Signal Processing Blockset™ 6

Reference

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Signal Processing Blockset™ Reference

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Contents

1 Block Reference

Estimation 1-2
Linear Prediction 1-2
Parametric Estimation 1-3
Power Spectrum Estimation 1-3

Filtering 1-4
Adaptive Filters 1-4
Filter Design Toolbox 1-4
Filter Designs 1-5
Multirate Filters 1-6

Math Functions 1-7
Math Operations 1-7
Matrices and Linear Algebra 1-7
Polynomial Functions 1-11

Quantizers 1-11

Signal Management 1-12
Buffers 1-12
Indexing 1-13
Signal Attributes 1-13
Switches and Counters 1-14

Signal Operations 1-14

Signal Processing Sinks 1-15

Signal Processing Sources 1-16

Statistics 1-17

Transforms 1-18

2 Blocks — Alphabetical List

3 Function Reference

Glossary

Index

Block Reference

Estimation (p. 1-2)	Perform spectrum estimates and autoregressive modeling
Filtering (p. 1-4)	Design, create, and work with filters
Math Functions (p. 1-7)	Perform linear algebra and basic math calculations
Quantizers (p. 1-11)	Design and implement quantization schemes
Signal Management (p. 1-12)	Perform basic signal processing operations
Signal Operations (p. 1-14)	Control signal attributes, buffer signals, and index signals
Signal Processing Sinks (p. 1-15)	View or log signals
Signal Processing Sources (p. 1-16)	Generate discrete-time signals
Statistics (p. 1-17)	Perform statistical computations on signals
Transforms (p. 1-18)	Compute transforms

Estimation

Linear Prediction (p. 1-2)	Compute or work with linear predictive representations
Parametric Estimation (p. 1-3)	Compute estimates of autoregressive model parameters
Power Spectrum Estimation (p. 1-3)	Compute parametric and nonparametric spectral estimates

Linear Prediction

Autocorrelation LPC	Determine coefficients of Nth-order forward linear predictors
Levinson-Durbin	Solve linear system of equations using Levinson-Durbin recursion
LPC to LSF/LSP Conversion	Convert linear prediction coefficients to line spectral pairs or line spectral frequencies
LPC to/from Cepstral Coefficients	Convert linear prediction coefficients to cepstral coefficients or cepstral coefficients to linear prediction coefficients
LPC to/from RC	Convert linear prediction coefficients to reflection coefficients or reflection coefficients to linear prediction coefficients
LPC/RC to Autocorrelation	Convert linear prediction coefficients or reflection coefficients to autocorrelation coefficients
LSF/LSP to LPC Conversion	Convert line spectral frequencies or line spectral pairs to linear prediction coefficients

Parametric Estimation

Burg AR Estimator	Compute estimate of autoregressive (AR) model parameters using Burg method
Covariance AR Estimator	Compute estimate of autoregressive (AR) model parameters using covariance method
Modified Covariance AR Estimator	Compute estimate of autoregressive (AR) model parameters using modified covariance method
Yule-Walker AR Estimator	Compute estimate of autoregressive (AR) model parameters using Yule-Walker method

Power Spectrum Estimation

Burg Method	Compute parametric spectral estimate using Burg method
Covariance Method	Compute parametric spectral estimate using covariance method
Magnitude FFT	Compute nonparametric estimate of spectrum using periodogram method
Modified Covariance Method	Compute parametric spectral estimate using modified covariance method
Periodogram	Compute nonparametric estimate of spectrum
Yule-Walker Method	Compute parametric estimate of tspectrum using Yule-Walker autoregressive (AR) method

Filtering

Adaptive Filters (p. 1-4)	Use adaptive filter algorithms
Filter Design Toolbox (p. 1-4)	Design and implement single- and multirate FIR and IIR filters
Filter Designs (p. 1-5)	Design and implement filters
Multirate Filters (p. 1-6)	Implement multirate filters

Adaptive Filters

Block LMS Filter	Compute filtered output, filter error, and filter weights for given input and desired signal using Block LMS adaptive filter algorithm
Fast Block LMS Filter	Compute filtered output, filter error, and filter weights for given input and desired signal using Fast Block LMS adaptive filter algorithm
Kalman Filter	Predict or estimate states of dynamic systems
LMS Filter	Compute filtered output, filter error, and filter weights for given input and desired signal using LMS adaptive filter algorithm
RLS Filter	Compute filtered output, filter error, and filter weights for given input and desired signal using RLS adaptive filter algorithm

Filter Design Toolbox

Arbitrary Magnitude Filter	Design arbitrary response filter
Bandpass Filter	Design bandpass filter

Bandstop Filter	Design bandstop filter
CIC Compensator	Design CIC compensator
CIC Filter	Design Cascaded Integrator-Comb (CIC) Filter
Differentiator Filter	Design differentiator filter
Fractional Delay Filter	Design fractional delay filter
Halfband Filter	Design halfband filter
Highpass Filter	Design highpass filter
Hilbert Filter	Design Hilbert filter
Inverse Sinc Filter	Design inverse sinc filter
Lowpass Filter	Design lowpass Filter
Nyquist Filter	Design Nyquist filter
Octave Filter	Design octave filter
Parametric Equalizer	Design parametric equalizer
Peak-Notch Filter	Design peak or notch filter

Filter Designs

Analog Filter Design	Design and implement analog filters
Digital Filter	Filter each channel of input over time using static or time-varying digital filter implementations
Digital Filter Design	Design and implement digital FIR and IIR filters
Filter Realization Wizard	Construct filter realizations using Digital Filter block or Sum, Gain, and Delay blocks

Overlap-Add FFT Filter	Implement overlap-add method of frequency-domain filtering
Overlap-Save FFT Filter	Implement overlap-save method of frequency-domain filtering

Multirate Filters

CIC Decimation	Decimate signal using Cascaded Integrator-Comb filter
CIC Interpolation	Interpolate signal using Cascaded Integrator-Comb filter
Dyadic Analysis Filter Bank	Decompose signals into subbands with smaller bandwidths and slower sample rates
Dyadic Synthesis Filter Bank	Reconstruct signals from subbands with smaller bandwidths and slower sample rates
FIR Decimation	Filter and downsample input signals
FIR Interpolation	Upsample and filter input signals
FIR Rate Conversion	Upsample, filter, and downsample input signals
Two-Channel Analysis Subband Filter	Decompose signal into high-frequency subband and low-frequency subband
Two-Channel Synthesis Subband Filter	Reconstruct signal from high-frequency subband and low-frequency subband

Math Functions

Math Operations (p. 1-7)	Use specialized math operations for signal processing applications
Matrices and Linear Algebra (p. 1-7)	Work with matrices
Polynomial Functions (p. 1-11)	Work with polynomials

Math Operations

Complex Exponential	Compute complex exponential function
Cumulative Product	Compute cumulative product of channel, column, or row elements
Cumulative Sum	Compute cumulative sum of channel, column, or row elements
dB Conversion	Convert magnitude data to decibels (dB or dBm)
dB Gain	Apply decibel gain
Difference	Compute element-to-element difference along specified dimension of input
Normalization	Perform vector normalization along rows, columns, or specified dimension

Matrices and Linear Algebra

Linear System Solvers (p. 1-8)	Solve matrix equation AX = B for X
Matrix Factorizations (p. 1-8)	Factor matrices
Matrix Inverses (p. 1-9)	Invert matrices
Matrix Operations (p. 1-9)	Perform basic matrix operations

Linear System Solvers

Backward Substitution	Solve *UX*=*B* for *X* when *U* is upper triangular matrix
Cholesky Solver	Solve *S*X=*B* for X when *S* is square Hermitian positive definite matrix
Forward Substitution	Solve *LX*=*B* for *X* when *L* is lower triangular matrix
LDL Solver	Solve *SX*=*B* for *X* when *S* is square Hermitian positive definite matrix
Levinson-Durbin	Solve linear system of equations using Levinson-Durbin recursion
LU Solver	Solve *AX*=*B* for *X* when *A* is square matrix
QR Solver	Find minimum-norm-residual solution to A*X*=B
SVD Solver	Solve *AX*=*B* using singular value decomposition

Matrix Factorizations

Cholesky Factorization	Factor square Hermitian positive definite matrix into triangular components
LDL Factorization	Factor square Hermitian positive definite matrices into lower, upper, and diagonal components
LU Factorization	Factor square matrix into lower and upper triangular components

QR Factorization	Factor rectangular matrix into unitary and upper triangular components
Singular Value Decomposition	Factor matrix using singular value decomposition

Matrix Inverses

Cholesky Inverse	Compute inverse of Hermitian positive definite matrix using Cholesky factorization
LDL Inverse	Compute inverse of Hermitian positive definite matrix using LDL factorization
LU Inverse	Compute inverse of square matrix using LU factorization
Pseudoinverse	Compute Moore-Penrose pseudoinverse of matrix

Matrix Operations

Array-Vector Add	Add vector to array along specified dimension
Array-Vector Divide	Divide array by vector along specified dimension
Array-Vector Multiply	Multiply array by vector along specified dimension
Array-Vector Subtract	Subtract vector from array along specified dimension
Constant Diagonal Matrix	Generate square, diagonal matrix
Create Diagonal Matrix	Create square diagonal matrix from diagonal elements

Extract Diagonal	Extract main diagonal of input matrix
Extract Triangular Matrix	Extract lower or upper triangle from input matrices
Identity Matrix	Generate matrix with ones on main diagonal and zeros elsewhere
Matrix 1-Norm	Compute 1-norm of matrix
Matrix Concatenate	Concatenate input signals of same data type to create contiguous output signal
Matrix Exponential	Compute matrix exponential
Matrix Multiply	Multiply or divide inputs
Matrix Product	Multiply matrix elements along rows, columns, or entire input
Matrix Square	Compute square of input matrix
Matrix Sum	Sum matrix elements along rows, columns, or entire input
Overwrite Values	Overwrite submatrix or subdiagonal of input
Permute Matrix	Reorder matrix rows or columns
Reciprocal Condition	Compute reciprocal condition of square matrix in 1-norm
Submatrix	Select subset of elements (submatrix) from matrix input
Toeplitz	Generate matrix with Toeplitz symmetry
Transpose	Compute matrix transpose

Polynomial Functions

Least Squares Polynomial Fit	Compute polynomial coefficients that best fit input data in least-squares sense
Polynomial Evaluation	Evaluate polynomial expression
Polynomial Stability Test	Use Schur-Cohn algorithm to determine whether all roots of input polynomial are inside unit circle

Quantizers

G711 Codec	Quantize narrowband speech input signals
Quantizer	Discretize input at specified interval
Scalar Quantizer Decoder	Convert each index value into quantized output value
Scalar Quantizer Design	Start Scalar Quantizer Design Tool (SQDTool) to design scalar quantizer using Lloyd algorithm
Scalar Quantizer Encoder	Encode each input value by associating it with index value of quantization region
Uniform Decoder	Decode integer input into floating-point output
Uniform Encoder	Quantize and encode floating-point input into integer output
Vector Quantizer Decoder	Find vector quantizer codeword that corresponds to given, zero-based index value

Vector Quantizer Design	Design vector quantizer using Vector Quantizer Design Tool (VQDTool)
Vector Quantizer Encoder	For given input, find index of nearest codeword based on Euclidean or weighted Euclidean distance measure

Signal Management

Buffers (p. 1-12)	Change sample rate or frame rate of signals by buffering or unbuffering
Indexing (p. 1-13)	Manipulate ordering of signals
Signal Attributes (p. 1-13)	Inspect or modify signal attributes
Switches and Counters (p. 1-14)	Perform actions when events occur

Buffers

Buffer	Buffer input sequence to smaller or larger frame size
Delay Line	Rebuffer sequence of inputs with one-sample shift
Queue	Store inputs in FIFO register
Stack	Store inputs into LIFO register
Unbuffer	Unbuffer input frame into sequence of scalar outputs

Indexing

Flip	Flip input vertically or horizontally
Multiport Selector	Distribute arbitrary subsets of input rows or columns to multiple output ports
Overwrite Values	Overwrite submatrix or subdiagonal of input
Selector	Select input elements from vector, matrix, or multidimensional signal
Submatrix	Select subset of elements (submatrix) from matrix input
Variable Selector	Select subset of rows or columns from input

Signal Attributes

Check Signal Attributes	Generate error when input signal does or does not match selected attributes exactly
Convert 1-D to 2-D	Reshape 1-D or 2-D input to 2-D matrix with specified dimensions
Convert 2-D to 1-D	Convert 2-D matrix input to 1-D vector
Data Type Conversion	Convert input signal to specified data type
Frame Conversion	Specify sampling mode of output signal
Inherit Complexity	Change complexity of input to match reference signal

Switches and Counters

Counter	Count up or down through specified range of numbers
Edge Detector	Detect transition from zero to nonzero value
Event-Count Comparator	Detect threshold crossing of accumulated nonzero inputs
Multiphase Clock	Generate multiple binary clock signals
N-Sample Enable	Output ones or zeros for specified number of sample times
N-Sample Switch	Switch between two inputs after specified number of sample periods

Signal Operations

Constant Ramp	Generate ramp signal with length based on input dimensions
Convolution	Compute convolution of two inputs
Delay	Delay discrete-time input by specified number of samples or frames
Downsample	Resample input at lower rate by deleting samples
Interpolation	Interpolate values of real input samples
NCO	Generate real or complex sinusoidal signals
Offset	Truncate vectors by removing or keeping beginning or ending values

Pad	Pad or truncate specified dimension(s)
Peak Finder	Determine whether each value of input signal is local minimum or maximum
Repeat	Resample input at higher rate by repeating values
Sample and Hold	Sample and hold input signal
Triggered Signal From Workspace	Import signal samples from MATLAB® workspace when triggered
Unwrap	Unwrap signal phase
Upsample	Resample input at higher rate by inserting zeros
Variable Fractional Delay	Delay input by time-varying fractional number of sample periods
Variable Integer Delay	Delay input by time-varying integer number of sample periods
Window Function	Compute and/or apply window to input signal
Zero Crossing	Count number of times signal crosses zero in single time step

Signal Processing Sinks

Display	Show value of input
Matrix Viewer	Display matrices as color images
Signal To Workspace	Write simulation data to array in MATLAB® workspace
Spectrum Scope	Compute and display periodogram of each input signal

Time Scope	Display signals generated during simulation
To Audio Device	Write audio data to computer's audio device
To Multimedia File	Write video frames and/or audio samples to multimedia file
To Wave File	Write audio data to file in Microsoft® Wave (`.wav`) format
Triggered To Workspace	Write input sample to MATLAB workspace when triggered
Vector Scope	Display vector or matrix of time-domain, frequency-domain, or user-defined data
Waterfall	View vectors of data over time

Signal Processing Sources

Chirp	Generate swept-frequency cosine (chirp) signal
Constant Diagonal Matrix	Generate square, diagonal matrix
Discrete Impulse	Generate discrete impulse
DSP Constant	Generate discrete- or continuous-time constant signal
From Audio Device	Read audio data from computer's audio device
From Multimedia File	Read video frames and/or audio samples from compressed multimedia file
From Wave File	Read audio data from Microsoft® Wave (`.wav`) file

Identity Matrix	Generate matrix with ones on main diagonal and zeros elsewhere
Multiphase Clock	Generate multiple binary clock signals
N-Sample Enable	Output ones or zeros for specified number of sample times
Random Source	Generate randomly distributed values
Signal From Workspace	Import signal from MATLAB® workspace
Sine Wave	Generate continuous or discrete sine wave

Statistics

Autocorrelation	Compute autocorrelation of vector inputs
Correlation	Compute cross-correlation of two inputs
Detrend	Remove linear trend from vectors
Histogram	Generate histogram of input or sequence of inputs
Maximum	Find maximum values in input or sequence of inputs
Mean	Find mean value of input or sequence of inputs
Median	Find median value of input
Minimum	Find minimum values in input or sequence of inputs
RMS	Compute root-mean-square value of input or sequence of inputs

Sort	Sort input elements by value
Standard Deviation	Find standard deviation of input or sequence of inputs
Variance	Compute variance of input or sequence of inputs

Transforms

Analytic Signal	Compute analytic signals of discrete-time inputs
Complex Cepstrum	Compute complex cepstrum of input
DCT	Compute discrete cosine transform (DCT) of input
DWT	Compute discrete wavelet transform (DWT) of input
FFT	Compute fast Fourier transform (FFT) of input
IDCT	Compute inverse discrete cosine transform (IDCT) of input
IDWT	Compute inverse discrete wavelet transform (IDWT) of input
IFFT	Compute inverse fast Fourier transform (IFFT) of input
Inverse Short-Time FFT	Recover time-domain signals by performing inverse short-time, fast Fourier transform (FFT)
Magnitude FFT	Compute nonparametric estimate of spectrum using periodogram method

Real Cepstrum	Compute real cepstrum of input
Short-Time FFT	Compute nonparametric estimate of spectrum using short-time, fast Fourier transform (FFT) method

2

Blocks — Alphabetical List

Analog Filter Design

Purpose

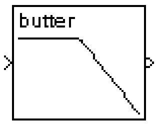
Design and implement analog filters

Library

Filtering / Filter Designs

`dsparch4`

Description



The Analog Filter Design block designs and implements a Butterworth, Chebyshev type I, Chebyshev type II, or elliptic filter in a highpass, lowpass, bandpass, or bandstop configuration.

The input must be a sample-based, continuous-time, real-valued, scalar signal.

The design and band configuration of the filter are selected from the **Design method** and **Filter type** pop-up menus in the dialog box. For each combination of design method and band configuration, an appropriate set of secondary parameters is displayed.

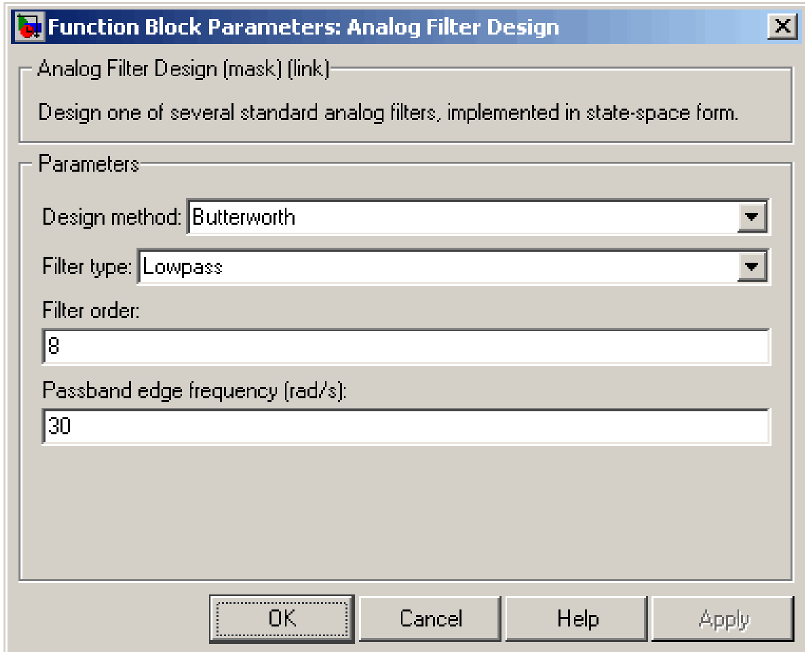
Filter Design	Description
`Butterworth`	The magnitude response of a Butterworth filter is maximally flat in the passband and monotonic overall.
`Chebyshev type I`	The magnitude response of a Chebyshev type I filter is equiripple in the passband and monotonic in the stopband.
`Chebyshev type II`	The magnitude response of a Chebyshev type II filter is monotonic in the passband and equiripple in the stopband.
`Elliptic`	The magnitude response of an elliptic filter is equiripple in both the passband and the stopband.

The following table lists the available parameters for each design/band combination. For lowpass and highpass band configurations, these parameters include the passband edge frequency Ω_p, the stopband edge frequency Ω_s, the passband ripple R_p, and the stopband attenuation R_s. For bandpass and bandstop configurations, the parameters include the lower and upper passband edge frequencies, Ω_{p1} and Ω_{p2}, the lower and upper stopband edge frequencies, Ω_{s1} and Ω_{s2}, the passband ripple R_p, and the stopband attenuation R_s. Frequency values are in rad/s, and ripple and attenuation values are in dB.

	Lowpass	Highpass	Bandpass	Bandstop
`Butterworth`	Order, Ω_p	Order, Ω_p	Order, Ω_{p1}, Ω_{p2}	Order, Ω_{p1}, Ω_{p2}
`Chebyshev Type I`	Order, Ω_p, R_p	Order, Ω_p, R_p	Order, Ω_{p1}, Ω_{p2}, R_p	Order, Ω_{p1}, Ω_{p2}, R_p
`Chebyshev Type II`	Order, Ω_s, R_s	Order, Ω_s, R_s	Order, Ω_{s1}, Ω_{s2}, R_s	Order, Ω_{s1}, Ω_{s2}, R_s
`Elliptic`	Order, Ω_p, R_p, R_s	Order, Ω_p, R_p, R_s	Order, Ω_{p1}, Ω_{p2}, R_p, R_s	Order, Ω_{p1}, Ω_{p2}, R_p, R_s

The analog filters are designed using the filter design commands in Signal Processing Toolbox™ software's `buttap`, `cheb1ap`, `cheb2ap`, and `ellipap` functions, and are implemented in state-space form. Filters of order 8 or less are implemented in controller canonical form for improved efficiency.

Dialog Box



The parameters displayed in the dialog box vary for different design/band combinations. Only some of the parameters listed below are visible in the dialog box at any one time.

Design method

The filter design method: `Butterworth`, `Chebyshev type I`, `Chebyshev type II`, or `Elliptic`. Tunable.

Filter type

The type of filter to design: `Lowpass`, `Highpass`, `Bandpass`, or `Bandstop`. Tunable.

Filter order

The order of the filter, for lowpass and highpass configurations. For bandpass and bandstop configurations, the order of the final filter is *twice* this value.

Passband edge frequency
The passband edge frequency, in rad/s, for the highpass and lowpass configurations of the Butterworth, Chebyshev type I, and elliptic designs. Tunable.

Lower passband edge frequency
The lower passband frequency, in rad/s, for the bandpass and bandstop configurations of the Butterworth, Chebyshev type I, and elliptic designs. Tunable.

Upper passband edge frequency
The upper passband frequency, in rad/s, for the bandpass and bandstop configurations of the Butterworth, Chebyshev type I, or elliptic designs. Tunable.

Stopband edge frequency
The stopband edge frequency, in rad/s, for the highpass and lowpass band configurations of the Chebyshev type II design. Tunable.

Lower stopband edge frequency
The lower stopband edge frequency, in rad/s, for the bandpass and bandstop configurations of the Chebyshev type II design. Tunable.

Upper stopband edge frequency
The upper stopband edge frequency, in rad/s, for the bandpass and bandstop filter configurations of the Chebyshev type II design. Tunable.

Passband ripple in dB
The passband ripple, in dB, for the Chebyshev Type I and elliptic designs. Tunable.

Stopband attenuation in dB
The stopband attenuation, in dB, for the Chebyshev Type II and elliptic designs. Tunable.

References

Antoniou, A. *Digital Filters: Analysis, Design, and Applications*. 2nd ed. New York, NY: McGraw-Hill, 1993.

Supported Data Types

- Double-precision floating point

See Also

Digital Filter Design	Signal Processing Blockset
`buttap`	Signal Processing Toolbox
`cheb1ap`	Signal Processing Toolbox
`cheb2ap`	Signal Processing Toolbox
`ellipap`	Signal Processing Toolbox

See the following sections for related information:

- "Filters"
- "Analog Filter Design Block"

Purpose

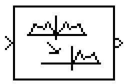
Compute analytic signals of discrete-time inputs

Library

Transforms

`dspxfrm3`

Description



The Analytic Signal block computes the complex analytic signal corresponding to each channel of the real M-by-N input, u

$$y = u + jH\{u\}$$

where $j = \sqrt{-1}$ and $H\{\ \}$ denotes the Hilbert transform. The real part of the output in each channel is a replica of the real input in that channel; the imaginary part is the Hilbert transform of the input. In the frequency domain, the analytic signal retains the positive frequency content of the original signal while zeroing-out negative frequencies and doubling the DC component.

The block computes the Hilbert transform using an equiripple FIR filter with the order specified by the **Filter order** parameter, n. The linear phase filter is designed using the Remez exchange algorithm, and imposes a delay of $n/2$ on the input samples.

The output has the same dimension and frame status as the input.

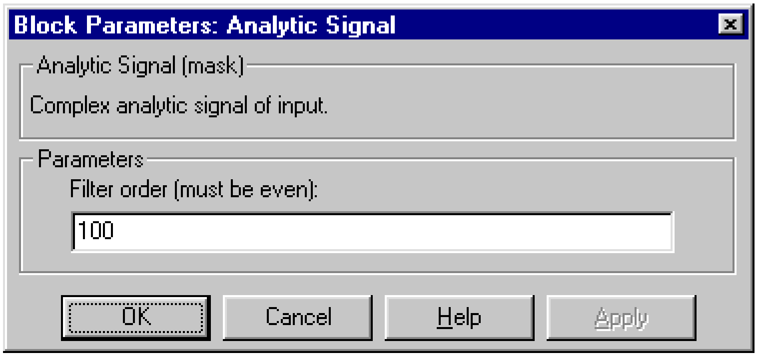
Sample-Based Operation

When the input is sample based, each of the $M*N$ matrix elements represents an independent channel. Thus, the block computes the analytic signal for each channel (matrix element) over time.

Frame-Based Operation

When the input is frame based, each of the N columns in the matrix contains M sequential time samples from an independent channel, and the block computes the analytic signal for each channel over time.

Dialog Box



Filter order

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

Supported Data Types

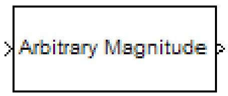
- Double-precision floating point
- Single-precision floating point

Purpose Design arbitrary response filter

Library Filtering / Filter Design Toolbox

`dspfdesign`

Description This block brings the functionality of the Fixed-Point Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox™ product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.



Dialog Box See "Arbitrary Response Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Add vector to array along specified dimension

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description

The Array-Vector Add block adds the values in the specified dimension of the N-dimensional input array A to the values in the input vector V.

The length of the input V must be the same as the length of the specified dimension of A. The Array-Vector Add block adds each element of V to the corresponding element along that dimension of A.

Consider a 3-dimensional M-by-N-by-P input array $A(i,j,k)$ and a P-by-1 input vector V. When the **Add along dimension** parameter is set to `2`, the output of the block $Y(i,j,k)$ is

$$Y(i,j,k) = A(i,j,k) + V(j)$$

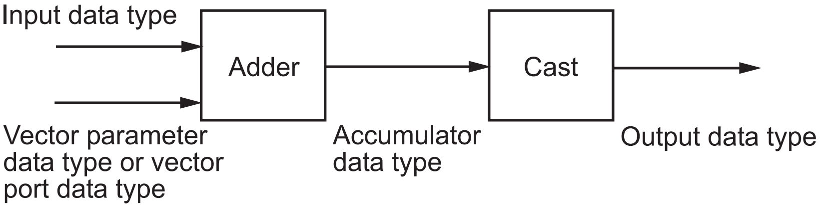
where

$$1 \le i \le M$$
$$1 \le j \le N$$
$$1 \le k \le P$$

The output of the Array-Vector Add block is the same size as the input array, A. When both inputs are sample based, the output is sample based; otherwise, the output is frame based. This block accepts real and complex floating-point and fixed-point inputs.

Fixed-Point Data Types

The following diagram shows the data types used within the Array-Vector Add block for fixed-point signals.

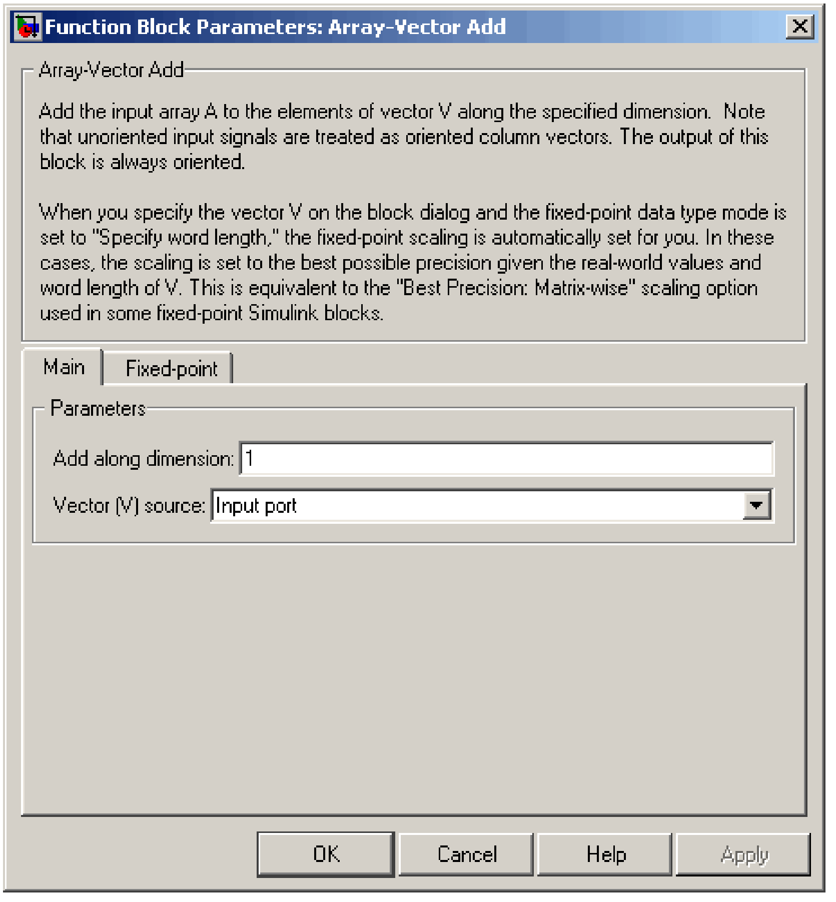


When the vector *V* is designated in the block mask, its elements have the data type and scaling that you specify in the **Vector (V)** parameters on the **Fixed-point** tab. When the vector comes in through the block port, its elements inherit their data type and scaling from the driving block.

You can set the vector, accumulator, and output data types in the block dialog as discussed below.

Dialog Box

The **Main** pane of the Array-Vector Add block dialog appears as follows.



Add along dimension

Specify the dimension along which to add the input array *A* to the elements of vector *V*.

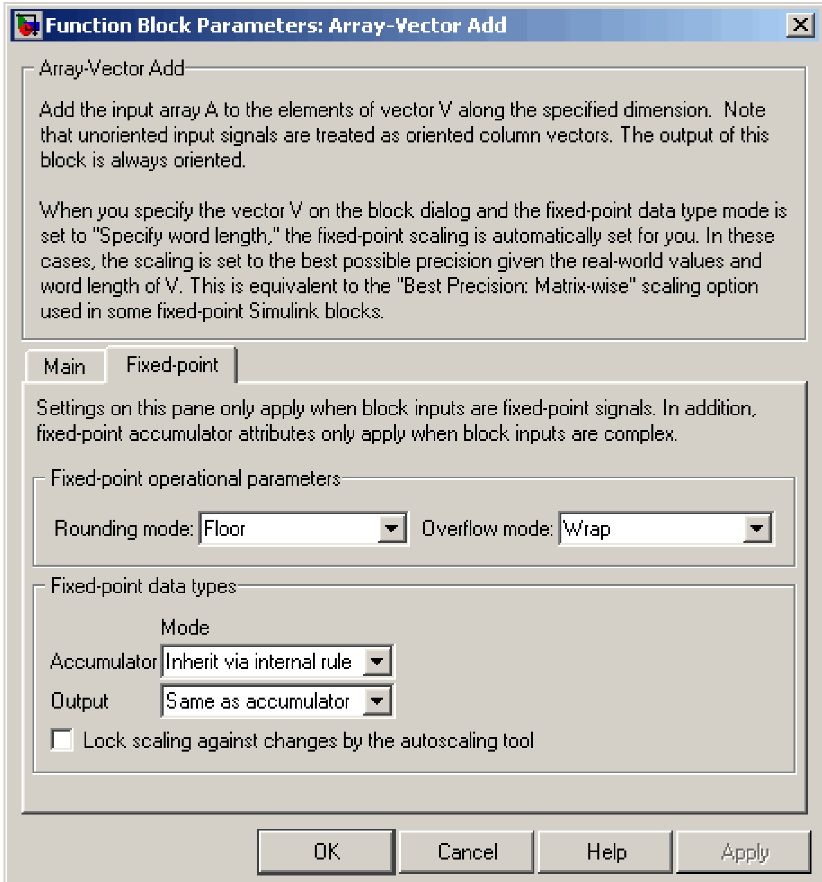
Vector (V) source

Specify the source of the vector, *V*. The vector can come from the `Input port` or from a `Dialog parameter`.

Vector (V)

Specify the vector, *V*. This parameter is visible only when you select `Dialog parameter` for the **Vector (V) source** parameter.

The **Fixed-point** pane of the Array-Vector Add block dialog appears as follows.



Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Vector (V)
Use this parameter to specify how you would like to designate the word and fraction lengths of the elements of the vector, *V*:

- When you select `Same word length as input`, the word length of the vector values match that of the input to the block.
- When you select `Specify word length`, you can enter the word length of the vector values, in bits. In this mode, the fraction length of the vector values is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the values.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the vector elements, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the vector element. This block requires power-of-two slope and a bias of zero.

Note The **Vector (V)** parameters on the **Fixed-point** pane are only applicable when you specify the vector through the **Vector (V)** parameter on the **Main** pane of the block mask. When the vector comes in through the block port, the data type and scaling of its elements are inherited from the driving block.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-11 for an illustration depicting the use of the accumulator data type in this block.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as first input`, these characteristics match those of the first input to the block.

- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output
Choose how you specify the output word length and fraction length:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.
- When you select `Same as first input`, these characteristics match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

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

Supported Data Types

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

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

See Also

Array-Vector Divide	Signal Processing Blockset
Array-Vector Multiply	Signal Processing Blockset
Array-Vector Subtract	Signal Processing Blockset

Purpose

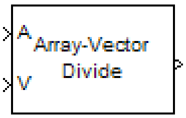
Divide array by vector along specified dimension

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description



The Array-Vector Divide block divides the values in the specified dimension of the N-dimensional input array *A* by the values in the input vector *V*.

The length of the input *V* must be the same as the length of the specified dimension of *A*. The Array-Vector Divide block divides each element of *V* by the corresponding element along that dimension of *A*.

Consider a 3-dimensional *M*-by-*N*-by-*P* input array *A(i,j,k)* and a *P*-by-1 input vector *V*. When the **Divide along dimension** parameter is set to `2`, the output of the block *Y(i,j,k)* is

$$Y(i,j,k) = \frac{A(i,j,k)}{V(j)}$$

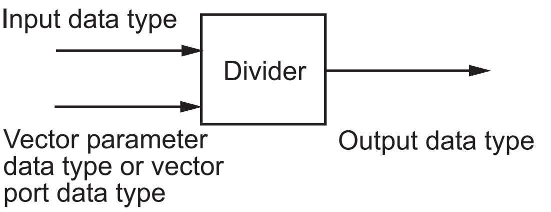
where

$$1 \le i \le M$$
$$1 \le j \le N$$
$$1 \le k \le P$$

The output of the Array-Vector Divide block is the same size as the input array, *A*. When both inputs are sample based, the output is sample based; otherwise, the output is frame based. This block accepts real and complex floating-point and fixed-point input arrays, and real floating-point and fixed-point input vectors.

Fixed-Point Data Types

The following diagram shows the data types used within the Array-Vector Divide block for fixed-point signals.

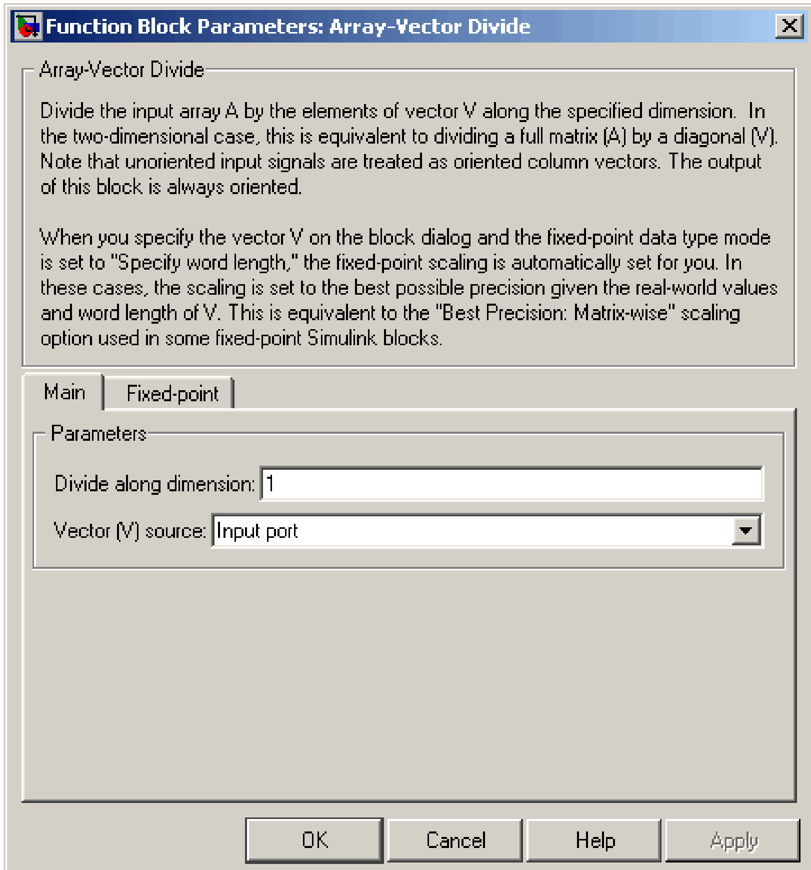


When the vector *V* is designated in the block mask, its elements have the data type and scaling that you specify in the **Vector (V)** parameters on the **Fixed-point** tab. When the vector comes in through the block port, its elements inherit their data type and scaling from the driving block.

You can set the vector and output data types in the block dialog as discussed below.

Dialog Box

The **Main** pane of the Array-Vector Divide block dialog appears as follows.



Divide along dimension

Specify the dimension along which to divide the input array *A* by the elements of vector *V*.

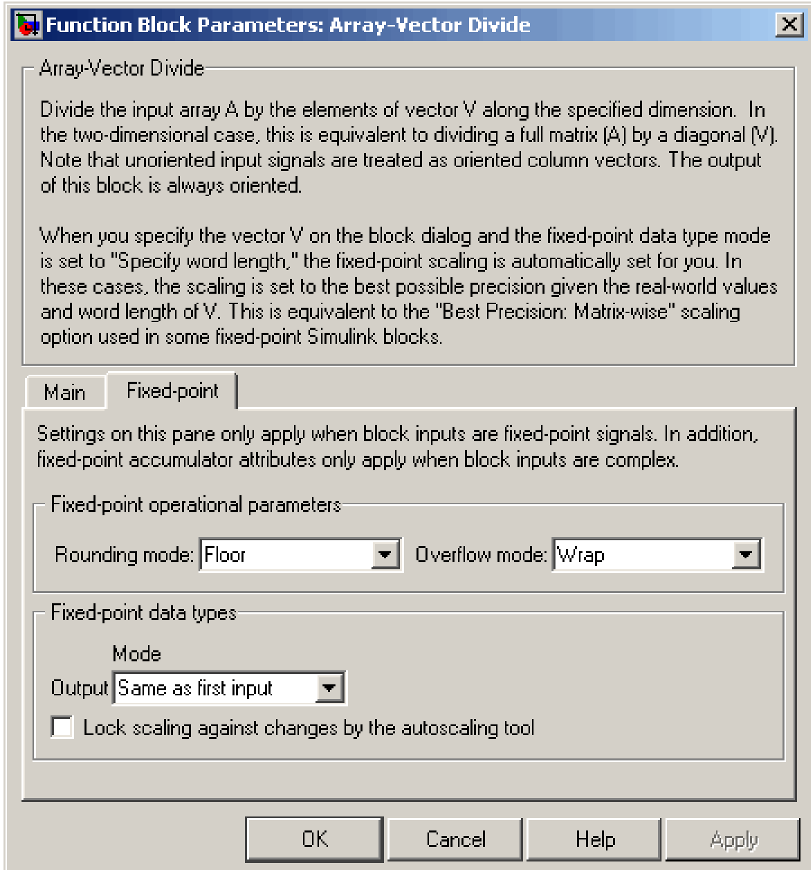
Vector (V) source

Specify the source of the vector, *V*. The vector can come from the `Input port` or from a `Dialog parameter`.

Vector (V)

Specify the vector, *V*. This parameter is visible only when you select `Dialog parameter` for the **Vector (V) source** parameter.

The **Fixed-point** pane of the Array-Vector Divide block dialog appears as follows.



Rounding mode
: Select the rounding mode for fixed-point operations.

Overflow mode
: Select the overflow mode for fixed-point operations.

Vector (V)
: Use this parameter to specify how you would like to designate the word and fraction lengths of the elements of the vector, *V*:

- When you select `Same word length as input`, the word length of the vector values match that of the input to the block.
- When you select `Specify word length`, you can enter the word length of the vector values, in bits. In this mode, the fraction length of the vector values is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the values.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the vector elements, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the vector element. This block requires power-of-two slope and a bias of zero.

Note The **Vector (V)** parameters on the **Fixed-point** pane are only applicable when you specify the vector through the **Vector (V)** parameter on the **Main** pane of the block mask. When the vector comes in through the block port, the data type and scaling of its elements are inherited from the driving block.

Output

Choose how you specify the output word length and fraction length:

- When you select `Same as first input`, these characteristics match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

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

Supported Data Types

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

See Also

Array-Vector Add	Signal Processing Blockset
Array-Vector Multiply	Signal Processing Blockset
Array-Vector Subtract	Signal Processing Blockset

Array-Vector Multiply

Purpose

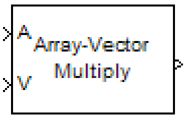
Multiply array by vector along specified dimension

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description



The Array-Vector Multiply block multiplies the values in the specified dimension of the N-dimensional input array A by the values in the input vector V.

The length of the input V must be the same as the length of the specified dimension of A. The Array-Vector Multiply block multiplies each element of V by the corresponding element along that dimension of A.

Consider a 3-dimensional M-by-N-by-P input array $A(i,j,k)$ and a P-by-1 input vector V. When the **Multiply along dimension** parameter is set to `2`, the output of the block $Y(i,j,k)$ is

$$Y(i,j,k) = A(i,j,k) * V(j)$$

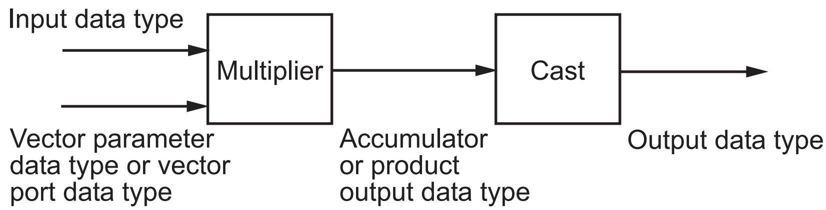
where

$$1 \leq i \leq M$$
$$1 \leq j \leq N$$
$$1 \leq k \leq P$$

The output of the Array-Vector Multiply block is the same size as the input array, A. When both inputs are sample based, the output is sample based; otherwise, the output is frame based. This block accepts real and complex floating-point and fixed-point inputs.

Fixed-Point Data Types

The following diagram shows the data types used within the Array-Vector Multiply block for fixed-point signals.



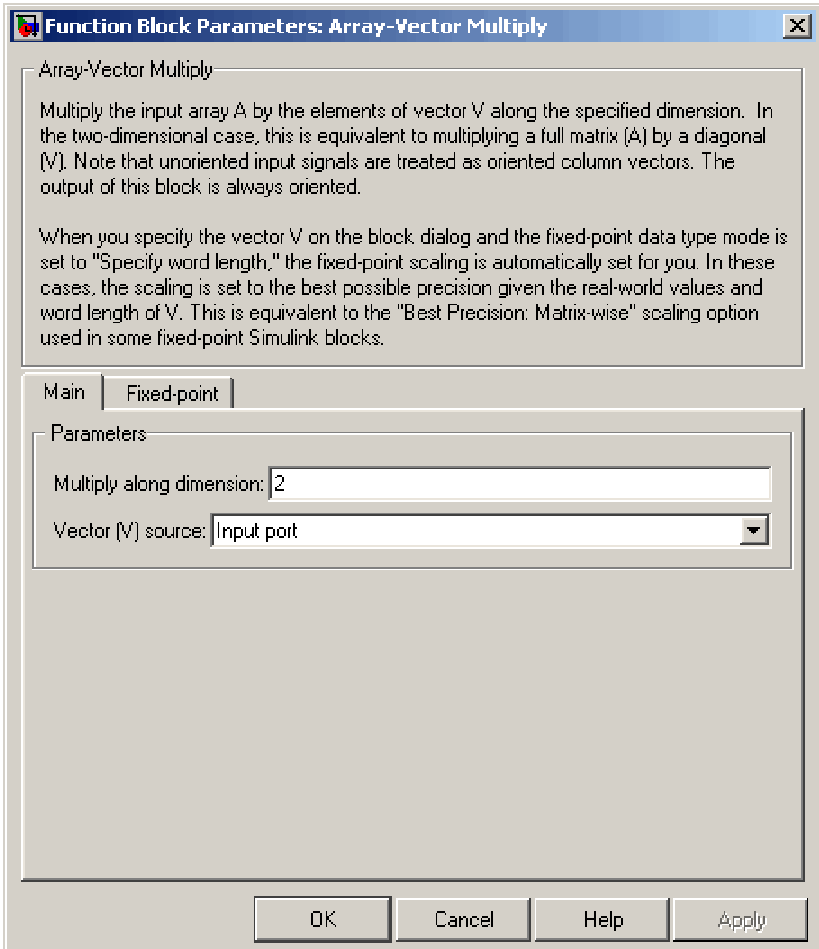
When the vector *V* is designated in the block mask, its elements have the data type and scaling that you specify in the **Vector (V)** parameters on the **Fixed-point** tab. When the vector comes in through the block port, its elements inherit their data type and scaling from the driving block.

The output of the multiplier is in the product output data type when at least one of the inputs to the multiplier is real. When both of the inputs to the multiplier are complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see "Multiplication Data Types".

You can set the vector, accumulator, product output, and output data types in the block dialog as discussed below.

Dialog Box

The **Main** pane of the Array-Vector Multiply block dialog appears as follows.



Multiply along dimension

Specify the dimension along which to multiply the input array A by the elements of vector V.

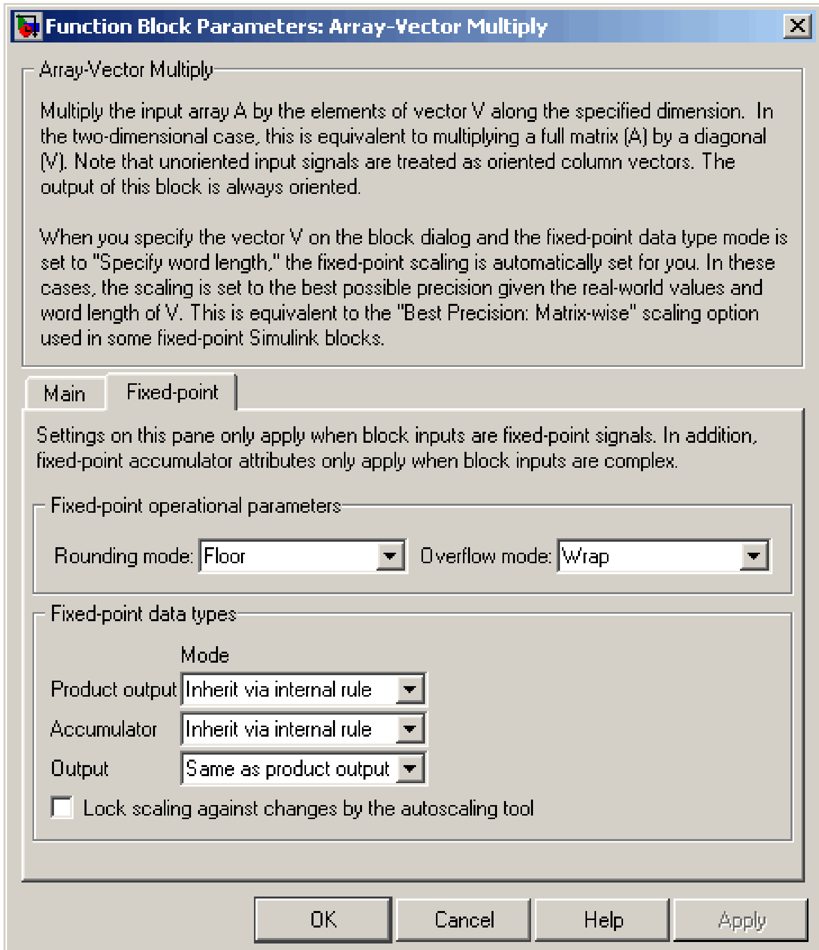
Vector (V) source

Specify the source of the vector, *V*. The vector can come from the `Input port` or from a `Dialog parameter`.

Vector (V)

Specify the vector, *V*. This parameter is visible only when you select `Dialog parameter` for the **Vector (V) source** parameter.

The **Fixed-point** pane of the Array-Vector Multiply block dialog appears as follows.



Rounding mode
: Select the rounding mode for fixed-point operations.

Overflow mode
: Select the overflow mode for fixed-point operations.

Vector (V)

Use this parameter to specify how you would like to designate the word and fraction lengths of the elements of the vector, *V*:

- When you select `Same word length as input`, the word length of the vector values match that of the input to the block.
- When you select `Specify word length`, you can enter the word length of the vector values, in bits. In this mode, the fraction length of the vector values is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the values.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the vector elements, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the vector element. This block requires power-of-two slope and a bias of zero.

Note The **Vector (V)** parameters on the **Fixed-point** pane are only applicable when you specify the vector through the **Vector (V)** parameter on the **Main** pane of the block mask. When the vector comes in through the block port, the data type and scaling of its elements are inherited from the driving block.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-26 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".

- When you select `Same as first input`, these characteristics match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-26 for an illustration depicting the use of the accumulator data type in this block. Note that the accumulator data type is only used when both inputs to the multiplier are complex:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as first input`, these characteristics match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Same as product output`, these characteristics match those of the product.
- When you select `Same as first input`, these characteristics match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

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

Supported Data Types

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

See Also

Array-Vector Add	Signal Processing Blockset
Array-Vector Divide	Signal Processing Blockset
Array-Vector Subtract	Signal Processing Blockset

Purpose

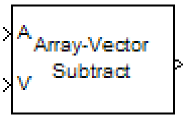
Subtract vector from array along specified dimension

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description



The Array-Vector Subtract block subtracts the values in the input vector *V* from the values in the specified dimension of the N-dimensional input array *A*.

The length of the input *V* must be the same as the length of the specified dimension of *A*. The Array-Vector Subtract block subtracts each element of *V* from the corresponding element along that dimension of *A*.

Consider a 3-dimensional *M*-by-*N*-by-*P* input array *A(i,j,k)* and a *P*-by-1 input vector *V*. When the **Subtract along dimension** parameter is set to `2`, the output of the block *Y(i,j,k)* is

$$Y(i,j,k) = A(i,j,k) - V(j)$$

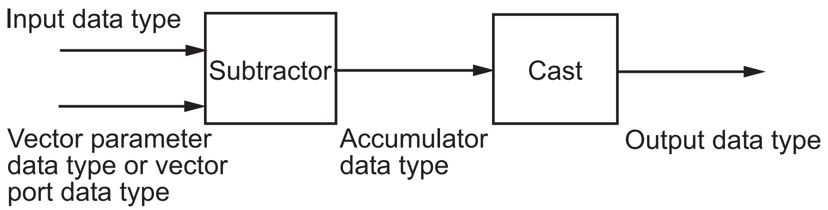
where

$$1 \le i \le M$$
$$1 \le j \le N$$
$$1 \le k \le P$$

The output of the Array-Vector Subtract block is the same size as the input array, *A*. When both inputs are sample based, the output is sample based; otherwise, the output is frame based. This block accepts real and complex floating-point and fixed-point inputs.

Fixed-Point Data Types

The following diagram shows the data types used within the Array-Vector Subtract block for fixed-point signals.



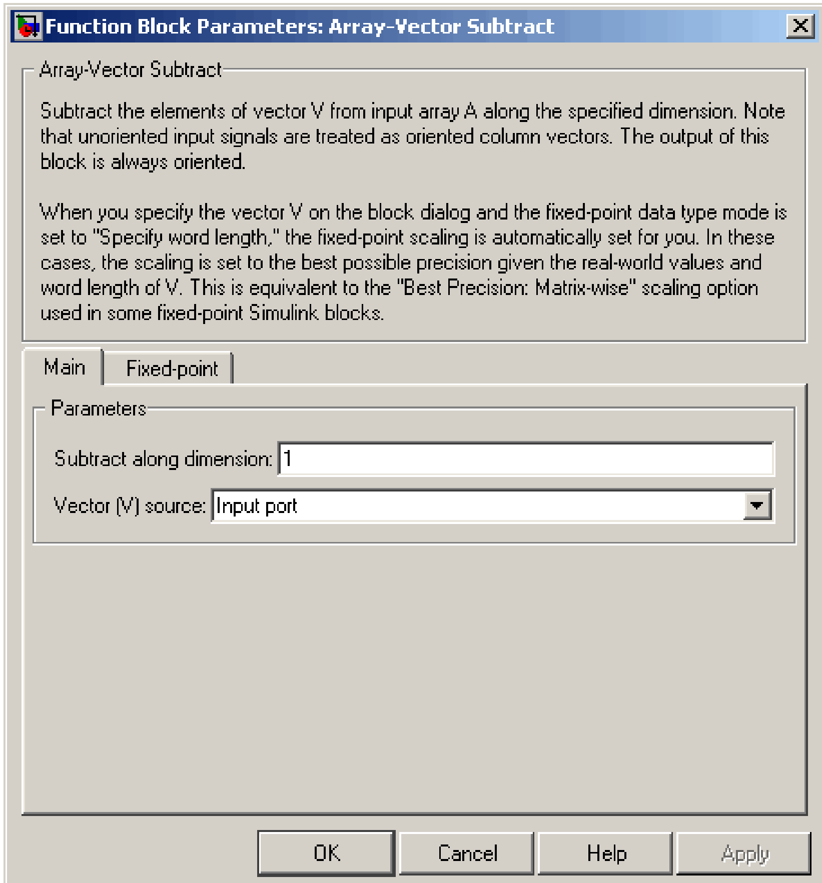
When the vector *V* is designated in the block mask, its elements have the data type and scaling that you specify in the **Vector (V)** parameters on the **Fixed-point** tab. When the vector comes in through the block port, its elements inherit their data type and scaling from the driving block.

The output of the subtractor is in the accumulator data type.

You can set the vector, accumulator, and output data types in the block dialog as discussed below.

Dialog Box

The **Main** pane of the Array-Vector Subtract block dialog appears as follows.



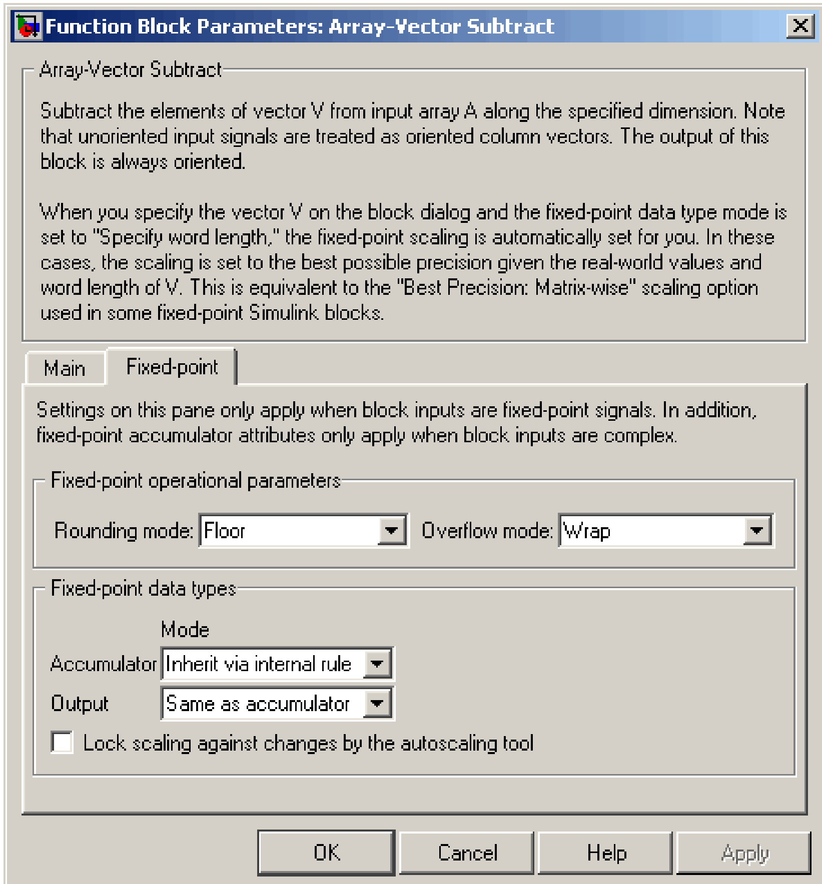
Subtract along dimension

Specify the dimension along which to subtract the elements of vector V from the input array A.

Vector (V) source
: Specify the source of the vector, *V*. The vector can come from the `Input port` or from a `Dialog parameter`.

Vector (V)
: Specify the vector, *V*. This parameter is visible only when you select `Dialog parameter` for the **Vector (V) source** parameter.

The **Fixed-point** pane of the Array-Vector Subtract block dialog appears as follows.



Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Vector (V)
Use this parameter to specify how you would like to designate the word and fraction lengths of the elements of the vector, *V*:

- When you select `Same word length as input`, the word length of the vector values match that of the input to the block.
- When you select `Specify word length`, you can enter the word length of the vector values, in bits. In this mode, the fraction length of the vector values is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the values.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the vector elements, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the vector element. This block requires power-of-two slope and a bias of zero.

Note The **Vector (V)** parameters on the **Fixed-point** pane are only applicable when you specify the vector through the **Vector (V)** parameter on the **Main** pane of the block mask. When the vector comes in through the block port, the data type and scaling of its elements are inherited from the driving block.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-35 for an illustration depicting the use of the accumulator data type in this block.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as first input`, these characteristics match those of the first input to the block.

- When you select Binary point scaling, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select Slope and bias scaling, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output
Choose how you specify the output word length and fraction length:

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

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

Supported Data Types

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

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

See Also

Array-Vector Add	Signal Processing Blockset
Array-Vector Divide	Signal Processing Blockset
Array-Vector Multiply	Signal Processing Blockset

Purpose

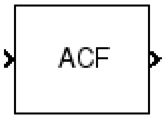
Compute autocorrelation of vector inputs

Library

Statistics

`dspstat3`

Description



The Autocorrelation block computes the autocorrelation along each column of a frame-based input, and computes along the first dimension of an N-D sample-based input. The output of the block is always sample-based.

The Autocorrelation block accepts both real and complex fixed-point and floating-point inputs. Fixed-point signals are not supported for the frequency domain.

Autocorrelation of Frame-Based Inputs

When the input to the Autocorrelation block is a frame-based M-by-N matrix u, the output, y, is a sample-based (l+1)-by-N matrix whose jth column has elements

$$y_{i,j} = \sum_{k=1}^{M} u^{*}_{k,j} u_{(k+i-1),j} \qquad 1 \le i \le (l+1)$$

where * denotes the complex conjugate, and l represents the maximum lag. $y_{1,j}$ is the zero-lag element in the jth column. When you select **Compute all non-negative lags**, l=M–1. Otherwise, l is specified as a nonnegative integer by the **Maximum non-negative lag (less than input length)** parameter.

Input u is zero when indexed outside of its valid range. When the input is real, the output is real; otherwise, the output is complex.

Autocorrelation of Sample-Based Inputs

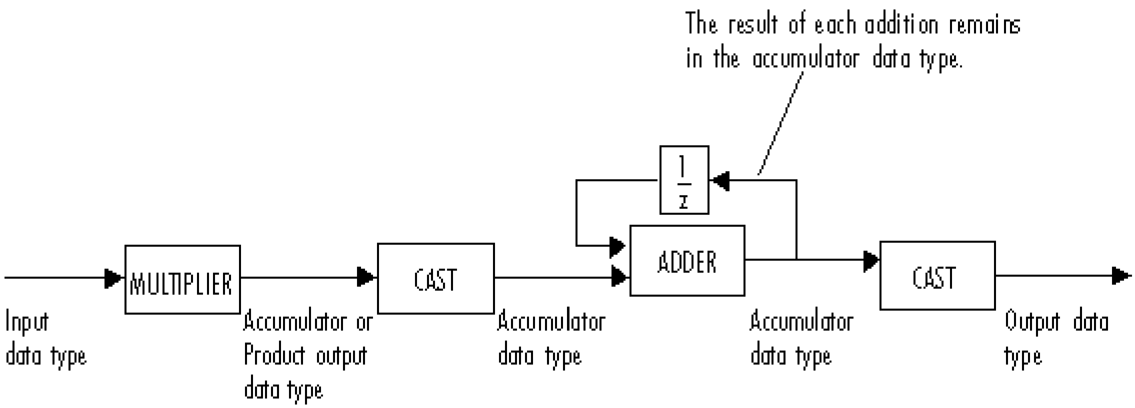
When the input is a sample-based N-D array, the block computes the autocorrelation along the first dimension of the input. The output is a sample based N-D array, where the size of the first dimension is l+1, and the sizes of all other dimensions match those of the input array. For example, when the input is an M-by-N-by-P array, the Autocorrelation block outputs an $(l+1)$-by-N-by-P sample-based array.

When you select **Compute all non-negative lags**, $l=M-1$. Otherwise, l is specified as a nonnegative integer by the **Maximum non-negative lag (less than input length)** parameter.

Input u is zero when indexed outside of its valid range. When the input is real, the output is real; otherwise, the output is complex.

Fixed-Point Data Types

The following diagram shows the data types used within the Autocorrelation block for fixed-point signals (time domain only).

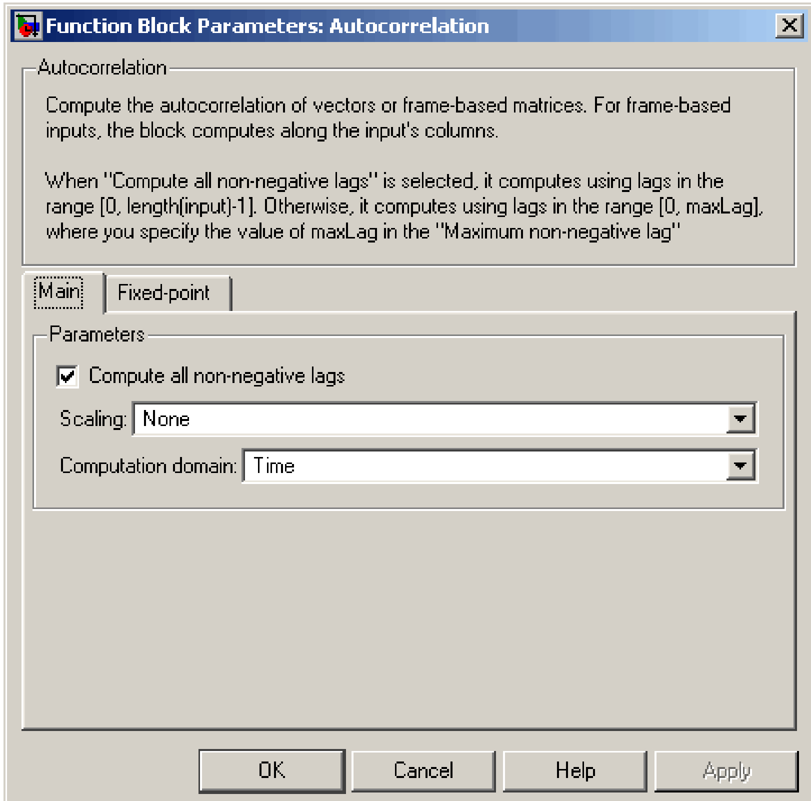


You can set the product output, accumulator, and output data types in the Fixed-Point pane of the block dialog as discussed in the next section.

The output of the multiplier is in the product output data type when the input is real. When the input is complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see "Multiplication Data Types".

Dialog Box

The **Main** pane of the Autocorrelation block dialog appears as follows.



Compute all non-negative lags
: Select to compute the autocorrelation over all nonnegative lags in the range [0, `length(input)-1`].

Maximum non-negative lag (less than input length)
: Specify the maximum positive lag, l, for the autocorrelation. This parameter is enabled when you do not select the **Compute all non-negative lags** check box.

Scaling

This parameter controls the scaling that is applied to the output. The following options are available:

- `None` — Generates the raw autocorrelation $y_{i,j}$ without normalization.
- `Biased` — Generates the biased estimate of the autocorrelation.

  $$y_{i,j}^{biased} = \frac{y_{i,j}}{M}$$

- `Unbiased` — Generates the unbiased estimate of the autocorrelation.

  $$y_{i,j}^{unbiased} = \frac{y_{i,j}}{M-i}$$

- `Unity at zero-lag` — Normalizes the estimate of the autocorrelation for each channel so that the zero-lag sum is identically 1.

  $$y_{1,j} = 1$$

Note The **Scaling** parameter must be set to `None` for fixed-point signals.

This parameter is tunable, except in the Simulink® external mode.

Computation domain

This parameter sets the domain in which the block computes convolutions to one of the following settings:

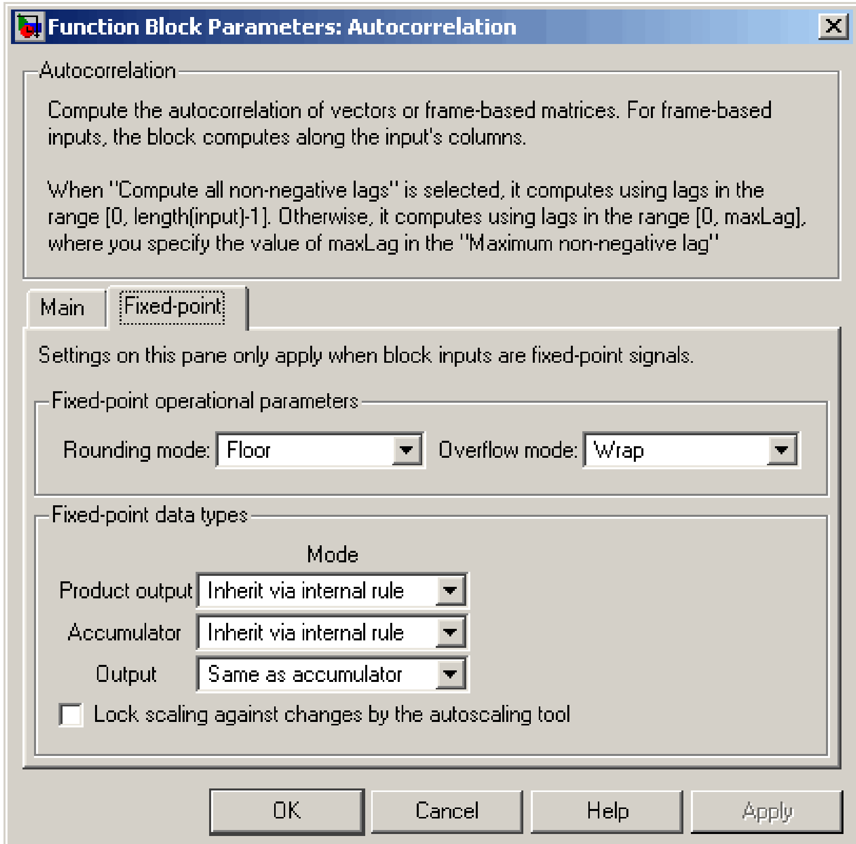
- `Time` — Computes in the time domain, which minimizes memory use

- `Frequency` — Computes in the frequency domain, which might require fewer computations than computing in the time domain, depending on the input length

---

Note This parameter must be set to `Time` for fixed-point signals.

The **Fixed-point** pane of the Autocorrelation block dialog appears as follows.



Note Fixed-point signals are only supported for the time domain. To use the parameters on this pane, make sure `Time` is selected for the **Computation domain** parameter on the **Main** pane.

Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

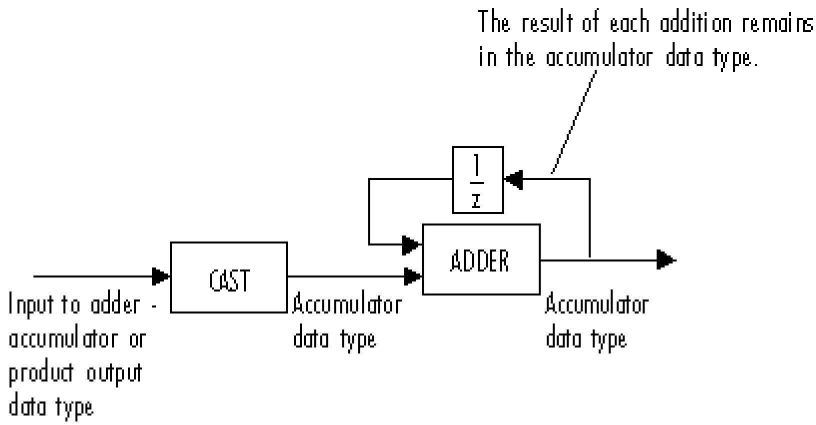
Select the overflow mode for fixed-point operations.

Product output

Use this parameter to specify how you want to designate the product output word and fraction lengths. See “Fixed-Point Data Types” on page 2-44 and “Multiplication Data Types” for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see “Inherit via Internal Rule”.
- When you select `Same as input`, these characteristics match those of the block input.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator



As depicted in this figure, inputs to the accumulator are cast to the accumulator data type. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you want to designate this accumulator word and fraction lengths.

You also use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. See “Multiplication Data Types” for more information.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see “Inherit via Internal Rule”.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the block input.

- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.

  A special case occurs when `Inherit via internal rule` is specified for **Accumulator**, and the block input is complex. In that case, the output word length is one less than the accumulator word length.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

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

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)

- 8-, 16-, and 32-bit signed integers

See Also

Correlation	Signal Processing Blockset
`xcorr`	Signal Processing Toolbox

Purpose

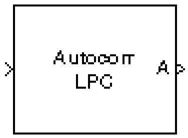
Determine coefficients of Nth-order forward linear predictors

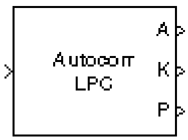
Library

Estimation / Linear Prediction

`dsplp`

Description





The Autocorrelation LPC block determines the coefficients of an *N-step forward linear predictor* for the time-series in each length-M input channel, u, by minimizing the prediction error in the least squares sense. A linear predictor is an FIR filter that predicts the next value in a sequence from the present and past inputs. This technique has applications in filter design, speech coding, spectral analysis, and system identification.

The Autocorrelation LPC block can output the prediction error for each channel as polynomial coefficients, reflection coefficients, or both. It can also output the prediction error power for each channel. The input u can be a scalar, 1-D vector, frame- or sample-based column vector, a sample-based row vector, or a channel-based matrix. Frame-based row vectors are not valid inputs. Frame-based matrices of size M-by-N are treated as N channels of length M.

When you select **Inherit prediction order from input dimensions**, the prediction order, N, is inherited from the input dimensions. Otherwise, you can use the **Prediction order** parameter to specify the value of N. Note that N must be a scalar with a value less than the length of the input channels or the block produces an error.

When **Output(s)** is set to `A`, port A is enabled. For each channel, port A outputs an (N+1)-by-1 column vector, $a = [1\ a_2\ a_3\ \ldots\ a_{N+1}]^T$, containing the coefficients of an Nth-order moving average (MA) linear process that predicts the next value, $\hat{u}_{M+1}$, in the input time-series.

$$\hat{u}_{M+1} = -(a_2 u_M) - (a_3 u_{M-1}) - \ldots - (a_{N+1} u_{M-N+1})$$

When **Output(s)** is set to `K`, port K is enabled. For each channel, port K outputs a length-N column vector whose elements are the prediction error reflection coefficients. When **Output(s)** is set to `A and K`, both port A and K are enabled, and each port outputs its respective set of prediction coefficients for each channel.

When you select **Output prediction error power (P)**, port P is enabled. The prediction error power is output at port P as a vector whose length is the number of input channels.

