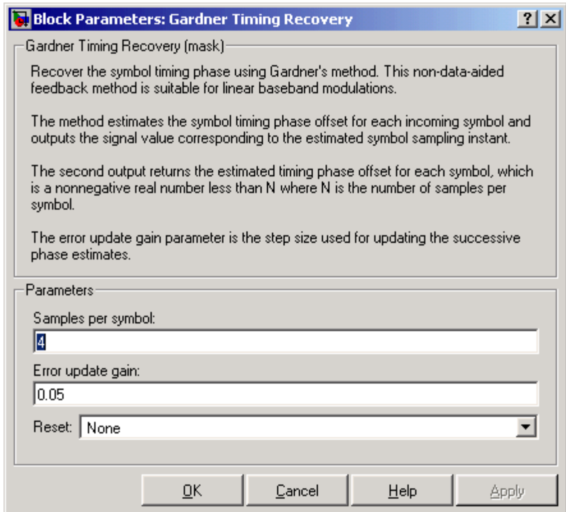
  The `Ph` output contains nonnegative real numbers less than N. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time or frame period of the `Ph` output is the same as that of the `Sym` output.

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

Delays

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

Dialog Box



Samples per symbol

The number of samples, N, that represent each symbol in the input signal. This must be greater than 1.

Error update gain

A positive real number representing the step size that the block uses for updating successive phase estimates. Typically, this number is less than 1/N, which corresponds to a slowly varying phase.

Reset

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are `None`, `Every frame`, and `On nonzero input via port`. The last option causes the block to have a second input port, labeled Rst.

Algorithm

This block uses a timing error detector whose result for the kth symbol is e(k), given by

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

where

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

Examples

The `gardner_vfracdelay` demonstration model uses this block.

References

[1] Gardner, F. M., "A BPSK/QPSK Timing-Error Detector for Sampled Receivers", *IEEE Transactions on Communications*, Vol. COM-34, No. 5, May 1986, pp. 423-429.

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

[3] Meyr, Heinrich, Marc Moeneclaey, and Stefan A. Fechtel, *Digital Communication Receivers*, Vol 2, New York, Wiley, 1998.

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

See Also

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

Purpose

Filter input signal, possibly downsampling, using Gaussian FIR filter

Library

Comm Filters

Description

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

Characteristics of the Filter

The impulse response of the Gaussian filter is

$$h(t) = \frac{\sqrt{\pi}}{\alpha}\exp\left(-\pi^2 t^2 / \alpha^2\right)$$

where

$$\alpha = \frac{\sqrt{\log 2}}{\sqrt{2}B}$$

and B is the filter's 3-dB bandwidth. The **BT product** parameter is B times the input signal's symbol period.

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.

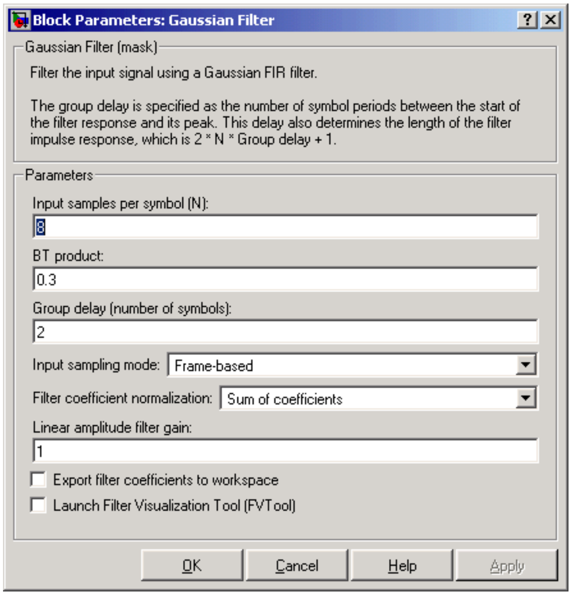
Input and Output Signals

The input signal must be a scalar or a frame-based column vector. Set the **Input sampling mode** parameter according to whether the input is sample-based or frame-based.

Exporting Filter Coefficients to the MATLAB Workspace

To examine or manipulate the coefficients of the filter that this block designs, select **Export filter coefficients to workspace**. Then set the **Coefficient variable name** parameter to the name of a variable that you want the block to create in the MATLAB workspace. Running the simulation causes the block to create the variable, overwriting any previous contents in case the variable already exists.

Dialog Box



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

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

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

Input sampling mode
The type of input signal: `Frame-based` or `Sample-based`.

Filter coefficient normalization
The block scales the set of filter coefficients so that this quantity equals 1. Choices are `Sum of coefficients`, `Filter energy`, and `Peak amplitude`.

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.

Launch Filter Visualization Tool
If you check this box, then MATLAB launches the Filter Visualization Tool (`fvtool`) to analyze the Gaussian filter whenever you apply any changes to the block's parameters.

See Also

Raised Cosine Receive Filter, `firgauss`

References

[1] Rappaport, Theodore S., *Wireless Communications: Principles and Practice*, Upper Saddle River, N.J., Prentice Hall, 1996.

Purpose

Generate Gaussian distributed noise with given mean and variance values

Library

Noise Generators sublibrary of Comm Sources

Description

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

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

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

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

The probability density function of n-dimensional Gaussian noise is

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

where x is a length-n vector, K is the n-by-n covariance matrix, μ 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 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'srandseed function. At the MATLAB prompt, enter

```
randseed
```

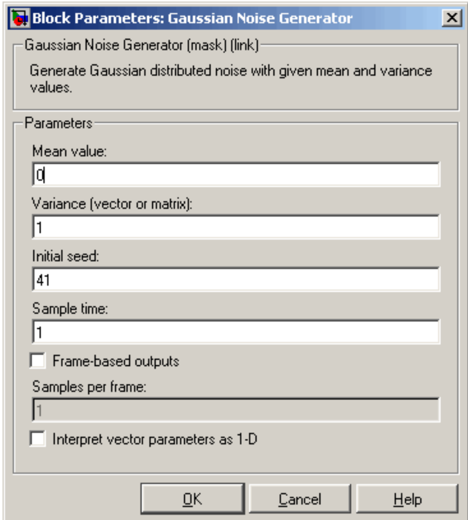
This returns a random prime number greater than 30. Entering `randseed` again produces a different prime number. If you supply an integer argument, `randseed` always returns the same prime for that integer. For example, `randseed(5)` always returns the same answer.

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 Using the Communications Blockset 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.

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.

See Also

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

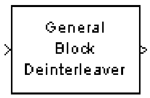
Purpose

Restore ordering of symbols in input vector

Library

Block sublibrary of Interleaving

Description



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

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

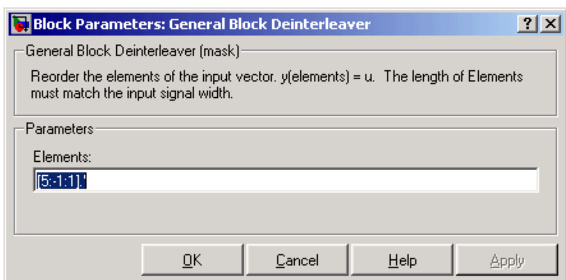
The **Elements** parameter must contain unique integers between 1 and N.

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 theGeneral 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



Elements

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

Examples

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

Pair Block

General Block Interleaver

See Also

`perms` (MATLAB function)

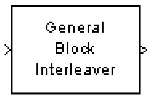
Purpose

Reorder symbols in input vector

Library

Block sublibrary of Interleaving

Description



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

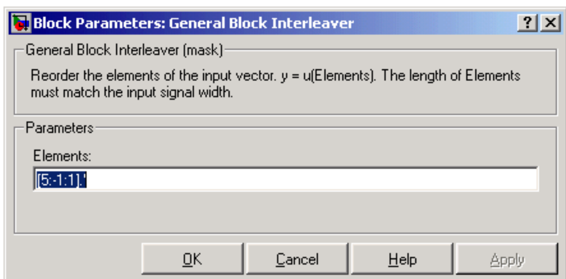
Output(k) = Input(**Elements**(k))

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.

Examples

If **Elements** is `[4,1,3,2]` and the input vector is `[40;32;59;1]`, then the output vector is `[1;40;59;32]`. Notice that all of these vectors have the same length and that the vector **Elements** is a permutation of the vector `[1:4]`.

Pair Block

General Block Deinterleaver

See Also

`perms` (MATLAB function)

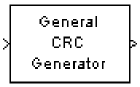
Purpose

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

Library

CRC sublibrary of Error Correction and Detection

Description



The General CRC Generator block generates cyclic redundancy code (CRC) bits for each input data frame and appends them to the frame. You specify the generator polynomial for the CRC algorithm using the **Generator polynomial** parameter. This block is general in the sense that the degree of the polynomial does not need to be a power of two. You represent the polynomial in one of these ways:

- As a binary row vector containing the coefficients in descending order of powers. For example, `[1 1 0 1]` represents the polynomial $x^3 + x^2 + 1$.
- As an integer row vector containing the powers of nonzero terms in the polynomial, in descending order. For example, `[3 2 0]` represents the polynomial $x^3 + x^2 + 1$.

You specify the initial state of the internal shift register by the **Initial states** parameter. The **Initial states** parameter is either a scalar or a binary row vector of length equal to the degree of the generator polynomial. A scalar value is expanded to a row vector of length equal to the degree of the generator polynomial. For example, the default initial state of `[0]` is expanded to a row vector of all zeros.

You specify the number of checksums that the block calculates for each input frame by the **Checksums per frame** parameter. The **Checksums per frame** value must evenly divide the size of the input frame. If the value of **Checksums per frame** is k, the block does the following:

1. Divides each input frame into k subframes of equal size
2. Prefixes the **Initial states** vector to each of the k subframes
3. Applies the CRC algorithm to each augmented subframe

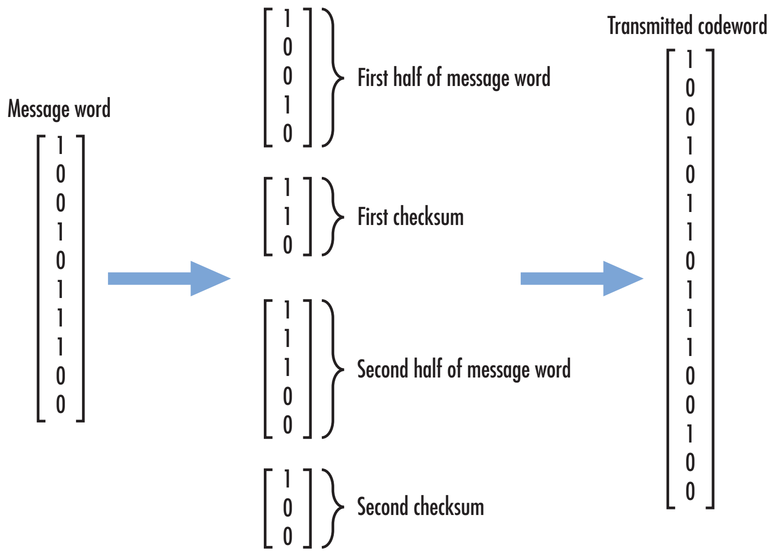
4 Appends the resulting checksums at the end of each subframe

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.

Example

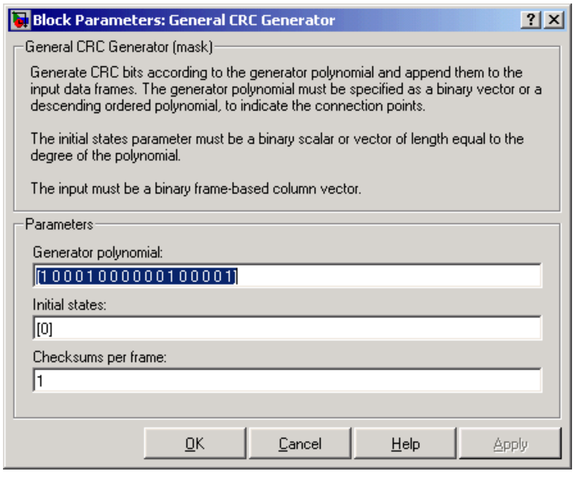
Suppose the size of the input frame is 10, the degree of the generator polynomial is 3, **Initial states** is `[0]`, and **Checksums per frame** is `2`. The block divides each input frame into two subframes of size 5 and appends a checksum of size 3 to each subframe, as shown below. The initial states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



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 Using the Communications Blockset.

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

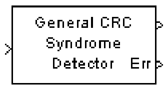
Purpose

Detect errors in input data frames according to generator polynomial

Library

CRC sublibrary of Error Correction and Detection

Description



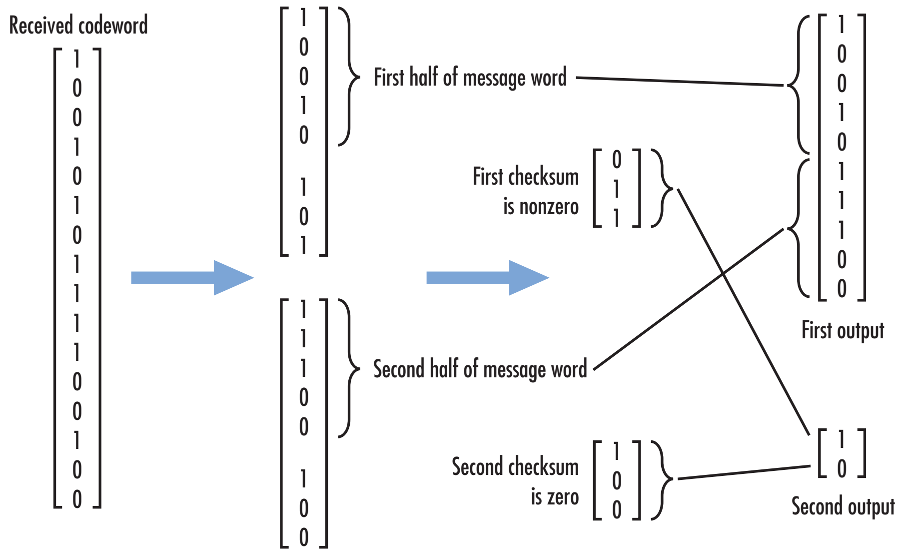
The General CRC Syndrome Detector block computes checksums for its entire input frame. The block's second output is a vector whose size is the number of checksums, and whose entries are 0 if the checksum computation yields a zero value, and 1 otherwise. The block's first output is the set of message words with the checksums removed.

The block's parameter settings should agree with those in the General CRC Generator block.

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

Example

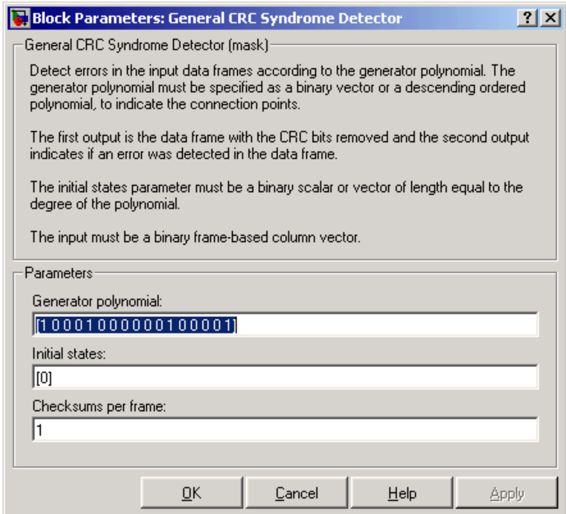
Suppose the received codeword has size 16, the generator polynomial has degree 3, **Initial states** is `[0]`, and **Checksums per frame** is `2`. The block computes the two checksums of size 3, one from the first half of the received codeword, and the other from the second half of the received codeword, 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.



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 Using the Communications Blockset.

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 Generator

See Also

CRC-N Generator, CRC-N Syndrome Detector

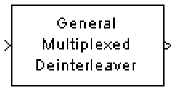
Purpose

Restore ordering of symbols using specified-delay shift registers

Library

Convolutional sublibrary of Interleaving

Description



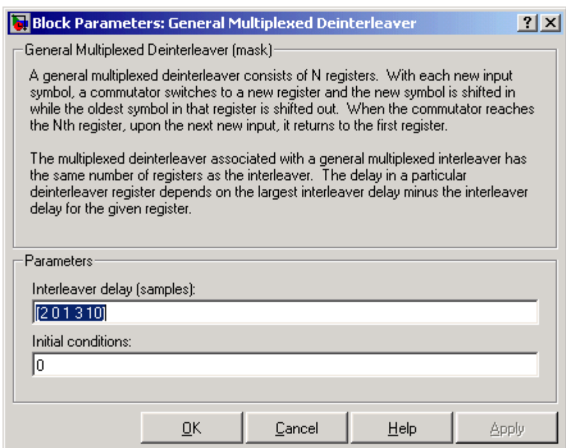
The General Multiplexed Deinterleaver block restores the original ordering of a sequence that was interleaved using theGeneral Multiplexed Interleaver block.

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



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

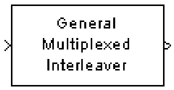
References

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

Purpose Permute input symbols using set of shift registers with specified delays

Library Convolutional sublibrary of Interleaving

Description



The General Multiplexed Interleaver block permutes the symbols in the input signal. Internally, it uses a set of shift registers, each with its own delay value.

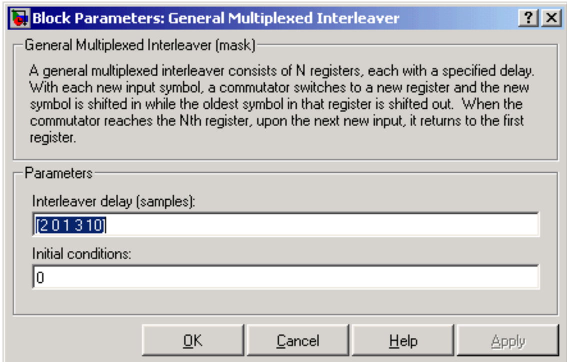
The input can be either a scalar or a frame-based column vector. It can be real or complex. The input and output signals share the same sample time.

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

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

The **Initial conditions** parameter indicates the values that fill each shift register at the beginning of the simulation. If **Initial conditions** is a scalar, then its value fills all shift registers; if **Initial conditions** is a column vector, then each entry fills the corresponding shift register. (In sample-based mode, **Initial conditions** can also be a row vector.) If a given shift register has zero delay, then the value of the corresponding entry in the **Initial conditions** vector is unimportant.

Dialog Box



Interleaver delay (samples)

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

Initial conditions

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

Pair Block

General Multiplexed Deinterleaver

See Also

Convolutional Interleaver, Helical Interleaver

References

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

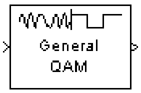
Purpose

Demodulate QAM-modulated data

Library

AM, in Digital Baseband sublibrary of Modulation

Description

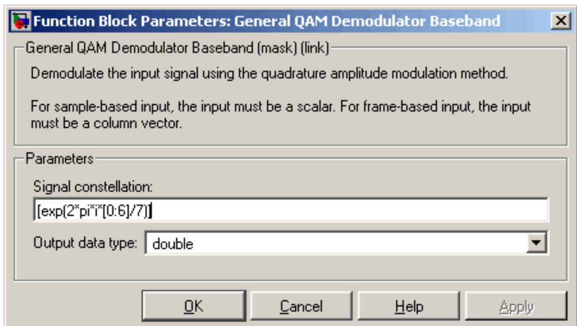


The General QAM Demodulator Baseband block demodulates a signal that was modulated using quadrature amplitude modulation. The input is a baseband representation of the modulated signal.

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

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

Dialog Box



Signal constellation
: A real or complex vector that lists the constellation points.

Output data type
: For integer inputs, this block can output the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. For bit inputs, output can be `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, or `double`.

Pair Block

General QAM Modulator Baseband

See Also

Rectangular QAM Demodulator Baseband

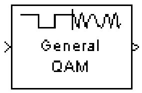
Purpose

Modulate using quadrature amplitude modulation

Library

AM, in Digital Baseband sublibrary of Modulation

Description

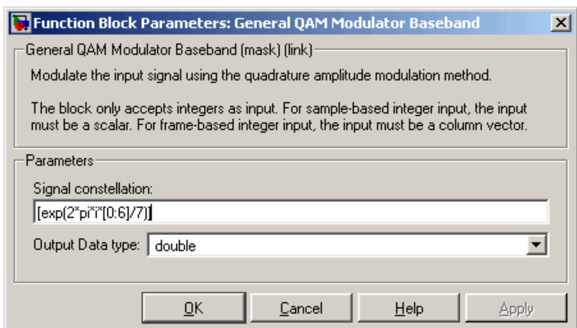


The General QAM Modulator Baseband block modulates using quadrature amplitude modulation. The output is a baseband representation of the modulated signal.

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

Dialog Box



Signal constellation
: A real or complex vector that lists the constellation points.

Output Data type
: The output data type can be either `single` or `double`.

Pair Block

General QAM Demodulator Baseband

See Also

Rectangular QAM Modulator Baseband

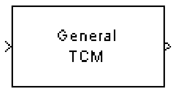
Purpose

Decode trellis-coded modulation data, mapped using arbitrary constellation

Library

Trellis-Coded Modulation

Description



The General TCM Decoder block uses the Viterbi algorithm to decode a trellis-coded modulation (TCM) signal that was previously modulated using an arbitrary signal constellation.

The **Trellis structure** and **Signal constellation** parameters in this block should match those in theGeneral 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.

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.

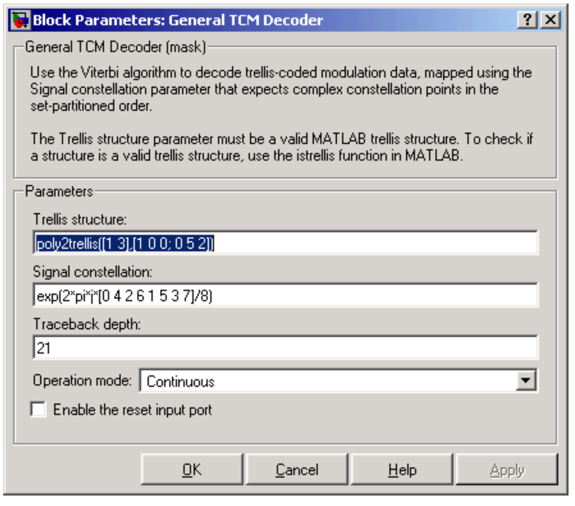
- In `Truncated` mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In `Terminated` mode, the block treats each frame independently. The traceback path always starts at the all-zeros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

Pair Block

General TCM Encoder

See Also

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

References

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

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

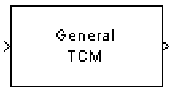
Purpose

Convolutionally encode binary data and map using arbitrary constellation

Library

Trellis-Coded Modulation

Description



The General TCM Encoder block implements trellis-coded modulation (TCM) by convolutionally encoding the binary input signal and mapping the result to an arbitrary signal constellation. The points in the signal constellation are listed in set-partitioned order in the **Signal constellation** parameter. This parameter is a complex vector whose length, M, equals the number of possible output symbols from the convolutional encoder. (That is, $\log_2 M$ is equal to n for a rate k/n convolutional code.)

Input and Output Signals

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

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

Specifying the Encoder

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

- If you want to specify the encoder using its constraint length, generator polynomials, and possibly feedback connection polynomials, then use a `poly2trellis` command within the **Trellis structure** field. For example, to use an encoder with a constraint length of 7, code generator polynomials of 171 and 133 (in octal numbers), and a feedback connection of 171 (in octal), set the **Trellis structure** parameter to

  ```
  poly2trellis(7,[171 133],171)
  ```

- If you have a variable in the MATLAB workspace that contains the trellis structure, then enter its name as the **Trellis structure** parameter. This way is faster because it causes Simulink to spend less time updating the diagram at the beginning of each simulation, compared to the usage in the previous bulleted item.

Signal Constellations

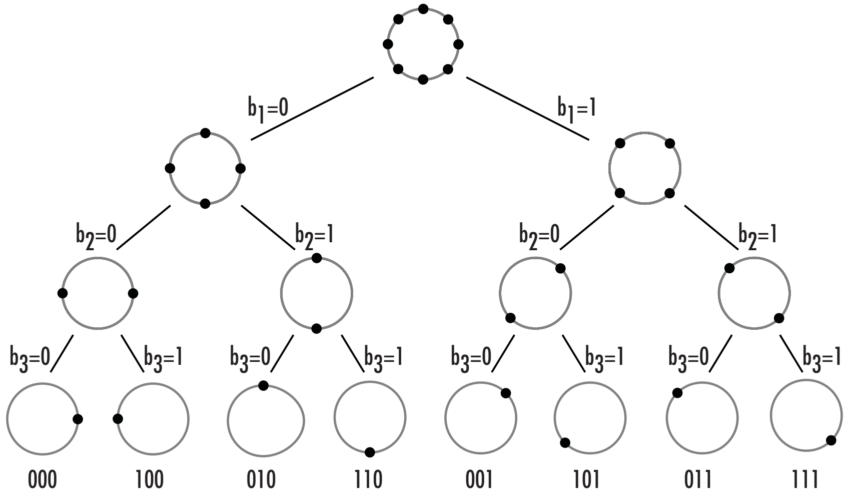
The trellis-coded modulation technique partitions the constellation into subsets called cosets so as to maximize the minimum distance between pairs of points in each coset.

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

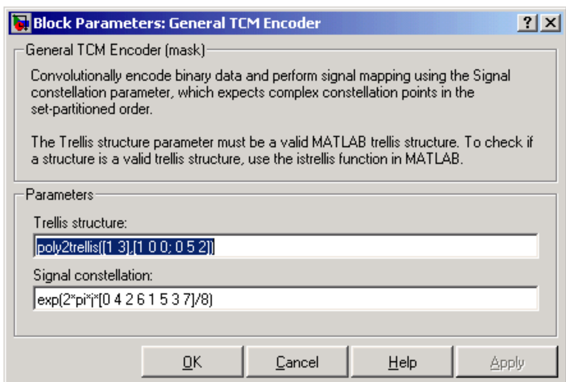
```
exp(2*pi*j*[0 4 2 6 1 5 3 7]/8)
```

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 theM-PSK TCM Encoder andRectangular QAM TCM Encoder blocks.

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.

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.

GMSK Demodulator Baseband

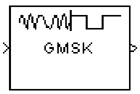
Purpose

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 theGMSK Modulator Baseband block.

Traceback Length and Output Delays

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The **Traceback length** parameter, D, in this block is the number of trellis branches used to construct each traceback path. D influences the output delay, which is the number of zero symbols that precede the first meaningful demodulated value in the output.

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

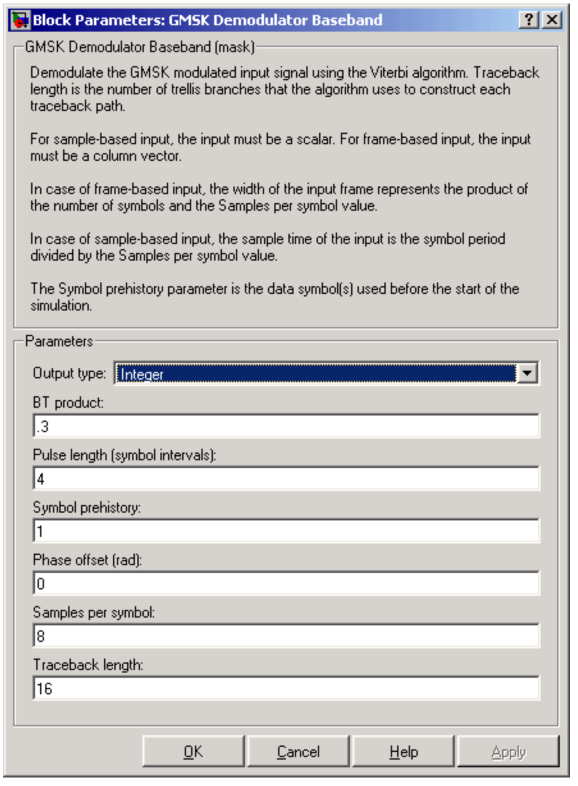
Inputs and Outputs

The input can be either a scalar or a frame-based column vector. 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 Using the Communications Blockset.

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.

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

Phase offset (rad)
The initial phase of the modulated waveform.

Samples per symbol
The number of input samples that represent each modulated symbol.

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

Pair Block

GMSK Modulator Baseband

See Also

CPM Demodulator Baseband, Viterbi Decoder

References

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

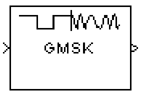
Purpose

Modulate using Gaussian minimum shift keying method

Library

CPM, in Digital Baseband sublibrary of Modulation

Description



The GMSK Modulator Baseband block modulates using the Gaussian minimum shift keying method. The output is a baseband representation of the modulated signal.

The **BT product** parameter represents bandwidth multiplied by time. This parameter is a nonnegative scalar. It is used to reduce the bandwidth at the expense of increased intersymbol interference. The **Pulse length** parameter measures the length of the Gaussian pulse shape, in symbol intervals. For an explanation of the pulse shape, see the work by Anderson, Aulin, and Sundberg among the references listed below. The frequency pulse shape is defined by the following equations.

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

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

The **Symbol prehistory** parameter is a scalar or vector that specifies the data symbols used before the start of the simulation, in reverse chronological order. If it is a vector, then its length must be one less than the **Pulse length** parameter.

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

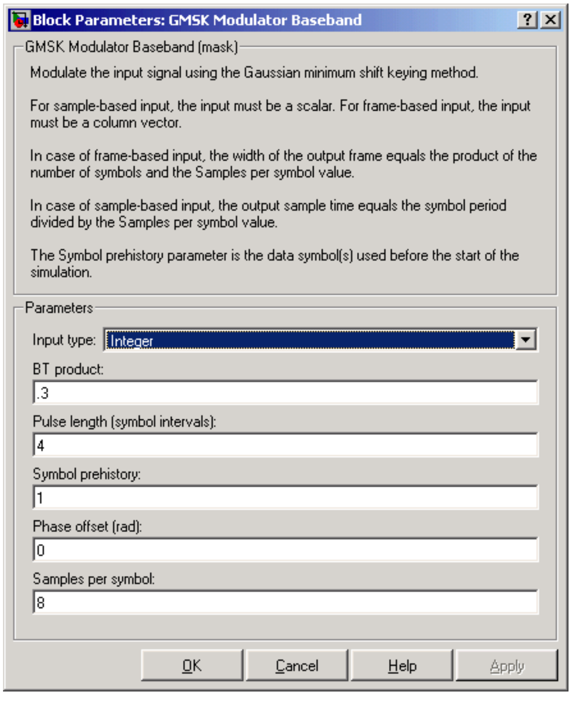
Input Attributes

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

Dialog Box



Input type
Indicates whether the input consists of bipolar or binary values.

BT product
The product of bandwidth and time.

Pulse length (symbol intervals)
The length of the frequency pulse shape.

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

Phase offset (rad)
The initial phase of the output waveform.

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

Pair Block

GMSK Demodulator Baseband

See Also

CPM Modulator Baseband

References

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

Gold Sequence Generator

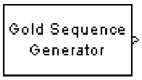
Purpose

Generate Gold sequence from set of sequences

Library

Sequence Generators sublibrary of Comm Sources

Description



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

The Gold sequences are defined using a specified pair of sequences *u* and *v*, of period $N = 2^n - 1$, called a *preferred pair*, as defined in "Preferred Pairs of Sequences" on page 2-239 below. The set $G(u, v)$ of Gold sequences is defined by

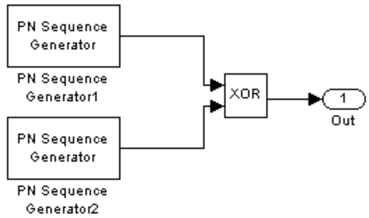
$$G(u,v) = \{u, v, u \oplus v, u \oplus Tv, u \oplus T^2 v, \ldots, u \oplus T^{N-1} v\}$$

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	$u \oplus T^2 v$
...	...
2^n-2	$u \oplus T^{2^n-2} v$

You can shift the starting point of the Gold sequence with the **Shift** parameter, which is an integer representing the length of the shift.

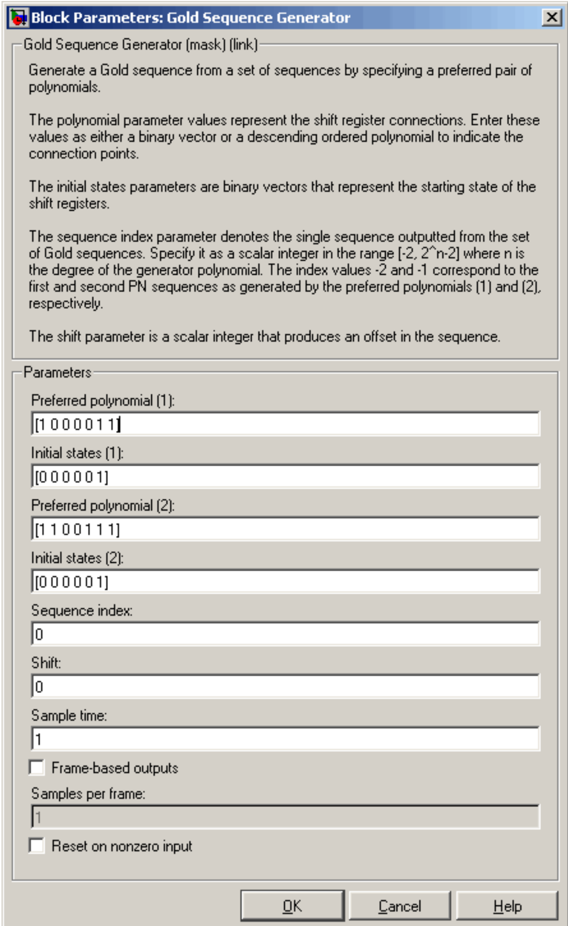
You can use an external signal to reset the values of the internal shift register to the initial state by selecting the **Reset on nonzero input** check box. This creates an input port for the external signal in the Gold Sequence Generator block. The way the block resets the internal shift register depends on whether its output signal and the reset signal are sample-based or frame-based. The following example demonstrates the possible alternatives. See "Example: Resetting a Signal" on page 2-431 for an example.

Preferred Pairs of Sequences

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

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

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

See Also

Kasami Sequence Generator, PN Sequence Generator

References

[1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.