Algorithm

The Autocorrelation LPC block computes the least squares solution to

$$\min_{i \in \Re^n} \|U\tilde{a} - b\|$$

where $\|\cdot\|$ indicates the 2-norm and

$$U = \begin{bmatrix} u_1 & 0 & \cdots & 0 \\ u_2 & u_1 & \ddots & \vdots \\ \vdots & u_2 & \ddots & 0 \\ \vdots & \vdots & \ddots & u_1 \\ \vdots & \vdots & \vdots & u_2 \\ \vdots & \vdots & \vdots & \vdots \\ u_M & \vdots & \vdots & \vdots \\ 0 & \ddots & \vdots & \vdots \\ \vdots & \ddots & \ddots & \vdots \\ 0 & \cdots & 0 & u_M \end{bmatrix}, \tilde{a} = \begin{bmatrix} a_2 \\ \vdots \\ a_n + 1 \end{bmatrix}, b = \begin{bmatrix} u_2 \\ u_3 \\ \vdots \\ u_M \\ 0 \\ \vdots \\ 0 \end{bmatrix}$$

Solving the least squares problem via the normal equations

$$U^* U\tilde{a} = U^* b$$

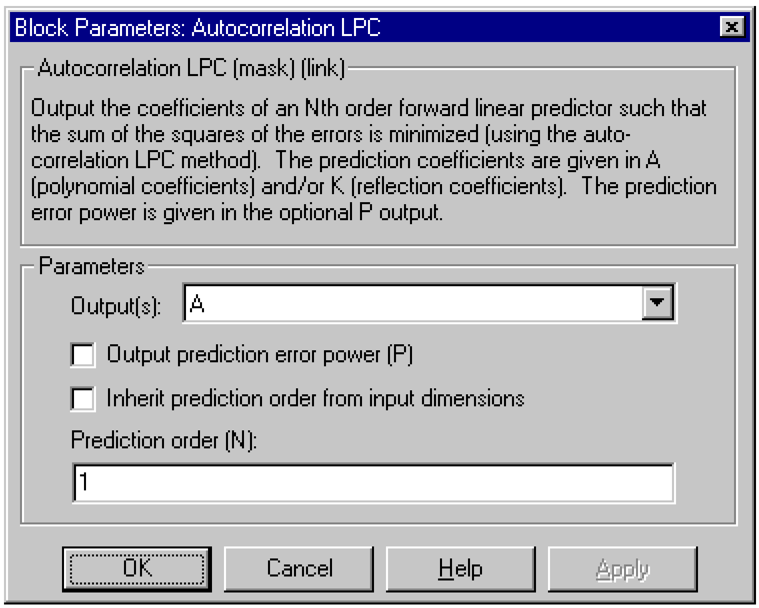
leads to the system of equations

$$\begin{bmatrix} r_1 & r_2^* & \cdots & r_n^* \\ r_2 & r_1 & \ddots & \vdots \\ \vdots & \ddots & \ddots & r_2^* \\ r_n & \cdots & r_2 & r_1 \end{bmatrix} \begin{bmatrix} a_2 \\ a_3 \\ \vdots \\ a_{n+1} \end{bmatrix} = \begin{bmatrix} -r_2 \\ -r_3 \\ \vdots \\ -r_{n+1} \end{bmatrix}$$

where $r = [r_1\ r_2\ r_3\ \ldots\ r_{n+1}]^T$ is an autocorrelation estimate for u computed using the Autocorrelation block, and * indicates the complex conjugate transpose. The normal equations are solved in $O(n^2)$ operations by the Levinson-Durbin block.

Note that the solution to the LPC problem is very closely related to the Yule-Walker AR method of spectral estimation. In that context, the normal equations above are referred to as the Yule-Walker AR equations.

Dialog Box



Output(s)
The type of prediction coefficients output by the block. The block can output polynomial coefficients (`A`), reflection coefficients (`K`), or both (`A and K`).

Output prediction error power (P)
When selected, enables port `P`, which outputs the output prediction error power.

Inherit prediction order from input dimensions
When selected, the block inherits the prediction order from the input dimensions.

Prediction order (N)
Specify the prediction order, *N*, which must be a scalar. This parameter is disabled when you select the **Inherit prediction order from input dimensions** parameter.

References

Haykin, S. *Adaptive Filter Theory*. 3rd ed. Englewood Cliffs, NJ: Prentice Hall, 1996.

Ljung, L. *System Identification: Theory for the User*. Englewood Cliffs, NJ: Prentice Hall, 1987. Pgs. 278-280.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Autocorrelation	Signal Processing Blockset
Levinson-Durbin	Signal Processing Blockset
Yule-Walker Method	Signal Processing Blockset
`lpc`	Signal Processing Toolbox

Purpose

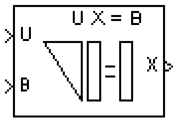
Solve $UX=B$ for X when U is upper triangular matrix

Library

Math Functions / Matrices and Linear Algebra / Linear System Solvers

`dspsolvers`

Description



The Backward Substitution block solves the linear system $UX=B$ by simple backward substitution of variables, where U is the upper triangular M-by-M matrix input to the U port, and B is the M-by-N matrix input to the B port. The output is the solution of the equations, the M-by-N matrix X, and is always sample based. The block does not check the rank of the inputs.

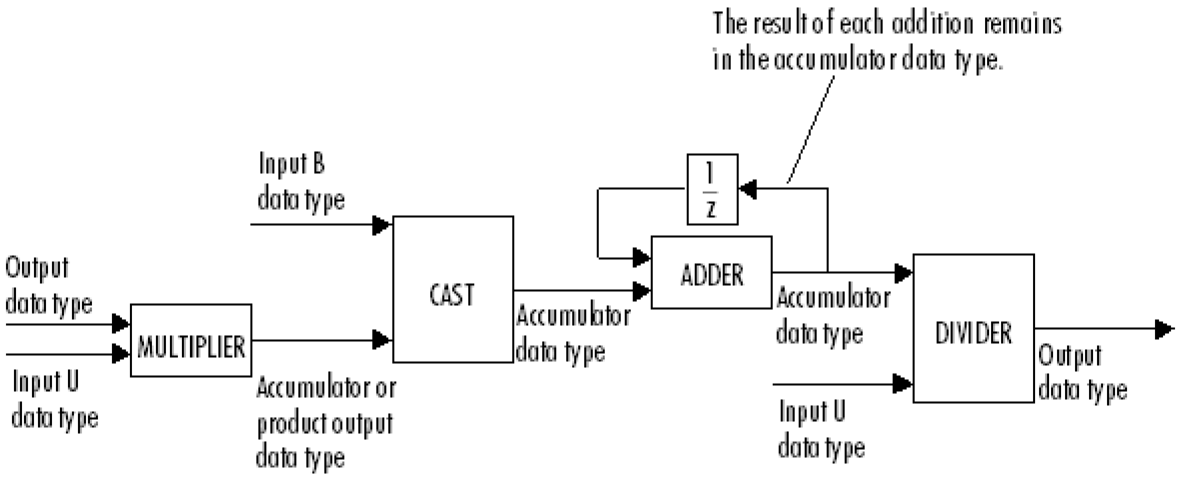
The block uses only the elements in the *upper triangle* of input U; the lower elements are ignored. When you select the **Input U is unit-upper triangular** check box, the block replaces the elements on the diagonal of U with 1's. This is useful when matrix U is the result of another operation, such as an LDL decomposition, that uses the diagonal elements to represent the D matrix.

A length-M vector input at port B is treated as an M-by-1 matrix.

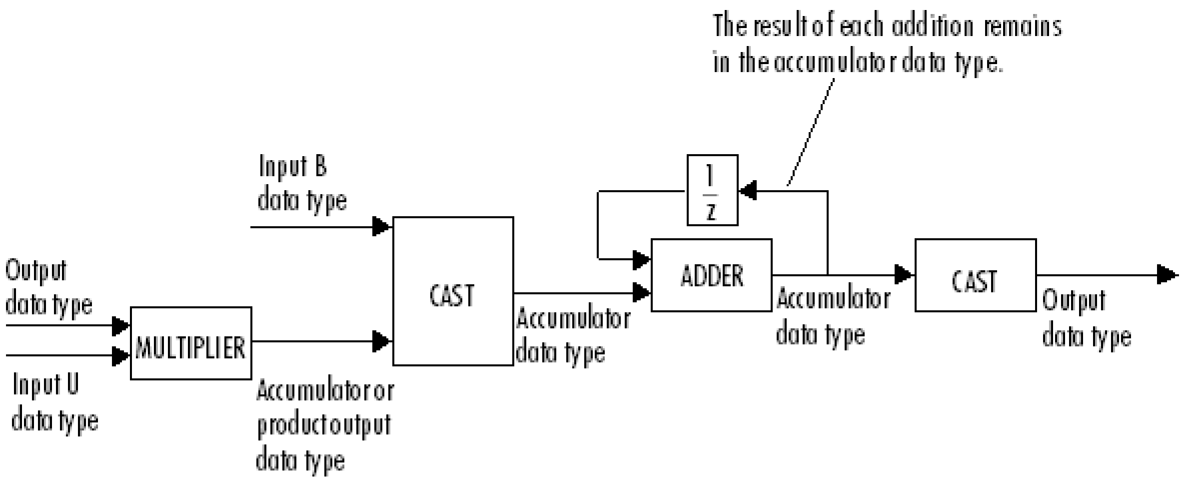
Fixed-Point Data Types

The following diagram shows the data types used within the Backward Substitution block for fixed-point signals.

When input U is not unit-upper triangular:



When input U is unit-upper triangular:

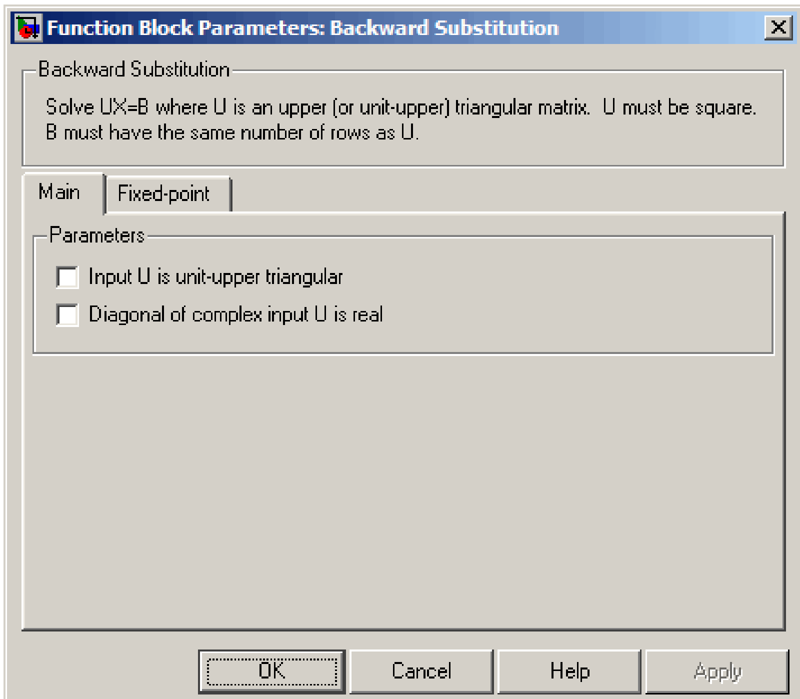


You can set the product output, accumulator, and output data types in the block dialog as discussed below.

The output of the multiplier is in the product output data type when the input is real. When the input is complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”.

Dialog Box

The **Main** pane of the Backward Substitution block dialog appears as follows.



Input U is unit-upper triangular

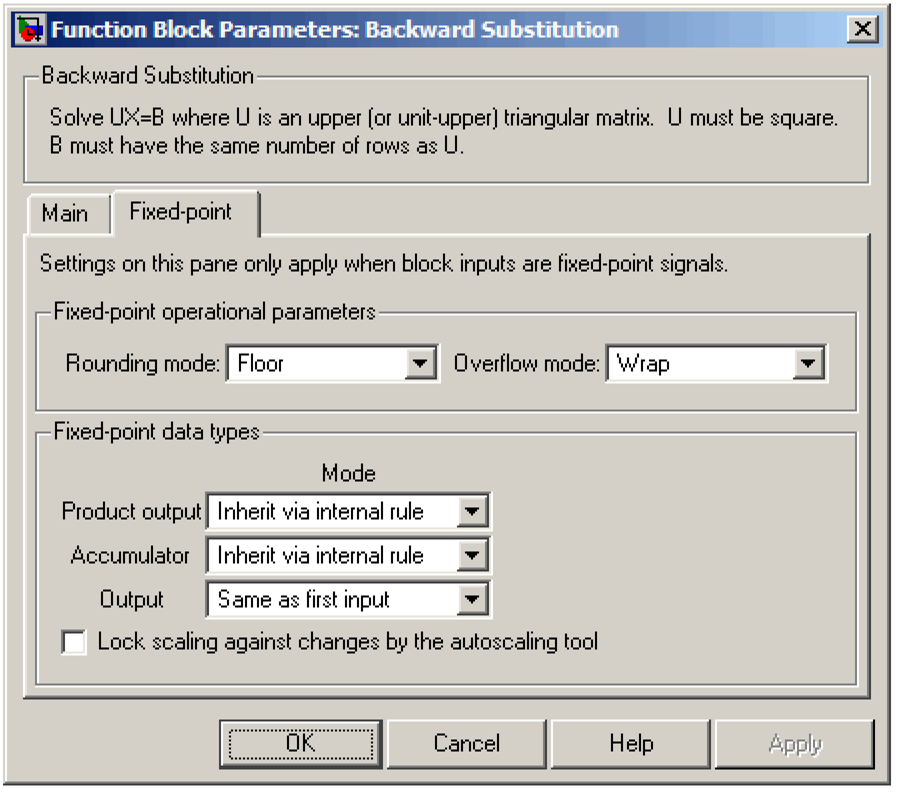
Select to replace the elements on the diagonal of U with 1’s.

Diagonal of complex input U is real

Select to optimize simulation speed when the diagonal elements of complex input U are real. This parameter is only visible when **Input U is unit-upper triangular** is not selected.

Note When U is a complex fixed-point signal, you must select either **Input U is unit-upper triangular** or **Diagonal of complex input U is real**. In such a case, any imaginary part of the diagonal of U is ignored.

The **Fixed-point** pane of the Backward Substitution block dialog appears as follows.



Rounding mode
: Select the rounding mode for fixed-point operations.

Overflow mode
: Select the overflow mode for fixed-point operations.

Product output
: Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point

Data Types" on page 2-58 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as first input`, these characteristics match those of the input U to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-58 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as first input`, these characteristics match those of the input U to the block.

- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Use this parameter to specify how you would like to designate the output word and fraction lengths. See "Fixed-Point Data Types" on page 2-58 for an illustration depicting the use of the output data type in this block:

- When you select `Same as first input`, these characteristics match those of the input U to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

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

Supported Data Types

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

Port	Supported Data Types
B	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
X	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers

See Also

Cholesky Solver	Signal Processing Blockset
Forward Substitution	Signal Processing Blockset
LDL Solver	Signal Processing Blockset
Levinson-Durbin	Signal Processing Blockset
LU Solver	Signal Processing Blockset
QR Solver	Signal Processing Blockset

See "Linear System Solvers" for related information.

Purpose

Design bandpass filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description

This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See "Bandpass Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Design bandstop filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description

This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See "Bandstop Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Apply cascade of biquadratic (second-order section) filters to input

Library

`dspobslib`

Description

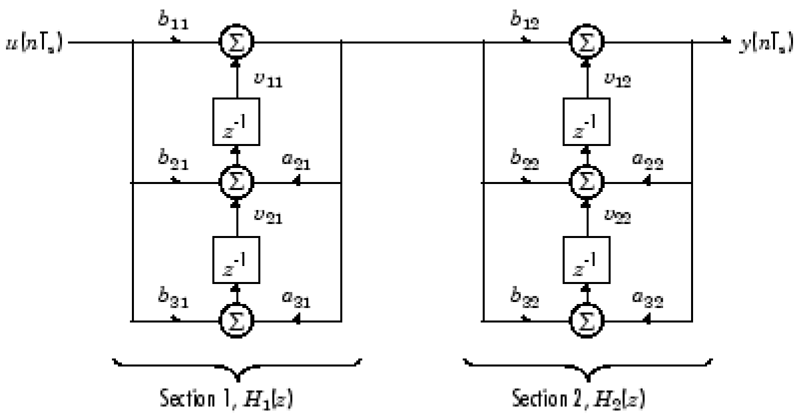
Note The Biquadratic Filter block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Biquadratic Filter block applies a cascade of biquadratic filters independently to each input channel. Biquadratic filters are useful for reduced precision implementations because the coefficients are bounded between ±2 for typical minimum-phase designs. This may reduce scaling and coefficient sensitivity problems.

The filter is constructed from L second-order sections, each having a quadratic numerator and denominator.

$$H(z) = \prod_{k=1}^{L} H_k(z) = \prod_{k=1}^{L} \frac{b_{1k} + b_{2k}z^{-1} + b_{3k}z^{-2}}{a_{1k} + a_{2k}z^{-1} + a_{3k}z^{-2}}$$

The figure below illustrates the structure of a 4th-order biquadratic filter (L=2) with states v_{ik}, where k is the section number.



An M-by-N sample-based matrix input is treated as $M*N$ independent channels, and an M-by-N frame-based matrix input is treated as N independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

The **SOS matrix** parameter specifies the filter coefficients as a second-order section matrix of the type produced by the Signal Processing Toolbox™ functions `ss2sos` and `tf2sos`.

$$\begin{bmatrix} b_{11} & b_{21} & b_{31} & a_{11} & a_{21} & a_{31} \\ b_{12} & b_{22} & b_{32} & a_{12} & a_{22} & a_{32} \\ \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\ b_{1L} & b_{2L} & b_{3L} & a_{1L} & a_{2L} & a_{3L} \end{bmatrix}$$

$$a_{11} = a_{12} = \ldots = a_{1L}$$

This is an L-by-6 matrix whose rows contain the numerator and denominator coefficients b_{ik} and a_{ik} of each second-order section in $H(z)$. Use the `ss2sos` and `tf2sos` functions to convert a state-space or transfer-function description of the filter into the second-order section description used by this block. Note that the filter uses a value of 1 for the zero-delay denominator coefficients (a_{11} to a_{1L}) regardless of the value specified in the **SOS matrix** parameter.

The **Initial conditions** parameter sets the initial filter states, and can be specified in the following different forms:

- *Scalar* to be used for all filter states (v_{11}, v_{12}, ..., v_{1L}, v_{21} v_{22}, ..., v_{2L}) in all channels. An empty vector, `[]`, is the same as the scalar value `0`.
- *Vector* of length 2*L (row or column) to initialize the filter states for all channels.

$$[\underbrace{v_{11}\ v_{21}}_{H_1(z)}\ \underbrace{v_{12}\ v_{22}}_{H_2(z)}\ \cdots\ \underbrace{v_{1L}\ v_{2L}}_{H_L(z)}]$$

  Each pair of elements specifies v_1k and v_2k for second-order section k in every channel.

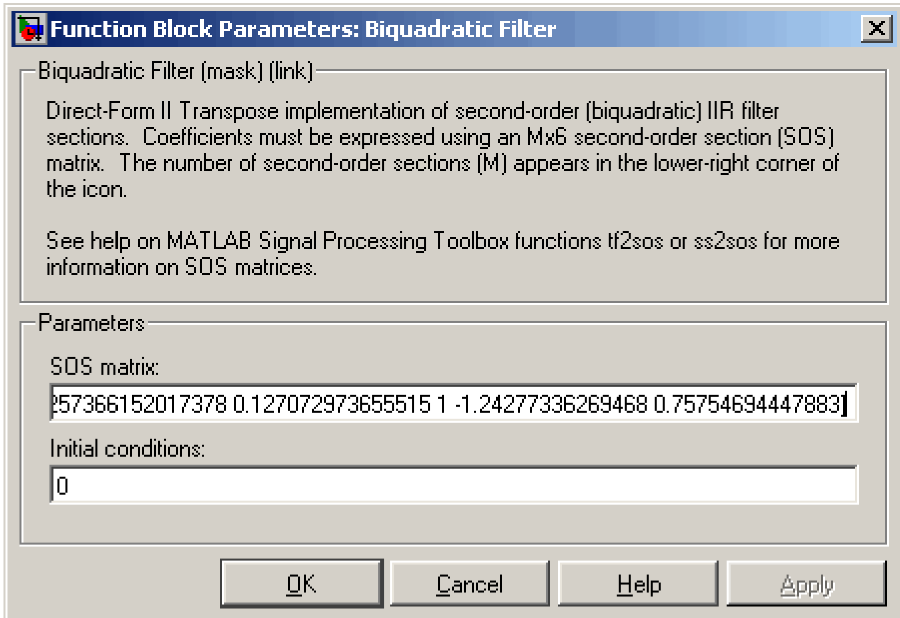
- *Matrix* of dimension (2*L)-by-N containing the initial filter states for each of the N channels.

  +

$$\begin{array}{c} H_1(z)\left\{\begin{array}{c} \\ \\ \end{array}\right. \\ H_2(z)\left\{\begin{array}{c} \\ \\ \end{array}\right. \\ \\ H_L(z)\left\{\begin{array}{c} \\ \\ \end{array}\right. \end{array}
\begin{bmatrix} v_{11}^{ch1} & v_{11}^{ch2} & \cdots & v_{11}^{chN} \\ v_{21}^{ch1} & v_{21}^{ch2} & \cdots & v_{21}^{chN} \\ v_{12}^{ch1} & v_{12}^{ch2} & \cdots & v_{12}^{chN} \\ v_{22}^{ch1} & v_{22}^{ch2} & \cdots & v_{22}^{chN} \\ \vdots & \vdots & \ddots & \vdots \\ v_{1L}^{ch1} & v_{1L}^{ch2} & \cdots & v_{1L}^{chN} \\ v_{2L}^{ch1} & v_{2L}^{ch2} & \cdots & v_{2L}^{chN} \end{bmatrix}$$

  Each pair of elements in a *column* specifies v_1k and v_2k for second-order section k of the corresponding channel.

Dialog Box



SOS matrix

The second-order section matrix specifying the filter's coefficients. This matrix can be generated from state-space or transfer-function descriptions by using the Signal Processing Toolbox functions `ss2sos` and `tf2sos`.

Initial conditions

The filter's initial conditions, a scalar, vector, or matrix.

Purpose

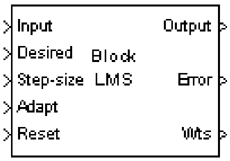
Compute filtered output, filter error, and filter weights for given input and desired signal using Block LMS adaptive filter algorithm

Library

Filtering / Adaptive Filters

`dspadpt3`

Description



Block LMS Filter

The Block LMS Filter block implements an adaptive least mean-square (LMS) filter, where the adaptation of filter weights occurs once for every block of samples. The block estimates the filter weights, or coefficients, needed to minimize the error, $e(n)$, between the output signal, $y(n)$, and the desired signal, $d(n)$. Connect the signal you want to filter to the Input port. This input signal can be a sample-based scalar or a single-channel frame-based signal. Connect the signal you want to model to the `Desired` port. The desired signal must have the same data type, frame status, complexity, and dimensions as the input signal. The `Output` port outputs the filtered input signal, which can be sample or frame based. The `Error` port outputs the result of subtracting the output signal from the desired signal.

The block calculates the filter weights using the Block LMS adaptive filter algorithm. This algorithm is defined by the following equations.

$$\begin{aligned} n &= kN + i \\ y(n) &= \mathbf{w}^T(k-1)\mathbf{u}(n) \\ e(n) &= d(n) - y(n) \\ \mathbf{w}(k) &= \mathbf{w}(k-1) + f(\mathbf{u}(n), e(n), \mu) \end{aligned}$$

The weight update function for the Block LMS adaptive filter algorithm is defined as

$$f(\mathbf{u}(n), e(n), \mu) = \mu \sum_{i=0}^{N-1} \mathbf{u}^*(kN+i)e(kN+i)$$

The variables are as follows.

Variable	Description
n	The current time index
i	The iteration variable in each block, $0 \le i \le N-1$
k	The block number
N	The block size
$\mathbf{u}(n)$	The vector of buffered input samples at step n
$\mathbf{w}(n)$	The vector of filter-tap estimates at step n
$y(n)$	The filtered output at step n
$e(n)$	The estimation error at time n
$d(n)$	The desired response at time n
μ	The adaptation step size

Use the **Filter length** parameter to specify the length of the filter weights vector.

The **Block size** parameter determines how many samples of the input signal are acquired before the filter weights are updated. The input frame length must be a multiple of the **Block size** parameter.

The adaptation **Step-size (mu)** parameter corresponds to µ in the equations. You can either specify a step-size using the input port, Step-size, or enter a value in the Block Parameters: Block LMS Filter dialog box.

Use the **Leakage factor (0 to 1)** parameter to specify the leakage factor, $0 < 1 - \mu\alpha \le 1$, in the leaky LMS algorithm shown below.

$$\mathbf{w}(k) = (1 - \mu\alpha)\mathbf{w}(k-1) + f(\mathbf{u}(n), e(n), \mu)$$

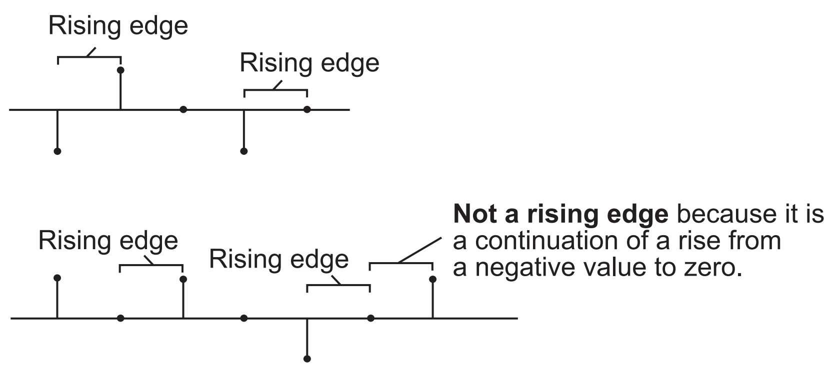
Enter the initial filter weights as a vector or a scalar in the **Initial value of filter weights** text box. When you enter a scalar, the block uses the scalar value to create a vector of filter weights. This vector has length equal to the filter length and all of its values are equal to the scalar value

When you select the **Adapt port** check box, an `Adapt` port appears on the block. When the input to this port is greater than zero, the block continuously updates the filter weights. When the input to this port is zero, the filter weights remain at their current values.

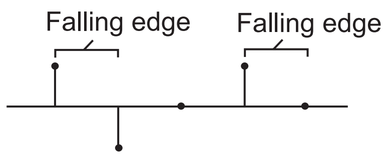
When you want to reset the value of the filter weights to their initial values, use the **Reset input** parameter. The block resets the filter weights whenever a reset event is detected at the Reset port. The reset signal rate must be the same rate as the data signal input.

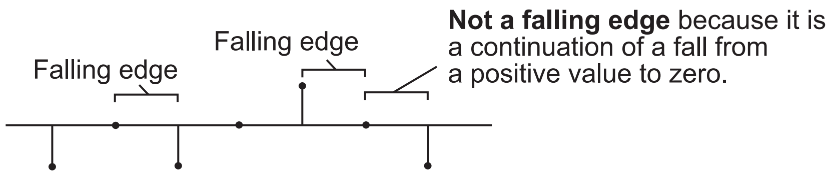
From the **Reset input** list, select `None` to disable the Reset port. To enable the Reset port, select one of the following from the **Reset input** list:

- `Rising edge` — Triggers a reset operation when the Reset input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure).



- `Falling edge` — Triggers a reset operation when the Reset input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)





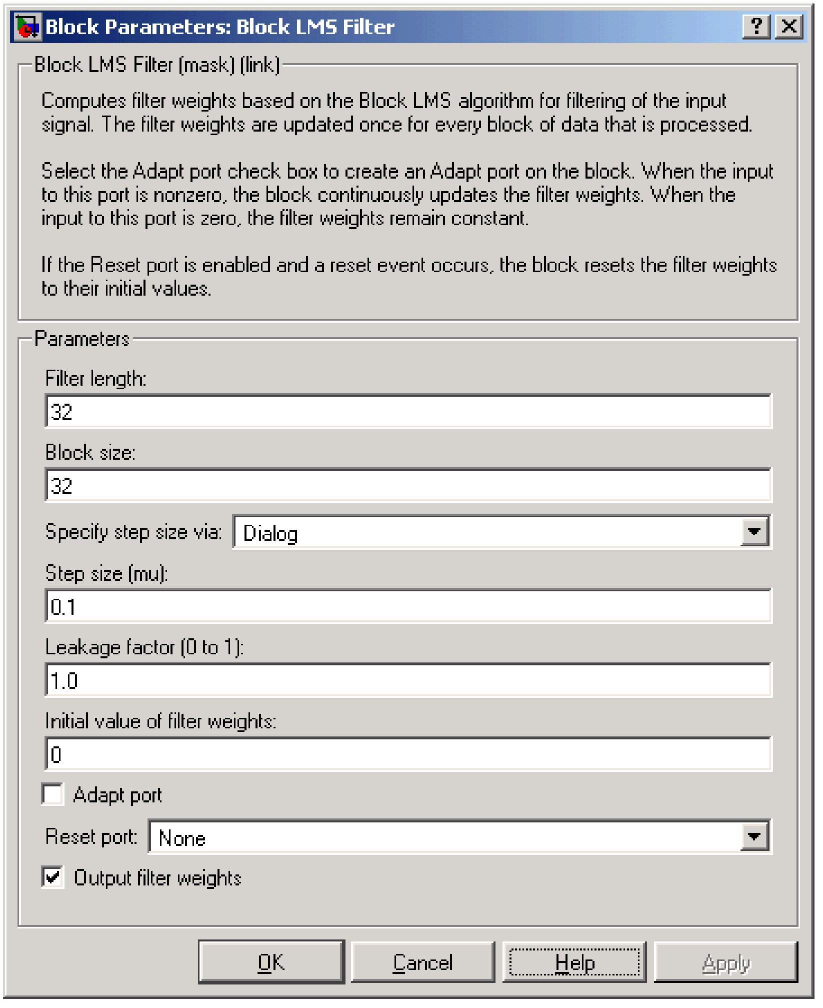
- `Either edge` — Triggers a reset operation when the Reset input is a `Rising edge` or `Falling edge` (as described above)
- `Non-zero sample` — Triggers a reset operation at each sample time that the Reset input is not zero

---

Note When running simulations in the Simulink® `MultiTasking` mode, sample-based reset signals have a one-sample latency, and frame-based reset signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a reset event, and when it applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Select the **Output filter weights** check box to create a `Wts` port on the block. For each iteration, the block outputs the current updated filter weights from this port.

Dialog Box



Filter length
Enter the length of the FIR filter weights vector.

Block size
Enter the number of samples to acquire before the filter weights are updated. The input frame length must be an integer multiple of the block size.

Specify step-size via
Select `Dialog` to enter a value for mu in the Block parameters: LMS Filter dialog box. Select `Input port` to specify mu using the Step-size input port.

Step-size (mu)
Enter the step-size. Tunable.

Leakage factor (0 to 1)
Enter the leakage factor, $0 < 1 - \mu\alpha \leq 1$. Tunable.

Initial value of filter weights
Specify the initial values of the FIR filter weights.

Adapt port
Select this check box to enable the Adapt input port.

Reset input
Select this check box to enable the Reset input port.

Output filter weights
Select this check box to export the filter weights from the Wts port.

References

Hayes, M. H. *Statistical Digital Signal Processing and Modeling.* New York: John Wiley & Sons, 1996.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point

Port	Supported Data Types
Desired	• Double-precision floating point • Single-precision floating point
Step-size	• Double-precision floating point • Single-precision floating point
Adapt	• Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Reset	• Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Output	• Double-precision floating point • Single-precision floating point
Error	• Double-precision floating point • Single-precision floating point
Wts	• Double-precision floating point • Single-precision floating point

See Also

Fast Block LMS Filter	Signal Processing Blockset
Kalman Adaptive Filter	Signal Processing Blockset
LMS Filter	Signal Processing Blockset
RLS Filter	Signal Processing Blockset

See “Adaptive Filters” for related information.

Purpose

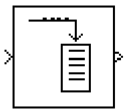
Buffer input sequence to smaller or larger frame size

Library

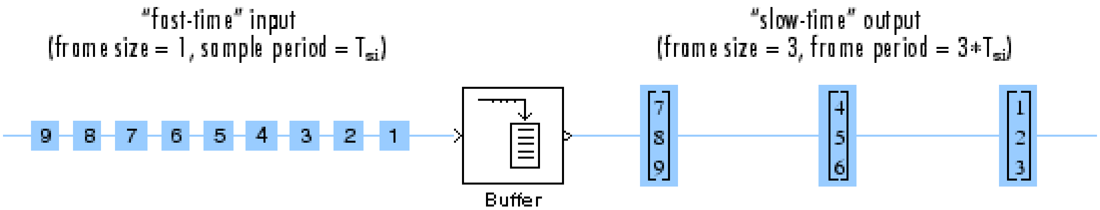
Signal Management / Buffers

`dspbuff3`

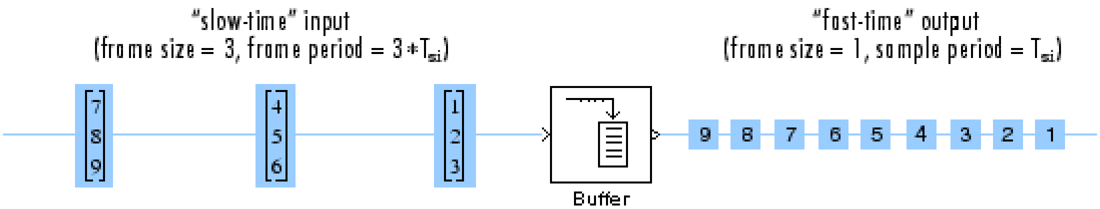
Description



The Buffer block redistributes the input samples to a new frame size. Buffering to a larger frame size yields an output with a *slower* frame rate than the input, as illustrated below for scalar input.



Buffering to a smaller frame size yields an output with a *faster* frame rate than the input, as illustrated below for scalar output.



The block coordinates the output *frame size* and *frame rate* of nonoverlapping buffers such that the sample period of the signal is the same at both the input and output: $T_{so} = T_{si}$.

This block supports triggered subsystems when the block's input and output rates are the same.

Sample-based full-dimension matrix inputs are not accepted.

Sample-Based Operation

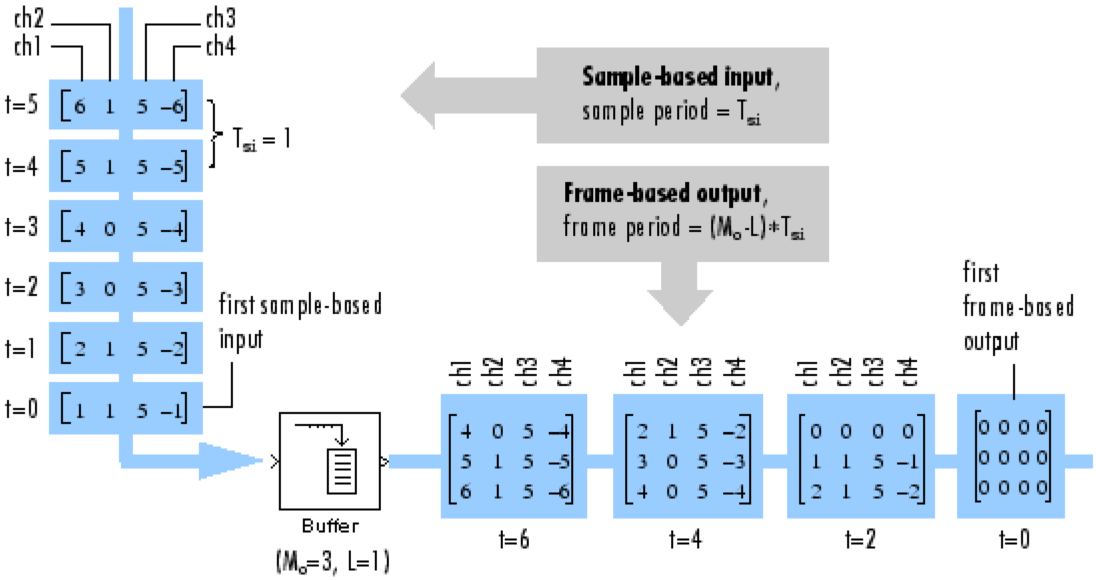
Sample-based inputs are interpreted by the Buffer block as independent channels of data. Therefore, a sample-based length-N vector input is interpreted as N independent samples.

In sample-based operation, the Buffer block creates frame-based outputs from sample-based inputs. A sequence of sample-based length-N vector inputs is buffered into an M_o-by-N matrix, where M_o is specified by the **Output buffer size** parameter (M_o>1). That is, each input vector becomes a *row* in the N-channel frame-based output matrix. When M_o=1, the input is simply passed through to the output, and retains the same dimension.

The **Buffer overlap** parameter, L, specifies the number of samples (rows) from the current output to repeat in the next output, where $L < M_o$. For $0 \le L < M_o$, the number of *new* input samples that the block acquires before propagating the buffered data to the output is the difference between the **Output buffer size** and **Buffer overlap**, or $M_o - L$.

The output frame period is $(M_o - L)T_{si}$, which is equal to the input sequence sample period, T_{si}, when the **Buffer overlap** is $M_o - 1$. For $L < 0$, the block simply discards L input samples after the buffer fills, and outputs the buffer with period $(M_o - L)T_{si}$, which is longer than the zero-overlap case.

In the model below, the block buffers a four-channel sample-based input using a **Output buffer size** of three and a **Buffer overlap** of one.



Notice that the input vectors do not begin appearing at the output until the second row of the second matrix. This is due to the block's latency. The first output matrix (all zeros in this example) reflects the block's **Initial conditions** setting, while the first row of zeros in the second output is a result of the one-sample overlap between consecutive output frames.

You can use the `rebuffer_delay` function with a frame size of 1 to precisely compute the delay (in samples) for sample-based signals. For the previous example,

```
d = rebuffer_delay(1,3,1)
d =
      4
```

This agrees with the four samples of delay (zeros) per channel shown in the previous figure.

Frame-Based Operation

In frame-based operation, the Buffer block redistributes the samples in the input frame to an output frame with a new size and rate. A sequence of M_i-by-N matrix inputs is buffered into a sequence of M_o-by-N frame-based matrix outputs, where M_o is the output frame size specified by the **Output buffer size** parameter. The output buffer size is the number of consecutive samples from the input frame that are buffered into the output frame. M_o can be greater or less than the input frame size, M_i. Each of the N input channels is buffered independently.

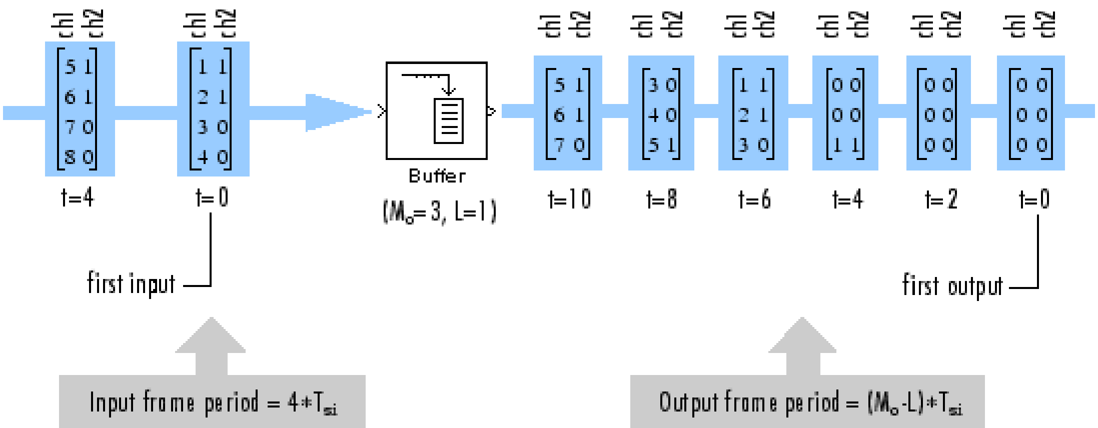
The **Buffer overlap** parameter, L, specifies the number of samples (rows) from the current output to repeat in the next output, where $L < M_o$. For $0 \le L < M_o$, the number of *new* input samples the block acquires before propagating the buffered data to the output is the difference between the **Output buffer size** and **Buffer overlap**, $M_o - L$.

The input frame period is $M_i T_{si}$, where T_{si} is the sample period. The output frame period is $(M_o - L)T_{si}$, which is equal to the sequence sample period when the **Buffer overlap** is $M_o - 1$. The output sample period is therefore related to the input sample period by

$$T_{so} = \frac{(M_o - L)T_{si}}{M_o}$$

Negative **Buffer overlap** values are not permitted.

In the model below, the block buffers a two-channel frame-based input using a **Output buffer size** of three and a **Buffer overlap** of one.



Notice that the sequence is delayed by eight samples, which is the latency of the block in the Simulink® multitasking mode for the parameter settings of this example. The first eight output samples therefore adopt the value specified for the **Initial conditions**, which is assumed here to be zero. Use the `rebuffer_delay` function to determine the block's latency for any combination of frame size and overlap.

Latency

Zero tasking latency means that the first input sample, received at t = 0, appears as the first output sample. In the Simulink single-tasking mode, the Buffer block has zero tasking latency for the following special cases:

- Scalar input and output ($M_o = M_i = 1$) with zero or negative **Buffer overlap** ($L \le 0$)
- Input frame size is an integer multiple of the output frame size

$$M_i = kM_o$$

where k is an integer with zero **Buffer overlap** ($L = 0$); notable cases of this include

- Any input frame size M_i with scalar output ($M_o = 1$) and zero **Buffer overlap** ($L = 0$)
- Equal input and output frame sizes ($M_o = M_i$) with zero **Buffer overlap** ($L = 0$)

For all cases of *sample-based single-tasking* operation other than those listed above, the Buffer block's buffer is initialized to the value(s) specified by the **Initial conditions** parameter, and the block reads from this buffer to generate the first D output samples, where

$$D = \begin{cases} M_o + L & (L \geq 0) \\ M_o & (L < 0) \end{cases}$$

The dimensions of the **Initial conditions** parameter depend on the **Buffer overlap**, L, and whether the input is single-channel or multichannel:

- When $L \neq 0$, the **Initial conditions** parameter must be a scalar.
- When $L = 0$, the **Initial conditions** parameter can be a scalar, or it can be a vector with the following constraints:
  - For single-channel inputs, the **Initial conditions** parameter can be a vector of length M_o if M_i is 1, or a vector of length M_i if M_o is 1.
  - For multichannel inputs, the **Initial conditions** parameter can be a vector of length $M_o * N$ if M_i is 1, or a vector of length $M_i * N$ if M_o is 1.

For *frame-based single-tasking* operation and all *multitasking* operation, use the `rebuffer_delay` function to compute the exact delay, in samples, that the Buffer block introduces for a given combination of buffer size and buffer overlap.

For general buffering between arbitrary frame sizes, the **Initial conditions** parameter must be a scalar value, which is then repeated across all elements of the initial output(s). However, in the special case where the input is 1-by-N, and the block's output is therefore an M_o-by-N matrix, **Initial conditions** can be

- An M_o-by-N matrix
- A length-M_o vector to be repeated across all columns of the initial output(s)
- A scalar to be repeated across all elements of the initial output(s)

In the special case where the output is 1-by-N , which is the result of unbuffering an M_i-by-N frame-based matrix, **Initial conditions** can be

- A vector containing M_i samples to output sequentially for each channel during the first M_i sample times
- A scalar to be repeated across all elements of the initial output(s)

Note For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Behavior in Enabled Subsystems

The Buffer block cannot be used in an enabled subsystem under the following conditions:

- In a multirate multitasking environment
- When the **Buffer overlap** parameter is set to a negative value

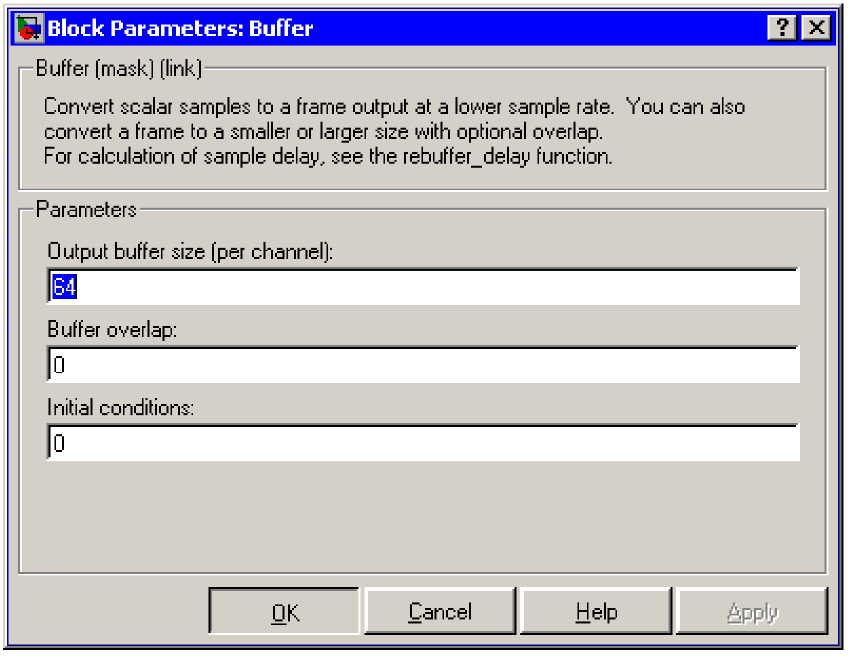
The Buffer block has an internal reservoir that temporarily stores data. When the Buffer block is used in an enabled subsystem, there is the possibility that the reservoir can overrun or underrun. The block implements safeguards against these occurrences.

Overrun occurs when more data enters into the buffer than it can hold. For example, consider buffering a sample-based input to a frame of size three with a buffer that accepts input every second and outputs every three seconds. If you place this buffer inside an enabled subsystem that is disabled every three seconds at t = 3s, t = 6s, and so on, the buffer accumulates data in its internal reservoir without being able to empty it. This condition results in overrun.

Underrun occurs when the buffer runs out of data to output. For example, again consider buffering a sample-based input to a frame size of three with a buffer that accepts input every second and outputs every three seconds. If you place this buffer inside an enabled subsystem that is disabled at t = 10s, t = 11s, t = 13s, t = 14s, t = 16s, and t = 17s, its internal reservoir becomes drained, and there is no data to output at t = 18s. This condition results in underrun.

To protect from overrun and underrun, the Buffer block keeps a record of the amount of data in its internal reservoir. When the Buffer block reads data, the amount of data in its reservoir goes up. When data is output from the Buffer block, the amount of data in its reservoir goes down. To protect from overrun, the oldest samples in the reservoir are discarded whenever amount of data in the reservoir is larger than the actual buffer size. To protect from underrun, the most recent samples are repeated whenever an output is due and there is no data in the reservoir.

Dialog Box



Output buffer size

Specify the number of consecutive samples, M_o, from each channel to buffer into the output frame.

Buffer overlap

Specify the number of samples, L, by which consecutive output frames overlap.

Initial conditions

Specify the value of the block's initial output for cases of nonzero latency; a scalar, vector, or matrix.

Supported Data Types

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

See Also

Delay Line	Signal Processing Blockset
Unbuffer	Signal Processing Blockset
`rebuffer_delay`	Signal Processing Blockset

See "Converting Sample and Frame Rates" and "Converting Frame Status" for more information.

Purpose

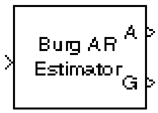
Compute estimate of autoregressive (AR) model parameters using Burg method

Library

Estimation / Parametric Estimation

`dspparest3`

Description



The Burg AR Estimator block uses the Burg method to fit an autoregressive (AR) model to the input data by minimizing (least squares) the forward and backward prediction errors while constraining the AR parameters to satisfy the Levinson-Durbin recursion.

The input is a sample-based vector (row, column, or 1-D) or frame-based vector (column only) representing a frame of consecutive time samples from a single-channel signal, which is assumed to be the output of an AR system driven by white noise. The block computes the normalized estimate of the AR system parameters, $A(z)$, independently for each successive input frame.

$$H(z) = \frac{G}{A(z)} = \frac{G}{1 + a(2)z^{-1} + \ldots + a(p+1)z^{-p}}$$

When you select the **Inherit estimation order from input dimensions** parameter, the order, p, of the all-pole model is one less that the length of the input vector. Otherwise, the order is the value specified by the **Estimation order** parameter.

The **Output(s)** parameter allows you to select between two realizations of the AR process:

- `A` — The top output, A, is a column vector of length p+1 with the same frame status as the input, and contains the normalized estimate of the AR model polynomial coefficients in descending powers of z.

  ```
  [1 a(2) ... a(p+1)]
  ```

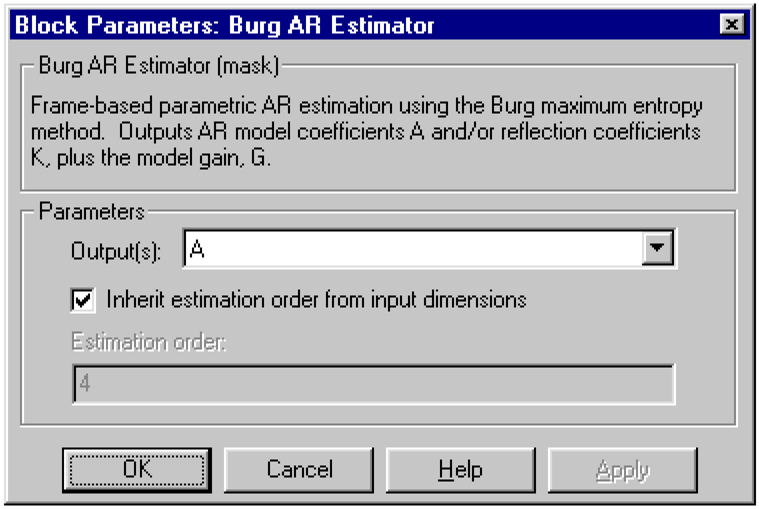
- `K` — The top output, K, is a column vector of length p with the same frame status as the input, and contains the reflection coefficients (which are a secondary result of the Levinson recursion).
- `A and K` — The block outputs both realizations.

The scalar gain, G, is provided at the bottom output (`G`).

The following table compares the features of the Burg AR Estimator block to the Covariance AR Estimator, Modified Covariance AR Estimator, and Yule-Walker AR Estimator blocks.

	Burg AR Estimator	Covariance AR Estimator	Modified Covariance AR Estimator	Yule-Walker AR Estimator
`Characteristics`	Does not apply window to data	Does not apply window to data	Does not apply window to data	Applies window to data
	Minimizes the forward and backward prediction errors in the least squares sense, with the AR coefficients constrained to satisfy the L-D recursion	Minimizes the forward prediction error in the least squares sense	Minimizes the forward and backward prediction errors in the least squares sense	Minimizes the forward prediction error in the least squares sense (also called “autocorrelation method”)
`Advantages`	Always produces a stable model			Always produces a stable model
`Disadvantages`		May produce unstable models	May produce unstable models	Performs relatively poorly for short data records
`Conditions for Nonsingularity`		Order must be less than or equal to half the input frame size	Order must be less than or equal to 2/3 the input frame size	Because of the biased estimate, the autocorrelation matrix is guaranteed to positive-definite, hence nonsingular

Dialog Box



Output(s)
: The realization to output, model coefficients, reflection coefficients, or both.

Inherit estimation order from input dimensions
: When selected, sets the estimation order p to one less than the length of the input vector.

Estimation order
: The order of the AR model, p. This parameter is enabled when you do not select **Inherit estimation order from input dimensions**.

References

Kay, S. M. *Modern Spectral Estimation: Theory and Application.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

Marple, S. L., Jr., *Digital Spectral Analysis with Applications.* Englewood Cliffs, NJ: Prentice-Hall, 1987.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
A	• Double-precision floating point • Single-precision floating point
G	• Double-precision floating point • Single-precision floating point

See Also

Burg Method	Signal Processing Blockset
Covariance AR Estimator	Signal Processing Blockset
Modified Covariance AR Estimator	Signal Processing Blockset
Yule-Walker AR Estimator	Signal Processing Blockset
`arburg`	Signal Processing Toolbox

Purpose

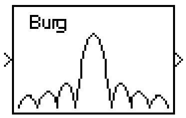
Compute parametric spectral estimate using Burg method

Library

Estimation / Power Spectrum Estimation

`dspspect3`

Description



The Burg Method block estimates the power spectral density (PSD) of the input frame using the Burg method. This method fits an autoregressive (AR) model to the signal by minimizing (least squares) the forward and backward prediction errors while constraining the AR parameters to satisfy the Levinson-Durbin recursion.

The input is a sample-based vector (row, column, or 1-D) or frame-based vector (column only) representing a frame of consecutive time samples from a single-channel signal. The block's output (a column vector) is the estimate of the signal's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where F_s is the signal's sample frequency.

When you select the **Inherit estimation order from input dimensions** parameter, the order of the all-pole model is one less that the input frame size. Otherwise, the order is the value specified by the **Estimation order** parameter. The spectrum is computed from the FFT of the estimated AR model parameters.

When you select the **Inherit FFT length from estimation order** parameter, N_{fft} is specified by the frame size of the input, which must be a power of 2. When you do *not* select **Inherit FFT length from estimation order**, N_{fft} is specified as a power of 2 by the **FFT length** parameter, and the block zero pads or wraps the input to N_{fft} before computing the FFT. The output is always sample based.

The Burg Method and Yule-Walker Method blocks return similar results for large frame sizes. The following table compares the features of the Burg Method block to the Covariance Method, Modified Covariance Method, and Yule-Walker Method blocks.

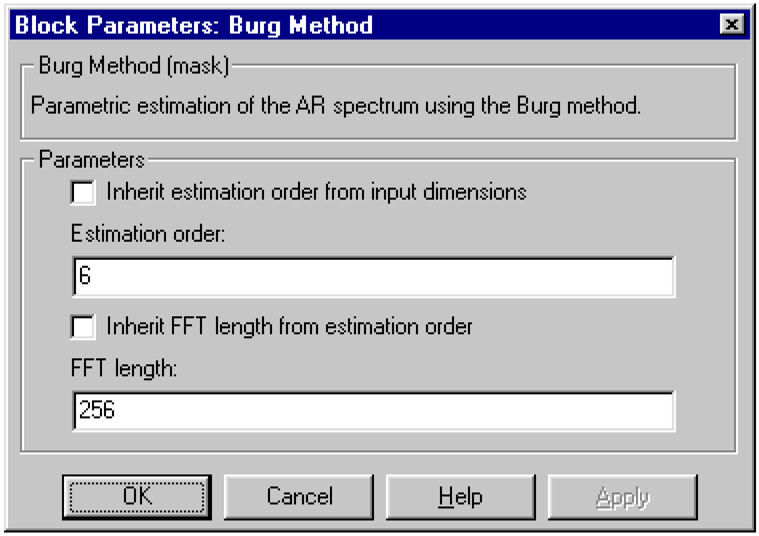
	Burg	Covariance	Modified Covariance	Yule-Walker
Characteristics	Does not apply window to data	Does not apply window to data	Does not apply window to data	Applies window to data
	Minimizes the forward and backward prediction errors in the least squares sense, with the AR coefficients constrained to satisfy the L-D recursion	Minimizes the forward prediction error in the least squares sense	Minimizes the forward and backward prediction errors in the least squares sense	Minimizes the forward prediction error in the least squares sense (also called “autocorrelation method”)
Advantages	High resolution for short data records	Better resolution than Y-W for short data records (more accurate estimates)	High resolution for short data records	Performs as well as other methods for large data records
	Always produces a stable model	Able to extract frequencies from data consisting of p or more pure sinusoids	Able to extract frequencies from data consisting of p or more pure sinusoids	Always produces a stable model
				Does not suffer spectral line-splitting

	Burg	Covariance	Modified Covariance	Yule-Walker
Disadvantages	Peak locations highly dependent on initial phase	May produce unstable models	May produce unstable models	Performs relatively poorly for short data records
	May suffer spectral line-splitting for sinusoids in noise, or when order is very large	Frequency bias for estimates of sinusoids in noise	Peak locations slightly dependent on initial phase	Frequency bias for estimates of sinusoids in noise
	Frequency bias for estimates of sinusoids in noise		Minor frequency bias for estimates of sinusoids in noise	
Conditions for Nonsingularity		Order must be less than or equal to half the input frame size	Order must be less than or equal to 2/3 the input frame size	Because of the biased estimate, the autocorrelation matrix is guaranteed to positive-definite, hence nonsingular

Examples

The `dspsacomp` demo compares the Burg method with several other spectral estimation methods.

Dialog Box



Inherit estimation order from input dimensions
When selected, sets the estimation order to one less than the length of the input vector.

Estimation order
The order of the AR model. This parameter is enabled when you do not select **Inherit estimation order from input dimensions**.

Inherit FFT length from estimation order
When selected, uses the input frame size as the number of data points, N_{fft}, on which to perform the FFT.

FFT length
Enter the number of data points on which to perform the FFT, N_{fft}. When N_{fft} is larger than the input frame size, each frame is zero-padded as needed. When N_{fft} is smaller than the input frame size, each frame is wrapped as needed. This parameter is enabled when you clear the **Inherit FFT length from input dimensions** check box.

References

Kay, S. M. *Modern Spectral Estimation: Theory and Application.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

Orfanidis, S. J. *Introduction to Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1995.

Orfanidis, S. J. *Optimum Signal Processing: An Introduction.* 2nd ed. New York, NY: Macmillan, 1985.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
Output	• Double-precision floating point • Single-precision floating point

See Also

Burg AR Estimator	Signal Processing Blockset
Covariance Method	Signal Processing Blockset
Modified Covariance Method	Signal Processing Blockset
Short-Time FFT	Signal Processing Blockset
Yule-Walker Method	Signal Processing Blockset
`pburg`	Signal Processing Toolbox

See "Power Spectrum Estimation" for related information.

Purpose

Generate error when input signal does or does not match selected attributes exactly

Library

Signal Management / Signal Attributes

`dspsigattribs`

Description

The Check Signal Attributes block terminates the simulation with an error when the input characteristics differ from those specified by the block parameters.

When the **Error when input** parameter is set to `Does not match attributes exactly`, the block generates an error only when the input possesses *none* of the attributes specified by the other parameters. Signals that possess *at least one* of the specified attributes are propagated to the output unaltered, and do not generate an error.

When the **Error when input** parameter is set to `Matches attributes exactly`, the block generates an error only when the input possesses *all* attributes specified by the other parameters. Signals that do not possess *all* of the specified attributes are propagated to the output unaltered, and do not generate an error.

Signal Attributes

The Check Signal Attributes block can test for up to five different signal attributes, as specified by the following parameters. When you select the `Ignore` in any parameter, the block does not check the signal for the corresponding attribute. For example, when **Complexity** is set to `Ignore`, neither real nor complex inputs cause the block to generate an error. The attributes are

- **Complexity**

  Checks whether the signal is real or complex. (Note that this information can be displayed in a model by attaching a Probe block with **Probe complex signal** selected. Alternatively, in the model window, from the **Format** menu, point to **Port/Signal Displays**, and select **Port Data Types**.)

- **Frame status**

  Checks whether the signal is frame based or sample based. (Note that the Simulink® environment displays sample-based signals using a single line, and frame-based signals using a double line.)

- **Dimensionality**

  Checks the dimension of signal for compliance (`Is...`) or noncompliance (`Is not...`) with the attributes in the subordinate **Dimension** menu, which are shown in the table below. *M* and *N* are positive integers unless otherwise indicated below.

Dimensions	Is...	Is not...
1-D	1-D vector, 1-D scalar	*M*-by-*N* matrix, 1-by-*N* matrix (row vector), *M*-by-1 matrix (column vector), 1-by-1 matrix (2-D scalar)
2-D	*M*-by-*N* matrix, 1-by-*N* matrix (row vector), *M*-by-1 matrix (column vector), 1-by-1 matrix (2-D scalar)	1-D vector, 1-D scalar
Scalar (1-D or 2-D)	1-D scalar, 1-by-1 matrix (2-D scalar)	1-D vector with length>1, *M*-by-*N* matrix with *M*>1 and/or *N*>1
Vector (1-D or 2-D)	1-D vector, 1-D scalar, 1-by-*N* matrix (row vector), *M*-by-1 matrix (column vector), 1-by-1 matrix (2-D scalar) Vector (1-D or 2-D) or scalar	*M*-by-*N* matrix with *M*>1 and *N*>1
Row Vector (2-D)	1-by-*N* matrix (row vector), 1-by-1 matrix (2-D scalar) Row vector (2-D) or scalar	1-D vector, 1-D scalar, *M*-by-*N* matrix with *M*>1

Dimensions	Is...	Is not...
Column Vector (2-D)	M-by-1 matrix (column vector), 1-by-1 matrix (2-D scalar) Column vector (2-D) or scalar	1-D vector, 1-D scalar, M-by-N matrix with N>1
Full matrix	M-by-N matrix with M>1 and N>1	1-D vector, 1-D scalar, 1-by-N matrix (row vector), M-by-1 matrix (column vector), 1-by-1 matrix (2-D scalar)
Square matrix	M-by-N matrix with M=N, 1-D scalar, 1-by-1 matrix (2-D scalar	M-by-N matrix with M≠N, 1-D vector, 1-by-N matrix (row vector), M-by-1 matrix (column vector)

If, in the model window, from the **Format** menu, you point to **Port/Signal Displays**, and select **Signal Dimensions**, Simulink displays the size of a 1-D vector signal as an unbracketed integer, and displays the dimension of a 2-D signal as a pair of bracketed integers, `[MxN]`. Simulink *does not display* any size information for a 1-D or 2-D scalar signal. Dimension information for a signal can also be displayed in a model by attaching a Probe block with **Probe signal dimensions** selected.

- **Data type**

  Checks the signal data type for compliance (`Is...`) or noncompliance (`Is not...`) with the attributes in the subordinate **General data type** menu, which are shown in the table below. Any of the specific data types listed in the `Is...` column below can be individually selected from the subordinate **Specific data type** menu.

General Data Type	Is...	Is not...
Boolean	`boolean`	`single, double, uint8, int8, uint16, int16, uint32, int32, fixed-point`

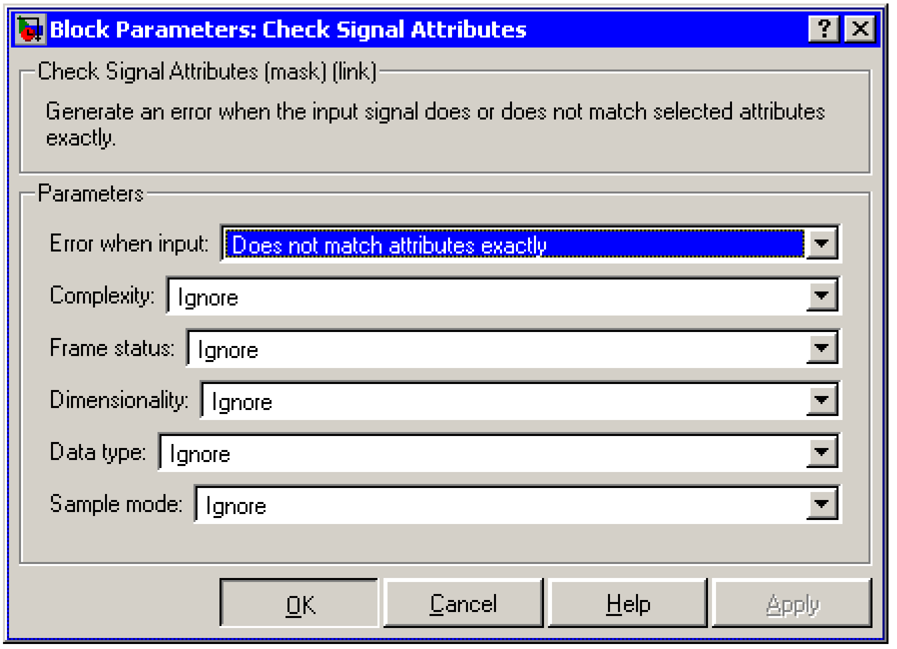
General Data Type	Is...	Is not...
Floating-point	`single`, `double`	`boolean`, `uint8`, `int8`, `uint16`, `int16`, `uint32`, `int32`, `fixed-point`
Fixed-point	fixed-point	`boolean`, `uint8`, `int8`, `uint16`, `int16`, `uint32`, `int32`, `single`, `double`
Integer	Signed integer `int8`, `int16`, `int32` Unsigned integer `uint8`, `uint16`, `uint32`	`boolean`, `single`, `double`

To display data type information, in your model window, from the **Format** menu, point to **Port/Signal Displays**, and select **Port Data Types**.

- **Sample mode**

  Checks whether the signal is discrete-time or continuous-time. (If, from the **Format** menu, you point to **Port/Signal Displays**, and select **Sample Time Colors**, Simulink displays continuous-time signal lines in black or grey and discrete-time signal lines in colors corresponding to the relative rate. When a Probe block with **Probe sample time** enabled is attached to a continuous-time signal, the block icon displays the string `Ts:[0 x]`, where `x` is the sample time offset. When a Probe block is attached to a discrete-time signal, the block icon displays the string `Ts:[t 0]` for a sample-based signal or `Tf:[t 0]` for a frame-based signal, where `t` is the nonzero sample period or frame period, respectively. Frame-based signals are almost always discrete time.

Dialog Box



Error when input

Specifies whether the block generates an error when the input possesses *none* of the required attributes (`Does not match attributes exactly`), or when the input possesses *all* of the required attributes (`Matches attributes exactly`).

Complexity

The complexity for which the input should be checked, `Real` or `Complex`. When you select `Ignore` from the list, the block does not check the input's complexity.

Frame status

The frame status for which the input should be checked, `Sample-based` or `Frame-based`. When you select `Ignore` from the list, the block does not check the input's frame status.

Dimensionality

Specifies whether the input should be checked for compliance (`Is...`) or noncompliance (`Is not...`) with the attributes in the subordinate **Dimension** menu. When you select `Ignore` from the list, the block does not check the input's dimensionality.

Data type

Specifies whether the input should be checked for compliance (`Is...`) or noncompliance (`Is not...`) with the attributes in the subordinate **General data type** menu. When you select `Ignore` from the list, the block does not check the input's data type.

Sample mode

The sample mode for which the input should be checked, `Discrete` or `Continuous`. When you select `Ignore` from the list, the block does not check the input's sample mode.

Supported Data Types

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

See Also

Buffer	Signal Processing Blockset
Convert 1-D to 2-D	Signal Processing Blockset
Convert 2-D to 1-D	Signal Processing Blockset
Data Type Conversion	Simulink
Frame Status Conversion	Signal Processing Blockset
Inherit Complexity	Signal Processing Blockset
Probe	Simulink
Reshape	Simulink
Submatrix	Signal Processing Blockset

Chirp

Purpose

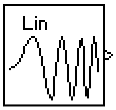
Generate swept-frequency cosine (chirp) signal

Library

Signal Processing Sources

`dspsrcs4`

Description



The Chirp block outputs a swept-frequency cosine (chirp) signal with unity amplitude and continuous phase. To specify the desired output chirp signal, you must define its instantaneous frequency function, also known as the output frequency sweep. The frequency sweep can be linear, quadratic, or logarithmic, and repeats once every **Sweep time** by default. See other sections of this reference page for more details about the block.

Sections of This Reference Page

- Variables Used in This Reference Page on page 2-109
- "Setting the Output Frame Status" on page 2-109
- "Shaping the Frequency Sweep by Setting Frequency Sweep and Sweep Mode" on page 2-110
- "Unidirectional and Bidirectional Sweep Modes" on page 2-111
- "Setting Instantaneous Frequency Sweep Values" on page 2-112
- "Block Computation Methods" on page 2-113
- "Cautions Regarding the Swept Cosine Sweep" on page 2-116
- "Dialog Box" on page 2-118
- "Examples" on page 2-120
- "Supported Data Types" on page 2-131
- "See Also" on page 2-132

Variables Used in This Reference Page

f_0	**Initial frequency** parameter (Hz)
$f_i(t_g)$	**Target frequency** parameter (Hz)
t_g	**Target time** parameter (seconds)
T_{sw}	**Sweep time** parameter (seconds)
ϕ_0	**Initial phase** parameter (radians)
$\psi(t)$	Phase of the chirp signal (radians)
$f_i(t)$	User-specified output instantaneous frequency function (Hz); user-specified sweep
$f_{i(actual)}(t)$	Actual output instantaneous frequency function (Hz); actual output sweep
$y_{chirp}(t)$	Output chirp function

Setting the Output Frame Status

Use **Samples per frame** parameter to set the block's output frame status, as summarized in the following table. The **Sample time** parameter sets the sample time of both sample- and frame-based outputs.

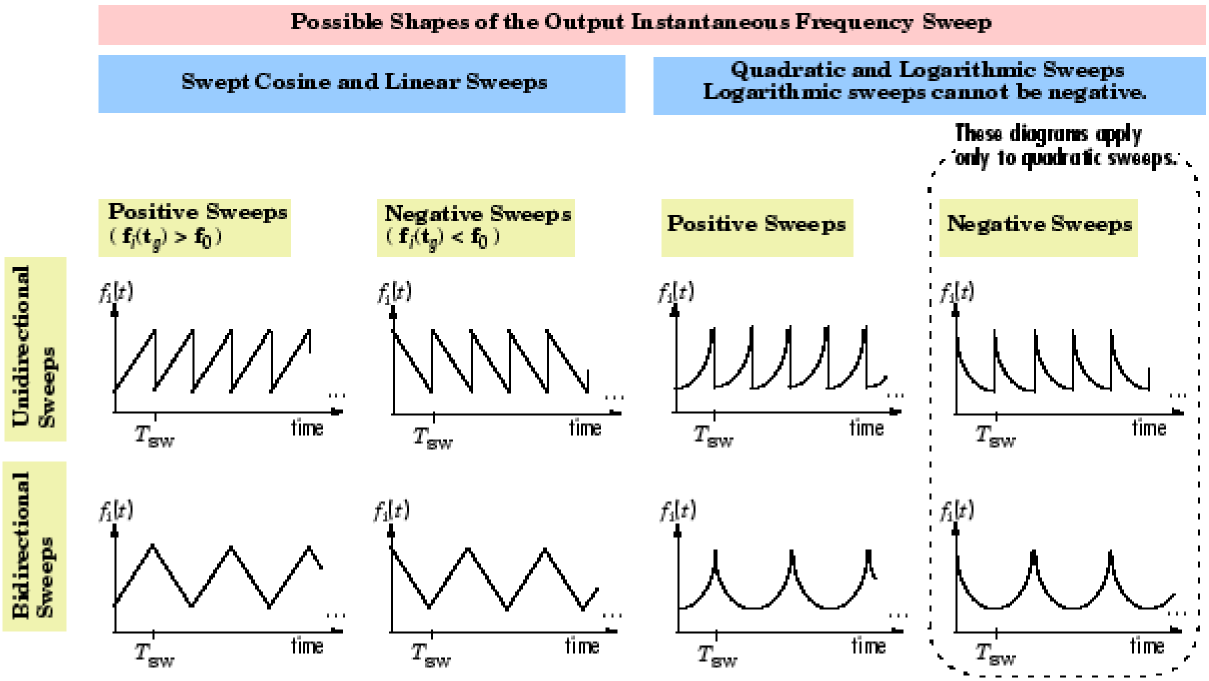
Setting of Samples Per Frame Parameter	Output Frame Status
1	Sample based
n (any integer greater than 1)	Frame based, frame size n

Shaping the Frequency Sweep by Setting Frequency Sweep and Sweep Mode

The basic shape of the output instantaneous frequency sweep, $f_i(t)$, is set by the **Frequency sweep** and **Sweep mode** parameters, described in the following table.

Parameters for Setting Sweep Shape	Possible Setting	Parameter Description
Frequency sweep	Linear Quadratic Logarithmic Swept cosine	Determines whether the sweep frequencies vary linearly, quadratically, or logarithmically. Linear and swept cosine sweeps both vary linearly.
Sweep mode	Unidirectional Bidirectional	Determines whether the sweep is unidirectional or bidirectional. For details, see "Unidirectional and Bidirectional Sweep Modes" on page 2-111

The following diagram illustrates the possible shapes of the frequency sweep that you can obtain by setting the **Frequency sweep** and **Sweep mode** parameters.



For information on how to set the frequency values in your sweep, see "Setting Instantaneous Frequency Sweep Values" on page 2-112.

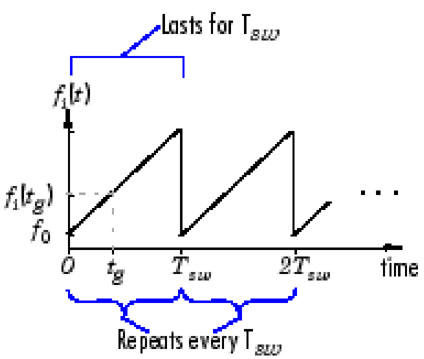
Unidirectional and Bidirectional Sweep Modes

The **Sweep mode** parameter determines whether your sweep is unidirectional or bidirectional, which affects the shape of your output frequency sweep (see "Shaping the Frequency Sweep by Setting Frequency Sweep and Sweep Mode" on page 2-110). The following table describes the characteristics of unidirectional and bidirectional sweeps.

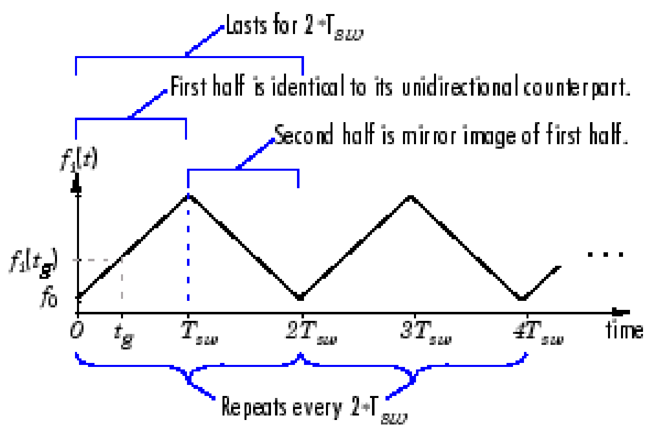
Sweep Mode Parameter Settings	Sweep Characteristics
`Unidirectional`	• Lasts for one **Sweep time**, T_{sw} • Repeats once every T_{sw}
`Bidirectional`	• Lasts for twice the **Sweep time**, 2^*T_{sw} • Repeats once every 2^*T_{sw} • First half is identical to its unidirectional counterpart. • Second half is a mirror image of the first half.

The following diagram illustrates a linear sweep in both sweep modes. For information on setting the frequency values in your sweep, see "Setting Instantaneous Frequency Sweep Values" on page 2-112.

Unidirectional Linear Sweep



Bidirectional Linear Sweep



Setting Instantaneous Frequency Sweep Values

Set the following parameters to tune the frequency values of your output frequency sweep. Note that because this is a source block, the simulation pauses while the block dialog box is open. You must close the dialog box by clicking **OK** to resume the simulation.

- **Initial frequency** (Hz), f_0
- **Target frequency** (Hz), $f_i(t_g)$
- **Target time** (seconds), t_g

The following table summarizes the sweep values at specific times for all **Frequency sweep** settings. For information on the formulas used to compute sweep values at other times, see "Block Computation Methods" on page 2-113.