[2] Gold, R., "Maximal Recursive Sequences with 3-valued Recursive Cross-Correlation Functions," *IEEE Trans. Infor. Theory*, Jan., 1968, pp. 154-156.

[3] Gold, R., "Optimal Binary Sequences for Spread Spectrum Multiplexing, *IEEE Trans. Infor. Theory*, Oct., 1967, pp. 619-621.

[4] Sarwate, D.V., and M.B. Pursley, "Crosscorrelation Properties of Pseudorandom and Related Sequences," *Proc. IEEE*, Vol. 68, No. 5, May, 1980, pp. 583-619.

Purpose

Generate Hadamard code from orthogonal set of codes

Library

Sequence Generators sublibrary of Comm Sources

Description

The Hadamard Code Generator block generates a Hadamard code from a Hadamard matrix, whose rows form an orthogonal set of codes. Orthogonal codes can be used for spreading in communication systems in which the receiver is perfectly synchronized with the transmitter. In these systems, the despreading operation is ideal, as the codes are decorrelated completely.

The Hadamard codes are the individual rows of a Hadamard matrix. Hadamard matrices are square matrices whose entries are +1 or -1, and whose rows and columns are mutually orthogonal. If N is a nonnegative power of 2, the N-by-N Hadamard matrix, denoted H_N, is defined recursively as follows.

$$H_1 = [1]$$

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

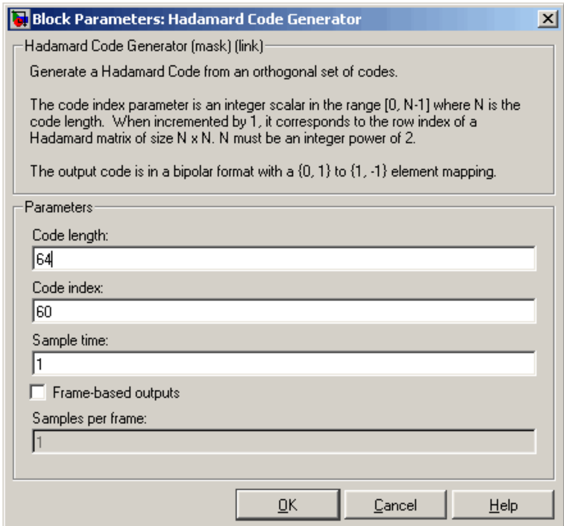
The N-by-N Hadamard matrix has the property that

$$H_N H_N^T = NI_N$$

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

The Hadamard Code Generator block outputs a row of H_N. The output is bipolar. You specify the length of the code, N, by the **Code length** parameter. The **Code length** must be a power of 2. You specify the index of the row of the Hadamard matrix, which is an integer in the range [0, 1, ... , 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.

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.

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 correspondingHamming Encoder block.

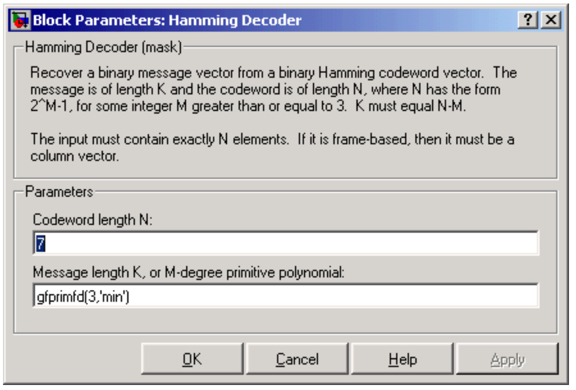
If the Hamming code has message length K and codeword length N, then N must have the form 2^M-1 for some integer M greater than or equal to 3. Also, K must equal N-M.

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

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

- To use the default primitive polynomial, simply enter N and K as the first and second dialog parameters, respectively. The algorithm uses `gfprimdf(M)` as the primitive polynomial for GF(2^M).
- To specify the primitive polynomial, enter N as the first parameter and a binary vector as the second parameter. The vector represents the primitive polynomial by listing its coefficients in order of ascending exponents. You can create primitive polynomials using the `gfprimfd` function in the Communications Toolbox.

Dialog Box



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

Message length K, or M-degree primitive polynomial
Either the message length, which is also the output vector length; or a binary vector that represents a primitive polynomial for GF(2^M).

Pair Block

Hamming Encoder

See Also

`hammgen` (Communications Toolbox)

Hamming Encoder

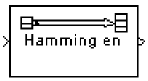
Purpose

Create Hamming code from binary vector data

Library

Block sublibrary of Channel Coding

Description



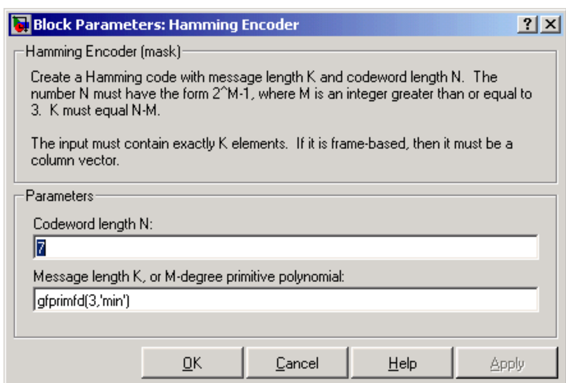
The Hamming Encoder block creates a Hamming code with message length K and codeword length N. The number N must have the form 2^M-1, where M is an integer greater than or equal to 3. Then K equals N-M.

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

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

- To use the default primitive polynomial, simply enter N and K as the first and second dialog parameters, respectively. The algorithm uses `gfprimdf(M)` as the primitive polynomial for GF(2^M).
- To specify the primitive polynomial, enter N as the first parameter and a binary vector as the second parameter. The vector represents the primitive polynomial by listing its coefficients in order of ascending exponents. You can create primitive polynomials using the `gfprimfd` function in the Communications Toolbox.

Dialog Box



Codeword length N

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

Message length K, or M-degree primitive polynomial

Either the message length, which is also the input vector length; or a binary vector that represents a primitive polynomial for GF(2^M).

Pair Block

Hamming Decoder

See Also

`hammgen` (Communications Toolbox)

Helical Deinterleaver

Purpose

Restore ordering of symbols permuted by helical interleaver

Library

Convolutional sublibrary of Interleaving

Description

The Helical Deinterleaver block permutes the symbols in the input signal by placing them in an array row by row and then selecting groups in a helical fashion to send to the output port.

The block uses the array internally for its computations. If C is the **Number of columns in helical array** parameter, then the array has C columns and unlimited rows. If N is the **Group size** parameter, then the block accepts an input of length C*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

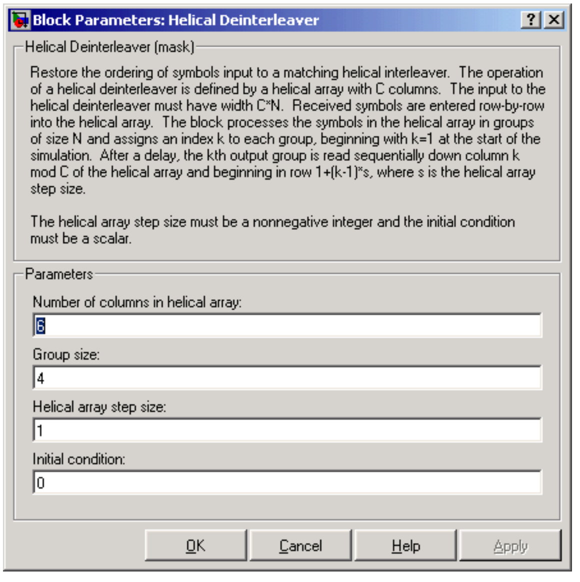
After processing a message with the Helical Interleaver block and the Helical Deinterleaver block, the deinterleaved data lags the original message by

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

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 Delay block in the Signal Processing Blockset to adjust delays manually, if necessary.

Dialog Box



Number of columns in helical array

The number of columns, C, in the helical array.

Group size

The size, N, of each group of symbols. The input width is C times N.

Helical array step size

The number of rows of separation between consecutive output groups as the block selects them from their respective columns of the helical array.

Initial condition

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

Pair Block

Helical Interleaver

See Also

General Multiplexed Deinterleaver

References

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

Purpose

Permute input symbols using helical array

Library

Convolutional sublibrary of Interleaving

Description

The Helical Interleaver block permutes the symbols in the input signal by placing them in an array in a helical fashion and then sending rows of the array to the output port.

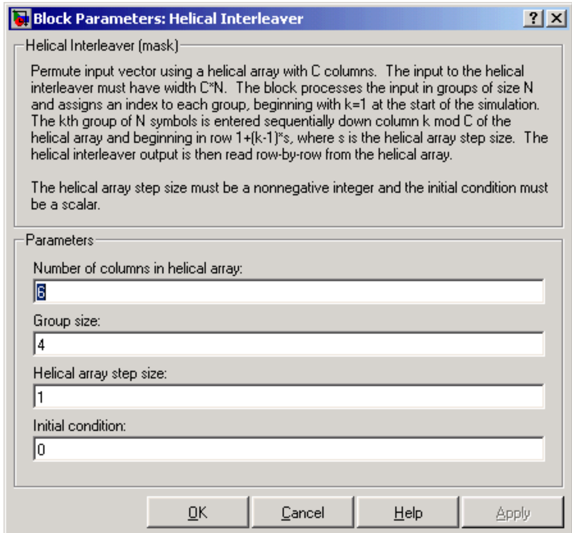
The block uses the array internally for its computations. If C is the **Number of columns in helical array** parameter, then the array has C columns and unlimited rows. If N is the **Group size** parameter, then the block accepts an input of length C*N at each time step and partitions the input into consecutive groups of N symbols. Counting from the beginning of the simulation, the block places the 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.

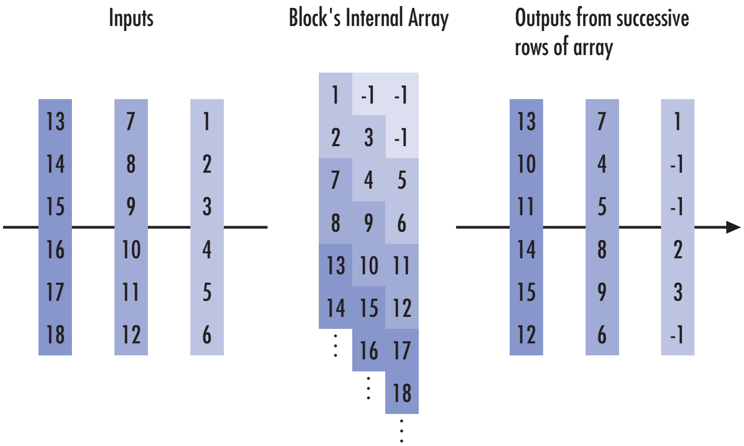
Group size
: The size, N, of each group of input symbols. The input width is C times N.

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.

Ideal Rectangular Pulse Filter

Purpose

Shape input signal using ideal rectangular pulses

Library

Comm Filters

Description

The Ideal Rectangular Pulse Filter block upsamples and shapes the input signal using rectangular pulses. The block replicates each input sample N times, where N is the **Pulse length** parameter. After replicating input samples, the block can also normalize the output signal and/or apply a linear amplitude gain.

If the **Pulse delay** parameter is nonzero, then the block outputs that number of zeros at the beginning of the simulation, before starting to replicate any of the input values.

Inputs and Outputs

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

- 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

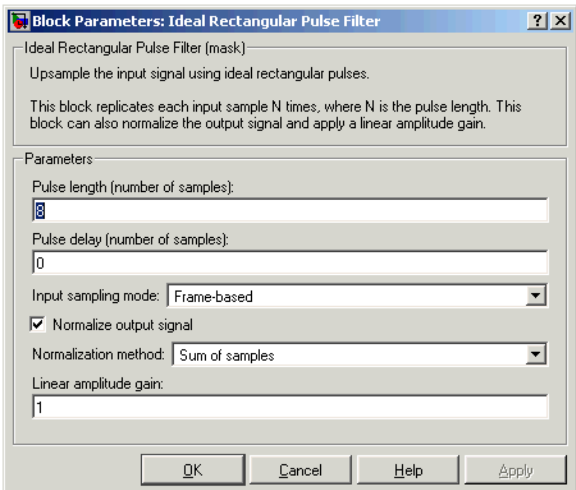
You determine the block's normalization behavior using the **Normalize output signal** and **Linear amplitude gain** parameters.

- 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.

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: `Frame-based` or `Sample-based`.

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

Normalization method
The quantity that the block considers when scaling the set of replicated values. Choices are `Sum of samples` and `Energy per pulse`. This field appears only if you select **Normalize output signal**.

Linear amplitude gain
A positive scalar used to scale the output signal.

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, 30,...

Normalization Method, If Any	Linear Amplitude Gain	First Several Output Values
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, 7.5, 7.5, 7.5, 7.5,...
Energy per pulse	1	0, 0, 0, 0.5, 0.5, 0.5, 0.5, 1.0, 1.0, 1.0, 1.0, 1.5, 1.5, 1.5, 1.5,..., where (0.5)^2*4=1^2
Energy per pulse	10	0, 0, 0, 5, 5, 5, 5, 10, 10, 10, 10, 15, 15, 15, 15,...

See Also

Upsample, Integrate and Dump

Insert Zero

Purpose Distribute input elements in output vector

Library Sequence Operations

Description

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

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

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

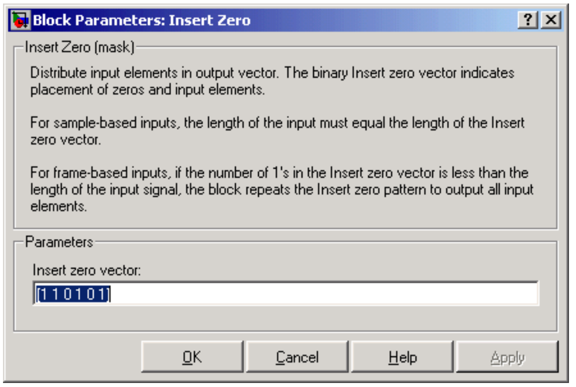
The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, `double`, and fixed-point. The data type of this output will be the same as that of the input signal.

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

Frame-Based Processing

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

Dialog Box



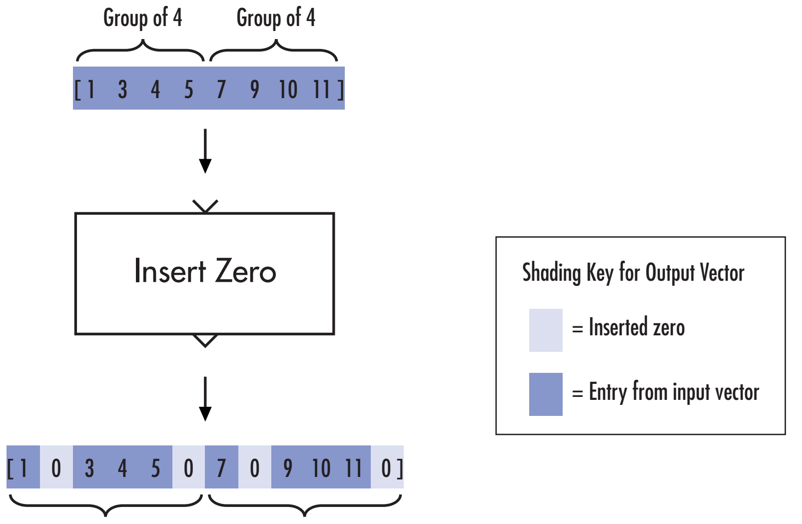
Insert zero vector

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

Examples

If the **Insert zero vector** parameter is the six-element vector `[1,0,1,1,1,0]`, then the block inserts zeros after the first and last elements of each consecutive grouping of four input elements. It considers groups of four elements because the **Insert zero vector** parameter has four 1s.

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



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

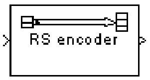
See Also

Puncture

Purpose Create Reed-Solomon code from integer vector data

Library Block sublibrary of Channel Coding

Description



The Integer-Input RS Encoder block creates a Reed-Solomon code with message length K and codeword length N. You specify both N and K directly in the block dialog. The symbols for the code are integers between 0 and 2^M-1, which represent elements of the finite field GF(2^M). Restrictions on M and N are described in "Restrictions on M and the Codeword Length N" on page 2-264 below. The difference N - K must be an even integer.

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

The default value of M is the smallest integer that is greater than or equal to log2(N+1), that is, `ceil(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 2-263 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 = ceil(log2(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 $GF(2^M)$ 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

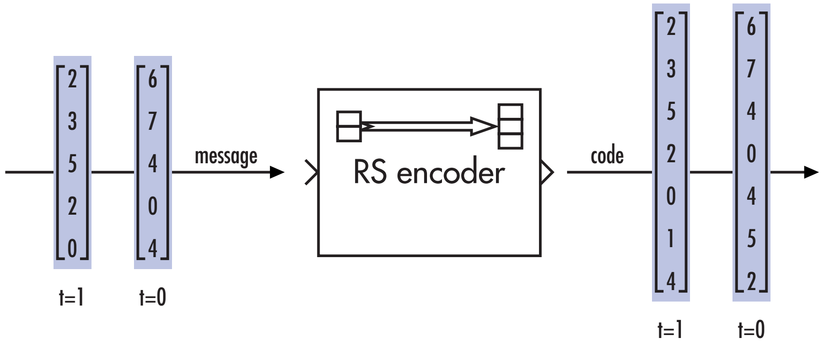
$$g(x) = (x+A^b)(x+A^{b+1})(x+A^{b+2})...(x+A^{b+N-K-1})$$

where A is the primitive element of the Galois field over which the input message is defined, and *b* is an integer.

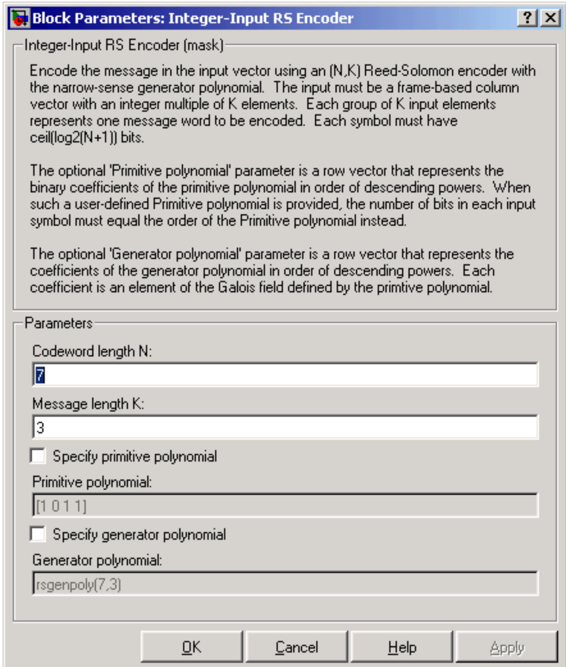
If you do not select **Specify generator polynomial**, the block uses the default generator polynomial, corresponding to *b*=1, for Reed-Solomon encoding. You can display the default generator polynomial by entering `rsgenpoly(N1,K1)`, where `N1 = 2^M-1` and `K1 = K+(N1-N)`, at the MATLAB prompt, if you are using the default primitive polynomial. If the **Specify primitive polynomial** box is selected, and you specify the primitive polynomial specified as `poly`, the default generator polynomial is `rsgenpoly(N1,K1,poly)`.

Examples

Suppose M = 3, N = 2^3-1 = 7, and K = 5. Then a message is a vector of length 5 whose entries are integers between 0 and 7. A corresponding codeword is a vector of length 7 whose entries are integers between 0 and 7. The following figure illustrates possible input and output signals to this block when **Codeword length N** is set to `7`, **Message length K** is set to `5`, and the default primitive and generator polynomials are used.



Dialog Box



Codeword length N
: The codeword length.

Message length K
: The message length.

Specify primitive polynomial
: When you select this box, you can specify the primitive polynomial as a binary row vector.

Primitive polynomial
: Binary row vector representing the primitive polynomial in descending order of powers.

Specify generator polynomial
: When you select this box, you can specify the generator polynomial as an integer row vector.

Generator polynomial

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

Pair Block

Integer-Output RS Decoder

See Also

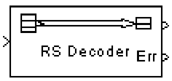
Binary-Input RS Encoder

Integer-Output RS Decoder

Purpose Decode Reed-Solomon code to recover integer vector data

Library Block sublibrary of Channel Coding

Description The Integer-Output RS Decoder block recovers a message vector from a Reed-Solomon codeword vector. For proper decoding, the parameter values in this block should match those in the correspondingInteger-Input RS Encoder block.



The Reed-Solomon code has message length K and codeword length N. You specify both N and K directly in the block dialog. The symbols for the code are integers between 0 and 2^M-1, which represent elements of the finite field GF(2^M). Restrictions on M and N are described in “Restrictions on M and the Codeword Length N” on page 2-264 below. The difference N - K must be an even integer.

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

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 2-263 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 2-264.

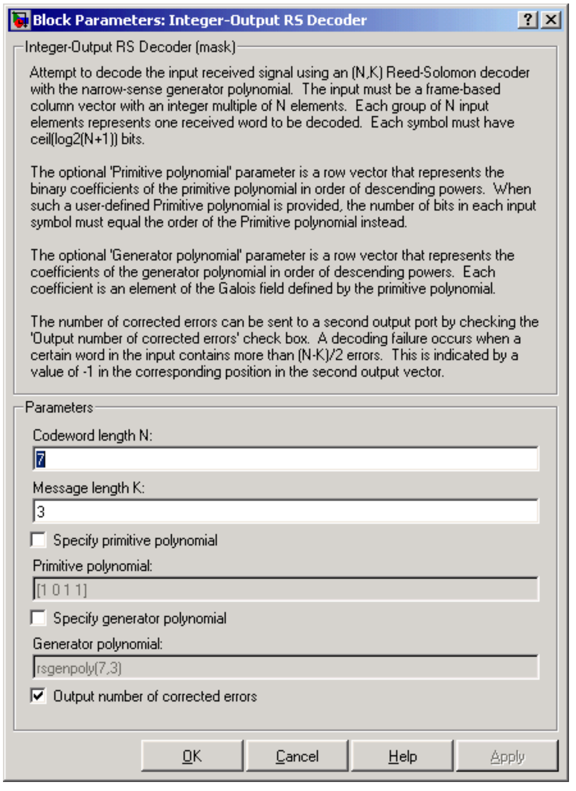
An (N, K) Reed-Solomon code can correct up to `floor((N-K)/2)` symbol errors (*not* bit errors) in each codeword.

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.

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

Dialog Box



Codeword length N
The codeword length.

Message length K
The message length.

Specify primitive polynomial
When you select this box, you can specify the primitive polynomial as a binary row vector.

Primitive polynomial
Binary row vector representing the primitive polynomial in descending order of powers.

Specify generator polynomial
When you select this box, you can specify the generator polynomial as an integer row vector.

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

Output number of corrected errors
When you select this box, the block outputs the number of corrected errors in each word through a second output port.

Algorithm

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

Pair Block

Integer-Input RS Encoder

References

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

[2] Berlekamp, Elwyn R., *Algebraic Coding Theory*, New York, McGraw-Hill, 1968.

See Also

Binary-Output RS Decoder

Integer to Bit Converter

Purpose Map vector of integers to vector of bits

Library Utility Blocks

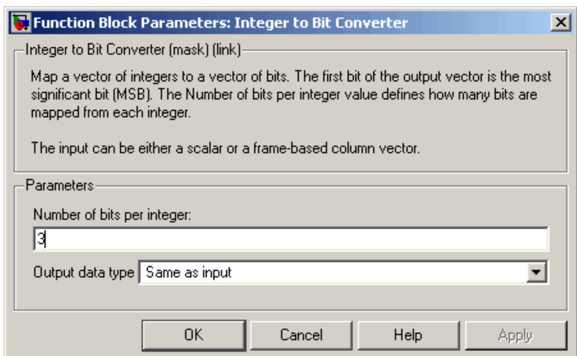
Description

The Integer to Bit Converter block maps each integer in the input vector to a group of bits in the output vector. If M is the **Number of bits per integer** parameter, then the input integers must be between 0 and 2^M-1. The block maps each integer to a group of M bits, using the first bit as the most significant bit. As a result, the output vector length is M times the input vector length.

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

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

Dialog Box



Number of bits per integer
: The number of bits the block uses to represent each integer of the input. This parameter must be an integer between 1 and 31.

Output data type
: The output data type can be set to `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, or `double`. If this field is set to `Same as input`, the output data type will be inherited from the input signal.

Examples

If the input is [7; 13] and the **Number of bits per integer** parameter is 4, then the output is [0; 1; 1; 1; 1; 1; 0; 1]. The first group of four bits (0, 1, 1, 1) represents 7 and the second group of four bits (1, 1, 0, 1) represents 13. Notice that the output length is four times the input length.

Pair Block

Bit to Integer Converter

Integrate and Dump

Purpose

Integrate discrete-time signal, resetting to zero periodically

Library

Comm Filters

Description

The Integrate and Dump block creates a cumulative sum of the discrete-time input signal, while resetting the sum to zero according to a fixed schedule. When the simulation begins, the block discards the number of samples specified in the **Offset** parameter. After this initial period, the block sums the input signal along columns and resets the sum to zero every N input samples, where N is the **Integration period** parameter value. The reset occurs after the block produces its output at that time step.

This block supports inputs and outputs of type `double` and `single`. The port data types are inherited from the signals that drive the block.

The integrate-and-dump operation is often used in a receiver model when the system's transmitter uses a simple square-pulse model. It can also be used in fiber optics and in spread-spectrum communication systems such as CDMA (code division multiple access) applications.

The input can be either a scalar or a frame-based matrix. If the input is frame-based, then it must have k*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.

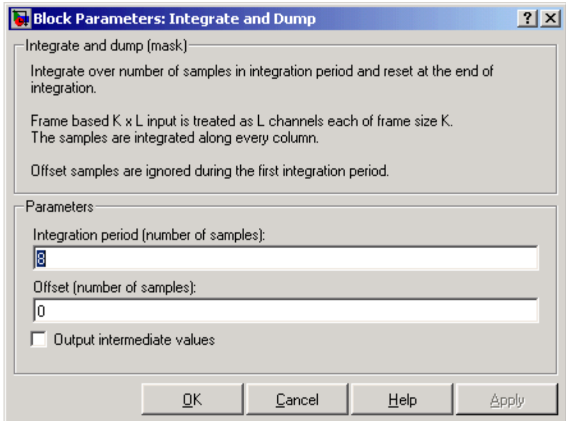
Transients and Delays

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 2-277.

Dialog Box



Integration period
: The number of input samples between resets.

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

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

Examples

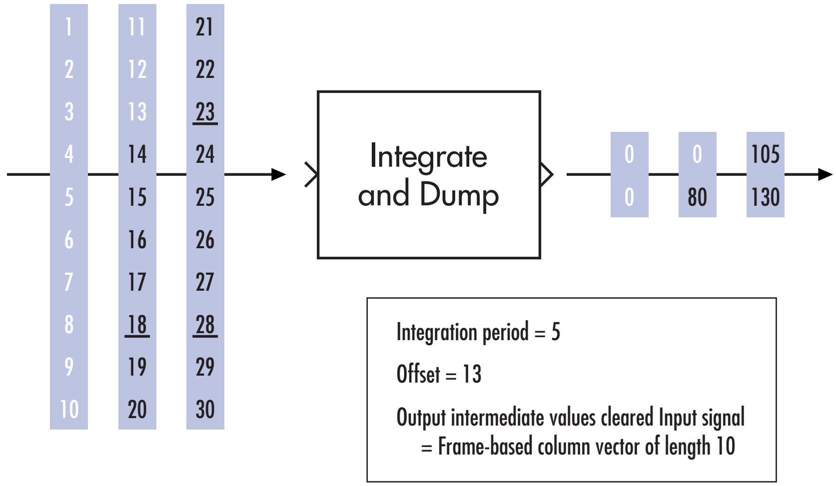
If **Integration period** is 4 and **Offset** is the scalar 3, then the table below shows how the block treats the beginning of a ramp (1, 2, 3, 4,...) in several situations. (The values shown in the table do not reflect vector sizes but merely indicate numerical values.)

Output intermediate values Check Box	Input Signal Properties	First Several Output Values
Cleared	Sample-based scalar	0, 0, 4+5+6+7, and 8+9+10+11, where one 0 is an initial transient value and the other 0 is a delay value that results from the cleared check box and sample-based input.
Cleared	Frame-based column vector of length 4	0, 4+5+6+7, and 8+9+10+11, where 0 is an initial delay value that results from the nonzero offset. The output is a frame-based scalar.
Selected	Sample-based scalar	0, 0, 0, 4, 4+5, 4+5+6, 4+5+6+7, 8, 8+9, 8+9+10, 8+9+10+11, and 12, where the three 0s are initial transient values.
Selected	Frame-based column vector of length 4	0, 0, 0, 4, 4+5, 4+5+6, 4+5+6+7, 8, 8+9, 8+9+10, 8+9+10+11, and 12, where the three 0s are initial transient values. The output is a frame-based column vector of length 4.

In all cases, the block discards the first three input samples (1, 2, and 3).

Example of Transient and Delay

The figure below illustrates a situation in which the block exhibits both a transient effect for three output samples, as well as a one-sample delay in alternate subsequent output samples for the rest of the simulation. 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), Ideal Rectangular Pulse Filter

Purpose

Alternately select elements from two input vectors to generate output vector

Library

Sequence Operations

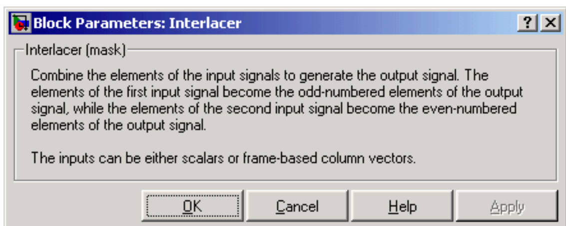
Description

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

The inputs can be either scalars or frame-based column vectors. The block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, `double`, and fixed-point. The data type of this output will be the same as that of the input signals.

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

Dialog Box



Examples

If the two input vectors are frame-based with values `[1; 2; 3; 4]` and `[5; 6; 7; 8]`, then the output vector is `[1; 5; 2; 6; 3; 7; 4; 8]`.

Pair Block

Deinterlacer

See Also

General Block Interleaver; Mux (Simulink)

I/Q Imbalance

Purpose

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

Library

RF Impairments

Description

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

The I/Q Imbalance block applies amplitude and phase imbalances to the in-phase and quadrature components of the input signal, and then combines the results into a complex signal. The block

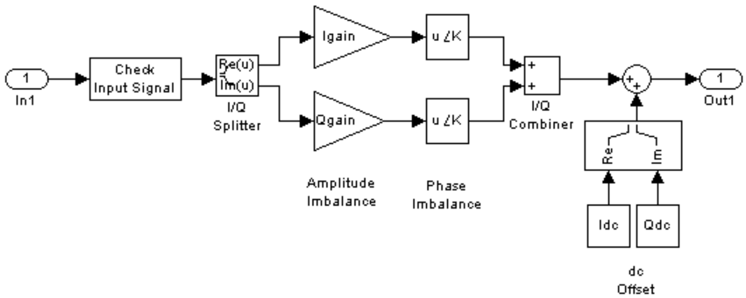
1. Separates the signal into its in-phase and quadrature components.

2. Applies amplitude and phase imbalances, specified by the **I/Q amplitude imbalance (dB)** and **I/Q phase imbalance (deg)** parameters, respectively, to both components.

3. Combines the in-phase and quadrature components into a complex signal.

4. Applies an in-phase dc offset, specified by the **I dc offset** parameter, and a quadrature offset, specified by the **Q dc offset** parameter, to the signal.

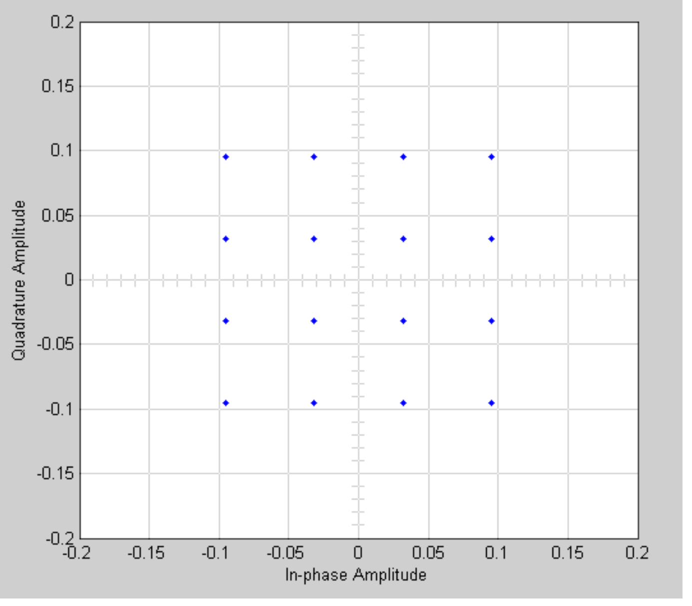
The block performs these operations in the subsystem shown in the following diagram, which you can view by right-clicking the block and selecting **Look under mask**:



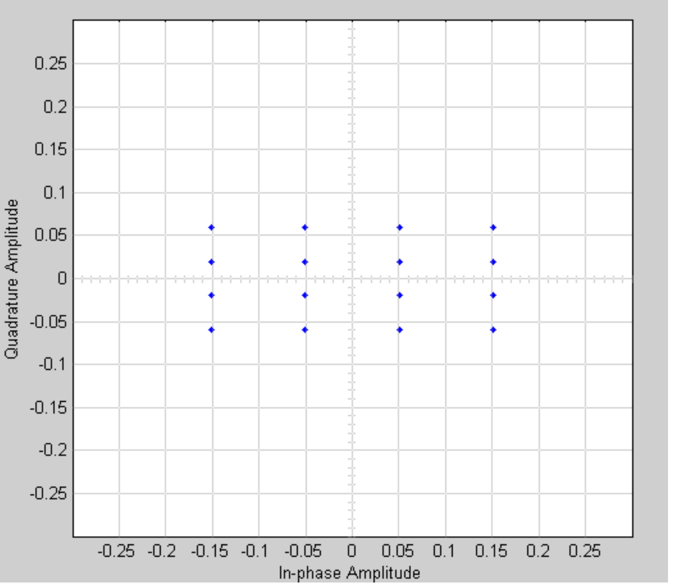
The value of the **I/Q amplitude imbalance (dB)** parameter is divided between the in-phase and quadrature components:

- If you enter a positive value *X* for the **I/Q amplitude imbalance (dB)**, the block applies a gain of +*X/2* dB to the in-phase component and a gain of -*X/2* dB to the quadrature component.
- If you enter a negative value *X* for the **I/Q amplitude imbalance (dB)**, the block applies a gain of -*X/2* dB to the in-phase component and a gain of +*X/2* dB to the quadrature component.

The effects of changing the block's parameters are illustrated by the following scatter plots of a signal modulated by 16-ary quadrature amplitude modulation (QAM) with an average power of `0.01` watts. The usual 16-ary QAM constellation without distortion is shown in the first scatter plot:

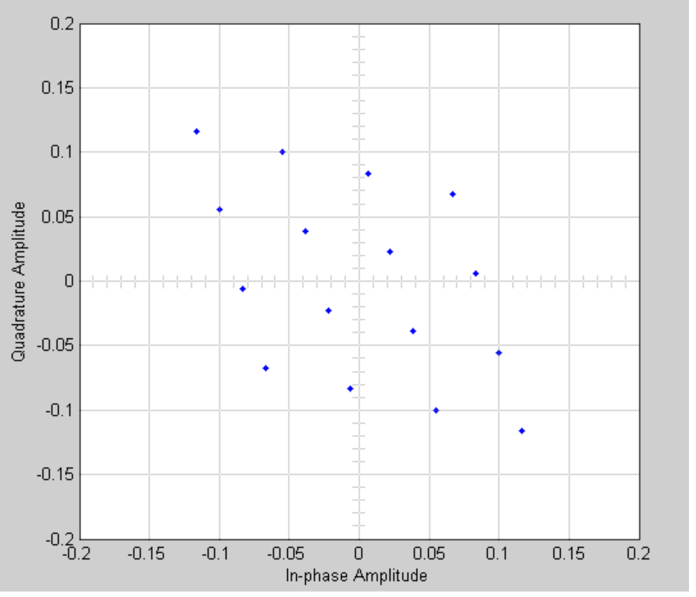


The following figure shows a scatter plot of an output signal, modulated by 16-ary QAM, from the I/Q block with **I/Q amplitude imbalance (dB)** set to `8` and all other parameters set to `0`:

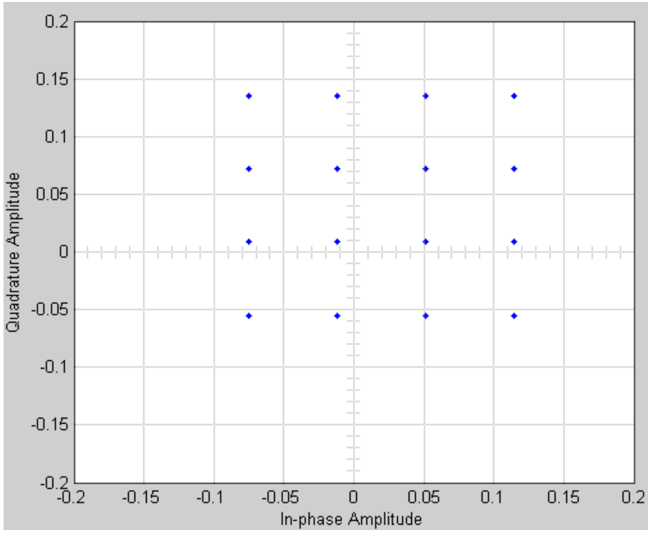


Observe that the scatter plot is stretched horizontally and compressed vertically compared to the undistorted constellation.

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:

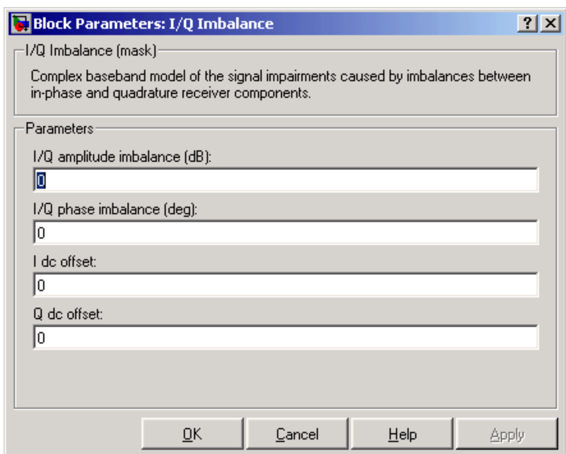


Setting the **I dc offset** to `0.02` and the **Q dc offset** to `0.04` shifts the constellation 0.02 to the right and 0.04 up, as shown below:



See "Scatter Plot Examples" for a description of the model that generates this plot.

Dialog Box



I/Q amplitude imbalance (dB)
Scalar specifying the I/Q amplitude imbalance in decibels.

I/Q phase imbalance (deg)
Scalar specifying the I/Q phase imbalance in degrees.

I dc offset
Scalar specifying the in-phase dc offset.

Q dc offset
Scalar specifying the amplitude dc offset.

See Also

Memoryless Nonlinearity

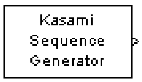
Purpose

Generate Kasami sequence from set of Kasami sequences

Library

Sequence Generators sublibrary of Comm Sources

Description



The Kasami Sequence Generator block generates a sequence from the set of Kasami sequences. The Kasami sequences are a set of sequences that have good cross-correlation properties.

There are two classes of Kasami sequences: the *small set* and the *large set*. The large set contains all the sequences in the small set. Only the small set is optimal in the sense of matching Welch's lower bound for correlation functions.

Kasami sequences have period $N = 2^n - 1$, where n is a nonnegative, even integer. Let u be a binary sequence of length N, and let w be the sequence obtained by decimating u by $2^{n/2} + 1$. The small set of Kasami sequences is defined by the following formulas, in which T denotes the left shift operator, m is the shift parameter for w, and $\oplus$ denotes addition modulo 2.

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

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.

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

Large Set of Kasami Sequences for mod(n, 4) = 2

The sequences described in the first three rows of the preceding figure correspond to the Gold sequences for mod(n, 4) = 2. See the reference page for the Gold Sequence Generator block for a description of Gold sequences. However, the Kasami sequences form a larger set than the Gold sequences.

The correlation functions for the sequences takes on the values

{-$t(n)$, -$s(n)$, -1, $s(n)$ -2 , $t(n)$ - 2}

where

$$t(n) = 1 + 2^{(n+2)/2},\ n \text{ even}$$
$$s(n) = \frac{1}{2}(t(n)+1)$$

Block Parameters

The **Generator polynomial** parameter specifies the generator polynomial, which determines the connections in the shift register that generates the sequence u. You can specify the **Generator polynomial** parameter using either of these formats:

- A vector that lists the coefficients of the polynomial in descending order of powers. The first and last entries must be 1. Note that the length of this vector is one more than the degree of the generator polynomial.

- 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`	$m = 0, \ldots , 2^{n/2} - 2$	$u \oplus T^m w$

- 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]`	$k = 0, 1, \ldots, 2^n - 2$ $m = -1$	$u \oplus T^k v$
`[-2 m]`	$k = -2$ $m = 0, 1, \ldots, 2^{n/2} - 2$	$u \oplus T^m w$
`[-1 m]`	$k = -1$ $m = 0, \ldots, 2^{n/2} - 2$	$v \oplus T^m w$
`[k m]`	$k = 0, \ldots, 2^n - 2$ $m = 0, \ldots, 2^{n/2} - 2$	$u \oplus T^k v \oplus T^m w$

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 2-431 for an example.

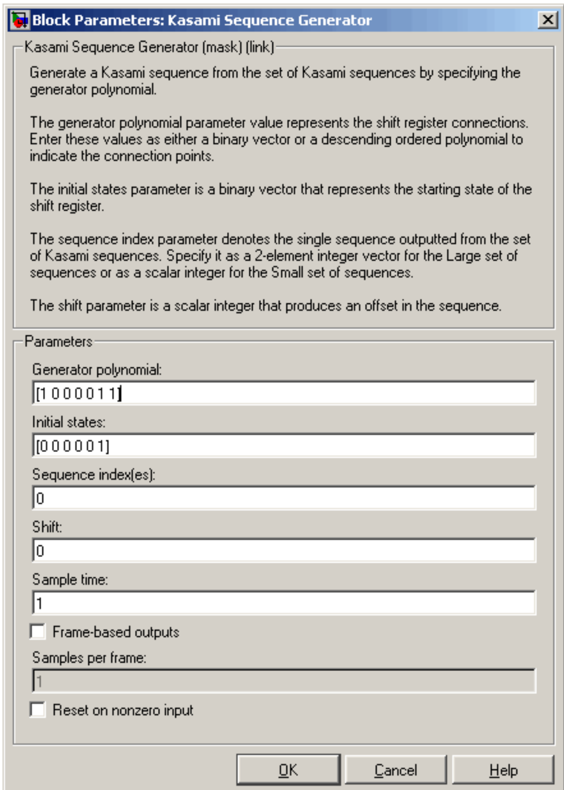
Polynomials for Generating Kasami Sequences

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.**

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.

Linearized Baseband PLL

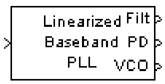
Purpose

Implement linearized version of a baseband phase-locked loop

Library

Components sublibrary of Synchronization

Description



The Linearized Baseband PLL block is a feedback control system that automatically adjusts the phase of a locally generated signal to match the phase of an input signal. Unlike thePhase-Locked Loop block, this block uses a baseband model method. Unlike theBaseband 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 the Signal Processing Toolbox. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.

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

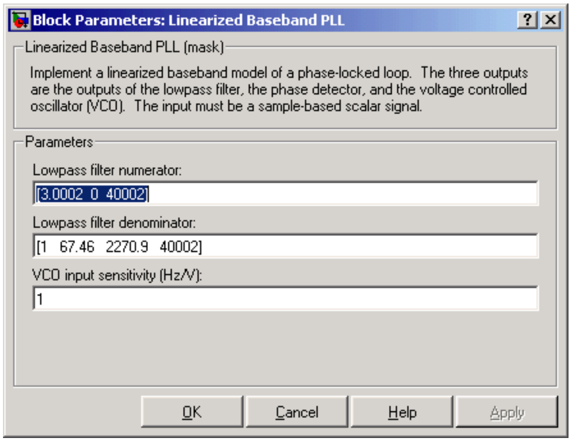
- A voltage-controlled oscillator (VCO). You specify the sensitivity of the VCO signal to its input using the **VCO input sensitivity** parameter. This parameter, measured in Hertz per volt, is a scale factor that determines how much the VCO shifts from its quiescent frequency.

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

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

- The output of the VCO

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 Using the Communications Blockset.

LMS Decision Feedback Equalizer

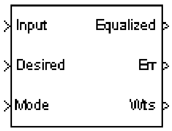
Purpose

Equalize using decision feedback equalizer that updates weights with LMS algorithm

Library

Equalizers

Description



The LMS Decision Feedback Equalizer block uses a decision feedback equalizer and the LMS algorithm to equalize a linearly modulated 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.

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 Using the Communications Blockset.
- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

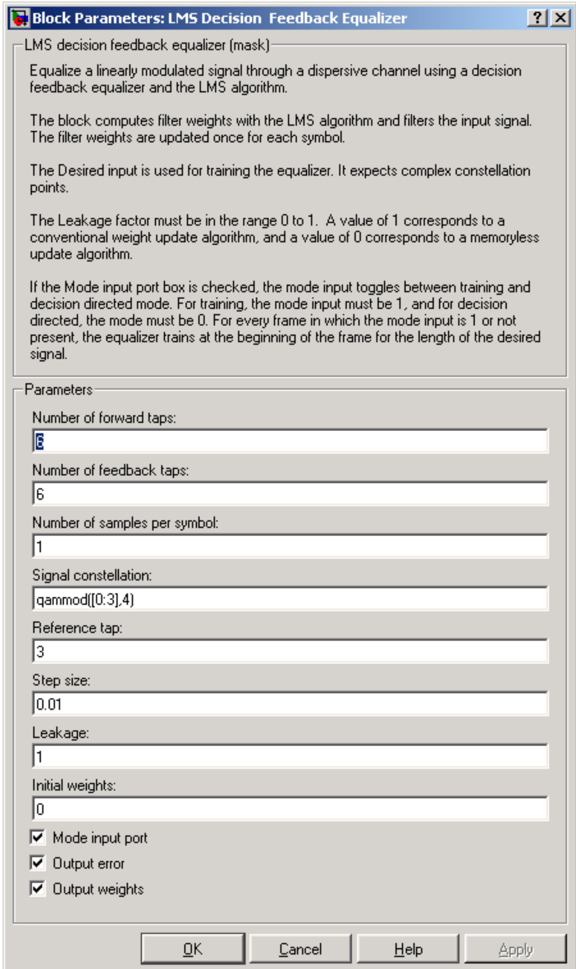
Equalizer Delay

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

LMS Linear Equalizer

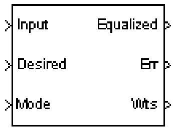
Purpose

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.

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 Using the Communications Blockset.
- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

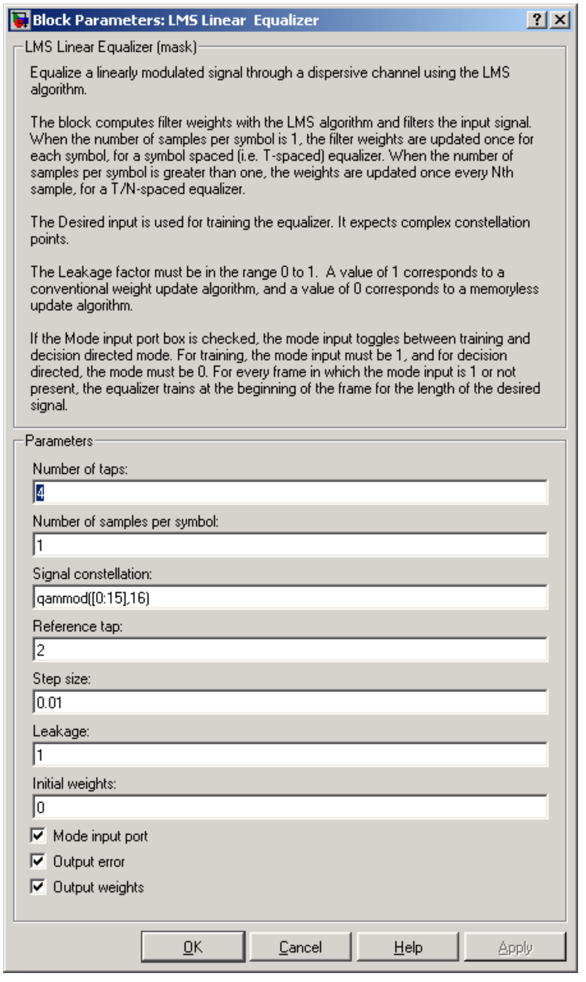
Equalizer Delay

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 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.

[2] Haykin, Simon, *Adaptive Filter Theory*, Third Ed., Upper Saddle River, N.J., Prentice-Hall, 1996.

[3] Kurzweil, Jack, *An Introduction to Digital Communications*, New York, Wiley, 2000.

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

See Also

LMS Decision Feedback Equalizer, Normalized LMS Linear Equalizer, Sign LMS Linear Equalizer, Variable Step LMS Linear Equalizer, RLS Linear Equalizer, CMA Equalizer

Purpose

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

Library

Block sublibrary of Interleaving

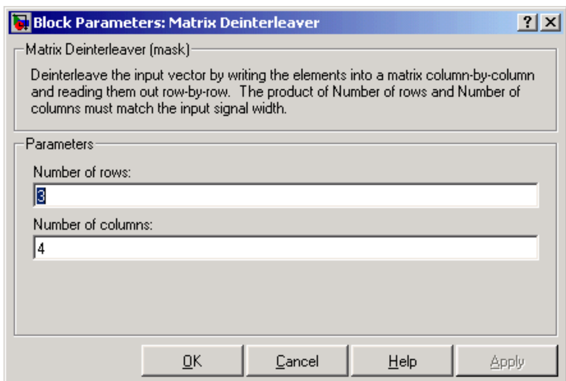
Description

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.

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 a matrix along diagonals

Library

Block sublibrary of Interleaving

Description

The Matrix Helical Scan Deinterleaver block performs block deinterleaving by filling a matrix with the input symbols in a helical fashion and then sending the matrix contents to the output port row by row. The **Number of rows** and **Number of columns** parameters are the dimensions of the matrix that the block uses internally for its computations.

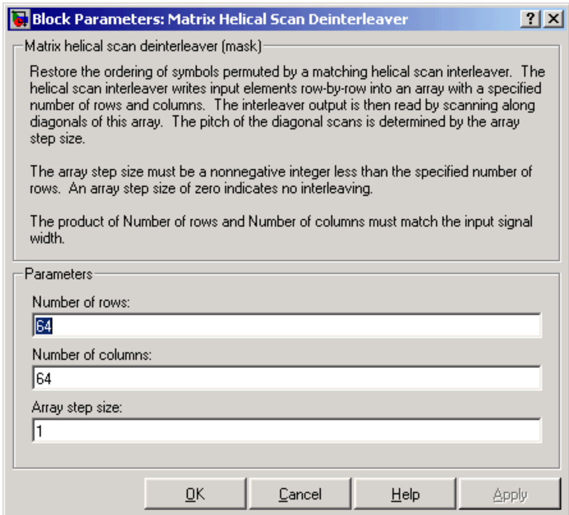
Helical fashion means that the block places input symbols along diagonals of the matrix. The number of elements in each diagonal matches the **Number of columns** parameter, after the block wraps past the edges of the matrix when necessary. The block traverses diagonals so that the row index and column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

The **Array step size** parameter is the slope of each diagonal, that is, the amount by which the row index increases as the column index increases by one. This parameter must be an integer between zero and the **Number of rows** parameter. If the **Array step size** parameter is zero, then the block does not deinterleave and the output is the same as the input.

The number of elements of the input vector must be the product of **Number of rows** and **Number of columns**. If the input is frame-based, then it must be a column vector.

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

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

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.

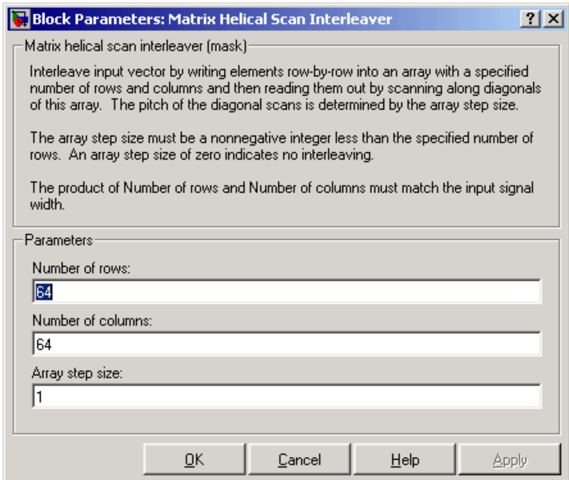
Helical fashion means that the block selects output symbols by selecting elements along diagonals of the matrix. The number of elements in each diagonal matches the **Number of columns** parameter, after the block wraps past the edges of the matrix when necessary. The block traverses diagonals so that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

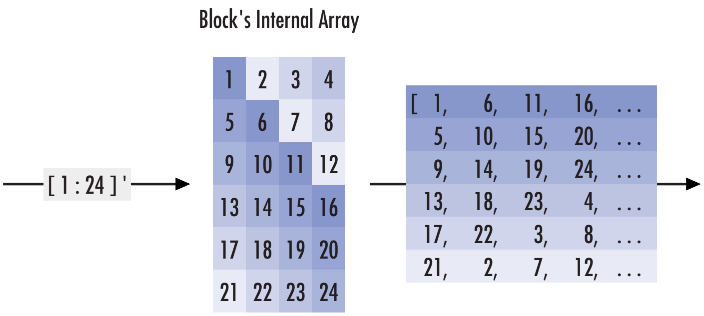
The slope of the diagonals that the block reads.

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 a matrix by rows and emptying it by columns

Library

Block sublibrary of Interleaving

Description

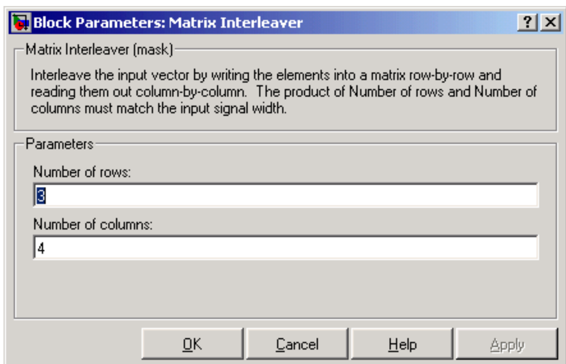
The Matrix Interleaver block performs block interleaving by filling a matrix with the input symbols row by row and then sending the matrix contents to the output port column by column.

The **Number of rows** and **Number of columns** parameters are the dimensions of the matrix that the block uses internally for its computations.

The number of elements of the input vector must be the product of **Number of rows** and **Number of columns**. If the input is frame-based, then it must be a column vector.

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

Dialog Box



Number of rows

The number of rows in the matrix that the block uses for its computations.

Number of columns
The number of columns in the matrix that the block uses for its computations.

Examples

If the **Number of rows** and **Number of columns** parameters are 2 and 3, respectively, then the interleaver uses a 2-by-3 matrix for its internal computations. Given an input signal of `[1; 2; 3; 4; 5; 6]`, the block produces an output of `[1; 4; 2; 5; 3; 6]`.

Pair Block

Matrix Deinterleaver

See Also

General Block Interleaver

Purpose

Demodulate DPSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

Description

The M-DPSK Demodulator Baseband block demodulates a signal that was modulated using the M-ary differential phase shift keying method. The input is a baseband representation of the modulated signal. The input and output for this block are discrete-time signals. The input can be either a scalar or a frame-based column vector. The block accepts the input data types `single` and `double`.

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

Binary or Integer Outputs

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

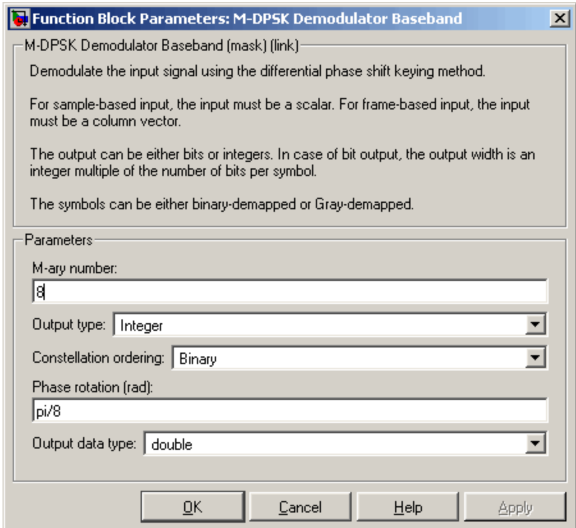
$$\theta + 2\pi k/M$$

to k, where θ 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 theM-DPSK Modulator Baseband andM-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
: For integer inputs, this block can output the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. For bit inputs, output can be `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, or `double`.

Pair Block M-DPSK Modulator Baseband

See Also DBPSK Demodulator Baseband, DQPSK Demodulator Baseband, M-PSK Demodulator Baseband

References

[1] Pawula, R. F., "On M-ary DPSK Transmission Over Terrestrial and Satellite Channels," *IEEE Transactions on Communications*, Vol. COM-32, July 1984, 752-761.

M-DPSK Modulator Baseband

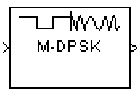
Purpose

Modulate using M-ary differential phase shift keying method

Library

PM, in Digital Baseband sublibrary of Modulation

Description



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

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

Inputs and Constellation Types

If the **Input type** parameter is set to `Integer`, then valid input values are integers between 0 and M-1. In this case, the input can be either a scalar or a frame-based column vector. If the first input is k_1, then the modulated symbol is

$$\exp\left(j\theta + j2\pi\frac{k_1}{m}\right)$$

where θ is the **Phase rotation** parameter. If a successive input is k, then the modulated symbol is

$$\exp\left(j\theta + j2\pi\frac{k}{m}\right)\cdot\text{(previous modulated symbol)}$$

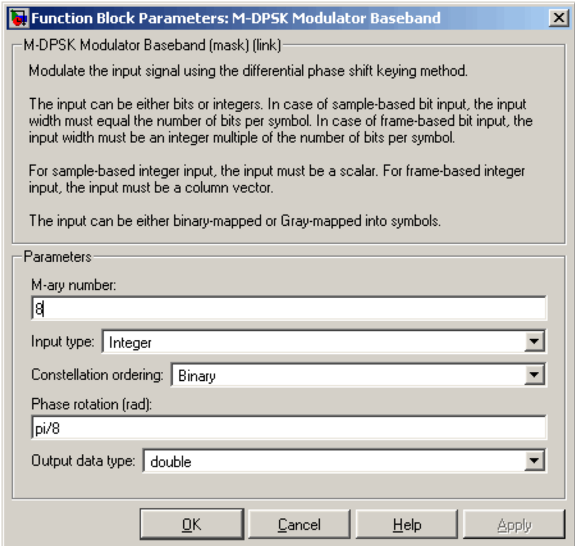
If the **Input type** parameter is set to `Bit` and the **M-ary number** parameter has the form 2^K for some positive integer K, then the block accepts binary representations of integers between 0 and M-1. It modulates each group of K bits, called a binary *word*. The input can be either a vector of length K or a frame-based column vector whose length is an integer multiple of K.

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. θ is the **Phase rotation** parameter. The phase difference is between the previous symbol and the current symbol.

Current Input	Binary-Coded Phase Difference	Gray-Coded Phase Difference
[0 0 0]	$j\theta$	$j\theta$
[0 0 1]	$j\theta + j\pi/4$	$j\theta + j\pi/4$
[0 1 0]	$j\theta + j\pi 2/4$	$j\theta + j\pi 3/4$
[0 1 1]	$j\theta + j\pi 3/4$	$j\theta + j\pi 2/4$
[1 0 0]	$j\theta + j\pi 4/4$	$j\theta + j\pi 7/4$
[1 0 1]	$j\theta + j\pi 5/4$	$j\theta + j\pi 6/4$
[1 1 0]	$j\theta + j\pi 6/4$	$j\theta + j\pi 4/4$
[1 1 1]	$j\theta + j\pi 7/4$	$j\theta + j\pi 5/4$

For more details about the `Binary` and `Gray` options, see the reference page for theM-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`.

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.

Memoryless Nonlinearity

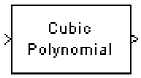
Purpose

Apply memoryless nonlinearity to complex baseband signal.

Library

RF Impairments

Description

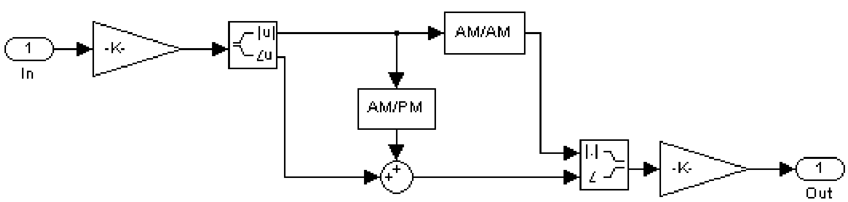


The Memoryless Nonlinearity block applies a memoryless nonlinearity to a complex, baseband signal. You can use the block to model radio frequency (RF) impairments to a signal at the receiver.

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 figure below.



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:

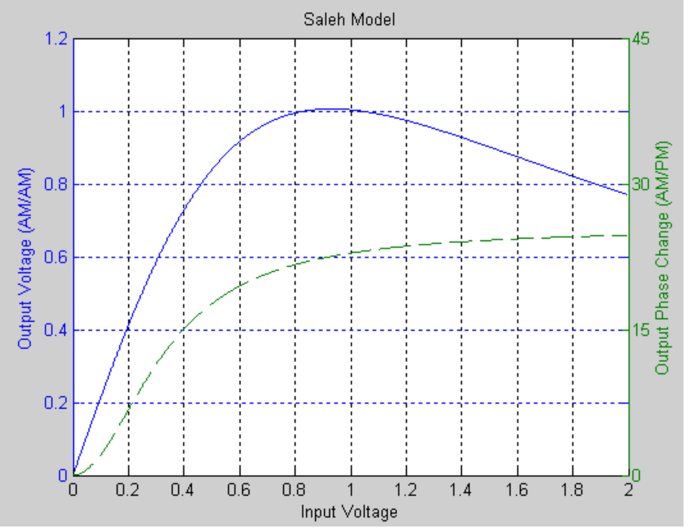
1 Right-click on the Memoryless Nonlinearity block and select **Look under mask**. This displays the block's configuration underneath the mask. The block contains five subsystems corresponding to the five nonlinearity methods.

2 Double-click the subsystem for the method you are interested in. This displays the subsystem shown in the preceding figure, Nonlinearity Subsytem on page 2-322.

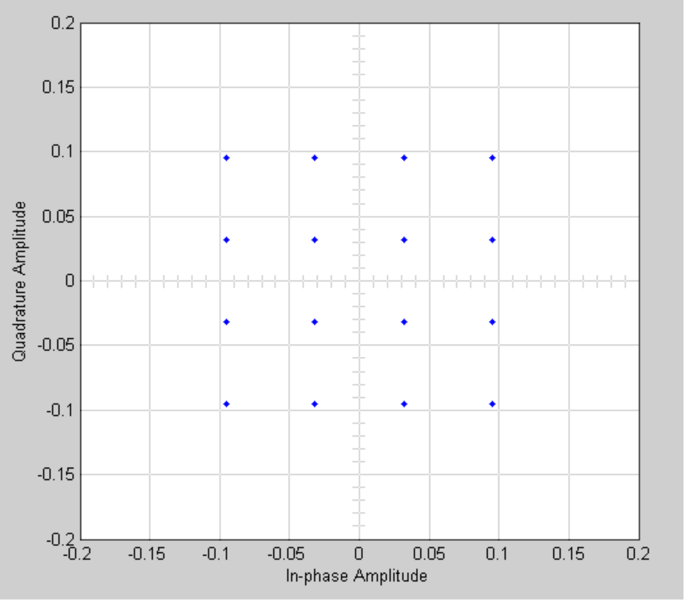
3 Double-click on one of the subsystems labeled AM/AM or AM/PM to view how the block implements the conversions.

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

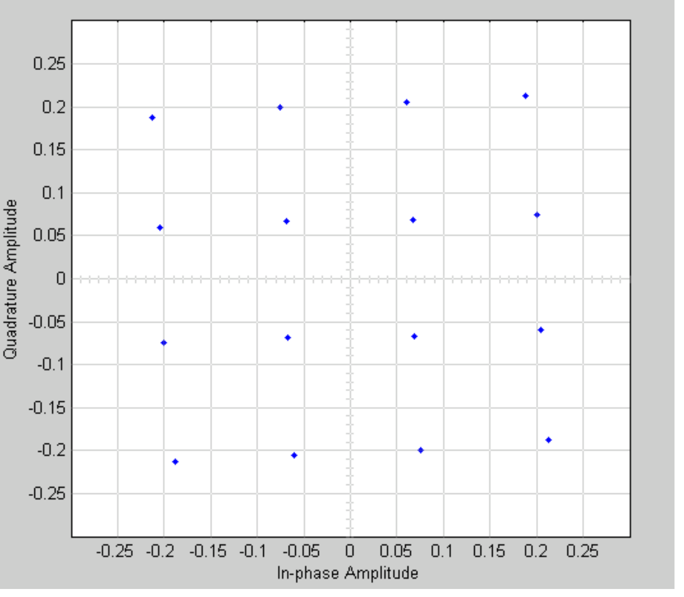
- Output voltage against input voltage for the AM/AM conversion
- Output phase against input voltage for the AM/PM conversion



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:

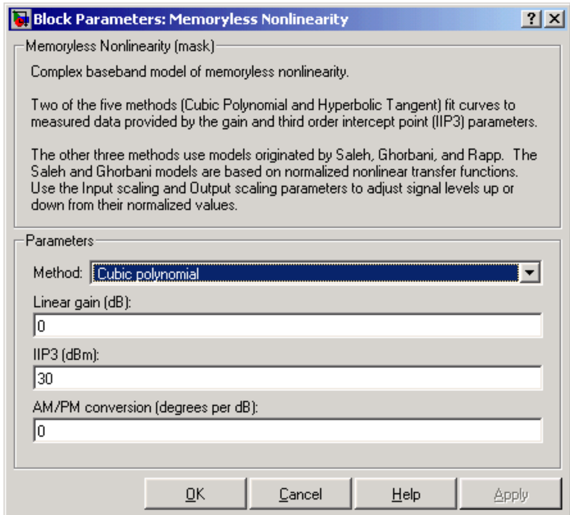
$$F_{AM/AM}(u) = \frac{u}{\left(1 + \left(\frac{u}{O_{sat}}\right)^{2S}\right)^{1/2S}}$$

where u is the magnitude of the scaled signal, S is the **Smoothness factor**, and O_{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.

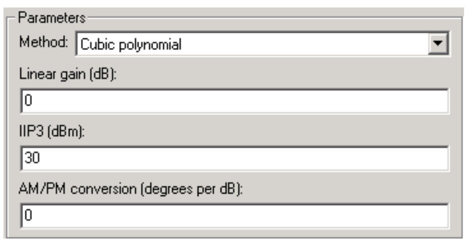
Dialog Box



Method

The nonlinearity method.

The following describes specific parameters for each method.



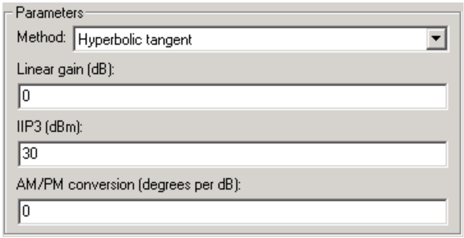
Linear gain (db)

Scalar specifying the linear gain for the output function.

IIP3 (dBm)

Scalar specifying the third order intercept.

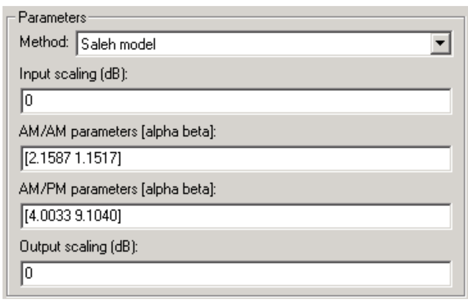
AM/PM conversion (degrees per dB)
Scaler specifying the AM/PM conversion in degrees per decibel.