Instantaneous Frequency Sweep Values

Frequency Sweep	Sweep Value at $t = 0$	Sweep Value at $t = t_g$	Time when Sweep Value Is Target Frequency, $f_i(t_g)$
Linear	f_0	$f_i(t_g)$	t_g
Quadratic	f_0	$f_i(t_g)$	t_g
Logarithmic	f_0	$f_i(t_g)$	t_g
Swept cosine	f_0	$2f_i(t_g) - f_0$	$t_g/2$

Block Computation Methods

The Chirp block uses one of two formulas to compute the block output, depending on the **Frequency Sweep** parameter setting. For details, see the following sections:

- "Equations for Output Computation" on page 2-114
- "Output Computation Method for Linear, Quadratic, and Logarithmic Frequency Sweeps" on page 2-115

- "Output Computation Method for Swept Cosine Frequency Sweep" on page 2-116

Equations for Output Computation

The following table shows the equations used by the block to compute the user-specified output frequency sweep, $f_i(t)$, the block output, $y_{chirp}(t)$, and the actual output frequency sweep, $f_{i(actual)}(t)$. The only time the user-specified sweep is not the actual output sweep is when the **Frequency sweep** parameter is set to `Swept cosine`.

Note The following equations apply only to unidirectional sweeps in which $f_i(0) < f_i(t_g)$. To derive equations for other cases, you might find it helpful to examine the following table and the diagram in "Shaping the Frequency Sweep by Setting Frequency Sweep and Sweep Mode" on page 2-110.

The table below contains the following variables:

- $f_i(t)$ — the user-specified frequency sweep
- $f_{i(actual)}(t)$ — the actual output frequency sweep, usually equal to $f_i(t)$
- $y(t)$ — the Chirp block output
- $\psi(t)$ — the phase of the chirp signal, where $\psi(0) = 0$, and $2\pi f_i(t)$ is the derivative of the phase

$$f_i(t) = \frac{1}{2\pi} \cdot \frac{d\psi(t)}{dt}$$

- ϕ_0 — the **Initial phase** parameter value, where $y_{chirp}(0) = \cos(\phi_0)$

Equations Used by the Chirp Block for Unidirectional Positive Sweeps

Frequency Sweep	Block Output Chirp Signal	User-Specified Frequency Sweep, $f_i(t)$	β	Actual Frequency Sweep, $f_{i(actual)}(t)$
`Linear`	$y(t) = \cos(\psi(t) + \phi_0)$	$f_i(t) = f_0 + \beta t$	$\beta = \frac{f_i(t_g) - f_0}{t_g}$	$f_{i(actual)}^{(t)} = f_i^{(t)}$
`Quadratic`	Same as `Linear`	$f_i(t) = f_0 + \beta t^2$	$\beta = \frac{f_i(t_g) - f_0}{2}$	$f_{i(actual)}^{(t)} = f_i^{(t)}$
`Logarithmic`	Same as `Linear`	$F_i(t) = f_0 \left(\frac{f_i(t_g)}{f_0} \right)^{\frac{t}{t_g}}$ Where $f_i(t_g) > f_0 > 0$	N/A	$f_{i(actual)}(t) = f_i(t)$
`Swept cosine`	$y(t) = \cos(2\pi f_i(t) t + \phi_0)$	Same as `Linear`	Same as `Linear`	$f_{i(actual)}(t) = f_i(t) + \beta t$

Output Computation Method for Linear, Quadratic, and Logarithmic Frequency Sweeps

The derivative of the phase of a chirp function gives the instantaneous frequency of the chirp function. The Chirp block uses this principle to calculate the chirp output when the **Frequency Sweep** parameter is set to `Linear`, `Quadratic`, or `Logarithmic`.

$y_{chirp}(t) = \cos(\psi(t) + \phi_0)$	Linear, quadratic, or logarithmic chirp signal with phase $\psi(t)$
$f_i(t) = \frac{1}{2\pi} \cdot \frac{d\psi(t)}{dt}$	Phase derivative is instantaneous frequency

For instance, if you want a chirp signal with a linear instantaneous frequency sweep, you should set the **Frequency Sweep** parameter to

`Linear`, and tune the linear sweep values by setting other parameters appropriately. Note that because this is a source block, the simulation pauses while the block dialog box is open. You must close the dialog box by clicking **OK** to resume the simulation. The block outputs a chirp signal, the phase derivative of which is the specified linear sweep. This ensures that the instantaneous frequency of the output is the linear sweep you desired. For equations describing the linear, quadratic, and logarithmic sweeps, see "Equations for Output Computation" on page 2-114.

Output Computation Method for Swept Cosine Frequency Sweep

To generate the swept cosine chirp signal, the block sets the swept cosine chirp output as follows.

$$y_{chirp}(t) = \cos(\psi(t) + \phi_0) = \cos(2\pi f_i(t)t + \phi_0)$$

Swept cosine chirp output (Instantaneous frequency equation, shown above, does not hold.)

Note that the instantaneous frequency equation, shown above, does not hold for the swept cosine chirp, so the user-defined frequency sweep, $f_i(t)$, is not the actual output frequency sweep, $f_{i(actual)}(t)$, of the swept cosine chirp. Thus, the swept cosine output might not behave as you expect. To learn more about swept cosine chirp behavior, see "Cautions Regarding the Swept Cosine Sweep" on page 2-116 and "Equations for Output Computation" on page 2-114.

Cautions Regarding the Swept Cosine Sweep

When you want a linearly swept chirp signal, we recommend you use a linear frequency sweep. Though a swept cosine frequency sweep also yields a linearly swept chirp signal, the output might have unexpected frequency content. For details, see the following two sections.

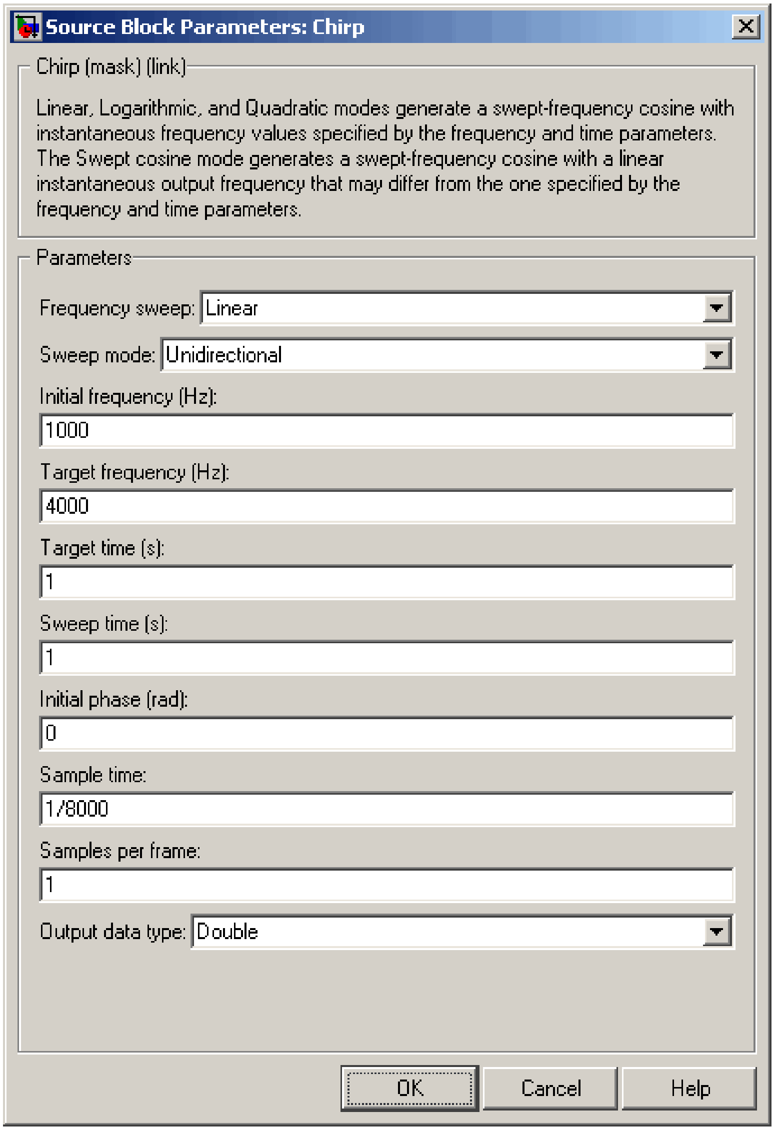
Swept Cosine Instantaneous Output Frequency at the Target Time is not the Target Frequency

The swept cosine sweep value at the **Target time** is not necessarily the **Target frequency**. This is because the user-specified sweep is not the actual frequency sweep of the swept cosine output, as noted in “Output Computation Method for Swept Cosine Frequency Sweep” on page 2-116. See the table Instantaneous Frequency Sweep Values on page 2-113 for the actual value of the swept cosine sweep at the **Target time**.

Swept Cosine Output Frequency Content May Greatly Exceed Frequencies in the Sweep

In **Swept cosine** mode, you should not set the parameters so that $1/T$sw is very large compared to the values of the **Initial frequency** and **Target frequency** parameters. In such cases, the actual frequency content of the swept cosine sweep might be closer to $1/T$sw, far exceeding the **Initial frequency** and **Target frequency** parameter values.

Dialog Box


Source Block Parameters: Chirp
Chirp (mask) (link)
Linear, Logarithmic, and Quadratic modes generate a swept-frequency cosine with instantaneous frequency values specified by the frequency and time parameters. The Swept cosine mode generates a swept-frequency cosine with a linear instantaneous output frequency that may differ from the one specified by the frequency and time parameters.
Parameters
Frequency sweep: Linear
Sweep mode: Unidirectional
Initial frequency (Hz):
1000
Target frequency (Hz):
4000
Target time (s):
1
Sweep time (s):
1
Initial phase (rad):
0
Sample time:
1/8000
Samples per frame:
1
Output data type: Double
OK
Cancel
Help

Opening this dialog box causes a running simulation to pause. See "Changing Source Block Parameters During Simulation" in the online Simulink® documentation for details.

Frequency sweep

The type of output instantaneous frequency sweep, $f_i(t)$: `Linear`, `Logarithmic`, `Quadratic`, or `Swept cosine`.

Sweep mode

The directionality of the chirp signal: `Unidirectional` or `Bidirectional`.

Initial frequency (Hz)

For `Linear`, `Quadratic`, and `Swept cosine` sweeps, the initial frequency, f_0, of the output chirp signal. For `Logarithmic` sweeps, **Initial frequency** is one less than the actual initial frequency of the sweep. Also, when the sweep is `Logarithmic`, you must set the **Initial frequency** to be less than the **Target frequency**. Tunable.

Target frequency (Hz)

For `Linear`, `Quadratic`, and `Logarithmic` sweeps, the instantaneous frequency, $f_i(t_g)$, of the output at the **Target time**, t_g. For a `Swept cosine` sweep, **Target frequency** is the instantaneous frequency of the output at half the **Target time**, $t_g/2$. When **Frequency sweep** is `Logarithmic`, you must set the **Target frequency** to be greater than the **Initial frequency**. Tunable.

Target time (s)

For `Linear`, `Quadratic`, and `Logarithmic` sweeps, the time, t_g, at which the **Target frequency**, $f_i(t_g)$, is reached by the sweep. For a `Swept cosine` sweep, **Target time** is the time at which the sweep reaches $2f_i(t_g) - f_0$. You must set **Target time** to be *no greater than*

Sweep time, $T_{sw} \geq t_g$. Tunable.

Sweep time (s)

In `Unidirectional` **Sweep mode**, the **Sweep time**, T_{sw}, is the period of the output frequency sweep. In `Bidirectional` **Sweep**

mode, the **Sweep time** is half the period of the output frequency sweep. You must set **Sweep time** to be no less than **Target time**,

$T_{sw} \geq t_g$. Tunable.

Initial phase (rad)

The phase, ϕ_0 , of the cosine output at t=0; $y_{chirp}(t) = \cos(\phi_0)$. Tunable.

Sample time

The sample period, T_s, of the output. The output frame period is M_o*T_s.

Samples per frame

The number of samples, M_o, to buffer into each output frame. When the value of this parameter is 1, the block outputs a sample-based signal.

Output data type

The data type of the output, single-precision or double-precision.

Examples

The first few examples demonstrate how to use the Chirp block's main parameters, how to view the output in the time domain, and how to view the output spectrogram:

- "Example 1: Setting a Final Frequency Value for Unidirectional Sweeps" on page 2-121
- "Example 2: Bidirectional Sweeps" on page 2-124
- "Example 3: When Sweep Time is Greater Than Target Time" on page 2-126

Examples 4 and 5 illustrate Chirp block settings that might produce unexpected outputs:

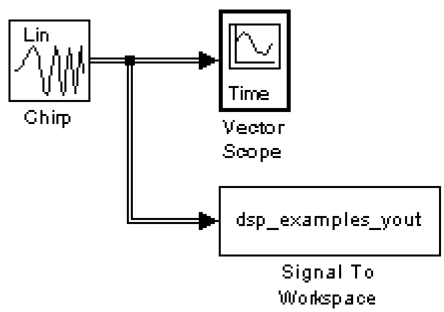
- "Example 4: Output Sweep with Negative Frequencies" on page 2-128
- "Example 5: Output Sweep with Frequencies Greater Than Half the Sampling Frequency" on page 2-130

Example 1: Setting a Final Frequency Value for Unidirectional Sweeps

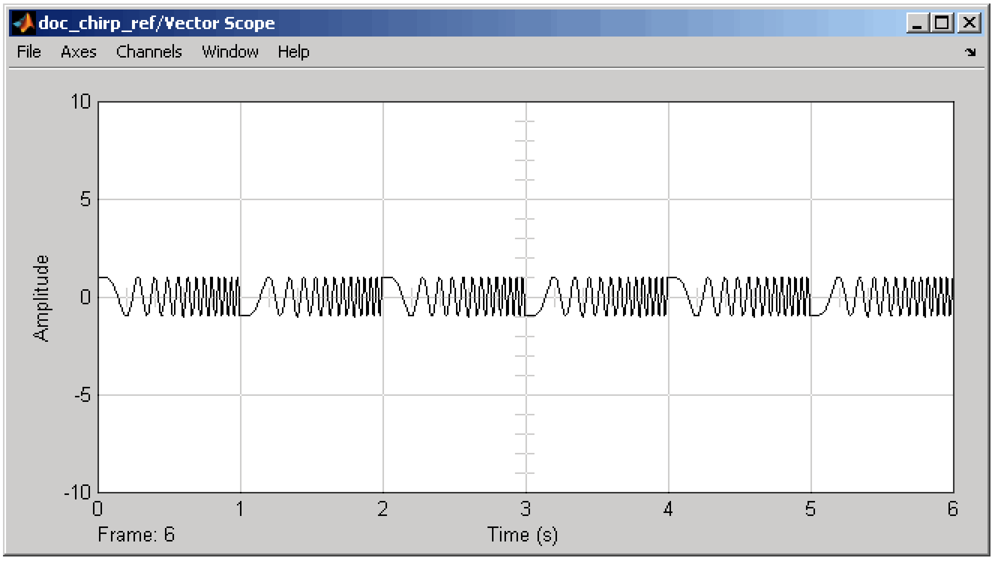
Often times, you might want a unidirectional sweep for which you know the initial and final frequency values. You can specify the final frequency of a unidirectional sweep by setting **Target time** equal to **Sweep time**, in which case the **Target frequency** becomes the final frequency in the sweep. The following model demonstrates this method.

This technique might not work for swept cosine sweeps. For details, see "Cautions Regarding the Swept Cosine Sweep" on page 2-116.

Open the Example 1 model by typing `doc_chirp_ref` at the MATLAB® command line. You can also rebuild the model yourself; see the following list for model parameter settings (leave unlisted parameters in their default states).

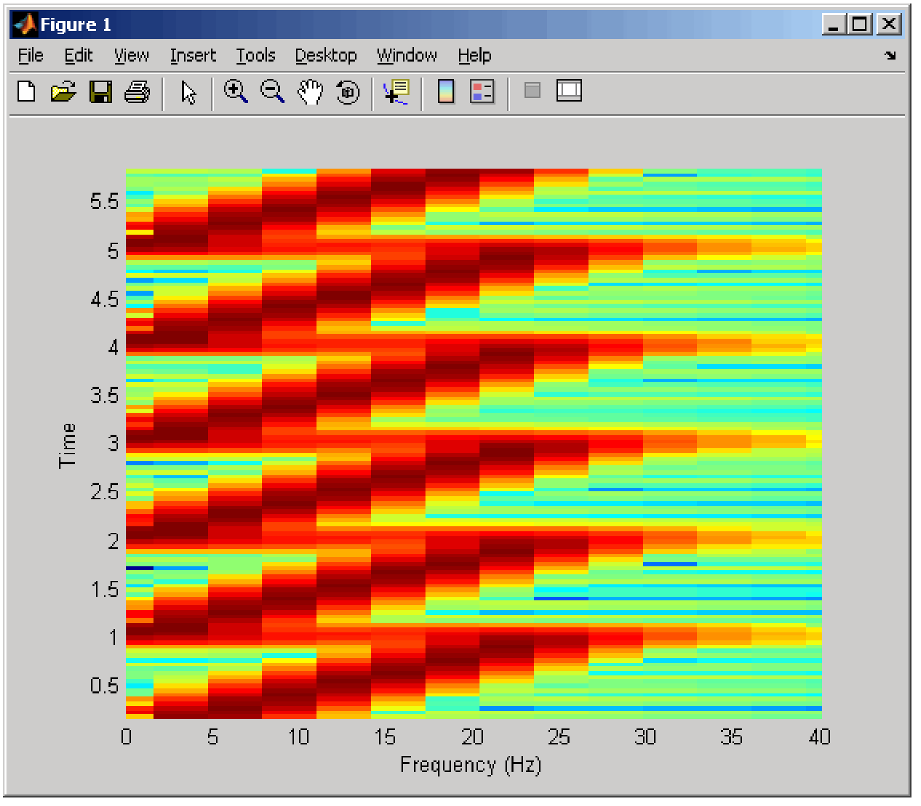


Since **Target time** is set to equal **Sweep time** (1 second), the **Target frequency** (25 Hz) is the final frequency of the unidirectional sweep. Run your model to see the time domain output:



Type the following command to view the chirp output spectrogram:

```
spectrogram(dsp_examples_yout,hamming(128),...
      110,[0:.01:40],400)
```



Chirp Block Parameters for Example 1	
Frequency sweep	`Linear`
Sweep mode	`Unidirectional`
Initial frequency	0

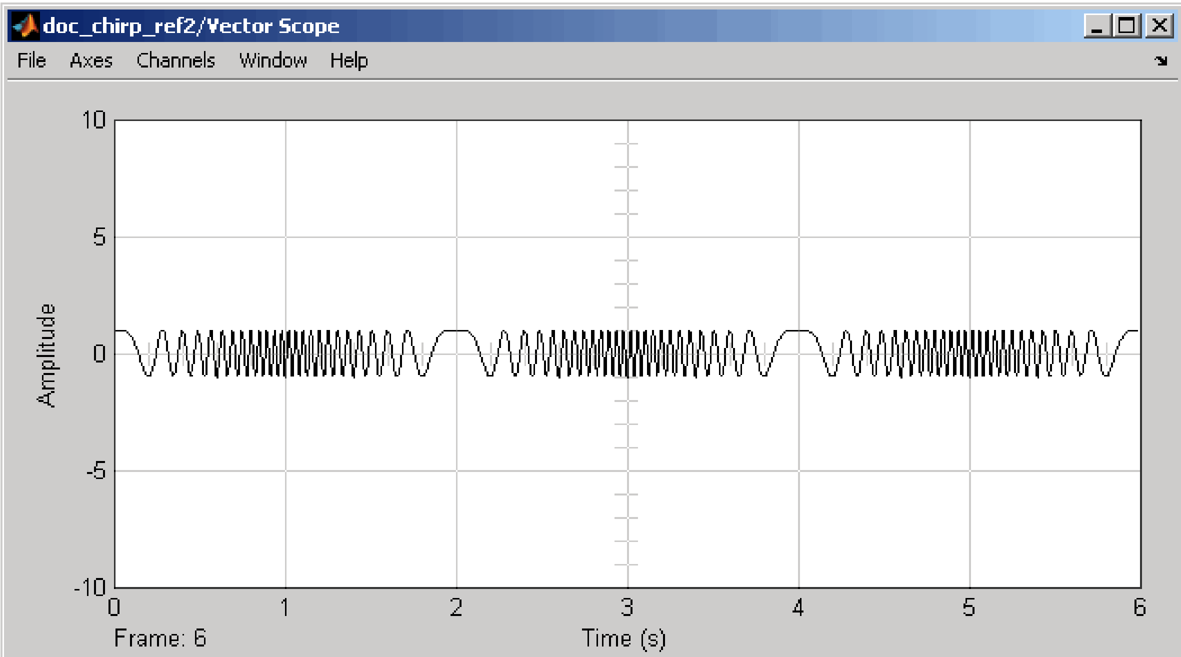
Target frequency	25
Target time	1
Sweep time	1
Initial phase	0
Sample time	1/400
Samples per frame	400
Vector Scope Block Parameters for Example 1	
Input domain	`Time`
Time display span	6
Signal To Workspace Block Parameters for Example 1	
Variable name	`dsp_examples_yout`
Configuration Dialog Parameters for Example 1	
Stop time	5

Example 2: Bidirectional Sweeps

Change the **Sweep mode** parameter in the Example 1 model to `Bidirectional`, and leave all other parameters the same to view the following bidirectional chirp. Note that in the bidirectional sweep, the period of the sweep is twice the **Sweep time** (2 seconds), whereas it was one **Sweep time** (1 second) for the unidirectional sweep in Example 1.

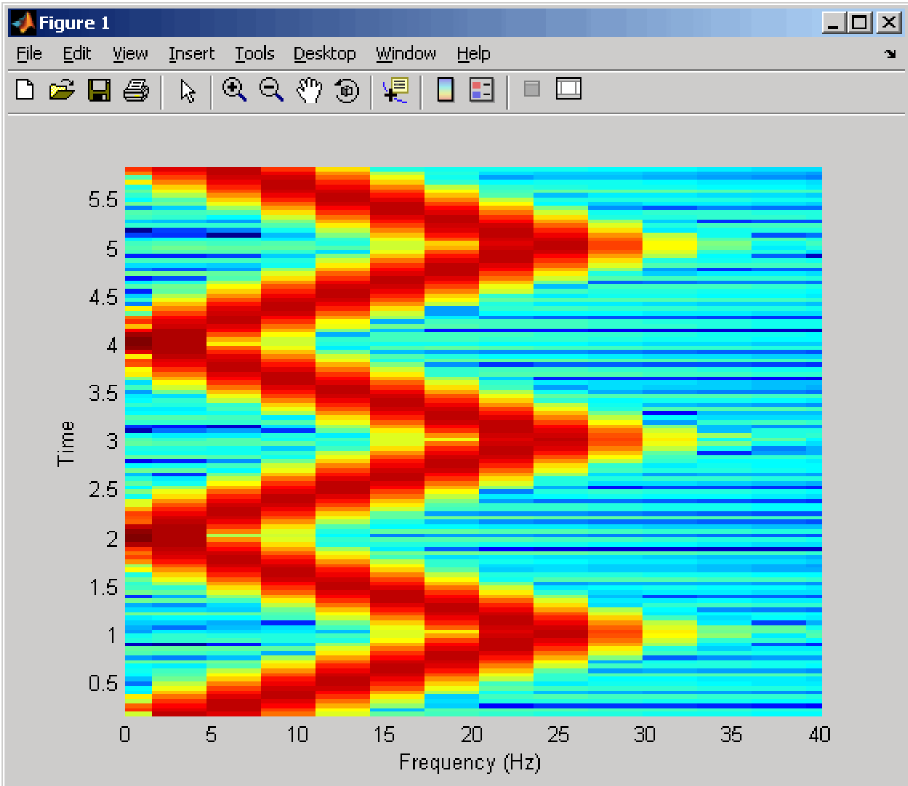
Open the Example 2 model by typing `doc_chirp_ref2` at the MATLAB command line.

Run your model to see the time domain output:



Type the following command to view the chirp output spectrogram:

```
spectrogram(dsp_examples_yout,hamming(128),...
      110,[0:.01:40],400)
```



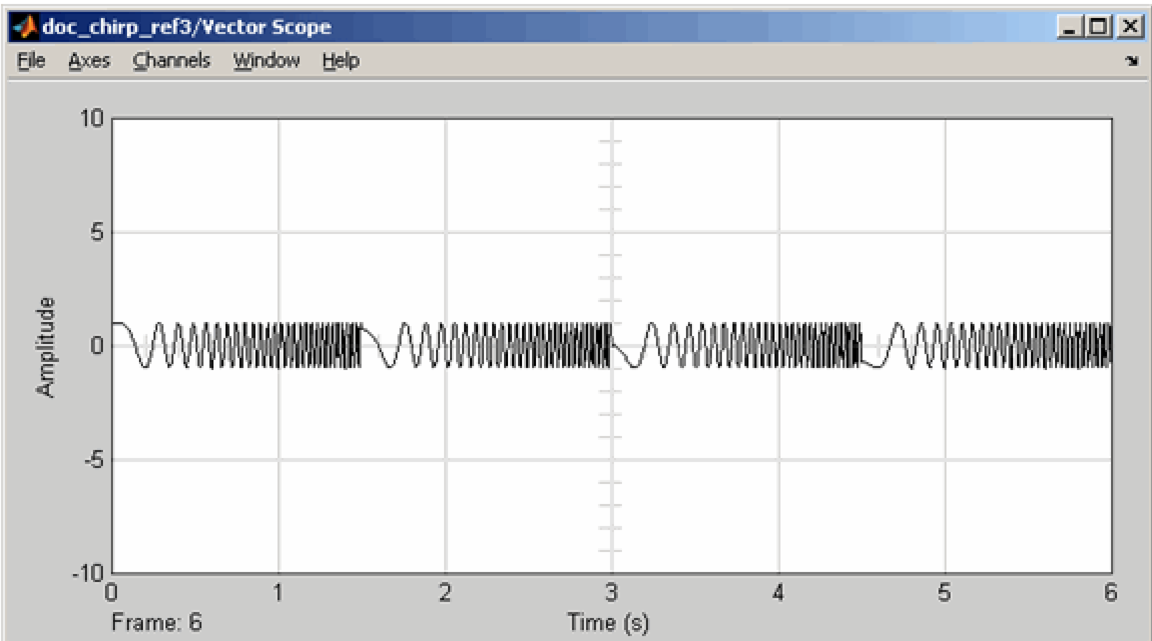
Example 3: When Sweep Time is Greater Than Target Time

Setting **Sweep time** to `1.5` and leaving the rest of the parameters as in the Example 1 model gives the following output. The sweep still reaches the **Target frequency** (25 Hz) at the **Target time** (1 second), but since **Sweep time** is greater than **Target time**, the sweep continues on its linear path until one **Sweep time** (1.5 seconds) is traversed.

Unexpected behavior might arise when you set **Sweep time** greater than **Target time**; see "Example 4: Output Sweep with Negative Frequencies" on page 2-128 for details.

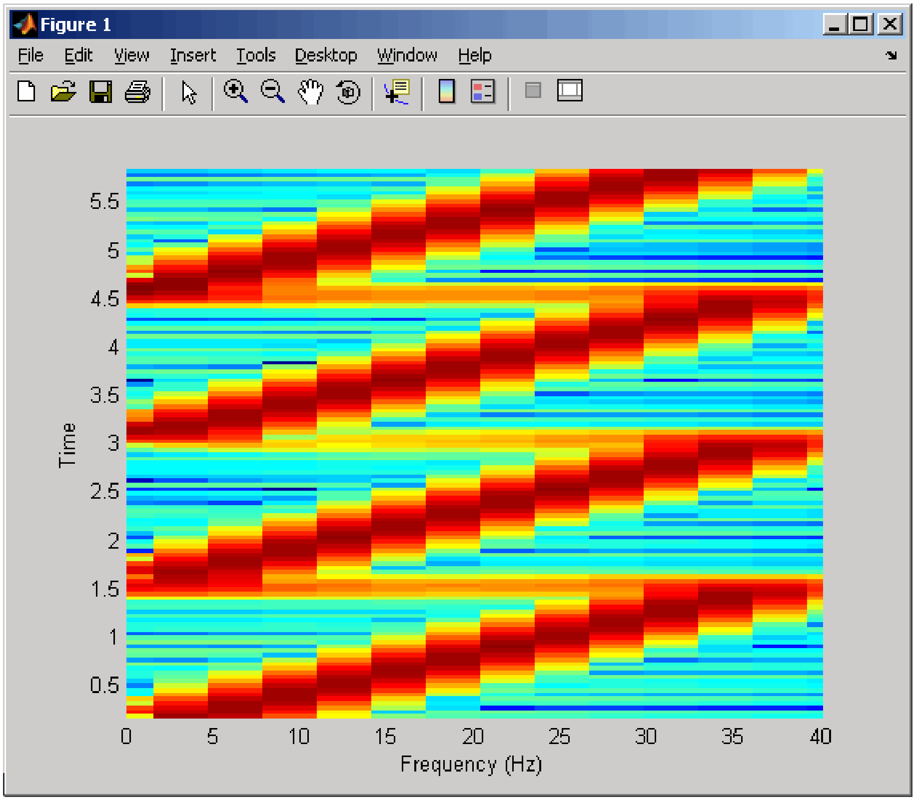
Open the Example 3 model by typing `doc_chirp_ref3` at the MATLAB command line.

Run your model to see the time domain output:



```
spectrogram(dsp_examples_yout,hamming(128),...
      110,[0:.01:40],400)
```

Type the following command to view the chirp output spectrogram:



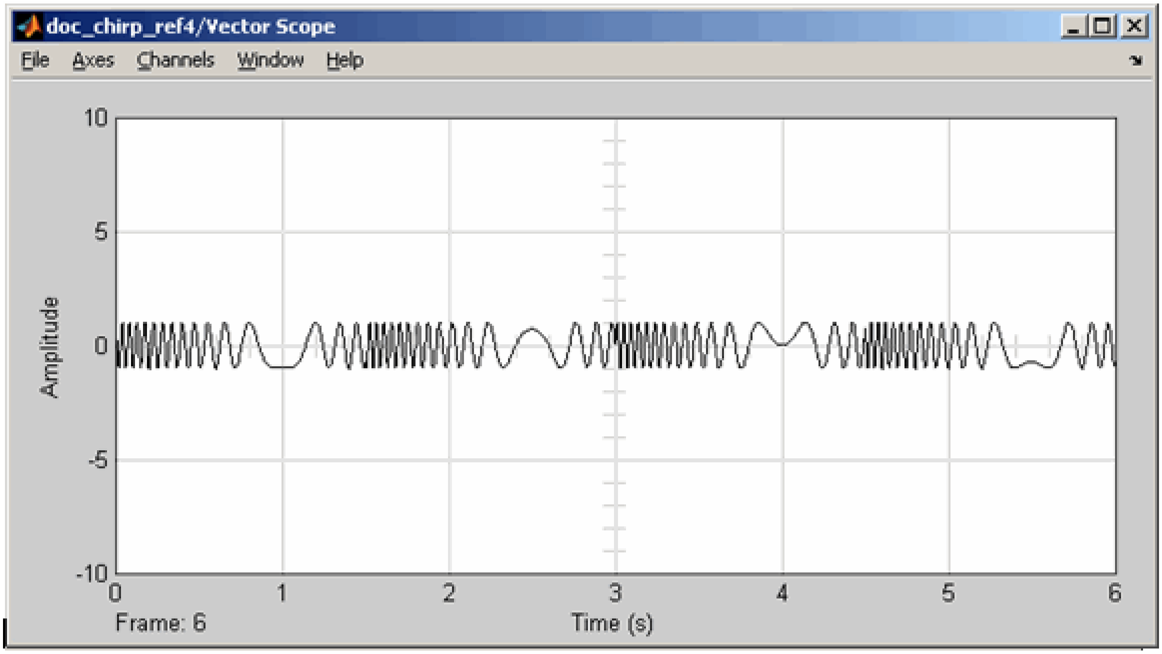
Example 4: Output Sweep with Negative Frequencies

Modify the Example 1 model by changing **Sweep time** to `1.5`, **Initial frequency** to `25`, and **Target frequency** to `0`. *The output chirp of this example might not behave as you expect* because the sweep contains negative frequencies between 1 and 1.5 seconds. The sweep reaches the **Target frequency** of 0 Hz at one second, then continues on its negative slope, taking on negative frequency values until it traverses one **Sweep time** (1.5 seconds).

The spectrogram might reflect negative sweep frequencies along the *x*-axis so they appear to be positive. If you unexpectedly get a chirp output with a spectrogram resembling the one below, your chirp's sweep might contain negative frequencies. See the next example for another possible unexpected chirp output.

Open the Example 4 model by typing `doc_chirp_ref4` at the MATLAB command line.

Run your model to see the time domain output:

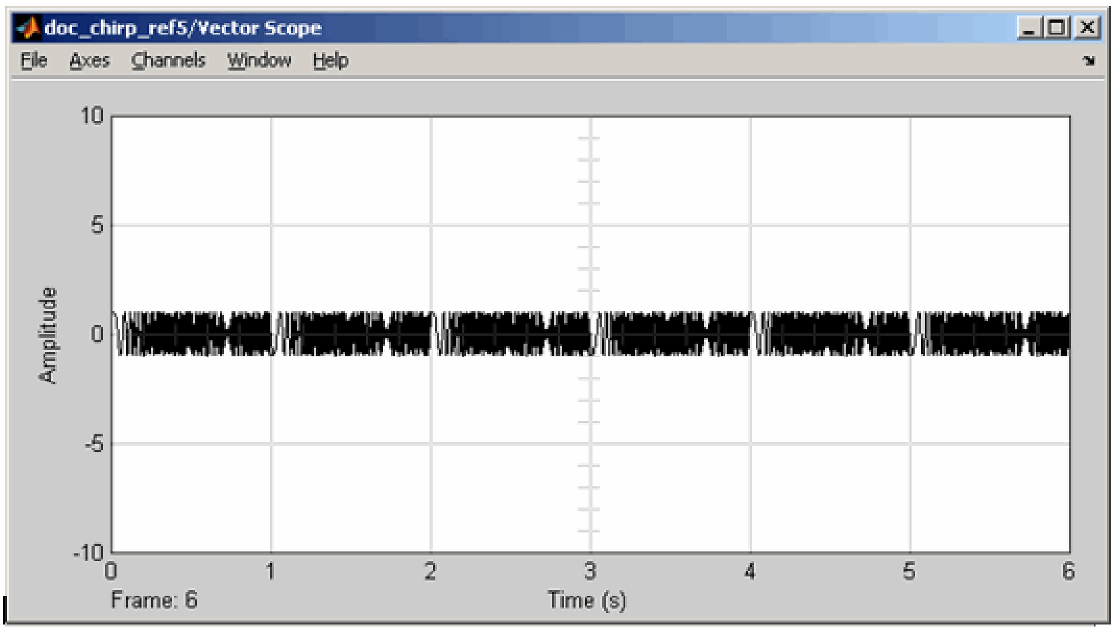


Example 5: Output Sweep with Frequencies Greater Than Half the Sampling Frequency

Modify the Example 1 model by changing the **Target frequency** parameter to `275`. *The output chirp of this model might not behave as you expect* because the sweep contains frequencies greater than half the sampling frequency (`200` Hz), which causes aliasing. If you unexpectedly get a chirp output with a spectrogram resembling the one following, your chirp's sweep might contain frequencies greater than half the sampling frequency. See the previous example for another possible unexpected chirp output.

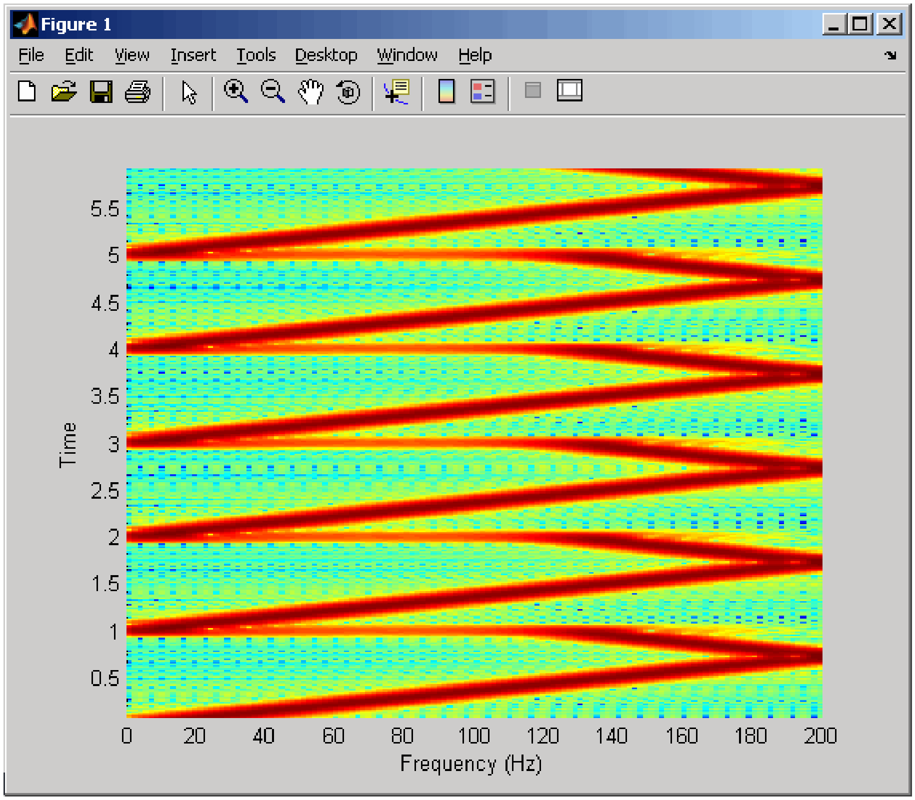
Open the Example 5 model by typing `doc_chirp_ref5` at the MATLAB command line.

Run your model to see the time domain output:



Type the following command to view the chirp output spectrogram:

```
spectrogram(dsp_examples_yout,hamming(64),...
        60,256,400)
```



Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Signal From Workspace	Signal Processing Blockset
Signal Generator	Simulink
Sine Wave	Signal Processing Blockset
`chirp`	Signal Processing Toolbox
`spectrogram`	Signal Processing Toolbox

Purpose

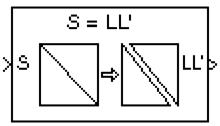
Factor square Hermitian positive definite matrix into triangular components

Library

Math Functions / Matrices and Linear Algebra / Matrix Factorizations

`dspfactors`

Description



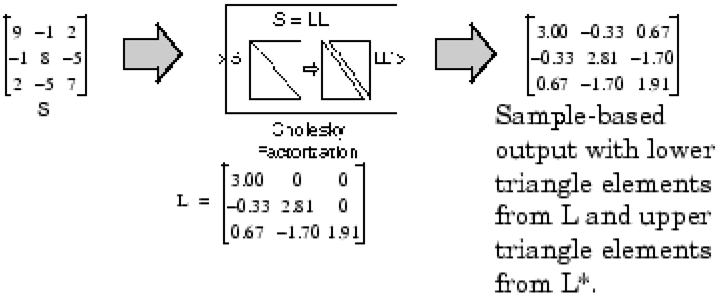
The Cholesky Factorization block uniquely factors the square Hermitian positive definite input matrix S as

$$S = LL^*$$

where L is a lower triangular square matrix with positive diagonal elements and L^* is the Hermitian (complex conjugate) transpose of L. The block outputs a matrix with lower triangle elements from L and upper triangle elements from L^*. The output is always sample based. The output is not in the same form as the output of the MATLAB® `chol` function. In order to convert the output of the Cholesky Factorization block to the MATLAB form, use the following equation:

```
R = triu(LL');
```

Here, `LL'` is the output of the Cholesky Factorization block. Due to roundoff error, these equations do not produce a result that is exactly the same as the MATLAB result.



Block Output Composed of L and L*

Input Requirements for Valid Output

The block output is valid only when its input has the following characteristics:

- Hermitian — The block does *not* check whether the input is Hermitian; it uses only the diagonal and upper triangle of the input to compute the output.
- Real-valued diagonal entries — The block disregards any imaginary component of the input's diagonal entries.
- Positive definite — Set the block to notify you when the input is not positive definite as described in "Response to Nonpositive Definite Input" on page 2-134.

Response to Nonpositive Definite Input

To generate a valid output, the block algorithm requires a positive definite input (see "Input Requirements for Valid Output" on page 2-134). Set the **Non-positive definite input** parameter to determine how the block responds to a nonpositive definite input:

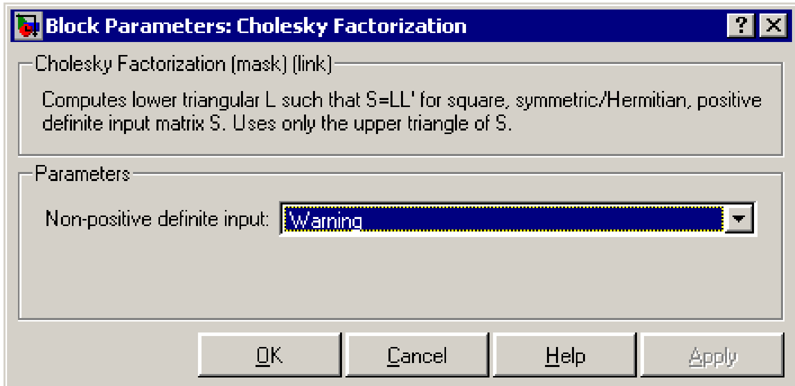
- `Ignore` — Proceed with the computation and *do not* issue an alert. The output is *not* a valid factorization. A partial factorization will be present in the upper left corner of the output.
- `Warning` — Display a warning message in the MATLAB Command Window, and continue the simulation. The output is *not* a valid factorization. A partial factorization will be present in the upper left corner of the output.
- `Error` — Display an error dialog and terminate the simulation.

Note The **Non-positive definite input** parameter is a diagnostic parameter. Like all diagnostic parameters on the Configuration Parameters dialog box, it is set to `Ignore` in the code generated for this block by Real-Time Workshop® code generation software.

Performance Comparisons with Other Blocks

Note that L and L^* share the same diagonal in the output matrix. Cholesky factorization requires half the computation of Gaussian elimination (LU decomposition), and is always stable.

Dialog Box



Non-positive definite input

Response to nonpositive definite matrix inputs: `Ignore`, `Warning`, or `Error`. See “Response to Nonpositive Definite Input” on page 2-134.

References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

Port	Supported Data Types
S	• Double-precision floating point • Single-precision floating point
LL'	• Double-precision floating point • Single-precision floating point

See Also

Autocorrelation LPC	Signal Processing Blockset
Cholesky Inverse	Signal Processing Blockset
Cholesky Solver	Signal Processing Blockset
LDL Factorization	Signal Processing Blockset
LU Factorization	Signal Processing Blockset
QR Factorization	Signal Processing Blockset
`chol`	MATLAB

See “Matrix Factorizations” for related information.

Cholesky Inverse

Purpose

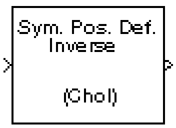
Compute inverse of Hermitian positive definite matrix using Cholesky factorization

Library

Math Functions / Matrices and Linear Algebra / Matrix Inverses

`dspinverses`

Description



The Cholesky Inverse block computes the inverse of the Hermitian positive definite input matrix S by performing Cholesky factorization.

$$S^{-1} = (LL^*)^{-1}$$

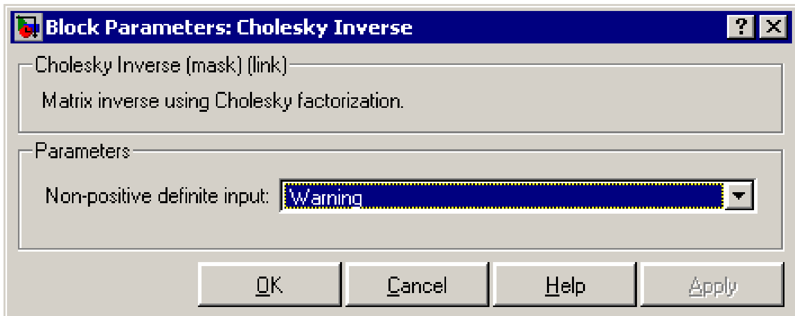
L is a lower triangular square matrix with positive diagonal elements and L^* is the Hermitian (complex conjugate) transpose of L. Only the diagonal and upper triangle of the input matrix are used, and any imaginary component of the diagonal entries is disregarded. Cholesky factorization requires half the computation of Gaussian elimination (LU decomposition), and is always stable. The output is always sample based.

The algorithm requires that the input be Hermitian positive definite. When the input is not positive definite, the block reacts with the behavior specified by the **Non-positive definite input** parameter. The following options are available:

- `Ignore` — Proceed with the computation and *do not* issue an alert. The output is *not* a valid inverse.
- `Warning` — Display a warning message in the MATLAB® Command Window, and continue the simulation. The output is *not* a valid inverse.
- `Error` — Display an error dialog box and terminate the simulation.

> **Note** The **Non-positive definite input** parameter is a diagnostic parameter. Like all diagnostic parameters on the Configuration Parameters dialog box, it is set to `Ignore` in the code generated for this block by Real-Time Workshop® code generation software.

Dialog Box



Non-positive definite input
Response to nonpositive definite matrix inputs.

References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Cholesky Factorization	Signal Processing Blockset
Cholesky Solver	Signal Processing Blockset
LDL Inverse	Signal Processing Blockset
LU Inverse	Signal Processing Blockset
Pseudoinverse	Signal Processing Blockset
`inv`	MATLAB

See "Matrix Inverses" for related information.

Cholesky Solver

Purpose

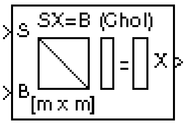
Solve *S*X=*B* for X when *S* is square Hermitian positive definite matrix

Library

Math Functions / Matrices and Linear Algebra / Linear System Solvers

`dspsolvers`

Description



The Cholesky Solver block solves the linear system *S*X=*B* by applying Cholesky factorization to input matrix at the `S` port, which must be square (*M*-by-*M*) and Hermitian positive definite. Only the diagonal and upper triangle of the matrix are used, and any imaginary component of the diagonal entries is disregarded. The input to the `B` port is the right side *M*-by-*N* matrix, *B*. The output is the unique solution of the equations, *M*-by-*N* matrix X, and is always sample based.

When the input is not positive definite, the block reacts with the behavior specified by the **Non-positive definite input** parameter. The following options are available:

- `Ignore` — Proceed with the computation and *do not* issue an alert. The output is *not* a valid solution.
- `Warning` — Proceed with the computation and display a warning message in the MATLAB® Command Window. The output is *not* a valid solution.
- `Error` — Display an error dialog box and terminate the simulation.

Note The **Non-positive definite input** parameter is a diagnostic parameter. Like all diagnostic parameters on the Configuration Parameters dialog box, it is set to `Ignore` in the code generated for this block by Real-Time Workshop® code generation software.

A length-*M* vector input for right side *B* is treated as an *M*-by-1 matrix.

Algorithm

Cholesky factorization uniquely factors the Hermitian positive definite input matrix S as

$$S = LL^*$$

where L is a lower triangular square matrix with positive diagonal elements.

The equation $SX=B$ then becomes

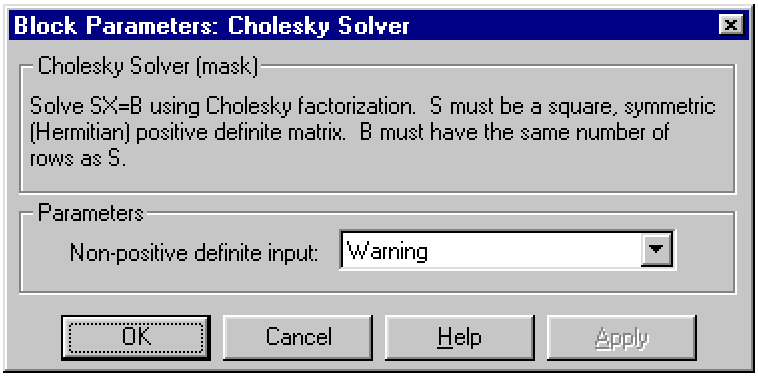
$$LL^*X = B$$

which is solved for X by making the substitution , and solving the following two triangular systems by forward and backward substitution, respectively.

$$LY = B$$

$$L^*X = Y$$

Dialog Box



Non-positive definite input
Response to nonpositive definite matrix inputs.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Autocorrelation LPC	Signal Processing Blockset
Cholesky Factorization	Signal Processing Blockset
Cholesky Inverse	Signal Processing Blockset
LDL Solver	Signal Processing Blockset
LU Solver	Signal Processing Blockset
QR Solver	Signal Processing Blockset
`chol`	MATLAB

See "Linear System Solvers" for related information.

Purpose

Design CIC compensator

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description

This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See “CIC Compensator Filter Design Dialog Box — Main Pane” in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Decimate signal using Cascaded Integrator-Comb filter

Library

Filtering / Multirate Filters

`dspmlti4`

Description

The CIC Decimation block performs a sample rate decrease (decimation) on an input signal by an integer factor. Cascaded Integrator-Comb (CIC) filters are a class of linear phase FIR filters comprised of a comb part and an integrator part.

The transfer function of a CIC decimator filter is

$$H(z) = H_I^{\,N}(z)H_c^{\,N}(z) = \frac{\left(1 - z^{-RM}\right)^N}{\left(1 - z^{-1}\right)^N} = \left[\sum_{k=0}^{RM-1} z^{-k}\right]^N$$

where

- H_I is the transfer function of the integrator part of the filter.
- H_C is the transfer function of the comb part of the filter.
- N is the number of sections. The number of sections in a CIC filter is defined as the number of sections in either the comb part *or* the integrator part of the filter, not as the total number of sections throughout the entire filter.
- R is the decimation factor.
- M is the differential delay.

The CIC Decimation block supports real and complex fixed-point inputs. Each channel of a complex input is treated as two real input channels.

CIC Filter Structures

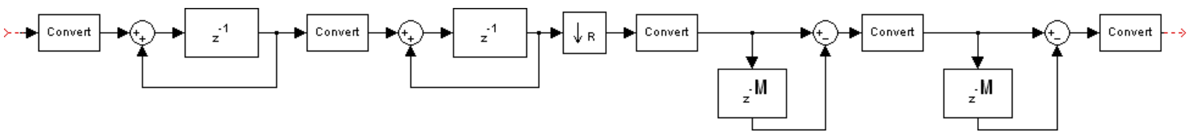
The filter structures supported by the CIC Decimation and CIC Interpolation blocks exactly match those created by Filter Design Toolbox™ `mfilt` CIC objects. If you have the Filter Design Toolbox product installed, you can create an `mfilt` object in any workspace to specify in the **Multirate filter variable** parameter of this block. Otherwise, you can specify the CIC filter completely using only block dialog parameters.

This block can be used to create either of the following CIC filter structures:

- "Decimator" on page 2-146
- "Zero-latency decimator" on page 2-146

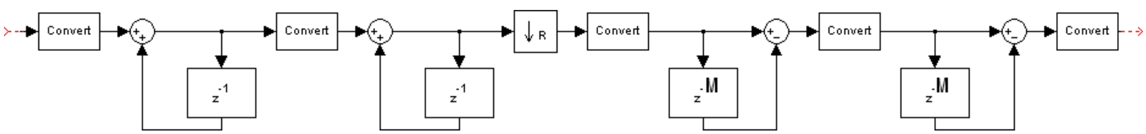
Decimator

This decimator has a latency of N, where N is the number of sections in either the comb or the integrator part of the filter.



Zero-latency decimator

This filter is the classical Hogenauer CIC decimator, which has zero latency.



Dialog Box

The CIC Decimation block can operate in two different modes. Select the mode in the **Coefficient source** group box. If you select

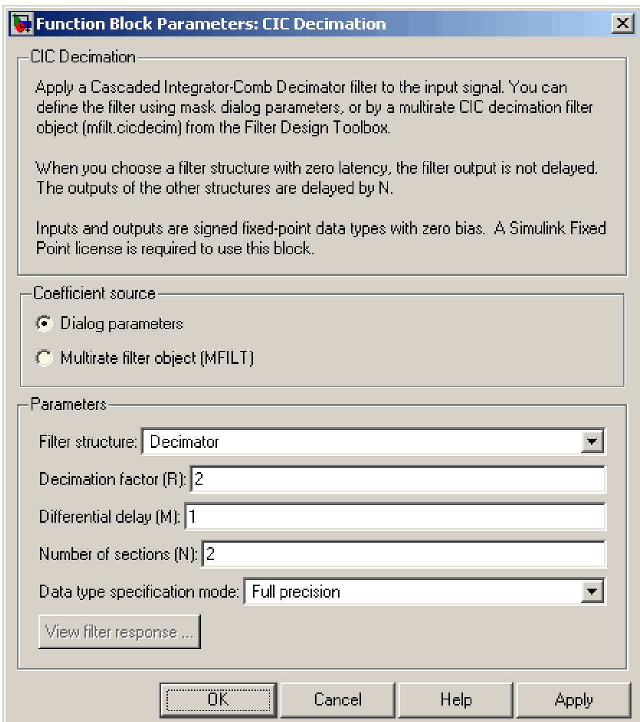
- **Dialog parameters**, you enter information about the filter such as structure and coefficients in the block mask.
- **Multirate filter object (MFILT)**, you specify the filter using a Filter Design Toolbox `mfilt` object.

Different items appear on the CIC Decimation block dialog depending on whether you select **Dialog parameters** or **Multirate filter object (MFILT)** in the **Coefficient source** group box. See the following sections for details:

- "Specify Filter Characteristics in Dialog" on page 2-147
- "Specify Multirate Filter Object" on page 2-151

Specify Filter Characteristics in Dialog

The **Main** pane of the CIC Decimation block dialog appears as follows when **Dialog parameters** is selected in the **Coefficient source** group box.



Filter structure

Select one of the following CIC filter structures:

- Decimator — CIC decimator with latency N
- Zero-latency decimator — Classical Hogenauer CIC decimator with zero latency

See "CIC Filter Structures" on page 2-145 for diagrams of these filter structures.

Decimation factor (R)

Specify the decimation factor of the filter.

Differential delay (M)

Specify the differential delay of the comb part of the filter, `M`, as shown in the diagrams in "CIC Filter Structures" on page 2-145.

Number of sections (N)

Specify the number of filter sections. This number is equal to the number of sections in either the comb part of the filter or in the integrator part of the filter. This value is not equal to the total number of sections in the comb and integrator parts combined.

Data type specification mode

Choose how you specify the fixed-point word length and fraction length of the filter sections and/or output.

- `Full precision` — In this mode, the word and fraction lengths of the filter sections and outputs are automatically selected for you. All word lengths are set to

  $$\text{word length} = \text{ceil}(N * \log_2(M * R)) + I$$

  where

  - I = input word length
  - M = differential delay
  - N = number of sections
  - R = decimation factor

All fraction lengths are set to the input fraction length.

- `Minimum section word lengths` — In this mode, you specify the word length of the filter output in the **Output word length** parameter. The word lengths of the filter sections and all fraction lengths are automatically selected for you such that each of the section word lengths is as small as possible. The precision of each filter section is less than in `Full precision` mode, but the range of each section is preserved.
- `Specify word lengths` — In this mode you specify the word lengths of the filter sections and output in the **Section word lengths** and **Output word length** parameters. The fraction lengths of the filter sections and output are automatically selected for you such that when least significant bits are discarded at each section, the range of that section is preserved.
- `Binary point scaling` — In this mode you fully specify the word and fraction lengths of the filter sections and output in the **Section word lengths**, **Section fraction lengths**, **Output word length**, and **Output fraction length** parameters.

Section word lengths

Specify the word length, in bits, of the filter sections.

This parameter is only visible if `Specify word lengths` or `Binary point scaling` is selected for the **Data type specification mode** parameter.

Section fraction lengths

Specify the fraction length of the filter sections.

This parameter is only visible if `Binary point scaling` is selected for the **Data type specification mode** parameter.

Output word length

Specify the word length, in bits, of the filter output.

This parameter is only visible if `Minimum section word lengths`, `Specify word lengths`, or `Binary point scaling` is selected for the **Data type specification mode** parameter.

Output fraction length
Specify the fraction length of the filter output.

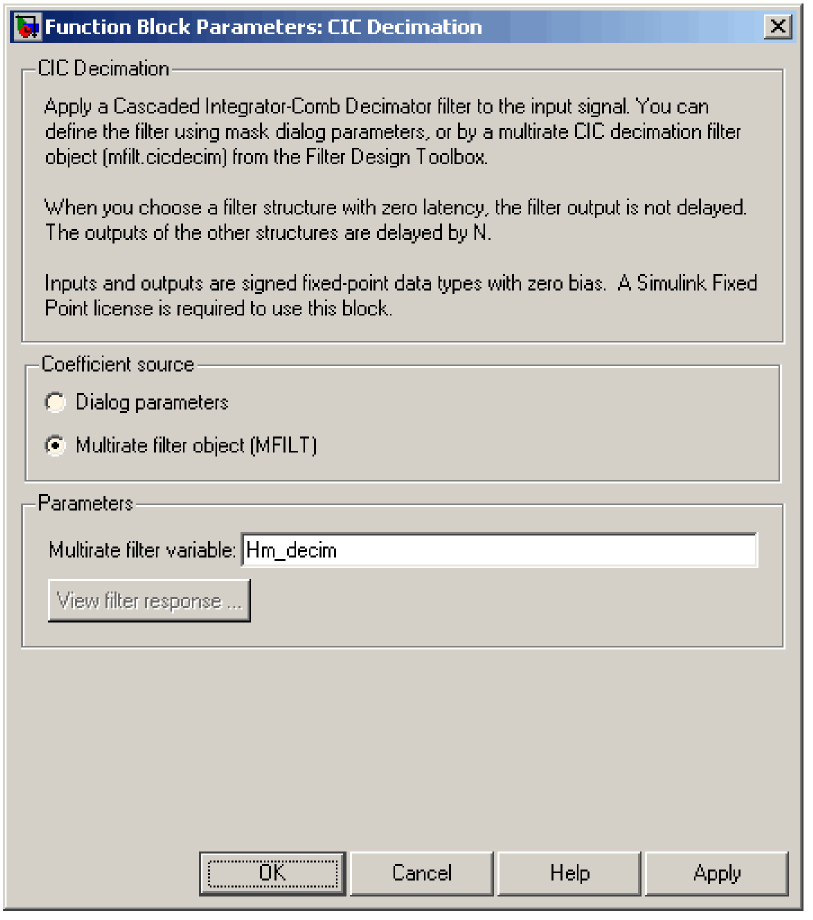
This parameter is only visible if `Binary point scaling` is selected for the **Data type specification mode** parameter.

View filter response
This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox™ product and displays the filter response of the filter defined in the block. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

Specify Multirate Filter Object

The **Main** pane of the CIC Decimation block dialog appears as follows when **Multirate filter object (MFILT)** is specified in the **Coefficient source** group box.



Multirate filter variable

Specify the multirate filter object (`mfilt`) that you would like the block to implement. You can do this in one of three ways:

- You can fully specify the `mfilt` object in the block mask.
- You can enter the variable name of a `mfilt` object that is defined in any workspace.
- You can enter a variable name for a `mfilt` object that is not yet defined, as shown in the default value.

For more information on creating `mfilt` objects, see the `mfilt` function reference page in the Filter Design Toolbox documentation.

View filter response

This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the `mfilt` object specified in the **Multirate filter variable** parameter. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

References

[1] Hogenauer, E.B., "An Economical Class of Digital Filters for Decimation and Interpolation," *IEEE Transactions on Acoustics, Speech and Signal Processing*, ASSP-29(2): pp. 155-162, 1981.

[2] Meyer-Baese, U., *Digital Signal Processing with Field Programmable Gate Arrays*, Springer Verlag, 2001.

[3] Harris, Fredric J., *Multirate Signal Processing for Communication Systems*, Prentice Hall PTR, 2004.

Supported Data Types

- Fixed point (signed only)
- 8-, 16-, and 32-bit signed integers

See Also

CIC Interpolation	Signal Processing Blockset
FIR Decimation	Signal Processing Blockset
FIR Interpolation	Signal Processing Blockset
`filter`	Filter Design Toolbox
`mfilt.cicdecim`	Filter Design Toolbox
`mfilt.cicinterp`	Filter Design Toolbox

Purpose

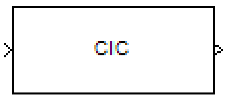
Design Cascaded Integrator-Comb (CIC) Filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description



This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See “CIC Filter Design Dialog Box — Main Pane” in the Filter Design Toolbox documentation for more information about the parameters of this block.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Interpolate signal using Cascaded Integrator-Comb filter

Library

Filtering / Multirate Filters

`dspmlti4`

Description

The CIC Interpolation block performs a sample rate increase (interpolation) on an input signal by an integer factor. Cascaded Integrator-Comb (CIC) filters are a class of linear phase FIR filters comprised of a comb part and an integrator part.

The transfer function of a CIC interpolator filter is

$$H(z) = {H_I}^N(z){H_C}^N(z) = \frac{(1-z^{-RM})^N}{(1-z^{-1})^N} = \left[\sum_{k=0}^{RM-1} z^{-k}\right]^N$$

where

- H_I is the transfer function of the integrator part of the filter.
- H_C is the transfer function of the comb part of the filter.
- N is the number of sections. The number of sections in a CIC filter is defined as the number of sections in either the comb part *or* the integrator part of the filter, not as the total number of sections throughout the entire filter.
- R is the interpolation factor.
- M is the differential delay.

The CIC Interpolation block supports real and complex fixed-point inputs. Each channel of a complex input is treated as two real input channels.

CIC Filter Structures

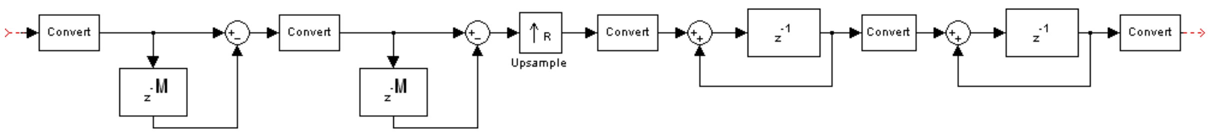
The filter structures supported by the CIC Interpolation and CIC Decimation blocks exactly match those created by Filter Design Toolbox™ `mfilt` CIC objects. If you have the Filter Design Toolbox product installed, you can create an `mfilt` object in any workspace to specify in the **Multirate filter variable** parameter of this block. Otherwise, you can specify the CIC filter completely using only block dialog parameters.

This block can be used to create either of the following CIC filter structures:

- "Interpolator" on page 2-157
- "Zero-latency interpolator" on page 2-157

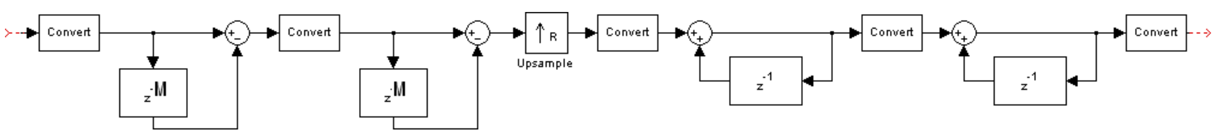
Interpolator

This interpolator has a latency of *N*, where *N* is the number of sections in either the comb or the integrator part of the filter.



Zero-latency interpolator

This filter is the classical Hogenauer CIC interpolator, which has zero latency.



Dialog Box

The CIC Interpolation block can operate in two different modes. Select the mode in the **Coefficient source** group box. If you select

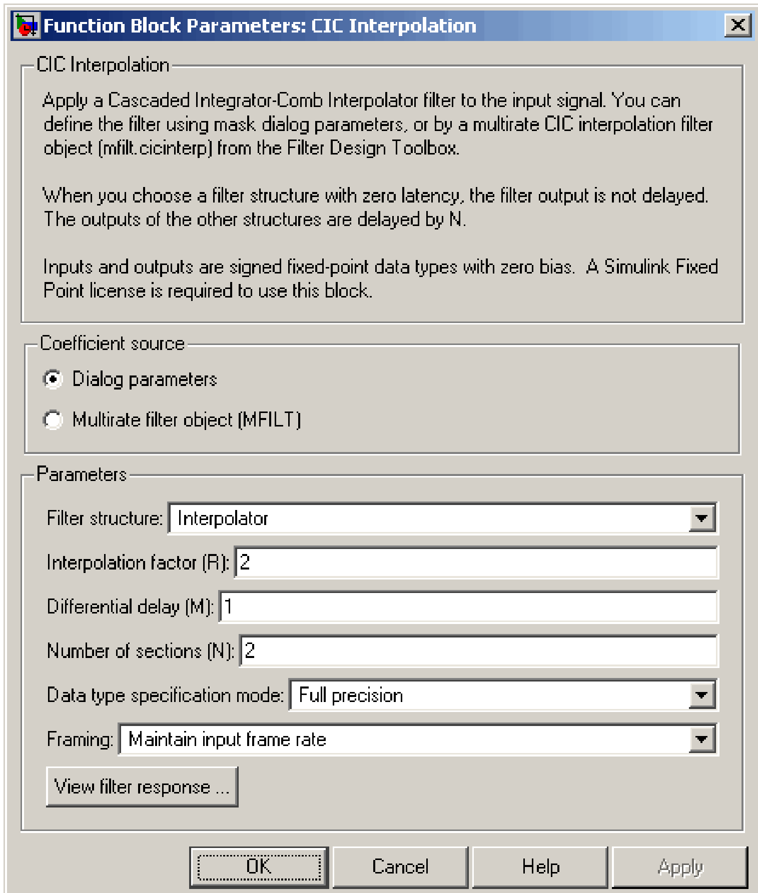
- **Dialog parameters**, you enter information about the filter such as structure and coefficients in the block mask
- **Multirate filter object (MFILT)**, you specify the filter using a Filter Design Toolbox `mfilt` object

Different items appear on the CIC Interpolation block dialog depending on whether you select **Dialog parameters** or **Multirate filter object (MFILT)** in the **Coefficient source** group box. See the following sections for details:

- "Specify Filter Characteristics in Dialog" on page 2-159
- "Specify Multirate Filter Object" on page 2-163

Specify Filter Characteristics in Dialog

The **Main** pane of the CIC Interpolation block dialog appears as follows when **Dialog parameters** is selected in the **Coefficient source** group box.



Filter structure

Select one of the following CIC filter structures:

- Interpolator — CIC interpolator with latency N
- Zero-latency interpolator — Classical Hogenauer CIC interpolator with zero latency

See "CIC Filter Structures" on page 2-156 for diagrams of these filter structures.

Interpolation factor (R)
Specify the interpolation factor of the filter.

Differential delay (M)
Specify the differential delay of the comb portion of the filter, `M`, as shown in the diagrams in "CIC Filter Structures" on page 2-145.

Number of sections (N)
Specify the number of filter sections. This number is equal to the number of sections in either the comb part of the filter or in the integrator part of the filter. This value is not equal to the total number of sections in the comb and integrator parts combined.

Data type specification mode
Choose how you specify the fixed-point word length and fraction length of the filter sections and/or output.

- `Full precision` — In this mode, the word and fraction lengths of the filter sections and outputs are automatically selected for you. The output and last section word lengths are set to

$$\text{word length} = \text{ceil}(\log_2(\frac{(R*M)^N}{R})) + I$$

where

- I = input word length
- M = differential delay
- N = number of sections
- R = interpolation factor

The other section word lengths are set is such a way as to accommodate the bit growth, as described in Hogenauer's paper [1]. All fraction lengths are set to the input fraction length.

- `Minimum section word lengths` — In this mode, you specify the word length of the filter output in the **Output word length** parameter. The word lengths of the filter sections are set in the same way as in `Full precision` mode.

  The section fraction lengths are set to the input fraction length. The output fraction length is set to the input fraction length minus the difference between the last section and output word lengths.

- `Specify word lengths` — In this mode you specify the word lengths of the filter sections and output in the **Section word lengths** and **Output word length** parameters. The fraction lengths of the filter sections are set such that the spread between word length and fraction length is the same as in full-precision mode. The output fraction length is set to the input fraction length minus the difference between the last section and output word lengths.

- `Binary point scaling` — In this mode you fully specify the word and fraction lengths of the filter sections and output in the **Section word lengths**, **Section fraction lengths**, **Output word length**, and **Output fraction length** parameters.

Section word lengths

Specify the word length, in bits, of the filter sections.

This parameter is only visible if `Specify word lengths` or `Binary point scaling` is selected for the **Data type specification mode** parameter.

Section fraction lengths

Specify the fraction length of the filter sections.

This parameter is only visible if `Binary point scaling` is selected for the **Data type specification mode** parameter.

Output word length

Specify the word length, in bits, of the filter output.

This parameter is only visible if `Minimum section word lengths`, `Specify word lengths`, or `Binary point scaling` is selected for the **Data type specification mode** parameter.

Output fraction length

Specify the fraction length of the filter output.

This parameter is only visible if `Binary point scaling` is selected for the **Data type specification mode** parameter.

Framing

For frame-based operation, specify the method by which to implement the interpolation; increase the output frame rate, or increase the output frame size. This parameter cannot be set to `Maintain input frame rate` for sample-based signals.

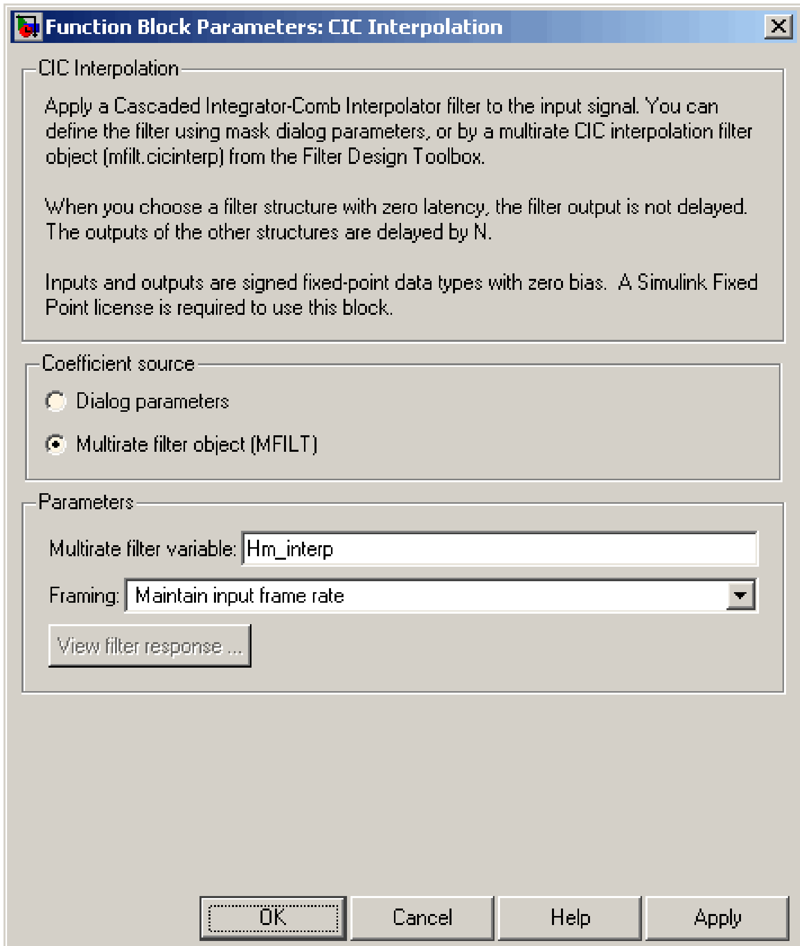
View filter response

This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox™ product and displays the filter response of the `mfilt` object specified in the **Multirate filter variable** parameter. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

Specify Multirate Filter Object

The **Main** pane of the CIC Interpolation block dialog appears as follows when **Multirate filter object (MFILT)** is specified in the **Coefficient source** group box.



Multirate filter variable

Specify the multirate filter object (`mfilt`) that you would like the block to implement. You can do this in one of three ways:

- You can fully specify the `mfilt` object in the block mask.
- You can enter the variable name of a `mfilt` object that is defined in any workspace.
- You can enter a variable name for a `mfilt` object that is not yet defined, as shown in the default value.

For more information on creating `mfilt` objects, see the `mfilt` function reference page in the Filter Design Toolbox documentation.

Framing

For frame-based operation, specify the method by which to implement the interpolation; reduce the output frame rate, or reduce the output frame size. This parameter cannot be set to `Maintain input frame rate` for sample-based signals.

View filter response

This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the `mfilt` object specified in the **Multirate filter variable** parameter. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

References

[1] Hogenauer, E.B., “An Economical Class of Digital Filters for Decimation and Interpolation,” *IEEE Transactions on Acoustics, Speech and Signal Processing*, ASSP-29(2): pp. 155-162, 1981.

[2] Meyer-Baese, U., *Digital Signal Processing with Field Programmable Gate Arrays*, Springer Verlag, 2001.

[3] Harris, Fredric J., *Multirate Signal Processing for Communication Systems*, Prentice Hall PTR, 2004.

Supported Data Types

- Fixed point (signed only)
- 8-, 16-, and 32-bit signed integers

See Also

CIC Decimation	Signal Processing Blockset
FIR Decimation	Signal Processing Blockset
FIR Interpolation	Signal Processing Blockset
`filter`	Filter Design Toolbox
`mfilt.cicdecim`	Filter Design Toolbox
`mfilt.cicinterp`	Filter Design Toolbox

Complex Cepstrum

Purpose

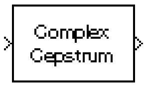
Compute complex cepstrum of input

Library

Transforms

`dspxfrm3`

Description



The Complex Cepstrum block computes the complex cepstrum of each channel in the real-valued M-by-N input matrix, u. For both sample-based and frame-based inputs, the block assumes that each input column is a frame containing M consecutive samples from an independent channel. The block does not accept complex-valued inputs.

The input is altered by the application of a linear phase term so that there is no phase discontinuity at $\pm\pi$ radians. That is, each input channel is independently zero padded and circularly shifted to have zero phase at π radians.