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.



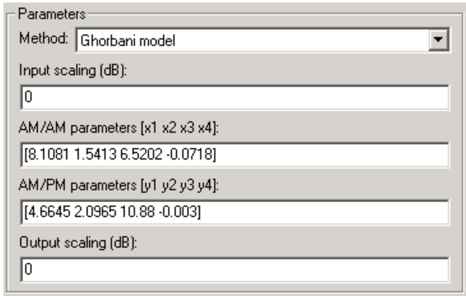
Input scaling (dB)
Number that scales the input signal level.

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]

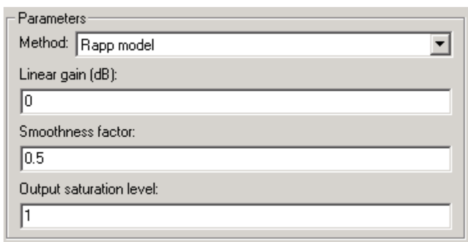
Vector specifying the AM/AM parameters.

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.

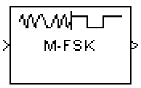
Purpose

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.

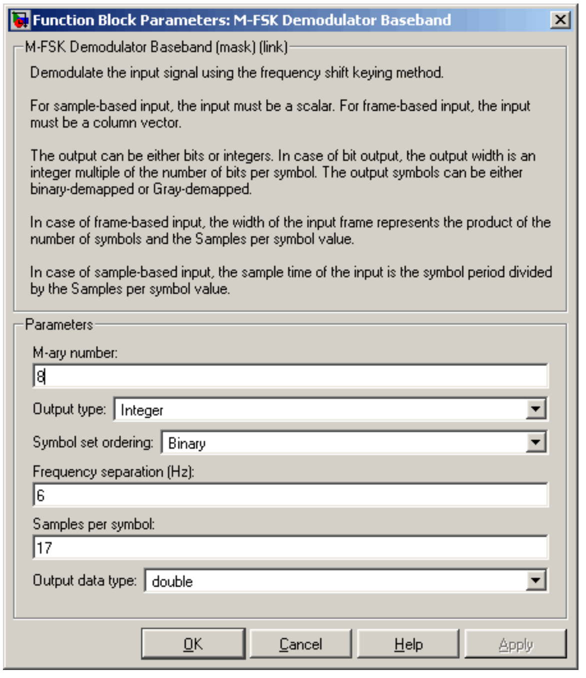
If the **Output type** parameter is set to `Bit` and the **M-ary number** parameter has the form 2^K for some positive integer K, then the block outputs binary representations of integers between 0 and M-1. It outputs a group of K bits, called a binary *word*, for each symbol.

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 theM-FSK Modulator Baseband andM-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.

Frequency separation (Hz)
The distance between successive frequencies in the modulated signal.

Output data type
The output type of the block can be specified here as `boolean`, `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, or `double`. By default, the block sets this to `double`.

Pair Block

M-FSK Modulator Baseband

See Also

CPFSK Demodulator Baseband

M-FSK Modulator Baseband

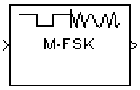
Purpose

Modulate using M-ary frequency shift keying method

Library

FM, in Digital Baseband sublibrary of Modulation

Description



The M-FSK Modulator Baseband block modulates using the M-ary frequency shift keying method. The output is a baseband representation of the modulated signal.

The **M-ary number** parameter, M, is the number of frequencies in the modulated signal. The **Frequency separation** parameter is the distance, in Hz, between successive frequencies of the modulated signal. 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.

Input Signal Values

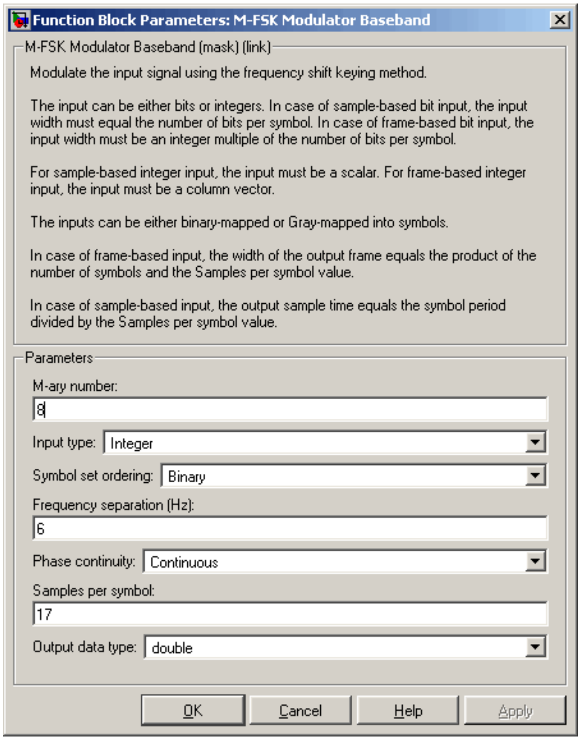
The input and output for this block are discrete-time signals. The **Input type** parameter determines whether the block accepts integers between 0 and M-1, or binary representations of integers:

- If **Input type** is set to `Integer`, then the block accepts integers. The input can be either a scalar or a frame-based column vector of type `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, or a `double` with an integer value. They can also be `boolean` if the size of the alphabet is 2 (i.e. M = 2).
- If **Input type** is set to `Bit`, then the block accepts groups of K bits, called binary words. The input can be either a vector of length K or a frame-based column vector (whose length is an integer multiple of K), and must be `boolean` or `double` typed, valued from the set {0, 1}. The **Symbol set ordering** parameter indicates how the block assigns binary words to corresponding integers.
  - If **Symbol set ordering** is set to `Binary`, then the block uses a natural binary-coded ordering.

- If **Symbol set ordering** is set to `Gray`, then the block uses a Gray-coded ordering. For details about the Gray coding, see the reference page for theM-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.

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.

Output data type

The output type of the block can be specified as either a `double` or a `single`. By default, the block sets this to `double`.

Pair Block

M-FSK Demodulator Baseband

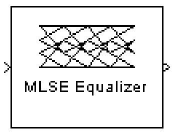
See Also

CPFSK Modulator Baseband

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.

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

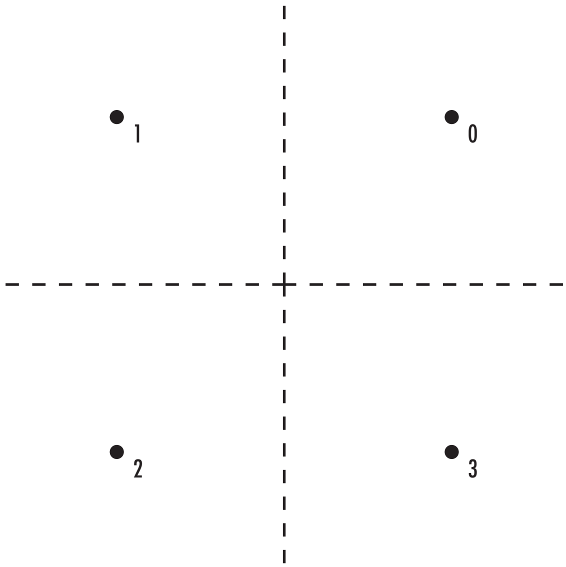
The **Signal constellation** parameter specifies the constellation for the modulated signal, as determined by the modulator in your model. **Signal constellation** is a vector of complex numbers, where the kth complex number in the vector is the constellation point to which the modulator maps the integer 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.

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 using the Matrix Concatenation block.

To learn how the block uses the preamble and postamble, see “"Reset Every Frame" Operation Mode” on page 2-341 below.

"Reset Every Frame" Operation Mode

One way that the Viterbi algorithm can transition between successive frames is called `Reset every frame` mode. You can choose this mode using the **Operation mode** parameter.

In `Reset every frame` mode, the block decodes each frame of data independently, resetting the state metric at the end of each frame. The traceback decoding always starts at the state with the minimum state metric.

The initialization of state metrics depends on whether you specify a preamble and/or postamble:

- If you do not specify a preamble, the decoder initializes the metrics of all states to 0 at the beginning of each frame of data.
- If you specify a preamble, the block uses it to initialize the state metrics at the beginning of each frame of data. More specifically, the block decodes the preamble and assigns a metric of 0 to the decoded

state. If the preamble does not decode to a unique state – that is, if the length of the preamble is less than the channel memory – the decoder assigns a metric of 0 to all states that can be represented by the preamble. Whenever you specify a preamble, the traceback path ends at one of the states represented by the preamble.

- If you do not specify a postamble, the traceback path starts at the state with the smallest metric.
- If you specify a postamble, the traceback path begins at the state represented by the postamble. If the postamble does not decode to a unique state, the decoder identifies the smallest of all possible decoded states that are represented by the postamble and begins traceback decoding at that state.

Note In `Reset every frame` mode, the input to the MLSE Equalizer block must contain at least T symbols, not including an optional preamble, where T is the **Traceback depth** parameter.

Continuous Operation Mode

An alternative way that the Viterbi algorithm can transition between successive frames is called `Continuous with reset option` mode. You can choose this mode using the **Operation mode** parameter.

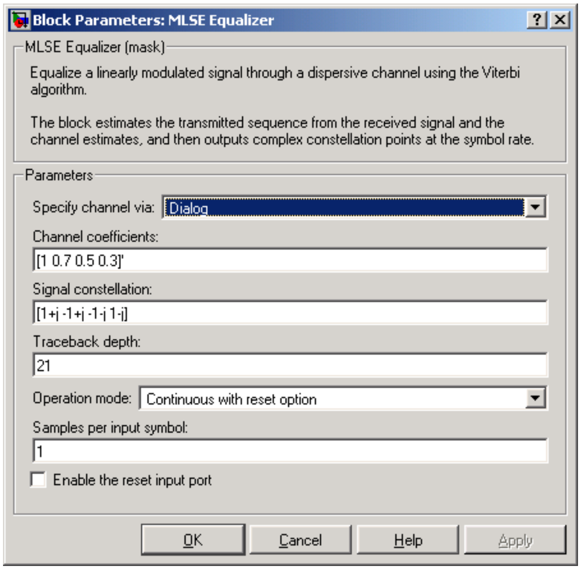
In `Continuous with reset option` mode, the block initializes the metrics of all states to 0 at the beginning of the simulation. At the end of each frame, the block saves the internal state metric for use in computing the traceback paths in the next frame.

If you select 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.

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.

M-PAM Demodulator Baseband

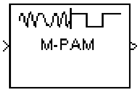
Purpose

Demodulate PAM-modulated data

Library

AM, in Digital Baseband sublibrary of Modulation

Description



The M-PAM Demodulator Baseband block demodulates a signal that was modulated using the M-ary pulse amplitude modulation. The input is a baseband representation of the modulated signal.

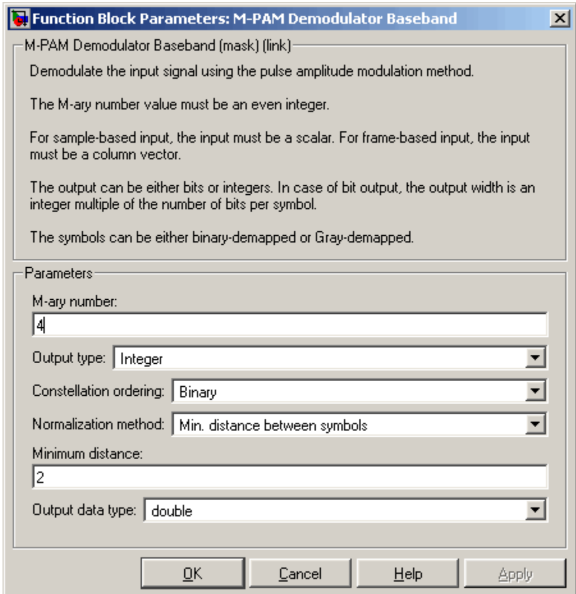
The signal constellation has M points, where M is the **M-ary number** parameter. M must be an even integer. The block scales the signal constellation based on how you set the **Normalization method** parameter. For details on the constellation and its scaling, see the reference page for theM-PAM Modulator Baseband block.

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

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 theM-PAM Modulator Baseband block.

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 (watts)
The average power of the symbols in the constellation. This field appears only when **Normalization method** is set to `Average Power`.

Peak power (watts)
The maximum power of the symbols in the constellation. This field appears only when **Normalization method** is set to `Peak Power`.

Output data type
For integer inputs, this block can output the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. For bit inputs, output can be `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, or `double`.

Pair Block

M-PAM Modulator Baseband

See Also

General QAM Demodulator Baseband

M-PAM Modulator Baseband

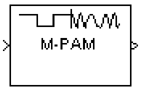
Purpose

Modulate using M-ary pulse amplitude modulation

Library

AM, in Digital Baseband sublibrary of Modulation

Description



The M-PAM Modulator Baseband block modulates using M-ary pulse amplitude modulation. The output is a baseband representation of the modulated signal. The **M-ary number** parameter, M, is the number of points in the signal constellation. It must be an even integer.

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 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 table below 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 **Minimum distance** parameter

Value of Normalization method Parameter	Scaling Condition
`Average Power`	The average power of the symbols in the constellation is the **Average power** parameter
`Peak Power`	The maximum power of the symbols in the constellation is the **Peak power** parameter

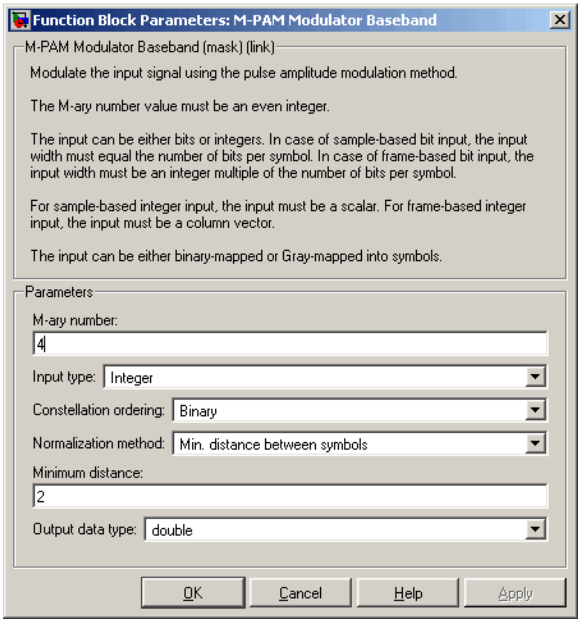
Input Signal Values

The input and output for this block are discrete-time signals. The **Input type** parameter determines whether the block accepts integers between 0 and M-1, or binary representations of integers.

- If **Input type** is set to `Integer`, then the block accepts integers. The input can be either a scalar or a frame-based column vector of data type `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, or `double`.
- If **Input type** is set to `Bit`, then the block accepts groups of K bits, called binary words. The input can be either a vector of length K or a frame-based column vector whose length is an integer multiple of K. For bit inputs, the block can accept `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, and `double`. The **Constellation ordering** parameter indicates how the block assigns binary words to points of the signal constellation.
  - If **Constellation ordering** is set to `Binary`, then the block uses a natural binary-coded constellation.
  - If **Constellation ordering** is set to `Gray`, then the block uses a Gray-coded constellation.

For details about the Gray coding, see the reference page for theM-PSK Modulator Baseband block.

Dialog Box



M-ary number
: The number of points in the signal constellation. It must be an even integer.

Input type
: Indicates whether the input consists of integers or groups of bits. If this parameter is set to `Bit`, then the **M-ary number** parameter must be 2^K for some positive integer K.

Constellation ordering
: Determines how the block maps each group of input bits to a corresponding integer. This field is active only when **Input 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 (watts)
The average power of the symbols in the constellation. This field appears only when **Normalization method** is set to `Average Power`.

Peak power (watts)
The maximum power of the symbols in the constellation. This field appears only when **Normalization method** is set to `Peak Power`.

Output data type
The output data type can be either `single` or `double`.

Pair Block

M-PAM Demodulator Baseband

See Also

General QAM Modulator Baseband

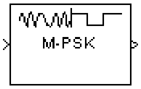
Purpose

Demodulate PSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

Description



The M-PSK Demodulator Baseband block demodulates a signal that was modulated using the M-ary phase shift keying method. The input is a baseband representation of the modulated signal. The input and output for this block are discrete-time signals. The input can be either a scalar or a frame-based column vector of data types `single` or `double`. The **M-ary number** parameter, M, is the number of points in the signal constellation.

Binary or Integer Outputs

If the **Output type** parameter is set to `Integer`, then the block maps the point

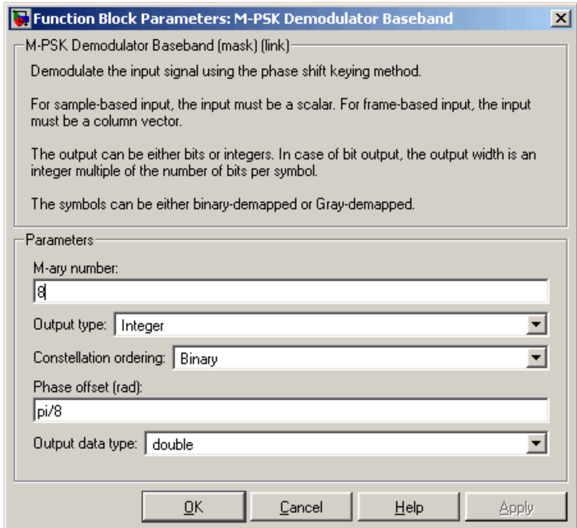
$$\exp(j\theta + j2\pi m/M)$$

to m, where θ is the **Phase offset** parameter and m 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 page for theM-PSK Modulator Baseband block for details.

Dialog Box



M-ary number

The number of points in the signal constellation.

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.

Phase offset (rad)

The phase of the zeroth point of the signal constellation.

Output data type

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

Pair Block

M-PSK Modulator Baseband

See Also

BPSK Demodulator Baseband, QPSK Demodulator Baseband, M-DPSK Demodulator Baseband

M-PSK Modulator Baseband

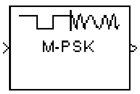
Purpose

Modulate using M-ary phase shift keying method

Library

PM, in Digital Baseband sublibrary of Modulation

Description



The M-PSK Modulator Baseband block modulates using the M-ary phase shift keying method. The output is a baseband representation of the modulated signal. The **M-ary number** parameter, M, is the number of points in the signal constellation.

Baseband M-ary phase shift keying modulation with a phase offset of θ maps an integer m between 0 and M-1 to the complex value

$$\exp(j\theta + j2\pi m/M)$$

The input and output for this block are discrete-time signals. To use integers between 0 and M-1 as input values, set the **Input type** parameter to `Integer`. In this case, the input can be either a scalar or a frame-based column vector. For integer inputs, the block can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. For bit inputs, the block can accept `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, and `double`.

Alternative configurations of the block determine how the block interprets its input and arranges its output, as explained in the sections below.

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, 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.

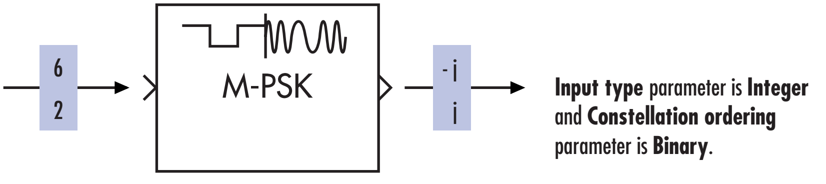
The **Constellation ordering** parameter indicates how the block maps a group of K input bits to a corresponding integer. Choices are `Binary` and `Gray`. For more information, see "Binary-Valued and Integer-Valued Signals" in Using the Communications Blockset.

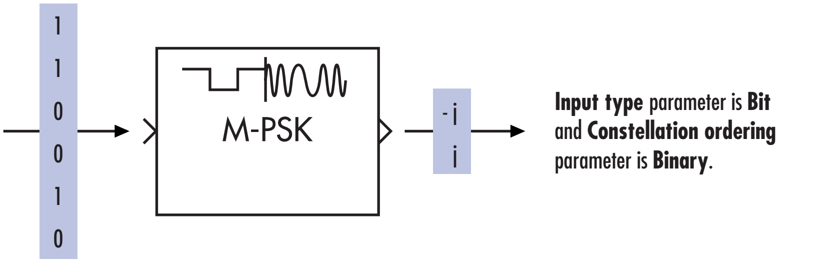
If **Constellation ordering** is set to `Gray`, then the block uses a Gray-coded signal constellation; as a result, binary representations that differ in more than one bit cannot map to consecutive integers modulo M. The explicit mapping is described in "Algorithm" on page 2-359 below.

Frame-Based Inputs

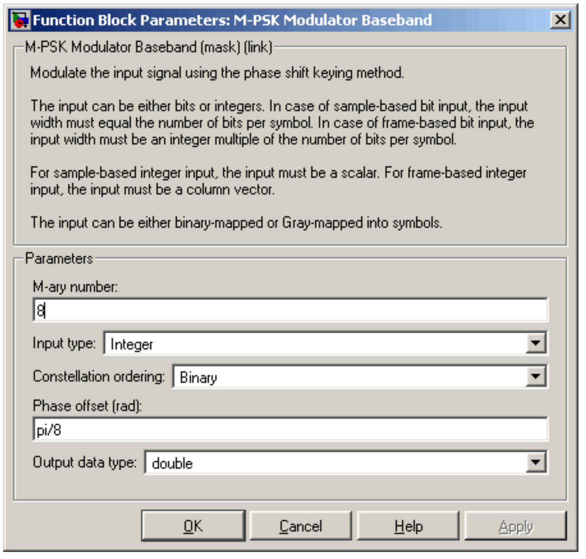
If the input is a frame-based column vector, then the block processes several integers or several binary words, in each time step. (If the **Input type** parameter is set to `Bit`, then a binary word consists of $\log_2(M)$ bits.)

For example, the schematics below 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.





Dialog Box



M-ary number
The number of points in the signal constellation.

Input type
Indicates whether the input consists of integers or groups of bits. If this parameter is set to `Bit`, 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 offset (rad)
The phase of the zeroth point of the signal constellation.

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

Algorithm

If the **Constellation ordering** parameter is set to Gray, then 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, then the block output has phase

$$j\theta + j2\pi m/M$$

where θ 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, then the binary representations that correspond to the zeroth through seventh phases are below.

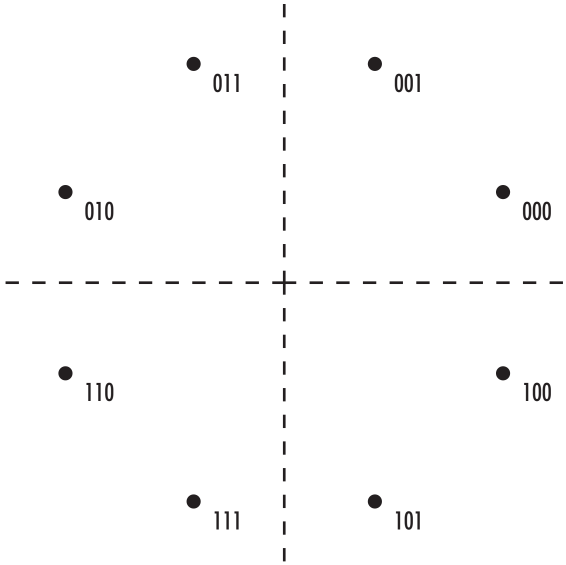
```
M = 8; m = [0:M-1]';
de2bi(bitxor(m,floor(m/2)), log2(M),'left-msb')

ans =

     0     0     0
     0     0     1
```

0	1	1
0	1	0
1	1	0
1	1	1
1	0	1
1	0	0

Below is the 8-ary Gray-coded constellation that the block uses if the **Phase offset** parameter is $\pi/8$.



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. The input signal represents a baseband signal at the symbol rate, so it must be complex-valued and must contain one sample per symbol.

The outputs are as follows:

- The output port labeled `Sig` gives the result of rotating the input signal counterclockwise, where the amount of rotation equals the carrier phase estimate. The `Sig` output is thus a corrected version of the input signal, and has the same sample time and vector size as the input signal.

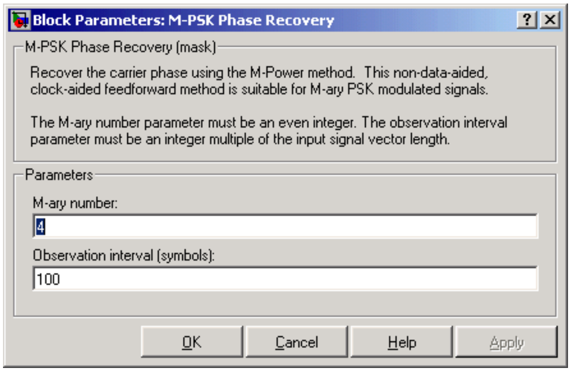
- The output port labeled `Ph` outputs the carrier phase estimate, in degrees, for all symbols in the observation interval. The `Ph` output is a scalar signal.

Note Because the block internally computes the argument of a complex number, the carrier phase estimate has an inherent ambiguity. The carrier phase estimate is between -180/M and 180/M degrees and might differ from the actual carrier phase by an integer multiple of 360/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 Using the Communications Blockset.

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

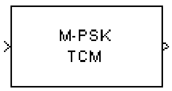
Purpose

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, $\log_2$(**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 theM-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.

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.

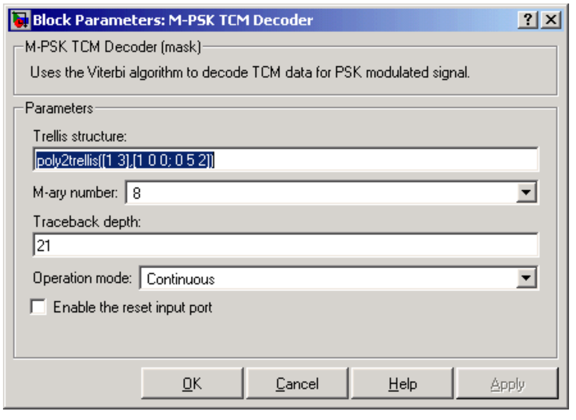
- In `Truncated` mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In `Terminated` mode, the block treats each frame independently. The traceback path always starts at the all-zeros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

Pair Block

M-PSK TCM Encoder

See Also

General TCM Decoder, `poly2trellis`

References

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

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

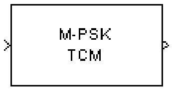
Purpose

Convolutionally encode binary data and modulate using PSK method

Library

Trellis-Coded Modulation

Description



The M-PSK TCM Encoder block implements trellis-coded modulation (TCM) by convolutionally encoding the binary input signal and mapping the result to a PSK signal constellation.

The **M-ary number** parameter is the number of points in the signal constellation, which also equals the number of possible output symbols from the convolutional encoder. (That is, $\log_2$(**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.

Specifying the Encoder

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

- If you want to specify the encoder using its constraint length, generator polynomials, and possibly feedback connection polynomials, then use a `poly2trellis` command within the **Trellis structure** field. For example, to use an encoder with a constraint length of 7, code generator polynomials of 171 and 133 (in octal numbers), and a feedback connection of 171 (in octal), set the **Trellis structure** parameter to

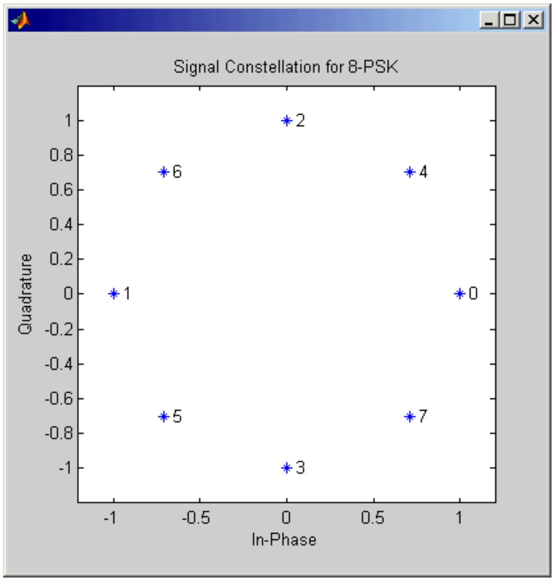
  ```
  poly2trellis(7,[171 133],171)
  ```

- If you have a variable in the MATLAB workspace that contains the trellis structure, then enter its name as the **Trellis structure** parameter. This way is faster because it causes Simulink to spend less time updating the diagram at the beginning of each simulation, compared to the usage in the previous bulleted item.

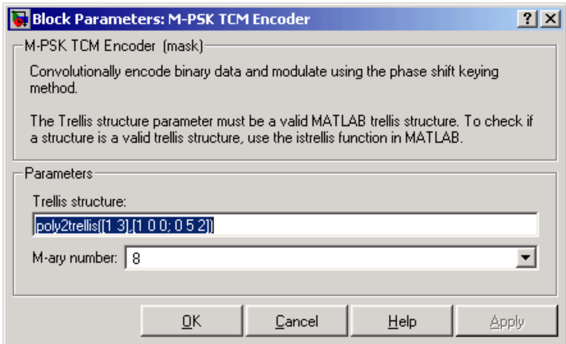
Signal Constellations

The trellis-coded modulation technique partitions the constellation into subsets called cosets, so as to maximize the minimum distance between pairs of points in each coset. This block internally forms a valid partition based on the value you choose for the **M-ary number** parameter.

The figure below shows the labeled set-partitioned signal constellation that the block uses when **M-ary number** is 8. For constellations of other sizes, see [1].



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.

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.

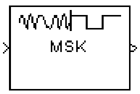
[2] Proakis, John G., Digital Communications, Fourth edition, New York, McGraw-Hill, 2001

MSK Demodulator Baseband

Purpose Demodulate MSK-modulated data

Library CPM, in Digital Baseband sublibrary of Modulation

Description



The MSK Demodulator Baseband block demodulates a signal that was modulated using the minimum shift keying method. The input is a baseband representation of the modulated signal. The **Phase offset** parameter is the initial phase of the modulated waveform.

Traceback Length and Output Delays

Internally, this block creates a trellis description of the modulation scheme and uses the Viterbi algorithm. The **Traceback length** parameter, D, in this block is the number of trellis branches used to construct each traceback path. D influences the output delay, which is the number of zero symbols that precede the first meaningful demodulated value in the output.

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

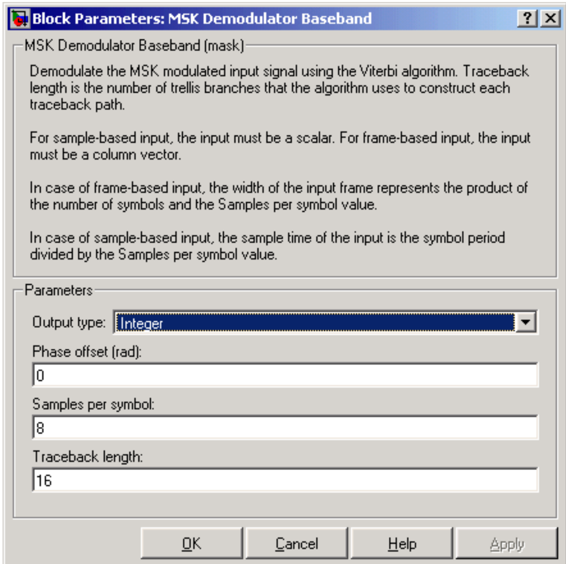
Inputs and Outputs

The input can be either a scalar or a frame-based column vector. 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 Using the Communications Blockset.

Dialog Box



Output type
: Determines whether the output consists of bipolar or binary values.

Phase offset (rad)
: The initial phase of the modulated waveform.

Samples per symbol
: The number of input samples that represent each modulated symbol.

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

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.

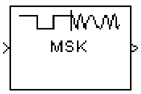
Purpose

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 **Modulation index** parameter times π 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.

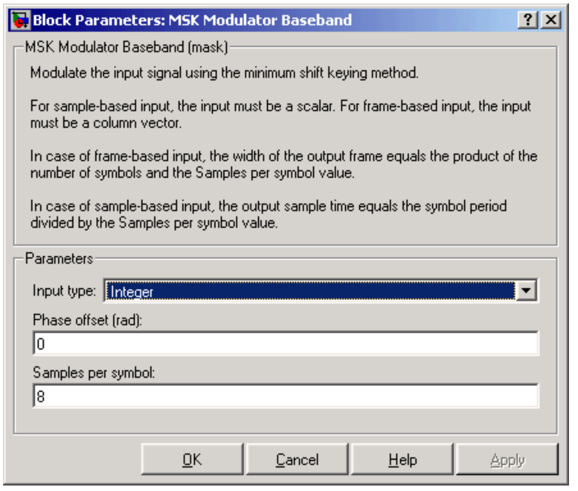
Input Attributes

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 Using the Communications Blockset.

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.

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

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.

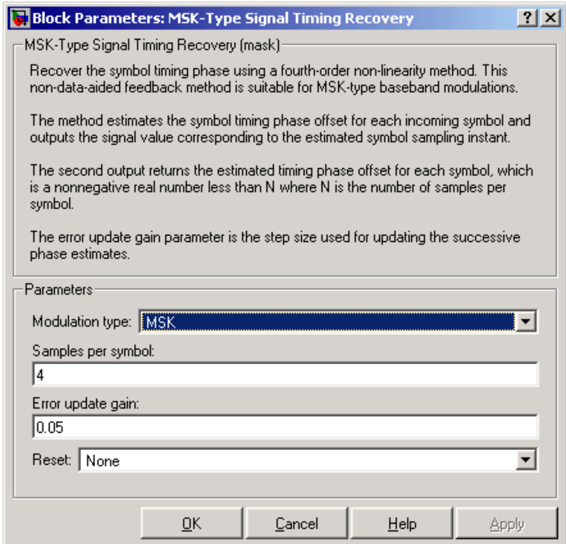
  The `Ph` output contains nonnegative real numbers less than N. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time or frame period of the `Ph` output is the same as that of the `Sym` output.

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

Delays

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

Dialog Box



Modulation type
: The type of modulation in the system. Choices are `MSK` and `GMSK`.

Samples per symbol
: The number of samples, N, that represent each symbol in the input signal. This must be greater than 1.