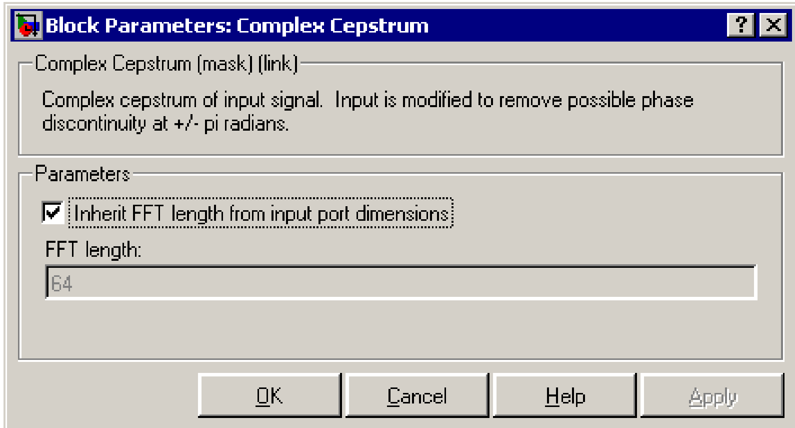
The output is a real M_o-by-N matrix, where M_o is specified by the **FFT length** parameter. Each output column contains the length-M_o complex cepstrum of the corresponding input column.

```
y = cceps(u,Mo)     % Equivalent MATLAB code
```

When you select the **Inherit FFT length from input port dimensions** check box, the output frame size matches the input frame size (M_o=M). In this case, a *sample-based* length-M row vector input is processed as a single channel (that is, as an M-by-1 column vector), and the output is a length-M row vector. A 1-D vector input is *always* processed as a single channel, and the output is a 1-D vector.

The output is always sample based, and the output port rate is the same as the input port rate.

Dialog Box



Inherit FFT length from input port dimensions
When selected, matches the output frame size to the input frame size.

FFT length
The number of frequency points at which to compute the FFT, which is also the output frame size, M_o. This parameter is available when you do not select **Inherit FFT length from input port dimensions**.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

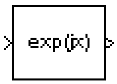
DCT	Signal Processing Blockset
FFT	Signal Processing Blockset
Real Cepstrum	Signal Processing Blockset
`cceps`	Signal Processing Toolbox

Purpose Compute complex exponential function

Library Math Functions / Math Operations

`dspmathops`

Description

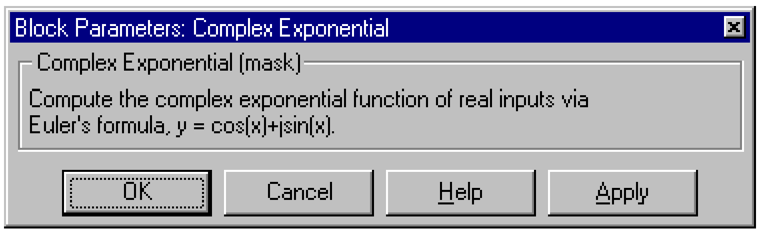


The Complex Exponential block computes the complex exponential function for each element of the real input, u.

$$y = e^{ju} = \cos u + j\sin u$$

where $j = \sqrt{-1}$. The output is complex, with the same size and frame status as the input.

Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Math Function	Simulink
Sine Wave	Signal Processing Blockset
`exp`	MATLAB

Purpose

Generate square, diagonal matrix

Library

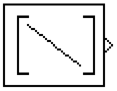
- Signal Processing Sources

  `dspsrcs4`

- Math Functions / Matrices and Linear Algebra / Matrix Operations

  `dspmtrx3`

Description

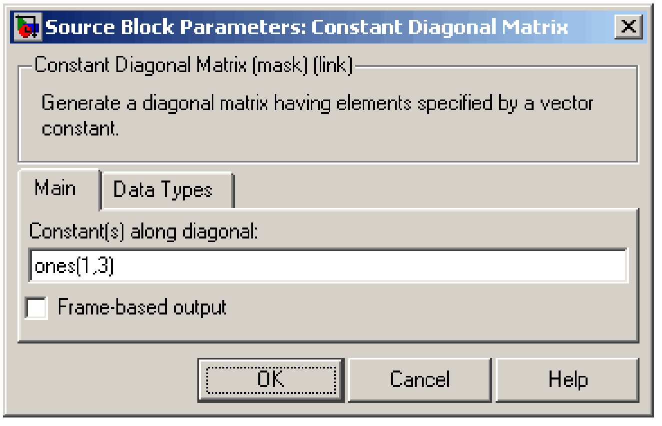


The Constant Diagonal Matrix block outputs a square diagonal matrix constant. The **Constant along diagonal** parameter determines the values along the matrix diagonal. This parameter can be a scalar to be repeated for all elements along the diagonal, or a vector containing the values of the diagonal elements. To generate the identity matrix, set the **Constant along diagonal** to `1`, or use the Identity Matrix block.

The output is frame based when you select the **Frame-based output** check box; otherwise, the output is sample based.

Dialog Box

The **Main** pane of the Constant Diagonal Matrix block dialog appears as follows.



Constant Diagonal Matrix

Opening this dialog box causes a running simulation to pause. See "Changing Source Block Parameters During Simulation" in the online Simulink® documentation for details.

Constant(s) along diagonal

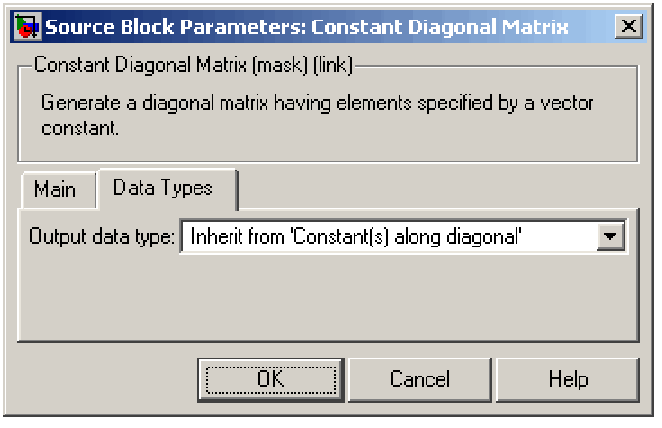
Specify the values of the elements along the diagonal. You can input a scalar or a vector. Tunable.

When you specify any data type information in this field, it is overridden by the value of the **Output data type** parameter on the **Data Types** pane, unless you select `Inherit from 'Constant(s) along diagonal'`.

Frame-based output

Select to cause the output of the block to be frame based. Otherwise, the output is sample based.

The **Data types** pane of the Constant Diagonal Matrix block dialog appears as follows.



Output data type

Specify the output data type in one of the following ways:

- Choose one of the built-in data types from the list.

- Choose `Fixed-point` to specify the output data type and scaling in the **Signed**, **Word length**, **Set fraction length in output to,** and **Fraction length** parameters.
- Choose `User-defined` to specify the output data type and scaling in the **User-defined data type**, **Set fraction length in output to,** and **Fraction length** parameters.
- Choose `Inherit from 'Constant(s) along diagonal'` to set the output data type and scaling to match the values of the **Constant(s) along diagonal** parameter on the **Main** pane.
- Choose `Inherit via back propagation` to set the output data type and scaling to match the next block downstream.

The value of this parameter overrides any data type information specified in the **Constant(s) along diagonal** parameter on the **Main** pane, except when you select `Inherit from 'Constant(s) along diagonal'`.

Signed
Select to output a signed fixed-point signal. Otherwise, the signal is unsigned. This parameter is only visible when you select `Fixed-point` for the **Output data type** parameter.

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

User-defined data type
Specify any built-in or fixed-point data type. You can specify fixed-point data types using the following Simulink® Fixed Point™ functions: `sfix`, `ufix`, `sint`, `uint`, `sfrac`, and `ufrac`. This parameter is only visible when you select `User-defined` for the **Output data type** parameter.

Set fraction length in output to
Specify the scaling of the fixed-point output by either of the following two methods:

- Choose `Best precision` to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose `User-defined` to specify the output scaling in the **Fraction length** parameter.

This parameter is only visible when you select `Fixed-point` for the **Output data type** parameter, or when you select `User-defined` and the specified output data type is a fixed-point data type.

Fraction length

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

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Create Diagonal Matrix	Signal Processing Blockset
Constant	Simulink
Identity Matrix	Signal Processing Blockset
`diag`	MATLAB

Purpose

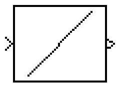
Generate ramp signal with length based on input dimensions

Library

Signal Operations

`dspsigops`

Description



The Constant Ramp block generates the constant ramp signal

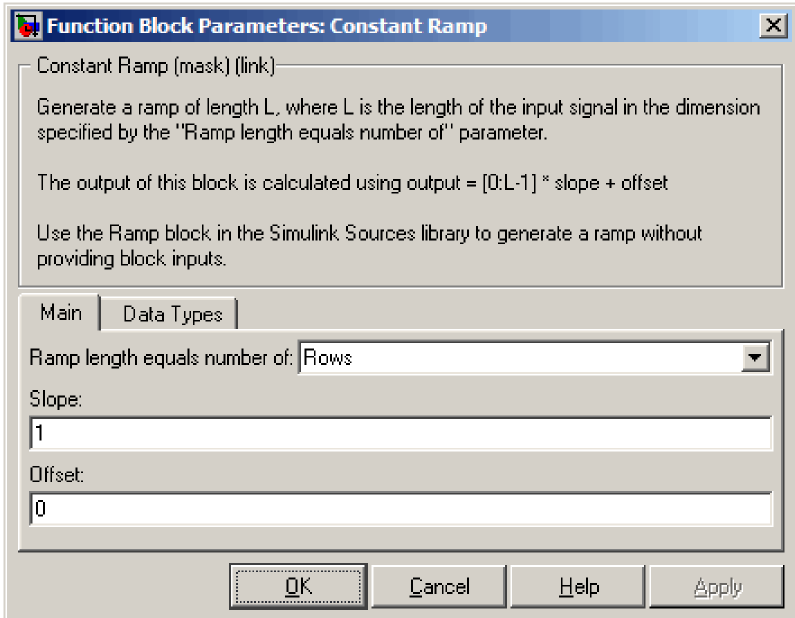
```
y = (0:L-1)*m + b
```

where m is the slope specified by the scalar **Slope** parameter, and b is the y-intercept specified by the scalar **Offset** parameter.

For a 1-D vector input, L is equal to the length of the input vector. For an N-D array input, the length L of the output ramp is equal to the length of the input in the dimension specified by the **Ramp length equals number of** or **Dimension** parameter. The output, y, is always a 1-D vector.

Dialog Box

The **Main** pane of the Constant Ramp block dialog appears as follows.



Ramp length equals number of
Specify whether the length of the output ramp is the number of rows, number of columns, or the length of the specified dimension of the input.

Dimension
Specify the one-based dimension of the input array that determines the length of the output ramp.

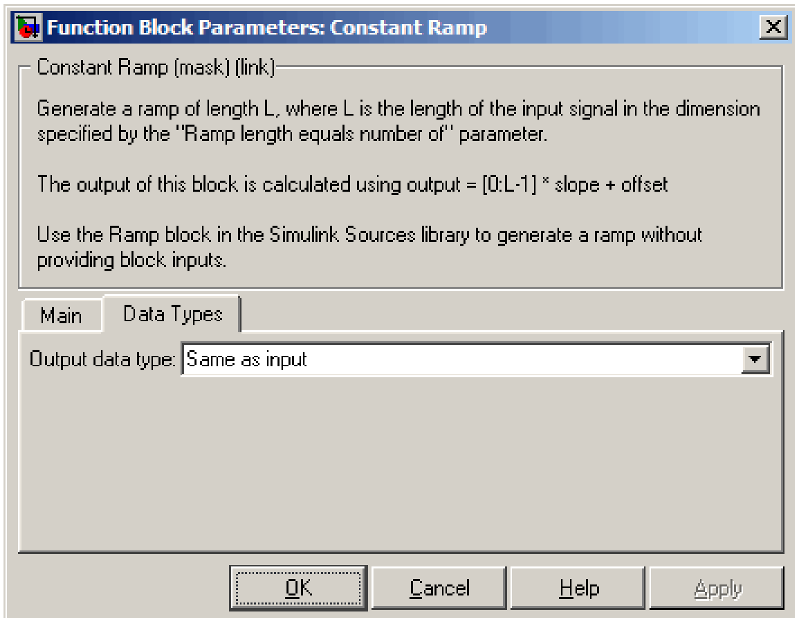
This parameter is only visible when you select `Elements in specified dimension` for the **Ramp length equals number of** parameter.

Slope
Specify the scalar slope of the ramp.

Offset

Specify the scalar *y*-intercept of the ramp.

The **Data types** pane of the Constant Ramp block dialog appears as follows.



Output data type

Specify the output data type in one of the following ways:

- Select `Same as input` to force the data type of the output to be the same as the data type of the input to the block.
- Select one of the built-in data types from the list.
- Select `Fixed-point` to specify the output data type and scaling in the **Signed**, **Word length**, **Set fraction length in output to,** and **Fraction length** parameters.

- Select `User-defined` to specify the output data type and scaling in the **User-defined data type**, **Set fraction length in output to,** and **Fraction length** parameters.
- Select `Inherit via back propagation` to set the output data type and scaling to match the next block downstream.

This block differs from other Signal Processing Blockset™ blocks in that unless you choose `Same as input` for this parameter, the data types of the input and the output do not need to be the same.

Signed
Select to output a signed fixed-point signal. Otherwise, the signal is unsigned. This parameter is only visible when you select `Fixed-point` for the **Output data type** parameter.

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

User-defined data type
Specify any built-in or fixed-point data type. You can specify fixed-point data types using the following Simulink® Fixed Point™ functions: `sfix`, `ufix`, `sint`, `uint`, `sfrac`, and `ufrac`. This parameter is only visible when you select `User-defined` for the **Output data type** parameter.

Set fraction length in output to
Specify the scaling of the fixed-point output by either of the following two methods:

- Select `Best precision` to have the output scaling automatically set such that the output signal has the best possible precision.
- Select `User-defined` to specify the output scaling in the **Fraction length** parameter.

This parameter is only visible when you select `Fixed-point` for the **Output data type** parameter, or when you select

User-defined and the specified output data type is a fixed-point data type.

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

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

This block differs from other Signal Processing Blockset blocks in that unless you choose Same as input for the **Output data type** parameter, the data types of the input and the output do not need to be the same.

See Also

Create Diagonal Matrix	Signal Processing Blockset
Constant	Simulink
Identity Matrix	Signal Processing Blockset

Contiguous Copy

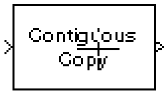
Purpose

Create discontiguous input in contiguous block of memory

Library

`dspobslib`

Description



Contiguous
Copy

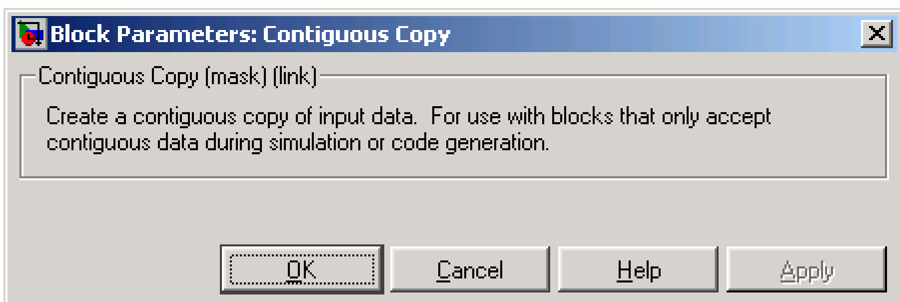
Note The Contiguous Copy block is still supported but is likely to be obsoleted in a future release.

The Contiguous Copy block copies the input to a contiguous block of memory, and passes this new copy to the output. The output is identical to the input, but is guaranteed to reside in a contiguous section of memory.

Because Simulink® software employs an efficient copy-by-reference method for propagating data in a model, some operations produce outputs with discontiguous memory locations.

Although this does not present a problem during simulation, blocks linked to versions of DSP Blockset prior to 4.0 may require contiguous inputs for code generation with the Real-Time Workshop® product. When such blocks are used in a model intended for code generation, they should be preceded by the Contiguous Copy block to ensure that their inputs are contiguous.

Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point
- Boolean
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

Convert 1-D to 2-D

Purpose

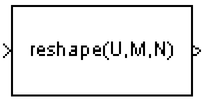
Reshape 1-D or 2-D input to 2-D matrix with specified dimensions

Library

Signal Management / Signal Attributes

`dspsigattribs`

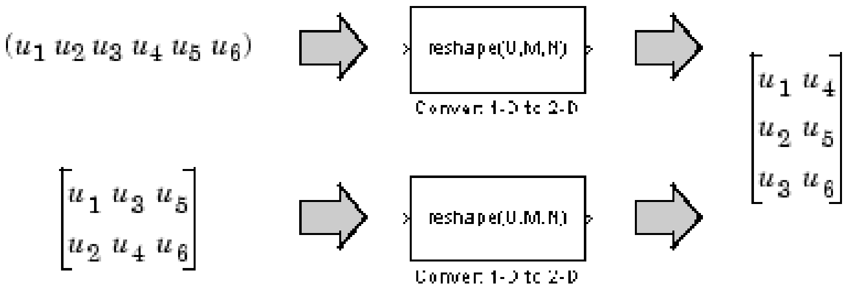
Description



The Convert 1-D to 2-D block reshapes a length-M_i 1-D vector or an M_i-by-N_i matrix to an M_o-by-N_o matrix, where M_o is specified by the **Number of output rows** parameter, and N_o is specified by the **Number of output columns** parameter.

```
y = reshape(u,Mo,No)       % Equivalent MATLAB code
```

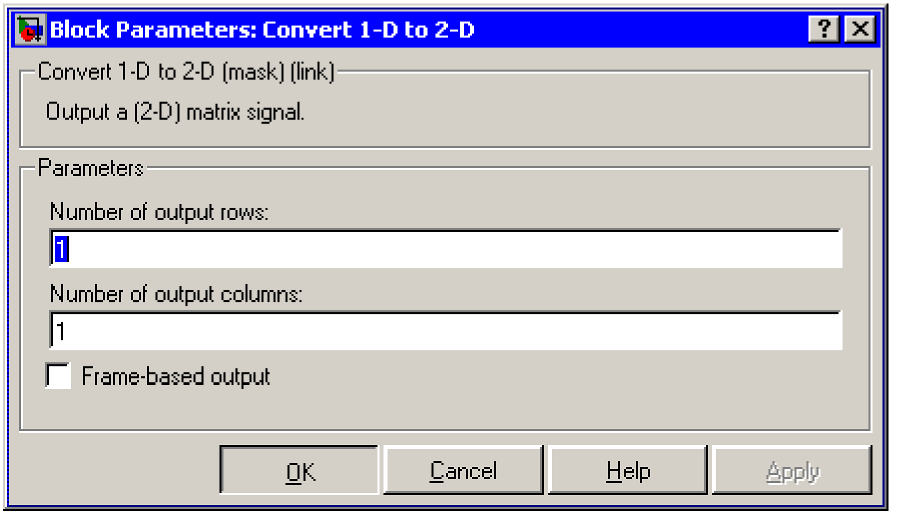
The input is reshaped *columnwise*, as shown in the two cases below. The length-6 vector and the 2-by-3 matrix are both reshaped to the same 3-by-2 output matrix.



An error is generated when $(M_o{*}N_o) \neq (M_i{*}N_i)$. That is, the total number of input elements must be conserved in the output.

The output is frame based when you select the **Frame-based output** check box; otherwise, the output is sample based.

Dialog Box



Number of output rows
: The number of rows, M_o, in the output matrix.

Number of output columns
: The number of rows, N_o, in the output matrix.

Frame-based output
: Creates a frame-based output when selected.

Supported Data Types

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

See Also

Buffer	Signal Processing Blockset
Convert 2-D to 1-D	Signal Processing Blockset
Frame Status Conversion	Signal Processing Blockset
Reshape	Simulink
Submatrix	Signal Processing Blockset

Purpose

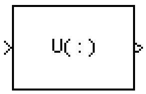
Convert 2-D matrix input to 1-D vector

Library

Signal Management / Signal Attributes

`dspsigattribs`

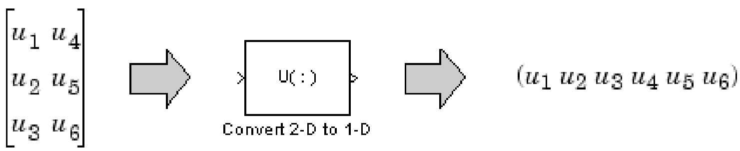
Description



The Convert 2-D to 1-D block reshapes an *M*-by-*N* matrix input to a 1-D vector with length *M***N*.

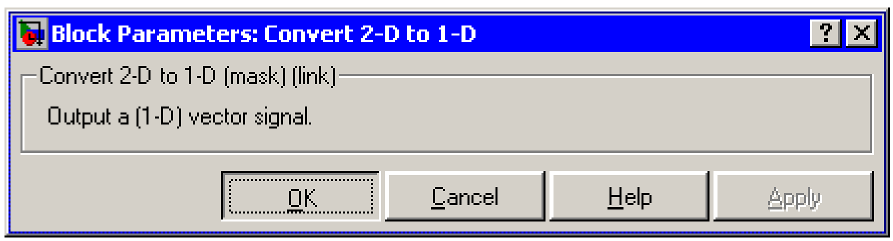
```
y = u(:)     % Equivalent MATLAB code
```

The input is reshaped *columnwise*, as shown below for a 3-by-2 matrix.



The output is always sample-based.

Dialog Box



Supported Data Types

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

See Also

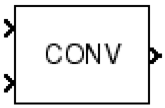
Buffer	Signal Processing Blockset
Convert 1-D to 2-D	Signal Processing Blockset
Frame Status Conversion	Signal Processing Blockset
Reshape	Simulink
Submatrix	Signal Processing Blockset

Purpose Compute convolution of two inputs

Library Signal Operations

dspsigops

Description



The Convolution block convolves the first dimension of a sample-based N-D input array u, with the first dimension of a sample-based N-D input array v. The block can also independently convolve a sample-based vector with the first-dimension of an N-D input array. For frame-based inputs, the Convolution block convolves analogous columns of an M_u-by-N input matrix u and an M_v-by-N input matrix v. The Convolution block can also independently convolve a single-channel frame-based column vector with each column of a multiple-channel frame-based matrix.

The frame status of both inputs to the Convolution block must be the same. The output of the block is always sample-based.

The Convolution block accepts both real and complex floating-point and fixed-point inputs. Fixed-point signals are not supported for the frequency domain.

Convolving Frame-Based Inputs

When the inputs to the Convolution block are a frame based M_u-by-N input matrix u and an M_v-by-N input matrix v, the output, y, is a sample-based (M_u+M_v-1)-by-N matrix whose jth column has elements

$$y_{i,j} = \sum_{k=1}^{\max(M_u,M_v)} u_{k,j} v_{(i-k+1),j} \qquad 1 \le i \le (M_u + M_v - 1)$$

Inputs u and v are zero when indexed outside of their valid ranges. When both inputs are real, the output is real; when one or both inputs are complex, the output is complex.

When one input is a column vector (single channel) and the other is a matrix (multiple channels), the single-channel input is independently convolved with each channel of the multichannel input. For example, when u is a M_u-by-1 column vector and v is an M_v-by-N matrix, the output is an (M_u+M_v-1)-by-N matrix whose jth column has elements

$$y_{i,j} = \sum_{k=1}^{\max(M_u,M_v)} u_k v_{(i-k+1),j} \quad 1 \le i \le (M_u + M_v - 1)$$

Convolving Sample-Based Inputs

The Convolution block supports sample-based N-D input arrays. The convolution of N-D array input is always computed across the first dimension. If both inputs are N-D arrays, the size of their first dimension can differ, but the size of all other dimensions must be equal. For example, when u is an M_u-by-N-by-P array, and v is an M_v-by-N-by-P array, the output is an (M_u+M_v-1)-by-N-by-P array.

When one input is an N-D sample-based array and the other is a vector, the vector is independently convolved with the first dimension of the N-D input. For example, when u is a M_u-by-1 column vector and v is an M_v-by-N-by-P array, the output is an (M_u+M_v-1)-by-N-by-P array.

The Convolution block also accepts two vector inputs. When u and v are sample-based vectors with lengths M_u and M_v, the Convolution block performs the vector convolution

$$y_i = \sum_{k=1}^{\max(M_u,M_v)} u_k v_{(i-k+1)} \quad 1 \le i \le (M_u + M_v - 1)$$

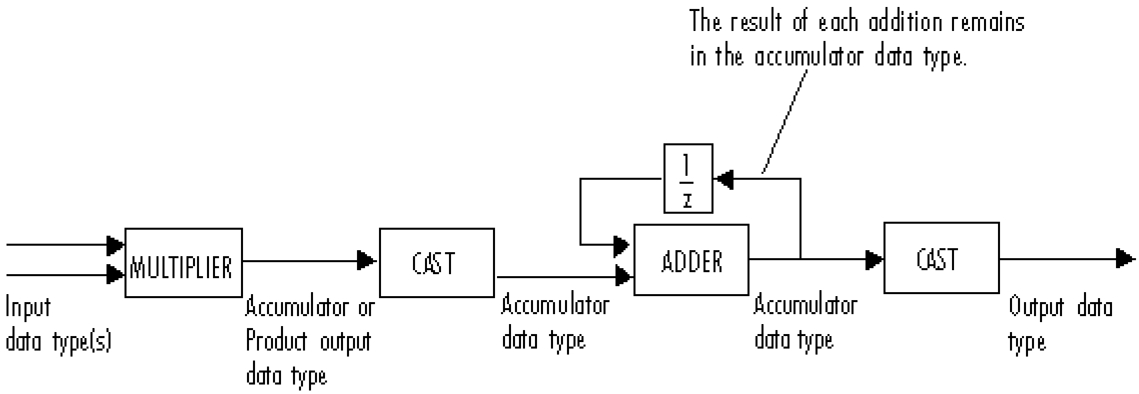
The dimensions of the sample-based output vector are determined by the dimensions of the input vectors:

- When both inputs are row vectors, or when one input is a row vector and the other is a 1-D vector, the output is a 1-by-(M_u+M_v-1) row vector.
- When both inputs are column vectors, or when one input is a column vector and the other is a 1-D vector, the output is a (M_u+M_v-1)-by-1 column vector.

- When both inputs are 1-D vectors, the output is a 1-D vector of length M_u+M_v-1.

Fixed-Point Data Types

The following diagram shows the data types used within the Convolution block for fixed-point signals (time domain only).

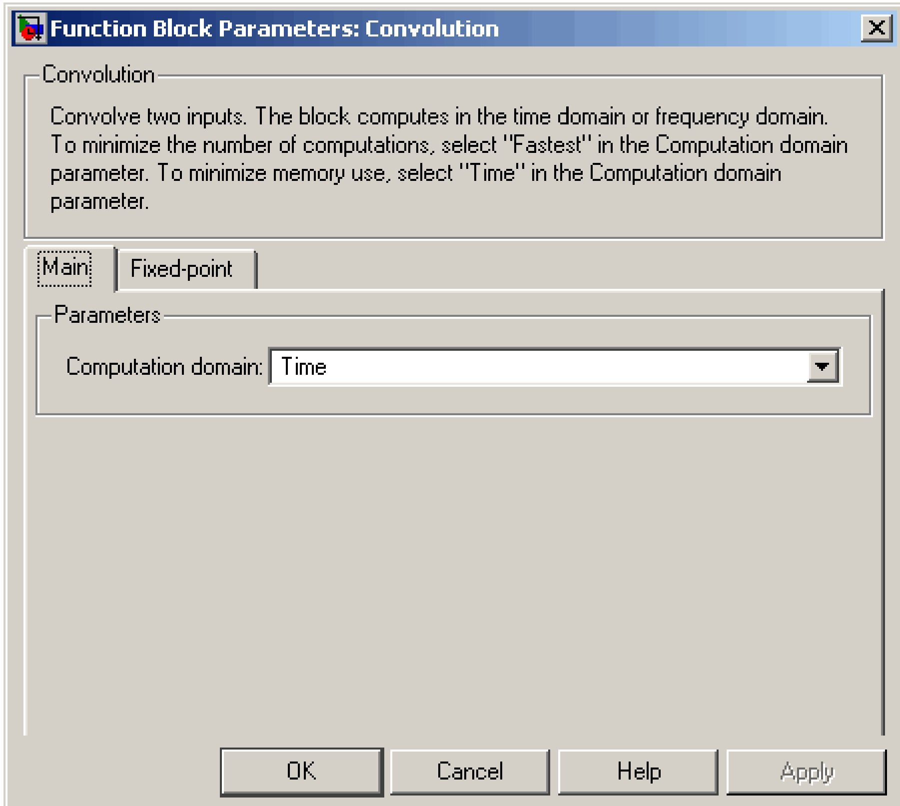


You can set the product output, accumulator, and output data types in the block dialog as discussed in the next section.

The output of the multiplier is in the product output data type when the input is real. When the input is complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”.

Dialog Box

The **Main** pane of the Convolution block dialog appears as follows.



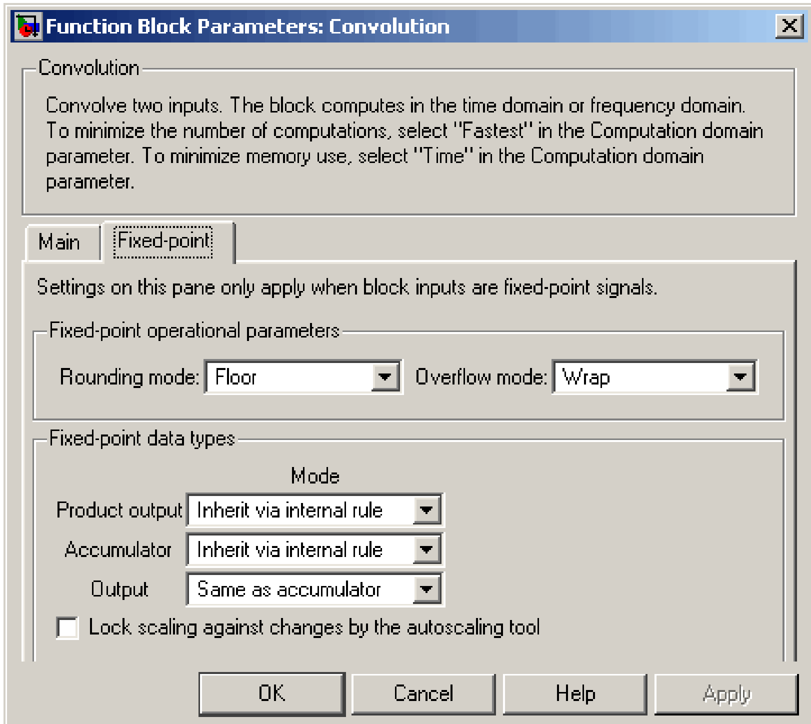
Computation domain

Set the domain in which the block computes convolutions:

- `Time` — The block computes in the time domain, which minimizes memory use.
- `Frequency` — The block computes in the frequency domain, which might require fewer computations than computing in the time domain, depending on the input length.

- `Fastest` — The block computes in the domain, which minimizes the number of computations.

The **Fixed-point** pane of the Convolution block dialog appears as follows.



Note Fixed-point signals are only supported for the time domain. To use the parameters on this pane, make sure `Time` is selected for the **Computation domain** parameter on the **Main** pane.

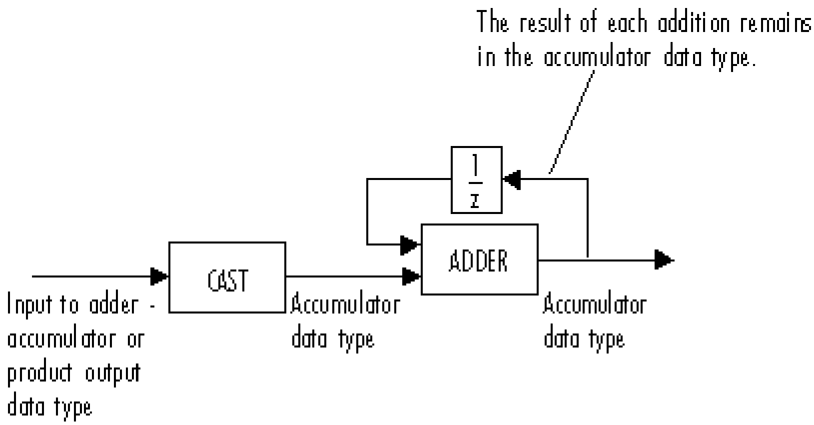
Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Product output
Use this parameter to specify how you want to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-44 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as first input`, these characteristics match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator



As depicted in this figure, inputs to the accumulator are cast to the accumulator data type. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you want to designate this accumulator word and fraction lengths.

You also use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. See “Multiplication Data Types” for more information.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see “Inherit via Internal Rule”.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as first input`, these characteristics match those of the first input to the block.

- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the data type and scaling of the output of the block:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.

  A special case occurs when `Inherit via internal rule` is specified for **Accumulator**, and both block inputs are complex. In that case, the output word length is one less than the accumulator word length.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as first input`, these characteristics match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

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

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)

- 8-, 16-, and 32-bit signed integers

See Also

Correlation	Signal Processing Blockset
`conv`	MATLAB

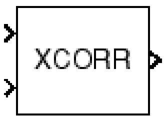
Correlation

Purpose Compute cross-correlation of two inputs

Library Statistics

`dspstat3`

Description



The Correlation block computes the cross-correlation of the first dimension of a sample-based N-D input array u, and the first dimension of a sample-based N-D input array v. The block can also independently cross-correlate a sample-based vector with the first-dimension of an N-D input array. For frame-based inputs, the Correlation block computes the cross-correlation of analogous columns of an M_u-by-N input matrix u and an M_v-by-N input matrix v. The Correlation block can also independently cross-correlate a single-channel frame-based column vector with each column of a multiple-channel frame-based matrix.

The frame status of both inputs to the Correlation block must be the same. The output of the block is always sample-based.

The Correlation block accepts both real and complex floating-point and fixed-point inputs. Fixed-point signals are not supported for the frequency domain.

Correlating Frame-Based Inputs

When the inputs to the Correlation block are an M_u-by-N frame-based input matrix u and an M_v-by-N frame-based input matrix v, the output, y, is a sample-based (M_u+M_v-1)-by-N matrix whose jth column has elements

$$y_{(i+M_v),j} = \sum_{k=1}^{max(M_u,M_v)} u_{k,j} v^*_{(k-i),j} \qquad -M_u < i < M_v$$

where * denotes the complex conjugate. Inputs u and v are zero when indexed outside of their valid ranges. When both inputs are real, the output is real; when one or both inputs are complex, the output is complex.

When one input is a column vector (single channel) and the other is a matrix (multiple channels), the single-channel input is independently cross-correlated with each channel of the multichannel input. Each column of the input represents a separate channel. For example, when u is a M_u-by-1 column vector and v is an M_v-by-N matrix, the output is an (M_u+M_v-1)-by-N matrix whose jth column has elements

$$y_{(i+M_v),j} = \sum_{k=1}^{max(M_u,M_v)} u_k v^*_{(k-i),j} \qquad -M_u < i < M_v$$

Correlating Sample-Based Inputs

The Correlation block supports sample-based N-D array input. The cross-correlation for sample-based N-D inputs is always computed across the first dimension. If both inputs are N-D arrays, the size of their first dimensions can differ, but the size of all other dimensions must be equal. For example, when u is an M_u-by-N-by-P array and v is an M_v-by-N-by-P array, the output, y, is a sample-based (M_u+M_v-1)-by-N-by-P array.

When one input is an N-D sample-based array and the other is a vector, the vector is independently cross-correlated with each column of the N-D input. For example, when u is a M_u-by-1 column vector and v is an M_v-by-N-by-P array, the output is an $(M_u+M_v$-1)-by-N-by-P array.

The Correlation block also accepts two vector inputs. When u and v are sample-based column vectors with lengths M_u and M_v, the Correlation block performs the vector cross-correlation according to the following equation:

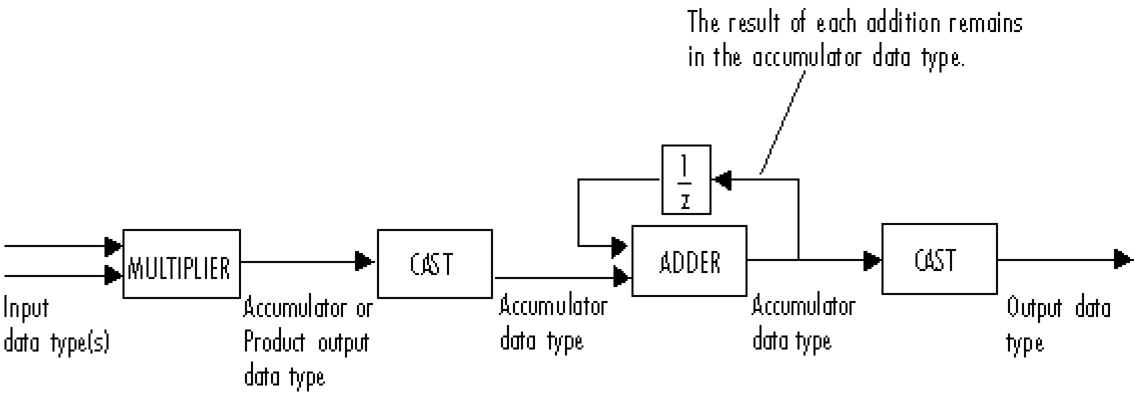
$$y_{(i+M_v)} = \sum_{k=1}^{max(M_u,M_v)} u_k v^*_{(k-i)} \qquad -M_u < i < M_v$$

The dimensions of the sample-based output vector are determined by the dimensions of the input vectors:

- When both inputs are column vectors, or when one input is a column vector and the other is a 1-D vector, the output is a (M_u+M_v-1)-by-1 column vector.
- When both inputs are row vectors, or when one input is a row vector and the other is a 1-D vector, the output is a 1-by-(M_u+M_v-1) row vector.
- When both inputs are 1-D vectors, the output is a 1-D vector of length M_u+M_v-1.

Fixed-Point Data Types

The following diagram shows the data types used within the Correlation block for fixed-point signals (time domain only).

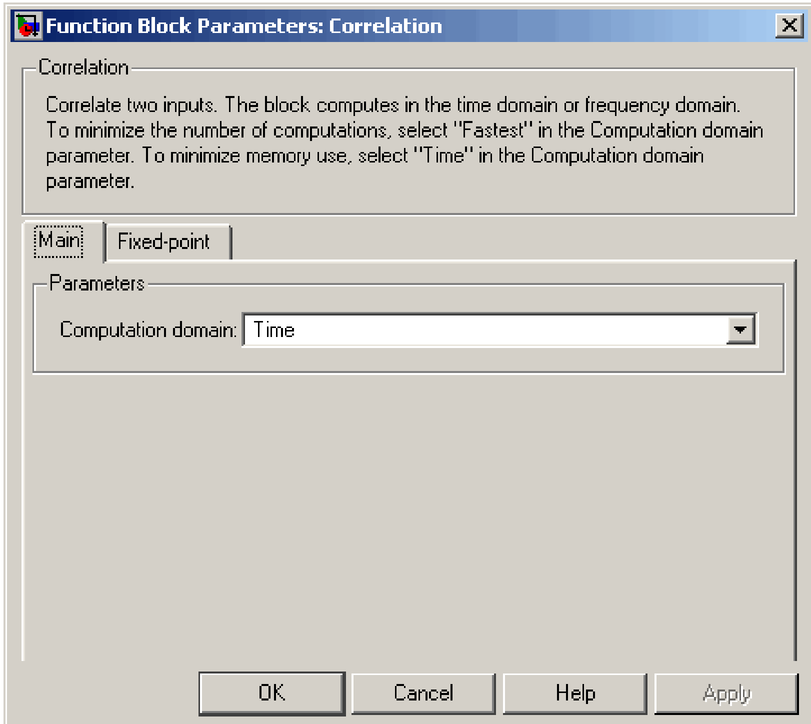


You can set the product output, accumulator, and output data types in the block dialog as discussed in the next section.

The output of the multiplier is in the product output data type when the input is real. When the input is complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”.

Dialog Box

The **Main** pane of the Correlation block dialog appears as follows.

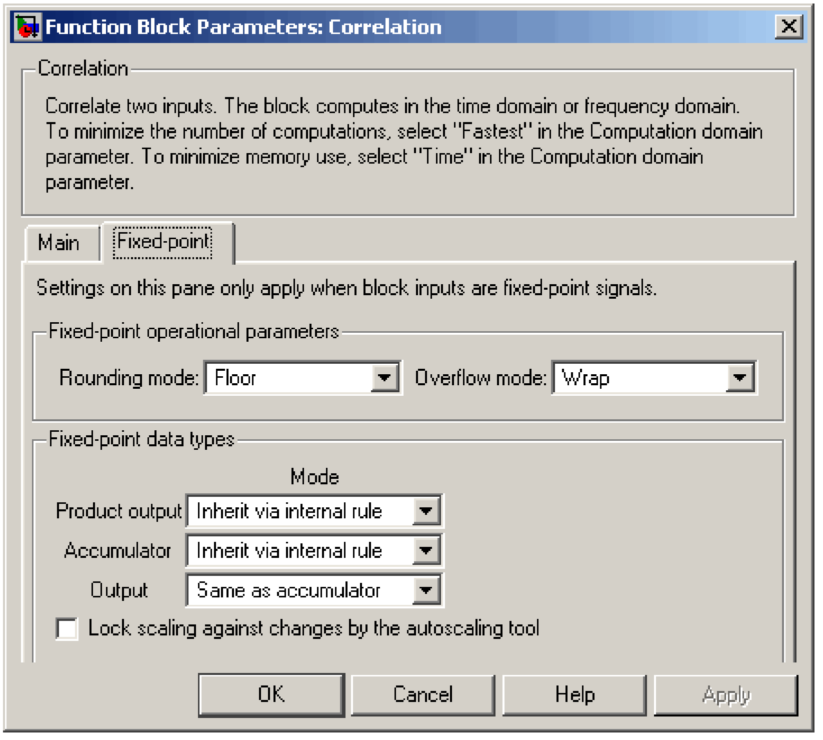


Computation domain

Set the domain in which the block computes correlations:

- `Time` — The block computes in the time domain, which minimizes memory use.
- `Frequency` — The block computes in the frequency domain, which might require fewer computations than computing in the time domain, depending on the input length.
- `Fastest` — The block computes in the domain, which minimizes the number of computations.

The **Fixed-point** pane of the Correlation block dialog appears as follows.



Note Fixed-point signals are only supported for the time domain. To use the parameters on this pane, make sure `Time` is selected for the **Computation domain** parameter on the **Main** pane.

Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode

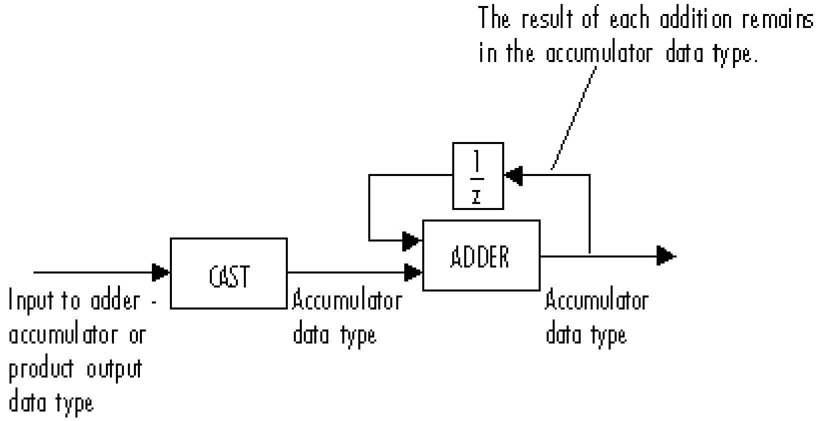
Select the overflow mode for fixed-point operations.

Product output

Use this parameter to specify how you want to designate the product output word and fraction lengths. See “Fixed-Point Data Types” on page 2-44 and “Multiplication Data Types” for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see “Inherit via Internal Rule”.
- When you select `Same as first input`, these characteristics match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator



As depicted above, inputs to the accumulator are cast to the accumulator data type. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you want to designate this accumulator word and fraction lengths.

You also use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. See "Multiplication Data Types" for more information.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as first input`, these characteristics match those of the first input to the block.

- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the word length and fraction length of the output of the block:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.

  A special case occurs when `Inherit via internal rule` is specified for **Accumulator**, and both block inputs are complex. In that case, the output word length is one less than the accumulator word length.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as first input`, these characteristics match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

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

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)

- 8-, 16-, and 32-bit signed integers

See Also

Autocorrelation	Signal Processing Blockset
Convolution	Signal Processing Blockset
`xcorr`	Signal Processing Toolbox

Purpose

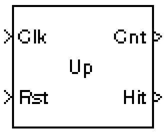
Count up or down through specified range of numbers

Library

Signal Management / Switches and Counters

`dspswit3`

Description



The Counter block increments or decrements an internal counter each time it receives a trigger event at the `Clk` port. A trigger event at the `Rst` port resets the counter to its initial state.

The input to the `Rst` port must be a real sample based scalar. The input to the `Clk` port can be a real sample-based scalar, or a real frame-based vector (that is, single channel). When both inputs are sample based, they must have the same sample period. When the `Clk` input is frame based, the frame period must equal the sample period of the `Rst` input.

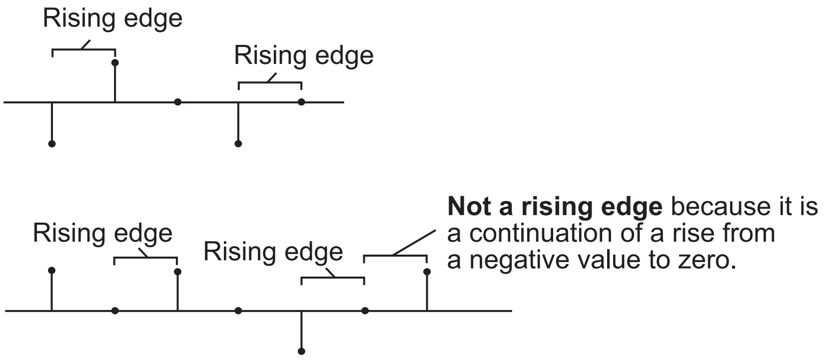
Sections of This Reference Page

- “Setting the Count Event Parameter” on page 2-203
- “Setting the Counter Size and Initial Count Parameters” on page 2-206
- “Sample-Based Operation” on page 2-206
- “Frame-Based Operation” on page 2-207
- “Free-Running Operation” on page 2-208
- “Examples” on page 2-208
- “Dialog Box” on page 2-211
- “Supported Data Types” on page 2-213
- “See Also” on page 2-214

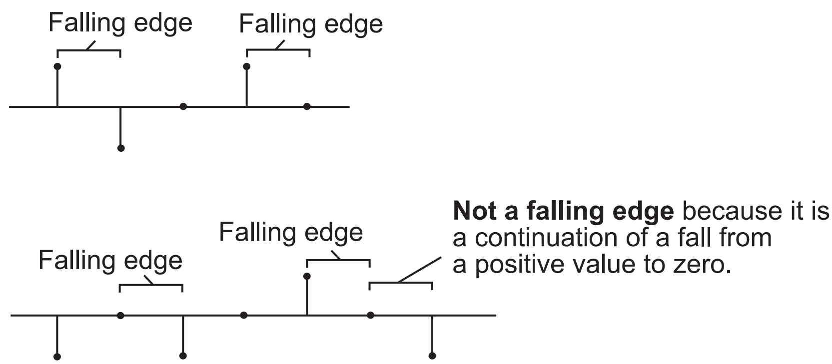
Setting the Count Event Parameter

The trigger event for both inputs is specified by the **Count event** parameter, and can be one of the following:

- `Rising edge` — Triggers a count or reset operation when the `Clk` or `Rst` input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers a count or reset operation when the `Clk` or `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)



- `Either edge` — Triggers a count or reset operation when the `Clk` or `Rst` input is a `Rising edge` or `Falling edge` (as described above).
- `Non-zero sample` — Triggers a count or reset operation at each sample time when the `Clk` or `Rst` input is not zero.
- `Free running` disables the `Clk` port, and enables the **Samples per output frame** and **Sample time** parameters. The block increments or decrements the counter at a constant interval, T_s, specified by the **Sample time** parameter (for more information, see "Free-Running Operation" on page 2-208). The `Rst` port behaves as if the **Count event** parameter were set to `Non-zero sample`.

Note When running simulations in the Simulink® `MultiTasking` mode, sample-based reset signals have a one-sample latency, and frame-based reset and clock signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a trigger event at the `Clk` or `Rst` port, and when it applies the trigger. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide* .

When running simulations in the Simulink `MultiTasking` mode, sample-based reset signals have a one-sample latency, and frame-based reset signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a reset event, and when it applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop User's Guide*.

Setting the Counter Size and Initial Count Parameters

At the start of the simulation, the block sets the counter to the value specified by the **Initial count** parameter, which can be any integer in the range defined by the **Counter size** parameter. The **Counter size** parameter allows you to choose from three standard counter ranges, or to specify an arbitrary counter limit:

- `8 bits` specifies a counter with a range of 0 to 255.
- `16 bits` specifies a counter with a range of 0 to 65535.
- `32 bits` specifies a counter with a range of 0 to 2^{32}-1.
- `User defined` enables the supplementary **Maximum count** parameter, which allows you to specify an arbitrary integer as the upper count limit. The range of the counter is then 0 to the **Maximum count** value.

Sample-Based Operation

The block operates in sample-based mode when the `Clk` input is a sample-based scalar. Sample-based vectors and matrices are not accepted.

When the **Count direction** parameter is set to `Up`, a sample-based trigger event at the `Clk` input causes the block to increment the counter by one. The block continues incrementing the counter when triggered until the counter value reaches the upper count limit (that is 255 for an 8-bit counter). At the next `Clk` trigger event, the block resets the counter to 0, and resumes incrementing the counter with the subsequent `Clk` trigger event.

When the **Count direction** parameter is set to `Down`, a sample-based trigger event at the `Clk` input causes the block to decrement the counter by one. The block continues decrementing the counter when triggered until the counter value reaches 0. At the next `Clk` trigger event, the block resets the counter to the upper count limit (that is 255 for an 8-bit counter), and resumes decrementing the counter with the subsequent `Clk` trigger event.

Between triggering events the block holds the output at its most recent value. The block resets the counter to its initial state when the trigger event specified in the **Count event** menu is received at the optional `Rst` input. When trigger events are received simultaneously at the `Clk` and `Rst` ports, the block first resets the counter, and then increments or decrements appropriately. (If you do not need to reset the counter during the simulation, you can disable the `Rst` port by clearing the **Reset input** check box.)

The **Output** pop-up menu provides three options for the output port configuration of the block icon:

- `Count` configures the block icon to show a `Cnt` port, which produces the current value of the counter as a sample-based scalar with the same sample period as the inputs.
- `Hit` configures the block icon to show a `Hit` port. The `Hit` port produces zeros while the value of the counter does not equal the integer **Hit value** parameter setting. When the counter value *does* equal the **Hit value** setting, the block generates a value of `1` at the `Hit` port. The output is sample based with the same sample period as the inputs.
- `Count and Hit` configures the block icon with both ports.

Frame-Based Operation

The block operates in frame-based mode when the `Clk` input is a frame-based vector (that is, single channel). Multichannel frame-based inputs are not accepted.

Frame-based operation is the same as sample-based operation, except that the block increments or decrements the counter by the total number of trigger events contained in the `Clk` input frame. A trigger event that is split across two consecutive frames is counted in the frame that contains the conclusion of the event. When a trigger event is received at the `Rst` port, the block first resets the counter, and then increments or decrements the counter by the number of trigger events contained in the `Clk` frame.

The `Cnt` and `Hit` outputs are sample-based scalars with sample period equal to the `Clk` input frame period.

Free-Running Operation

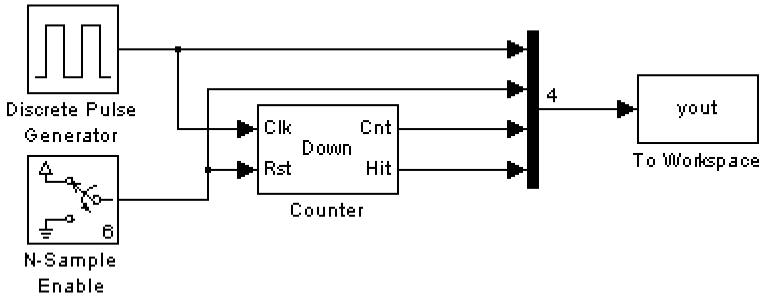
The block operates in free-running mode when you select `Free running` from the **Count event** menu.

The `Rst` port behaves as if the **Count event** parameter were set to `Non-zero sample` (triggers a reset at each sample time that the `Rst` input is not zero).

The `Clk` input port is disabled in this mode, and the block simply increments or decrements the counter using the constant sample period specified by the **Sample time** parameter, T_s. The `Cnt` output is a frame-based M-by-1 matrix containing the count value at each of M consecutive sample times, where M is specified by the **Samples per output frame** parameter. The `Hit` output is a frame-based M-by-1 matrix containing the hit status (`0` or `1`) at each of those M consecutive sample times. Both outputs have a frame period of $M*T_s$.

Examples

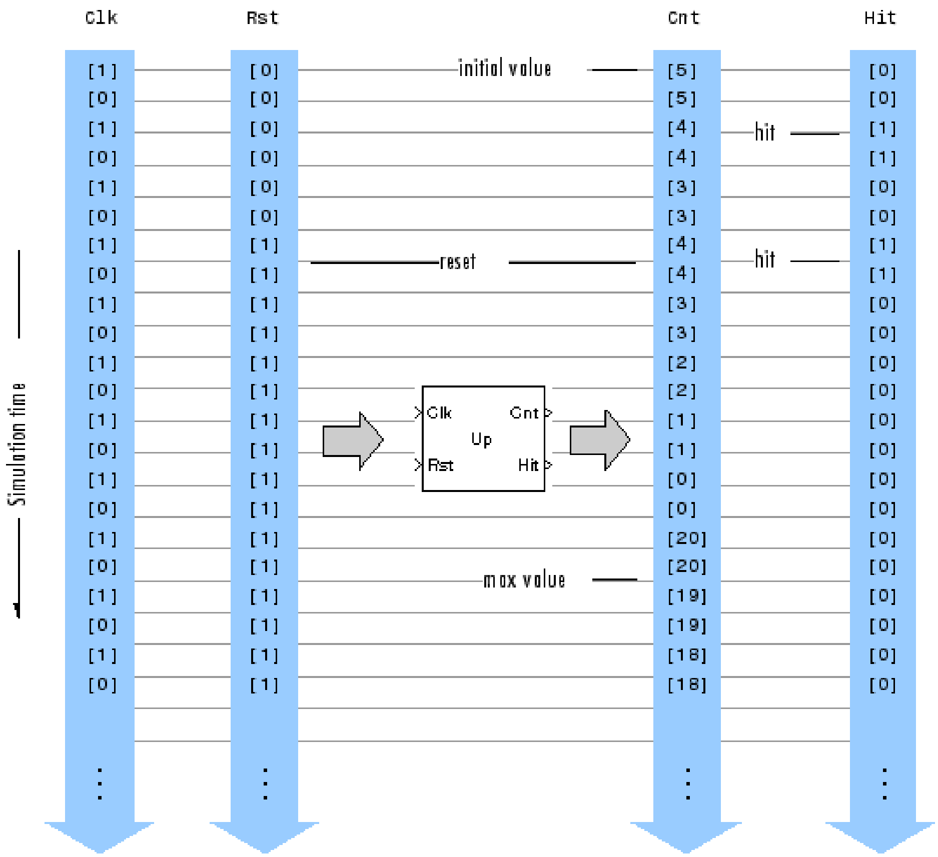
In the model below, the `Clk` port of the Counter block is driven by the Simulink Pulse Generator block, and the `Rst` port is triggered by an N-Sample Enable block. All of the Counter block's inputs and outputs are multiplexed into a single To Workspace block using a 4-port Mux block.



To run the model, first select Configuration Parameters from the **Simulation** menu. In the **Select** pane, click **Solver**, and set the **Stop time** to `30`. Then adjust the block parameters as described below. (Use the default settings for the Pulse Generator and To Workspace blocks.)

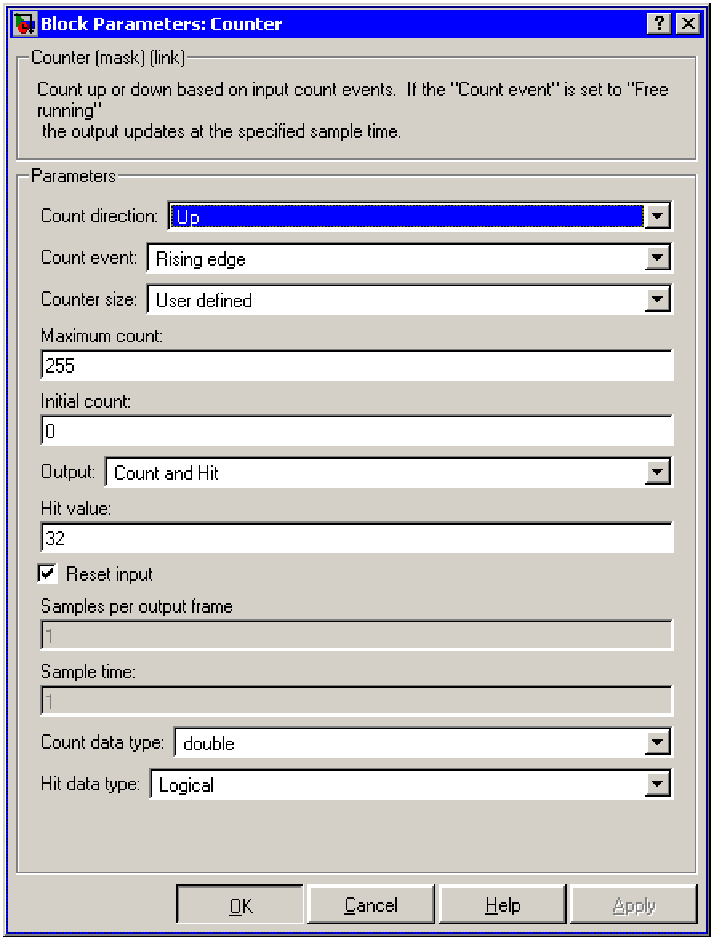
- Set the N-Sample Enable block parameters as follows:
  - **Trigger count** = `6`
  - **Active level** = `High (1)`
- Set the Counter block parameters as follows:
  - **Count direction:** `Down`
  - **Count event:** `Rising edge`
  - **Counter size:** `User defined`
  - **Maximum count:** `20`
  - **Initial count:** `5`
  - **Output:** `Count and Hit`
  - **Hit value:** `4`
  - **Reset input** ☑
  - **Count data type:** `Double`
  - **Hit data type:** `Logical`
- Set the **Number of inputs** parameter of the Mux block to `4`.

The figure below shows the first 22 samples of the model's four-column output, `yout`. The first column is the Counter block's `Clk` input, the second column is the block's `Rst` input, the third column is the block's `Cnt` output, and the fourth column is the block's `Hit` output.



You can see that the seventh input samples to both the `Clk` and `Rst` ports of the Counter block represent trigger events (rising edges), so at this time step the block first resets the counter to its initial value of 5, and then immediately decrements the count to 4. When the counter reaches its minimum value of 0, it rolls over to its maximum value of 20 with the following trigger event at the `Cnt` port.

Dialog Box



Count direction

The counter direction, `Up` or `Down`. Tunable, except in the Simulink external mode.

Count event
The type of event that triggers the block to increment, decrement, or reset the counter when received at the `Clk` or `Rst` ports. `Free running` disables the `Clk` port, and counts continuously with the period specified by the **Sample time** parameter. For more information on all the possible settings, see "Setting the Count Event Parameter" on page 2-203.

Counter size
The range of integer values the block should count through before recycling to zero. For more information, see "Setting the Counter Size and Initial Count Parameters" on page 2-206.

Maximum count
The counter's maximum value when **Counter size** is set to `User defined`. Tunable.

Initial count
The counter's initial value at the start of the simulation and after reset. Tunable, except in the Simulink external mode.

Output
Selects the output port(s) to enable: `Cnt`, `Hit`, or both.

Hit value
The scalar value whose occurrence in the count should be flagged by a `1` at the (optional) `Hit` output. This parameter is available when `Hit` or `Count and Hit` are selected in the **Output** menu. Tunable.

Reset input
Enables the `Rst` input port when selected.

Samples per output frame
The number of samples, M, in each output frame. This parameter is available when you select `Free running` in the **Count event** menu.

Sample time
The output sample period, T_s, in free-running mode. This parameter is available when you select `Free running` in the **Count event** menu.

Count data type
The data type of the output from the `Cnt` output port. This parameter is available when the **Output** parameter is set to `Count` or `Count and Hit`.

Hit data type
The data type of the output from the `Hit` output port. For information on the `Logical` and `Boolean` options of this parameter, see "Effects of Enabling and Disabling Boolean Support". This parameter is available when the **Output** parameter is set to `Hit` or the **Output** parameter is set to `Count and Hit` and the **Count data type** parameter is set to `Double`.

Supported Data Types

Port	Supported Data Types
Clk	• Double-precision floating point • Single-precision floating point • Boolean
Rst	• Double-precision floating point • Single-precision floating point • Boolean

Port	Supported Data Types
Cnt	• Double-precision floating point • Single-precision floating point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Hit	• Logical • Boolean — The block might output Boolean values from the `Hit` output port depending on the **Hit data type** parameter setting, as described in "Effects of Enabling and Disabling Boolean Support". To learn how to disable Boolean output support, see "Steps to Disabling Boolean Support".

See Also

Edge Detector	Signal Processing Blockset
N-Sample Enable	Signal Processing Blockset
N-Sample Switch	Signal Processing Blockset

Purpose

Compute estimate of autoregressive (AR) model parameters using covariance method

Library

Estimation / Parametric Estimation

`dspparest3`

Description

The Covariance AR Estimator block uses the covariance method to fit an autoregressive (AR) model to the input data. This method minimizes the forward prediction error in the least squares sense.

The input is a sample-based vector (row, column, or 1-D) or frame-based vector (column only) representing a frame of consecutive time samples from a single-channel signal, which is assumed to be the output of an AR system driven by white noise. The block computes the normalized estimate of the AR system parameters, $A(z)$, independently for each successive input frame.

$$H(z) = \frac{G}{A(z)} = \frac{G}{1 + a(2)z^{-1} + \ldots + a(p+1)z^{-p}}$$

The order, p, of the all-pole model is specified by the **Estimation order** parameter. To guarantee a valid output, you must set the **Estimation order** parameter to be less than or equal to half the input vector length.

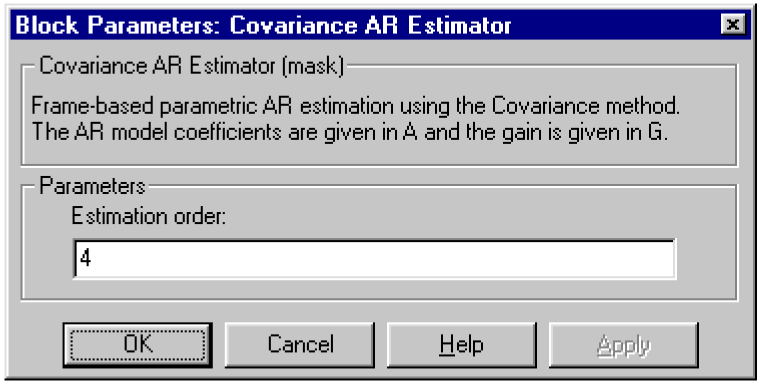
The top output, `A`, is a column vector of length p+1 with the same frame status as the input, and contains the normalized estimate of the AR model coefficients in descending powers of z.

```
[1 a(2) ... a(p+1)]
```

The scalar gain, G, is provided at the bottom output (`G`).

See the Burg AR Estimator block reference page for a comparison of the Burg AR Estimator, Covariance AR Estimator, Modified Covariance AR Estimator, and Yule-Walker AR Estimator blocks.

Dialog Box



Estimation order

The order of the AR model, p. To guarantee a nonsingular output, you must set p to be less than the input length. Otherwise, the output might be singular.

References

Kay, S. M. *Modern Spectral Estimation: Theory and Application.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

Marple, S. L., Jr., *Digital Spectral Analysis with Applications.* Englewood Cliffs, NJ: Prentice-Hall, 1987.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
A	• Double-precision floating point • Single-precision floating point
G	• Double-precision floating point • Single-precision floating point

See Also

Burg AR Estimator	Signal Processing Blockset
Covariance Method	Signal Processing Blockset
Modified Covariance AR Estimator	Signal Processing Blockset
Yule-Walker AR Estimator	Signal Processing Blockset
`arcov`	Signal Processing Toolbox

Purpose

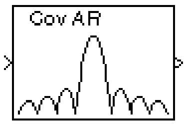
Compute parametric spectral estimate using covariance method

Library

Estimation / Power Spectrum Estimation

`dspspect3`

Description



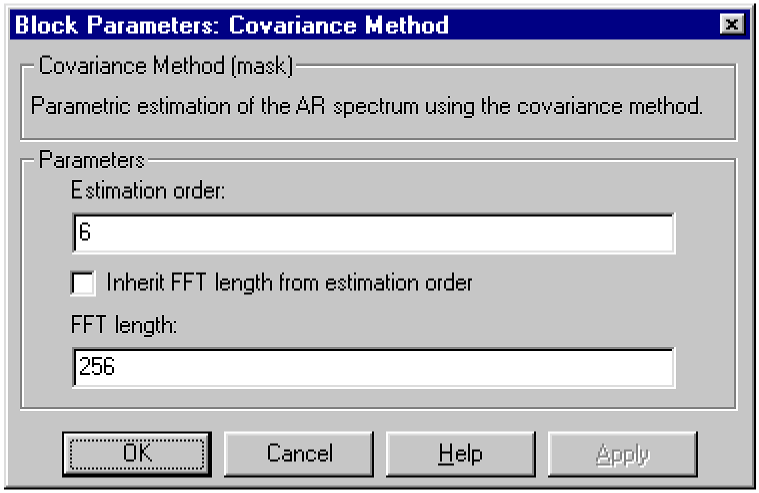
The Covariance Method block estimates the power spectral density (PSD) of the input using the covariance method. This method fits an autoregressive (AR) model to the signal by minimizing the forward prediction error in the least squares sense. The order of the all-pole model is the value specified by the **Estimation order** parameter, and the spectrum is computed from the FFT of the estimated AR model parameters. To guarantee a valid output, you must set the **Estimation order** parameter to be less than or equal to half the input vector length.

The input is a sample-based vector (row, column, or 1-D) or frame-based vector (column only) representing a frame of consecutive time samples from a single-channel signal. The block's output (a column vector) is the estimate of the signal's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where F_s is the signal's sample frequency.

When you select **Inherit FFT length from input dimensions**, N_{fft} is specified by the frame size of the input, which must be a power of 2. When you do *not* select **Inherit FFT length from input dimensions**, N_{fft} is specified as a power of 2 by the **FFT length** parameter, and the block zero pads or wraps the input to N_{fft} before computing the FFT. The output is always sample based.

See the Burg Method block reference for a comparison of the Burg Method, Covariance Method, Modified Covariance Method, and Yule-Walker Method blocks.

Dialog Box



Estimation order
The order of the AR model. To guarantee a nonsingular output, you must set the value of this parameter to be less than the input length. Otherwise, the output might be singular.

Inherit FFT length from input dimensions
When selected, uses the input frame size as the number of data points, N_{fft}, on which to perform the FFT. Tunable.

FFT length
Enter the number of data points on which to perform the FFT, N_{fft}. When N_{fft} is larger than the input frame size, each frame is zero-padded as needed. When N_{fft} is smaller than the input frame size, each frame is wrapped as needed. This parameter is enabled when you clear the **Inherit FFT length from input dimensions** check box.

References

Kay, S. M. *Modern Spectral Estimation: Theory and Application.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

Marple, S. L., Jr., *Digital Spectral Analysis with Applications.* Englewood Cliffs, NJ: Prentice-Hall, 1987.

Orfanidis, S. J. *Introduction to Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1995.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
Output	• Double-precision floating point • Single-precision floating point

See Also

Burg Method	Signal Processing Blockset
Covariance AR Estimator	Signal Processing Blockset
Modified Covariance Method	Signal Processing Blockset
Short-Time FFT	Signal Processing Blockset
Yule-Walker Method	Signal Processing Blockset
`pcov`	Signal Processing Toolbox

See “Power Spectrum Estimation” for related information.

Purpose

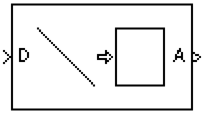
Create square diagonal matrix from diagonal elements

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description

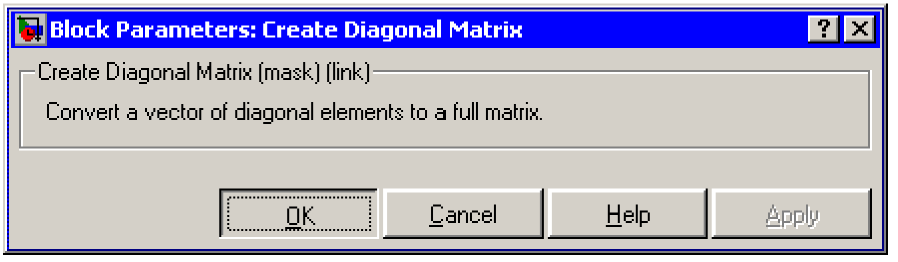


The Create Diagonal Matrix block populates the diagonal of the *M*-by-*M* matrix output with the elements contained in the length-*M* vector input, D. The elements off the diagonal are zero.

```
A = diag(D)     Equivalent MATLAB code
```

The output is always sample based.

Dialog Box



Supported Data Types

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

See Also

Constant Diagonal Matrix	Signal Processing Blockset
Extract Diagonal	Signal Processing Blockset
`diag`	MATLAB

Purpose

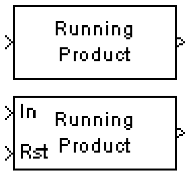
Compute cumulative product of channel, column, or row elements

Library

Math Functions / Math Operations

`dspmathops`

Description



The Cumulative Product block computes the cumulative product of elements in each channel, column, or row of the M-by-N input matrix.

The inputs can be sample-based or frame-based vectors and matrices. The output always has the same dimensions, rate, frame status, data type, and complexity as the input.

The Cumulative Product block accepts real and complex fixed-point and floating-point inputs except for complex unsigned fixed-point inputs.

- "Valid Input" on page 2-223
- "Valid Reset Signal" on page 2-224
- "Output Characteristics" on page 2-224
- "Multiplying Along Channels of Frame-Based Inputs" on page 2-224
- "Multiplying Along Channels of Sample-Based Inputs" on page 2-225
- "Resetting the Cumulative Product Along Channels" on page 2-226
- "Multiplying Along Columns" on page 2-228
- "Multiplying Along Rows" on page 2-229
- "Dialog Box" on page 2-231
- "Supported Data Types" on page 2-236
- "See Also" on page 2-236

Valid Input

The block computes the cumulative product of both sample- and frame-based vector and matrix inputs. Inputs can be real or complex. When multiplying along channels or columns, 1-D unoriented vectors are treated as column vectors. When multiplying along rows, 1-D vectors are treated as row vectors.

Valid Reset Signal

The optional reset port, `Rst`, accepts scalar values, which can be any built-in Simulink® data type including `boolean`. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input.

Output Characteristics

The output always has the same dimensions, rate, frame status, data type, and complexity as the data signal input.

Multiplying Along Channels of Frame-Based Inputs

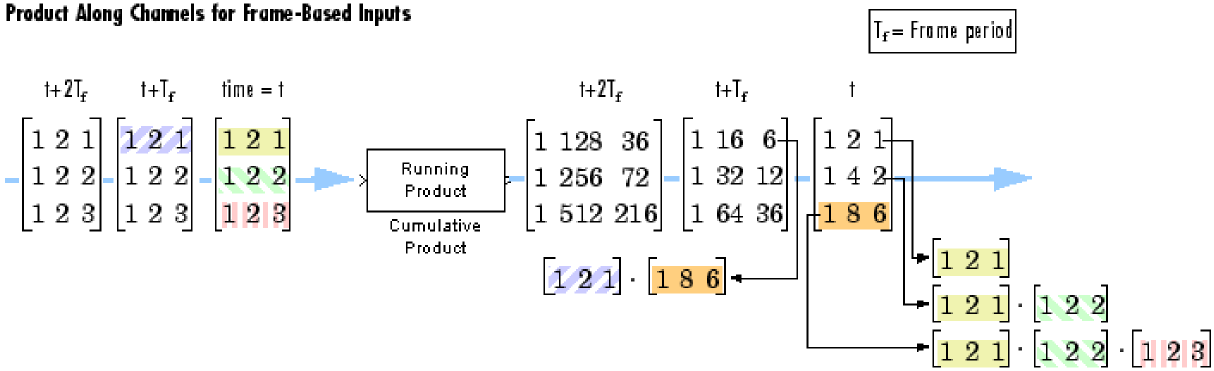
For frame-based inputs, the block treats each input column as an independent channel. As the following figure and equation illustrate, the output has the following characteristics:

- The first row of the first output is the same as the first row of the first input.
- The first row of each subsequent output is the element-wise product of the first row of the current input (time t), and the last row of the previous output (time t - T_f, where T_f is the frame period).
- The output has the same size, dimension, frame status, data type, and complexity as the input.

Given an M-by-N frame-based input, u, the output, y, is a frame-based M-by-N matrix whose first row has elements

$$y_{1,j}(t) = u_{1,j}(t) \cdot y_{M,j}\left(t - T_f\right)$$

Product Along Channels for Frame-Based Inputs



Multiplying Along Channels of Sample-Based Inputs

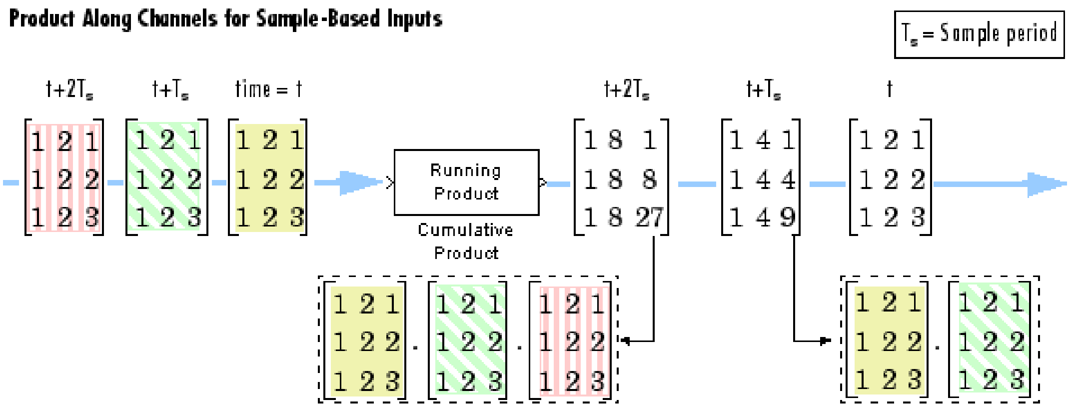
For sample-based inputs, the block treats each element of the input matrix as an independent channel. As the following figure and equation illustrate, the output has the following characteristics:

- The first output is the same as the first input.
- Each subsequent output is the element-wise product of the current input (time t) and the previous output (time t - T_s, where T_s is the sample period).
- The output has the same size, dimension, frame status, data type, and complexity as the input.

Given an M-by-N sample-based input, u, the output, y, is a sample-based M-by-N matrix with the elements

$$y_{i,j}(t) = u_{i,j}(t) \cdot y_{i,j}(t - T_s) \quad \begin{array}{l} 1 \le i \le M \\ 1 \le j \le N \end{array}$$

For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors when multiplying along channels, and the output is a length-M 1-D vector.



Resetting the Cumulative Product Along Channels

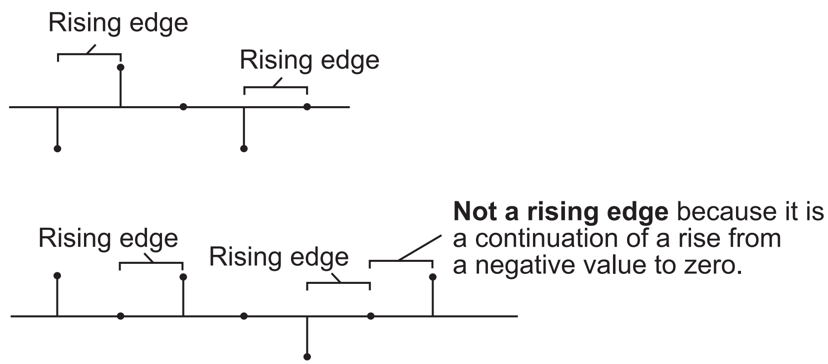
When you set the **Multiply input along** parameter to `Channels (running product)`, you can set the block to reset the running product whenever it detects a reset event at the optional `Rst` port. The reset sample time must be a positive integer multiple of the input sample time. The input to the `Rst` port can be of the Boolean data type.

When the block is reset for sample-based inputs, the block initializes the current output to the values of the current input. For frame-based inputs, the block initializes the first row of the current output to the values in the first row of the current input.

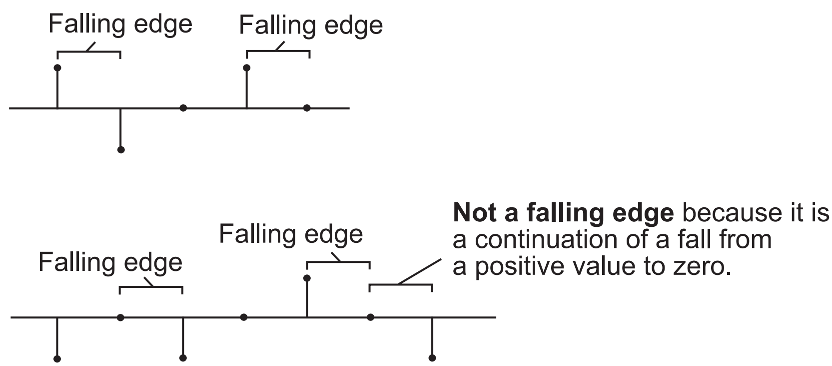
The **Reset port** parameter specifies the reset event, which can be one of the following:

- `None` disables the `Rst` port.
- `Rising edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Rises from a negative value to a positive value or zero

- Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)



- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described above)
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero

---

Note When running simulations in the Simulink `MultiTasking` mode, reset signals have a one-sample latency. Therefore, when the block detects a reset event, there is a one-sample delay at the reset port rate before the block applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Multiplying Along Columns

When the **Multiply input along** parameter is set to `Columns`, the block computes the cumulative product of each column of the input, where the current cumulative product is independent of the cumulative products of previous inputs.

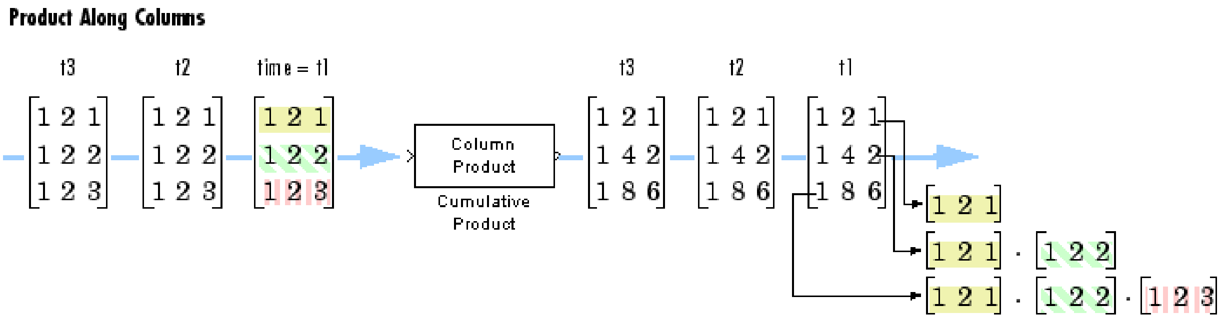
```
y = cumprod(u)      % Equivalent MATLAB code
```

The output has the same size, dimension, frame status, data type, and complexity as the input. The mth output row is the element-wise product of the first m input rows.

Given an M-by-N input, u, the output, y, is an M-by-N matrix whose jth column has elements

$$y_{i,j} = \prod_{k=1}^{i} u_{k,j} \qquad 1 \le i \le M$$

The block treats length-M 1-D vector inputs as M-by-1 column vectors when multiplying along columns.



Multiplying Along Rows

When the **Multiply input along** parameter is set to `Rows`, the block computes the cumulative product of the row elements, where the current cumulative product is independent of the cumulative products of previous inputs.

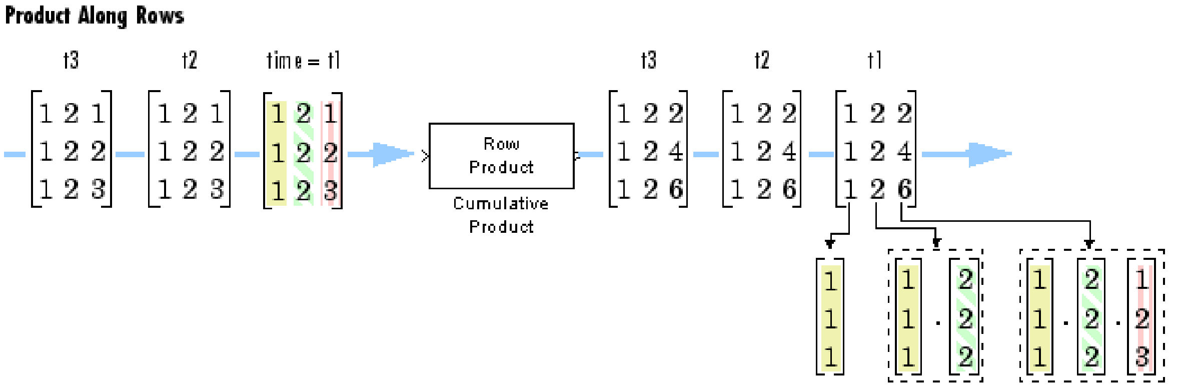
```
y = cumprod(u,2)      % Equivalent MATLAB code
```

The output has the same size, dimension, frame status, and data type as the input. The nth output column is the element-wise product of the first n input columns.

Given an M-by-N input, u, the output, y, is an M-by-N matrix whose ith row has elements

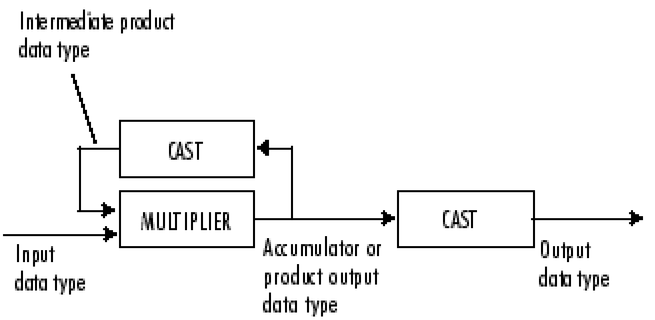
$$y_{i,j} = \prod_{k=1}^{j} u_{i,k} \qquad 1 \le j \le N$$

The block treats length-N 1-D vector inputs as 1-by-N row vectors when multiplying along rows.



Fixed-Point Data Types

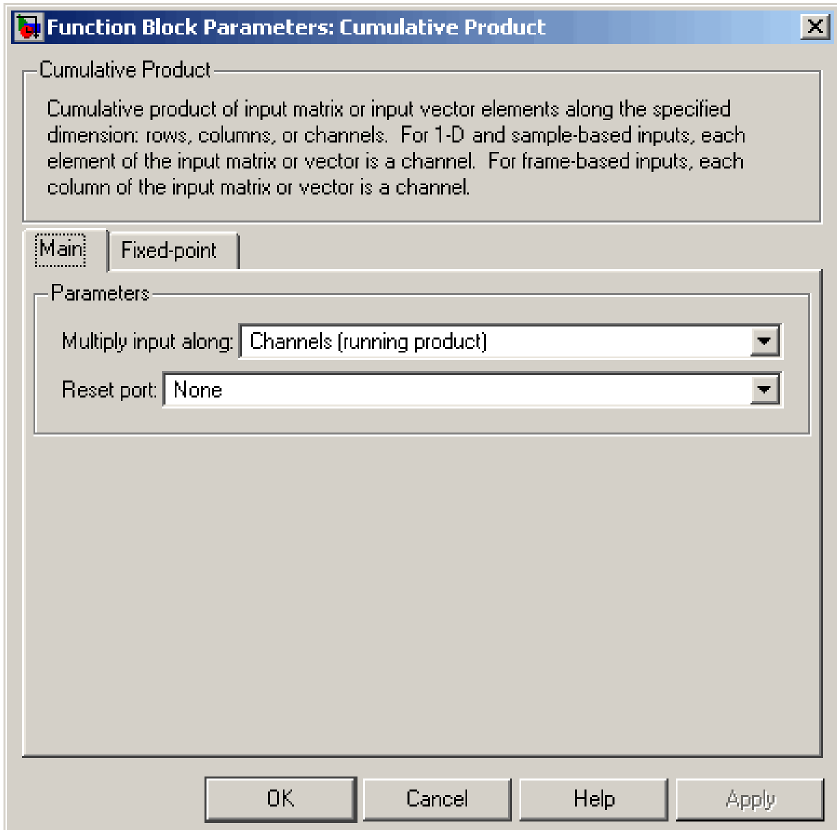
The following diagram shows the data types used within the Cumulative Product block for fixed-point signals.



The output of the multiplier is in the product output data type when at least one of the inputs to the multiplier is real. When both of the inputs to the multiplier are complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see "Multiplication Data Types". You can set the accumulator, product output, intermediate product, and output data types in the block dialog as discussed in "Dialog Box" on page 2-231.

Dialog Box

The **Main** pane of the Cumulative Product block dialog appears as follows.



Multiply input along

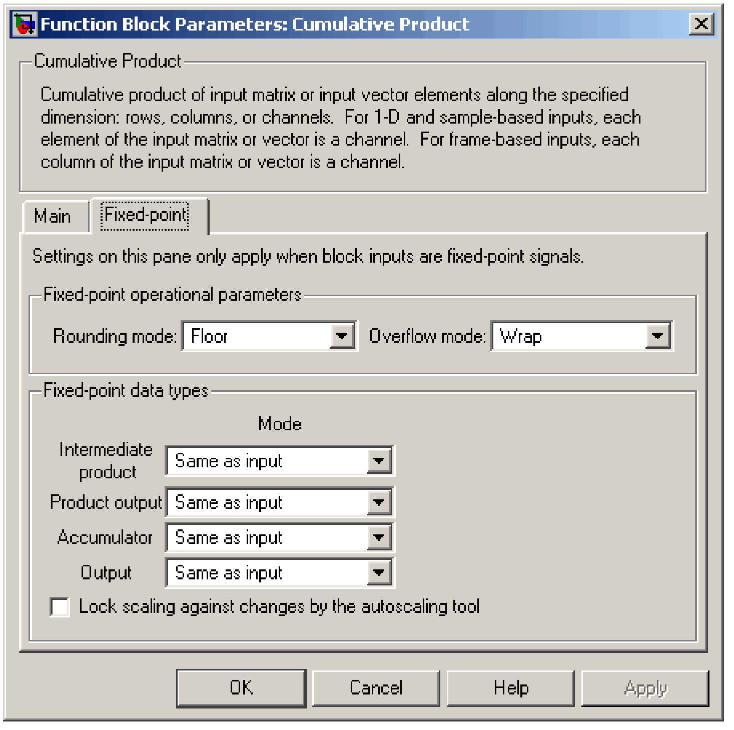
The dimension along which to compute the cumulative products. The options allow you to multiply along `Channels (running product)`, `Columns`, and `Rows`. For more information, see the following sections:

- “Multiplying Along Channels of Frame-Based Inputs” on page 2-224
- “Multiplying Along Channels of Sample-Based Inputs” on page 2-225
- “Multiplying Along Columns” on page 2-228
- “Multiplying Along Rows” on page 2-229

Reset port

Determines the reset event that causes the block to reset the product along channels. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input. This parameter is enabled only when you set the **Multiply input along** parameter to `Channels (running product)`. For more information, see “Resetting the Cumulative Product Along Channels” on page 2-226.

The **Fixed-point** pane of the Cumulative Product block dialog appears as follows.



Rounding mode
: Select the rounding mode for fixed-point operations.

Overflow mode
: Select the overflow mode for fixed-point operations.

Intermediate product
: As shown in “Fixed-Point Data Types” on page 2-230, the output of the multiplier is cast to the intermediate product data type before the next element of the input is multiplied into it. Use

this parameter to specify how you would like to designate the intermediate product word and fraction lengths:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the intermediate product, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the intermediate product. This block requires power-of-two slope and a bias of zero.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-230 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-230 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block. Note that the accumulator data type is only used when both inputs to the multiplier are complex:

- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output
Choose how you specify the word length and fraction length of the output of the block:

- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

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

Supported Data Types

Input and Output Ports	Supported Data Types
Data input port, `In`	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Reset input port, `Rst`	All built-in Simulink data types: • Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Output port	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers

See Also

Cumulative Sum	Signal Processing Blockset
Matrix Product	Signal Processing Blockset
`cumprod`	MATLAB

Purpose

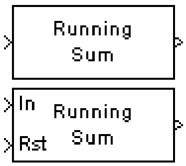
Compute cumulative sum of channel, column, or row elements

Library

Math Functions / Math Operations

`dspmathops`

Description



The Cumulative Sum block computes the cumulative sum of the elements in each channel, column, or row of the M-by-N input matrix.

The inputs can be sample-based or frame-based vectors and matrices. The output always has the same dimensions, rate, frame status, data type, and complexity as the input.

The Cumulative Sum block accepts real and complex fixed-point and floating-point inputs except for complex unsigned fixed-point inputs.

Sections of This Reference Page

- "Input and Output Characteristics" on page 2-237
- "Summing Along Channels" on page 2-238
- "Resetting the Cumulative Sum Along Channels" on page 2-240
- "Summing Along Columns" on page 2-242
- "Summing Along Rows" on page 2-243
- "Dialog Box" on page 2-245
- "Supported Data Types" on page 2-248
- "See Also" on page 2-248

Input and Output Characteristics

Valid Input

The block computes the cumulative sum of both sample- and frame-based vector and matrix inputs. Inputs can be real or complex. When summing along channels or columns, 1-D unoriented vectors are treated as column vectors. When summing along rows, 1-D vectors are treated as row vectors.

Valid Reset Signal

The optional reset port, `Rst`, accepts scalar values, which can be any built-in Simulink® data type including `boolean`. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input.

Output Characteristics

The output always has the same dimensions, rate, frame status, data type, and complexity as the data signal input.

Summing Along Channels

When the **Sum input along** parameter is set to `Channels (running sum)`, the block computes the cumulative sum of the elements in each input channel. The running sum of the current input takes into account the running sum of all previous inputs. See the following sections for more information:

- "Summing Along Channels of Frame-Based Inputs" on page 2-238
- "Summing Along Channels of Sample-Based Inputs" on page 2-239
- "Resetting the Cumulative Sum Along Channels" on page 2-240

Summing Along Channels of Frame-Based Inputs

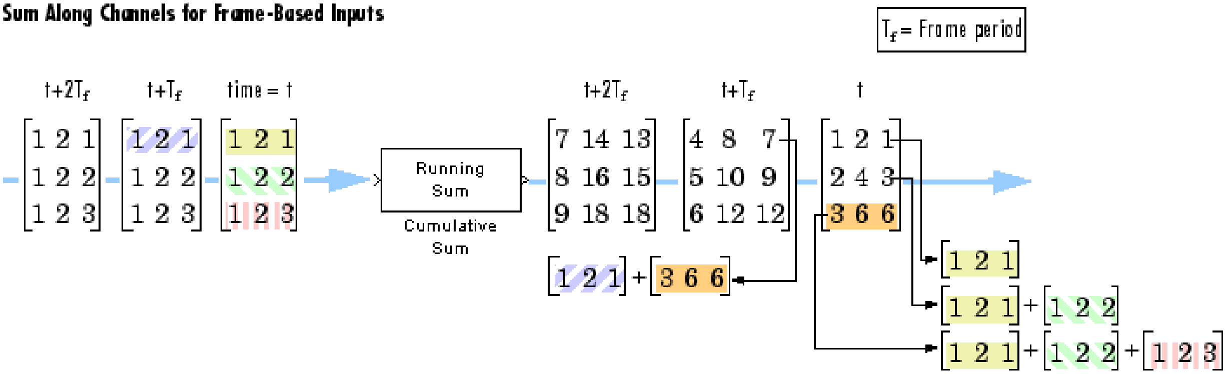
For frame-based inputs, the block treats each input column as an independent channel. As the following figure and equation illustrate, the output has the following characteristics:

- The first row of the first output is the same as the first row of the first input.
- The first row of each subsequent output is the sum of the first row of the current input (time t), and the last row of the previous output (time t - T_f, where T_f is the frame period).
- The output has the same size, dimension, frame status, data type, and complexity as the input.

Given an M-by-N frame-based input, u, the output, y, is a frame-based M-by-N matrix whose first row has elements

$$y_{1,j}(t) = u_{1,j}(t) + y_{M,j}(t - T_f)$$

Sum Along Channels for Frame-Based Inputs



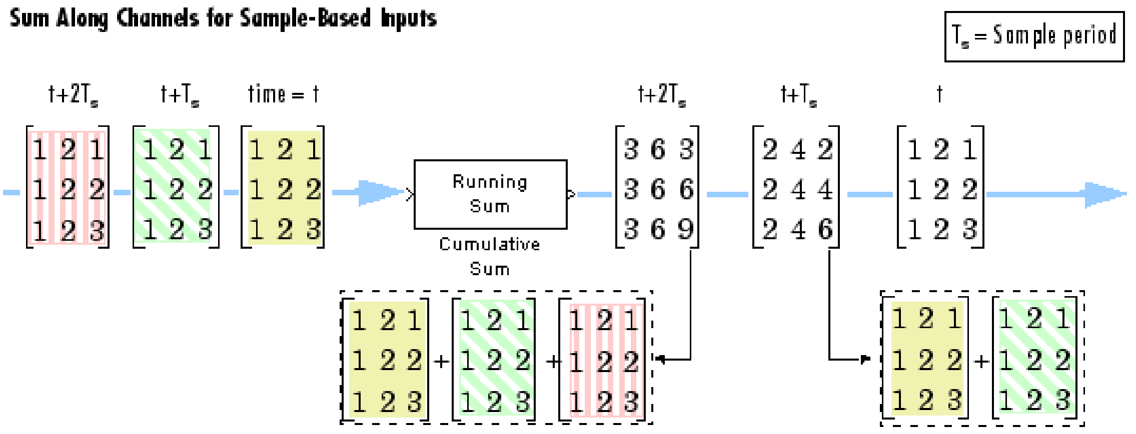
Summing Along Channels of Sample-Based Inputs

For sample-based inputs, the block treats each element of the input matrix as an independent channel. As the following figure and equation illustrate, the output has the following characteristics:

- The first output is the same as the first input.
- Each subsequent output is the sum of the current input (time t) and the previous output (time t - T_s, where T_s is the sample period).
- The output has the same size, dimension, frame status, data type, and complexity as the input.

Given an M-by-N sample-based input, u, the output, y, is a sample-based M-by-N matrix with the elements

$$y_{i,j}(t) = u_{i,j}(t) + y_{i,j}(t - T_s) \qquad \begin{array}{l} 1 \le i \le M \\ 1 \le j \le N \end{array}$$



Resetting the Cumulative Sum Along Channels

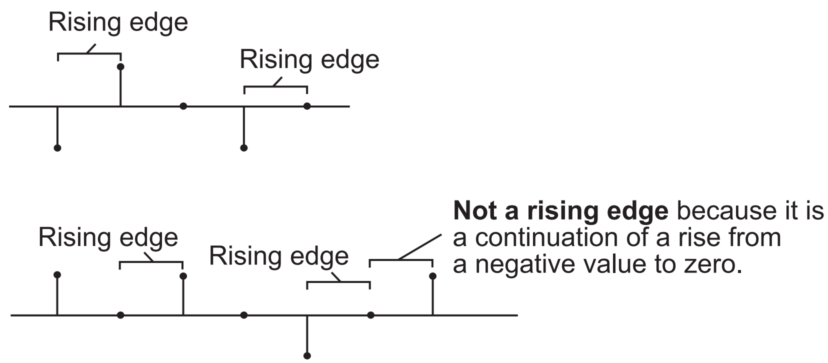
When you set the **Sum input along** parameter to `Channels (running sum)`, you can set the block to reset the running sum whenever it detects a reset event at the optional `Rst` port. The reset sample time must be a positive integer multiple of the input sample time. The input to the `Rst` port can be of the `boolean` data type.

When the block is reset for sample-based inputs, the block initializes the current output to the values of the current input. For frame-based inputs, the block initializes the first row of the current output to the values in the first row of the current input.

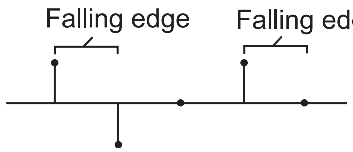
The **Reset port** parameter specifies the reset event, which can be one of the following:

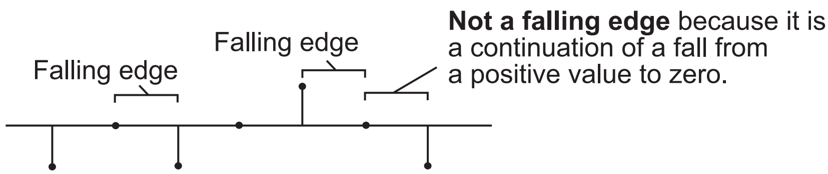
- `None` disables the `Rst` port.

- `Rising edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)





- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described above)
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero

---

Note When running simulations in the Simulink `MultiTasking` mode, reset signals have a one-sample latency. Therefore, when the block detects a reset event, there is a one-sample delay at the reset port rate before the block applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and the topic onmodels with multiple sample rates in the Real-Time Workshop® documentation.

Summing Along Columns

When the **Sum input along** parameter is set to `Columns`, the block computes the cumulative sum of each column of the input, where the current cumulative sum is independent of the cumulative sums of previous inputs.

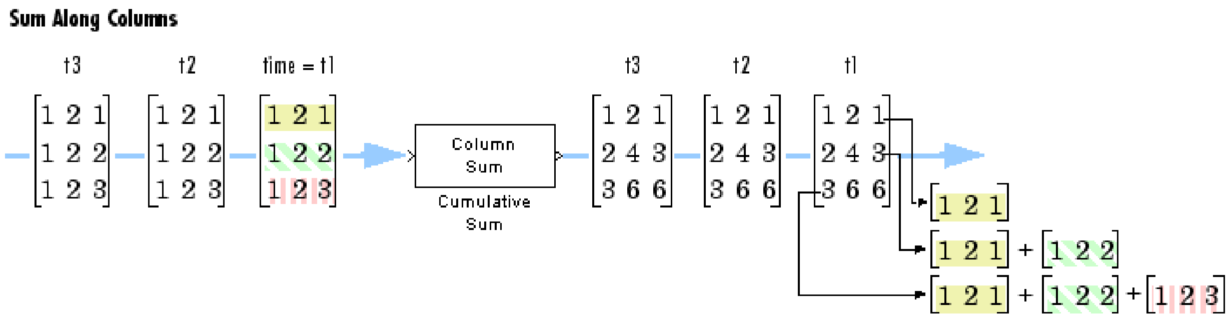
```
y = cumsum(u)      % Equivalent MATLAB code
```

The output has the same size, dimension, frame status, data type, and complexity as the input. The mth output row is the sum of the first m input rows.

Given an M-by-N input, u, the output, y, is an M-by-N matrix whose jth column has elements

$$y_{i,j} = \sum_{k=1}^{j} u_{k,j} \qquad 1 \le i \le M$$

The block treats length-M 1-D vector inputs as M-by-1 column vectors when summing along columns.



Summing Along Rows

When the **Sum input along** parameter is set to `Rows`, the block computes the cumulative sum of the row elements, where the current cumulative sum is independent of the cumulative sums of previous inputs.

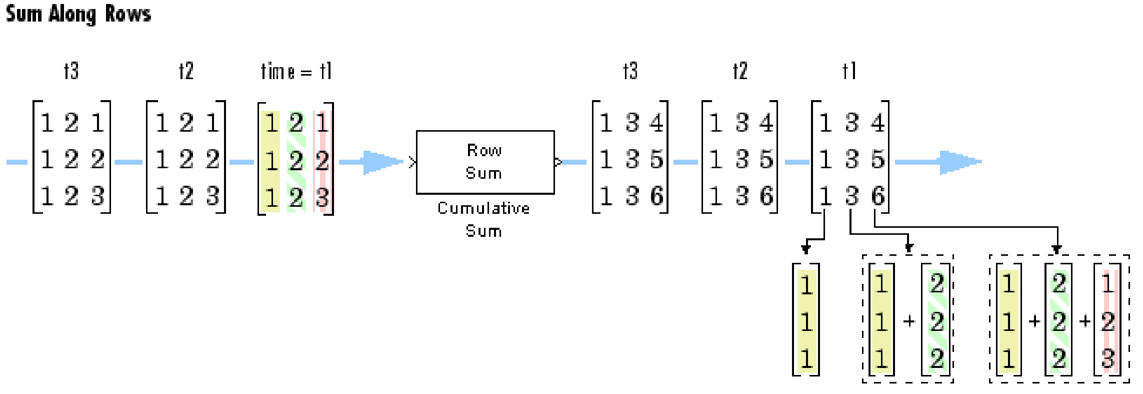
```
y = cumsum(u,2)      % Equivalent MATLAB code
```

The output has the same size, dimension, frame status, and data type as the input. The nth output column is the sum of the first n input columns.

Given an M-by-N input, u, the output, y, is an M-by-N matrix whose ith row has elements

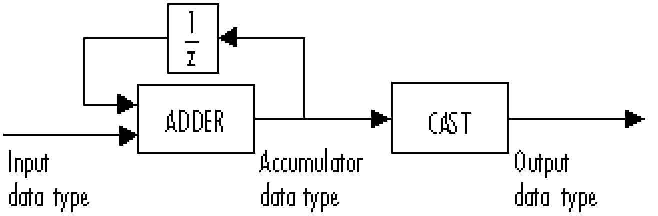
$$y_{i,j} = \sum_{k=1}^{j} u_{i,k} \qquad 1 \le j \le N$$

The block treats length-N 1-D vector inputs as 1-by-N row vectors when summing along rows.



Fixed-Point Data Types

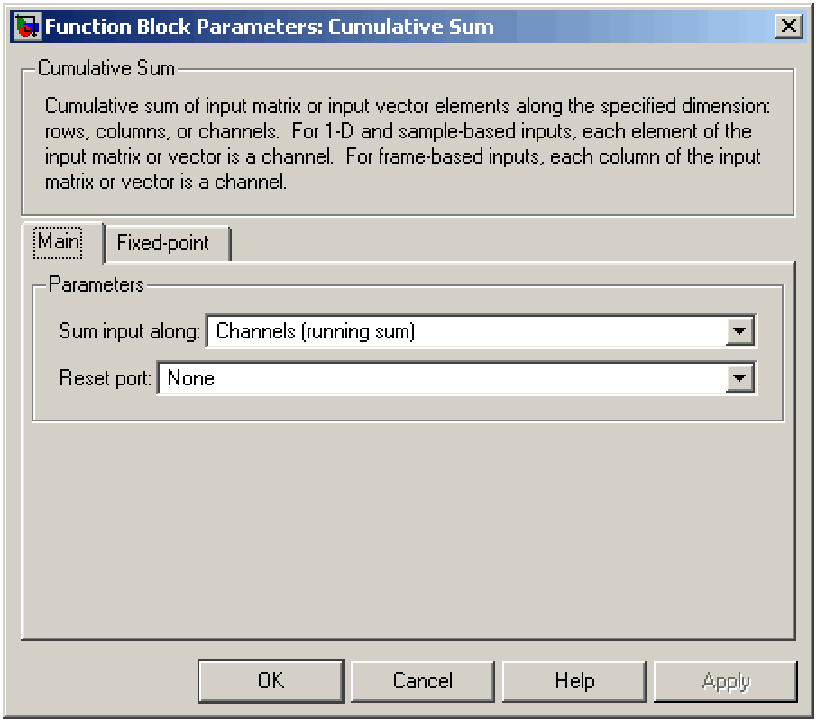
The following diagram shows the data types used within the Cumulative Sum block for fixed-point signals.



You can set the accumulator and output data types in the block dialog as discussed in "Dialog Box" on page 2-245.

Dialog Box

The **Main** pane of the Cumulative Sum block dialog appears as follows.



Sum input along

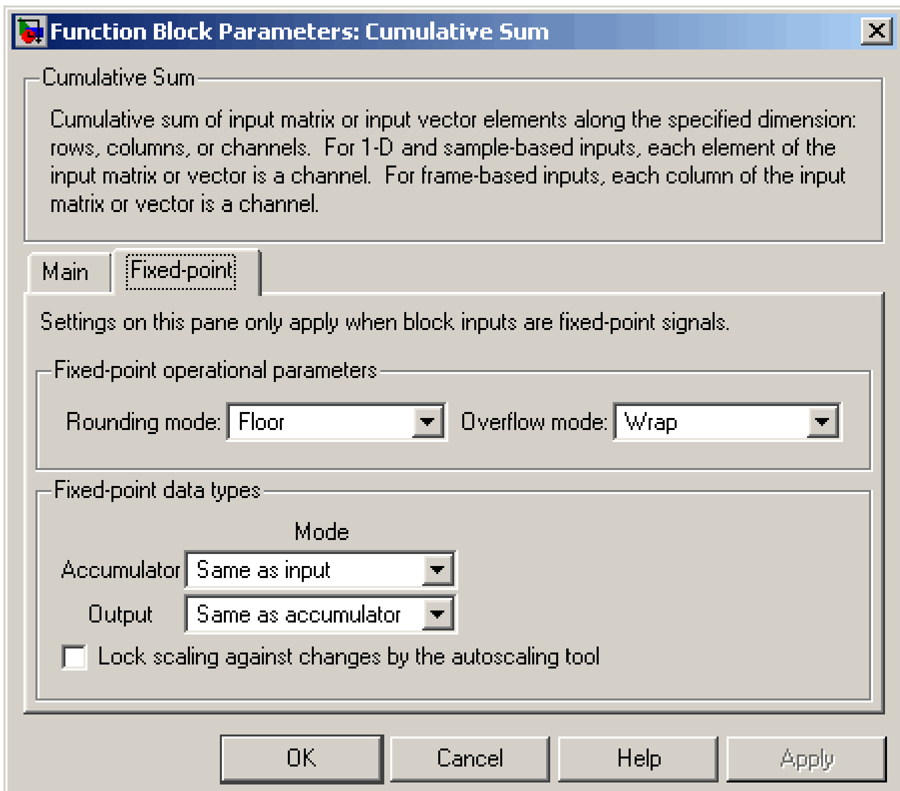
The dimension along which to compute the cumulative summations. The options allow you to sum along `Channels (running sum)`, `Columns`, and `Rows`. For more information, see the following sections:

- "Summing Along Channels" on page 2-238
- "Summing Along Columns" on page 2-242
- "Summing Along Rows" on page 2-243

Reset port

Determines the reset event that causes the block to reset the sum along channels. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input. This parameter is enabled only when you set the **Sum input along** parameter to `Channels (running sum)`. For more information, see "Resetting the Cumulative Sum Along Channels" on page 2-240.

The **Fixed-point** pane of the Cumulative Sum block dialog appears as follows.



Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Accumulator
Use this parameter to specify how you would like to designate this accumulator word and fraction lengths:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output
Choose how you specify the output word length and fraction length:

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

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

Supported Data Types

Input and Output Ports	Supported Data Types
Data input port, `In`	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Reset input port, `Rst`	All built-in Simulink data types: • Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Output port	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers

See Also

Cumulative Product	Signal Processing Blockset
Difference	Signal Processing Blockset
Matrix Sum	Signal Processing Blockset
`cumsum`	MATLAB

Purpose

Convert input signal to specified data type

Library

Signal Management / Signal Attributes

`dspsigattribs`

Description

Refer to the Simulink® Data Type Conversion reference page for more information.

dB Conversion

Purpose

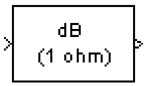
Convert magnitude data to decibels (dB or dBm)

Library

Math Functions / Math Operations

`dspmathops`

Description



The dB Conversion block converts a linearly scaled power or amplitude input to dB or dBm. The reference power is 1 Watt for conversions to dB and 1 mWatt for conversions to dBm. The **Input signal** parameter specifies whether the input is a power signal or a voltage signal, and the **Convert to** parameter controls the scaling of the output. When selected, the **Add eps to input to protect against "log(0) = -inf"** parameter adds a value of `eps` to all power and voltage inputs. When this option is not enabled, zero-valued inputs produce `-inf` at the output.

The size and frame status of the output are the same as the input.

Power Inputs

Select `Power` as the **Input signal** parameter when the input, `u`, is a real, nonnegative, power signal (units of watts). When the **Convert to** parameter is set to `dB`, the block performs the dB conversion

```
y = 10*log10(u)       % Equivalent MATLAB code
```

When the **Convert to** parameter is set to `dBm`, the block performs the dBm conversion

```
y = 10*log10(u) + 30
```

The dBm conversion is equivalent to performing the dB operation *after* converting the input to milliwatts.

Voltage Inputs

Select `Amplitude` as the **Input signal** parameter when the input, `u`, is a real voltage signal (units of volts). The block uses the scale factor specified in ohms by the **Load resistance** parameter, `R`, to convert the voltage input to units of power (watts) before converting to dB or dBm.

When the **Convert to** parameter is set to `dB`, the block performs the dB conversion

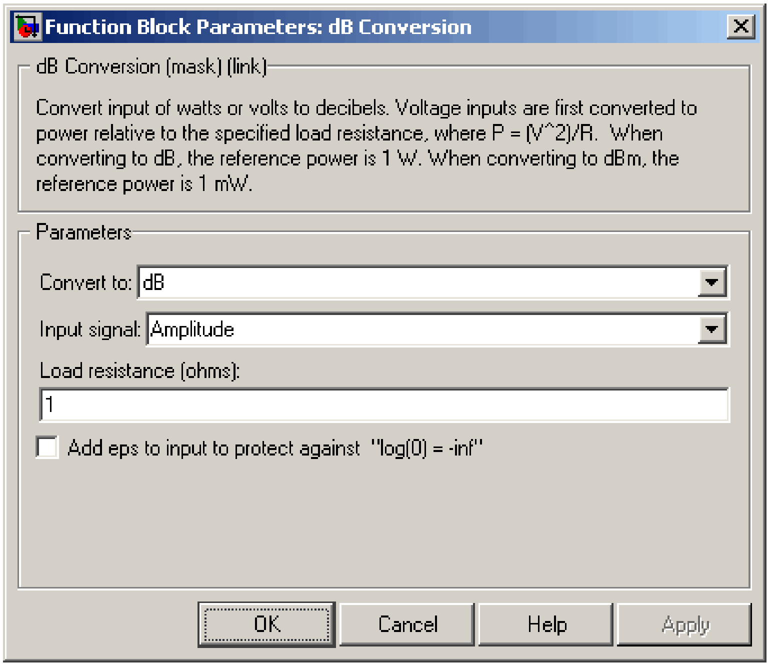
```
y = 10*log10(abs(u)^2/R)
```

When the **Convert to** parameter is set to `dBm`, the block performs the dBm conversion

```
y = 10*log10(abs(u)^2/R) + 30
```

The dBm conversion is equivalent to performing the dB operation *after* converting the `(abs(u)^2/R)` result to milliwatts.

Dialog Box



Convert to

The logarithmic scaling to which the input is converted, `dB` or `dBm`. The reference power is 1 W for conversions to dB and 1 mW for conversions to dBm. Tunable.

Input signal

The type of input signal, `Power` or `Amplitude`.

Load resistance

The scale factor used to convert voltage inputs to units of power. Tunable.

Add eps to input to protect against "log(0) = -inf"

When selected, adds `eps` to all input values (power or voltage). Tunable.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

dB Gain	Signal Processing Blockset
Math Function	Simulink
`log10`	MATLAB

Purpose

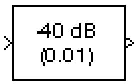
Apply decibel gain

Library

Math Functions / Math Operations

`dspmathops`

Description



The dB Gain block multiplies the input by the decibel values specified in the **Gain** parameter. For an M-by-N input matrix u with elements u_{ij}, the **Gain** parameter can be a real M-by-N matrix with elements g_{ij} to be multiplied element-wise with the input, or a real scalar.

$$y_{ij} = 10u_{ij}^{(g_{ij}/k)}$$

The value of k is 10 for power signals (select `Power` as the **Input signal** parameter) and 20 for voltage signals (select `Amplitude` as the **Input signal** parameter).

The value of the equivalent linear gain

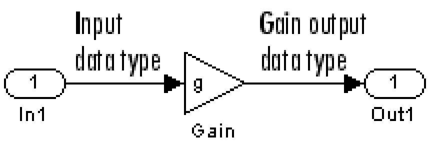
$$g_{ij}^{lin} = 10^{(g_{ij}/k)}$$

is displayed in the block icon below the dB gain value. The size and frame status of the output are the same as the input.

The dB Gain block supports real and complex floating-point and fixed-point data types.

Fixed-Point Data Types

The following diagram shows the data types used within the dB Gain subsystem block for fixed-point signals.

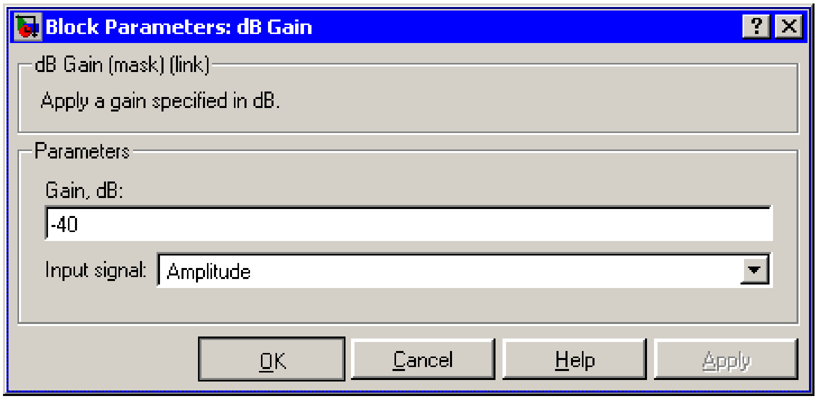


The settings for the fixed-point parameters of the Gain block in the diagram above are as follows:

- Round integer calculations toward: `Floor`
- Saturate on integer overflow — unselected
- Parameter data type mode — `Inherit via internal rule`
- Output data type mode — `Inherit via internal rule`

See the Gain reference page for more information.

Dialog Box



Gain

The dB gain to apply to the input, a scalar or a real M-by-N matrix. Tunable.

Input signal

The type of input signal: `Power` or `Amplitude`. Tunable.

Note This block does not support tunability in generated code.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)

- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

dB Conversion	Signal Processing Blockset
Math Function	Simulink
`log10`	MATLAB

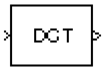
DCT

Purpose Compute discrete cosine transform (DCT) of input

Library Transforms

`dspxfrm3`

Description



The DCT block computes the unitary discrete cosine transform (DCT) of each channel in the M-by-N input matrix, u.

```
y = dct(u)     % Equivalent MATLAB code
```

For both sample-based and frame-based inputs, the block assumes that each input column is a frame containing M consecutive samples from an independent channel. The frame size, M, must be a power of two. To work with other frame sizes, use the Pad block to pad or truncate the frame size to a power-of-two length.

The output is an M-by-N matrix whose lth column contains the length-M DCT of the corresponding input column.

$$y(k,l) = w(k) \sum_{m=1}^{M} u(m,l) \cos \frac{\pi(2m-1)(k-1)}{2M}, \quad k = 1,\ldots,M$$

where

$$w(k) = \begin{cases} \dfrac{1}{\sqrt{M}}, & k = 1 \\ \sqrt{\dfrac{2}{M}}, & 2 \le k \le M \end{cases}$$

The output is always sample based, and the output port rate and data type (real/complex) are the same as those of the input port.

For convenience, length-M 1-D vector inputs and *sample-based* length-M row vector inputs are processed as single channels (that is, as M-by-1 column vectors), and the output has the same dimension as the input.

The **Sine and cosine computation** parameter determines how the block computes the necessary sine and cosine values. This parameter has two settings, each with its advantages and disadvantages, as described in the following table.

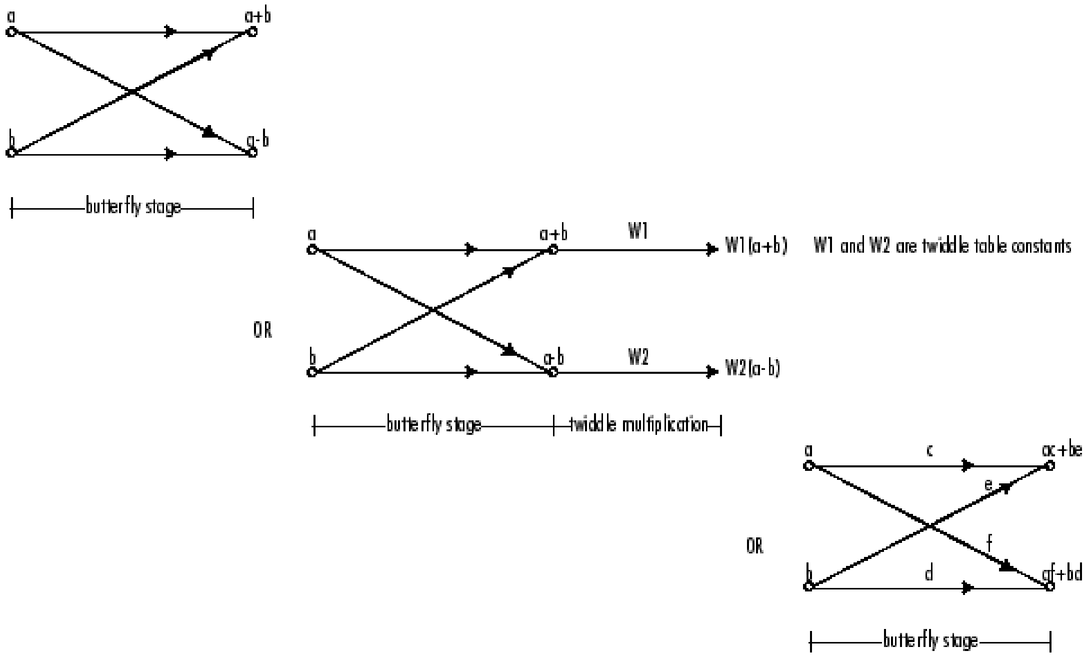
Sine and Cosine Computation Parameter Setting	Sine and Cosine Computation Method	Effect on Block Performance
`Table lookup`	The block computes and stores the trigonometric values before the simulation starts, and retrieves them during the simulation. When you generate code from the block, the processor running the generated code stores the trigonometric values computed by the block in a speed-optimized table, and retrieves the values during code execution.	The block usually runs much more quickly, but requires extra memory for storing the precomputed trigonometric values.
`Trigonometric fcn`	The block computes sine and cosine values during the simulation. When you generate code from the block, the processor running the generated code computes the sine and cosine values while the code runs.	The block usually runs more slowly, but does not need extra data memory. For code generation, the block requires a support library to emulate the trigonometric functions, increasing the size of the generated code.

This block supports Simulink® virtual buses.

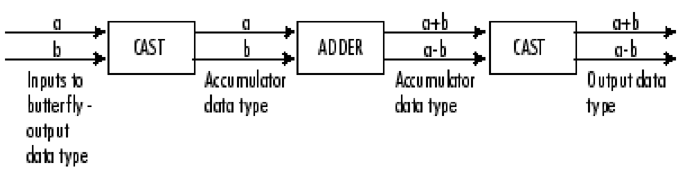
Fixed-Point Data Types

The diagrams below show the data types used within the DCT block for fixed-point signals. You can set the sine table, accumulator, product output, and output data types displayed in the diagrams in the DCT block dialog as discussed in "Dialog Box" on page 2-259.

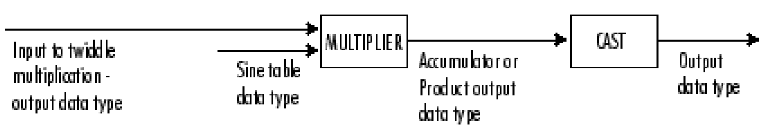
Inputs to the DCT block are first cast to the output data type and stored in the output buffer. Each butterfly stage processes signals in the accumulator data type, with the final output of the butterfly being cast back into the output data type.



Butterfly Stage Data Types



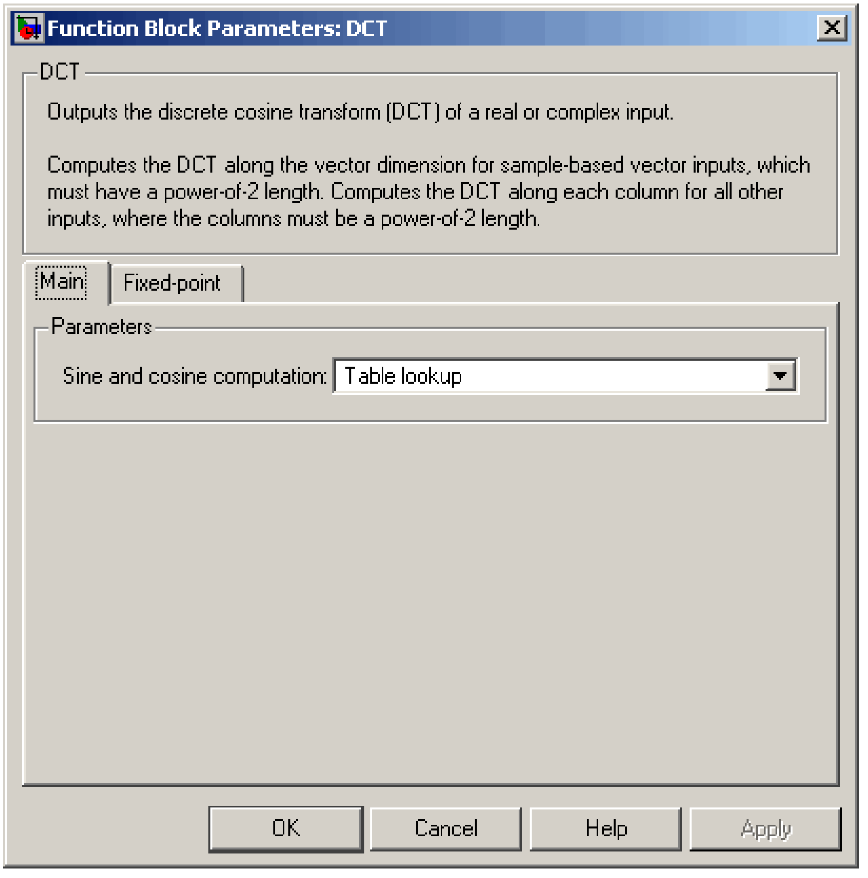
Twiddle Multiplication Data Types



The output of the multiplier is in the product output data type when at least one of the inputs to the multiplier is real. When both of the inputs to the multiplier are complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see "Multiplication Data Types".

Dialog Box

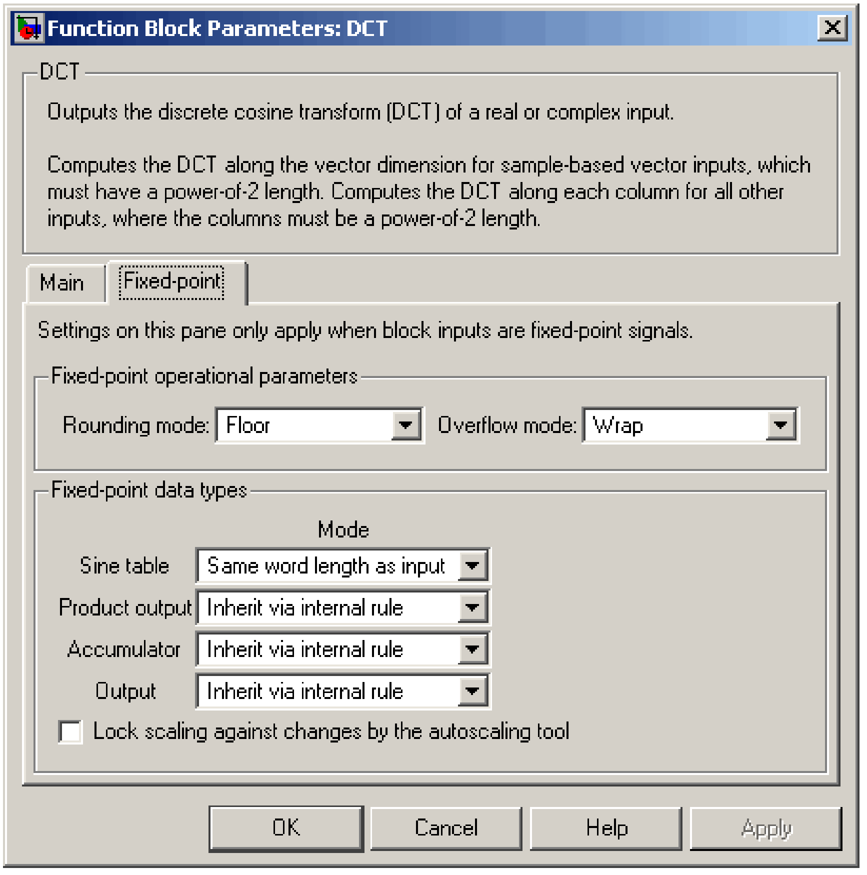
The **Main** pane of the DCT block dialog appears as follows.



Sine and cosine computation

Sets the block to compute sines and cosines by either looking up sine and cosine values in a speed-optimized table (`Table lookup`), or by making sine and cosine function calls (`Trigonometric fcn`). See the previous table.

The **Fixed-point** pane of the DCT block dialog appears as follows.



Rounding mode
Select the rounding mode for fixed-point operations. The sine table values do not obey this parameter; they always round to `Nearest`.

Overflow mode
Select the overflow mode for fixed-point operations.

Sine table
Choose how you specify the word length of the values of the sine table. The fraction length of the sine table values is always equal to the word length minus one:

- When you select `Same word length as input`, the word length of the sine table values match that of the input to the block.
- When you select `Specify word length`, you can enter the word length of the sine table values, in bits.

The sine table values do not obey the **Rounding mode** and **Overflow mode** parameters; they are always saturated and rounded to `Nearest`.

Product output
Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-257 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-257 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Inherit via internal rule`, the output word length and fraction length are calculated automatically. The internal rule first calculates an ideal output word length and fraction length using the following equations:

$$WL_{ideal\ output} = WL_{input} + floor(\log_2(DCT\ length - 1)) + 1$$

$$FL_{ideal\ output} = FL_{input}$$

Using these ideal results, the internal rule then selects word lengths and fraction lengths that are appropriate for your hardware. For more information, see "Inherit via Internal Rule".

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Complex Cepstrum	Signal Processing Blockset
FFT	Signal Processing Blockset
IDCT	Signal Processing Blockset
Real Cepstrum	Signal Processing Blockset
`dct`	Signal Processing Toolbox

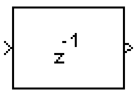
Delay

Purpose Delay discrete-time input by specified number of samples or frames

Library Signal Operations

`dspsigops`

Description



Delay

The Delay block delays a discrete-time input by the number of samples or frames specified in the **Delay units** and **Delay** parameters. The **Delay** value must be an integer value greater than or equal to zero. When you enter a value of zero for the **Delay** parameter, any initial conditions you might have entered have no effect on the output.

The Delay block allows you to set the initial conditions of the signal that is being delayed. The initial conditions must be numeric.

This block reference contains the following topics:

- "Sample-Based Operation" on page 2-264 — Use the Delay block with a sample-based input signal
- "Frame-Based Operation" on page 2-265 — Use the Delay block with a frame-based input signal

Sample-Based Operation

When the input is a sample-based N-D array, each sample of the input is treated as an independent channel. Thus, the total number of channels is equal to the product of the input dimensions. The dimension of the output is the same as that of the input.

When the input is a sample-based N-D array, the **Delay** parameter can be a scalar integer by which to equally delay all channels or an N-D array of the same dimensions as the input array, containing nonnegative integers that specify the number of sample intervals to delay each channel of the input.

There are four different choices for initial conditions. The initial conditions can be the same or different for each channel. They can also be the same or different within a channel. See the "Sample-Based Operation Examples" on page 2-268 section for more information.

Frame-Based Operation

When the input is a frame-based M-by-N matrix, the block treats each of the N columns as an independent channel, and delays each channel as specified by the **Delay** parameter.

When the input is frame based, the **Delay** parameter can be a scalar integer by which to equally delay all channels or a vector whose length is equal to the number of channels.

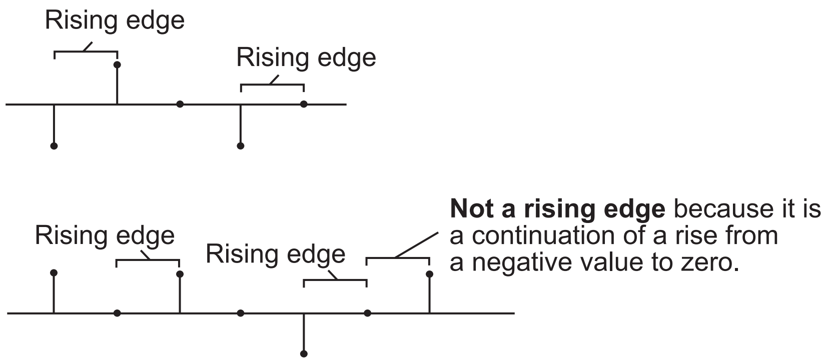
There are four different choices for initial conditions. The initial conditions can be the same or different for each channel. They can also be constant or varying along each channel. See the "Frame-Based Operation Examples" on page 2-273 section for more information.

Resetting the Delay

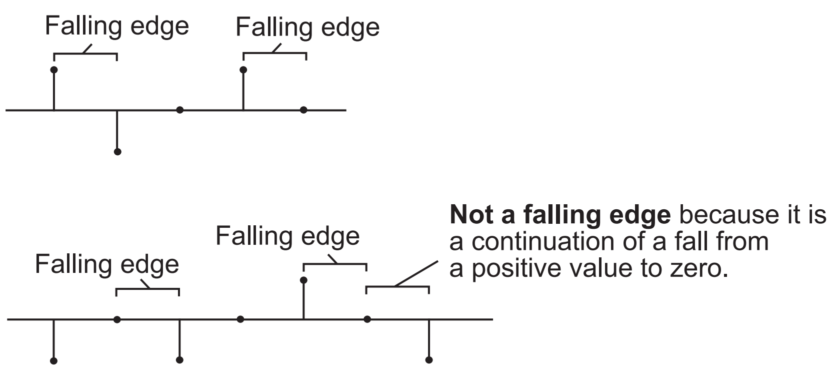
The Delay block resets the delay whenever it detects a reset event at the optional `Rst` port. The reset sample time must be a positive integer multiple of the input sample time.

The reset event is specified by the **Reset port** parameter, and can be one of the following:

- `None` disables the `Rst` port.
- `Rising edge` triggers a reset operation when the `Rst` input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)

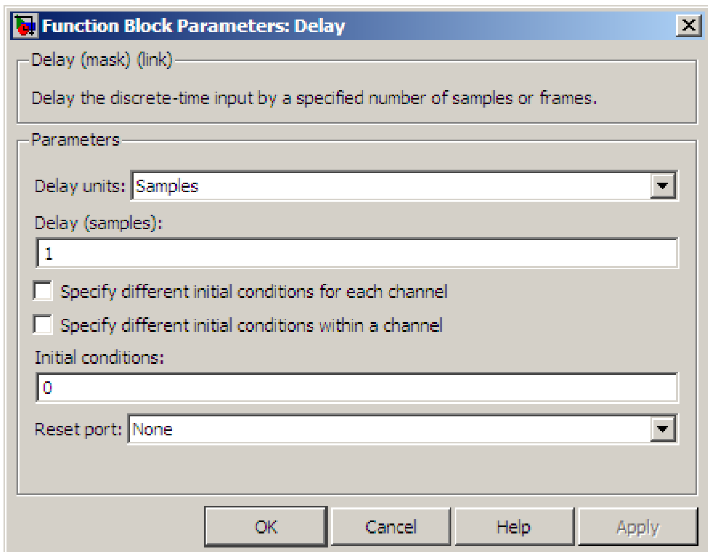


- `Either edge` triggers a reset operation when the `Rst` input is `Rising edge` or `Falling edge` (as described earlier).
- `Non-zero sample` triggers a reset operation at each sample time that the `Rst` input is not zero.

Note When running simulations in the Simulink® MultiTasking mode, reset signals have a one-sample latency. Therefore, when the block detects a reset event, there is a one-sample delay at the reset port rate before the block applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

This block supports Simulink virtual buses.

Dialog Box



Delay units

Select whether you want to delay your input by a specified number of `Samples` or `Frames`. You can choose to delay your signal by a certain number of samples or frames, regardless of whether your input is sample or frame based.

Delay (samples) or Delay (frames)

See "Sample-Based Operation" on page 2-264 and "Frame-Based Operation" on page 2-265 for a description of what format to use for each configuration of the block dialog.

Specify different initial conditions for each channel

Select this check box when you want the initial conditions to vary across the channels. When you do not select this check box, the initial conditions are the same across the channels.

Specify different initial conditions within a channel

Select this check box when you want the initial conditions to vary within the channels. When you do not select this check box, the initial conditions are the same within the channels.

Initial conditions

Enter a scalar, vector, matrix, or cell array of initial condition values, depending on your choice for the **Specify different initial conditions for each channel** and **Specify different initial conditions within a channel** check boxes. See "Sample-Based Operation" on page 2-264 and "Frame-Based Operation" on page 2-265 for a description of what format to use for each configuration of the block dialog.

Reset port

Determines the reset event that causes the block to reset the delay. For more information, see "Resetting the Delay" on page 2-265.

Examples

Sample-Based Operation Examples

There are four different choices for initial conditions. The initial conditions can be the same or different for each channel. They can also be the same or different along each channel. The next sections describe the behavior of the block for each of these four cases:

- "Case 1 — Use the Same Initial Conditions for Each Channel and Within a Channel" on page 2-269
- "Case 2 — Use Different Initial Conditions for Each Channel and the Same Initial Conditions Within a Channel" on page 2-270

- "Case 3 — Use the Same Initial Conditions for Each Channel and Different Initial Conditions Within a Channel" on page 2-271
- "Case 4 — Use Different Initial Conditions for Each Channel and Within a Channel" on page 2-272

Case 1 — Use the Same Initial Conditions for Each Channel and Within a Channel

Enter a scalar value for the initial conditions. This value is used as the constant initial condition value for each of the channels.

For example, suppose your input is a sample-based matrix.

$$\begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix}, \begin{bmatrix} 2 & 2 \\ 2 & 2 \end{bmatrix}, \begin{bmatrix} 3 & 3 \\ 3 & 3 \end{bmatrix}, \ldots$$

You want the initial conditions of your four-channel signal to be identical and zero for the first two samples:

1. For the **Delay (samples)** parameter, type `2`.
2. Clear the **Specify different initial conditions for each channel** and **Specify different initial conditions within a channel** check boxes.
3. For the **Initial conditions** parameter, specify a scalar value of `0`.

   The output of the delay block is

   $$\begin{bmatrix} 0 & 0 \\ 0 & 0 \end{bmatrix}, \begin{bmatrix} 0 & 0 \\ 0 & 0 \end{bmatrix}, \begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix}, \begin{bmatrix} 2 & 2 \\ 2 & 2 \end{bmatrix}, \begin{bmatrix} 3 & 3 \\ 3 & 3 \end{bmatrix}, \ldots$$

   `0`, the scalar initial condition value, is used for each channel and within the channels. It is the output at sample time zero and sample time one.

Case 2 — Use Different Initial Conditions for Each Channel and the Same Initial Conditions Within a Channel

The initial conditions must be an N-D array for N-D input. The initial conditions must have the same dimensions as the input data. These initial condition values are used as the constant initial condition value for each of the channels.

For example, suppose your input is a sample-based matrix.

$$\begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix}, \begin{bmatrix} 2 & 2 \\ 2 & 2 \end{bmatrix}, \begin{bmatrix} 3 & 3 \\ 3 & 3 \end{bmatrix}, \ldots$$

You want the initial conditions of your four-channel signal to be

$$\begin{bmatrix} 7 & 9 \\ 11 & 13 \end{bmatrix}$$

for the first two samples:

1 For the **Delay (samples)** parameter, type `2`.

2 Select the **Specify different initial conditions for each channel** check box.

3 Clear the **Specify different initial conditions within a channel** check box.

4 For the **Initial conditions** parameter, type `[7 9; 11 13]`.

   The output of the delay block is

$$\begin{bmatrix} 7 & 9 \\ 11 & 13 \end{bmatrix}, \begin{bmatrix} 7 & 9 \\ 11 & 13 \end{bmatrix}, \begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix}, \begin{bmatrix} 2 & 2 \\ 2 & 2 \end{bmatrix}, \begin{bmatrix} 3 & 3 \\ 3 & 3 \end{bmatrix}, \ldots$$

The initial condition matrix is the output at sample time zero and sample time one. Different initial conditions are used for each channel; the same initial condition value is used within a channel.

Case 3 — Use the Same Initial Conditions for Each Channel and Different Initial Conditions Within a Channel

In this case, for N-D sample-based inputs, the initial conditions parameter must be a vector whose length is equal to the delay value, specified by the **Delay** parameter. The values in this vector are used as the initial condition values along each of the channels to be delayed.

For example, suppose your input is a sample-based matrix.

$$\begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix}, \begin{bmatrix} 2 & 2 \\ 2 & 2 \end{bmatrix}, \begin{bmatrix} 3 & 3 \\ 3 & 3 \end{bmatrix}, \ldots$$

You want the initial conditions of your four channel signal to be the same along each of the channels to be delayed:

1. For the **Delay (samples)** parameter, type `2`.
2. Clear the **Specify different initial conditions for each channel** check box.
3. Select the **Specify different initial conditions within a channel** check box.
4. For the **Initial conditions** parameter, type `[10 20]`.

   The output of the delay block is

   $$\begin{bmatrix} 10 & 10 \\ 10 & 10 \end{bmatrix}, \begin{bmatrix} 20 & 20 \\ 20 & 20 \end{bmatrix}, \begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix}, \begin{bmatrix} 2 & 2 \\ 2 & 2 \end{bmatrix}, \begin{bmatrix} 3 & 3 \\ 3 & 3 \end{bmatrix}, \ldots$$

   The first element of the initial conditions vector is the output, for all channels, at sample time zero. The second element of the initial conditions vector is the output, for all channels, at sample time one.

The same initial conditions are used for each channel, but different initial condition values are used within a channel.

Case 4 — Use Different Initial Conditions for Each Channel and Within a Channel

Enter a cell array for your initial condition values. The cell array must be the same size as your input signal. Each cell of the cell array represents the delay values for one channel, and must be a vector of size equal to the delay value. If you have a vector or scalar input and a scalar delay value, you can enter the initial conditions as a matrix.

For example, suppose your input is a sample-based vector.

$$[1\quad 1],\quad [2\quad 2],\quad [3\quad 3],\ldots$$

You want the initial conditions of your two channel signal to be different for each channel and along each channel:

1. For the **Delay (samples)** parameter, type `2`.

2. Select the **Specify different initial conditions for each channel** and **Specify different initial conditions within a channel** check boxes.

3. For the **Initial conditions** parameter, type `[10 20; 30 40]`

   The output of the delay block is

   $$[10\quad 20],\quad [30\quad 40],\quad [1\quad 1],\quad [2\quad 2]\ldots$$

   The first row of the initial conditions vector is the output at sample time zero. The second row of the initial conditions vector is the output at sample time one. Different initial conditions are used for each channel and within the channels.

In addition, suppose your input is a sample-based matrix.

$$\begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix}, \begin{bmatrix} 2 & 2 \\ 2 & 2 \end{bmatrix}, \begin{bmatrix} 3 & 3 \\ 3 & 3 \end{bmatrix}, \ldots$$

You want the initial conditions of your two-channel signal to be different for each channel and along each channel:

1 For the **Delay (samples)** parameter, type `2`.

2 Select the **Specify different initial conditions for each channel** and the **Specify different initial conditions within a channel** check boxes.

3 For the **Initial conditions** parameter, type `{[11 15] [12 16]; [13 17] [14 18]}`. The dimensions of the cell array match the dimensions of the input. Also, each element of the cell array represents the initial conditions within one channel.

   The output of the delay block is

$$\begin{bmatrix} 11 & 12 \\ 13 & 14 \end{bmatrix}, \begin{bmatrix} 15 & 16 \\ 17 & 18 \end{bmatrix}, \begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix}, \begin{bmatrix} 2 & 2 \\ 2 & 2 \end{bmatrix}, \ldots$$

   Each element of the cell array represents the initial conditions within a channel. The first element, a vector, represents the initial conditions within channel 1. The second element, a vector, represents the initial conditions within channel 2, and so on. Different initial conditions are used for each channel and within the channels.

Frame-Based Operation Examples

There are four different choices for initial conditions. The initial conditions can be the same or different for each channel. They can also be constant or varying along each channel. The next sections describe the behavior of the block for each of these four cases:

- "Case 1 — Use the Same Initial Conditions for Each Channel and Within a Channel" on page 2-274
- "Case 2 — Use Different Initial Conditions for Each Channel and the Same Initial Conditions Within a Channel" on page 2-275
- "Case 3 — Use the Same Initial Conditions for Each Channel and Different Initial Conditions Within a Channel" on page 2-276
- "Case 4 — Use Different Initial Conditions for Each Channel and Within a Channel" on page 2-277

Case 1 — Use the Same Initial Conditions for Each Channel and Within a Channel

Enter a scalar value for the initial conditions. This value is used as the constant initial condition value for each of the channels.

For example, suppose your input is a frame-based matrix.

$$\begin{bmatrix} 1 & 1 & 1 \\ 2 & 2 & 2 \\ 3 & 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 & 4 \\ 5 & 5 & 5 \\ 6 & 6 & 6 \end{bmatrix}, \begin{bmatrix} 7 & 7 & 7 \\ 8 & 8 & 8 \\ 9 & 9 & 9 \end{bmatrix}, \ldots$$

You want the initial conditions of your three-channel signal to be identical and zero for the first frame:

1. For the **Delay (frames)** parameter, type `1`.
2. Clear the **Specify different initial conditions for each channel** and the **Specify different initial conditions within a channel** check boxes.
3. For the **Initial conditions** parameter, specify a scalar value of `0`.

   The output of the delay block is

$$\begin{bmatrix} 0 & 0 & 0 \\ 0 & 0 & 0 \\ 0 & 0 & 0 \end{bmatrix}, \begin{bmatrix} 1 & 1 & 1 \\ 2 & 2 & 2 \\ 3 & 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 & 4 \\ 5 & 5 & 5 \\ 6 & 6 & 6 \end{bmatrix}, \begin{bmatrix} 7 & 7 & 7 \\ 8 & 8 & 8 \\ 9 & 9 & 9 \end{bmatrix}, \ldots$$

`0`, the scalar initial condition value, is used across the channels and within the channels for the first frame. This frame is the output at sample time zero.

Case 2 — Use Different Initial Conditions for Each Channel and the Same Initial Conditions Within a Channel

The initial conditions must be a vector of length N, where $N \geq 1$. N is also equal to the number of channels in your signal. These initial condition values are used as the constant initial condition value for each of the channels.

For example, suppose your input is a frame-based matrix.

$$\begin{bmatrix} 1 & 1 & 1 \\ 2 & 2 & 2 \\ 3 & 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 & 4 \\ 5 & 5 & 5 \\ 6 & 6 & 6 \end{bmatrix}, \begin{bmatrix} 7 & 7 & 7 \\ 8 & 8 & 8 \\ 9 & 9 & 9 \end{bmatrix}, \ldots$$

You want the initial conditions of your three-channel signal to be `[0 10 20]` for the first frame:

1. For the **Delay (frames)** parameter, type `1`.
2. Select the **Specify different initial conditions for each channel** check box.
3. Clear the **Specify different initial conditions within a channel** check box.
4. For the **Initial conditions** parameter, type `[0 10 20]`.

   The output of the delay block is

$$\begin{bmatrix} 0 & 10 & 20 \\ 0 & 10 & 20 \\ 0 & 10 & 20 \end{bmatrix}, \begin{bmatrix} 1 & 1 & 1 \\ 2 & 2 & 2 \\ 3 & 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 & 4 \\ 5 & 5 & 5 \\ 6 & 6 & 6 \end{bmatrix}, \begin{bmatrix} 7 & 7 & 7 \\ 8 & 8 & 8 \\ 9 & 9 & 9 \end{bmatrix}, \ldots$$

The initial condition vector expands to create the frame that is output at sample time zero. Different initial conditions are used for each channel, but the same initial condition value is used with a channel.

Case 3 — Use the Same Initial Conditions for Each Channel and Different Initial Conditions Within a Channel

In this case, the **Delay** parameter can be a scalar integer by which to equally delay all channels or a vector whose length is equal to the number of channels. All the values of this vector must be equal.

Enter the initial conditions as a vector. These values are used as the initial condition value along each of the channels to be delayed. The initial condition vector must have length equal to the value of the **Delay (frames)** parameter multiplied by the frame length. For example, if you want to delay your signal by two frames with frame length two and an initial condition value of 3, enter your initial condition vector as `[3 3 3 3]`.

For example, suppose your input is a frame-based matrix.

$$\begin{bmatrix} 1 & 1 & 1 \\ 2 & 2 & 2 \\ 3 & 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 & 4 \\ 5 & 5 & 5 \\ 6 & 6 & 6 \end{bmatrix}, \begin{bmatrix} 7 & 7 & 7 \\ 8 & 8 & 8 \\ 9 & 9 & 9 \end{bmatrix}, \ldots$$

You want the initial conditions of your three-channel signal to be the same along each of the channels to be delayed:

1 For the **Delay (frame)** parameter, type `1`.

2 Clear the **Specify different initial conditions for each channel** check box.

3 Select the **Specify different initial conditions within a channel** check box.

4 For the **Initial conditions** parameter, type `[10 20 30]`.

The output of the delay block is

$$\begin{bmatrix} 10 & 10 & 10 \\ 20 & 20 & 20 \\ 30 & 30 & 30 \end{bmatrix}, \begin{bmatrix} 1 & 1 & 1 \\ 2 & 2 & 2 \\ 3 & 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 & 4 \\ 5 & 5 & 5 \\ 6 & 6 & 6 \end{bmatrix}, \begin{bmatrix} 7 & 7 & 7 \\ 8 & 8 & 8 \\ 9 & 9 & 9 \end{bmatrix}, \ldots$$

The initial condition vector defines the initial condition values within each of the three channels. The same initial conditions are used for each channel, but different initial condition values are used with a channel.

Case 4 — Use Different Initial Conditions for Each Channel and Within a Channel

Enter a cell array for your initial condition values. Or, when you have a scalar delay value, you can enter the initial conditions as a matrix.

For example, suppose your input is a frame-based matrix.

$$\begin{bmatrix} 1 & 1 & 1 \\ 2 & 2 & 2 \\ 3 & 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 & 4 \\ 5 & 5 & 5 \\ 6 & 6 & 6 \end{bmatrix}, \begin{bmatrix} 7 & 7 & 7 \\ 8 & 8 & 8 \\ 9 & 9 & 9 \end{bmatrix}, \ldots$$

You want the initial conditions of your three-channel signal to be different for each channel and along each channel.

1 For the **Delay (frames)** parameter, type `1`.

2 Select the **Specify different initial conditions for each channel** and the **Specify different initial conditions within a channel** check boxes.