Error update gain
: A positive real number representing the step size that the block uses for updating successive phase estimates. Typically, this number is less than 1/N, which corresponds to a slowly varying phase.

Reset
: Determines whether and under what circumstances the block restarts the phase estimation process. Choices are `None`, `Every frame`, and `On nonzero input via port`. The last option causes the block to have a second input port, labeled `Rst`.

MSK-Type Signal Timing Recovery

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

$$\begin{aligned} e(k) = {} & (-1)^{D+1}\,\mathrm{Re}\{r^2(kT - T_s + d_{k-1})r^{*2}((k-1)T - T_s + d_{k-2})\} \\ & -(-1)^{D+1}\,\mathrm{Re}\{r^2(kT + T_s + d_{k-1})r^{*2}((k-1)T + T_s + d_{k-1})\} \end{aligned}$$

where

- r is the block's input signal
- T is the symbol period
- T_s is the sampling period
- * means complex conjugate
- d_k is the phase estimate for the 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 Using the Communications Blockset.

References

[1] D'Andrea, A. N., U. Mengali, and R. Reggiannini, "A Digital Approach to Clock Recovery in Generalized Minimum Shift Keying," *IEEE Transactions on Vehicular Technology*, Vol. 39, No. 3, August 1990, pp. 227-234.

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

See Also

Early-Late Gate Timing Recovery, Squaring Timing Recovery

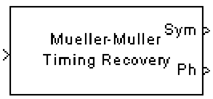
Purpose

Recover symbol timing phase using Mueller-Muller method

Library

Timing Phase Recovery sublibrary of Synchronization

Description



The Mueller-Muller Timing Recovery block recovers the symbol timing phase of the input signal using the Mueller-Muller method. This block implements a decision-directed, data-aided feedback method that requires prior recovery of the carrier phase.

Inputs

By default, the block has one input port. Typically, the input signal is the output of a receive filter that is matched to the transmitting pulse shape. The input must be a scalar or a frame-based column vector. The input uses N samples to represent each symbol, where N > 1 is the **Samples per symbol** parameter. If the input is frame-based, then its vector length is N*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.

- If the input signal is a sample-based scalar with sample time T/N, then the `Sym` output is a sample-based scalar with sample time T.

- The `Ph` output gives the phase estimate for each symbol in the input signal.

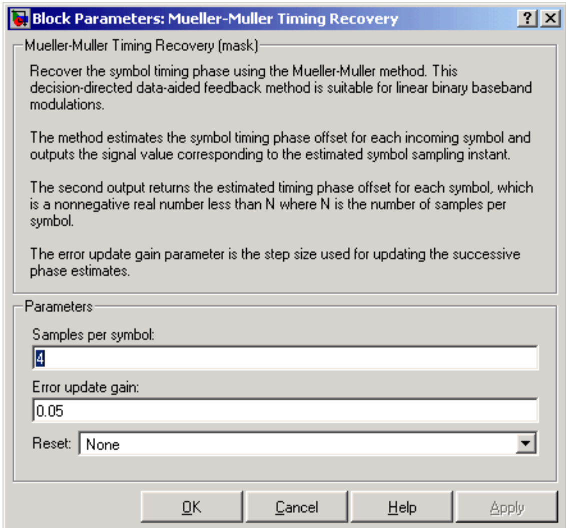
  The `Ph` output contains nonnegative real numbers less than N. Noninteger values for the phase estimate correspond to interpolated values that lie between two values of the input signal. The sample time or frame period of the `Ph` output is the same as that of the `Sym` output.

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

Delays

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

Dialog Box



Samples per symbol

The number of samples, N, that represent each symbol in the input signal. This must be greater than 1.

Error update gain

A positive real number representing the step size that the block uses for updating successive phase estimates. Typically, this number is less than 1/N, which corresponds to a slowly varying phase.

Reset

Determines whether and under what circumstances the block restarts the phase estimation process. Choices are `None`, `Every frame`, and `On nonzero input via port`. The last option causes the block to have a second input port, labeled `Rst`.

Algorithm

This block uses a timing error detector whose result for the kth symbol is e(k), given by

$$e(k) = \mathrm{Re}\{c_{k-1}^{*}y(kT + d_k) - c_k^{*}y((k-1)T + d_{k-1})\}$$

where

- y is the block's input signal
- c_k is the decision based on the sample value $y(kT+d_k)$
- T is the symbol period
- d_k is the phase estimate for the 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 Using the Communications Blockset.

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

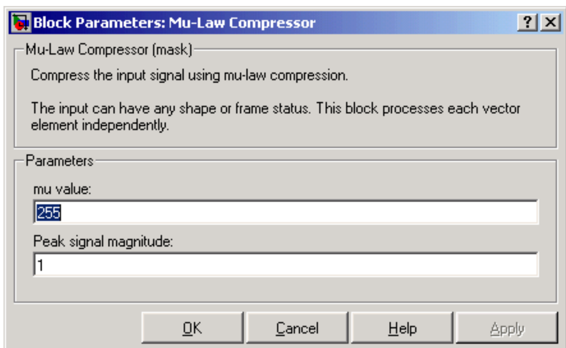
The Mu-Law Compressor block implements a μ-law compressor for the input signal. The formula for the μ-law compressor is

$$y = \frac{V \log(1 + \mu|x|/V)}{\log(1 + \mu)} \operatorname{sgn}(x)$$

where μ is the μ-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.

Pair Block

Mu-Law Expander

Mu-Law Compressor

See Also

A-Law Compressor

References

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

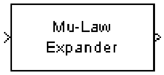
Purpose

Implement μ-law expander for source coding

Library

Source Coding

Description

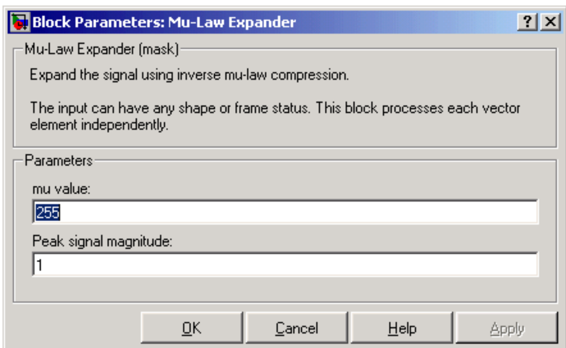


The Mu-Law Expander block recovers data that the Mu-Law Compressor block compressed. The formula for the μ-law expander, shown below, is the inverse of the compressor function.

$$x = \frac{V}{\mu}\left(e^{|y|\log(1+\mu)/V} - 1\right)\operatorname{sgn}(y)$$

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.

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.

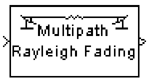
Purpose

Simulate multipath Rayleigh fading propagation channel

Library

Channels

Description



The Multipath Rayleigh Fading Channel block implements a baseband simulation of a multipath Rayleigh fading propagation channel. This block is useful for modeling mobile wireless communication systems. For details about fading channels, see the references listed below.

This block accepts only frame-based complex signals at its input. To work with sample-based inputs, use the `Frame conversion` block of the Signal Processing blockset to reformat the signal.

Relative motion between the transmitter and receiver causes Doppler shifts in the signal frequency. The Jakes PSD (power spectral density) determines the spectrum of the Rayleigh process.

Since a multipath channel reflects signals at multiple places, a transmitted signal travels to the receiver along several paths that may have different lengths and hence different associated time delays. Fading occurs when signals traveling along different paths interfere with each other. In the block's parameter dialog, the **Delay vector** specifies the time delay for each path. If the **Normalize gain vector to 0 dB overall gain** box is unchecked, then the **Gain vector** specifies the gain for each path. If the box is checked, then the block uses a multiple of **Gain vector** instead of the **Gain vector** itself, choosing the scaling factor so that the channel's effective gain considering all paths is 0 dB.

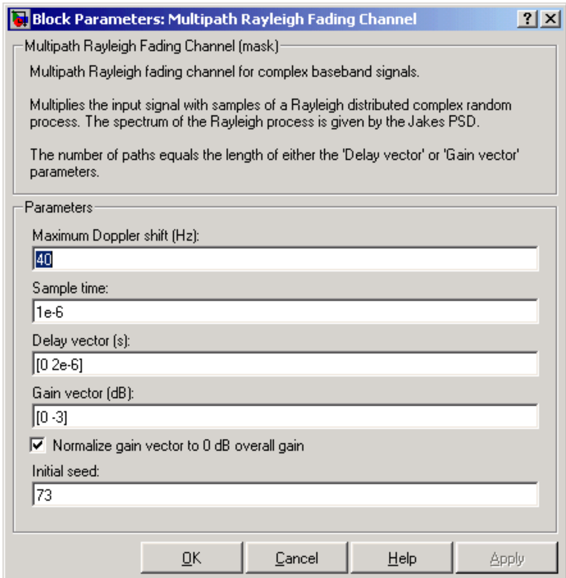
The number of paths is the length of **Delay vector** or **Gain vector**, whichever is larger. If both of these parameters are vectors, then they must have the same length; if exactly one of these parameters is a scalar, then the block expands it into a vector whose size matches that of the other **vector** parameter.

The block multiplies the input signal by samples of a Rayleigh-distributed complex random process. The scalar **Initial seed** parameter seeds the random number generator.

Double clicking this block during simulation or checking the block dialog's check-box labeled **Open channel visualization at start of**

simulation will plot the channel characteristics using the channel visualization tool. See in the Communications Toolbox User's Guide for details.

Dialog Box



Maximum Doppler shift (Hz)
: A positive scalar that indicates the maximum Doppler shift.

Delay vector (s)
: A vector that specifies the propagation delay for each path.

Gain vector (dB)
: A vector that specifies the gain for each path.

Normalize gain vector to 0 dB overall gain
: Checking this box causes the block to scale the **Gain vector** parameter so that the channel's effective gain (considering all paths) is 0 decibels.

Initial seed
The scalar seed for the Gaussian noise generator.

Open channel visualization at start of simulation
Checking this box will open the channel visualization tool when a simulation is started.

Complex path gains port
Checking this box will create a port that outputs the complex path gains data. This is an N by M multichannel frame, where N is the number of samples per frame and M is the number of discrete paths (number of delays).

Channel filter delay port
Checking this box will create a port that outputs the filter delay data.

Algorithm

This implementation is based on the direct form simulator described in Reference [1] below.

Some wireless applications, such as standard GSM (Global System for Mobile Communication) systems, prefer to specify Doppler shifts in terms of the speed of the mobile. If the mobile moves at speed v making an angle of θ with the direction of wave motion, then the Doppler shift is

$$f_{\mathrm{d}} = (vf/c)\cos\theta$$

where f is the transmission carrier frequency and c is the speed of light. The Doppler frequency is the maximum Doppler shift arising from motion of the mobile.

See Also

Rayleigh Noise Generator, Rician Fading Channel

References

[1] Jeruchim, Michel C., Balaban, Philip, and Shanmugan, K. Sam, *Simulation of Communication Systems*, Second edition, New York, Kluwer Academic/Plenum, 2000.

[2] Jakes, William C., ed. *Microwave Mobile Communications*, New York, IEEE Press, 1974.

[3] Lee, William C. Y., *Mobile Communications Design Fundamentals*, 2nd Ed. New York, Wiley, 1993.

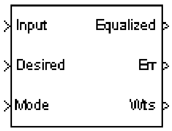
Purpose

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.

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 Using the Communications Blockset.
- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

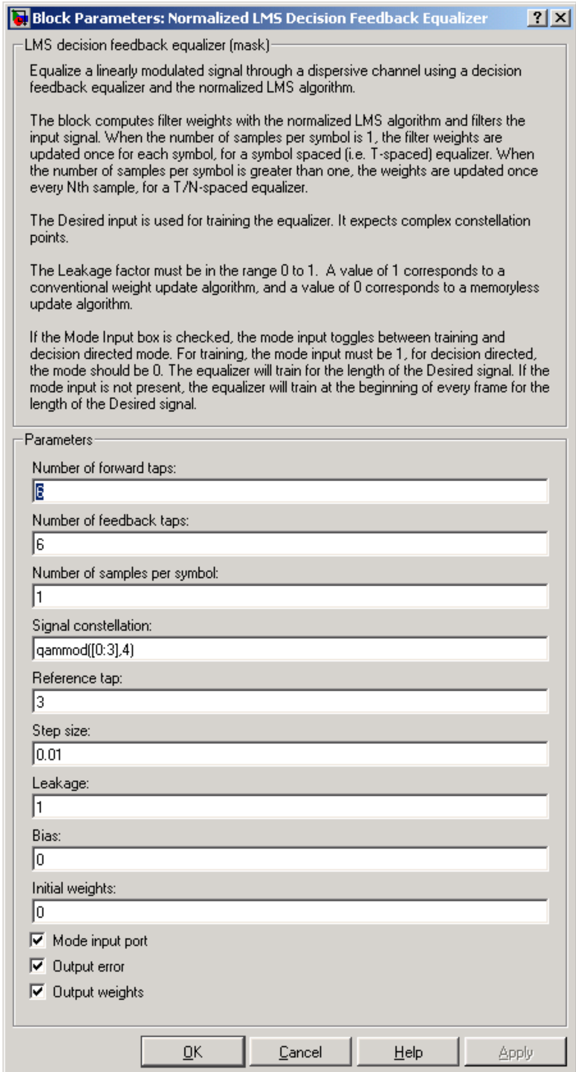
Equalizer Delay

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


Block Parameters: Normalized LMS Decision Feedback Equalizer
LMS decision feedback equalizer (mask)
Equalize a linearly modulated signal through a dispersive channel using a decision feedback equalizer and the normalized LMS algorithm.
The block computes filter weights with the normalized LMS algorithm and filters the input signal. When the number of samples per symbol is 1, the filter weights are updated once for each symbol, for a symbol spaced (i.e. T-spaced) equalizer. When the number of samples per symbol is greater than one, the weights are updated once every Nth sample, for a T/N-spaced equalizer.
The Desired input is used for training the equalizer. It expects complex constellation points.
The Leakage factor must be in the range 0 to 1. A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.
If the Mode Input box is checked, the mode input toggles between training and decision directed mode. For training, the mode input must be 1, for decision directed, the mode should be 0. The equalizer will train for the length of the Desired signal. If the mode input is not present, the equalizer will train at the beginning of every frame for the length of the Desired signal.
Parameters
Number of forward taps:
6
Number of feedback taps:
6
Number of samples per symbol:
1
Signal constellation:
qammod([0:3],4)
Reference tap:
3
Step size:
0.01
Leakage:
1
Bias:
0
Initial weights:
0
Mode input port
Output error
Output weights
OK
Cancel
Help
Apply

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

Normalized LMS Linear Equalizer

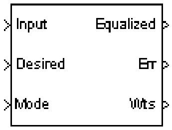
Purpose

Equalize using linear equalizer that updates weights with normalized LMS algorithm

Library

Equalizers

Description



The Normalized LMS Linear Equalizer block uses a linear equalizer and the normalized LMS algorithm to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the normalized LMS algorithm to update the weights, once per symbol. If the **Number of samples per symbol** parameter is `1`, then the block implements a symbol-spaced equalizer; otherwise, the block implements a fractionally spaced equalizer.

Input and Output Signals

The port labeled `Input` receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled `Desired` receives a training sequence whose length is less than or equal to the number of symbols in the `Input` signal. Valid training symbols are those listed in the **Signal constellation** vector.

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 Using the Communications Blockset.
- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

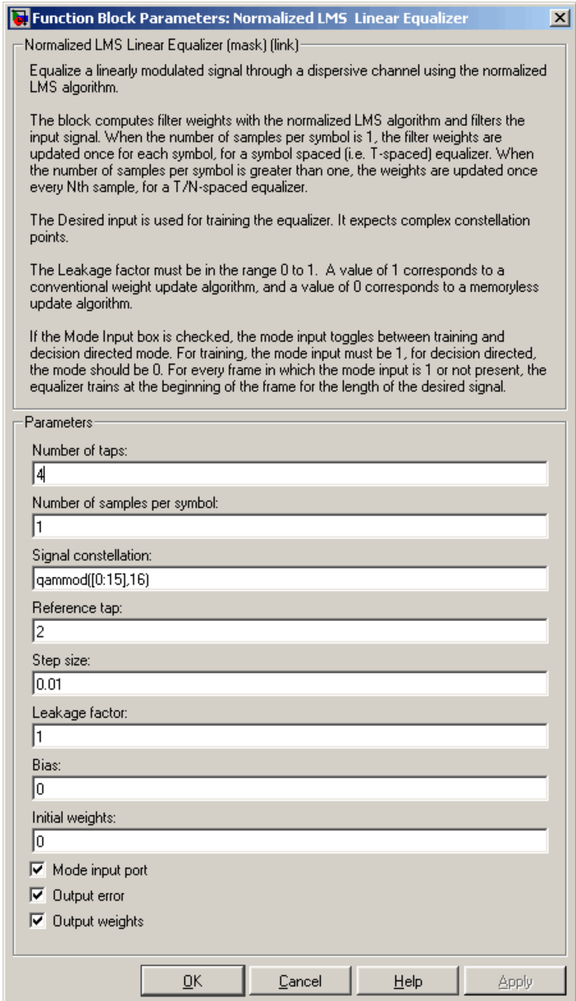
Equalizer Delay

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

Purpose

Demodulate OQPSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

Description

The OQPSK Demodulator Baseband block demodulates a signal that was modulated using the offset quadrature phase shift keying method. The input is a baseband representation of the modulated signal.

The input must be a discrete-time complex signal. The input can be either a scalar or a frame-based column vector. The block accepts the input data types `single` and `double`.

If the **Output type** parameter is set to `Integer`, then the block outputs integers between 0 and 3. If the **Output type** parameter is set to `Bit`, then the block outputs binary representations of such integers, in a binary-valued vector whose length is an even number.

The input symbol period is half the period of each output integer or bit pair. The constellation used to map bit pairs to symbols is on the reference page for theOQPSK Modulator Baseband block.

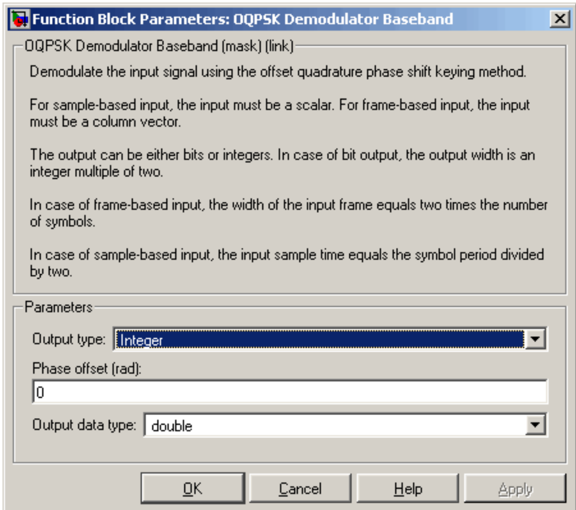
Frame-Based Inputs

If the input is a frame-based column vector, then the block processes several integers or several pairs of bits, in each time step. In this case, the output sample time equals the input sample time, even though the symbol period is half the output period.

Delays

The modulator-demodulator pair incurs a delay, as described in "Delays in Digital Modulation".

Dialog Box



Output type
: Determines whether the output consists of integers or pairs of bits.

Phase offset (rad)
: The amount by which the phase of the zeroth point of the signal constellation is shifted from $\pi/4$.

Output data type
: For integer inputs, this block can output the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. For bit inputs, output can be `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, or `double`.

Pair Block

OQPSK Modulator Baseband

See Also

QPSK Demodulator Baseband

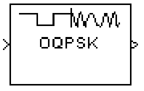
Purpose

Modulate using offset quadrature phase shift keying method

Library

PM, in Digital Baseband sublibrary of Modulation

Description



The OQPSK Modulator Baseband block modulates using the offset quadrature phase shift keying method. The output is a baseband representation of the modulated signal.

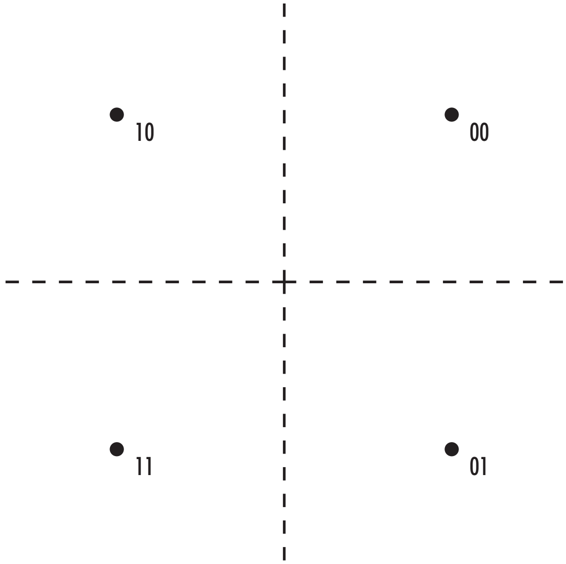
If the **Input type** parameter is set to `Integer`, then valid input values are 0, 1, 2, and 3. In this case, the input can be either a scalar or a frame-based column vector.

If the **Input type** parameter is set to `Bit`, then the input must be a binary-valued vector. In this case, the input can be either a vector of length two or a frame-based column vector whose length is an even integer.

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

The symbol period is half the input period. The first output symbol is an initial condition of zero that is unrelated to the input values.

The constellation used to map bit pairs to symbols is in the figure below. If the block's **Phase offset** parameter is nonzero, then this constellation is rotated by that parameter value.



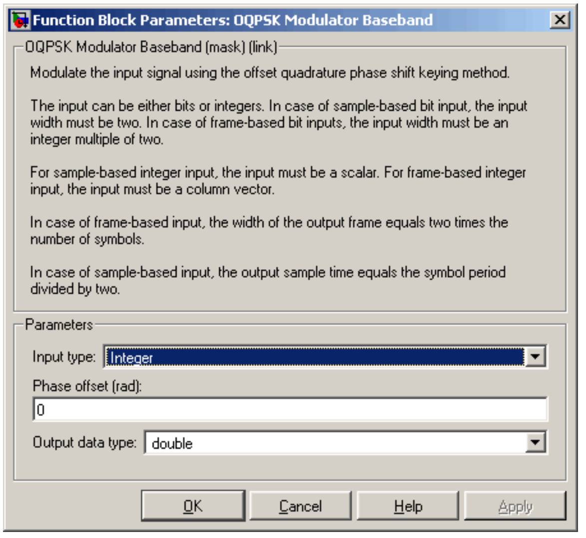
Frame-Based Inputs

If the input is a frame-based column vector, then the block processes several integers or several pairs of bits in each time step. In this case, the output sample time equals the input sample time, even though the period of each output symbol is half the period of each integer or bit pair in the input.

Delays

The modulator-demodulator pair incurs a delay, as described in "Delays in Digital Modulation".

Dialog Box



Input type
: Indicates whether the input consists of integers or pairs of bits.

Phase offset (rad)
: The amount by which the phase of the zeroth point of the signal constellation is shifted from $\pi/4$.

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

Pair Block

OQPSK Demodulator Baseband

See Also

QPSK Modulator Baseband

OVSF Code Generator

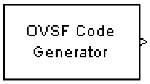
Purpose

Generate orthogonal variable spreading factor (OVSF) code from set of orthogonal codes

Library

Spreading Codes

Description



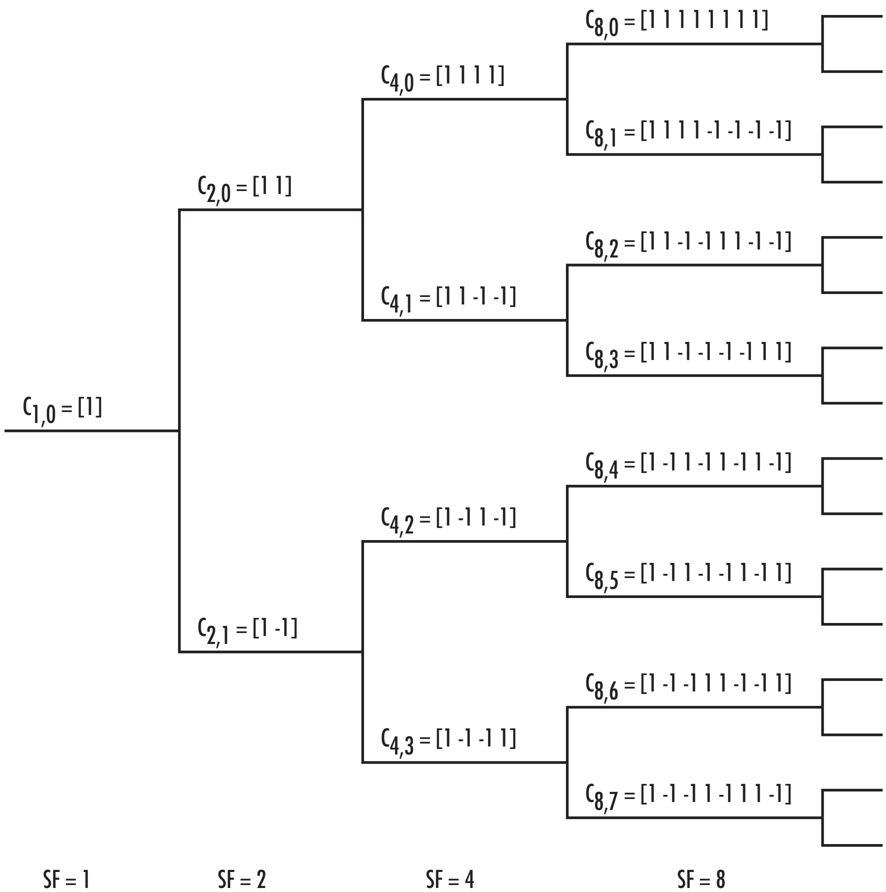
The OVSF Code Generator block generates an OVSF code from a set of orthogonal codes. OVSF codes were first introduced for 3G communication systems. OVSF codes are primarily used to preserve orthogonality between different channels in a communication system.

OVSF codes are defined as the rows of an N-by-N matrix, C_N, which is defined recursively as follows. First, define $C_1 = [1]$. Next, assume that C_N is defined and let $C_N(k)$ denote the 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_N is only defined for N a power of 2. It follows by induction that the rows of C_N are orthogonal.

The OVSF codes can also be defined recursively by a tree structure, as shown in the following figure.



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^r. The **Code index** specifies how far down the column of the tree at depth r the code appears, counting from 0 to N - 1. For $C_{N,k}$ in the preceding diagram, N is the **Spreading factor** and k is the **Code index**.

You can recover the code from the **Spreading factor** and the **Code index** as follows. Convert the **Code index** to the corresponding binary number, and then add 0s to the left, if necessary, so that the resulting binary sequence $x_1\, x_2 \ldots x_r$ has length r, where r is the logarithm base 2 of the **Spreading factor**. This sequence describes the path from the root to the code. The path takes the upper branch from the code at depth i if $x_i = 0$, and the lower branch if $x_i = 1$.

To reconstruct the code, recursively define a sequence of codes C_i for as follows. Let C_0 be the root [1]. Assuming that C_i has been defined, for $i < r$, define C_{i+1} by

$$C_{i+1} = \begin{cases} C_i C_i & \text{if } x_i = 0 \\ C_i(-C_i) & \text{if } x_i = 1 \end{cases}$$

The code C_N has the specified **Spreading factor** and **Code index**.

For example, to find the code with **Spreading factor** `16` and **Code index** `6`, do the following:

1. Convert 6 to the binary number 110.

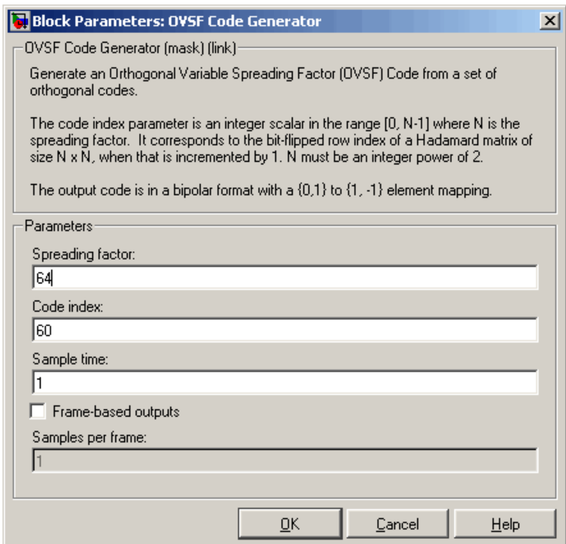
2. Add one 0 to the left to obtain 0110, which has length $4 = \log_2 16$.

3. Construct the sequences C_i according to the following table.

i	x_i	C_i
0		C_0 = [1]
1	0	$C_1 = C_0\ C_0$ = [1] [1]
2	1	$C_2 = C_1\ \text{-}C_1$ = [1 1] [-1 -1]
3	1	$C_3 = C_2\ \text{-}C_2$ = [1 1 -1 -1] [-1 -1 1 1]
4	0	$C_4 = C_3\ C_3$ = [1 1 -1 -1 -1 -1 1 1] [1 1 -1 -1 -1 -1 1 1]

The code C_4 has **Spreading factor** 16 and **Code index** 6.

Dialog Box



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

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.

See Also

Hadamard Code Generator, Walsh Code Generator

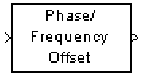
Purpose

Apply phase and frequency offsets to complex baseband signal.

Library

RF Impairments

Description



The Phase/Frequency Offset block first applies a phase offset and then a frequency offset to a complex, baseband 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**:

You can view the implementation of the phase or frequency offsets by double-clicking the Phase Offset or Frequency Offset subsystems under the mask.

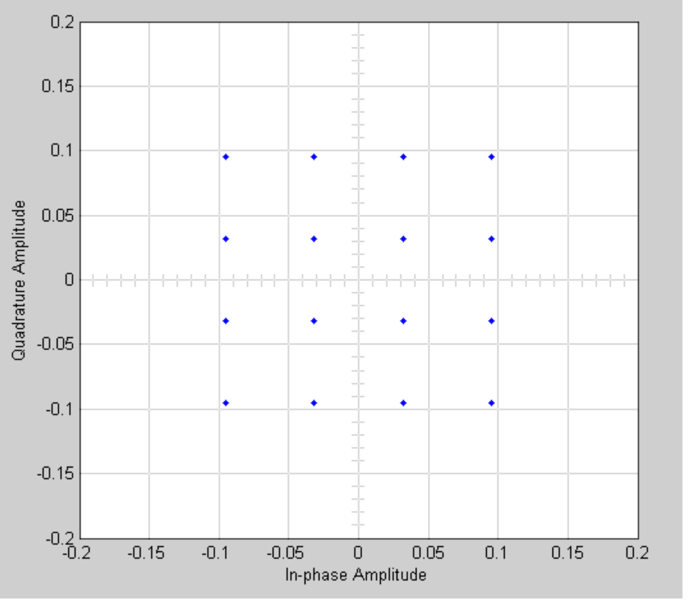
Phase Offset

The block applies a phase offset to the input signal, specified by the **Phase offset (deg)** parameter.

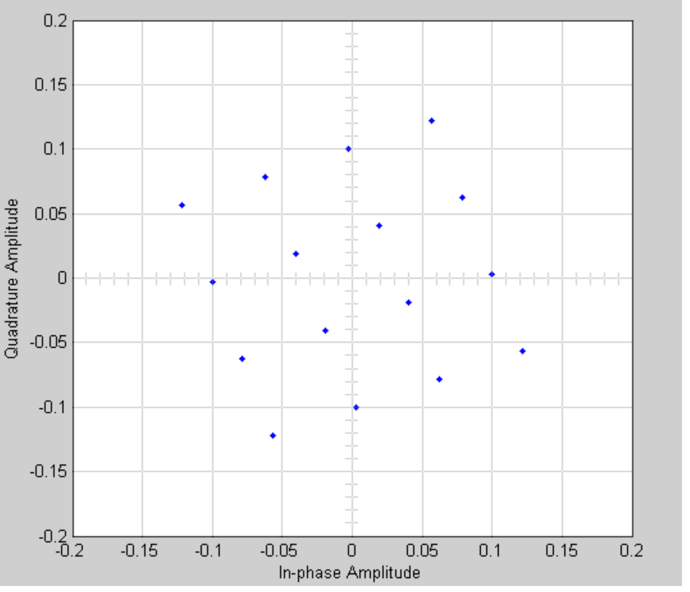
Frequency Offset

The block applies a frequency offset to the signal that is specified by the **Frequency offset (Hz)** parameter.

The effects of changing the block's parameters are illustrated by the following scatter plots of a signal modulated by 16-ary quadrature amplitude modulation (QAM). The usual 16-ary QAM constellation without the effect of the Phase/Frequency Offset block is shown in the first scatter plot:

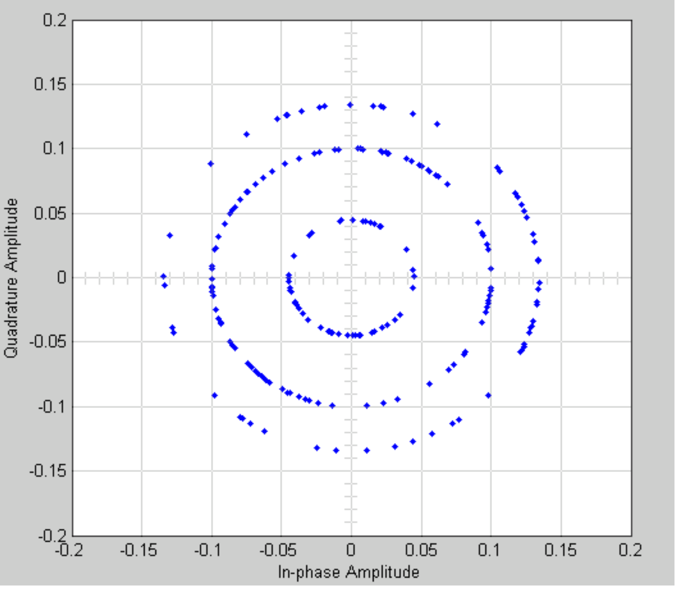


The following figure shows a scatter plot of an output signal, modulated by 16-ary QAM, from the Phase/Frequency Offset block with **Phase offset (deg)** set to 20 and **Frequency offset (Hz)** set to 0:



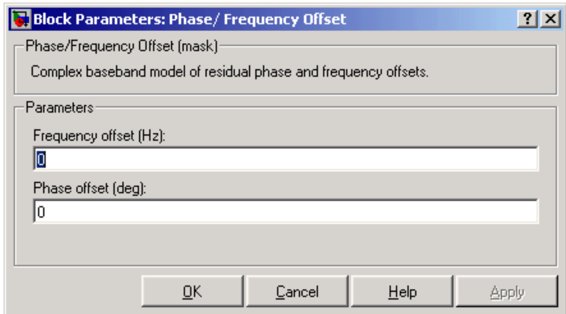
Observe that each point in the constellation is rotated by a 20 degree angle counterclockwise.

If you set **Frequency offset (Hz)** to `2` and **Phase offset (deg)** 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:



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.



Dialog Box

Frequency offset (hz)
: Scalar specifying the frequency offset in Hertz.

Phase offset (deg)
: Scalar specifying the phase offset in degrees.

See Also

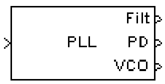
Phase Noise

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.

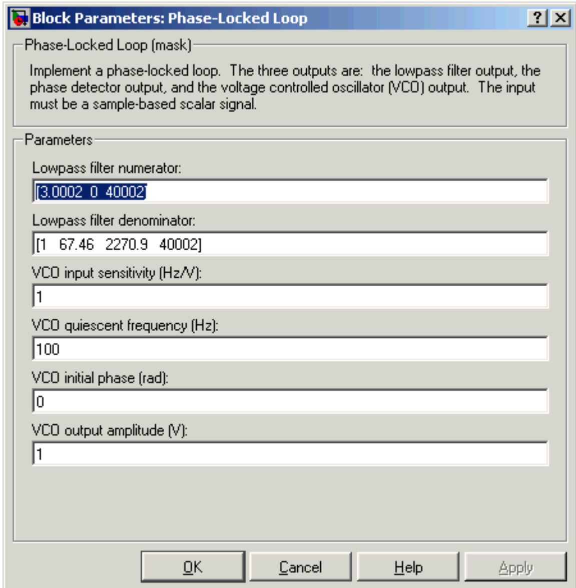
  To design a filter, you can use functions such as `butter`, `cheby1`, and `cheby2` in the Signal Processing Toolbox. The default filter is a Chebyshev type II filter whose transfer function arises from the command below.

  ```
  [num, den] = cheby2(3,40,100,'s')
  ```
- A voltage-controlled oscillator (VCO). You specify characteristics of the VCO using the **VCO quiescent frequency**, **VCO initial phase**, and **VCO output amplitude** parameters.

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

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

Dialog Box



Lowpass filter numerator

The numerator of the lowpass filter's transfer function, represented as a vector that lists the coefficients in order of descending powers of *s*.

Lowpass filter denominator

The denominator of the lowpass filter's transfer function, represented as a vector that lists the coefficients in order of descending powers of *s*.

VCO input sensitivity (Hz/V)

This value scales the input to the VCO and, consequently, the shift from the **VCO quiescent frequency** value. The units of **VCO input sensitivity** are Hertz per volt.

VCO quiescent frequency (Hz)

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

VCO initial phase (rad)
The initial phase of the VCO signal.

VCO output amplitude
The amplitude of the VCO signal.

See Also

Baseband PLL, Linearized Baseband PLL, Charge Pump PLL

References

For more information about phase-locked loops, see the works listed in "Selected Bibliography for Synchronization" in Using the Communications Blockset.

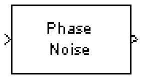
Purpose

Apply receiver phase noise to complex baseband signal

Library

RF Impairments

Description

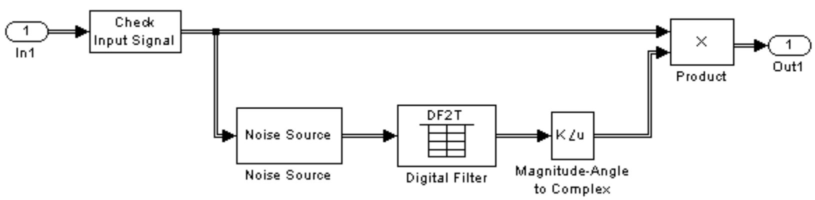


The Phase Noise block appies phase noise to a complex, baseband signal. The block applies the phase noise as follows:

1. Generates additive white Gaussian noise (AWGN) and filters it with a digital filter.

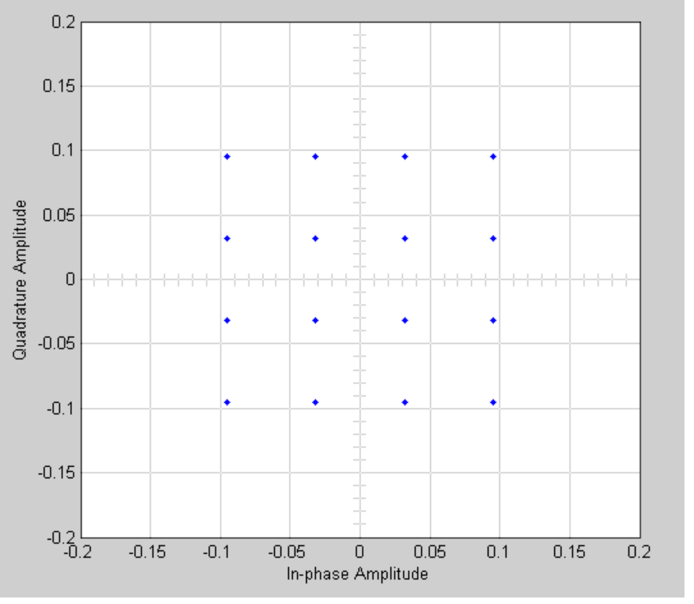
2. Adds the resulting noise to the angle component of the input signal.

You can view the block's implementation of phase noise by right-clicking on the block and selecting **Look under mask**. This displays the following figure:

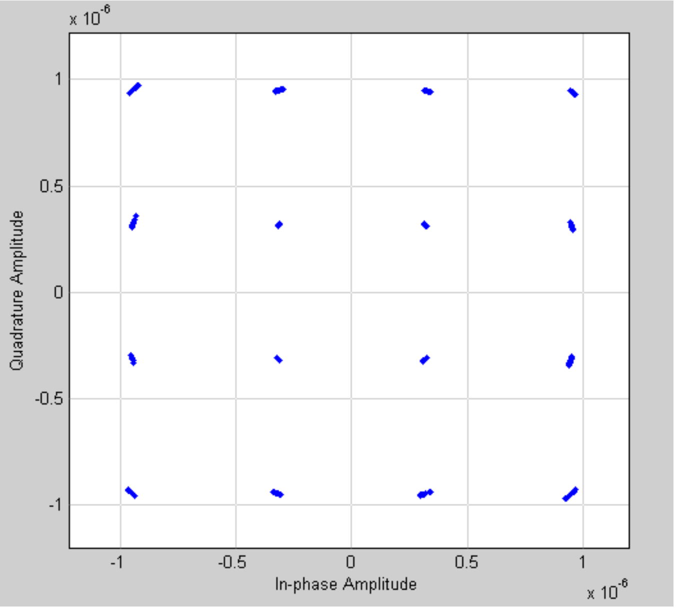


You can view the construction of the Noise Source subsystem by double-clicking it.

The effects of changing the block's parameters are illustrated by the following scatter plots of a signal modulated by 16-ary quadrature amplitude modulation (QAM). The usual 16-ary QAM constellation without distortion is shown in the first scatter plot:



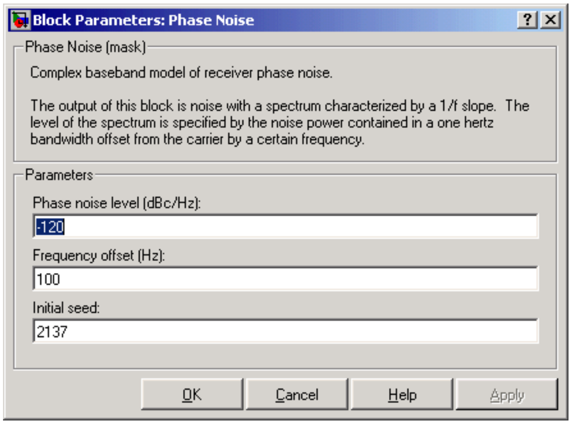
The following figure shows a scatter plot of an output signal, modulated by 16-ary QAM, from the Phase Noise block with **Phase noise level (dBc/Hz)** set to `-70` and **Frequency offset (Hz)** set to `100`:



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

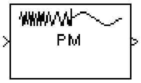
Purpose

Demodulate PM-modulated data

Library

Analog Passband Modulation, in Modulation

Description



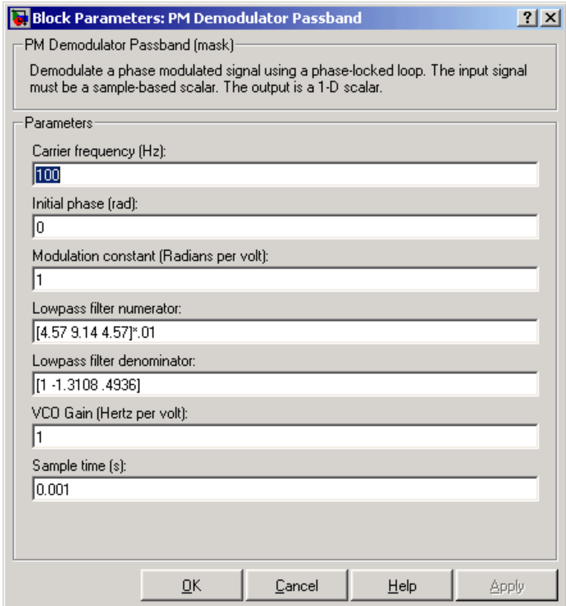
The PM Demodulator Passband block demodulates a signal that was modulated using phase modulation. The input is a passband representation of the modulated signal. Both the input and output signals are real sample-based scalar signals.

This block uses a phase-locked loop containing a voltage-controlled oscillator (VCO). The **VCO Gain** parameter specifies the input sensitivity of the VCO.

In the course of demodulating, the block uses a filter whose transfer function is described by the **Lowpass filter numerator** and **Lowpass filter denominator** parameters.

By the Nyquist sampling theorem, the reciprocal of the **Sample time** parameter must exceed twice the **Carrier frequency** parameter.

Dialog Box



Carrier frequency (Hz)
: The carrier frequency in the corresponding PM Modulator Passband block.

Initial phase (rad)
: The carrier signal's initial phase in the corresponding PM Modulator Passband block.

Modulation constant (Radians per volt)
: The modulation constant in the corresponding PM Modulator Passband block.

Lowpass filter numerator
: The numerator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s.

Lowpass filter denominator
The denominator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s. For an FIR filter, set this parameter to `1`.

VCO Gain (Hertz per volt)
The input sensitivity of the voltage-controlled oscillator.

Sample time
The sample time of the output signal.

Pair Block

PM Modulator Passband

PM Modulator Passband

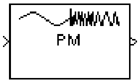
Purpose

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.

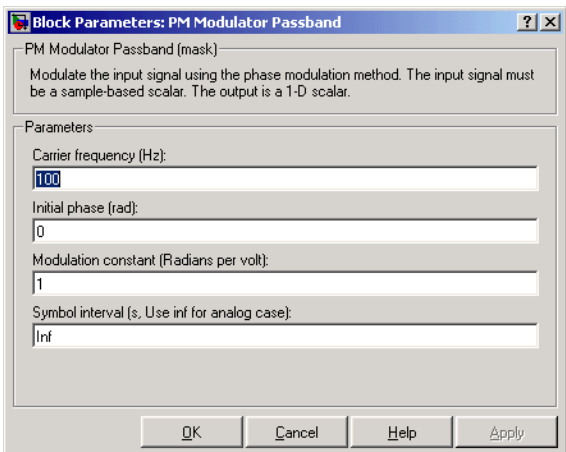
If the input is $u(t)$ as a function of time t, then the output is

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

where f_c is the **Carrier frequency** parameter, θ 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.

Dialog Box



Carrier frequency (Hz)
The frequency of the carrier.

Initial phase (rad)
The initial phase of the carrier in radians.

Modulation constant (Radians per volt)
The modulation constant K_c.

Symbol interval
Typically set to `Inf`.

Pair Block

PM Demodulator Passband

PN Sequence Generator

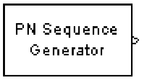
Purpose

Generate pseudonoise sequence

Library

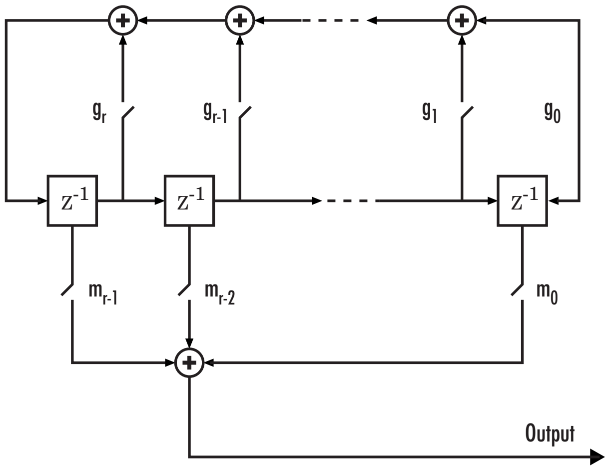
Sequence Generators sublibrary of Comm Sources

Description



The PN Sequence Generator block generates a sequence of pseudorandom binary numbers. 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*, $g_r z^r + g_{r-1} z^{r-1} + g_{r-2} z^{r-2} + ... + g_0$. 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.

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 $p(z) = z^8 + z^2 + 1$.

Quantity	Example 1	Example 2
Generator polynomial	`g1 = [1 0 0 0 0 0 1 0 1]`	`g2 = [8 2 0]`

Quantity	Example 1	Example 2
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]`

The **Shift** parameter 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 the **Shift** parameter. 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 **Shift (or mask)** to 0, versus a positive integer d, is shown in the following table.

	T = 0	T = 1	T = 2	...	T = d	T = d+1
Shift = 0	x_0	x_1	x_2	...	x_d	x_{d+1}
Shift = d	x_d	x_{d+1}	x_{d+2}	...	x_{2d}	x_{2d+1}

Alternatively, you can set the **Shift** parameter to a binary vector, corresponding to a polynomial in *z*, $m_{r-1}z^{r-1} + m_{r-2}z^{r-2} + \ldots + m_1z + m_0$, 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 the **Shift (or mask)** parameter is implemented when you specify it as a mask vector. The default setting for the **Shift (or**

mask) parameter is `[0 0 0 1]`, which corresponds to d = 0. You can calculate the mask vector using the Communications Toolbox function `shift2mask`.

You can use an external signal to reset the values of the internal shift register to the initial state by selecting the **Reset on nonzero input** check box. This creates an input port for the external signal in the PN Sequence Generator block. The way the block resets the internal shift register depends on whether its output signal and the reset signal are sample-based or frame-based. The following example demonstrates the possible alternatives.

Example: Resetting a Signal

Suppose that the PN Sequence Generator block outputs `[1 0 0 1 1 0 1 1]` when there is no reset. You then select the **Reset on nonzero input** check box and input a reset signal [0 0 0 1]. The following table shows three possibilities for the properties of the reset signal and the PN Sequence Generator block.

Reset Signal Properties	PN Sequence Generator block	Reset Signal, Output Signal
Sample-based **Sample time** = 1	Sample-based **Sample time** = 1	Reset 0 0 0 1 1 0 0 1 0 0 1 1 0 1 1

Reset Signal Properties	PN Sequence Generator block	Reset Signal, Output Signal
Frame-based **Sample time** =1 **Samples per frame** = 2	Frame-based **Sample time** = 1 **Samples per frame** = 2	Reset 0 0 0 $\boxed{1}$ 1 0 0 $\boxed{1}$ 0 0 1 1 0 1 1
Sample-based **Sample time** = 2 **Samples per frame** = 1	Frame-based **Sample time** = 1 **Samples per frame** = 2	Reset 0 0 0 $\boxed{1}$ 1 0 0 1 0 0 $\boxed{1}$ 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.

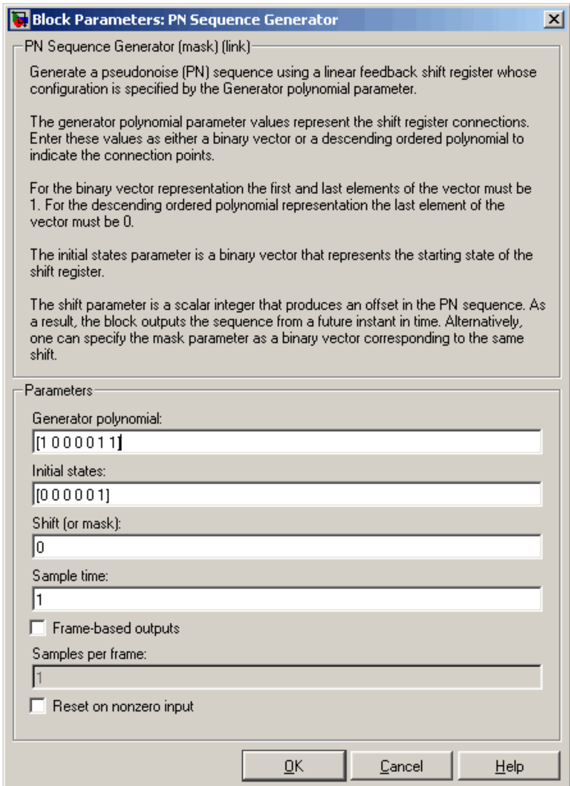
Sequences of Maximum Length

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.

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]

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.

Shift (or mask)
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.

See Also

Kasami Sequence Generator, Scrambler

References

[1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.

[2] Lee, J. S., and L. E. Miller, *CDMA Systems Engineering Handbook*, Artech House, 1998.

[3] Golomb, S.W., *Shift Register Sequences*, Aegean Park Press, 1967.

Poisson Integer Generator

Purpose

Generate Poisson-distributed random integers

Library

Data Sources sublibrary of Comm Sources

Description

The Poisson Integer Generator block generates random integers using a Poisson distribution. The probability of generating a nonnegative integer k is

$$\lambda^k \exp(-\lambda)/(k!)$$

where λ is a positive number known as the Poisson parameter.

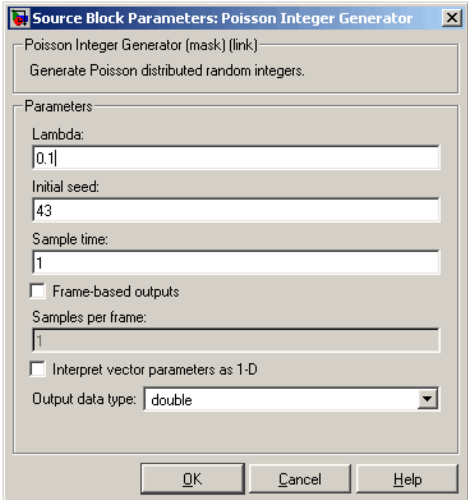
You can use the Poisson Integer Generator to generate noise in a binary transmission channel. In this case, the Poisson parameter **Lambda** should be less than 1, usually much less.

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 Using the Communications Blockset 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 λ. 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.

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)

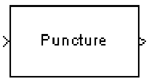
Purpose

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.

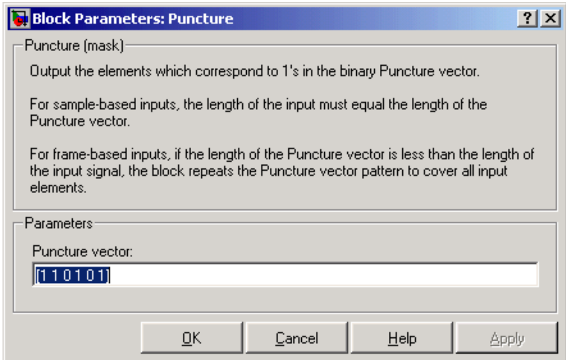
Frame-Based Processing

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

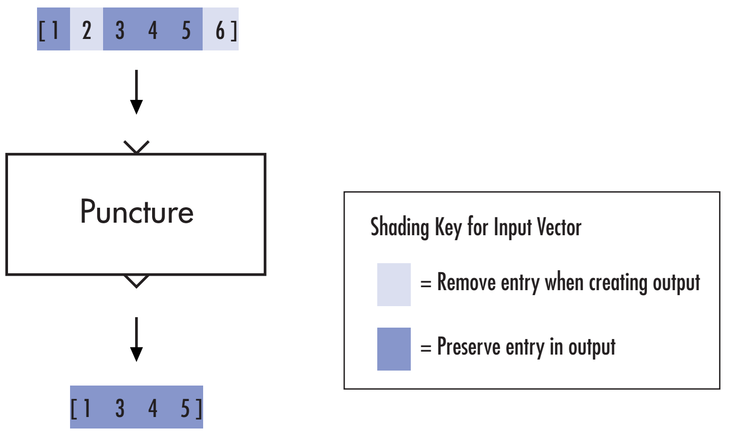
A binary vector whose pattern of 0s (1s) indicates which elements of the input the block should remove (preserve).

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.



See Also

Insert Zero

QPSK Demodulator Baseband

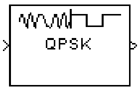
Purpose

Demodulate QPSK-modulated data

Library

PM, in Digital Baseband sublibrary of Modulation

Description



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

The input must be a discrete-time complex signal. The input can be either a scalar or a frame-based column vector. The block accepts the input data types `single` and `double`.

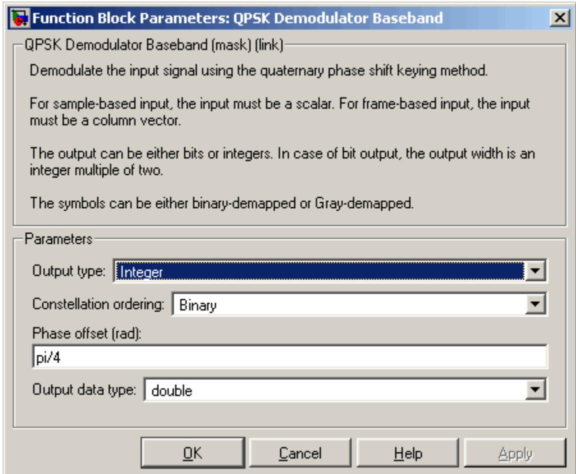
If the **Output type** parameter is set to `Integer`, then the block maps the point

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

to m, where θ is the **Phase offset** 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 theQPSK Modulator Baseband block shows the signal constellations 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 offset (rad)

The phase of the zeroth point of the signal constellation.

Output data type

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

Pair Block

QPSK Modulator Baseband

See Also

M-PSK Demodulator Baseband, BPSK Demodulator Baseband, DQPSK Demodulator Baseband

QPSK Modulator Baseband

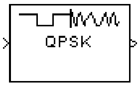
Purpose

Modulate using the 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

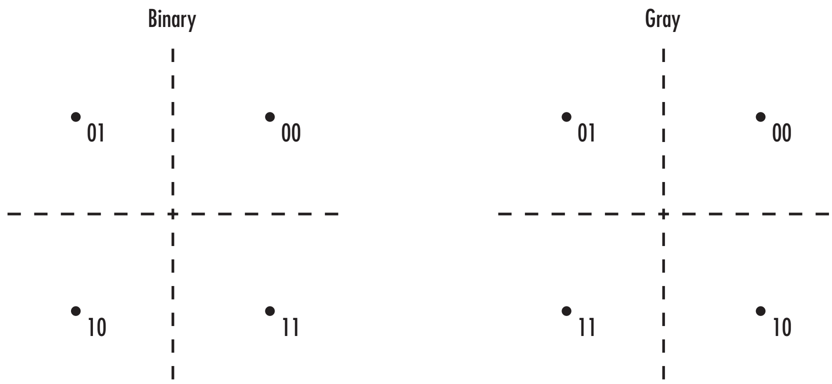
If the **Input type** parameter is set to `Integer`, then valid input values are 0, 1, 2, and 3. If the input is m, then the output symbol is

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

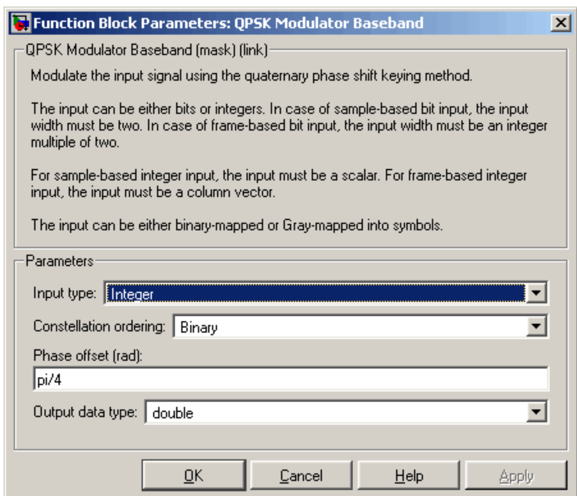
where θ is the **Phase offset** parameter. In this case, the input can be either a scalar or a frame-based column vector.

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

If the **Input type** parameter is set to `Bit`, then the input contains pairs of binary values. The input can be either a vector of length two 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 figure below, depending on whether the **Constellation ordering** parameter is set to `Binary` or `Gray`.



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 offset (rad)
The phase of the zeroth point of the signal constellation.

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

Pair Block

QPSK Demodulator Baseband

See Also

M-PSK Modulator Baseband, BPSK Modulator Baseband, DQPSK Modulator Baseband

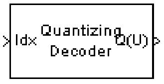
Purpose

Decode quantization index according to codebook

Library

Source Coding

Description

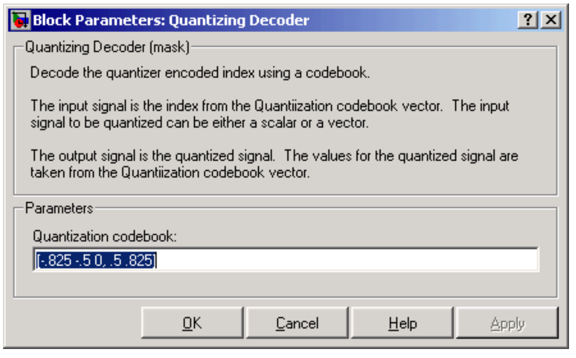


The Quantizing Decoder block converts quantization indices to the corresponding codebook values. The **Quantization codebook** parameter, a vector of length N, prescribes the possible output values. If the input is an integer k between 0 and N-1, then the output is the (k+1)st element of **Quantization codebook**.

The input can be either a scalar or a vector. The input must be a discrete-time signal. This block processes each vector element independently.

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



Quantization codebook

A real vector that prescribes the output value corresponding to each nonnegative integer of the input.

Quantizing Decoder

Pair Block Quantizing Encoder

See Also Scalar Quantizer (Signal Processing Blockset)

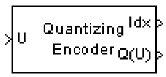
Purpose

Quantize signal using partition and codebook

Library

Source Coding

Description



The Quantizing Encoder block quantizes the input signal according to the **Partition** vector and encodes the input signal according to the **Codebook** vector. The input signal can be either a scalar or a vector. This block processes each vector element independently.

The first output is the quantization index. The second output is the quantized signal. The values for the quantized signal are taken from the **Codebook** vector.

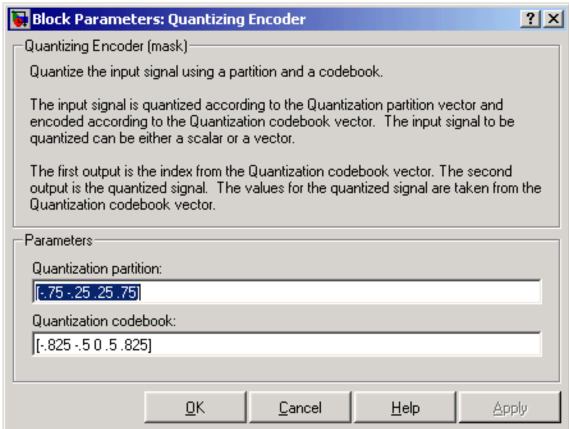
The **Quantization partition** parameter, P, is a real vector of length n whose entries are in strictly ascending order. The quantization index (second output signal value) corresponding to an input value of x is

- 0 if x P(1)
- m if P(m) < x P(m+1)
- n if P(n) < x

The **Quantization codebook** parameter, whose length is n+1, prescribes a value for each partition in the quantization. The first element of **Quantization codebook** is the value for the interval between negative infinity and the first element of P. The second output signal from this block contains the quantization of the input signal based on the quantization indices and prescribed values.

You can use the function `lloyds` in the 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.

Pair Block

Quantizing Decoder

See Also

Scalar Quantizer (Signal Processing Blockset), `lloyds` (Communications Toolbox)

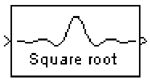
Purpose

Filter input signal, possibly downsampling, using raised cosine FIR filter

Library

Comm Filters

Description



The Raised Cosine Receive Filter block filters the input signal using a normal raised cosine FIR filter or a square root raised cosine FIR filter. It also downsamples the filtered signal if you set the **Output mode** parameter to `Downsampling`. The block's icon shows the filter's impulse response."

Characteristics of the Filter

Characteristics of the raised cosine filter are the same as in theRaised 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).

Downsampling the Filtered Signal

To have the block downsample the filtered signal, set the **Output mode** parameter to `Downsampling`. If L is the **Downsampling factor** parameter value, then the block retains 1/L of the samples, choosing them as follows:

- If the **Sample offset** parameter is zero, then the block selects the samples of the filtered signal indexed by 1, L+1, 2*L+1, 3*L+1, etc.
- If the **Sample offset** parameter is a positive integer less than L, then the block initially discards that number of samples from the filtered signal and downsamples the remaining data as in the case above.

To preserve the entire filtered signal and avoid downsampling, set **Output mode** to `None`. This setting is appropriate, for example, when the output from the filter block forms the input to a timing phase recovery block such as Squaring Timing Recovery. The timing phase recovery block performs the downsampling in that case.

Input and Output Signals

The input signal must be a scalar or a frame-based column vector. Set the **Input sampling mode** parameter according to whether the input is sample-based or frame-based.

If **Output mode** is set to `None`, then the input and output signals share the same sampling mode, sample time, and vector length.

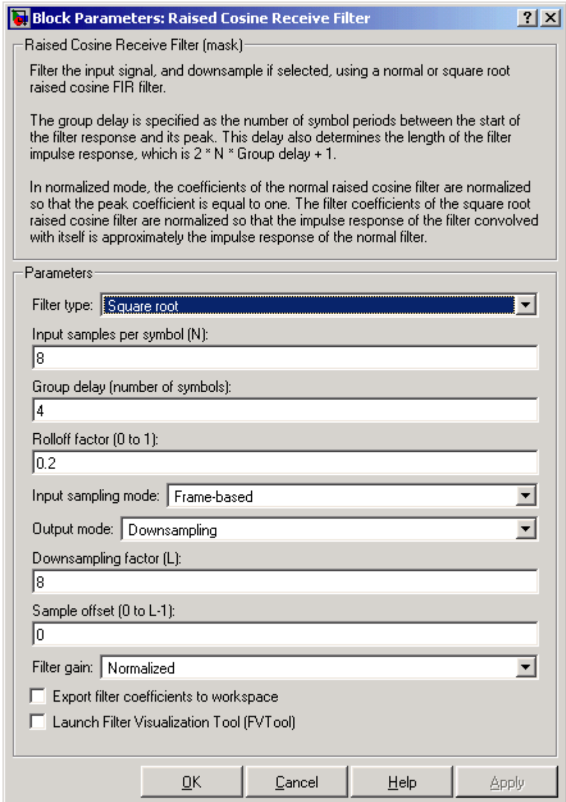
If **Output mode** is set to `Downsampling` and **Downsampling factor** is L, then L and the input sampling mode determine characteristics of the output signal:

- If the input is sample-based, then the output is sample-based and the output sample time is 1/L times the input sample time.
- If the input is frame-based, then the output is a frame-based vector whose length is 1/L times the length of the input vector. The output frame period equals the input frame period.

Exporting Filter Coefficients to the MATLAB Workspace

To examine or manipulate the coefficients of the filter that this block designs, select **Export filter coefficients to workspace**. Then set the **Coefficient variable name** parameter to the name of a variable that you want the block to create in the MATLAB workspace. Running the simulation causes the block to create the variable, overwriting any previous contents in case the variable already exists.

Dialog Box



Filter type

The type of raised cosine filter: `Square root` or `Normal`.

Input samples per symbol

An integer greater than 1 representing the number of samples per symbol in the input signal.

Group delay

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

Rolloff factor
The rolloff factor for the filter, a real number between 0 and 1.

Input sampling mode
The type of input signal: `Frame-based` or `Sample-based`.

Output mode
Determines whether or not the block downsamples the signal after filtering. Choices are `Downsampling` and `None`.

Downsampling factor
The factor by which the block downsamples the signal after filtering. This field appears only if **Output mode** is set to `Downsampling`.

Sample offset
The number of filtered samples the block discards before downsampling. This field appears only if **Output mode** is set to `Downsampling`.

Filter gain
Determines how the block scales the filter coefficients. Choices are `Normalized` and `User-specified`.

Linear amplitude filter gain
A positive scalar used to scale the filter coefficients. This field appears only if **Filter gain** is set to `User-specified`.

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

Coefficient variable name
The name of the variable to create in the MATLAB workspace. This field appears only if **Export filter coefficients to workspace** is selected.

Launch Filter Visualization Tool
If you check this box, then MATLAB launches the Filter Visualization Tool (`fvtool`) to analyze the raised cosine filter whenever you apply any changes to the block's parameters.

Pair Block

Raised Cosine Transmit Filter

See Also

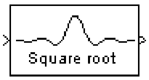
Gaussian Filter, `rcosine`, `rcosflt`

Raised Cosine Transmit Filter

Purpose Upsample and filter input signal using raised cosine FIR filter

Library Comm Filters

Description The Raised Cosine Transmit Filter block upsamples and filters the input signal using a normal raised cosine FIR filter or a square root raised cosine FIR filter. The block's icon shows the filter's impulse response."



Characteristics of the Filter

The **Filter type** parameter determines which type of filter the block uses; choices are `Normal` and `Square root`.