3 For the **Initial conditions** parameter, type either `[10 20 30; 40 50 60; 70 80 90]` or `{[10 40 70];[20 50 80];[30 60 90]}`. Each cell of the cell array represents the delay along one channel.

Regardless of whether you use a matrix or cell array, the output of the delay block is

$$\begin{bmatrix} 10 & 20 & 30 \\ 40 & 50 & 60 \\ 70 & 80 & 90 \end{bmatrix}, \begin{bmatrix} 1 & 1 & 1 \\ 2 & 2 & 2 \\ 3 & 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 & 4 \\ 5 & 5 & 5 \\ 6 & 6 & 6 \end{bmatrix}, \begin{bmatrix} 7 & 7 & 7 \\ 8 & 8 & 8 \\ 9 & 9 & 9 \end{bmatrix} \ldots$$

The initial condition matrix is the output at sample time zero. The elements of the initial condition cell array define the initial condition values within each channel. The first element, a vector, represents the initial conditions within channel 1. The second element, a vector, represents the initial conditions within channel 2, and so on. Different initial conditions are used for each channel and within the channels.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- Boolean
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Unit Delay	Simulink
Variable Fractional Delay	Signal Processing Blockset
Variable Integer Delay	Signal Processing Blockset

Purpose

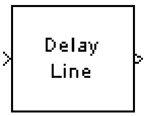
Rebuffer sequence of inputs with one-sample shift

Library

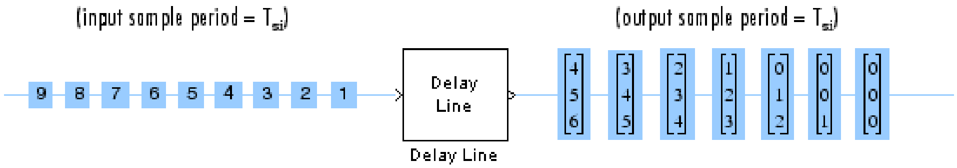
Signal Management / Buffers

`dspbuff3`

Description



The Delay Line block buffers the input samples into a sequence of overlapping or underlapping matrix outputs. In the most typical use (sample-based inputs), each output differs from the preceding output by only one sample, as illustrated below for scalar input.



Note that the first output of the block in the example above is all zeros; this is because the **Initial Conditions** parameter is set to zero. Due to the latency of the Delay Line block, all outputs are delayed by one frame, the entries of which are defined by the **Initial Conditions** parameter.

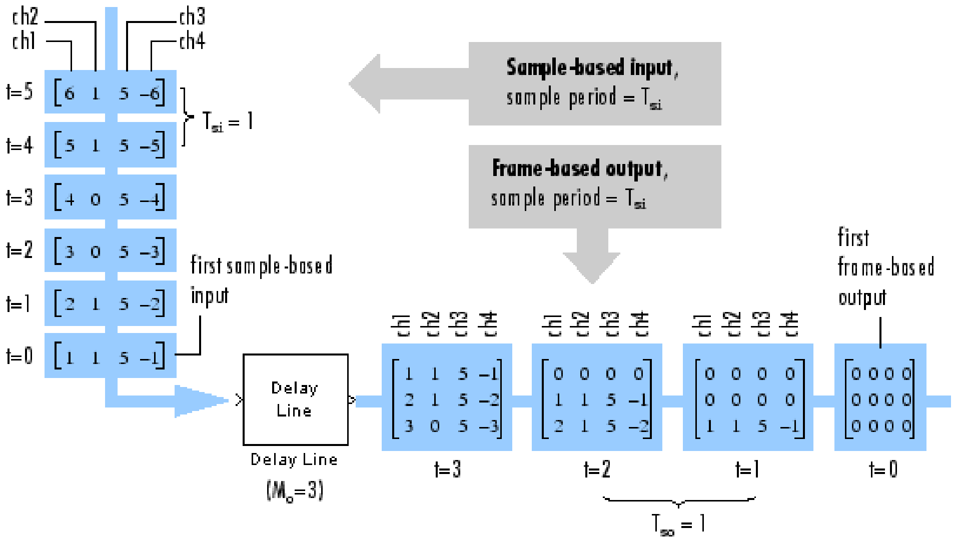
Sample-Based Operation

In sample-based operation, the Delay Line block buffers a sequence of sample-based length-N vector inputs (1-D, row, or column) into a sequence of overlapping frame-based M_o-by-N matrix outputs, where M_o is specified by the **Delay line size** parameter (M_o>1). That is, each input vector becomes a *row* in the frame-based output matrix.

At each sample time the new input vector is added in the last row of the output, so each output overlaps the previous output by M_o-1 samples. Therefore, the output sample period and frame period is the same as the input sample period ($T_{so}=T_{si}$, and $T_{fo}=T_{si}$). When M_o=1, the input is simply passed through to the output and retains the same dimension, but becomes frame based. The latency of the block always causes an initial delay in the output; the value of the first output is specified by the **Initial conditions** parameter (see “Initial Conditions” on page 2-282). Sample-based full-dimension matrix inputs are not accepted.

The Delay Line block's sample-based operation is similar to that of a Buffer block with **Buffer size** equal to M_o and **Buffer overlap** equal to M_o-1, except that the Buffer block has a different latency.

In the following model, the block operates on a sample-based input with a **Delay line size** of 3.



The input vectors in the example above do not begin appearing at the output until the second row of the second matrix due to the block's latency (see "Initial Conditions" on page 2-282). The first output matrix (all zeros in this example) reflects the block's **Initial conditions** setting. As for any sample-based input, the output frame rate and output sample rate are both equal to the input sample rate.

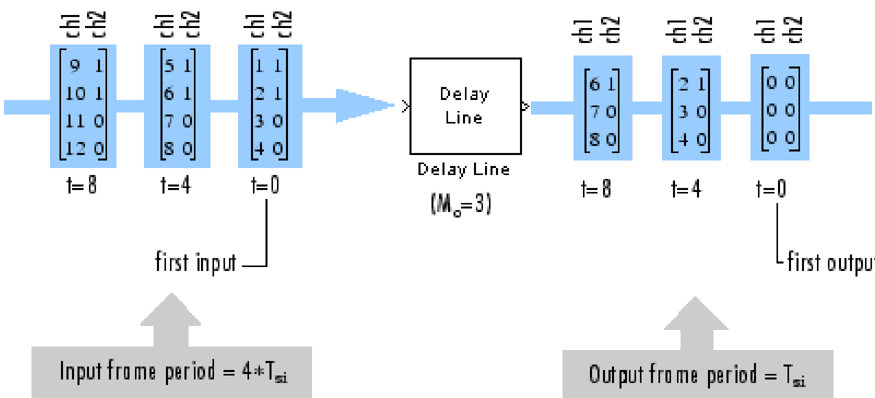
Frame-Based Operation

In frame-based operation, the Delay Line block rebuffers a sequence of frame-based M_i-by-N matrix inputs into a sequence of frame-based M_o-by-N matrix outputs, where M_o is the output frame size specified by the **Delay line size** parameter. Depending on whether M_o is greater than, less than, or equal to the input frame size, M_i, the output frames can be underlapped or overlapped. Each of the N input channels is rebuffered independently.

When $M_o > M_i$, the output frame overlap is the difference between the output and input frame size, M_o-M_i. When $M_o < M_i$, the output is underlapped; the Delay Line block discards the first M_i-M_o samples of each input frame so that only the last M_o samples are buffered into the corresponding output frame. When $M_o = M_i$, the output data is identical to the input data, but is delayed by the latency of the block. Due to the block's latency, the outputs are always delayed by one frame, the entries of which are specified by the **Initial conditions** (see "Initial Conditions" on page 2-282).

The output frame period is equal to the input frame period (T_{fo}=T_{fi}). The output sample period, T_{so}, is therefore equal to T_{fi}/M_o, or equivalently, $T_{si}(M_i/M_o)$

In the following model, the block rebuffers a two-channel frame-based input with a **Delay line size** of 3.

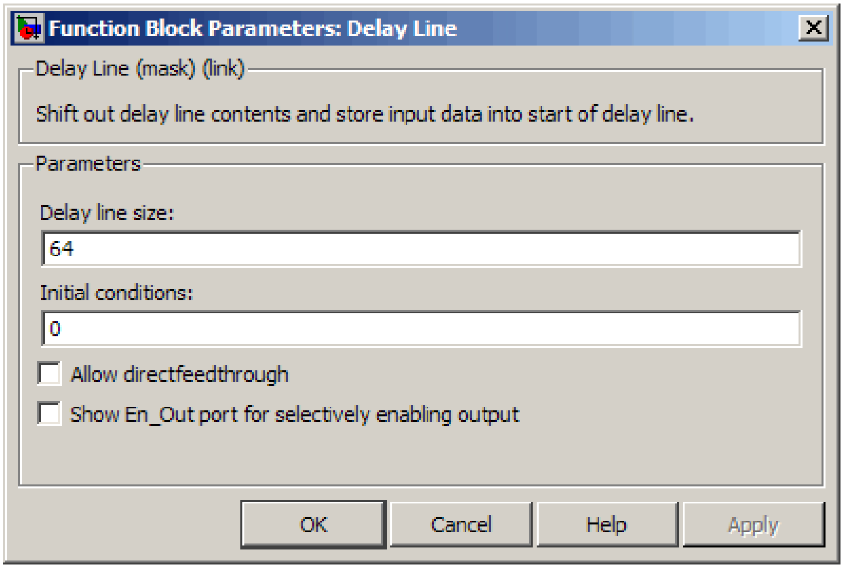


The first output frame in the example is a product of the latency of the Delay Line block; it is all zeros because the **Initial conditions** is set to be zero. Since the input frame size, 4, is larger than the output frame size, 3, only the last three samples in each input frame are propagated to the corresponding output frame. The frame periods of the input and output are the same, and the output sample period is $T_{si}(M_i/M_o)$, or 4/3 the input sample period.

Initial Conditions

The Delay Line block's buffer is initialized to the value specified by the **Initial condition** parameter. The block outputs this buffer at the first simulation step (t=0). When the block's output is a vector, the **Initial condition** can be a vector of the same size, or a scalar value to be repeated across all elements of the initial output. When the block's output is a matrix, the **Initial condition** can be a matrix of the same size, a vector (of length equal to the number of matrix rows) to be repeated across all columns of the initial output, or a scalar to be repeated across all elements of the initial output.

Dialog Box



Delay line size

The number of rows in output matrix, M_o.

Initial conditions
The value of the block's initial output, a scalar, vector, or matrix.

Allow direct feedthrough
When you select this check box, the input data is not delayed by an extra frame before it is available at the output buffer. Instead, the input data is available immediately at the output port of the block.

Show En_Out port for selectively enabling output
When you select this check box, the En_Out port appears on the block icon. The block internally uses a circular buffer, even though the output is linear. This means, for valid output, data from the circular buffer has to be linearized.

The En_Out port determines whether or not a valid output needs to be computed based on the value of the `Boolean` input. If the input value to the En_Out port is `1`, the block output is linearized, and thus is valid. Otherwise, the output is not linearized, and is invalid. This allows the block to be more efficient when the tapped Delay Line's output is not required at each sample time.

Hold previous value when the output is disabled
This parameter only appears and applies when the **Show En_Out port for selectively enabling output** parameter is selected. Use this parameter to specify the block output at those time steps when the internal state buffer is not being linearized to output valid data. If you select this parameter, the most recent valid value is held on the output port. If you do not select the parameter, the signal on the output port is invalid data.

Supported Data Types

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

See Also

Buffer	Signal Processing Blockset
Triggered Delay Line	Signal Processing Blockset

Purpose

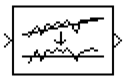
Remove linear trend from vectors

Library

Statistics

`dspstat3`

Description



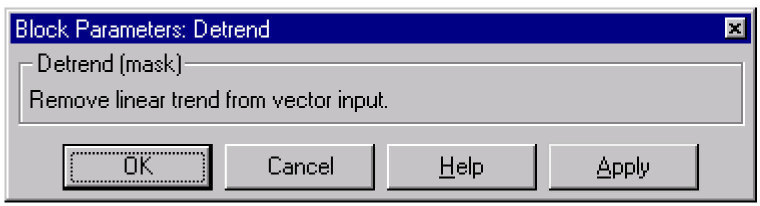
The Detrend block removes a linear trend from the length-M input vector, u, by subtracting the straight line that best fits the data in the least squares sense.

The least squares line, $\hat{u} = ax + b$, is the line with parameters a and b that minimizes the quantity

$$\sum_{i=1}^{M}(u_i - u_i)^2$$

for M evenly-spaced values of x, where u_i is the ith element in the input vector. The output, $y = u\text{-}\hat{u}$, is an M-by-1 column vector (regardless of the input vector dimension) with the same frame status as the input.

Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Cumulative Sum	Signal Processing Blockset
Difference	Signal Processing Blockset
Least Squares Polynomial Fit	Signal Processing Blockset
Unwrap	Signal Processing Blockset
`detrend`	MATLAB

Purpose

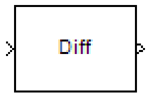
Compute element-to-element difference along specified dimension of input

Library

Math Functions / Math Operations

`dspmathops`

Description



The Difference block computes the difference between adjacent elements in rows, columns, or a specified dimension of the input array `u`. This block accepts real and complex fixed-point and floating-point inputs, except for complex unsigned fixed-point inputs.

Columnwise Differencing

When the **Difference along** parameter is set to `Columns`, the block computes differences between adjacent elements along each column.

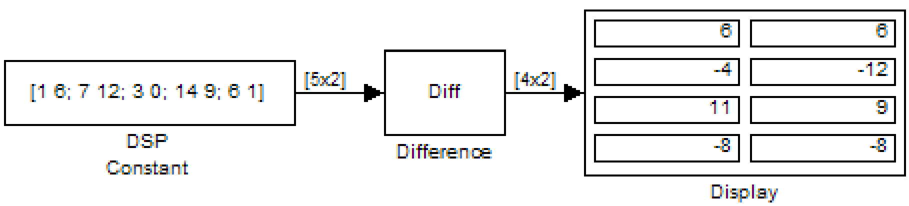
```
y = diff(u)      % Equivalent MATLAB code
```

For sample-based inputs, the output is a sample-based (M-1)-by-N matrix whose jth column has elements

$$y_{i,j} = u_{i+1,j} - u_{i,j} \quad 1 \le i \le (M-1)$$

For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors for columnwise differencing, and the output is 1-D.

For example, the following figure shows the block output for sample-based inputs:



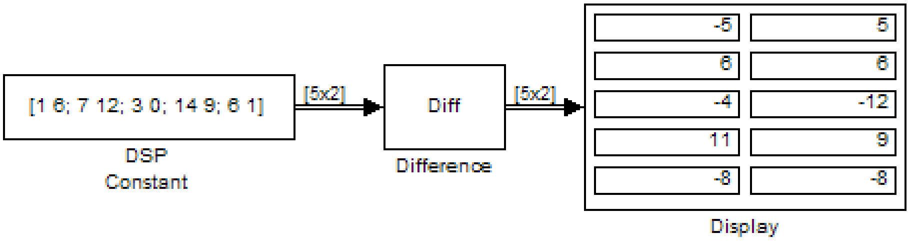
For frame-based inputs, the output is a frame-based M-by-N matrix whose jth column has elements

$$y_{i,j} = u_{i+1,j} - u_{i,j} \quad 2 \le i \le (M-1)$$

The first element of the output for each column is the first input element minus the last input element of the previous frame. For the first frame, zero is subtracted from the first input element.

$$y_{1,j}(t) = u_{1,j}(t) - u_{M,j}(t - T_f)$$

For example, the following figure shows the second frame of the block output for a frame-based input:



Rowwise Differencing

When the **Difference along** parameter is set to `Rows`, the block computes differences between adjacent elements along each row. The result is the same regardless of the frame status of the input signal.

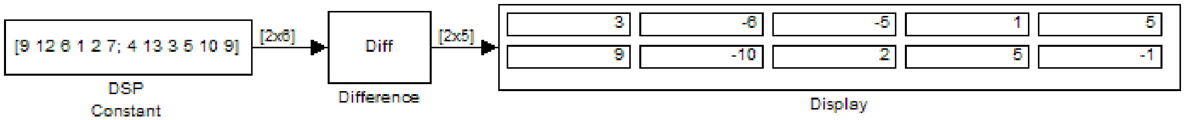
```
y = diff(u,[],2)      % Equivalent MATLAB code
```

The output is an M-by-(N-1) matrix whose ith row has elements

$$y_{i,j} = u_{i,j+1} - u_{i,j} \quad 1 \le j \le (N-1)$$

The frame status of the output is the same as the input. For convenience, length-N 1-D vector inputs are treated as 1-by-N row vectors for rowwise differencing, and the output is 1-D.

For example, the following figure shows the block output for sample-based inputs. The output is the same for frame-based inputs:



Differencing Along Arbitrary Dimensions

When the **Difference along** parameter is set to `Specified dimension`, the behavior of the block is an extension of the rowwise differencing described earlier. The block computes differences between adjacent elements along the dimension specified by the **Dimension** parameter.

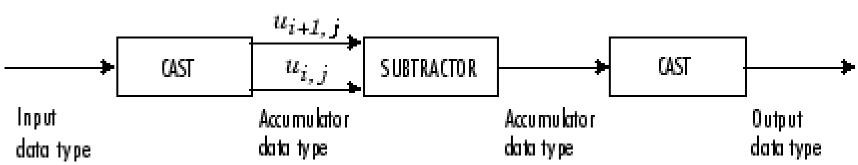
```
y = diff(u,[],d) % Equivalent MATLAB code where d is the dimension
```

The output is an array whose length in the specified dimension is one less than that of the input, and whose lengths in other dimensions are unchanged. For example, consider an M-by-N-by-P-by-R input array with elements *u(i,j,k,l)* and assume that **Dimension** is 3. The output of the block is an M-by-N-by-(P–1)-by-R array with elements

$$y_{i,j,k,l} = u_{i,j,k+1,l} - u_{i,j,k,l} \quad 1 \le k \le (P-1)$$

Fixed-Point Data Types

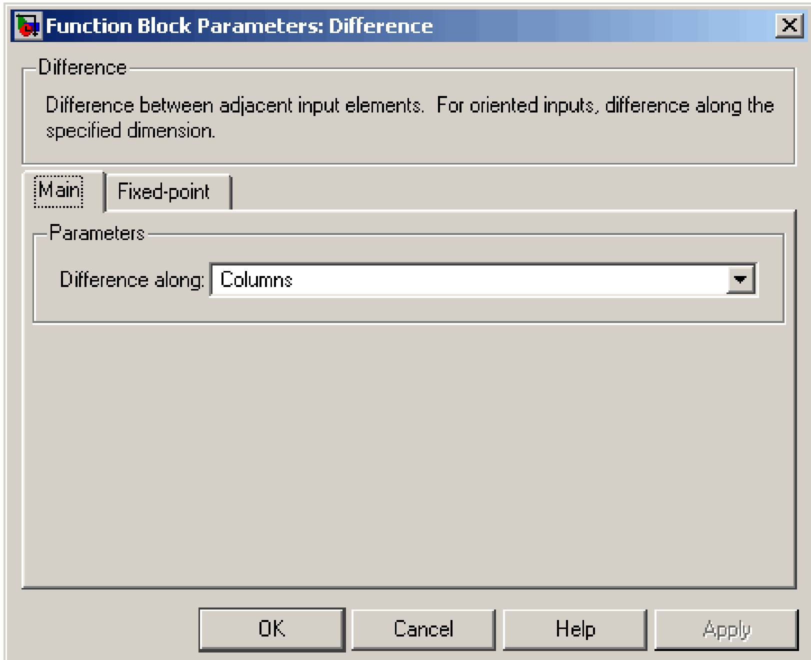
The following diagram shows the data types used within the Difference block for fixed-point signals.



You can set the accumulator and output data types in the block dialog as discussed in “Dialog Box” on page 2-290 .

Dialog Box

The **Main** pane of the Difference block appears as follows.

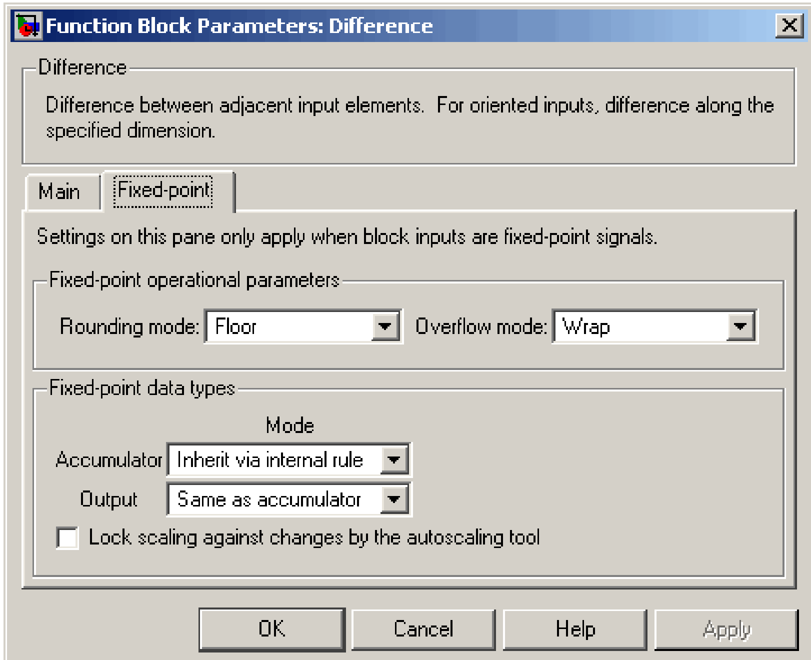


Difference along
: Specify whether the block performs columnwise differencing, rowwise differencing, or differencing along a specified dimension.

Dimension
: Specify the one-based dimension along which to compute element-to-element differences.

  This parameter is only visible when you select `Specified dimension` for the **Difference along** parameter.

The **Fixed-point** pane of the Difference block appears as follows.



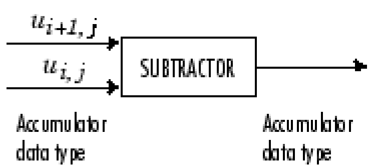
Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

Select the overflow mode for fixed-point operations.

Accumulator



Use this parameter to specify how you would like to designate the accumulator word and fraction lengths:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

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

Lock scaling against changes by the autoscaling tool

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

Supported Data Types

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

See Also

Cumulative Sum	Signal Processing Blockset
`diff`	MATLAB

Differentiator Filter

Purpose

Design differentiator filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description

This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See “Differentiator Filter Design Dialog Box — Main Pane” in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Digital Filter

Purpose

Filter each channel of input over time using static or time-varying digital filter implementations

Library

Filtering / Filter Designs

`dsparch4`

Description

Note Use this block to efficiently implement a floating-point or fixed-point filter for which you know the coefficients, or that is already defined in a Signal Processing Toolbox™ `dfilt` object or a Filter Design Toolbox™ `dfilt` object. The following Signal Processing Blockset™ blocks also implement digital filters, but serve slightly different purposes:

- Digital Filter Design — Use to design, analyze, and then efficiently implement floating-point filters. This block provides the same filter implementation as the Digital Filter block for floating-point signals.
- Filter Realization Wizard — Use to implement floating-point or fixed-point filters built from Sum, Gain, and Unit Delay blocks. You can either design the filter using block filter design and analysis parameters, or import the coefficients of a filter that you designed elsewhere.

The Digital Filter block independently filters each channel of the input signal with a specified digital IIR or FIR filter. The block can implement *static filters* with fixed coefficients, as well as *time-varying filters* with coefficients that change over time. You can tune the coefficients of a static filter during simulation.

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

The outputs of this block numerically match the outputs of the Digital Filter Design block and of the `dfilt` function in Signal Processing Toolbox software or Filter Design Toolbox software.

Note The Digital Filter block has direct feedthrough, so if you connect the output of this block back to its input you get an algebraic loop. For more information on direct feedthrough and algebraic loops, see "Algebraic Loops" in the Simulink® documentation.

Sections of This Reference Page

- "Coefficient Source" on page 2-297
- "Supported Filter Structures" on page 2-298
- "Specifying Initial Conditions" on page 2-300
- "State Logging" on page 2-304
- "Fixed-Point Data Types" on page 2-305
- "Dialog Box" on page 2-305
- "Filter Structure Diagrams" on page 2-320
- "Supported Data Types" on page 2-356
- "See Also" on page 2-356

Coefficient Source

The Digital Filter block can operate in three different modes. Select the mode in the **Coefficient source** group box. If you select

- **Dialog parameters**, you enter information about the filter such as structure and coefficients in the block mask.

- **Input port(s)**, you enter the filter structure in the block mask, and the filter coefficients come in through one or more block ports. This mode is useful for specifying time-varying filters.
- **Discrete-time filter object (DFILT)**, you specify the filter using a `dfilt` object from the Signal Processing Toolbox product or the Filter Design Toolbox product.

Supported Filter Structures

When you select **Discrete-time filter object (DFILT)**, the following `dfilt` structures are supported:

- `dfilt.df1`
- `dfilt.df1t`
- `dfilt.df2`
- `dfilt.df2t`
- `dfilt.df1sos`
- `dfilt.df1tsos`
- `dfilt.df2sos`
- `dfilt.df2tsos`
- `dfilt.dffir`
- `dfilt.dffirt`
- `dfilt.dfsymfir`
- `dfilt.dfasymfir`
- `dfilt.latticear`
- `dfilt.latticemamin`

When you select **Dialog parameters** or **Input port(s)**, the list of filter structures offered in the **Filter structure** parameter depends on whether you set the **Transfer function type** to `IIR (poles &`

zeros), IIR (all poles), or FIR (all zeros), as summarized in the following table.

Note Each structure listed in the table below supports both fixed-point and floating-point signals.

The table also shows the vector or matrix of filter coefficients you must provide for each filter structure. For more information on how to specify filter coefficients for various filter structures, see “Specifying Static Filters” and “Specifying Time-Varying Filters”.

Filter Structures and Filter Coefficients

Transfer Function Type	Supported Filter Structures	Filter Coefficient Specification
IIR (poles & zeros)	Direct form I Direct form I transposed Direct form II Direct form II transposed	• Numerator coefficients vector [b0, b1, b2, ..., bn] • Denominator coefficients vector [a0, a1, a2, ..., am]
	Biquadratic direct form I (SOS) Biquadratic direct form I transposed (SOS) Biquadratic direct form II (SOS) Biquadratic direct form II transposed (SOS)	• *M*-by-6 second-order section (SOS) matrix. • Scale values See “Specifying the SOS Matrix (Biquadratic Filter Coefficients)”.

Filter Structures and Filter Coefficients (Continued)

Transfer Function Type	Supported Filter Structures	Filter Coefficient Specification
IIR (all poles)	`Direct form` `Direct form transposed`	Denominator coefficients vector [a0, a1, a2, ..., am]
	`Lattice AR`	Reflection coefficients vector [k1, k2, ..., kn]

FIR (all zeros)	`Direct form`
	`Direct form symmetric`
	`Direct form antisymmetric`
	`Direct form transposed`
		Numerator coefficients vector [b0, b1, b2, ..., bn]

	`Lattice MA`	Reflection coefficients vector [k1, k2, ..., kn]

Specifying Initial Conditions

In **Dialog parameters** and **Input port(s)** modes, the block initializes the internal filter states to zero by default, which is equivalent to assuming past inputs and outputs are zero. You can optionally use the **Initial conditions** parameter to specify nonzero initial conditions for the filter delays.

To determine the number of initial condition values you must specify, and how to specify them, see the following table on Valid Initial Conditions and Number of Delay Elements (Filter States) on page 2-303. The **Initial conditions** parameter can take one of four forms as described in the following table.

Valid Initial Conditions

Initial Condition	Examples	Description
Scalar	5 Each delay element for each channel is set to 5.	The block initializes all delay elements in the filter to the scalar value.
Vector (for applying the same delay elements to each channel)	For a filter with two delay elements: $[d_1\ d_2]$ The delay elements for all channels are d1 and d2.	Each vector element specifies a unique initial condition for a corresponding delay element. The block applies the same vector of initial conditions to each channel of the input signal. The vector length must equal the number of delay elements in the filter (specified in the table Number of Delay Elements (Filter States) on page 2-303).

Valid Initial Conditions (Continued)

Initial Condition	Examples	Description
Vector or matrix (for applying different delay elements to each channel)	For a 3-channel input signal and a filter with two delay elements: $[d_1\ d_2\ D_1\ D_2\ d_1\ d_2]$ or $\begin{bmatrix} d_1 & D_1 & d_1 \\ d_2 & D_2 & d_2 \end{bmatrix}$ • The delay elements for channel 1 are d_1 and d_2. • The delay elements for channel 2 are D_1 and D_2. • The delay elements for channel 3 are $d_1$and d_2.	Each vector or matrix element specifies a unique initial condition for a corresponding delay element in a corresponding channel: • The vector length must be equal to the product of the number of input channels and the number of delay elements in the filter (specified in the table Number of Delay Elements (Filter States) on page 2-303). • The matrix must have the same number of rows as the number of delay elements in the filter (specified in the table Number of Delay Elements (Filter States) on page 2-303), and must have one column for each channel of the input signal.
Empty matrix	`[ ]` Each delay element for each channel is set to `0`.	The empty matrix, `[]`, is equivalent to setting the **Initial conditions** parameter to the scalar value `0`.

The number of delay elements (filter states) per input channel depends on the filter structure, as indicated in the following table.

Number of Delay Elements (Filter States)

Filter Structure	Number of Delay Elements per Channel

`Direct form`
`Direct form transposed`
`Direct form symmetric`
`Direct form antisymmetric`
	`#_of_filter_coeffs-1`

`Direct form I` `Direct form I transposed`	• `#_of_zeros-1` • `#_of_poles-1`
`Direct form II` `Direct form II transposed`	`max(#_of_zeros, #_of_poles)-1`

`Biquadratic direct form I (SOS)`
`Biquadratic direct form I transposed (SOS)`
`Biquadratic direct form II (SOS)`
`Biquadratic direct form II transposed (SOS)`
	`2 * #_of_filter_sections`

`Lattice AR` `Lattice MA`	`#_of_reflection_coeffs`

State Logging

Simulink enables you to log the states in your model to the MATLAB® workspace. The following table indicates which filter structures of the Digital Filter block support the Simulink state logging feature. See "States" in the Simulink User's Guide documentation for more information.

Transfer Function Type	Filter Structure	State Logging Supported
IIR (poles & zeros)	`Direct form I`	No
	`Direct form I transposed`	Yes
	`Direct form II`	No
	`Direct form II transposed`	Yes
	`Biquadratic direct form I (SOS)`	Yes
	`Biquadratic direct form I transposed (SOS)`	Yes
	`Biquadratic direct form II (SOS)`	Yes
	`Biquadratic direct form II transposed (SOS)`	Yes
IIR (all poles)	`Direct form`	No
	`Direct form transposed`	Yes
	`Lattice AR`	Yes
FIR (all zeros)	`Direct form`	No
	`Direct form symmetric`	No
	`Direct form antisymmetric`	No
	`Direct form transposed`	Yes
	`Lattice MA`	Yes

Fixed-Point Data Types

All structures supported by the Digital Filter block support fixed-point data types. You can specify intermediate fixed-point data types for quantities such as the coefficients, accumulator, and product output for each filter structure. See "Filter Structure Diagrams" on page 2-320 for diagrams depicting the use of these intermediate fixed-point data types in each filter structure.

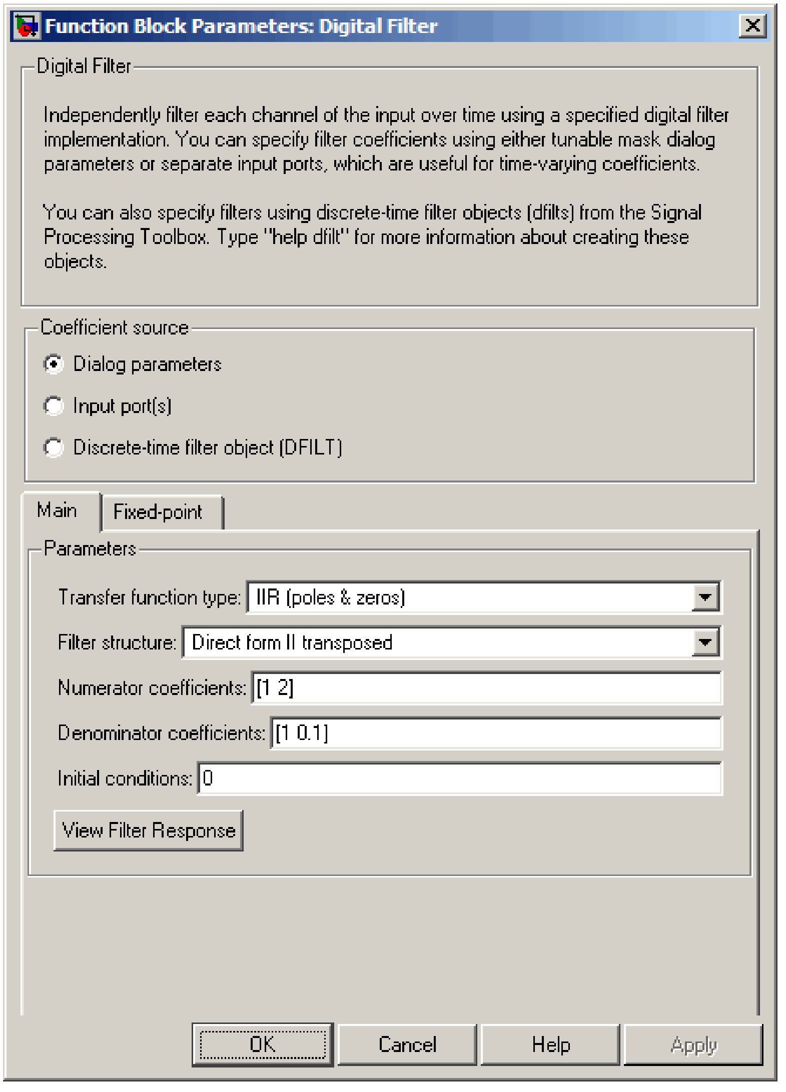
Dialog Box

Different items appear on the Digital Filter block dialog depending on whether you select **Dialog parameters**, **Input port(s)**, or **Discrete-time filter object (DFILT)** in the **Coefficient source** group box. See the following sections for details:

- "Specify Filter Characteristics in Dialog and/or Through Input Ports" on page 2-306
- "Specify Discrete-Time Filter Object" on page 2-317

Specify Filter Characteristics in Dialog and/or Through Input Ports

The **Main** pane of the Digital Filter block dialog appears as follows when **Dialog parameters** is specified in the **Coefficient source** group box. The parameters below can appear when **Dialog parameters** or **Input port(s)** is selected, as noted.



Transfer function type

Select the type of transfer function of the filter; `IIR (poles & zeros)`, `IIR (all poles)`, or `FIR (all zeros)`. See "Supported Filter Structures" on page 2-298 for more information.

Filter structure

Select the filter structure. The selection of available structures varies depending the setting of the **Transfer function type** parameter. See "Supported Filter Structures" on page 2-298 for more information.

Numerator coefficients

Specify the vector of numerator coefficients of the filter's transfer function.

This parameter is only visible when **Dialog parameters** is selected *and* when the selected filter structure lends itself to specification with numerator coefficients. Tunable.

Denominator coefficients

Specify the vector of denominator coefficients of the filter's transfer function.

This parameter is only visible when **Dialog parameters** is selected *and* when the selected filter structure lends itself to specification with denominator coefficients. Tunable.

Reflection coefficients

Specify the vector of reflection coefficients of the filter's transfer function.

This parameter is only visible when **Dialog parameters** is selected *and* when the selected filter structure lends itself to specification with reflection coefficients. Tunable.

SOS matrix (Mx6)

Specify an *M*-by-6 *SOS matrix* containing coefficients of a second-order section (SOS) filter, where *M* is the number of sections. You can use the `ss2sos` and `tf2sos` functions from

Signal Processing Toolbox software to check whether your SOS matrix is valid. For more on the requirements of the SOS matrix, see "Specifying the SOS Matrix (Biquadratic Filter Coefficients)".

This parameter is only visible when **Dialog parameters** is selected *and* when the selected filter structure is biquadratic. Tunable.

Scale values

Specify the scale values to be applied before and after each section of a biquadratic filter.

- If you specify a scalar, that value is applied before the first filter section. The rest of the scale values are set to 1.
- You can also specify a vector with M + 1 elements, assigning a different value to each scale. See "Filter Structure Diagrams" on page 2-320 for diagrams depicting the use of scale values in biquadratic filter structures.

This parameter is only visible when **Dialog parameters** is selected *and* when the selected filter structure is biquadratic. Tunable.

First denominator coefficient = 1, remove a0 term in the structure

Select this parameter to reduce the number of computations the block must make to produce the output by omitting the $1 / a_0$ term in the filter structure. The block output is invalid if you select this parameter when the first denominator filter coefficient is *not* always 1 for your time-varying filter.

This parameter is only enabled when the **Input port(s)** is selected *and* when the selected filter structure lends itself to this specification. See "Removing the a0 Term in the Filter Structure" for a diagram and details.

Coefficient update rate

Specify how often the block updates time-varying filters; once per sample or once per frame. This parameter only affects the output when the input signal is frame based.

This parameter is only visible when the **Input port(s)** is selected *and* when the selected filter structure lends itself to this specification. For more information, see "Specifying Time-Varying Filters".

Initial conditions

Specify the initial conditions of the filter states. To learn how to specify initial conditions, see "Specifying Initial Conditions" on page 2-300.

Initial conditions on zeros side

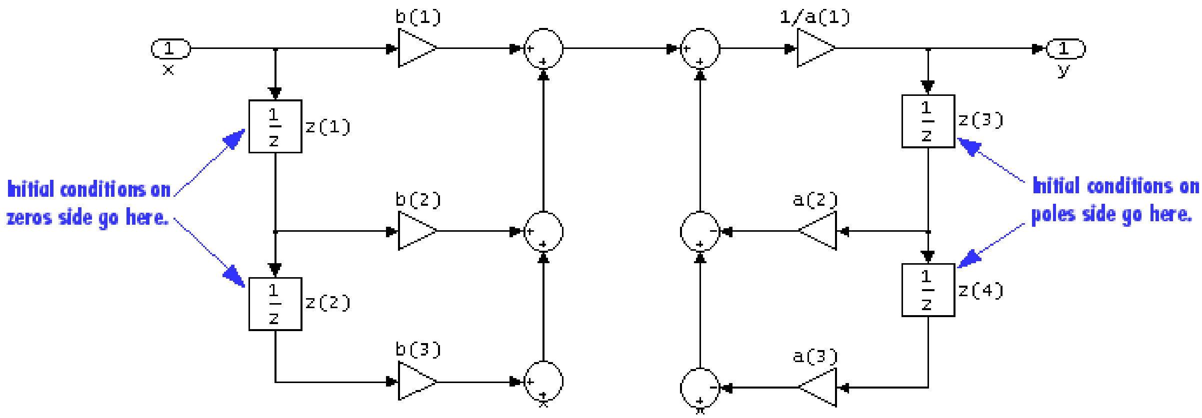
(Not shown in dialog above.) Specify the initial conditions for the filter states on the side of the filter structure with the zeros (b_0, b_1,b_2, ...); see the diagram below.

This parameter is enabled only when the filter has both poles and zeros, *and* when you select a structure such as direct form I, which has separate filter states corresponding to the poles (a_k) and zeros (b_k). To learn how to specify initial conditions, see "Specifying Initial Conditions" on page 2-300.

Initial conditions on poles side

(Not shown in dialog above). Specify the initial conditions for the filter states on the side of the filter structure with the poles (a_0, a_1,a_2, ...); see the diagram below.

This parameter is enabled only when the filter has both poles and zeros, *and* when you select a structure such as direct form I, which has separate filter states corresponding to the poles (a_k) and zeros (b_k). To learn how to specify initial conditions, see "Specifying Initial Conditions" on page 2-300.

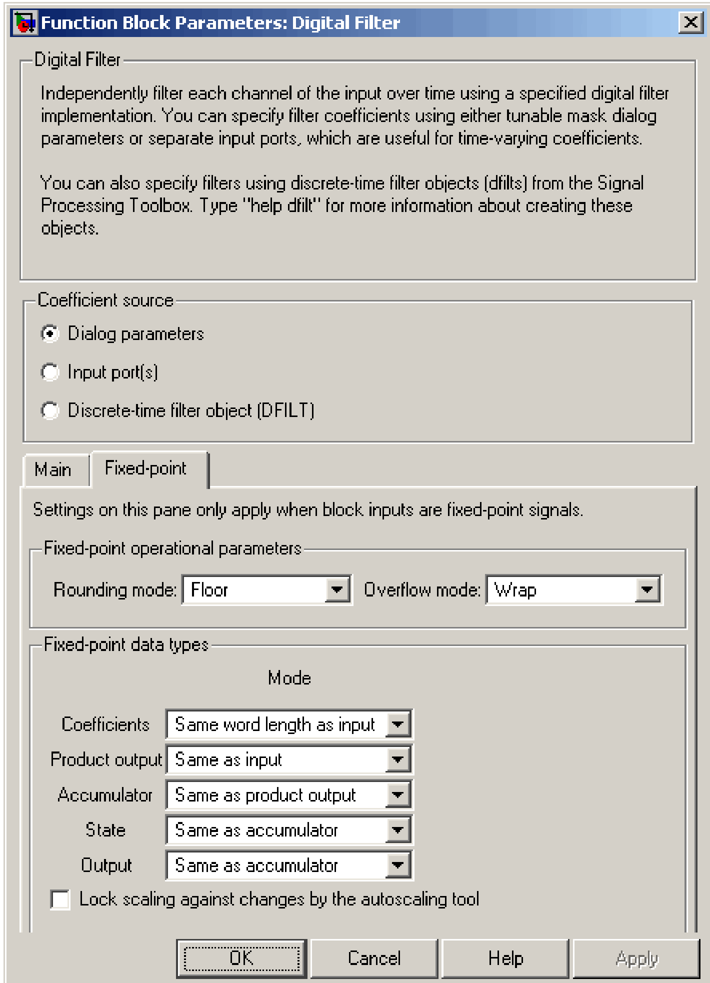


View filter response

This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the filter defined by the block. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Filter** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

The **Fixed point** pane of the Digital Filter block dialog appears as follows when **Dialog parameters** is specified in the **Coefficient source** group box. The parameters below can appear when **Dialog parameters** or **Input port(s)** is selected, depending on the filter structure and whether the coefficients are being entered via ports or on the block mask.



Rounding mode

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

Overflow mode

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

Section I/O

Choose how you specify the word length and the fraction length of the fixed-point data type going into and coming out of each section of a biquadratic filter. See "Filter Structure Diagrams" on page 2-320 for illustrations depicting the use of the section I/O data type in this block.

This parameter is only visible when the selected filter structure is biquadratic:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word and fraction lengths of the section input and output, in bits.
- When you select `Slope and bias scaling`, you can enter the word lengths, in bits, and the slopes of the section input and output. This block requires power-of-two slope and a bias of zero.

Tap sum

Choose how you specify the word length and the fraction length of the tap sum data type of a direct form symmetric or direct form antisymmetric filter. See "Filter Structure Diagrams" on page 2-320 for illustrations depicting the use of the tap sum data type in this block.

This parameter is only visible when the selected filter structure is either `Direct form symmetric` or `Direct form antisymmetric`:

- When you select `Same as input`, these characteristics match those of the input to the block.

- When you select `Binary point scaling`, you can enter the word length and the fraction length of the tap sum accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the tap sum accumulator. This block requires power-of-two slope and a bias of zero.

Multiplicand

Choose how you specify the word length and the fraction length of the multiplicand data type of a direct form I transposed or biquadratic direct form I transposed filter. See "Filter Structure Diagrams" on page 2-320 for illustrations depicting the use of the multiplicand data type in this block.

This parameter is only visible when the selected filter structure is either `Direct form I transposed` or `Biquad direct form I transposed (SOS)`:

- When you select `Same as output`, these characteristics match those of the output to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the multiplicand data type, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the multiplicand data type. This block requires power-of-two slope and a bias of zero.

Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients (numerator and/or denominator). See "Filter Structure Diagrams" on page 2-320 for illustrations depicting the use of the coefficient data types in this block:

- When you select `Same word length as input`, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.

- When you select `Specify word length`, you can enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the coefficients, in bits. If applicable, you can enter separate fraction lengths for the numerator and denominator coefficients.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the coefficients. If applicable, you can enter separate slopes for the numerator and denominator coefficients. This block requires power-of-two slope and a bias of zero.
- The filter coefficients do not obey the **Rounding mode** and the **Overflow mode** parameters; they are always saturated and rounded to `Nearest`.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See “Filter Structure Diagrams” on page 2-320 and “Multiplication Data Types” for illustrations depicting the use of the product output data type in this block:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Filter Structure Diagrams" on page 2-320 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

State

Use this parameter to specify how you would like to designate the state word and fraction lengths. See "Filter Structure Diagrams" on page 2-320 for illustrations depicting the use of the state data type in this block.

This parameter is not visible for direct form and direct form I filter structures.

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Same as accumulator`, these characteristics match those of the accumulator.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

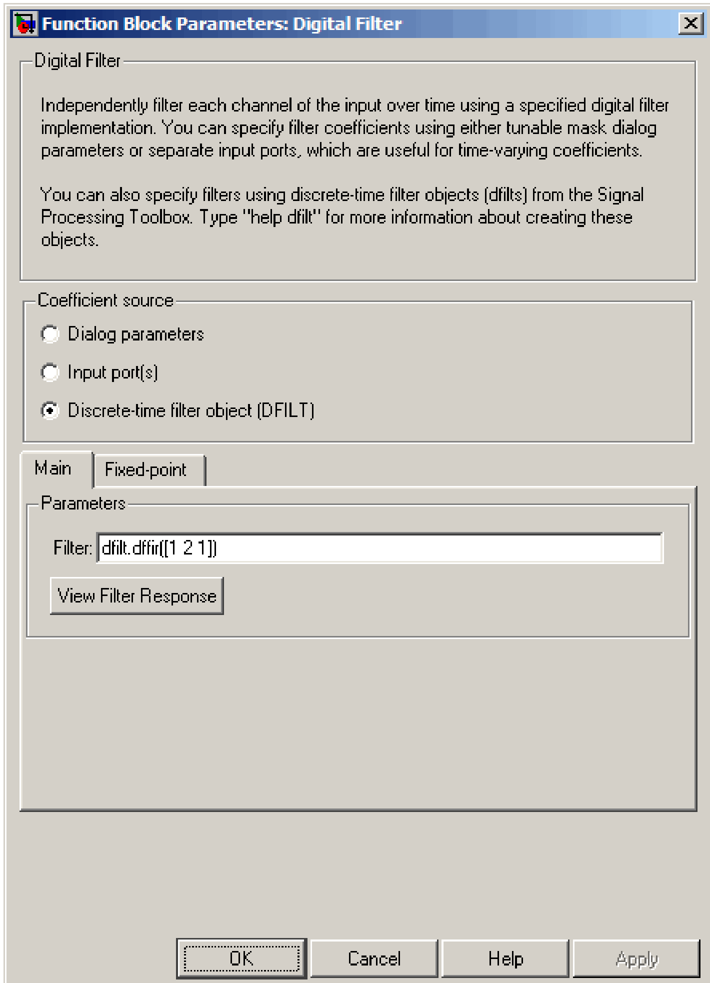
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Same as accumulator`, these characteristics match those of the accumulator.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output,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

Lock scaling against changes by the autoscaling tool

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

Specify Discrete-Time Filter Object

The **Main** pane of the Digital Filter block dialog appears as follows when **Discrete-time filter object (DFILT)** is specified in the **Coefficient source** group box.



Filter

Specify the discrete-time filter object (`dfilt`) that you would like the block to implement. You can do this in one of three ways:

- You can fully specify the `dfilt` object in the block mask, as shown in the default value.
- You can enter the variable name of a `dfilt` object that is defined in any workspace.
- You can enter a variable name for a `dfilt` object that is not yet defined.

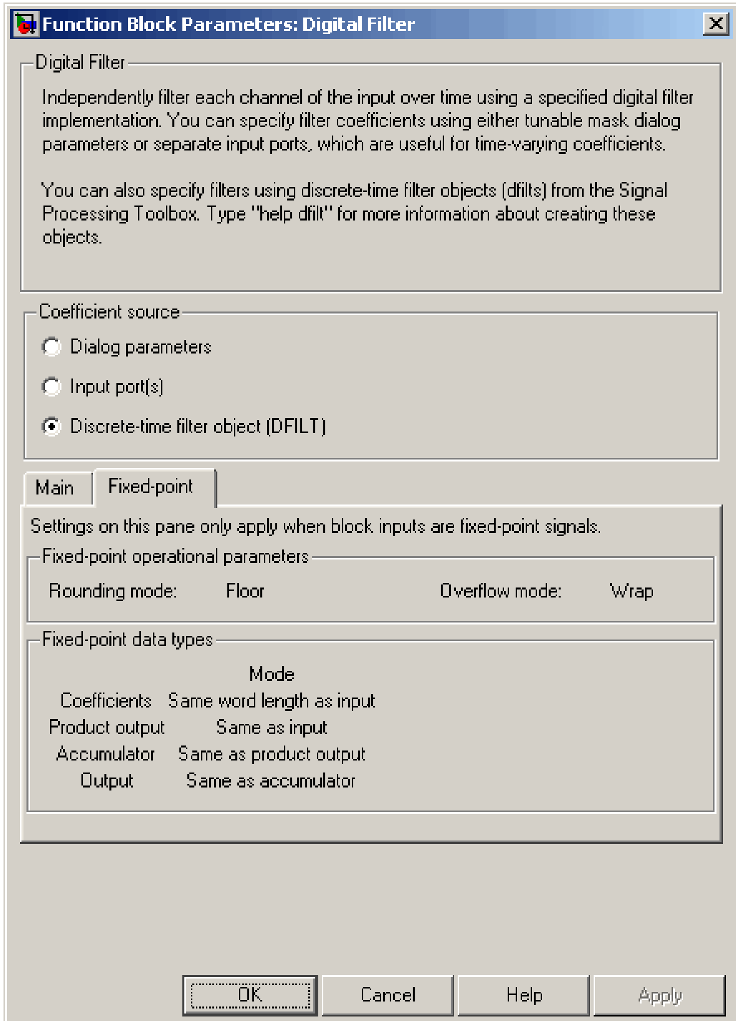
For more information on creating `dfilt` objects, see the `dfilt` function reference page in the Signal Processing Toolbox documentation or the Filter Design Toolbox documentation.

View filter response

This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the `dfilt` object specified in the **Filter** parameter. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Filter** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

The **Fixed-point** pane of the Digital Filter block dialog appears as follows when **Discrete-time filter object (DFILT)** is specified in the **Coefficient source** group box.



The fixed-point settings of the filter object specified on the **Main** pane are displayed on the **Fixed-point** pane. You cannot change these settings directly on the block mask. To change the fixed-point settings you must edit the filter object directly.

For more information on discrete-time filter objects, see the `dfilt` function reference page in the Signal Processing Toolbox documentation or the Filter Design Toolbox documentation.

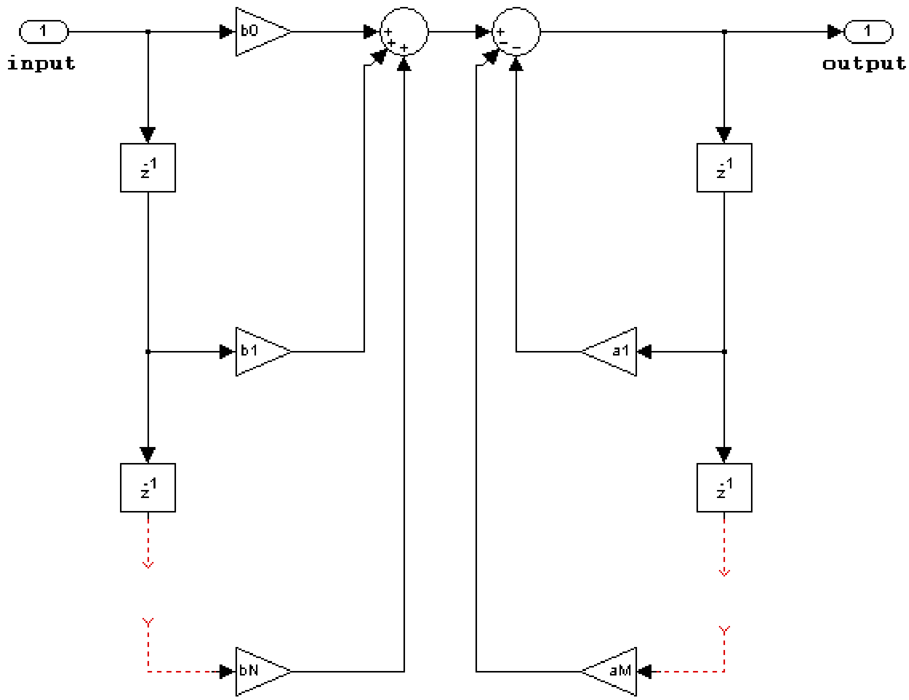
Filter Structure Diagrams

The diagrams in the following sections show the filter structures supported by the Digital Filter block. They also show the data types used in the filter structures for fixed-point signals. You can set the coefficient, output, accumulator, product output, and state data types shown in these diagrams in the block dialog. This is discussed in "Dialog Box" on page 2-305.

- "IIR direct form I" on page 2-321
- "IIR direct form I transposed" on page 2-323
- "IIR direct form II" on page 2-326
- "IIR direct form II transposed" on page 2-328
- "IIR biquadratic direct form I" on page 2-331
- "IIR biquadratic direct form I transposed" on page 2-334
- "IIR biquadratic direct form II" on page 2-337
- "IIR biquadratic direct form II transposed" on page 2-339
- "IIR (all poles) direct form" on page 2-342
- "IIR (all poles) direct form transposed" on page 2-344
- "IIR (all poles) direct form lattice AR" on page 2-346
- "FIR (all zeros) direct form" on page 2-347
- "FIR (all zeros) direct form symmetric" on page 2-349
- "FIR (all zeros) direct form antisymmetric" on page 2-351

- "FIR (all zeros) direct form transposed" on page 2-353
- "FIR (all zeros) lattice MA" on page 2-355

IIR direct form I

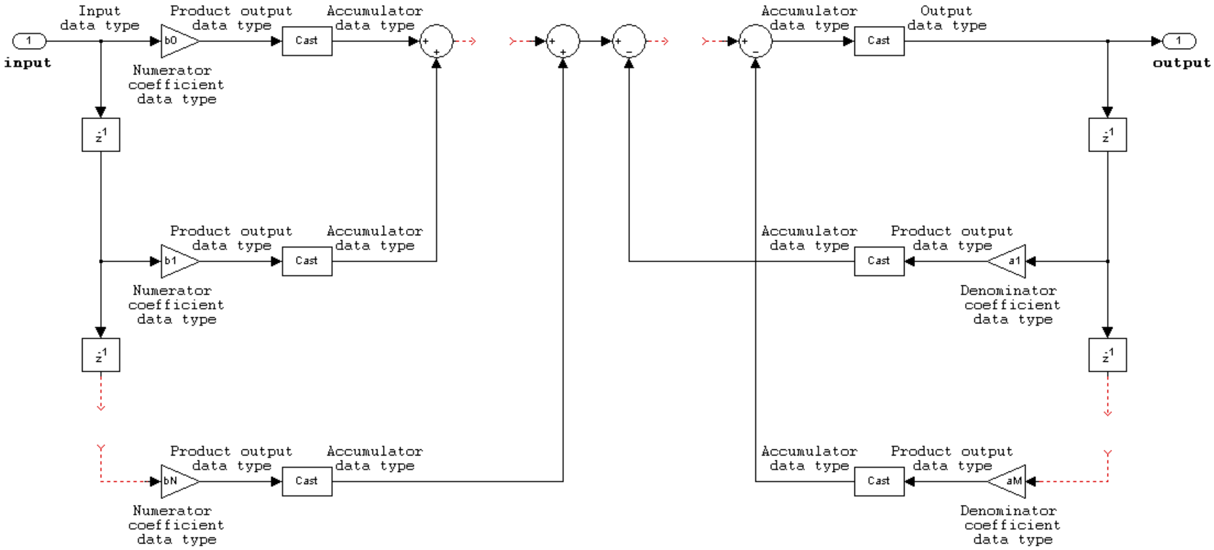


The following constraints are applicable when processing a fixed-point signal with this filter structure:

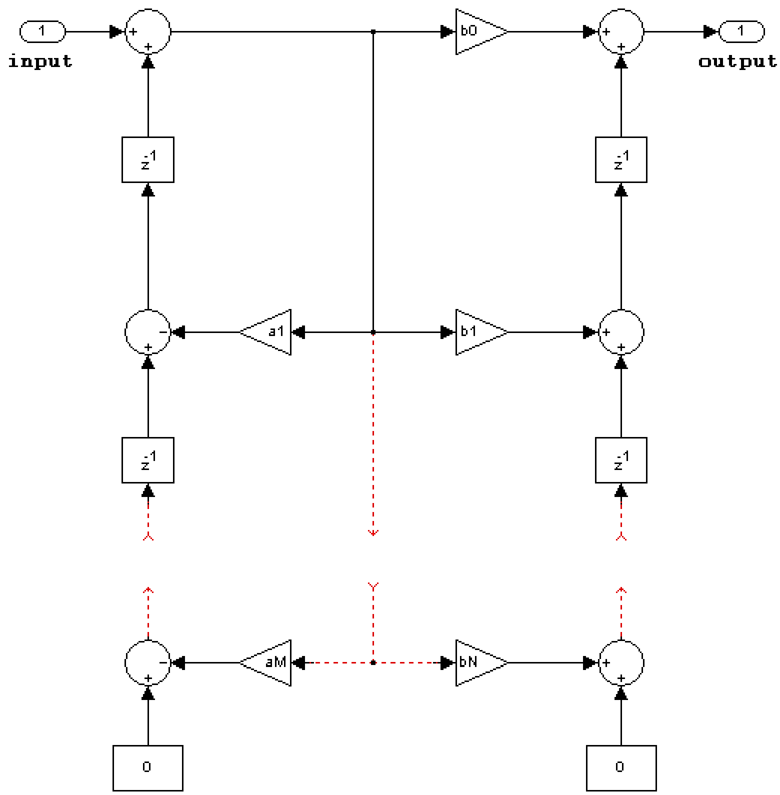
- Inputs can be real or complex.
- Numerator and denominator coefficients can be real or complex.
- Numerator and denominator coefficients must be the same complexity as each other.

- When the numerator and denominator coefficients are specified via input ports and have different complexities from each other, you get an error.
- When the numerator and denominator coefficients are specified in the dialog and have different complexities from each other, the block does not error, but instead processes the filter as if two sets of complex coefficients are provided. The coefficient set that is real-valued is treated as if it is a complex vector with zero-valued imaginary parts.

- Numerator and denominator coefficients must have the same word length. They can have different fraction lengths.
- The State data type cannot be specified on the block mask for this structure, because the input and output states have the same data types as the input and output buffers.



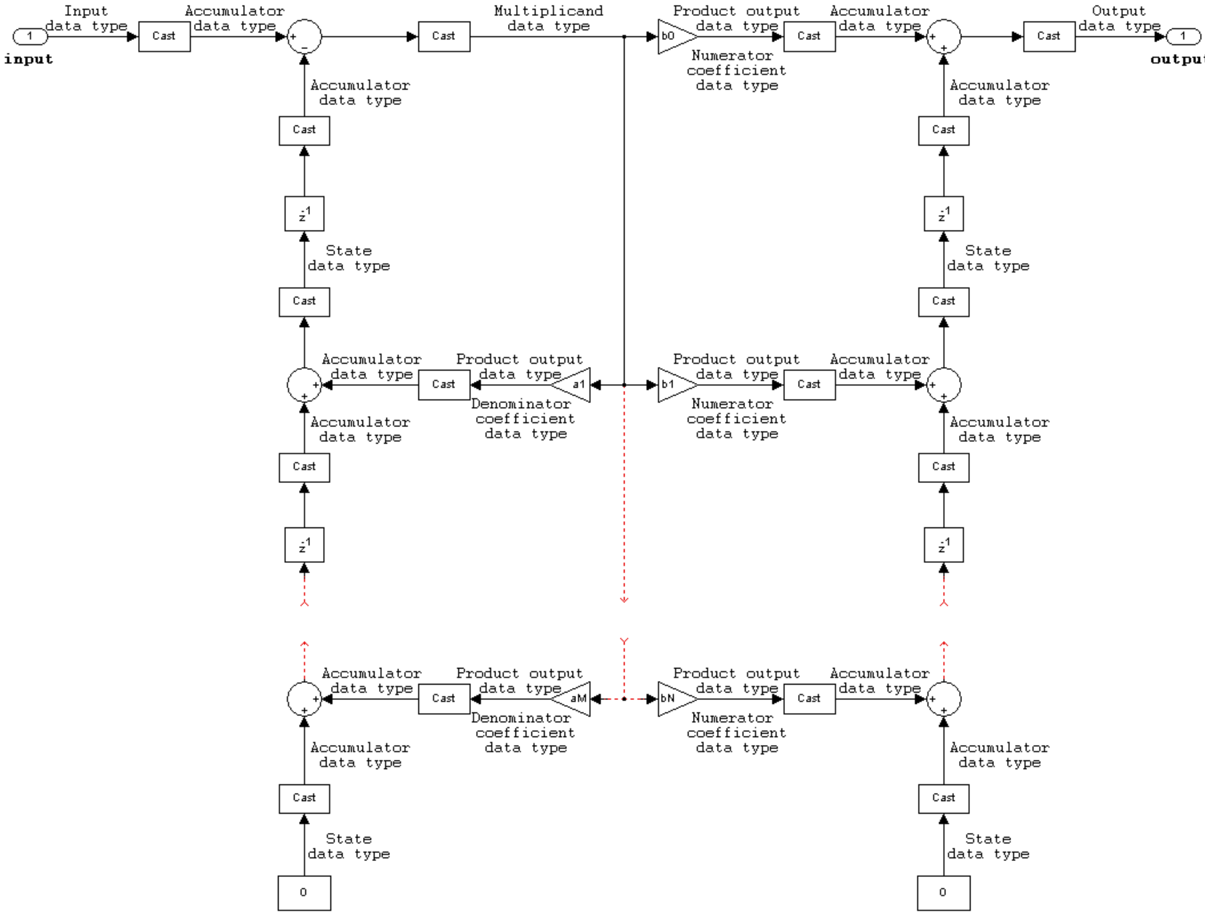
IIR direct form I transposed



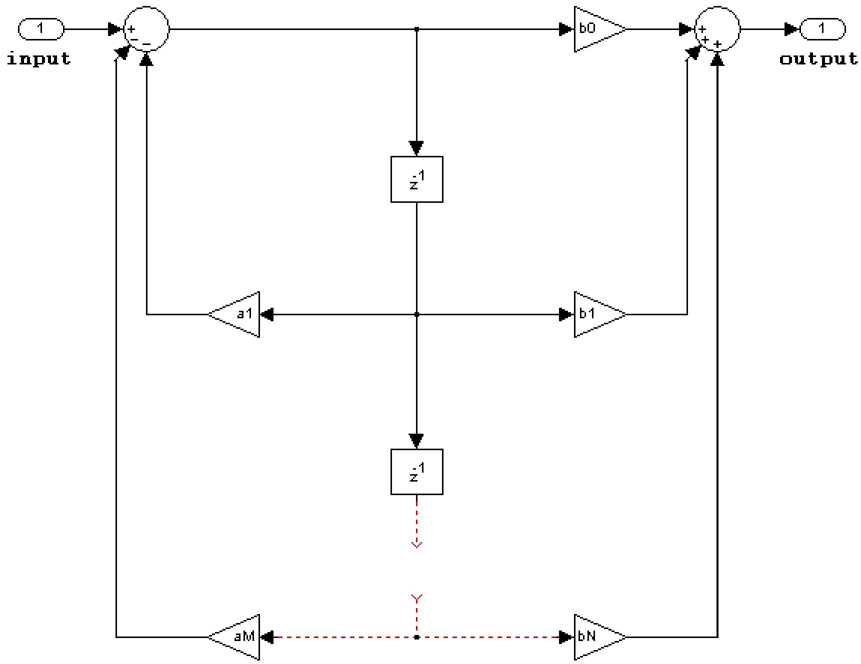
The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs can be real or complex.
- Numerator and denominator coefficients can be real or complex.
- Numerator and denominator coefficients must be the same complexity as each other.

  - When the numerator and denominator coefficients are specified via input ports and have different complexities from each other, you get an error.
  - When the numerator and denominator coefficients are specified in the dialog and have different complexities from each other, the block does not error, but instead processes the filter as if two sets of complex coefficients are provided. The coefficient set that is real-valued is treated as if it is a complex vector with zero-valued imaginary parts.
- States are complex when either the input or the coefficients are complex.
- Numerator and denominator coefficients must have the same word length. They can have different fraction lengths.


Input
data type
Accumulator
data type
Multiplicand
data type
Product output
data type
Accumulator
data type
Output
data type
input
output
Cast
Numerator
coefficient
data type
State
data type
Denominator
coefficient
data type
b0
b1
bN
a1
aM
0

IIR direct form II

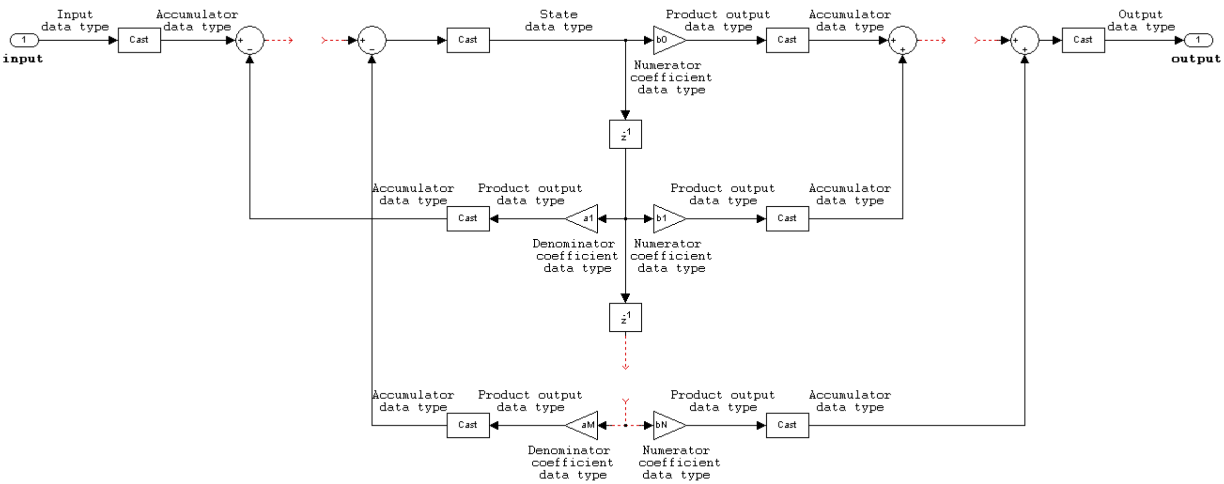


The following constraints are applicable when processing a fixed-point signal with this filter structure:

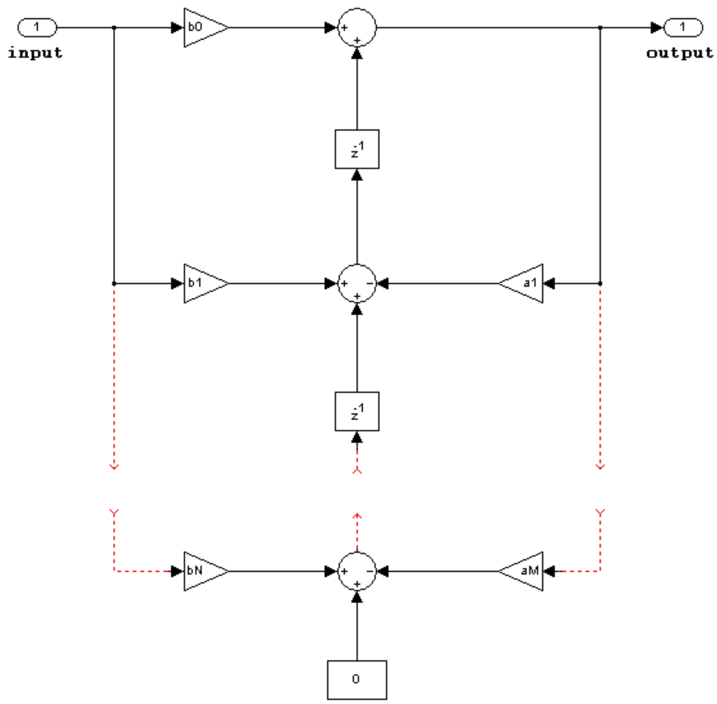
- Inputs can be real or complex.
- Numerator and denominator coefficients can be real or complex.
- Numerator and denominator coefficients must be the same complexity as each other.
  - When the numerator and denominator coefficients are specified via input ports and have different complexities from each other, you get an error.

- When the numerator and denominator coefficients are specified in the dialog and have different complexities from each other, the block does not error, but instead processes the filter as if two sets of complex coefficients are provided. The coefficient set that is real-valued is treated as if it is a complex vector with zero-valued imaginary parts.

- States are complex when either the inputs or the coefficients are complex.
- Numerator and denominator coefficients must have the same word length. They can have different fraction lengths.



IIR direct form II transposed

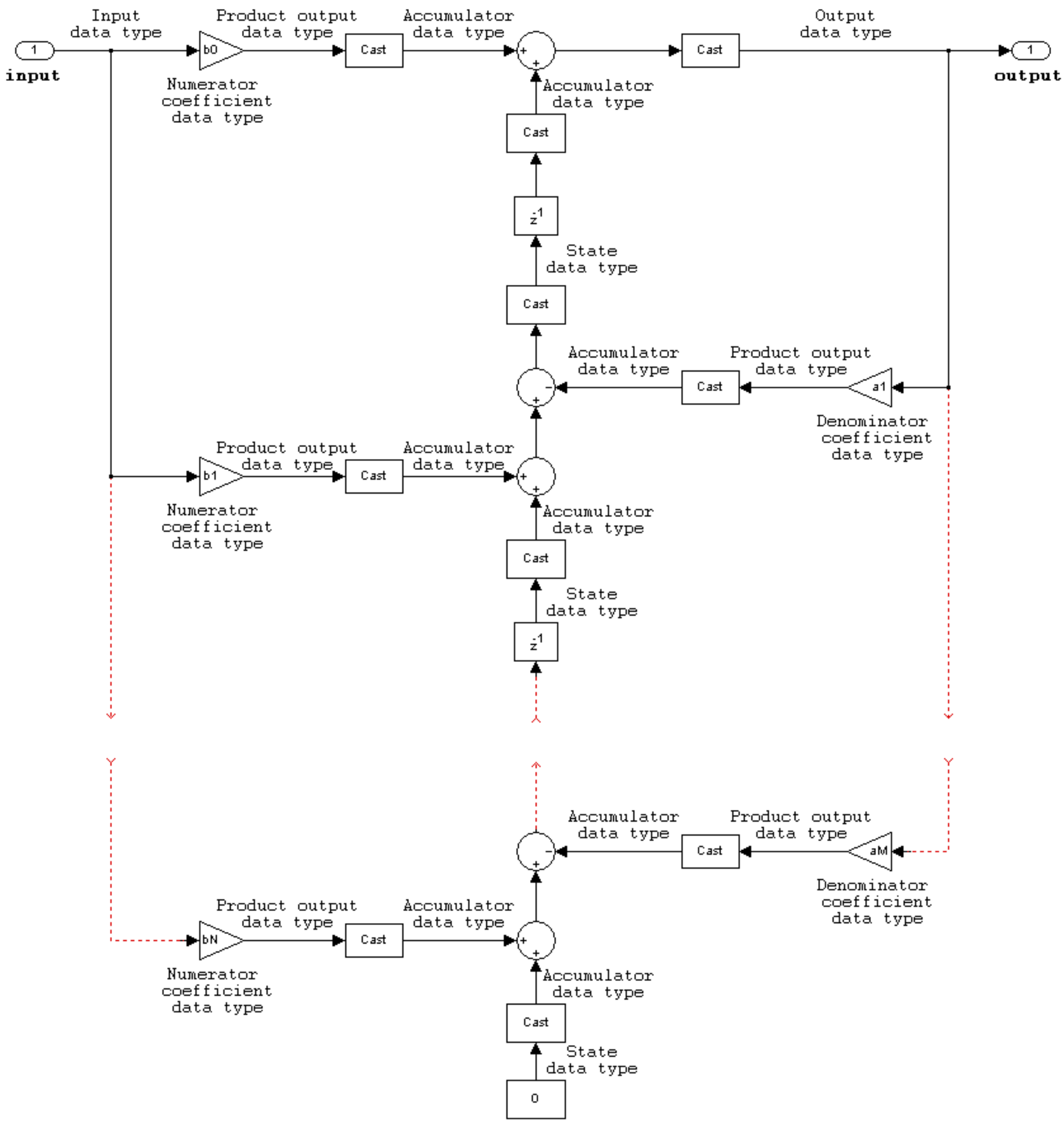


The following constraints are applicable when processing a fixed-point signal with this filter structure:

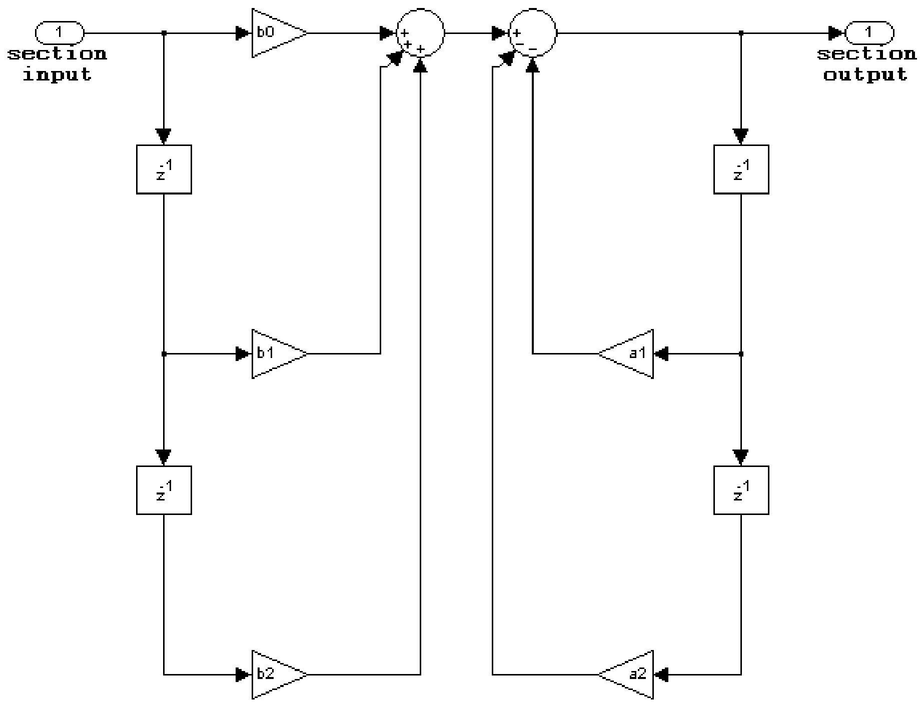
- Inputs can be real or complex.
- Numerator and denominator coefficients can be real or complex.
- Numerator and denominator coefficients must be the same complexity as each other.
  - When the numerator and denominator coefficients are specified via input ports and have different complexities from each other, you get an error.

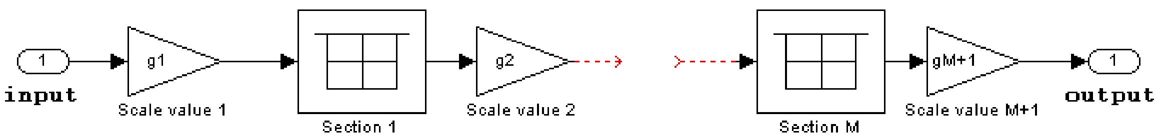
  - When the numerator and denominator coefficients are specified in the dialog and have different complexities from each other, the block does not error, but instead processes the filter as if two sets of complex coefficients are provided. The coefficient set that is real-valued is treated as if it is a complex vector with zero-valued imaginary parts.

- States are complex when either the inputs or the coefficients are complex.
- Numerator and denominator coefficients must have the same word length. They can have different fraction lengths.


1
input
Input
data type
b0
Product output
data type
Cast
Accumulator
data type
+
+
Numerator
coefficient
data type
Cast
Output
data type
1
output
Accumulator
data type
Cast
z-1
State
data type
Cast
Accumulator
data type
Product output
data type
a1
Denominator
coefficient
data type
b1
bN
aM
0

IIR biquadratic direct form I



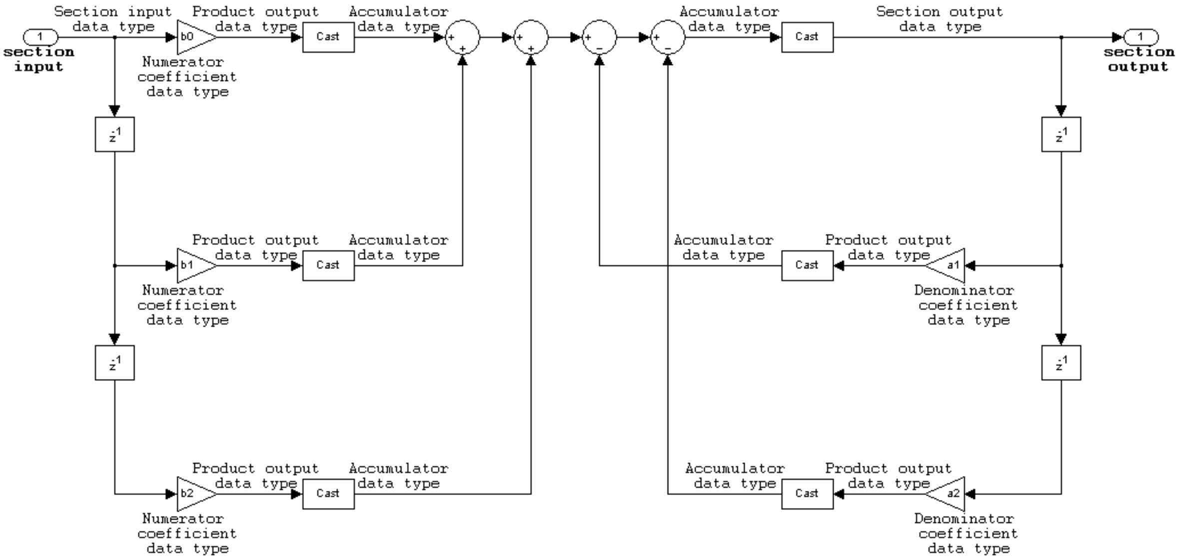


The following constraints are applicable when processing a fixed-point signal with this filter structure:

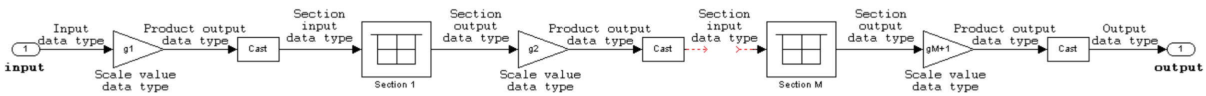
- Inputs and coefficients can be real or complex.
- Numerator and denominator coefficients can be real or complex.

- Specify the coefficients by a M-by-6 matrix in the block mask. You cannot specify coefficients by input ports for this filter structure.
- When the a0 element of any row is not equal to one, that row is normalized by a0 prior to filtering.
- States are complex when either the inputs or the coefficients are complex.
- You cannot specify the state data type on the block mask for this structure, because the input and output states have the same data types as the input.
- Scale values must have the same complexity as the coefficient SOS matrix.
- The scale value parameter must be a scalar or a vector of length M+1, where M is the number of sections.
- The **Section I/O** parameter determines the data type for the section input and output data types. The section input and stage output data type must have the same word length but can have different fraction lengths.

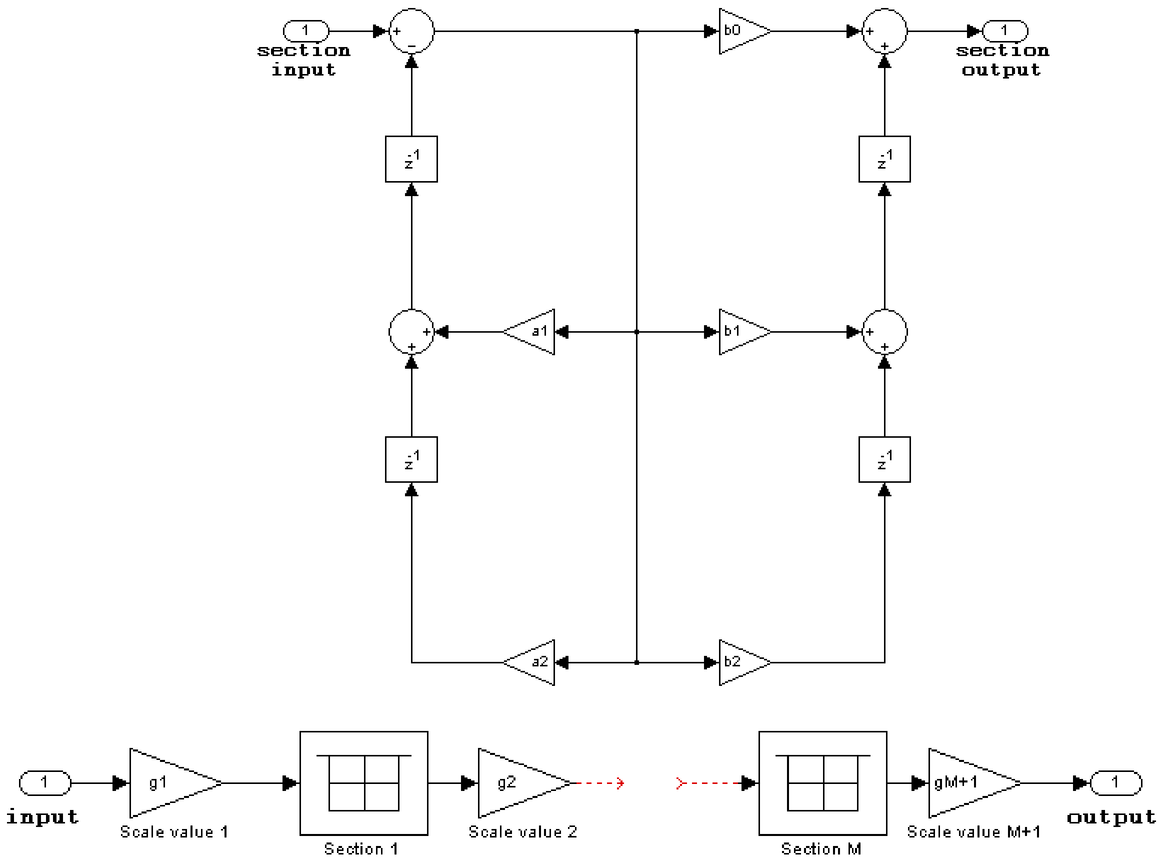
The following diagram shows the data types for one section of the filter.



The following diagram shows the data types between filter sections.



IIR biquadratic direct form I transposed

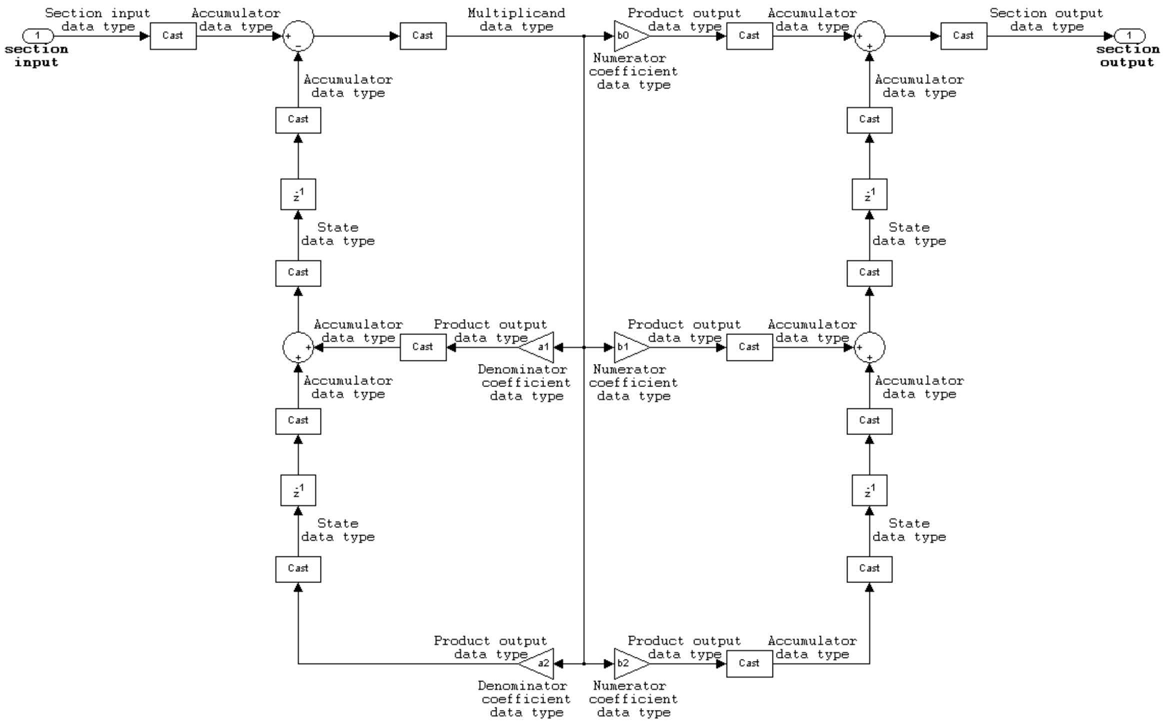


The following constraints are applicable when processing a fixed-point signal with this filter structure:

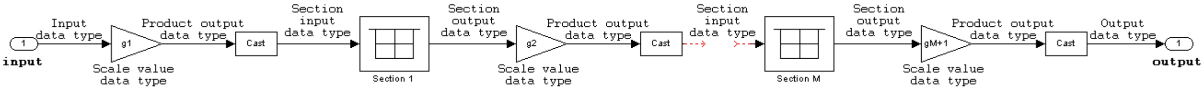
- Inputs and coefficients can be real or complex.
- Numerator and denominator coefficients can be real or complex.

- Specify the coefficients by a M-by-6 matrix in the block mask. You cannot specify coefficients by input ports for this filter structure.
- When the a0 element of any row is not equal to one, that row is normalized by a0 prior to filtering.
- States are complex when either the inputs or the coefficients are complex.
- Scale values must have the same complexity as the coefficient SOS matrix.
- The scale value parameter must be a scalar or a vector of length M+1, where M is the number of sections.
- The **Section I/O** parameter determines the data type for the section input and output data types. The section input and section output data type must have the same word length but can have different fraction lengths.

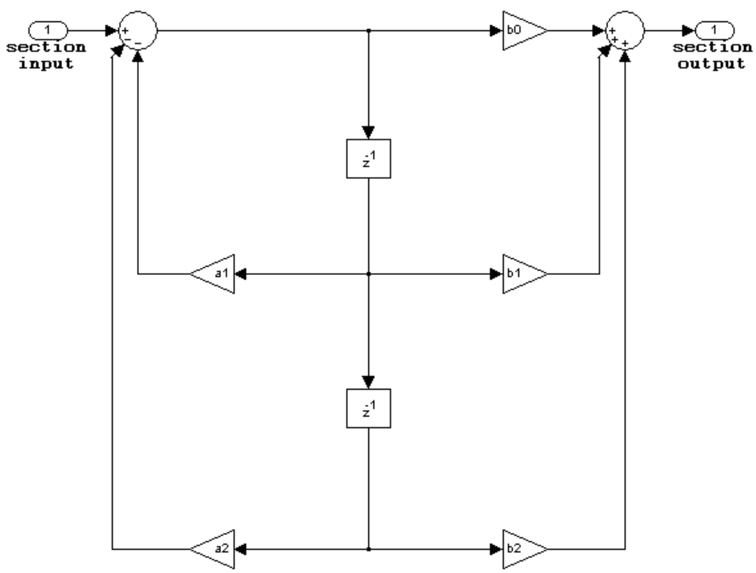
The following diagram shows the data types for one section of the filter.

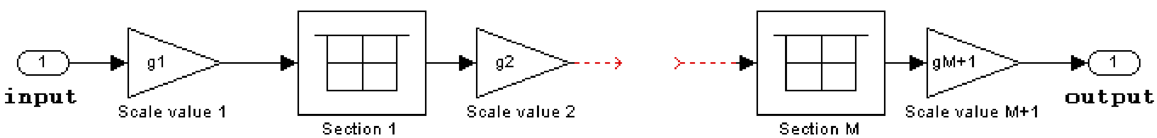


The following diagram shows the data types between filter sections.



IIR biquadratic direct form II



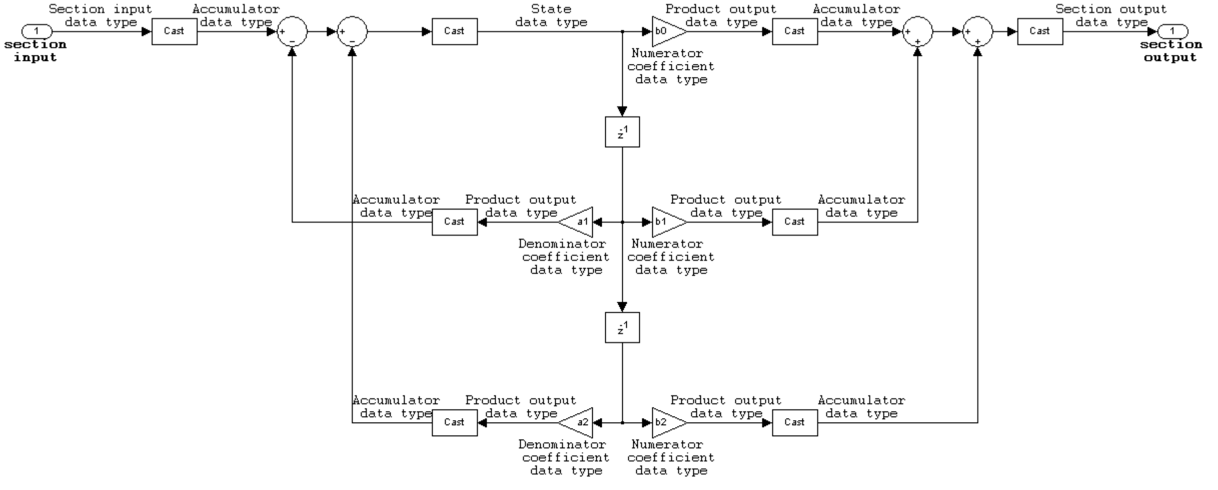


The following constraints are applicable when processing a fixed-point signal with this filter structure:

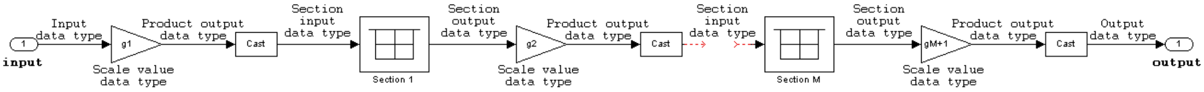
- Inputs and coefficients can be real or complex.
- Numerator and denominator coefficients can be real or complex.

- Specify the coefficients by a M-by-6 matrix in the block mask. You cannot specify coefficients by input ports for this filter structure.
- When the a0 element of any row is not equal to one, that row is normalized by a0 prior to filtering.
- States are complex when either the inputs or the coefficients are complex.
- Scale values must have the same complexity as the coefficient SOS matrix.
- The scale value parameter must be a scalar or a vector of length M+1, where M is the number of sections.
- The **Section I/O** parameter determines the data type for the section input and output data types. The section input and section output data type must have the same word length but can have different fraction lengths.

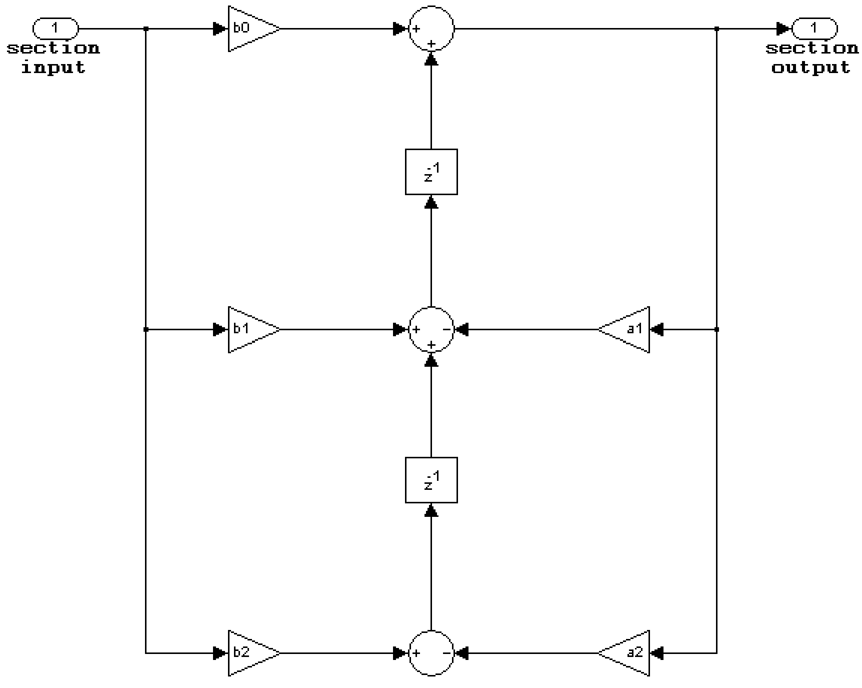
The following diagram shows the data types for one section of the filter.

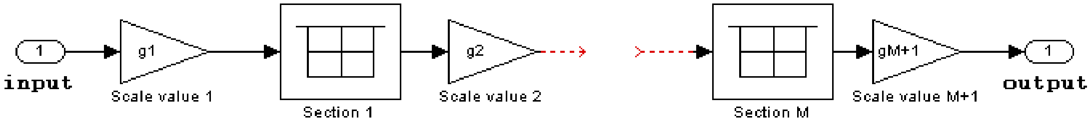


The following diagram shows the data types between filter sections.



IIR biquadratic direct form II transposed

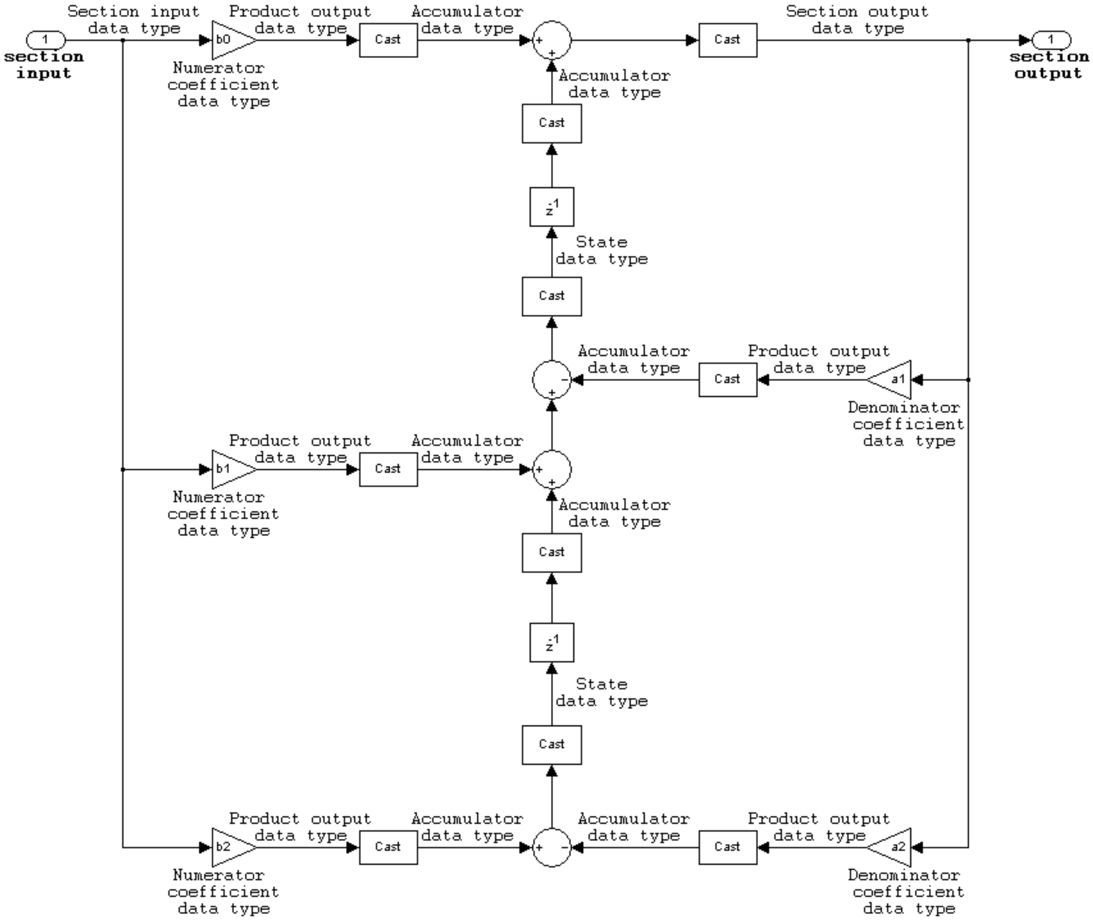




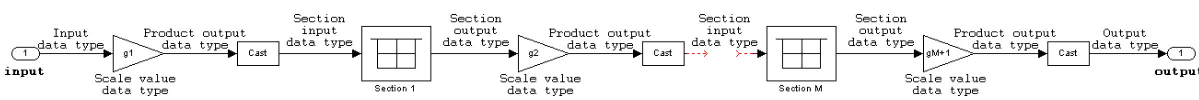
The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs and coefficients can be real or complex.
- Numerator and denominator coefficients can be real or complex.
- Specify the coefficients by a M-by-6 matrix in the block mask. You cannot specify coefficients by input ports for this filter structure.
- When the a0 element of any row is not equal to one, that row is normalized by a0 prior to filtering.
- States are complex when either the inputs or the coefficients are complex.
- Scale values must have the same complexity as the coefficient SOS matrix.
- The scale value parameter must be a scalar or a vector of length M+1, where M is the number of sections.
- The **Section I/O** parameter determines the data type for the section input and output data types. The section input and section output data type must have the same word length but can have different fraction lengths.

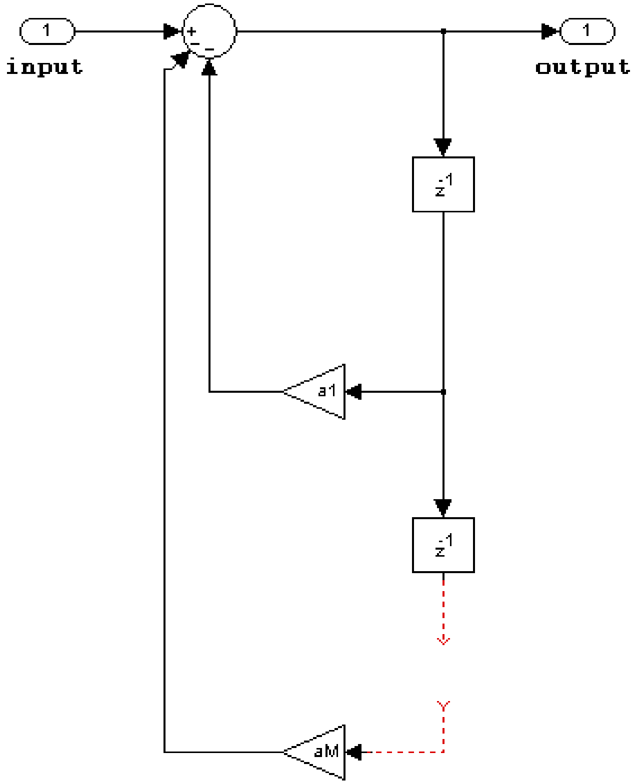
The following diagram shows the data types for one section of the filter.



The following diagram shows the data types between filter sections.

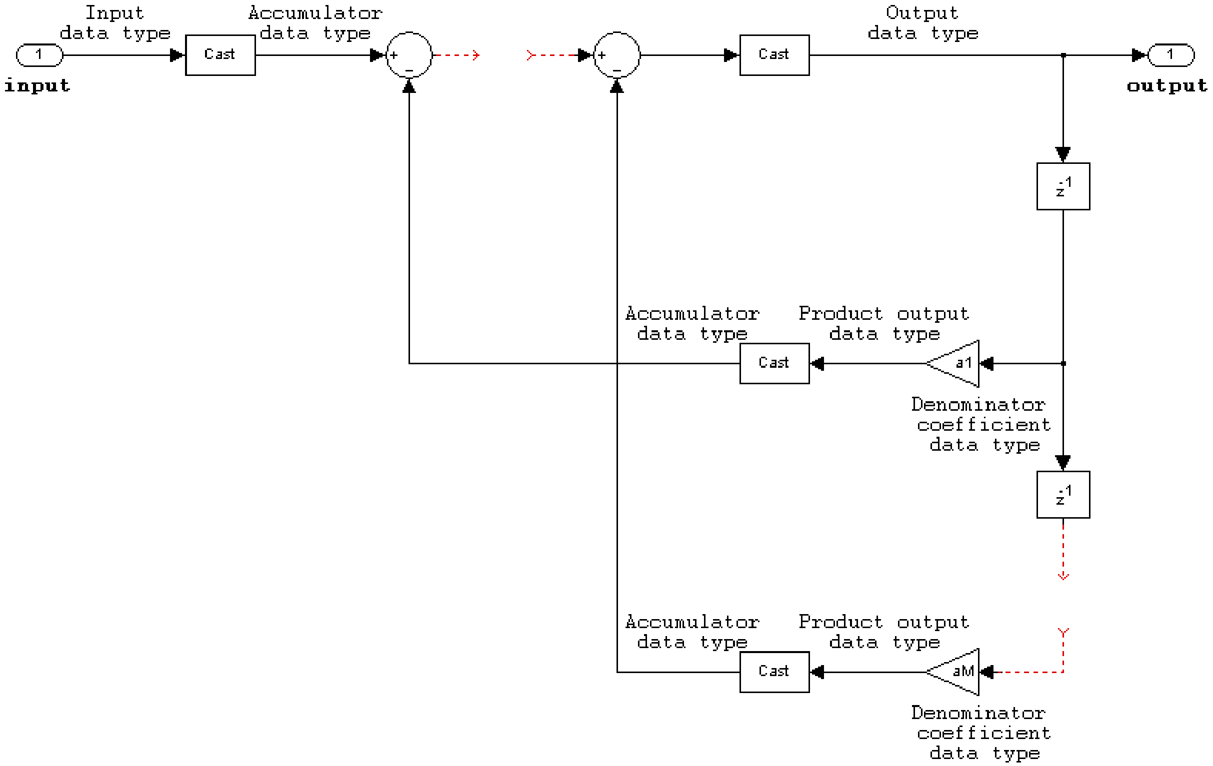


IIR (all poles) direct form

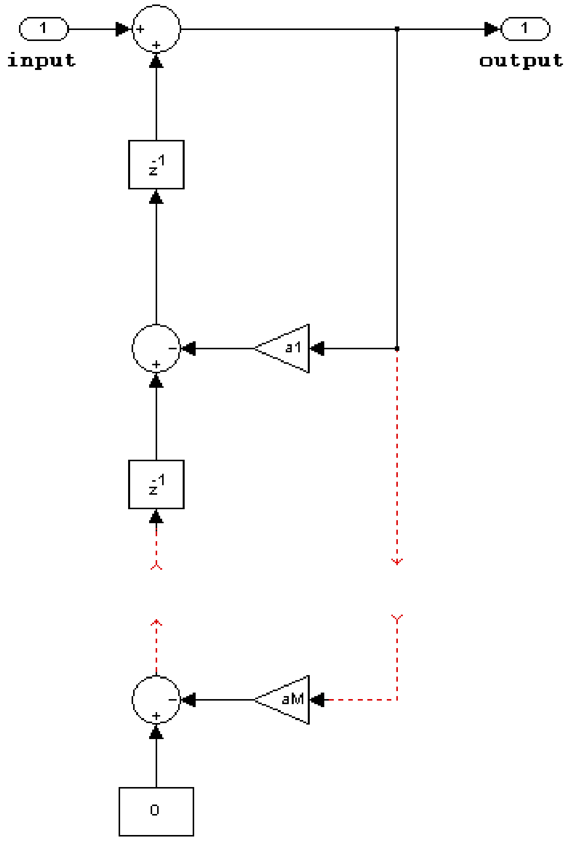


The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs and coefficients can be real or complex.
- Denominator coefficients can be real or complex.
- You cannot specify the state data type on the block mask for this structure, because the input and output states have the same data types as the input.

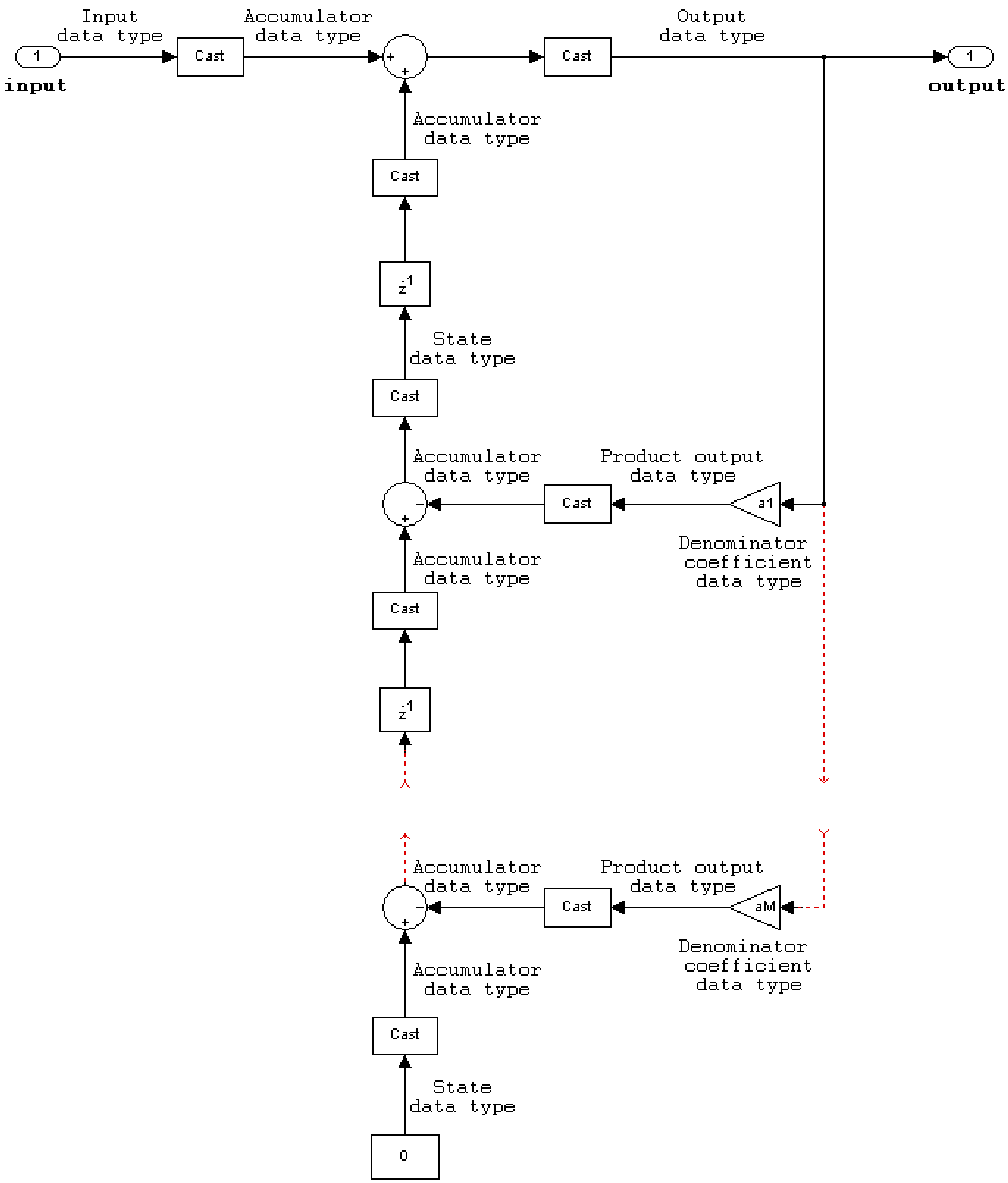

Input
data type
Accumulator
data type
Output
data type
Cast
Cast
1
input
1
output
z-1
Accumulator
data type
Product output
data type
Cast
a1
Denominator
coefficient
data type
z-1
Accumulator
data type
Product output
data type
Cast
aM
Denominator
coefficient
data type

IIR (all poles) direct form transposed

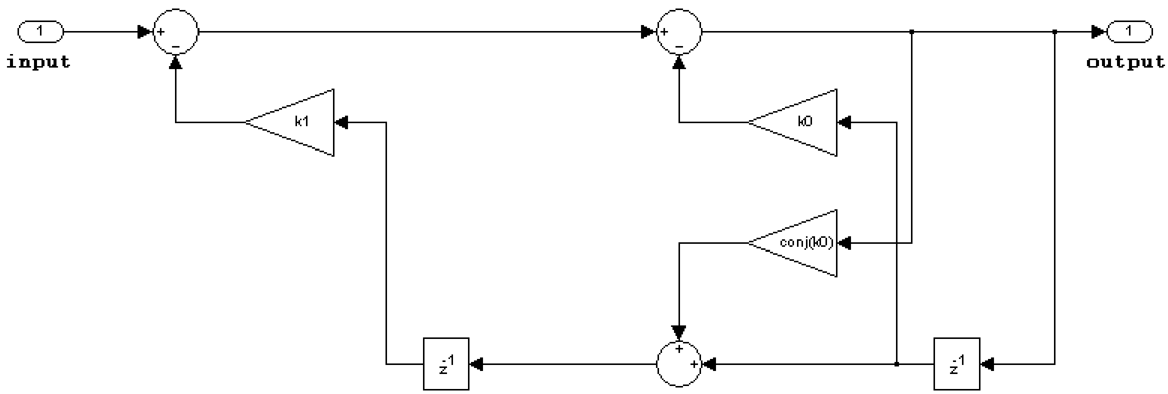


The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs and coefficients can be real or complex.
- Denominator coefficients can be real or complex.

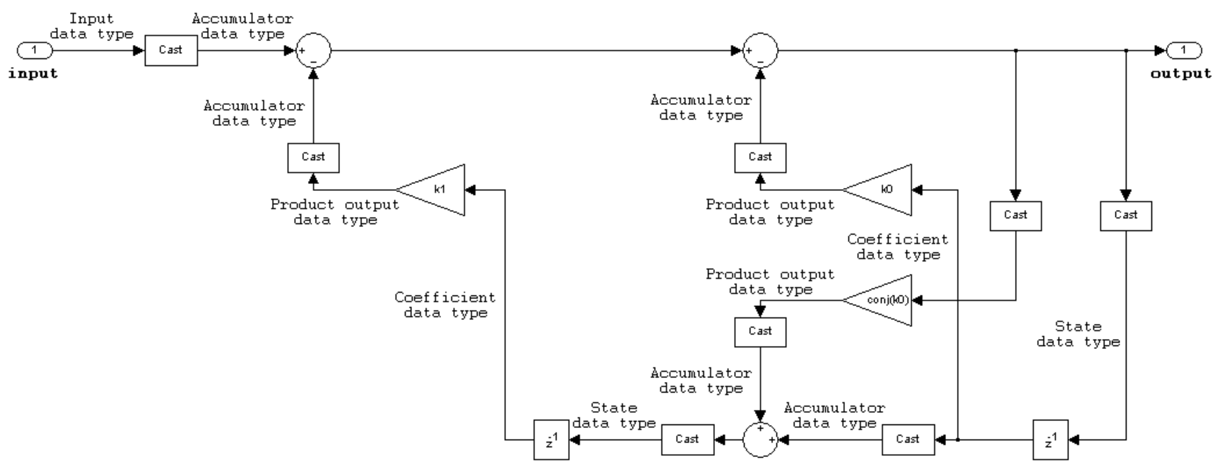

Input
data type
Accumulator
data type
Output
data type
1
input
Cast
Cast
1
output
Accumulator
data type
Cast
z-1
State
data type
Cast
Accumulator
data type
Product output
data type
Cast
a1
Denominator
coefficient
data type
Accumulator
data type
Cast
z-1
Accumulator
data type
Product output
data type
Cast
aM
Denominator
coefficient
data type
Accumulator
data type
Cast
State
data type
0

IIR (all poles) direct form lattice AR

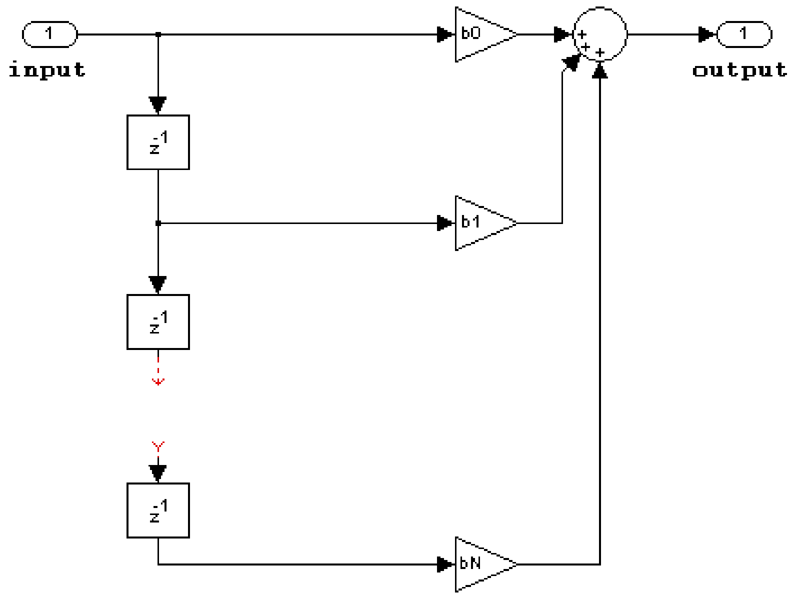


The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs and coefficients can be real or complex.
- Coefficients can be real or complex.

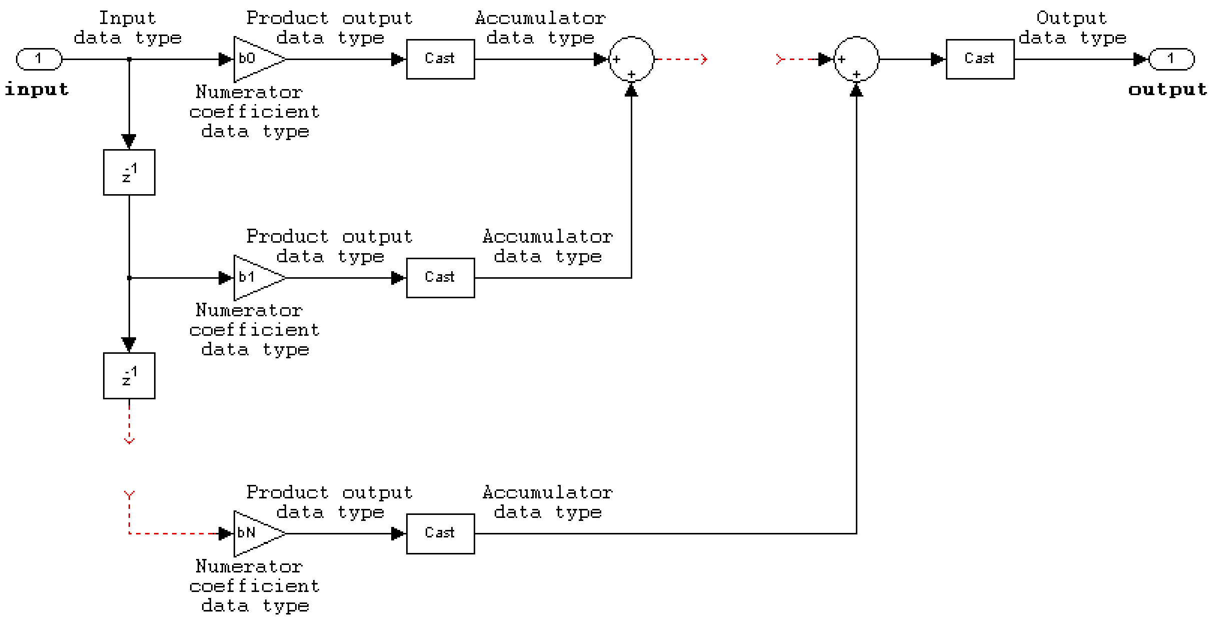


FIR (all zeros) direct form



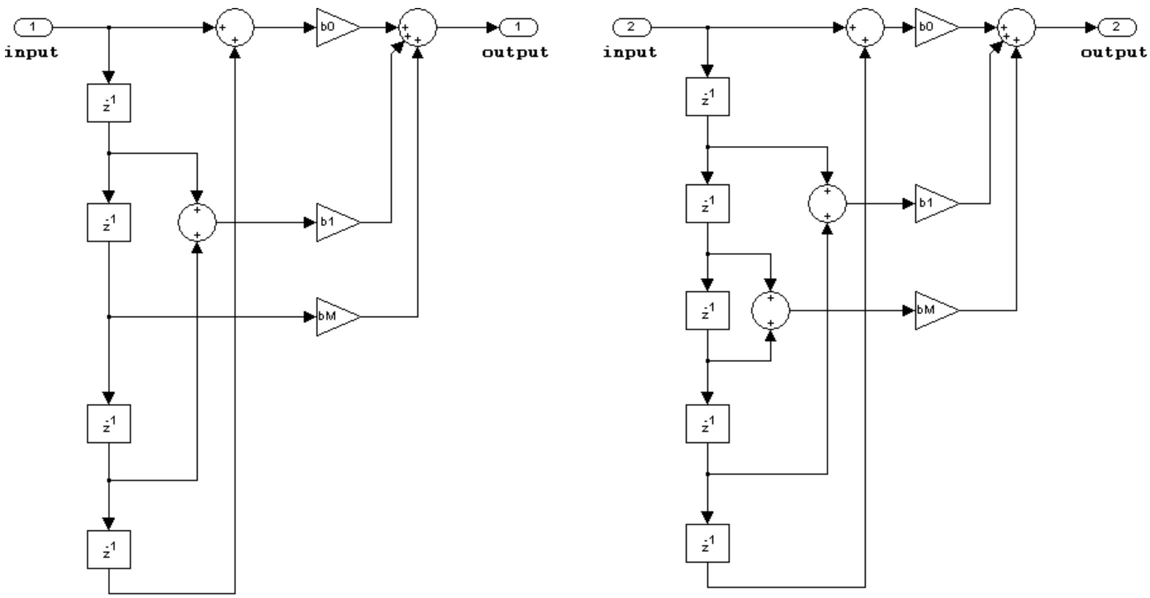
The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs can be real or complex.
- Numerator coefficients can be real or complex.
- You cannot specify the state data type on the block mask for this structure, because the input and output states have the same data types as the input.



1
input
Input
data type
b0
Product output
data type
Cast
Accumulator
data type
Numerator
coefficient
data type
z-1
b1
Product output
data type
Cast
Accumulator
data type
Numerator
coefficient
data type
z-1
bN
Product output
data type
Cast
Accumulator
data type
Numerator
coefficient
data type
Cast
Output
data type
1
output

FIR (all zeros) direct form symmetric

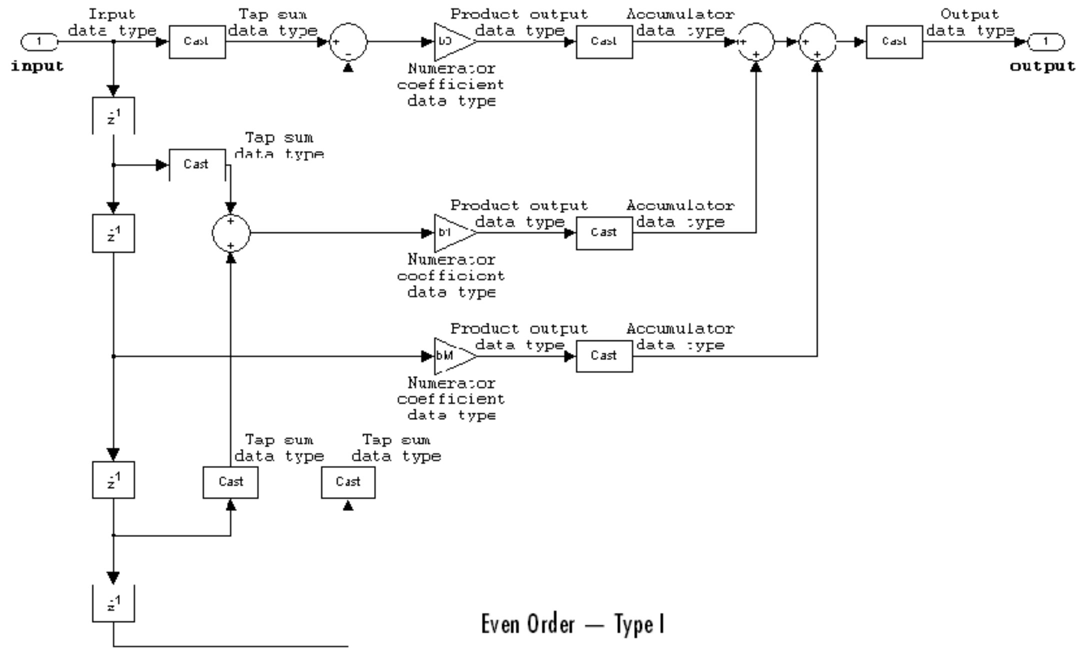


Even Order — Type I

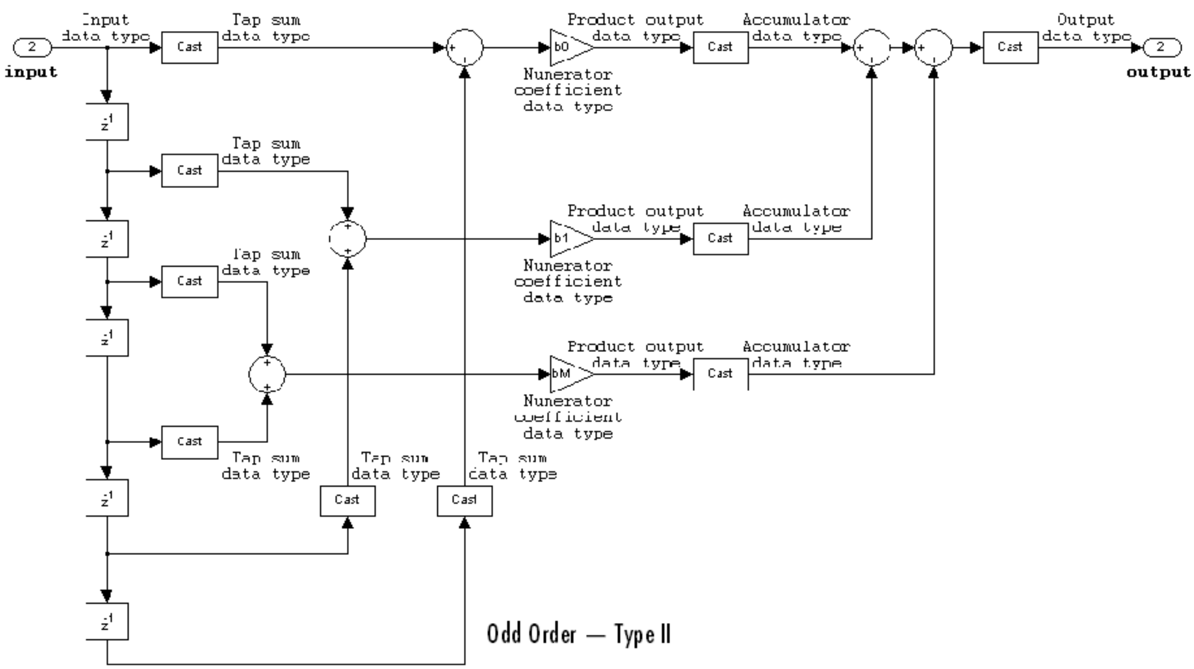
Odd Order — Type II

The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs can be real or complex.
- Numerator coefficients can be real or complex.
- You cannot specify the state data type on the block mask for this structure, because the input and output states have the same data types as the input.
- It is assumed that the filter coefficients are symmetric. Only the first half of the coefficients are used for filtering.
- The **Tap Sum** parameter determines the data type the filter uses when it sums the inputs prior to multiplication by the coefficients.

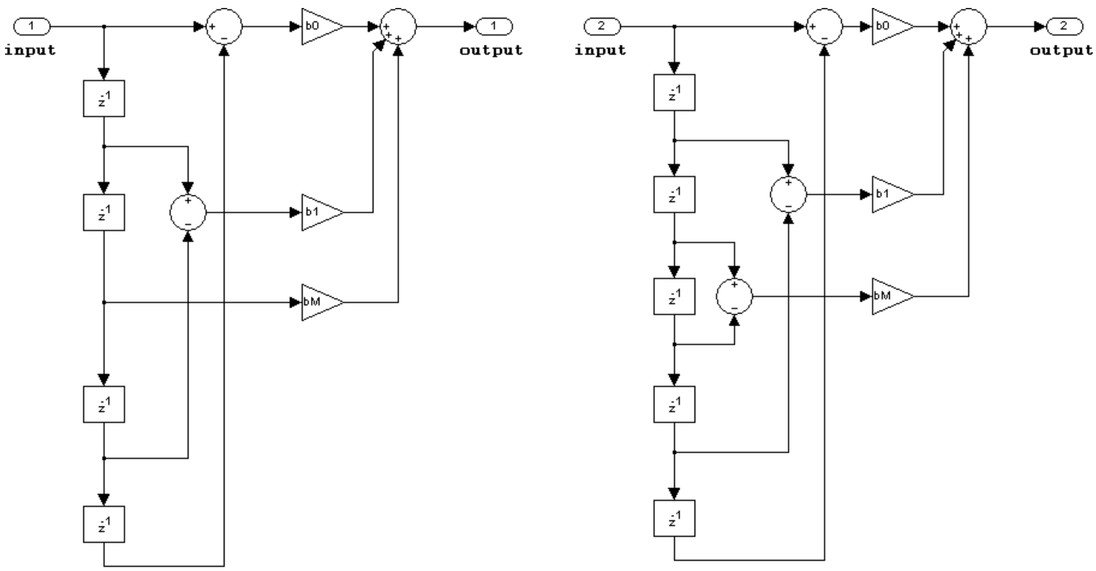


Even Order — Type I



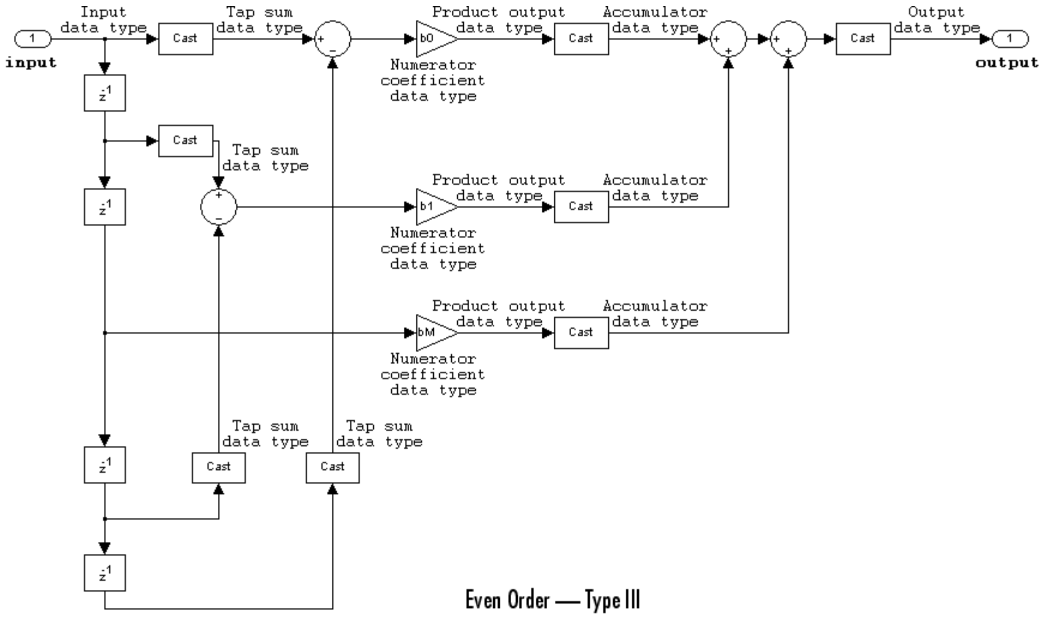
Odd Order — Type II

FIR (all zeros) direct form antisymmetric

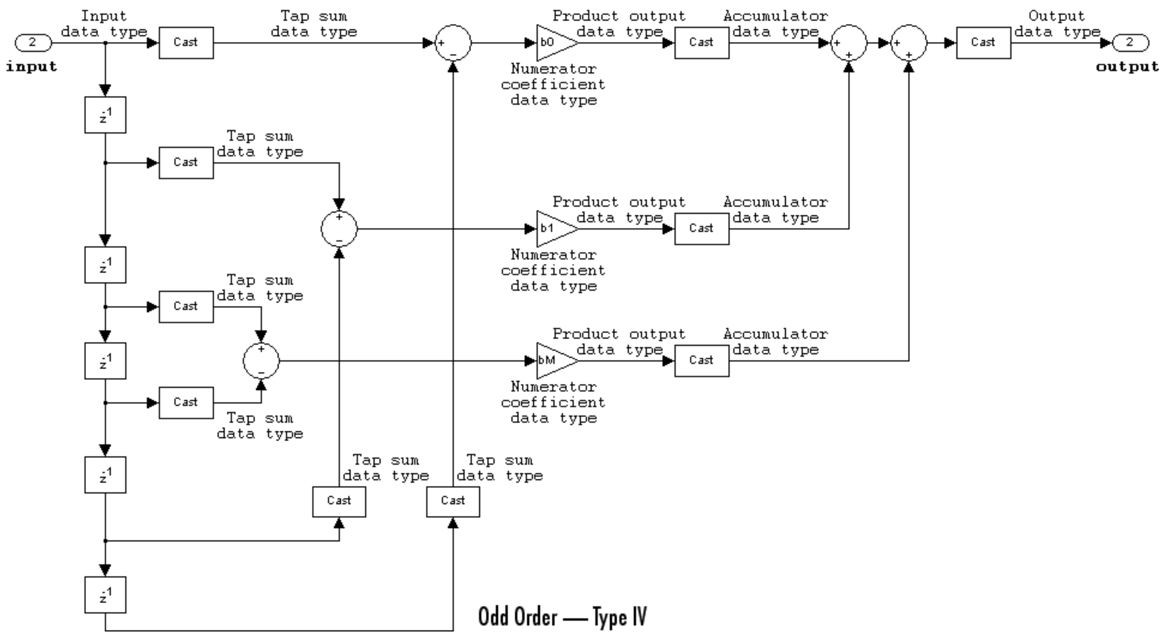


The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs can be real or complex.
- Numerator coefficients can be real or complex.
- You cannot specify the state data type on the block mask for this structure, because the input and output states have the same data types as the input.
- It is assumed that the filter coefficients are antisymmetric. Only the first half of the coefficients are used for filtering.
- The **Tap Sum** parameter determines the data type the filter uses when it sums the inputs prior to multiplication by the coefficients.

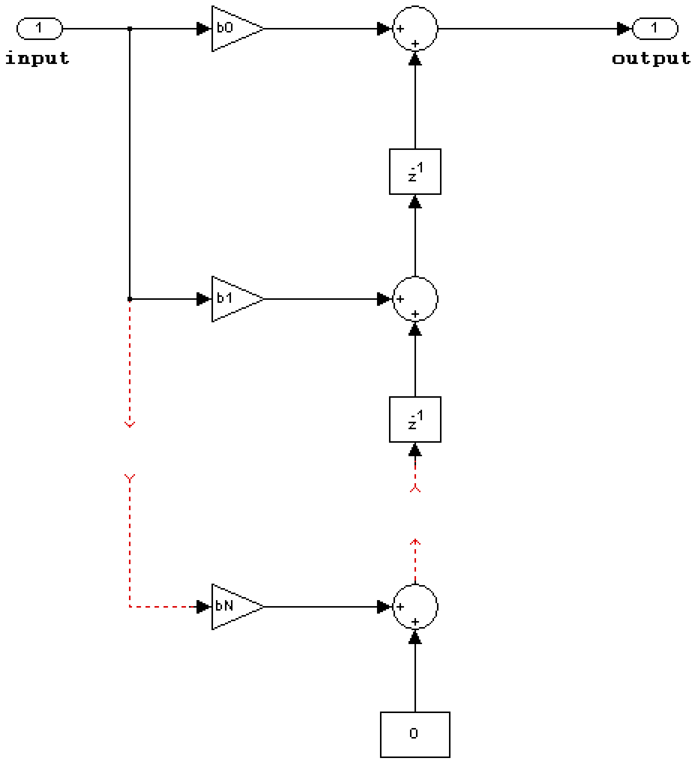


Even Order — Type III



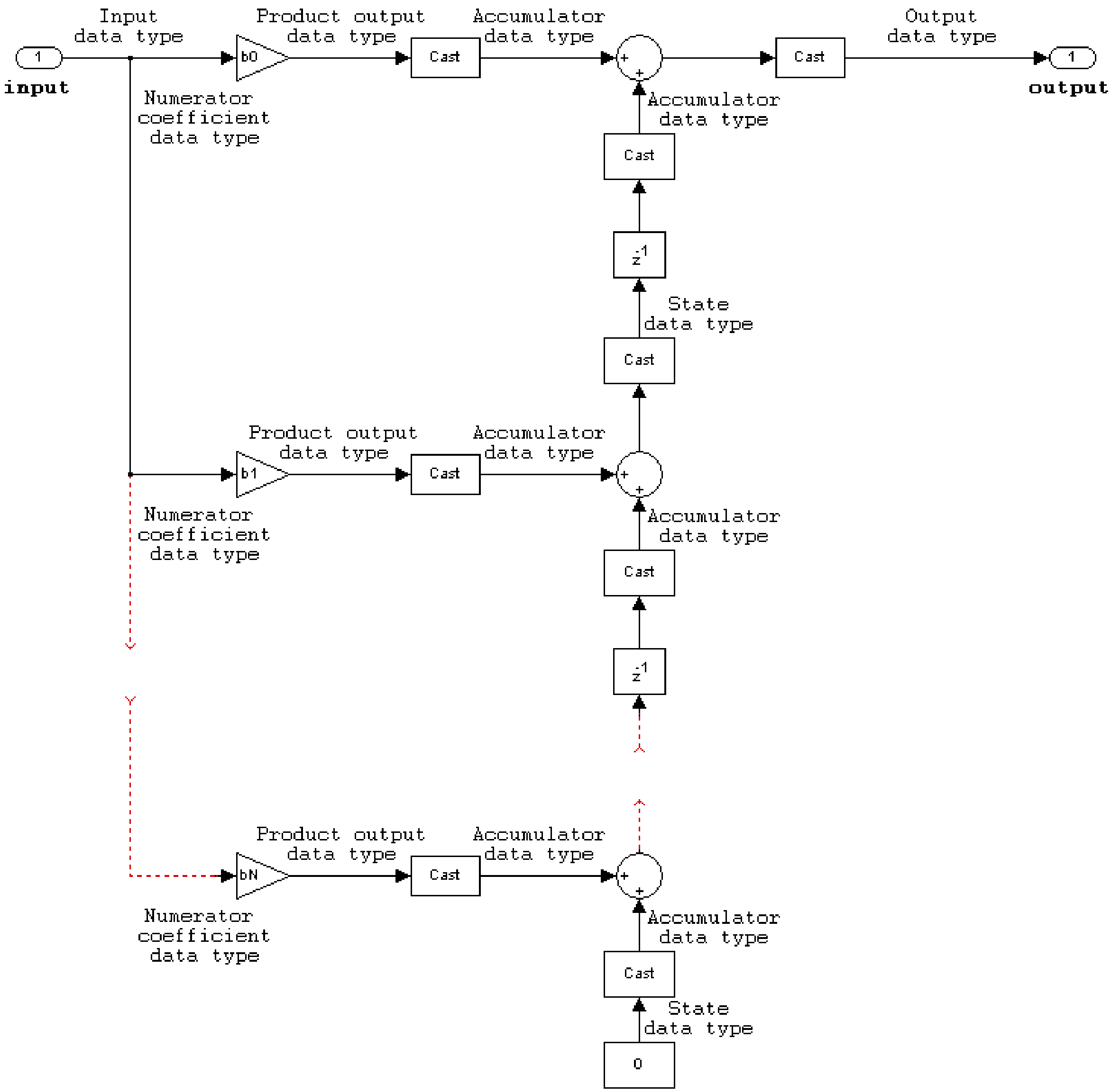
Odd Order — Type IV

FIR (all zeros) direct form transposed

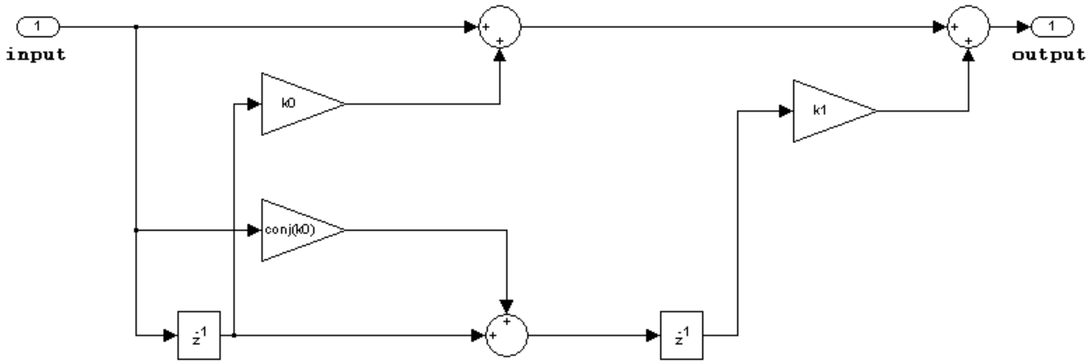


The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs can be real or complex.
- Coefficients can be real or complex.
- States are complex when either the inputs or the coefficients are complex.

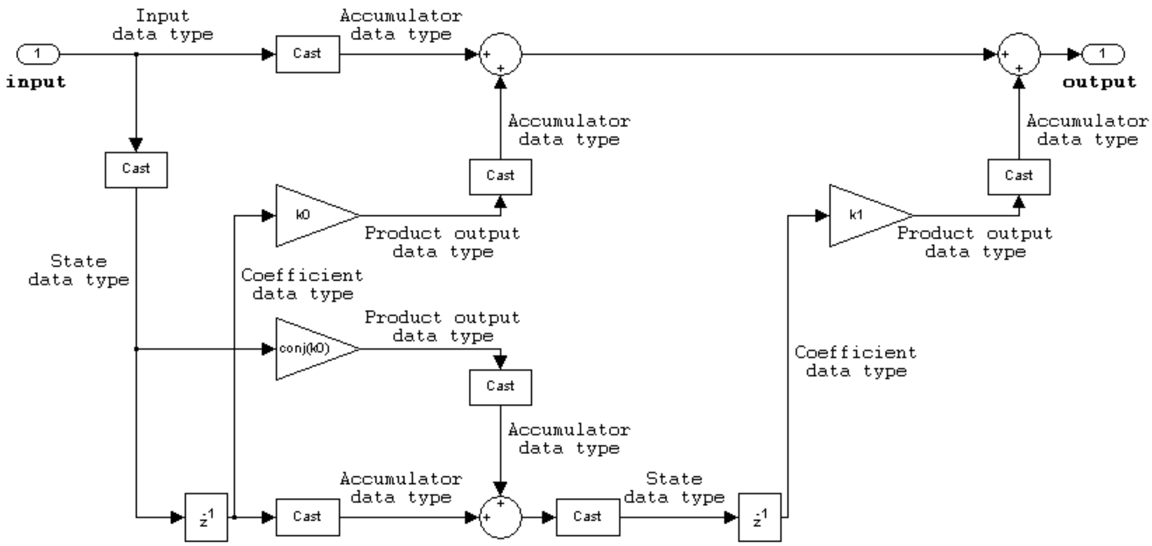

Input
data type
Product output
data type
Accumulator
data type
Output
data type
1
input
b0
Cast
Cast
1
output
Numerator
coefficient
data type
Accumulator
data type
Cast
z-1
State
data type
Cast
Product output
data type
Accumulator
data type
b1
Cast
Numerator
coefficient
data type
Accumulator
data type
Cast
z-1
Product output
data type
Accumulator
data type
bN
Cast
Numerator
coefficient
data type
Accumulator
data type
Cast
State
data type
0

FIR (all zeros) lattice MA



The following constraints are applicable when processing a fixed-point signal with this filter structure:

- Inputs and coefficients can be real or complex.
- Coefficients can be real or complex.



Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)
- 8-, 16-, and 32-bit signed integers

See Also

Digital Filter Design	Signal Processing Blockset
Filter Realization Wizard	Signal Processing Blockset
`dfilt`	Signal Processing Toolbox
`fdatool`	Signal Processing Toolbox
`fvtool`	Signal Processing Toolbox
`sptool`	Signal Processing Toolbox

Purpose

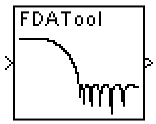
Design and implement digital FIR and IIR filters

Library

Filtering / Filter Designs

`dsparch4`

Description



Note Use this block to design, analyze, and then efficiently implement floating-point filters. The following blocks also implement digital filters, but serve slightly different purposes:

- Digital Filter — Use to efficiently implement floating-point or fixed-point filters that you have already designed. This block provides the same exact filter implementation as the Digital Filter Design block.
- Filter Realization Wizard — Use to implement floating-point or fixed-point filters built from Sum, Gain, and Unit Delay blocks. You can either design the filter within this block, or import the coefficients of a filter that you designed elsewhere.

The Digital Filter Design block implements a digital FIR or IIR filter that you design using the Filter Design and Analysis Tool (`fdatool`) GUI. This block provides the same exact filter implementation as the Digital Filter block.

The block applies the specified filter to each channel of a discrete-time input signal, and outputs the result. The outputs of the block numerically match the outputs of the Digital Filter block, the MATLAB® `filter` function, and the Filter Design Toolbox™ `filter` function.

The sampling frequency, Fs, that you specify in the FDATool GUI should be identical to the sampling frequency of the Digital Filter Design block's input block. When the sampling frequencies of these blocks do not match, the Digital Filter Design block returns a warning message and inherits the sampling frequency of the input block.

Valid Inputs and Corresponding Outputs

The block accepts inputs that are sample-based or frame-based vectors and matrices. The block filters each input channel independently over time, where

- Each *column* of a frame-based vector or matrix is an independent channel.
- Each *element* of a sample-based vector or matrix is an independent channel.

The output has the same dimensions and frame status as the input.

Designing the Filter

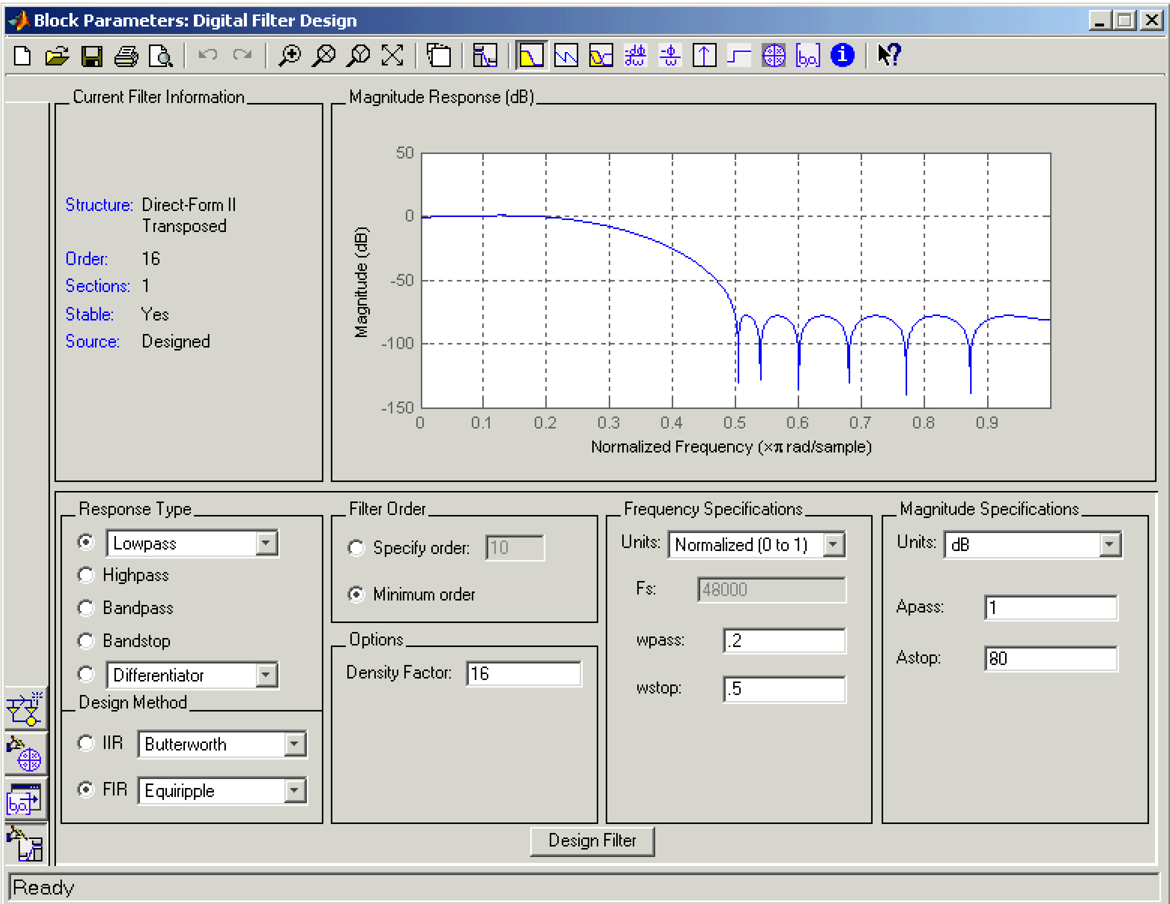
Double-click the Digital Filter Design block to open FDATool. Use FDATool to design or import a digital FIR or IIR filter. To learn how to design filters with this block and FDATool, see the following topics:

- "Digital Filter Design Block"
- `fdatool` reference page in the Signal Processing Toolbox™ documentation

Tuning the Filter During Simulation

You can tune the filter specifications in FDATool during simulations as long as your changes do not modify the filter length or filter order. The block's filter updates as soon as you apply any filter changes in FDATool.

Dialog Box



The FDATool GUI Opened from the Digital Filter Design Block

To get the **Transform Filter** button , install the Filter Design Toolbox product. To get the **Targets** menu, install the Target Support Package™ TC6 product.