The impulse response of a normal raised cosine filter with rolloff factor R and symbol period T is

$$h(t) = \frac{\sin(\pi t/T)}{(\pi t/T)} \cdot \frac{\cos(\pi R t/T)}{(1 - 4R^2 t^2/T^2)}$$

The impulse response of a square root raised cosine filter with rolloff factor R is

$$h(t) = 4R \frac{\cos\left((1+R)\pi t/T\right) + \dfrac{\sin\left((1-R)\pi t/T\right)}{(4Rt/T)}}{\pi\sqrt{T}\left(1-(4Rt/T)^2\right)}$$

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.

If you choose `User-specified`, then the block first uses the automatic scaling described above and then multiplies all coefficients by the **Linear amplitude filter gain** parameter. The **Linear amplitude filter gain** parameter appears after you set **Filter gain** to `User-specified`.

Input and Output Signals

The input signal must be a scalar or a frame-based column vector. Set the **Input sampling mode** parameter according to whether the input is sample-based or frame-based.

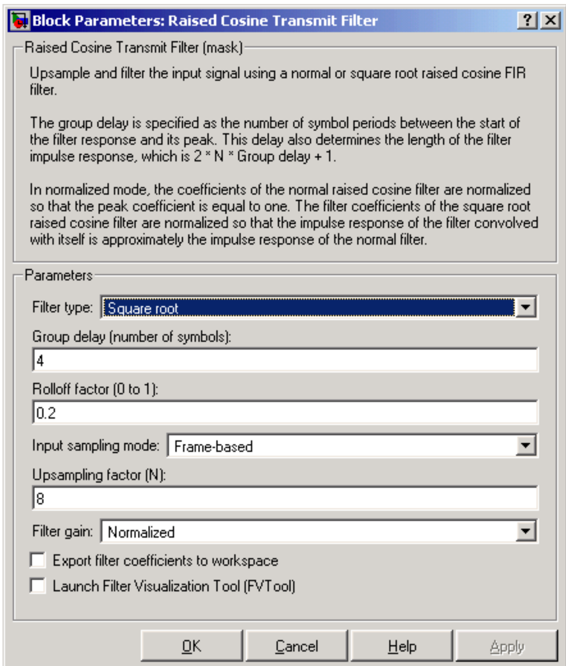
The input sampling mode and N, the value of the **Upsampling factor** parameter, determine characteristics of the output signal:

- If the input is a sample-based scalar, then the output is a sample-based scalar and the output sample time is N times the input sample time.
- If the input is frame-based, then the output is a frame-based vector whose length is N times the length of the input vector. The output frame period equals the input frame period.

Exporting Filter Coefficients to the MATLAB Workspace

To examine or manipulate the coefficients of the filter that this block designs, select **Export filter coefficients to workspace**. Then set the **Coefficient variable name** parameter to the name of a variable that you want the block to create in the MATLAB workspace. Running the simulation causes the block to create the variable, overwriting any previous contents in case the variable already exists.

Dialog Box



Filter type
: The type of raised cosine filter: `Square root` or `Normal`.

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

Rolloff factor
The rolloff factor for the filter, a real number between 0 and 1.

Input sampling mode
The type of input signal: `Frame-based` or `Sample-based`.

Upsampling factor
An integer greater than 1 representing the number of samples per symbol in the filtered output signal.

Filter gain
Determines how the block scales the filter coefficients. Choices are `Normalized` and `User-specified`.

Linear amplitude filter gain
A positive scalar used to scale the filter coefficients. This field appears only if **Filter gain** is set to `User-specified`.

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

Coefficient variable name
The name of the variable to create in the MATLAB workspace. This field appears only if **Export filter coefficients to workspace** is selected.

Launch Filter Visualization Tool
If you check this box, then MATLAB launches the Filter Visualization Tool (`fvtool`) to analyze the raised cosine filter whenever you apply any changes to the block's parameters.

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

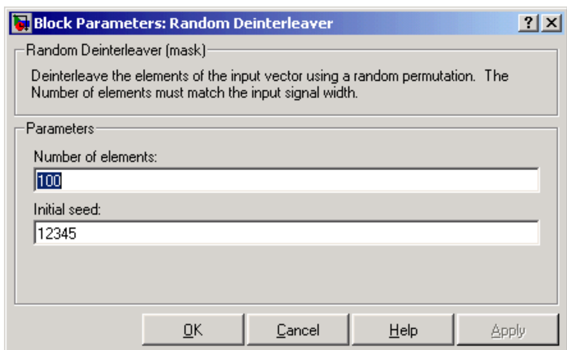
Description

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 theRandom 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

See Also

General Block Deinterleaver

Random Integer Generator

Purpose Generate integers randomly distributed in range [0, M-1]

Library Data Sources sublibrary of Comm Sources

Description

The Random Integer Generator block generates uniformly distributed random integers in the range [0, M-1], where M is the **M-ary number** defined in the dialog box.

The **M-ary number** can be either a scalar or a vector. If it is a scalar, then all output random variables are independent and identically distributed (i.i.d.). If the **M-ary number** is a vector, then its length must equal the length of the **Initial seed**; in this case each output has its own output range.

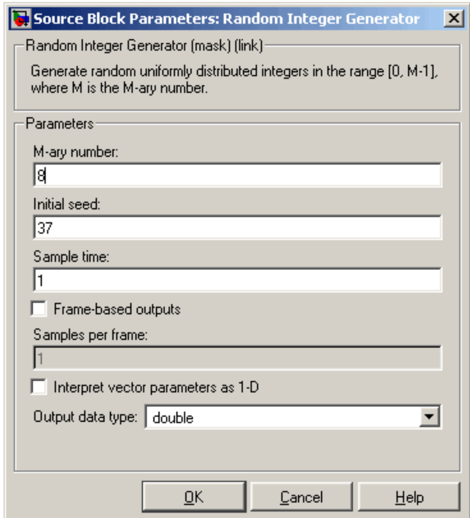
If the **Initial seed** parameter is a constant, then the resulting noise is repeatable.

Attributes of Output Signal

The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the **Frame-based outputs**, **Samples per frame**, and **Interpret vector parameters as 1-D** parameters. See "Signal Attribute Parameters for Random Sources" in Using the Communications Blockset 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.

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

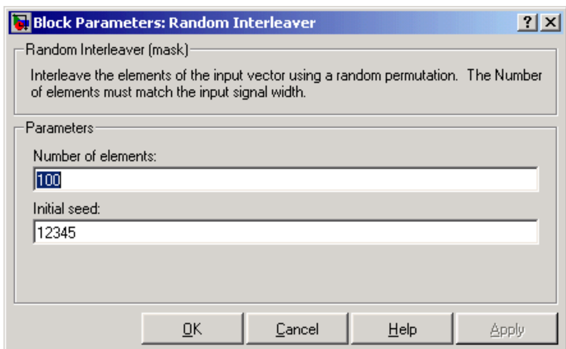
Description

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

See Also General Block Interleaver

Purpose

Generate Rayleigh distributed noise

Library

Noise Generators sublibrary of Comm Sources

Description

The Rayleigh Noise Generator block generates Rayleigh distributed noise. The Rayleigh probability density function is given by

$$f(x) = \begin{cases} \dfrac{x}{\sigma^2}\exp\left(-\dfrac{x^2}{2\sigma^2}\right) & x \geq 0 \\ 0 & x < 0 \end{cases}$$

where σ^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'srandseed function. At the MATLAB prompt, enter

```
randseed
```

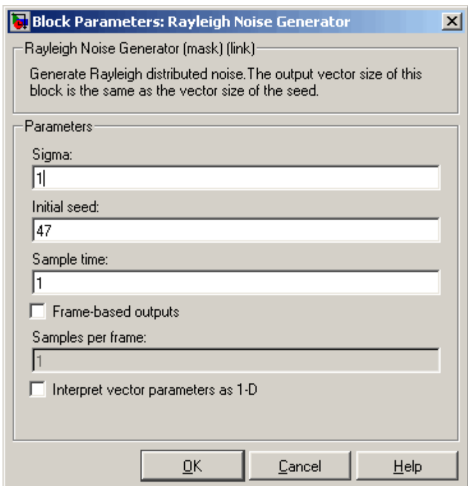
This returns a random prime number greater than 30. Entering `randseed` again produces a different prime number. If you supply an integer argument, `randseed` always returns the same prime for that integer. For example, `randseed(5)` always returns the same answer.

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 Using the Communications Blockset 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.

Sigma

Specify σ 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.

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

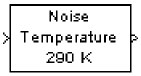
Purpose

Apply receiver thermal noise to complex baseband signal

Library

RF Impairments

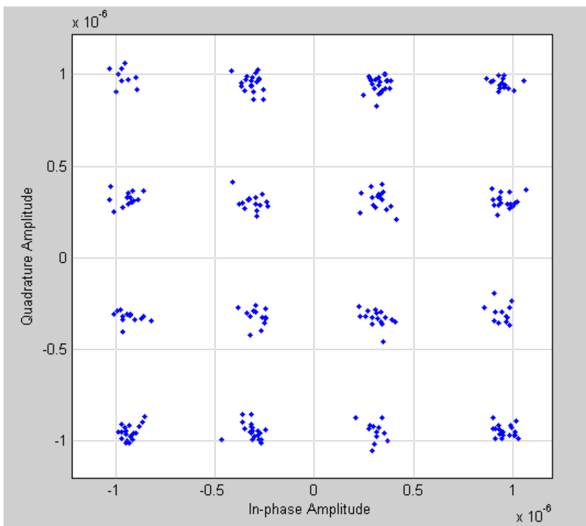
Description



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.

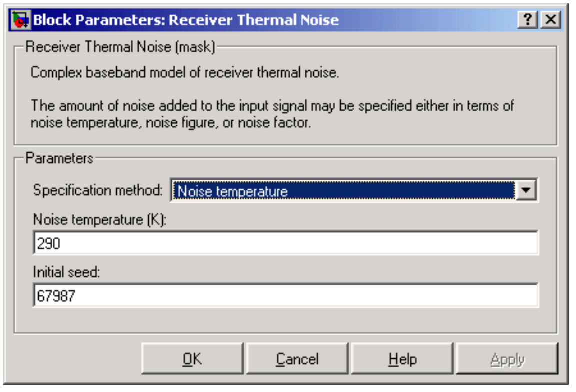
The following scatter plot shows the effect of the Receiver Thermal Noise block, with **Specification method** set to `Noise figure` and **Noise figure (dB)** set to `3.01`, on a signal modulated by 16-QAM.



This plot is generated by the model described in "Scatter Plot Examples" with the following parameter settings:

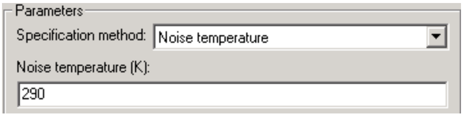
- Rectangular QAM Modulator Baseband
  - **Normalization method** set to `Average Power`
  - **Average power (watts)** set to `1e-12`
- Receiver Thermal Noise
  - **Specification method** set to `Noise figure`
  - **Noise figure (dB)** set to `3.01`

Dialog Box



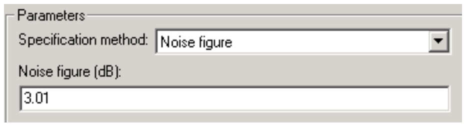
Specification method

The method by which you specify the amount of noise. The choices are `Noise temperature`, `Noise figure`, and `Noise factor`.



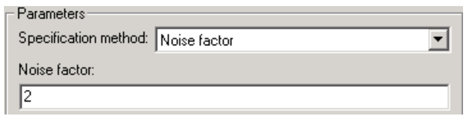
Noise temperature (K)

Scalar specifying the amount of noise in degrees Kelvin.



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

Rectangular QAM Demodulator Baseband

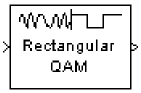
Purpose

Demodulate rectangular-QAM-modulated data

Library

AM, in Digital Baseband sublibrary of Modulation

Description



The Rectangular QAM Demodulator Baseband block demodulates a signal that was modulated using quadrature amplitude modulation with a constellation on a rectangular lattice.

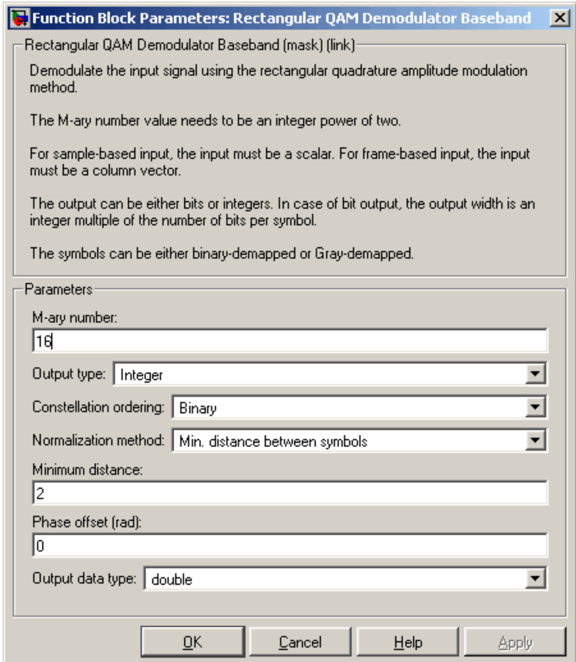
The signal constellation has M points, where M is the **M-ary number** parameter. M must have the form 2^K for some positive integer K. The block scales the signal constellation based on how you set the **Normalization method** parameter. For details, see the reference page for theRectangular QAM Modulator Baseband block.

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

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 theRectangular QAM Modulator Baseband block.

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.

Output type
: Indicates 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. 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 (watts)
The average power of the symbols in the constellation. This field appears only when **Normalization method** is set to `Average Power`.

Peak power (watts)
The maximum power among the symbols in the constellation. This field appears only when **Normalization method** is set to `Peak Power`.

Phase offset (rad)
The rotation of the signal constellation, in radians.

Output data type
For integer inputs, this block can output the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`. For bit inputs, output can be `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, or `double`.

Pair Block

Rectangular QAM Modulator Baseband

See Also

General QAM Demodulator Baseband

References

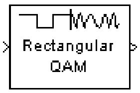
[1] Smith, Joel G., "Odd-Bit Quadrature Amplitude-Shift Keying," *IEEE Transactions on Communications*, Vol. COM-23, March 1975, 385-389.

Rectangular QAM Modulator Baseband

Purpose Modulate using rectangular quadrature amplitude modulation

Library AM, in Digital Baseband sublibrary of Modulation

Description The Rectangular QAM Modulator Baseband block modulates using M-ary quadrature amplitude modulation with a constellation on a rectangular lattice. The output is a baseband representation of the modulated signal.



Constellation Size and Scaling

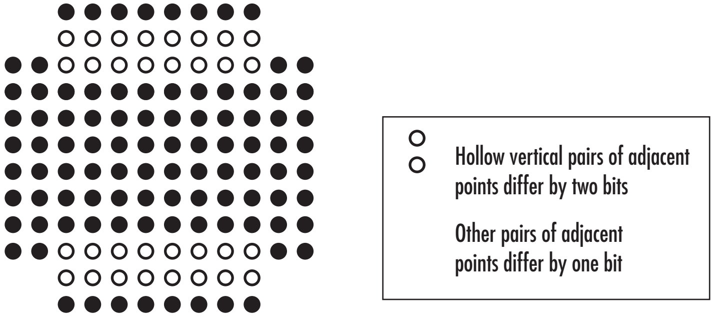
The signal constellation has M points, where M is the **M-ary number** parameter. M must have the form 2^K for some positive integer K. The block scales the signal constellation based on how you set the **Normalization method** parameter. The table below lists the possible scaling conditions.

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 **Minimum distance** parameter.
`Average Power`	The average power of the symbols in the constellation is the **Average power** parameter.
`Peak Power`	The maximum power of the symbols in the constellation is the **Peak power** parameter.

Input Signal Values

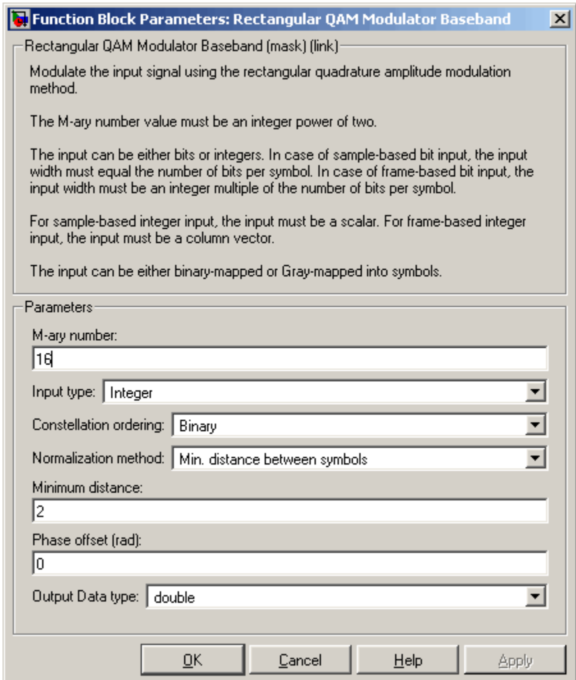
The input and output for this block are discrete-time signals. The **Input type** parameter determines whether the block accepts integers between 0 and M-1, or binary representations of integers:

- 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, and can accept the data types `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `single`, and `double`.
- If **Input type** is set to `Bit`, then the block accepts groups of K bits, called binary words. The input can be either a vector of length K or a frame-based column vector whose length is an integer multiple of K. For bit inputs, the block can accept `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, `boolean`, `single`, and `double`. The **Constellation ordering** parameter indicates how the block assigns binary words to points of the signal constellation. Such assignments apply independently to the in-phase and quadrature components of the input:
  - If **Constellation ordering** is set to `Binary`, then the block uses a natural binary-coded constellation.
  - If **Constellation ordering** is set to `Gray` and K is even, then the block uses a Gray-coded constellation.
  - If **Constellation ordering** is set to `Gray` and K is odd, then 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.



For details about the Gray coding, see the reference page for theM-PSK Modulator Baseband block and the paper among the references listed below. Note that since the in-phase and quadrature components are assigned independently, the Gray and binary orderings coincide when M = 4.

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 group of input bits to a corresponding integer. This field is active only when **Input 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 (watts)

The average power of the symbols in the constellation. This field appears only when **Normalization method** is set to `Average Power`.

Peak power (watts)

The maximum power of the symbols in the constellation. This field appears only when **Normalization method** is set to `Peak Power`.

Phase offset (rad)

The rotation of the signal constellation, in radians.

Output data type

The output data type can be either `single` or `double`.

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 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, $\log_2$(**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.

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.

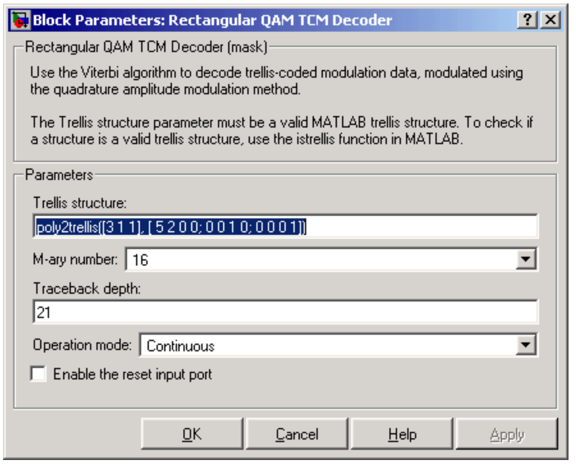
- In `Truncated` mode, the block treats each frame independently. The traceback path starts at the state with the lowest metric. D must be less than or equal to the vector length of the input.
- In `Terminated` mode, the block treats each frame independently. The traceback path always starts at the all-zeros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

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 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, $\log_2$(**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

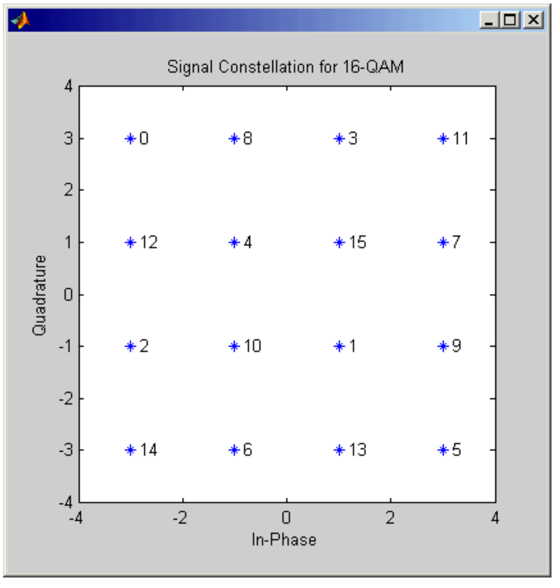
  ```
  poly2trellis(7,[171 133],171)
  ```

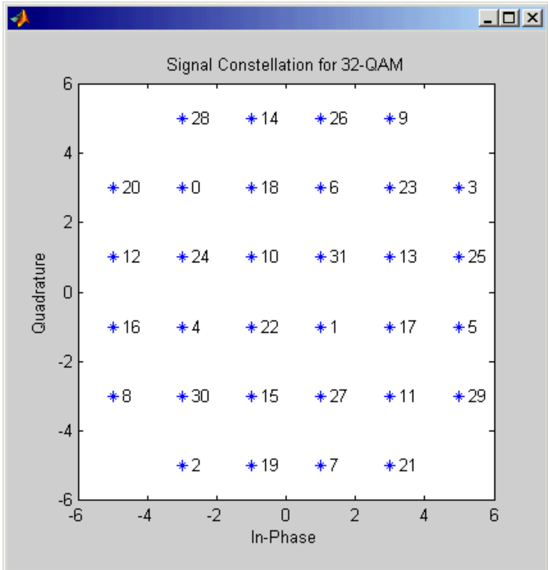
- If you have a variable in the MATLAB workspace that contains the trellis structure, then enter its name as the **Trellis structure** parameter. This way is faster because it causes Simulink to spend less time updating the diagram at the beginning of each simulation, compared to the usage in the previous bulleted item.

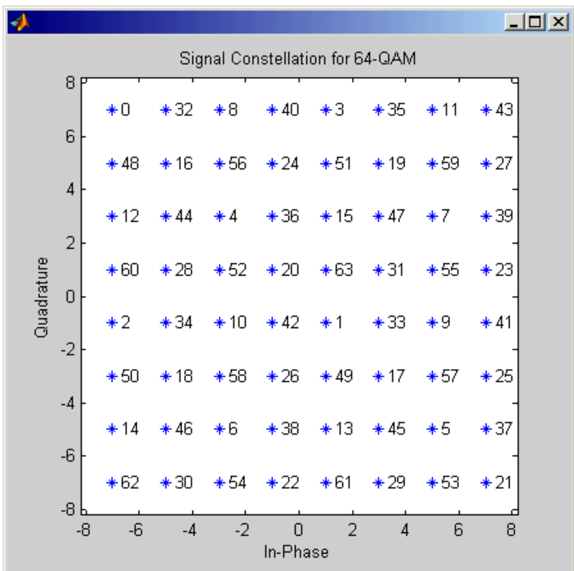
Signal Constellations

The trellis-coded modulation technique partitions the constellation into subsets called cosets, so as to maximize the minimum distance between pairs of points in each coset. This block internally forms a valid partition based on the value you choose for the **M-ary number** parameter.

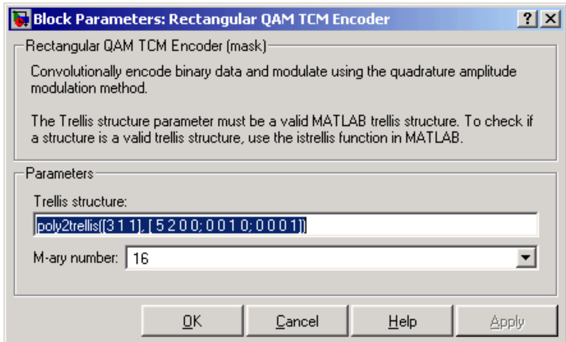
The figures below show the labeled set-partitioned signal constellations that the block uses when **M-ary number** is 16, 32, and 64. For constellations of other sizes, see [1].




Signal Constellation for 32-QAM
Quadrature
In-Phase
28 14 26 9
20 0 18 6 23 3
12 24 10 31 13 25
16 4 22 1 17 5
8 30 15 27 11 29
2 19 7 21


Signal Constellation for 64-QAM
Quadrature
In-Phase
0 32 8 40 3 35 11 43
48 16 56 24 51 19 59 27
12 44 4 36 15 47 7 39
60 28 52 20 63 31 55 23
2 34 10 42 1 33 9 41
50 18 58 26 49 17 57 25
14 46 6 38 13 45 5 37
62 30 54 22 61 29 53 21

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.

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

Rician Fading Channel

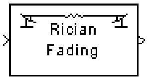
Purpose

Simulate Rician fading propagation channel

Library

Channels

Description



The Rician Fading Channel block implements a baseband simulation of a Rician fading propagation channel. This block is useful for modeling mobile wireless communication systems when the transmitted signal can travel to the receiver along a dominant line-of-sight or direct path. If the signal can travel along a line-of-sight path and also along other fading paths, then you can use this block in parallel with the Multipath Rayleigh Fading Channel block. For details about fading channels, see the references listed below.

The input can be either a scalar or a frame-based column vector. The input is a complex signal.

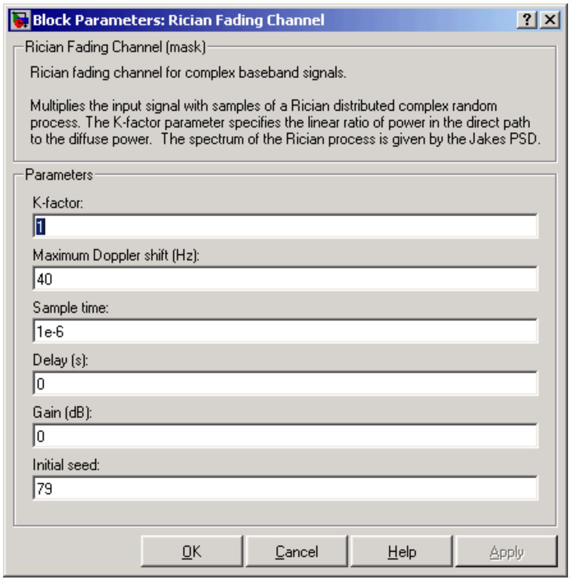
Fading causes the signal to spread and become diffuse. The **K-factor** parameter, which is part of the statistical description of the Rician distribution, represents the ratio between direct-path (unspread) power and diffuse power. The ratio is expressed linearly, not in decibels. While the **Gain** parameter controls the overall gain through the channel, the **K-factor** parameter controls the gain's partition into direct and diffuse components.

Relative motion between the transmitter and receiver causes Doppler shifts in the signal frequency. The Jakes PSD (power spectral density) determines the spectrum of the Rician process.

The **Sample time** parameter is the time between successive elements of the input signal. Note that if the input is a frame-based column vector of length n, then the frame period (as the Simulink Probe block reports, for example) is n***Sample time**.

The **Delay** parameter specifies a time delay in seconds and the **Gain** parameter specifies a gain that applies to the input signal. Both parameters are scalars.

Dialog Box



K-factor
: The ratio of power in the direct path to diffuse power. The ratio is expressed linearly, not in decibels.

Maximum Doppler shift (Hz)
: A positive scalar that indicates the maximum Doppler shift.

Sample time
: The period of each element of the input signal.

Delay (s)
: A scalar that specifies the propagation delay.

Gain (dB)
: A scalar that specifies the gain.

Initial seed
: The scalar seed for the Gaussian noise generator.

See Also

Rician Noise Generator, Multipath Rayleigh Fading Channel

References

[1] Jeruchim, Michel C., Balaban, Philip, and Shanmugan, K. Sam, *Simulation of Communication Systems*, Second edition, New York, Kluwer Academic/Plenum, 2000.

[2] Jakes, William C., ed. *Microwave Mobile Communications*. New York: IEEE Press, 1974.

[3] Lee, William C. Y. *Mobile Communications Design Fundamentals*, 2nd ed. New York: Wiley, 1993.

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

$$f(x) = \begin{cases} \dfrac{x}{\sigma^2} I_0\left(\dfrac{mx}{\sigma^2}\right) \exp\left(-\dfrac{x^2+m^2}{2\sigma^2}\right) & x \geq 0 \\ 0 & x < 0 \end{cases}$$

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
- 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 σ 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'srandseed function. At the MATLAB prompt, enter

```
randseed
```

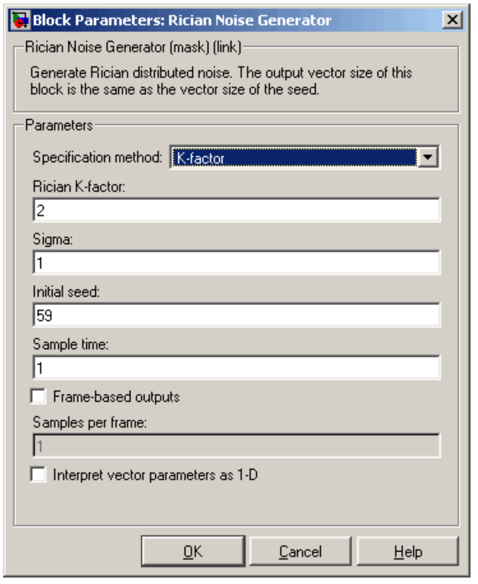
This returns a random prime number greater than 30. Entering `randseed` again produces a different prime number. If you supply an integer argument, `randseed` always returns the same prime for that integer. For example, `randseed(5)` always returns the same answer.

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 Using the Communications Blockset 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_I and m_Q, respectively, of the Gaussian components. These fields appear only if **Specification method** is `Quadrature components`.

Sigma
The variable σ 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.

See Also

Rician Fading Channel

References

[1] Proakis, John G., *Digital Communications*, Third edition, New York, McGraw Hill, 1995.

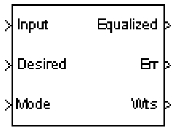
Purpose

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.

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 Using the Communications Blockset.
- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

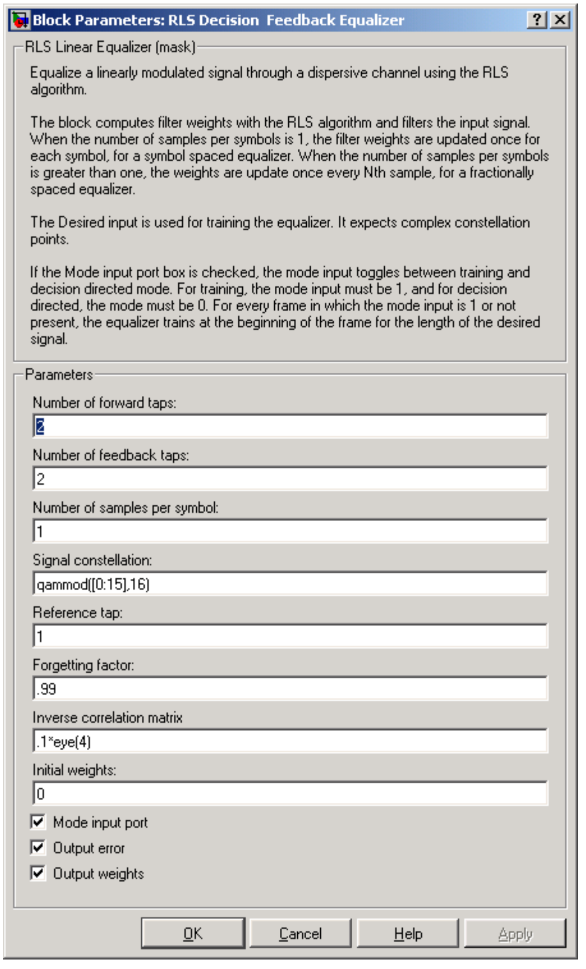
Equalizer Delay

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

RLS Linear Equalizer

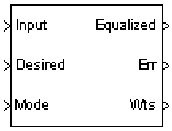
Purpose

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.

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 Using the Communications Blockset.
- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

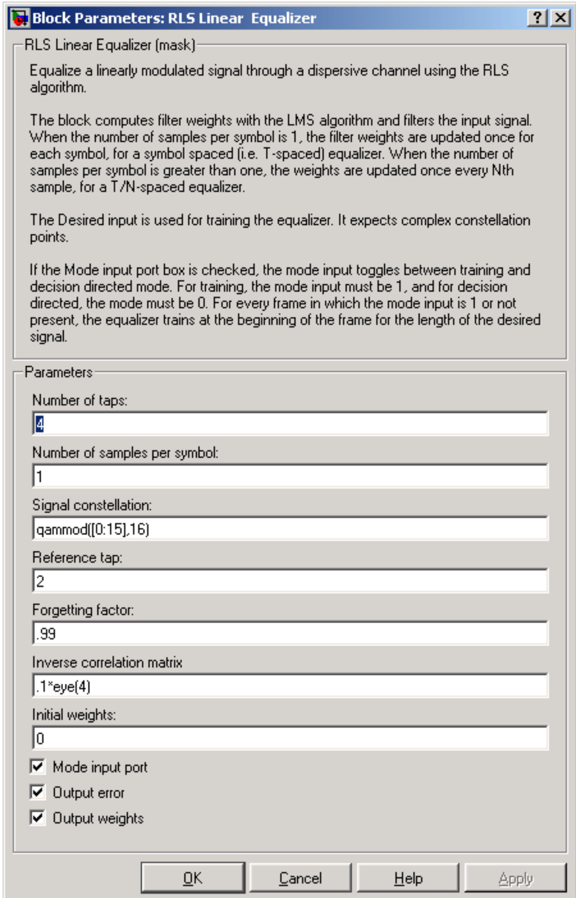
Equalizer Delay

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.

Forgetting factor
The forgetting factor of the RLS algorithm, a number between 0 and 1.

Inverse correlation matrix
The initial value for the inverse correlation matrix. The matrix must be N-by-N, where N is the number of taps.

Initial weights
A vector that lists the initial weights for the taps.

Mode input port
If you check this box, the block has an input port that enables you to toggle between training and decision-directed mode.

Output error
If you check this box, the block outputs the error signal, which is the difference between the equalized signal and the reference signal.

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

Examples

See the Adaptive Equalization demo.

References

[1] Farhang-Boroujeny, B., *Adaptive Filters: Theory and Applications*, Chichester, England, Wiley, 1998.

[2] Haykin, Simon, *Adaptive Filter Theory*, Third Ed., Upper Saddle River, N.J., Prentice-Hall, 1996.

[3] Kurzweil, Jack, *An Introduction to Digital Communications*, New York, Wiley, 2000.

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

See Also

RLS Decision Feedback Equalizer, LMS Linear Equalizer, CMA Equalizer

Purpose

Scramble the input signal

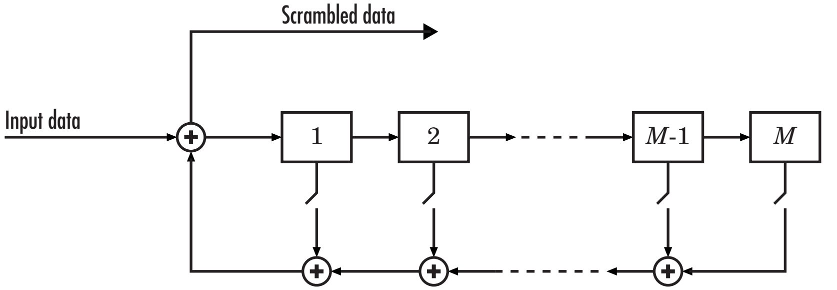
Library

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.

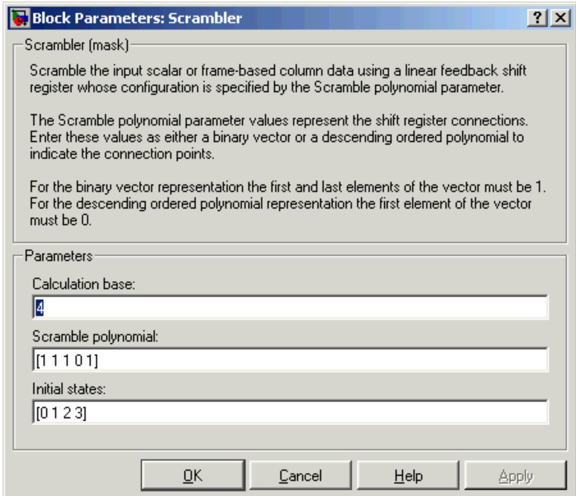
One purpose of scrambling is to reduce the length of strings of 0s or 1s in a transmitted signal, since a long string of 0s or 1s may cause transmission synchronization problems. Below is a schematic of the scrambler. All adders perform addition modulo N.



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}, 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 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

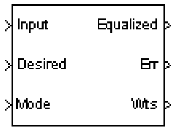
Purpose

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.

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 Using the Communications Blockset.

- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

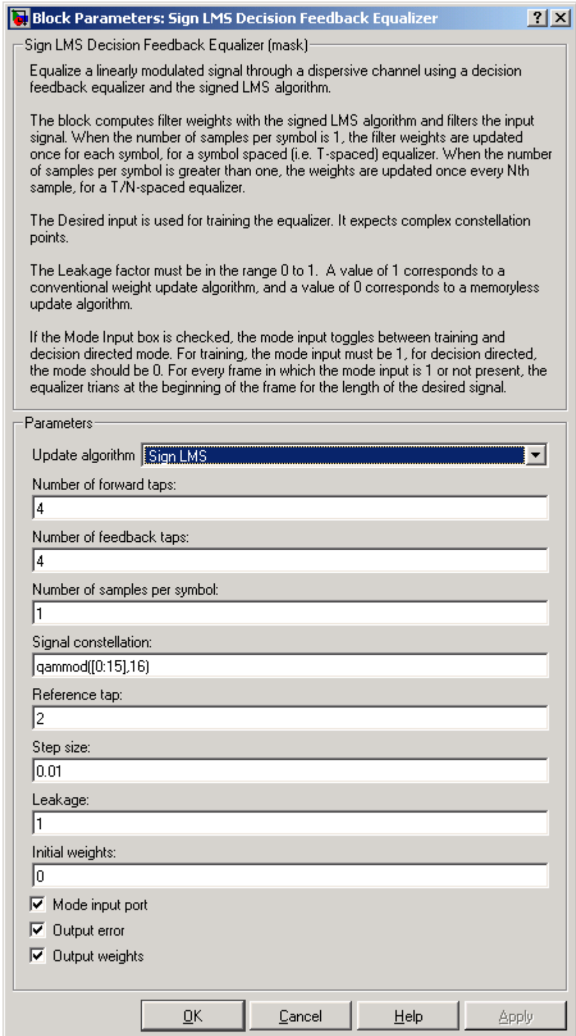
Equalizer Delay

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

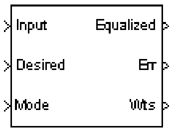
Purpose

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.

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 Using the Communications Blockset.

- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

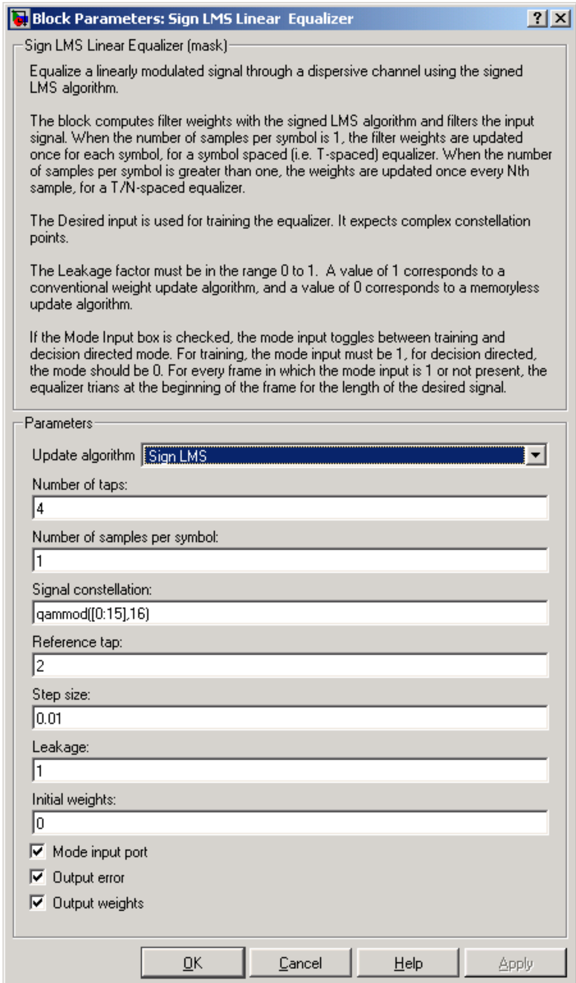
Equalizer Delay

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



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.

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

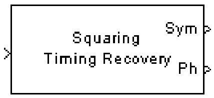
Purpose

Recover symbol timing phase using squaring method

Library

Timing Phase Recovery sublibrary of Synchronization

Description



The Squaring Timing Recovery block recovers the symbol timing phase of the input signal using a squaring method. This frame-based, feedforward, non-data-aided method is similar to the conventional squaring loop. This block is suitable for systems that use linear baseband modulation types such as pulse amplitude modulation (PAM), phase shift keying (PSK) modulation, and quadrature amplitude modulation (QAM).

Typically, the input to this block is the output of a receive filter that is matched to the transmitting pulse shape. The input to this block must be a frame-based column vector. 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.

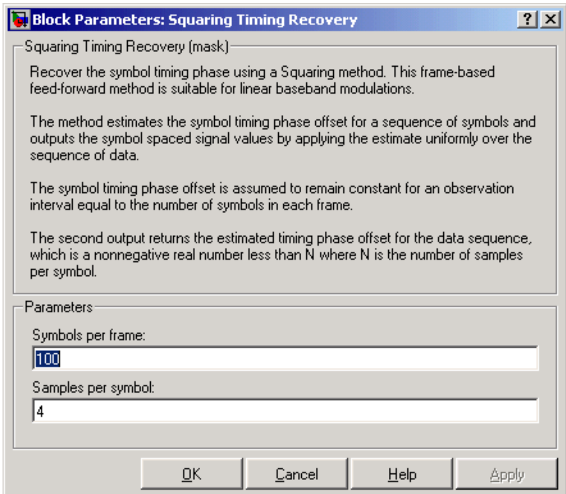
Note The block assumes that the phase offset is constant for all symbols in the entire input frame. If necessary, use the Buffer block to reorganize your data into frames over which the phase offset can be assumed constant. If the assumption of constant phase offset is valid, then a larger frame length 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

$$-\frac{1}{2\pi}\arg\left(\sum_{m=0}^{\mathrm{LN}-1} \left|x_{m+1}\right|^2 \exp(-\mathrm{j}2\pi m/\mathrm{N})\right)$$

as the normalized phase between -1/2 and 1/2, where x is the input vector, L is the **Symbols per frame** parameter and N is the **Samples per symbol** parameter.

For more information about the role that the timing estimator plays in this block's algorithm, see "Feedforward Method for Timing Phase Recovery" in Using the Communications Blockset.

Examples

See "Squaring Timing Phase Recovery Example" in Using the Communications Blockset.

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

Purpose

Demodulate SSB-AM-modulated data

Library

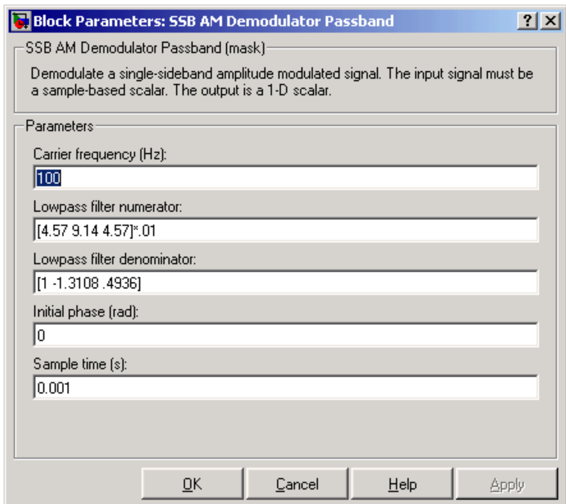
Analog Passband Modulation, in Modulation

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.

In the course of demodulating, this block uses a filter whose transfer function is described by the **Lowpass filter numerator** and **Lowpass filter denominator** parameters.

Dialog Box



Carrier frequency (Hz)
The carrier frequency in the corresponding SSB AM Modulator Passband block.

Lowpass filter numerator
The numerator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s.

Lowpass filter denominator
The denominator of the lowpass filter transfer function. It is represented as a vector that lists the coefficients in order of descending powers of s. For an FIR filter, set this parameter to `1`.

Initial phase (rad)
The initial phase of the carrier in radians.

Sample time
The sample time of the output signal.

Pair Block

SSB AM Modulator Passband

See Also

DSB AM Demodulator Passband, DSBSC AM Demodulator Passband

SSB AM Modulator 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.
- θ 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 Parameters

This block uses a Hilbert transform filter, possibly with a compensator. These dialog parameters relate to the Hilbert transform filter:

- The **Time delay for Hilbert transform filter** parameter specifies the delay in the filter design. You should choose a value of the form

  (N+1/2)*(**Sample time** parameter)

  where N is a positive integer.

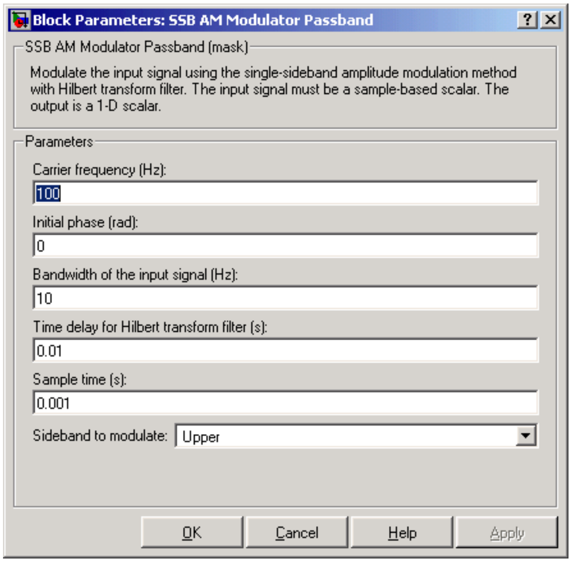
- The **Bandwidth of the input signal** parameter is the estimated highest frequency component in the input message signal.

  This parameter is used to design a compensator for the Hilbert transform filter, which would force the message signal amplitude to remain within the assigned range. If this parameter is either 0 or larger than 1/(2***Sample time**), then the block does not generate a compensator.

This block uses the `hilbiir` function in the Communications Toolbox to design the Hilbert transform filter.

Typically, an appropriate **Carrier frequency** value is much higher than the highest frequency of the input signal.

Dialog Box



Carrier frequency (Hz)
: The frequency of the carrier.

Initial phase (rad)
: The phase offset, θ, of the modulated signal.

Bandwidth of the input signal (Hz)
: The highest frequency component of the message signal. To avoid using a compensator in the Hilbert transform filter design, set this to `0`.

Time delay for Hilbert transform filter (s)
: The time delay in the design of the Hilbert transform filter.

Sample time
: The sample time of the Hilbert transform filtering.

Sideband to modulate
: This parameter specifies whether to transmit the upper or lower sideband.

Pair Block

SSB AM Demodulator Passband

See Also

DSB AM Modulator Passband, DSBSC AM Modulator Passband; `hilbiir` (Communications Toolbox)

References

[1] Peebles, Peyton Z, Jr. *Communication System Principles*. Reading, Mass.: Addison-Wesley, 1976.

Uniform Noise Generator

Purpose

Generate uniformly distributed noise between upper and lower bounds

Library

Noise Generators sublibrary of Comm Sources

Description

The Uniform Noise Generator block generates uniformly distributed noise. The output data of this block is uniformly distributed between the specified lower and upper bounds. The upper bound must be greater than or equal to the lower bound.

You must specify the **Initial seed** in the simulation. When it is a constant, the resulting noise is repeatable.

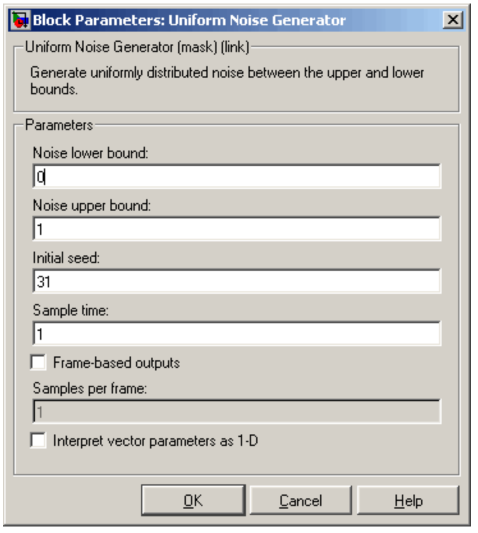
If all the elements of the output vector are to be independent and identically distributed (i.i.d.), then you can use a scalar for the **Noise lower bound** and **Noise upper bound** parameters. Alternatively, you can specify the range for each element of the output vector individually, by using vectors for the **Noise lower bound** and **Noise upper bound** parameters. If the bounds are vectors, then their length must equal the length of the **Initial seed** parameter.

Attributes of Output Signal

The output signal can be a frame-based matrix, a sample-based row or column vector, or a sample-based one-dimensional array. These attributes are controlled by the **Frame-based outputs**, **Samples per frame**, and **Interpret vector parameters as 1-D** parameters. See "Signal Attribute Parameters for Random Sources" in Using the Communications Blockset 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.

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.

See Also

Random Source (Signal Processing Blockset); `rand` (built-in MATLAB function)

Unipolar to Bipolar Converter

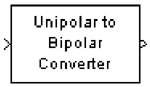
Purpose

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

Library

Utility Blocks

Description

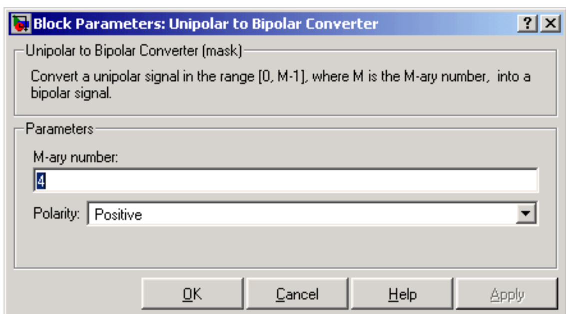


The Unipolar to Bipolar Converter block maps the unipolar input signal to a bipolar output signal. If the input consists of integers between 0 and M-1, where M is the **M-ary number** parameter, then the output consists of integers between -(M-1) and M-1. If M is even, then the output is odd, and vice-versa.

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

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

Dialog Box



M-ary number

The number of symbols in the bipolar or unipolar alphabet.

Polarity

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

Examples

If the input is [0; 1; 2; 3], the **M-ary number** parameter is `4`, and the **Polarity** parameter is `Positive`, then the output is [-3; -1; 1; 3]. Changing the **Polarity** parameter to `Negative` changes the output to [3; 1; -1; -3].

Pair Block

Bipolar to Unipolar Converter

Variable Step LMS Decision Feedback Equalizer

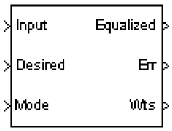
Purpose

Equalize using decision feedback equalizer that updates weights with variable-step-size LMS algorithm

Library

Equalizers

Description



The Variable Step LMS Decision Feedback Equalizer block uses a decision feedback equalizer and the variable-step-size LMS algorithm to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the variable-step-size LMS algorithm to update the weights, once per symbol. If the **Number of samples per symbol** parameter is `1`, then the block implements a symbol-spaced equalizer; otherwise, the block implements a fractionally spaced equalizer.

Input and Output Signals

The port labeled `Input` receives the signal you want to equalize, as a scalar or a frame-based column vector. The port labeled `Desired` receives a training sequence whose length is less than or equal to the number of symbols in the `Input` signal. Valid training symbols are those listed in the **Signal constellation** vector.

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 Using the Communications Blockset.
- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

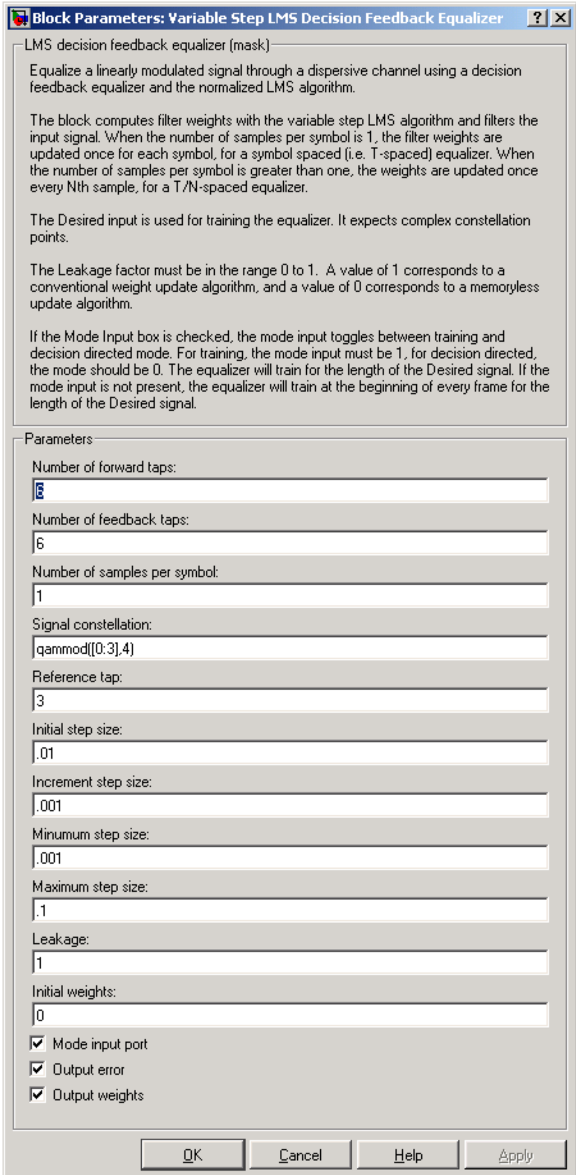
Equalizer Delay

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


Block Parameters: Variable Step LMS Decision Feedback Equalizer
LMS decision feedback equalizer (mask)
Equalize a linearly modulated signal through a dispersive channel using a decision feedback equalizer and the normalized LMS algorithm.
The block computes filter weights with the variable step LMS algorithm and filters the input signal. When the number of samples per symbol is 1, the filter weights are updated once for each symbol, for a symbol spaced (i.e. T-spaced) equalizer. When the number of samples per symbol is greater than one, the weights are updated once every Nth sample, for a T/N-spaced equalizer.
The Desired input is used for training the equalizer. It expects complex constellation points.
The Leakage factor must be in the range 0 to 1. A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.
If the Mode Input box is checked, the mode input toggles between training and decision directed mode. For training, the mode input must be 1, for decision directed, the mode should be 0. The equalizer will train for the length of the Desired signal. If the mode input is not present, the equalizer will train at the beginning of every frame for the length of the Desired signal.
Parameters
Number of forward taps:
Number of feedback taps:
6
Number of samples per symbol:
1
Signal constellation:
qammod([0:3],4)
Reference tap:
3
Initial step size:
.01
Increment step size:
.001
Minumum step size:
.001
Maximum step size:
.1
Leakage:
1
Initial weights:
0
Mode input port
Output error
Output weights
OK
Cancel
Help
Apply

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

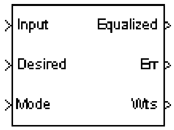
Purpose

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.

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 Using the Communications Blockset.
- `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 Using the Communications Blockset.

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 Using the Communications Blockset.

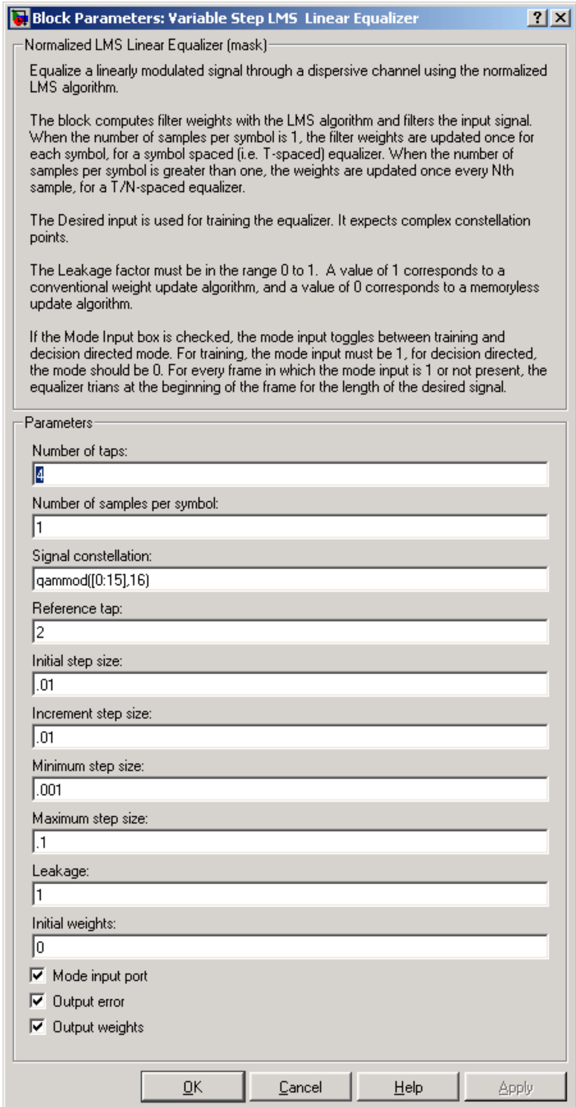
Equalizer Delay

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


Block Parameters: Variable Step LMS Linear Equalizer
Normalized LMS Linear Equalizer (mask)
Equalize a linearly modulated signal through a dispersive channel using the normalized LMS algorithm.
The block computes filter weights with the LMS algorithm and filters the input signal. When the number of samples per symbol is 1, the filter weights are updated once for each symbol, for a symbol spaced (i.e. T-spaced) equalizer. When the number of samples per symbol is greater than one, the weights are updated once every Nth sample, for a T/N-spaced equalizer.
The Desired input is used for training the equalizer. It expects complex constellation points.
The Leakage factor must be in the range 0 to 1. A value of 1 corresponds to a conventional weight update algorithm, and a value of 0 corresponds to a memoryless update algorithm.
If the Mode Input box is checked, the mode input toggles between training and decision directed mode. For training, the mode input must be 1, for decision directed, the mode should be 0. For every frame in which the mode input is 1 or not present, the equalizer trians at the beginning of the frame for the length of the desired signal.
Parameters
Number of taps:
4
Number of samples per symbol:
1
Signal constellation:
qammod([0:15],16)
Reference tap:
2
Initial step size:
.01
Increment step size:
.01
Minimum step size:
.001
Maximum step size:
.1
Leakage:
1
Initial weights:
0
Mode input port
Output error
Output weights
OK
Cancel
Help
Apply

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

Number of samples per symbol
The number of input samples for each symbol.

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

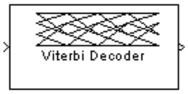
Purpose

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, then 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, then 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 non-double 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 the input data types `double`, `single`, `boolean`, `int8`, `uint8`, `int16`, `uint16`, `int32`, and `uint32`. For **Soft** decisions, the block accepts the input data types `double`, `single`, `int8`, `uint8`, `int16`, `uint16`, `int32`, and `uint32`.

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	+1: logical zero -1: logical one

Decision type Parameter	Possible Entries in Decoder Input	Interpretation of Values
`Hard Decision`	0, 1	0: logical zero 1: logical one
`Soft Decision`	Integers between 0 and 2^b-1, where *b* is the **Number of soft decision bits** parameter	0: most confident decision for logical zero 2^b-1: most confident decision for logical one Other values represent less confident decisions

To illustrate the soft decision situation more explicitly, the table below lists interpretations of values for 3-bit soft decisions.

Input Value	Interpretation
0	Most confident zero
1	Second most confident zero
2	Third most confident zero
3	Least confident zero
4	Least confident one
5	Third most confident one
6	Second most confident one
7	Most confident one

Operation Modes for Frame-Based Inputs

If the input signal is frame-based, then 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 **Reset** parameter set to `On each 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 encoder. If the encoder has `k` input streams and constraint length vector `constr` (using the polynomial description), then "enough" means `k*max(constr-1)`.

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, then the decoding delay consists of D zero symbols
- If the input signal is frame-based and the **Operation mode** parameter is set to `Continuous`, then the decoding delay consists of D zero symbols

- If the **Operation mode** parameter is set to `Truncated` or `Terminated`, then 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, then a typical **Traceback depth** value is about five times the constraint length of the code.

Reset Port

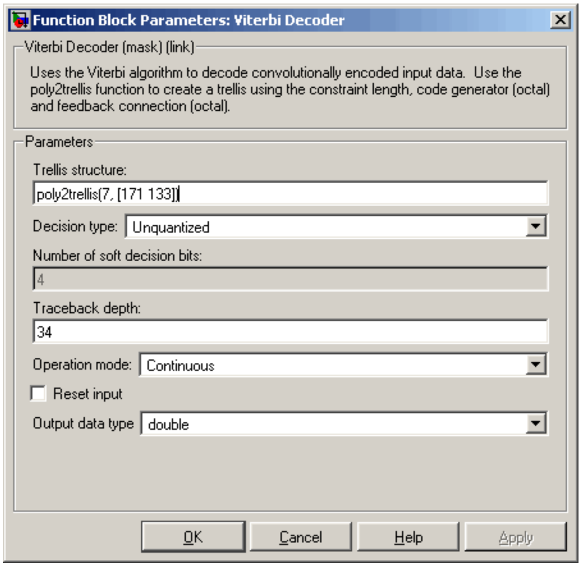
The reset port is usable only when the **Operation mode** parameter is set to `Continuous`. Checking the **Reset input** check box causes the block to have an additional input port, labeled `Rst`. When the `Rst` input is nonzero, the decoder returns to its initial state by configuring its internal memory 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 the **Reset** parameter in the Convolutional Encoder block to `On nonzero Rst input`.

The reset port supports `double` or `boolean` typed signals.

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.

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`.

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.

Reset input

When you check this box, the decoder 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.

Output data type

The output signal's data type can be `double`, `single`, `boolean`, `int8`, `uint8`, `int16`, `uint16`, `int32`, or `uint32`.

See Also

Convolutional Encoder, APP Decoder

References

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

[2] Gitlin, Richard D., Jeremiah F. Hayes, and Stephen B. Weinstein. *Data Communications Principles*. New York: Plenum, 1992.

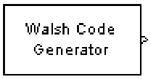
[3] Heller, Jerrold A. and Irwin Mark Jacobs. "Viterbi Decoding for Satellite and Space Communication." *IEEE Transactions on Communication Technology*, vol. COM-19, October 1971. 835-848.

Walsh Code Generator

Purpose Generate Walsh code from orthogonal set of codes

Library Sequence Generators sublibrary of Comm Sources

Description



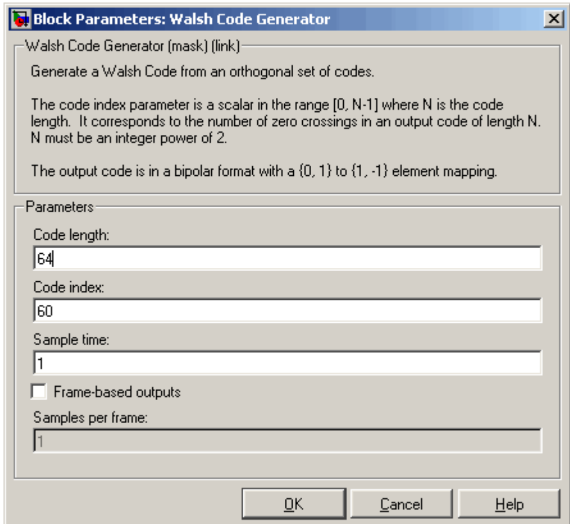
Walsh codes are defined as a set of N codes, denoted W_j, for $j = 0, 1, \ldots, N - 1$, which have the following properties:

- W_j takes on the values +1 and -1.
- $W_j[0] = 1$ for all j.
- W_j has exactly j zero crossings, for $j = 0, 1, \ldots, N - 1$.
- $W_j W_k^T = \begin{cases} 0 & j \neq k \\ N & j = k \end{cases}$
- 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 $[0, \ldots, N - 1]$. If you set **Walsh code index** equal to an integer j, the output code has exactly j zero crossings, for $j = 0, 1, \ldots, N - 1$.

Note, however, that the indexing in the Walsh Code Generator block is different than the indexing in the Hadamard Code Generator block. If you set the **Walsh code index** in the Walsh Code Generator block and the **Code index parameter** in the Hadamard Code Generator block, the two blocks output different codes.

Dialog Box



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

Code length
: Integer scalar that is a power of 2 specifying the length of the output code.

Code index
: Integer scalar in the range [0, 1, ... , N - 1], where N is the **Code length**, specifying the number of zero crossings in the output code.

Sample time
: A positive real scalar specifying the sample time of the output signal.

Frame-based outputs
: When checked, the block outputs a frame-based signal. When cleared, the block outputs a [1] unoriented scalar.

Samples per frame

The number of samples in a frame-based output signal. This field is active only if you select the **Frame-based outputs** check box. If **Samples per frame** is greater than the **Code length**, the code is cyclically repeated.

See also

Hadamard Code Generator, OVSF Code Generator

Purpose

Integrate over time window of fixed length

Library

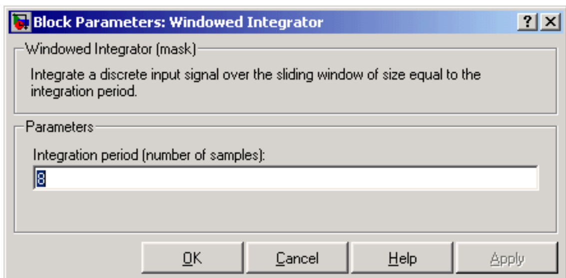
Comm Filters

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.

Dialog Box



Integration period
: The length of the interval of integration, measured in samples.

Examples

If **Integration period** is `3` and the input signal is a ramp (`1, 2, 3, 4`,...), then some of the sums that form the output of this block are as follows:

- 0+0+1 = 1
- 0+1+2 = 3
- 1+2+3 = 6
- 2+3+4 = 9

- 3+4+5 = 12
- 4+5+6 = 15
- etc.

The zeros in the first few sums represent initial conditions. If the input signal is a sample-based scalar, then the values 1, 3, 6,... are successive values of the scalar output signal. If the input signal is a frame-based column vector, then the values 1, 3, 6,... are organized into output frames that have the same vector length as the input frames.

See Also

Integrate and Dump, Discrete-Time Integrator (Simulink)

Functions — Alphabetical List

This section contains detailed references pages for each of the functions in the Communications Blockset.

comm_links

Purpose

Display and return library link information for Communications Blockset blocks.

Syntax

```
comm_links
comm_links(sys)
comm_links(sys,color)
```

Description

`comm_links` returns a structure with two elements. Each element contains a cell array of strings containing names of library blocks in the current system. The blocks are grouped into two categories: obsolete and current. Blocks at all levels of the model are analyzed.

`comm_links(sys)` works as above on the named system `sys`, instead of the current system.

`comm_links(sys,color)` additionally colors all obsolete blocks according to the specified `color`. `color` is one of the following strings: 'blue', 'green', 'red', 'cyan', 'magenta', 'yellow', or 'black'.

Obsolete blocks are blocks that are no longer supported. They might or might not work properly.

Current blocks are supported and represent the latest block functionality.

See Also

liblinks (Signal Processing Blockset), `commliblist`

Purpose

Open the main Communications Blockset library

Syntax

```
commlib
commlib(n)
commlib n
```

Description

`commlib` opens the current version of the main Communications Blockset library.

`commlib(n)` opens version number `n` of the main Communications Blockset library, where `n` can be either `'1.5'` or `'3.0.1'`. Version 1.5 refers to the Simulink portion of the Communications Toolbox 1.5 (Release 11.1).

`commlib n` is the same as `commlib(n)`.

See Also

simulink (Simulink), dsplib (Signal Processing Blockset)

commstartup

Purpose Default Simulink model settings for Communications Blockset

Syntax

```
commstartup
```

Description

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

Note The Signal Processing Blockset includes a similar dspstartup script, which assigns different model settings. For modeling communication systems, you should use `commstartup` alone.

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

To be more specific, the settings in `commstartup` cause models to:

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

See Also `startup`

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 is designed for producing random prime numbers that work well as seeds for random source blocks or noisy channel blocks in the Communications Blockset.

`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.

`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`

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