To learn how to use the FDATool GUI, see “Designing the Filter” on page 2-358.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Analog Filter Design	Signal Processing Blockset
Window Function	Signal Processing Blockset
`fdatool`	Signal Processing Toolbox
`filter`	Signal Processing Toolbox
`fvtool`	Signal Processing Toolbox
`sptool`	Signal Processing Toolbox
`filter`	Filter Design Toolbox

To learn how to use this block and FDATool, see the following:

- “Filters”
- “Digital Filter Design Block”
- `fdatool` reference page in the Signal Processing Toolbox documentation

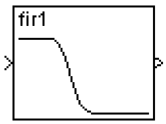
Purpose

Design and implement a variety of FIR filters

Library

`dspobslib`

Description



Note The Digital FIR Filter Design block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Digital FIR Filter Design block designs a discrete-time (digital) FIR filter in one of several different band configurations using a window method. Most of these filters are designed using the Signal Processing Toolbox™ `fir1` function, and are real with linear phase response. The block applies the filter to a discrete-time input using the Direct-Form II Transpose Filter block.

An *M*-by-*N* sample-based matrix input is treated as *M**N* independent channels, and an *M*-by-*N* frame-based matrix input is treated as *N* independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

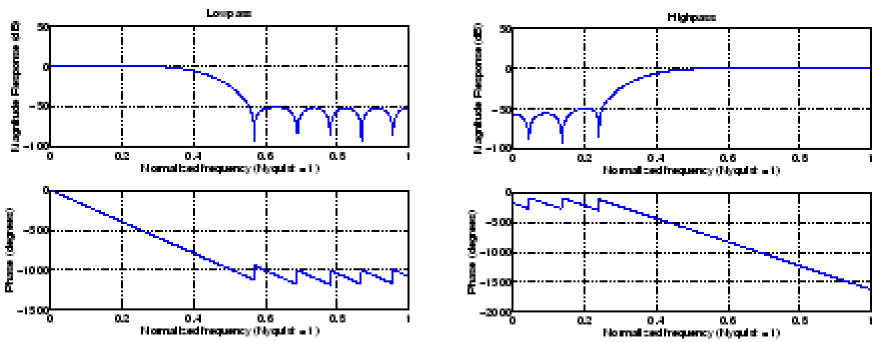
For complete details on the classical FIR filter design algorithm, see the description of the `fir1` and `fir2` functions in the Signal Processing Toolbox documentation.

Band Configurations

The band configuration for the filter is set from the **Filter type** pop-up menu. The band configuration parameters below this pop-up menu adapt appropriately to match the **Filter type** selection.

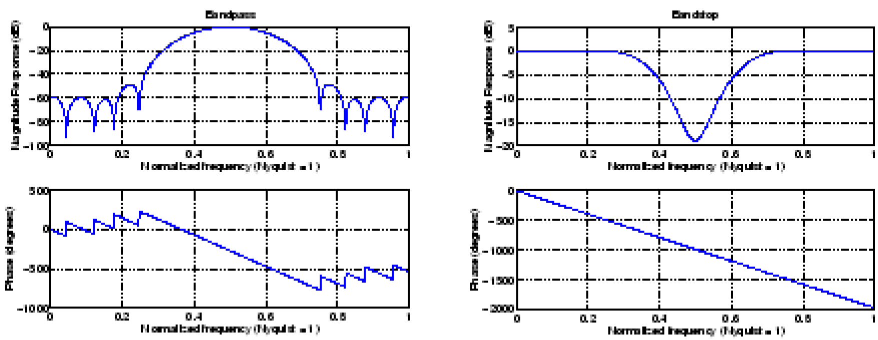
- **Lowpass** and **Highpass**

  In lowpass and highpass configurations, the **Filter order** and **Cutoff frequency** parameters specify the filter design. Frequencies are normalized to half the sample frequency. The figure below shows the frequency response of the default order-22 filter with cutoff at 0.4.



- **Bandpass** and **Bandstop**

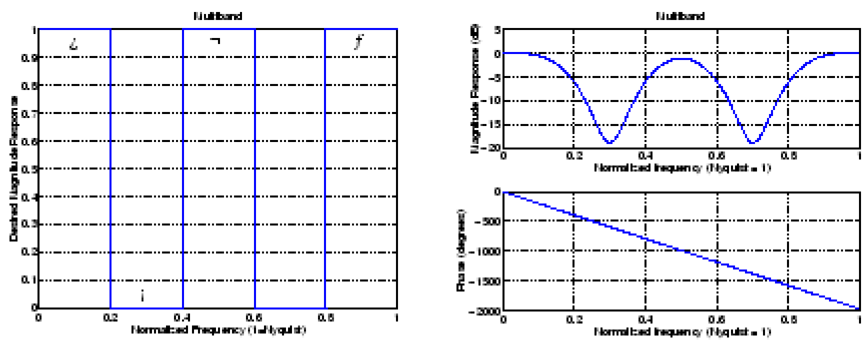
  In bandpass and bandstop configurations, the **Filter order**, **Lower cutoff frequency**, and **Upper cutoff frequency** parameters specify the filter design. Frequencies are normalized to half the sample frequency, and the actual filter order is twice the **Filter order** parameter value. The figure below shows the frequency response of the default order-22 filter with lower cutoff at 0.4, and upper cutoff at 0.6.



- **Multiband**

  In the multiband configuration, the **Filter order**, **Cutoff frequency vector**, and **Gain in the first band** parameters specify the filter design. The **Cutoff frequency vector** contains frequency points in the range `0` to `1`, where `1` corresponds to half the sample frequency. Frequency points must appear in ascending order. The **Gain in the first band** parameter specifies the gain in the first band: `0` indicates a stopband, and `1` indicates a passband. Additional bands alternate between passband and stopband. The figure below shows the frequency response of the default order-22 filter with five bands, the first a passband.

  Cutoff frequency vector = `[0.2 0.4 0.6 0.8]`



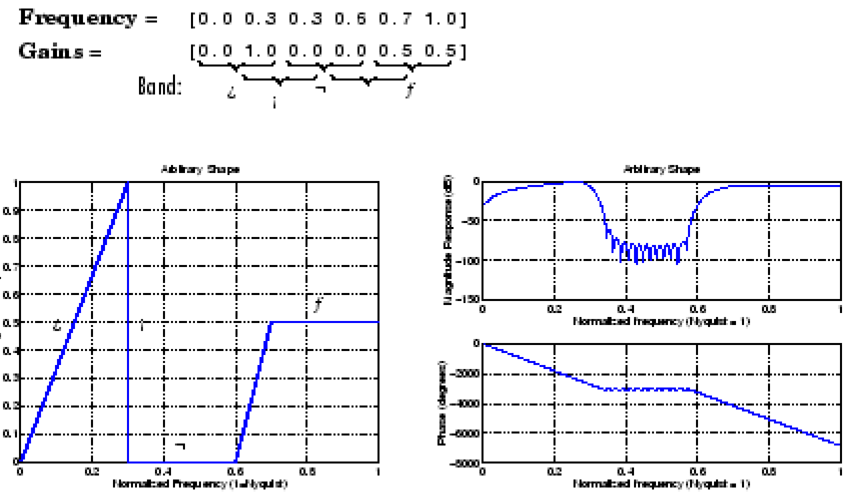
- **Arbitrary shape**

  In the arbitrary shape configuration, the **Filter order**, **Frequency vector**, and **Gains at these frequencies** parameters specify the filter design. The **Frequency vector**, `fn`, contains frequency points in the range `0` to `1` (inclusive) in ascending order, where `1` corresponds to half the sample frequency. The **Gains at these frequencies** parameter, `mn`, is a vector containing the desired magnitude response at the corresponding points in the **Frequency vector**. (Note that the specifications for the **Arbitrary shape** configuration are similar to those for the Yule-Walker IIR Filter Design block. Arbitrary-shape filters are designed using the Signal Processing Toolbox `fir2` function.)

The desired magnitude response of the design can be displayed by typing

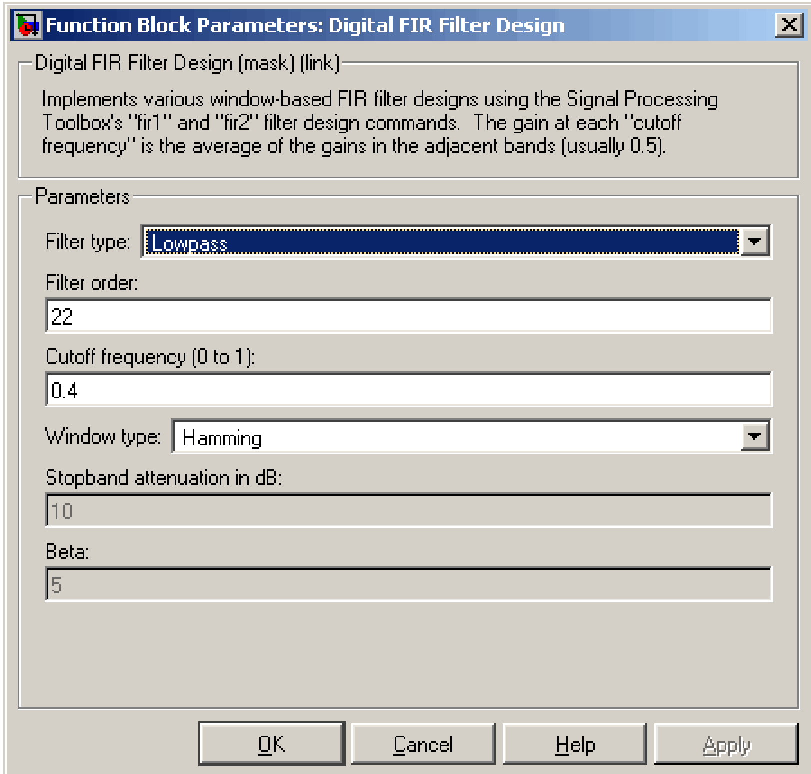
```
plot(fn,mn)
```

Duplicate frequencies can be used to specify a step in the response (such as band 2 below). The figure shows an order-100 filter with five bands.



The **Window type** parameter allows you to select from a variety of different windows. See the Window Function block reference for a complete description of the available options.

Dialog Box



The parameters displayed in the dialog box vary for different design/band combinations. Only some of the parameters listed below are visible in the dialog box at any one time.

Filter type

The type of filter to design: **Lowpass**, **Highpass**, **Bandpass**, **Bandstop**, **Multiband**, or **Arbitrary Shape**. Tunable.

Filter order

The order of the filter. The filter length is one more than this value. For the **Bandpass** and **Bandstop** configurations, the order of the final filter is twice this value.

Cutoff frequency
: The normalized cutoff frequency for the **Highpass** and **Lowpass** filter configurations. A value of 1 specifies half the sample frequency. Tunable.

Lower cutoff frequency
: The lower passband or stopband frequency for the **Bandpass** and **Bandstop** filter configurations. A value of 1 specifies half the sample frequency. Tunable.

Upper cutoff frequency
: The upper passband or stopband frequency for the **Bandpass** and **Bandstop** filter configurations. A value of 1 specifies half the sample frequency. Tunable.

Cutoff frequency vector
: A vector of ascending frequency points defining the cutoff edges for the **Multiband** filter. A value of 1 specifies half the sample frequency. Tunable.

Gain in the first band
: The gain in the first band of the **Multiband** filter: 0 specifies a stopband, 1 specifies a passband. Additional bands alternate between passband and stopband. Tunable.

Frequency vector
: A vector of ascending frequency points defining the frequency bands of the **Arbitrary shape** filter. The frequency range is 0 to 1 including the endpoints, where 1 corresponds to half the sample frequency. Tunable.

Gains at these frequencies
: A vector containing the desired magnitude response for the **Arbitrary shape** filter at the corresponding points in the **Frequency vector**. Tunable.

Window type
: The type of window to apply. See the Window Function block reference. Tunable.

Stopband ripple

The level (dB) of stopband ripple, R_s, for the **Chebyshev** window. Tunable.

Beta

The **Kaiser** window β parameter. Increasing **Beta** widens the mainlobe and decreases the amplitude of the window sidelobes in the window's frequency magnitude response. Tunable.

References

Antoniou, A. *Digital Filters: Analysis, Design, and Applications*. 2nd ed. New York, NY: McGraw-Hill, 1993.

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Digital FIR Raised Cosine Filter Design

Purpose

Design and implement raised cosine FIR filter

Library

`dspobslib`

Description

Note The Digital FIR Raised Cosine Filter Design block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Digital FIR Raised Cosine Filter Design block uses the Signal Processing Toolbox™ `firrcos` function to design a lowpass, linear-phase, digital FIR filter with a raised cosine transition band. The block applies the filter to a discrete-time input using the Direct-Form II Transpose Filter block.

An *M*-by-*N* sample-based matrix input is treated as *M*N* independent channels, and an *M*-by-*N* frame-based matrix input is treated as *N* independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

The frequency response of the raised cosine filter is

$$H(f) = \begin{cases} \dfrac{1}{2f_{n0}} & 0 \le |f| \le (1-R)f_{n0} \\[2ex] \left[\dfrac{1+\cos\dfrac{\pi}{2Rf_{n0}}|f|-(1-R)f_{n0}}{4f_{n0}}\right] & (1-R)f_{n0} \le |f| \le (1+R)f_{n0} \\[2ex] 0 & (1+R)f_{n0} \le |f| \le 1 \end{cases}$$

where $H(f)$ is the magnitude response at frequency f, f_{n0} is the normalized cutoff frequency (-6 dB) specified by the **Upper cutoff frequency** parameter, and R is a rolloff factor in the range `[0,1]` determining the passband-to-stopband transition width.

The **Square-root raised cosine filter** option designs a filter with magnitude response $\sqrt{H(f)}$. This is useful when the filter is part of a pair of matched filters.

When the **Design method** parameter is set to **Rolloff factor**, the secondary **Rolloff factor** parameter is enabled, and R can be directly specified. When **Design method** is set to **Transition bandwidth**, the secondary **Transition bandwidth** parameter is enabled, and the transition region bandwidth, Δf, can be specified in place of R. The transition region is centered on f_{n0} and must be sufficiently narrow to satisfy

$$0 < \left(f_{n0} \pm \frac{\Delta f}{2}\right) < 1$$

The **Upper cutoff frequency** and **Transition bandwidth** parameter values are normalized to half the sample frequency.

The **Window type** parameter allows you to apply a variety of different windows to the raised cosine filter. See the Window Function block reference for a complete description of the available options.

Algorithm

The filter output is computed by convolving the input with a truncated, delayed, windowed version of the filter's impulse response. The impulse response for the raised cosine filter is

$$h(kT_s) = \frac{1}{F_s}\operatorname{sinc}(2\pi kT_s f_{n0})\frac{\cos(2\pi RkT_s f_{n0})}{1-(4RkT_s f_{n0})^2} \quad -\infty < k < \infty$$

which has limits

$$h(0) = \frac{1}{F_s}$$

and

$$h\left(\pm\frac{1}{4Rf_{n0}}\right) = \frac{R}{2F_s}\sin\left(\frac{\pi}{2R}\right)$$

The impulse response for the square-root raised cosine filter is

$$h(kT_s) = \frac{4R\cos((1+R)2\pi kT_s f_{n0}) + \dfrac{\sin((1-R)2\pi kT_s f_{n0})}{8RkT_s f_{n0}}}{\pi F_s\sqrt{\dfrac{1}{2f_{n0}}}((8RkT_s f_{n0})^2 - 1)} \qquad -\infty < k < \infty$$

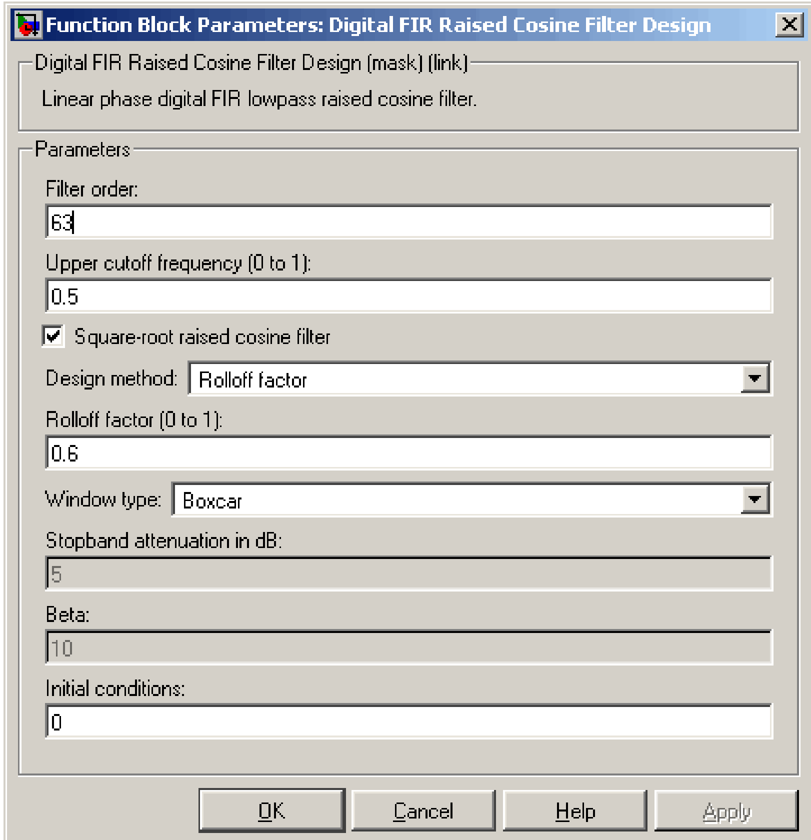
which has limits

$$h(0) = (-4R - \pi + \pi R)\frac{\sqrt{2f_{n0}}}{\pi F_s}$$

and

$$\left(\pm\frac{1}{8Rf_{n0}}\right) = \frac{\sqrt{2f_{n0}}}{2\pi F_s}\left[\pi(1+R)\sin\left(\frac{\pi(1+R)}{4R}\right) - 4R\sin\left(\frac{\pi(R-1)}{4R}\right) + \pi(R-1)\cos\left(\frac{\pi(R-1)}{4R}\right)\right]$$

Dialog Box



Filter order

The order of the filter. The filter length is one more than this value.

Upper cutoff frequency

The normalized cutoff frequency, f_{n0}. A value of 1 specifies half the sample frequency. Tunable.

Square-root raised cosine filter
Selects the square-root filter option, which designs a filter with magnitude response $\sqrt{H(f)}$. Tunable.

Design method
The method used to design the transition region of the filter, **Rolloff factor** or **Transition bandwidth**. Tunable.

Rolloff factor
The rolloff factor, R, enabled when **Rolloff factor** is selected in the **Design method** parameter. Tunable.

Transition bandwidth
The transition bandwidth, Δf, enabled when **Transition bandwidth** is selected in the **Design method** parameter. Tunable.

Window type
The type of window to apply. See the Window Function block reference. Tunable.

Stopband attenuation in dB
The level (dB) of stopband attenuation, R_s, for the Chebyshev window. Tunable.

Beta
The **Kaiser** window β parameter. Increasing β widens the mainlobe and decreases the amplitude of the window sidelobes in the window's frequency magnitude response. Tunable.

Initial conditions
The filter's initial conditions, a scalar, vector, or matrix. See the Direct-Form II Transpose Filter block reference for complete syntax information.

References

Proakis, J. G. *Digital Communications.* Third ed. New York, NY: McGraw-Hill, 1995.

Proakis, J. G. and M. Salehi. *Contemporary Communication Systems Using MATLAB.* Boston, MA: PWS Publishing, 1998.

Digital IIR Filter Design

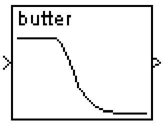
Purpose

Design and implement IIR filter

Library

`dspobslib`

Description



Note The Digital IIR Filter Design block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Digital IIR Filter Design block designs a discrete-time (digital) IIR filter in a lowpass, highpass, bandpass, or bandstop configuration, and applies it to the input using the Direct-Form II Transpose Filter block.

An M-by-N sample-based matrix input is treated as $M*N$ independent channels, and an M-by-N frame-based matrix input is treated as N independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

The **Design method** parameter allows you to specify Butterworth, Chebyshev type I, Chebyshev type II, and elliptic filter designs. Note that for the bandpass and bandstop configurations, the actual filter length is twice the **Filter order** parameter value.

Filter Design	Description
Butterworth	The magnitude response of a Butterworth filter is maximally flat in the passband and monotonic overall.
Chebyshev type I	The magnitude response of a Chebyshev type I filter is equiripple in the passband and monotonic in the stopband.

Filter Design	Description
Chebyshev type II	The magnitude response of a Chebyshev type II filter is monotonic in the passband and equiripple in the stopband.
Elliptic	The magnitude response of an elliptic filter is equiripple in both the passband and the stopband.

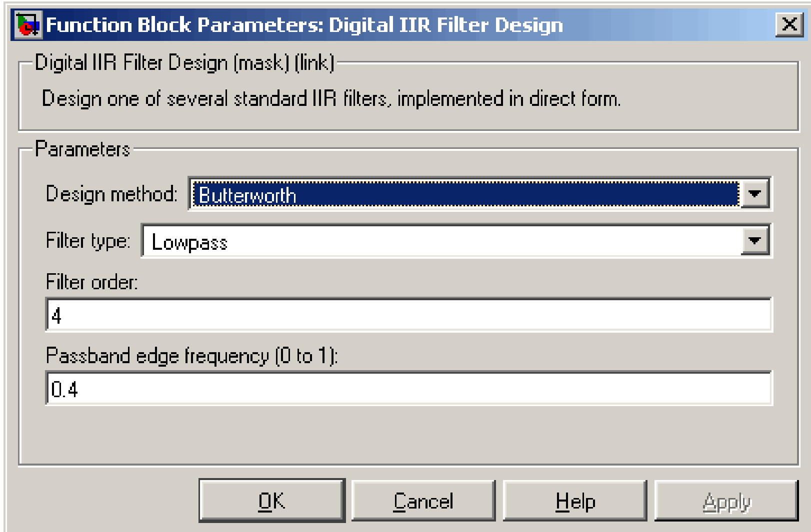
The design and band configuration of the filter are selected from the **Design method** and **Filter type** pop-up menus in the dialog box. For each combination of design method and band configuration, an appropriate set of secondary parameters is displayed.

The table below lists the available parameters for each design/band combination. For lowpass and highpass band configurations, these parameters include the passband edge frequency f_{np}, the stopband edge frequency f_{ns}, the passband ripple R_p, and the stopband attenuation R_s. For bandpass and bandstop configurations, the parameters include the lower and upper passband edge frequencies, f_{np1} and f_{np2}, the lower and upper stopband edge frequencies, f_{ns1} and f_{ns2}, the passband ripple R_p, and the stopband attenuation R_s. Frequency values are normalized to half the sample frequency, and ripple and attenuation values are in dB.

	Lowpass	Highpass	Bandpass	Bandstop
Butterworth	Order, f_{np}	Order, f_{np}	Order, f_{np1}, f_{np2}	Order, f_{np1}, f_{np2}
Chebyshev Type I	Order, f_{np}, R_p	Order, f_{np}, R_p	Order, f_{np1}, f_{np2}, $\mathrm{R_p}$	Order, f_{np1}, f_{np2}, R_p
Chebyshev Type II	Order, f_{ns}, R_s	Order, f_{ns}, R_s	Order, f_{ns1}, f_{ns2}, R_s	Order, f_{ns1}, f_{ns2}, R_s
Elliptic	Order, f_{np}, R_p, R_s	Order, f_{np}, R_p, R_s	Order, f_{np1}, f_{np2}, R_p, R_s	Order, f_{np1}, f_{np2}, R_p, R_s

The digital filters are designed using Signal Processing Toolbox™ software's filter design commands `butter`, `cheby1`, `cheby2`, and `ellip`.

Dialog Box



The parameters displayed in the dialog box vary for different design/band combinations. Only some of the parameters listed below are visible in the dialog box at any one time.

Design method

The filter design method: `Butterworth`, `Chebyshev type I`, `Chebyshev type II`, or `Elliptic`. Tunable.

Filter type

The type of filter to design: `Lowpass`, `Highpass`, `Bandpass`, or `Bandstop`. Tunable.

Filter order

The order of the filter for lowpass and highpass configurations. For bandpass and bandstop configurations, the length of the final filter is twice this value.

Passband edge frequency

The normalized passband edge frequency for the highpass and lowpass configurations of the Butterworth, Chebyshev type I, and elliptic designs. Tunable.

Lower passband edge frequency
The normalized lower passband frequency for the bandpass and bandstop configurations of the Butterworth, Chebyshev type I, and elliptic designs. Tunable.

Upper passband edge frequency
The normalized upper passband frequency for the bandpass and bandstop configurations of the Butterworth, Chebyshev type I, or elliptic designs. Tunable.

Stopband edge frequency
The normalized stopband edge frequency for the highpass and lowpass band configurations of the Chebyshev type II design. Tunable.

Lower stopband edge frequency
The normalized lower stopband frequency for the bandpass and bandstop configurations of the Chebyshev type II design. Tunable.

Upper stopband edge frequency
The normalized upper stopband frequency for the bandpass and bandstop filter configurations of the Chebyshev type II design. Tunable.

Passband ripple in dB
The passband ripple, in dB, for the Chebyshev type I and elliptic designs. Tunable.

Stopband attenuation in dB
The stopband attenuation, in dB, for the Chebyshev type II and elliptic designs. Tunable.

References

Antoniou, A. *Digital Filters: Analysis, Design, and Applications*. 2nd ed. New York, NY: McGraw-Hill, 1993.

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

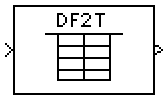
Purpose

Apply IIR filter to input

Library

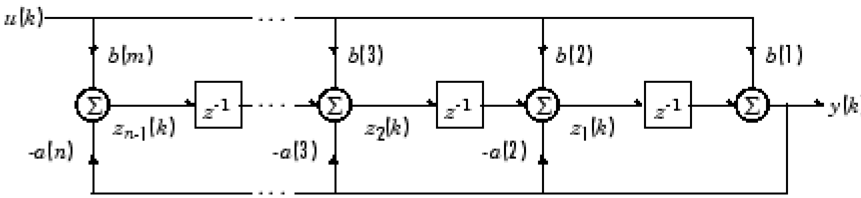
`dspobslib`

Description



Note The Direct-Form II Transpose Filter block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Direct-Form II Transpose Filter block applies a transposed direct-form II IIR filter to the input.



This is a canonical form that has the minimum number of delay elements. The filter order is `max(m,n)-1`.

An *M*-by-*N* sample-based matrix input is treated as *M***N* independent channels, and an *M*-by-*N* frame-based matrix input is treated as *N* independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

The filter is specified in the parameter dialog box by its transfer function,

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2 z^{-1} + \ldots + b_{m+1} z^{-(m-1)}}{a_1 + a_2 z^{-1} + \ldots + a_{n+1} z^{-(n-1)}}$$

where the **Numerator** parameter specifies the vector of numerator coefficients,

```
[b(1) b(2) ... b(m)]
```

and the **Denominator** parameter specifies the vector of denominator coefficients,

```
[a(1) a(2) ... a(n)]
```

The filter coefficients are normalized by a_1.

Initial Conditions

In its default form, the filter initializes the internal filter states to zero, which is equivalent to assuming past inputs and outputs are zero. The block also accepts optional nonzero initial conditions for the filter delays. Note that the number of filter states (delay elements) per input channel is

```
max(m,n)-1
```

The **Initial conditions** parameter may take one of four forms:

- Empty matrix

  The empty matrix, `[]`, causes a zero (0) initial condition to be applied to all delay elements in each filter channel.

- Scalar

  The scalar value is copied to all delay elements in each filter channel. Note that a value of zero is equivalent to setting the **Initial conditions** parameter to the empty matrix, `[]`.

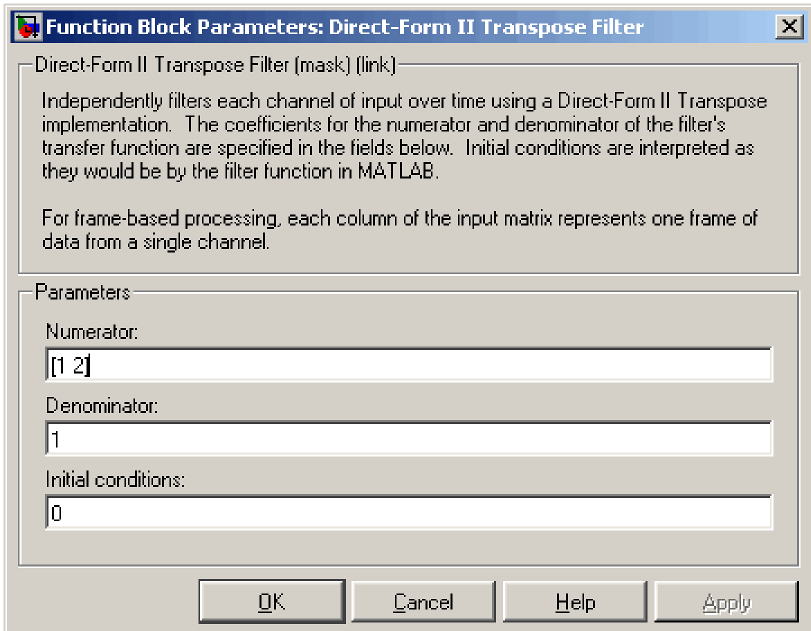
- Vector

  The vector has a length equal to the number of delay elements in each filter channel, `max(m,n)-1`, and specifies a unique initial condition for each delay element in the filter channel. This vector of initial conditions is applied to each filter channel.

- Matrix

The matrix specifies a unique initial condition for each delay element, and can specify different initial conditions for each filter channel. The matrix must have the same number of rows as the number of delay elements in the filter, `max(m,n)-1`, and must have one column per filter channel.

Dialog Box



Numerator
The filter numerator vector. Tunable; the numerator coefficients can be adjusted while the simulation runs, but the vector length (i.e., the filter order) must remain the same.

Denominator
The filter denominator vector. Tunable; the denominator coefficients can be adjusted while the simulation runs, but the vector length (i.e., the filter order) must remain the same.

Initial conditions
The filter's initial conditions, a scalar, vector, or matrix.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Purpose

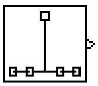
Generate discrete impulse

Library

Signal Processing Sources

`dspsrcs4`

Description



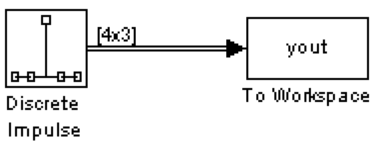
The Discrete Impulse block generates an impulse (the value `1`) at output sample `D+1`, where `D` is specified by the **Delay** parameter (`D` ≥`0`). All output samples preceding and following sample `D+1` are zero.

When `D` is a length-N vector, the block generates an M-by-N matrix output representing N distinct channels, where frame size M is specified by the **Samples per frame** parameter. The impulse for the `i`th channel appears at sample `D(i)+1`. For M=1, the output is sample based; otherwise, the output is frame based.

The **Sample time** parameter value, T_s, specifies the output signal sample period. The resulting frame period is $M*T_s$.

Examples

Construct the model below.



Configure the Discrete Impulse block to generate a frame-based three-channel output of type `double`, with impulses at samples 1, 4, and 6 of channels 1, 2, and 3, respectively. Use a sample period of 0.25 and a frame size of 4. The corresponding settings should be as follows:

- **Delay** = `[0 3 5]`
- **Sample time** = `0.25`
- **Samples per frame** = `4`
- **Output data type** = `double`

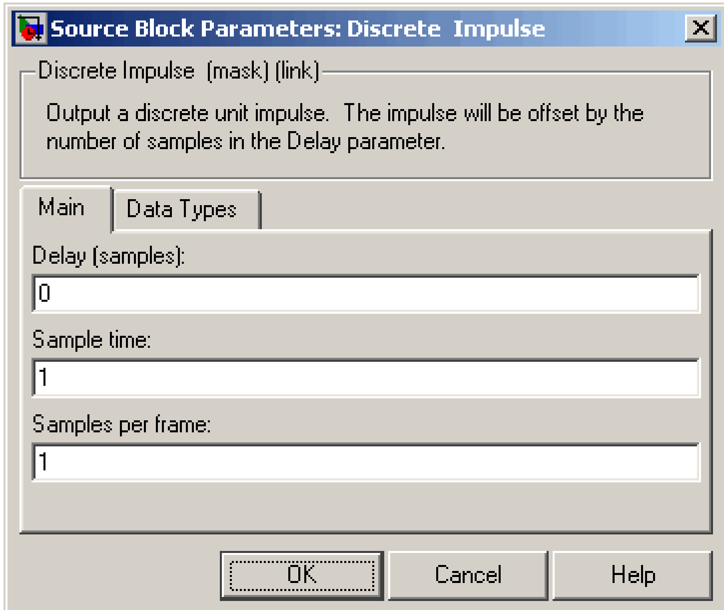
Run the model and look at the output, yout. The first few samples of each channel are shown below.

```
yout(1:10,:)
ans =
     1     0     0
     0     0     0
     0     0     0
     0     1     0
     0     0     0
     0     0     1
     0     0     0
     0     0     0
     0     0     0
     0     0     0
```

The block generates an impulse at sample 1 of channel 1 (first column), at sample 4 of channel 2 (second column), and at sample 6 of channel 3 (third column).

Dialog Box

The **Main** pane of the Discrete Impulse block dialog appears as follows.



Delay

The number of zero-valued output samples, `D`, preceding the impulse. A length-N vector specifies an N-channel output.

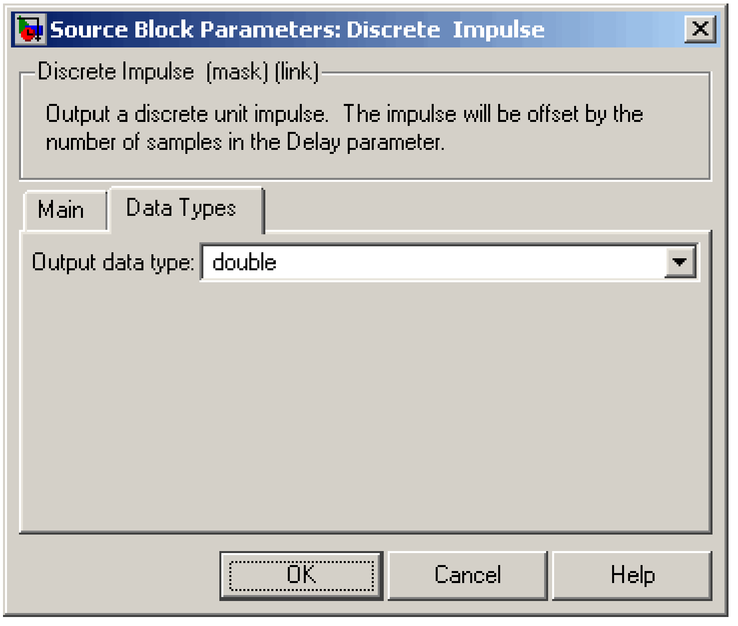
Sample time

The sample period, T_s, of the output signal. The output frame period is $M*T_s$.

Samples per frame

The number of samples, M, in each output frame. When the value of this parameter is `1`, the block outputs a sample-based signal.

The **Data Types** pane of the Discrete Impulse block dialog appears as follows.



Output data type

Specify the output data type in one of the following ways:

- Choose one of the built-in data types from the list.
- Choose `Fixed-point` to specify the output data type and scaling in the **Signed**, **Word length**, **Set fraction length in output to**, and **Fraction length** parameters.
- Choose `User-defined` to specify the output data type and scaling in the **User-defined data type**, **Set fraction length in output to**, and **Fraction length** parameters.
- Choose `Inherit via back propagation` to set the output data type and scaling to match the next block downstream.

Signed

Select to output a signed fixed-point signal. Otherwise, the signal is unsigned. This parameter is only visible when you select `Fixed-point` for the **Output data type** parameter.

Word length

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

User-defined data type

Specify any built-in or fixed-point data type. You can specify fixed-point data types using the following Simulink® Fixed Point™ functions: `sfix`, `ufix`, `sint`, `uint`, `sfrac`, and `ufrac`. This parameter is only visible when you select `User-defined` for the **Output data type** parameter.

Set fraction length in output to

Specify the scaling of the fixed-point output by either of the following two methods:

- Choose `Best precision` to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose `User-defined` to specify the output scaling in the **Fraction length** parameter.

This parameter is only visible when you select `Fixed-point` for the **Output data type** parameter, or when you select `User-defined` and the specified output data type is a fixed-point data type.

Fraction length

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

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point
- Boolean
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Data Type Conversion	Simulink
Constant	Simulink
Multiphase Clock	Signal Processing Blockset
N-Sample Enable	Signal Processing Blockset
Signal From Workspace	Signal Processing Blockset
`impz`	Signal Processing Toolbox

Purpose Show value of input

Library Signal Processing Sinks

`dspsnks4`

Description Refer to the Simulink® Display reference page for more information.

Purpose

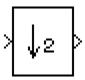
Resample input at lower rate by deleting samples

Library

Signal Operations

`dspsigops`

Description



The Downsample block resamples each channel of the M_i-by-N input at a rate K times lower than the input sample rate by discarding K-1 consecutive samples following each sample passed through to the output. The integer K is specified by the **Downsample factor** parameter.

The **Sample offset** parameter delays the output samples by an integer number of sample periods, D, where $0 \le D < (K\text{-}1)$, so that any of the K possible output phases can be selected. For example, when you downsample the sequence 1, 2, 3, ... by a factor of 4, you can select from the following four phases.

Input Sequence	Sample Offset, D	Output Sequence (K=4)
`1,2,3,...`	0	`1,5,9,13,17,21,25,29,...`
`1,2,3,...`	1	`0,2,6,10,14,18,22,26,...`
`1,2,3,...`	2	`0,3,7,11,15,19,23,27,...`
`1,2,3,...`	3	`0,4,8,12,16,20,24,28,...`

The initial zero in each of the latter three output sequences above is a result of the default zero **Initial condition** parameter setting for this example. See “Latency” on page 2-391 for more on the **Initial condition** parameter.

This block supports triggered subsystems if, for **Sample-based mode**, you select `Force single-rate` and, for **Frame-based mode**, you select `Maintain input frame rate`.

Sample-Based Operation

When the input is sample based, the block treats each of the $M*N$ matrix elements as an independent channel, and downsamples each channel over time. The input and output sizes are identical.

The **Sample-based mode** parameter determines how the block represents the new rate at the output. There are two available options:

- `Allow multirate`

  When you select `Allow multirate`, the sample period of the sample-based output is K times longer than the input sample period ($T_{so} = KT_{si}$). The block is therefore multirate.

- `Force single rate`

  When you select `Force single rate`, the block forces the output sample rate to match the input sample rate ($T_{so} = T_{si}$) by repeating every Kth input sample K times at the output. The block is therefore single-rate. (The block's operation when you select `Enforce single rate` is similar to the operation of a Sample and Hold block with a repeating trigger event of period KT_{si}.)

The setting of the **Frame-based mode** pop-up menu does not affect sample-based inputs.

Frame-Based Inputs

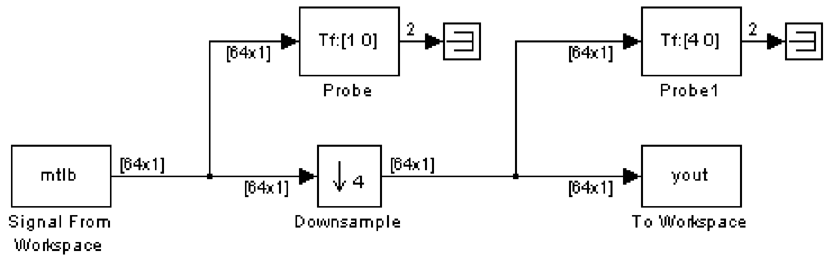
When the input is frame based, the block treats each of the N input columns as a frame containing M_i sequential time samples from an independent channel. The block downsamples each channel independently by discarding K-1 rows of the input matrix following each row that it passes through to the output.

The **Frame-based mode** parameter determines how the block adjusts the rate at the output to accommodate the reduced number of samples. There are two available options:

- `Maintain input frame size`

The block generates the output at the slower (downsampled) rate by using a proportionally longer frame *period* at the output port than at the input port. For downsampling by a factor of K, the output frame period is K times longer than the input frame period ($T_{fo} = KT_{fi}$), but the input and output frame sizes are equal.

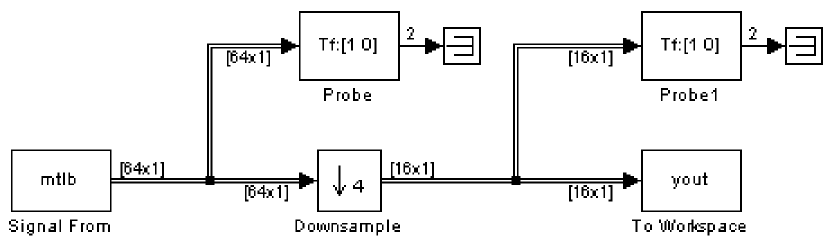
The model below shows a single-channel input with a frame period of 1 second being downsampled by a factor of 4 to a frame period of 4 seconds. The input and output frame sizes are identical.



- `Maintain input frame rate`

  The block generates the output at the slower (downsampled) rate by using a proportionally smaller frame *size* than the input. For downsampling by a factor of K, the output frame size is K times smaller than the input frame size ($M_o = M_i/K$), but the input and output frame rates are equal.

  The model below shows a single-channel input of frame size 64 being downsampled by a factor of 4 to a frame size of 16. The input and output frame rates are identical.



The setting of the **Sample-based mode** pop-up menu does not affect frame-based inputs.

Latency

The Downsample block has *zero tasking latency* in the following cases:

- The **Downsample factor** parameter, K, is `1`
- The block input is a frame-based signal and the **Maintain input frame rate** parameter is selected
- The block input is a sample-based signal and **Sample offset** parameter, D, is `0`

Zero tasking latency means that the block propagates input sample D+1 (received at t=0) as the first output sample, followed by input sample D+1+K, input sample D+1+2K, and so on. The **Initial condition** parameter value is not used.

For all other cases than those listed above the latency is nonzero:

- For sample-based input, the latency is one sample.
- For frame-based input, the latency is one frame.

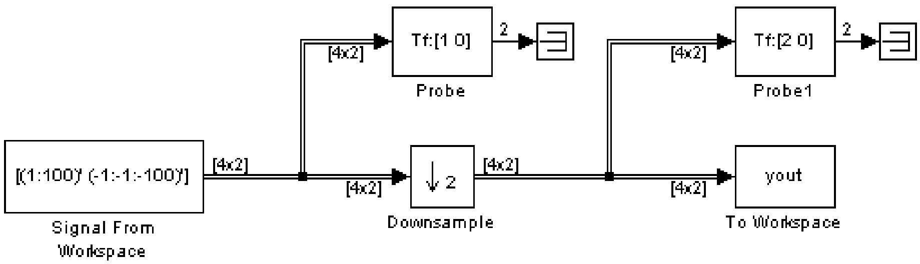
In all cases of *one-sample latency*, the initial condition for each channel appears as the first output sample. Input sample D+1 appears as the second output sample for each channel, followed by input sample D+1+K, input sample D+1+2K, and so on. The **Initial condition** parameter can be an M_i-by-N matrix containing one value for each channel, or a scalar to be applied to all signal channels.

In all cases of *one-frame latency*, the M_i rows of the initial condition matrix appear in sequence as the first M_i output rows. Input sample D+1 (i.e, row D+1 of the input matrix) appears in the output as sample M_i+1, followed by input sample D+1+K, input sample D+1+2K, and so on. The **Initial condition** value can be an M_i-by-N matrix, or a scalar to be repeated across all elements of the M_i-by-N matrix. See the following example for an illustration of this case.

Note For more information on latency and the Simulink® tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Examples

Construct the frame-based model shown below.



Adjust the block parameters as follows:

- Configure the Signal From Workspace block to generate a two-channel signal with frame size of 4 and sample period of 0.25 second. This represents an output frame period of 1 second (0.25*4). The first channel should contain the positive ramp signal 1, 2, ..., 100, and the second channel should contain the negative ramp signal -1, -2, ..., -100. The settings are
  - **Signal** = `[(1:100)' (-1:-1:-100)']`
  - **Sample time** = `0.25`
  - **Samples per frame** = `4`
- Configure the Downsample block to downsample the two-channel input by decreasing the output frame rate by a factor of 2 relative to the input frame rate. Set a sample offset of 1, and a 4-by-2 initial condition matrix of

$$\begin{bmatrix} 11 & -11 \\ 12 & -12 \\ 13 & -13 \\ 14 & -14 \end{bmatrix}$$

- **Downsample factor** = 2
- **Sample offset** = 1
- **Initial condition** = [11 -11;12 -12;13 -13;14 -14]
- **Frame-based mode** = Maintain input frame size

- Configure the Probe blocks by clearing the **Probe width** and **Probe complex signal** check boxes (if desired).

This model is multirate because there are at least two distinct frame rates, as shown by the two Probe blocks. To run this model in the Simulink multitasking mode, open the Configuration Parameters dialog box. From the list on the left side of the dialog box, click **Solver**. From the **Type** list, select Fixed-step, and from the **Solver** list, select discrete (no continuous states). From the **Tasking mode for periodic sample times** list, select MultiTasking. Additionally, set the **Stop time** parameter to 30.

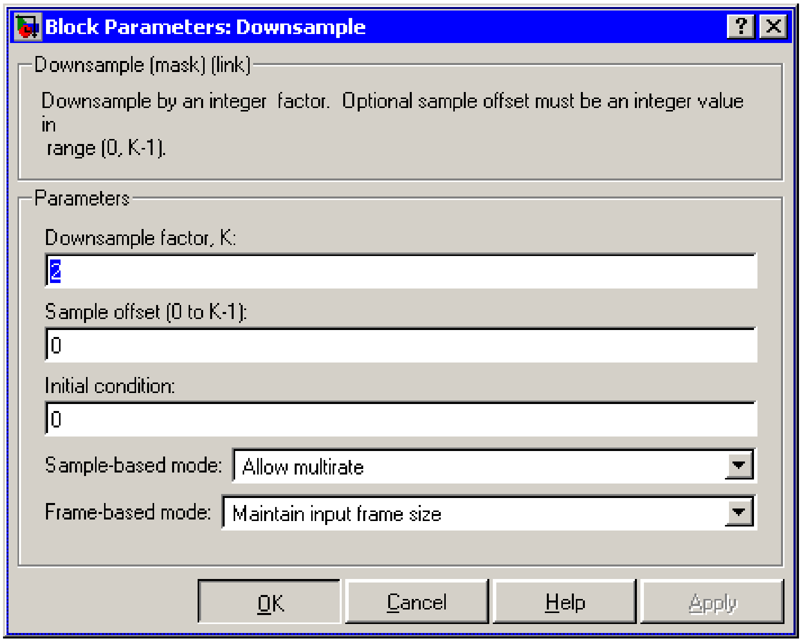
Run the model and look at the output, yout. The first few samples of each channel are shown below.

```
yout =
    11   -11
    12   -12
    13   -13
    14   -14
     2    -2
     4    -4
     6    -6
     8    -8
    10   -10
    12   -12
```

14 -14

Since we ran this frame based multirate model in multitasking mode, the first row of the initial condition matrix appears as the first output sample, followed by the other three initial condition rows. The second row of the first input matrix (that is, row D+1, where D is the **Sample offset**) appears in the output as sample 5 (that is sample M_i+1, where M_i is the input frame size).

Dialog Box



Downsample factor

The integer factor, K, by which to decrease the input sample rate.

Sample offset

The sample offset, D, which must be an integer in the range [0, K-1].

Initial condition
The value with which the block is initialized for cases of nonzero latency; a scalar or matrix.

Sample-based mode
The method by which to implement downsampling for sample-based inputs: `Allow multirate` (that is, decrease the output sample rate), or `Force single-rate` (that is, force the output sample rate to match the input sample rate by repeating every *K*th input sample *K* times at the output).

Frame-based mode
The method by which to implement downsampling for frame-based inputs: `Maintain input frame size` (that is, decrease the frame rate), or `Maintain input frame rate` (that is, decrease the frame size).

Supported Data Types

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

See Also

FIR Decimation	Signal Processing Blockset
FIR Rate Conversion	Signal Processing Blockset
Repeat	Signal Processing Blockset
Sample and Hold	Signal Processing Blockset
Upsample	Signal Processing Blockset

Purpose

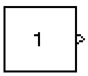
Generate discrete- or continuous-time constant signal

Library

Signal Processing Sources

`dspsrcs4`

Description



Note The DSP Constant block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Constantblock.

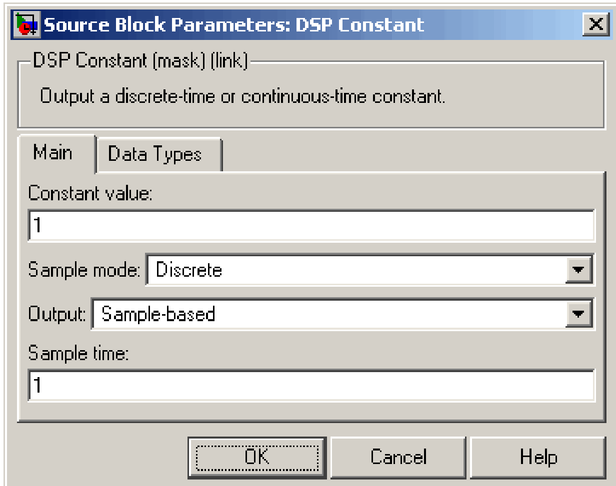
The DSP Constant block generates a signal whose value remains constant throughout the simulation. The **Constant value** parameter specifies the constant to output, and can be any valid MATLAB® expression that evaluates to a scalar, vector, or matrix.

When **Sample mode** is set to `Continuous`, the output is a continuous-time signal. When **Sample mode** is set to `Discrete`, the **Sample time** parameter is visible, and the signal has the discrete output period specified by the **Sample time** parameter.

You can set the output signal to `Frame-based`, `Sample-based`, or `Sample-based (interpret vectors as 1-D)` with the **Output** parameter.

Dialog Box

The **Main** pane of the DSP Constant block dialog box appears as follows.



Opening this dialog box causes a running simulation to pause. See "Changing Source Block Parameters During Simulation" in the online Simulink® documentation for details.

Constant value

Specify the constant to generate. This parameter is tunable; values entered here can be tuned, but their dimensions must remain fixed.

When you specify any data type information in this field, it is overridden by the value of the **Output data type** parameter in the **Data Types** pane, unless you select `Inherit from 'Constant value'`.

Sample mode

Specify the sample mode of the output, `Discrete` for a discrete-time signal or `Continuous` for a continuous-time signal.

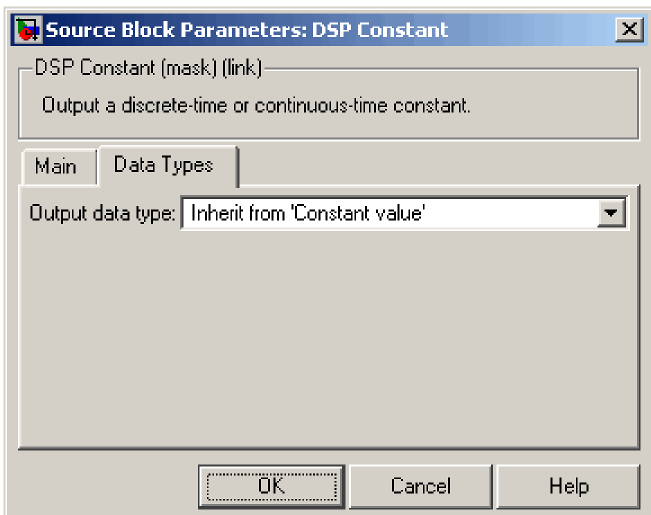
Output

Specify whether the output is `Sample-based (interpret vectors as 1-D)`, `Sample-based`, or `Frame-based`. When you select `Sample-based` and the output is a vector, its dimension is constrained to match the **Constant value** dimension (row or column). When you select `Sample-based (interpret vectors as 1-D)`, however, the output has no specified dimensionality.

Sample time

Specify the discrete sample period for sample-based outputs. When you select `Frame-based` for the **Output** parameter, this parameter is named **Frame period**, and is the discrete frame period for the frame-based output. This parameter is only visible when you select `Discrete` for the **Sample mode** parameter.

The **Data Types** pane of the DSP Constant block dialog box appears as follows.



Output data type

Specify the output data type in one of the following ways:

- Choose one of the built-in data types from the list.
- Choose `Fixed-point` to specify the output data type and scaling in the **Signed**, **Word length**, **Set fraction length in output to**, and **Fraction length** parameters.
- Choose `User-defined` to specify the output data type and scaling in the **User-defined data type**, **Set fraction length in output to,** and **Fraction length** parameters.
- Choose `Inherit from 'Constant value'` to set the output data type and scaling to match the values of the **Constant value** parameter in the **Main** pane.
- Choose `Inherit via back propagation` to set the output data type and scaling to match the following block.

The value of this parameter overrides any data type information specified in the **Constant value** parameter in the **Main** pane, except when you select `Inherit from 'Constant value'`.

Signed
Select to output a signed fixed-point signal. Otherwise, the signal is unsigned. This parameter is only visible when you select `Fixed-point` for the **Output data type** parameter.

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

User-defined data type
Specify any built-in or fixed-point data type. You can specify fixed-point data types using the following Simulink® Fixed Point™ functions: `sfix`, `ufix`, `sint`, `uint`, `sfrac`, and `ufrac`. This parameter is only visible when you select `User-defined` for the **Output data type** parameter.

Set fraction length in output to
Specify the scaling of the fixed-point output by either of the following two methods:

- Choose `Best precision` to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose `User-defined` to specify the output scaling in the **Fraction length** parameter.

This parameter is only visible when you select `Fixed-point` for the **Output data type** parameter, or when you select `User-defined` and the specified output data type is a fixed-point data type.

Fraction length

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

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point
- Boolean
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Constant	Simulink
Signal From Workspace	Signal Processing Blockset

Purpose

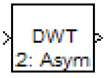
Compute discrete wavelet transform (DWT) of input

Library

Transforms

`dspxfrm3`

Description



Note The DWT block is the same as the Dyadic Analysis Filter Bank block in the Multirate Filters library, but with different default settings. See the Dyadic Analysis Filter Bank block reference page for more information on how to use the block.

The DWT block computes the discrete wavelet transform (DWT) of each column of a frame-based input. By default, the output is a sample-based vector or matrix with the same dimensions as the input. Each column of the output is the DWT of the corresponding input column.

You must install the Wavelet Toolbox™ product for the block to automatically design wavelet-based filters to compute the DWT. Otherwise, you must specify your own lowpass and highpass FIR filters by setting the **Filter** parameter to `User defined`.

For the same input, the DWT block and the Wavelet Toolbox function do not produce the same results. Because the blockset is designed for real-time implementation and the toolbox is designed for analysis, the products handle boundary conditions and filter states differently. To make the output of the `dwt` function and the DWT block match, complete the following steps:

1. For the `dwt` function, set the boundary condition to zero-padding by typing `dwtmode('zpd')` at the MATLAB® command prompt.

2 To match the latency of the DWT block, which is implemented using FIR filters, add zeros to the input of the `dwt` function. The number of zeros you add must be equal to the half the filter length.

For detailed information about how to use this block, see the Dyadic Analysis Filter Bank block reference page.

Examples

See "Examples" on page 2-410 in the Dyadic Analysis Filter Bank block reference page.

See Also

Dyadic Analysis Filter Bank	Signal Processing Blockset
IDWT	Signal Processing Blockset
`dwt`	Wavelet Toolbox

Dyadic Analysis Filter Bank

Purpose

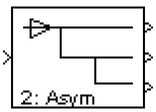
Decompose signals into subbands with smaller bandwidths and slower sample rates

Library

Filtering / Multirate Filters

`dspmlti4`

Description



Note This block decomposes frame-based signals with frame size a multiple of 2^n into either n+1 or 2^n subbands. To decompose sample-based signals or frame-based signals of different sizes, use the Two-Channel Analysis Subband Filter block. (You can connect multiple copies of the Two-Channel Analysis Subband Filter block to create a multilevel dyadic analysis filter bank.)

The Dyadic Analysis Filter Bank block decomposes a broadband signal into a collection of subbands with smaller bandwidths and slower sample rates. The block uses a series of highpass and lowpass FIR filters to repeatedly divide the input frequency range, as illustrated in the figure "n-Level Asymmetric Dyadic Analysis Filter Bank".

You can specify the filter bank's highpass and lowpass filters by providing vectors of filter coefficients. If you install the Wavelet Toolbox™ product, you can also specify wavelet-based filters by selecting a wavelet from the **Filter** parameter. You must set the filter bank structure to asymmetric or symmetric, and specify the number of levels in the filter bank.

Input Requirements

- Input can be a frame-based vector or frame-based matrix.
- The input frame size must be a multiple of 2^n, where n is the number of filter bank levels. For example, a frame size of 16 would be appropriate for a three-level tree (16 is a multiple of 2^3).
- The block always operates along the columns of the inputs.

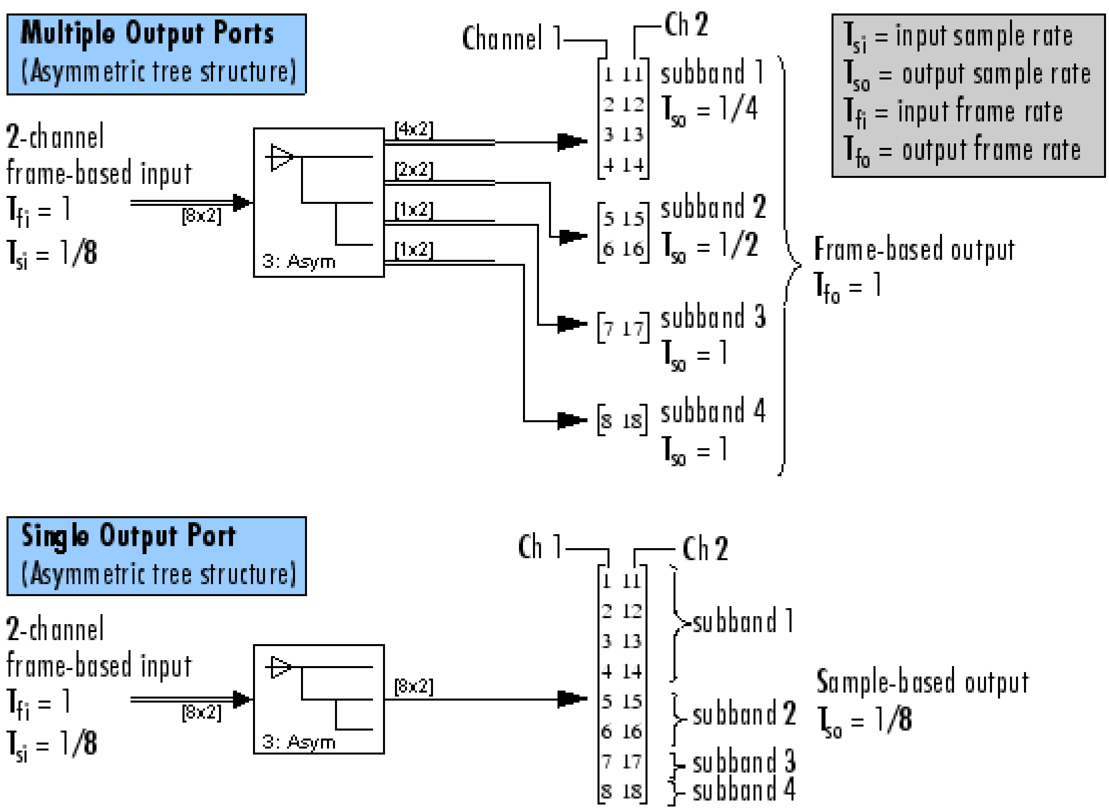
For an illustration of why the above input requirements exist, see the figure Outputs of a 3-Level Asymmetric Dyadic Analysis Filter Bank on page 2-406.

Output Characteristics

The output characteristics vary depending on the block's parameter settings, as summarized in the following list and figure:

- **Number of levels** parameter set to n
- **Tree structure** parameter setting:
  - `Asymmetric` — Block produces n+1 output subbands
  - `Symmetric` — Block produces 2n output subbands
- **Output** parameter setting can be `Multiple ports` or `Single port`. The following figure illustrates the difference between the two settings for a 3-level asymmetric dyadic analysis filter bank. For an explanation of the illustrated output characteristics, see the table Output Characteristics for an n-Level Dyadic Analysis Filter Bank on page 2-407.

For more information about the filter bank levels and structures, see "Dyadic Analysis Filter Banks".



Outputs of a 3-Level Asymmetric Dyadic Analysis Filter Bank

The following table summarizes the different output characteristics of the block when it is set to output from single or multiple ports.

Output Characteristics for an n-Level Dyadic Analysis Filter Bank

	Single Output Port	Multiple Output Ports
Output Description	Block concatenates all the subbands into one vector or matrix, and outputs the concatenated subbands from a single output port. Each output column contains subbands of the corresponding input channel.	Block outputs each subband from a separate output port. The topmost port outputs the subband with the highest frequencies. Each output column contains a subband for the corresponding input channel.
Output Frame Status	Sample-based	Frame-based
Output Frame Rate	*Not applicable*	Same as input frame rate (However, the output frame sizes can vary, so the output sample rates can vary.)

Output Characteristics for an n-Level Dyadic Analysis Filter Bank (Continued)

	Single Output Port	Multiple Output Ports
Output Dimensions (Frame Size)	Same number of rows and columns as the input.	The output has the same number of columns as the input. The number of output rows is the output frame size. For an input with frame size M_i output y_k has frame size $M_{o,k}$: • `Symmetric` — All outputs have the frame size, M_i / 2^n. • `Asymmetric` — The frame size of each output (except the last) is half that of the output from the previous level. The outputs from the last two output ports have the same frame size since they originate from the same level in the filter bank. $M_{o,k} = \begin{cases} M_i / 2^k & (1 \le k \le n) \\ M_i / 2^n & (k = n+1) \end{cases}$
Output Sample Rate	Same as input sample rate.	Though the outputs have the same frame rate as the input, they have different frame sizes than the input. Thus, the output sample rates, F_{so}, k, are different from the input sample rate, F_{si}: • `Symmetric` — All outputs have the sample rate F_{si} / 2^n. • `Asymmetric` — $F_{so,k} = \begin{cases} F_{si} / 2^k & (1 \le k \le n) \\ F_{si} / 2^n & (k = n+1) \end{cases}$

Filter Bank Filters

You must specify the highpass and lowpass filters in the filter bank by setting the **Filter** parameter to one of the following options:

- `User defined` — Allows you to explicitly specify the filters with two vectors of filter coefficients in the **Lowpass FIR filter coefficients** and **Highpass FIR filter coefficients** parameters. The block uses the same lowpass and highpass filters throughout the filter bank. The two filters should be halfband filters, where each filter passes the frequency band that the other filter stops.
- Wavelet such as `Biorthogonal` or `Daubechies` — The block uses the specified wavelet to construct the lowpass and highpass filters using the Wavelet Toolbox function `wfilters`. Depending on the wavelet, the block might enable either the **Wavelet order** or **Filter order [synthesis / analysis]** parameter. (The latter parameter allows you to specify different wavelet orders for the analysis and synthesis filter stages.) You must install the Wavelet Toolbox product to use wavelets.

Specifying Filters with the Filter Parameter and Related Parameters

Filter	Sample Setting for Related Filter Specification Parameters	Corresponding Wavelet Function Syntax
User-defined	Filters based on Daubechies wavelets with wavelet order 3: • **Highpass FIR filter coefficients =** `[-0.3327 0.8069 -0.4599 -0.1350 0.0854 0.0352]` • **Lowpass FIR filter coefficients =** `[0.0352 -0.0854 -0.1350 0.4599 0.8069 0.3327]`	None
Haar	None	`wfilters('haar')`

Specifying Filters with the Filter Parameter and Related Parameters (Continued)

Filter	Sample Setting for Related Filter Specification Parameters	Corresponding Wavelet Function Syntax
Daubechies	**Wavelet order =** `4`	`wfilters('db4')`
Symlets	**Wavelet order =** `3`	`wfilters('sym3')`
Coiflets	**Wavelet order =** `1`	`wfilters('coif1')`
Biorthogonal	**Filter order [synthesis / analysis] =** `[3/1]`	`wfilters('bior3.1')`
Reverse Biorthogonal	**Filter order [synthesis / analysis] =** `[3/1]`	`wfilters('rbio3.1')`
Discrete Meyer	None	`wfilters('dmey')`

Examples

Wavelets

The primary application for dyadic analysis filter banks and dyadic synthesis filter banks, is coding for data compression using wavelets.

At the transmitting end, the output of the dyadic analysis filter bank is fed to a lossy compression scheme, which typically assigns the number of bits for each filter bank output in proportion to the relative energy in that frequency band. This represents the more powerful signal components by a greater number of bits than the less powerful signal components.

At the receiving end, the transmission is decoded and fed to a dyadic synthesis filter bank to reconstruct the original signal. The filter coefficients of the complementary analysis and synthesis stages are designed to cancel aliasing introduced by the filtering and resampling.

Demos

See the following Signal Processing Blockset™ demos, which use the Dyadic Analysis Filter Bank block:

- Multi-level PR filter bank
- Denoising
- Wavelet transmultiplexer (WTM)

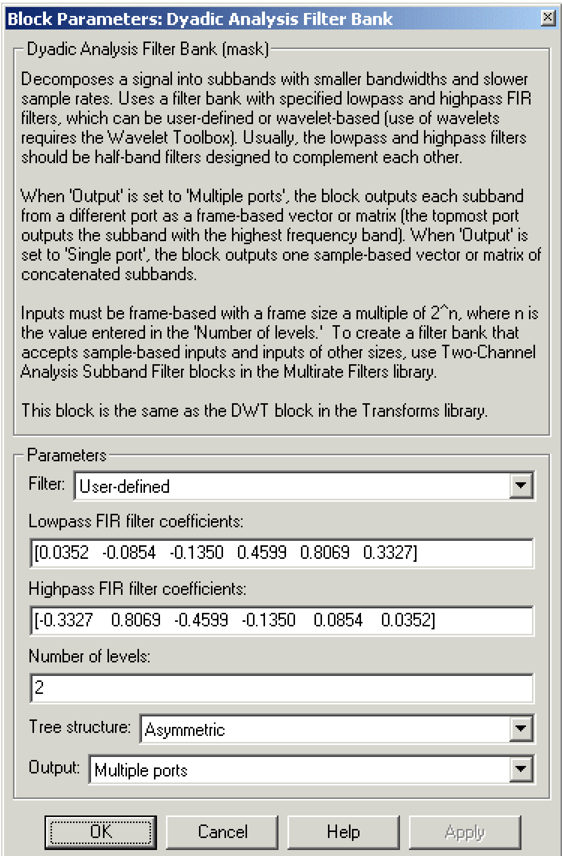
Note To see the version of the demos that use the Dyadic Analysis Filter Bank and Dyadic Synthesis Filter Bank blocks, click the **Frame-Based Demo** button in the demos.

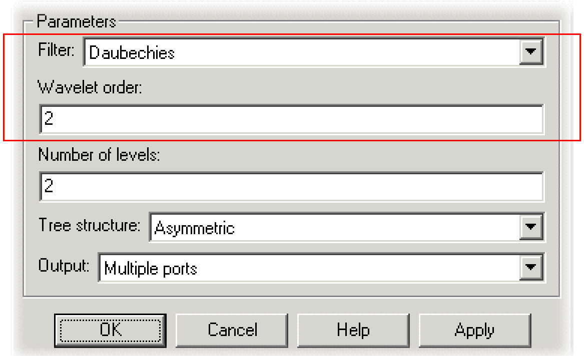
Open the demos using one of the following methods:

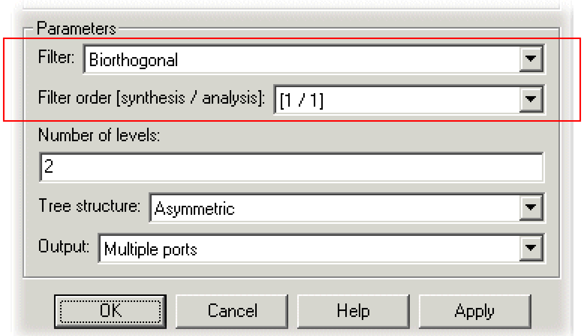
- Click the above links in the MATLAB® Help browser (*not* in a Web browser).
- Type `demo blockset dsp` at the MATLAB command line, and look in the `Wavelets` directory.

Dialog Box

The parameters displayed in the block dialog vary depending on the setting of the **Filter** parameter. Only some of the parameters described below are visible in the dialog box at any one time.







Filter

The type of filter used to determine the high- and low-pass FIR filters in the dyadic analysis filter bank:

Select `User defined` to explicitly specify the filter coefficients in the **Lowpass FIR filter coefficients** and **Highpass FIR filter coefficients** parameters.

Select a wavelet such as `Biorthogonal` or `Daubechies` to specify a wavelet-based filter. The block uses the Wavelet Toolbox function `wfilters` to construct the filters. Extra parameters such as **Wavelet order** or **Filter order [synthesis / analysis]** might become enabled. For a list of the supported wavelets, see Specifying Filters with the Filter Parameter and Related Parameters on page 2-409.

Lowpass FIR filter coefficients

A vector of filter coefficients (descending powers of z) that specifies coefficients used by all the lowpass filters in the filter bank. This parameter is enabled when you set **Filter** to `User defined`. The lowpass filter should be a half-band filter that passes the frequency band stopped by the filter specified in the **Highpass FIR filter coefficients** parameter. The default values of this parameter specify a filter based on a Daubechies wavelet with wavelet order `3`.

Highpass FIR filter coefficients

A vector of filter coefficients (descending powers of z) that specifies coefficients used by all the highpass filters in the filter bank. This parameter is enabled when you set **Filter** to `User defined`. The highpass filter should be a half-band filter that passes the frequency band stopped by the filter specified in the **Lowpass FIR filter coefficients** parameter. The default values of this parameter specify a filter based on a Daubechies wavelet with wavelet order `3`.

Wavelet order

The order of the wavelet selected in the **Filter** parameter. This parameter is enabled only when you set **Filter** to certain types of wavelets, as shown in the Specifying Filters with the Filter Parameter and Related Parameters table.

Filter order [synthesis / analysis]
: The order of the wavelet for the synthesis and analysis filter stages. For example, when you set the **Filter** parameter to `Biorthogonal` and set the **Filter order [synthesis / analysis]** parameter to `[2 / 6]`, the block calls the `wfilters` function with input argument `'bior2.6'`. This parameter is enabled only when you set **Filter** to certain types of wavelets, as shown in Specifying Filters with the Filter Parameter and Related Parameters on page 2-409.

Number of levels
: The number of filter bank levels. An n-level asymmetric structure has n+1 outputs, and an n-level symmetric structure has 2^n outputs, as shown in the figures "n-Level Asymmetric Dyadic Analysis Filter Bank" and "n-Level Symmetric Dyadic Analysis Filter Bank". The block's icon displays the value of this parameter in the lower-left corner.

Tree structure
: The structure of the filter bank: `Asymmetric`, or `Symmetric`. See the figures "n-Level Asymmetric Dyadic Analysis Filter Bank" and "n-Level Symmetric Dyadic Analysis Filter Bank".

Output
: Set to `Multiple ports` to output each output subband on a separate port (the topmost port outputs the subband with the highest frequency band). Set to `Single port` to concatenate the subbands into one vector or matrix and output the concatenated subbands on a single port. For more information, see "Output Characteristics" on page 2-405.

References

Fliege, N. J. *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets*. West Sussex, England: John Wiley & Sons, 1994.

Strang, G. and T. Nguyen. *Wavelets and Filter Banks*. Wellesley, MA: Wellesley-Cambridge Press, 1996.

Vaidyanathan, P. P. *Multirate Systems and Filter Banks*. Englewood Cliffs, NJ: Prentice Hall, 1993.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Dyadic Synthesis Filter Bank	Signal Processing Blockset
Two-Channel Analysis Subband Filter	Signal Processing Blockset

Dyadic Synthesis Filter Bank

Purpose

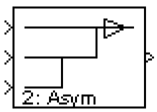
Reconstruct signals from subbands with smaller bandwidths and slower sample rates

Library

Filtering / Multirate Filters

`dspmlti4`

Description



Note This block always outputs frame-based signals, and its inputs must be of certain sizes. To get sample-based outputs or to use input subbands that do not fit the criteria of this block, use the Two-Channel Synthesis Subband Filter block. (You can connect multiple copies of the Two-Channel Synthesis Subband Filter block to create a multilevel dyadic synthesis filter bank.)

The Dyadic Synthesis Filter Bank block reconstructs a signal decomposed by the Dyadic Analysis Filter Bank block. The block takes in subbands of a signal, and uses them to reconstruct the signal by using a series of highpass and lowpass FIR filters as illustrated in the figure "n-Level Asymmetric Dyadic Synthesis Filter Bank". The reconstructed signal has a wider bandwidth and faster sample rate than the input subbands.

You can specify the filter bank's highpass and lowpass filters by providing vectors of filter coefficients. If you install the Wavelet Toolbox™ product, you can also specify wavelet-based filters by selecting a wavelet from the **Filter** parameter.

Note To use a dyadic synthesis filter bank to perfectly reconstruct the output of a dyadic analysis filter bank, the number of levels and tree structures of both filter banks *must* be the same. In addition, the filters in the synthesis filter bank *must* be designed to perfectly reconstruct the outputs of the analysis filter bank. Otherwise, the reconstruction is not perfect.

This block automatically computes wavelet-based perfect reconstruction filters when the wavelet selection in the **Filter** parameter of this block is the *same* as the **Filter** parameter setting of the corresponding Dyadic Analysis Filter Bank block. The use of wavelets requires the Wavelet Toolbox product. To learn how to design your own perfect reconstruction filters, see "References" on page 2-426.

Input Requirements

The inputs to this block are usually the outputs of a Dyadic Analysis Filter Bank block. Since the Dyadic Analysis Filter Bank block can output from either a single port or multiple ports, the Dyadic Synthesis Filter Bank block accepts inputs to either a single port or multiple ports.

The **Input** parameter sets whether the block accepts inputs from a single port or multiple ports, and thus determines the input requirements, as summarized in the following lists and figure.

Note Any output of a Dyadic Analysis Filter Bank block whose parameter settings match the corresponding settings of this block is a valid input to this block. For example, the setting of the Dyadic Analysis Filter Bank block parameter, **Output**, must be the same as this block's **Input** parameter (`Single port` or `Multiple ports`).

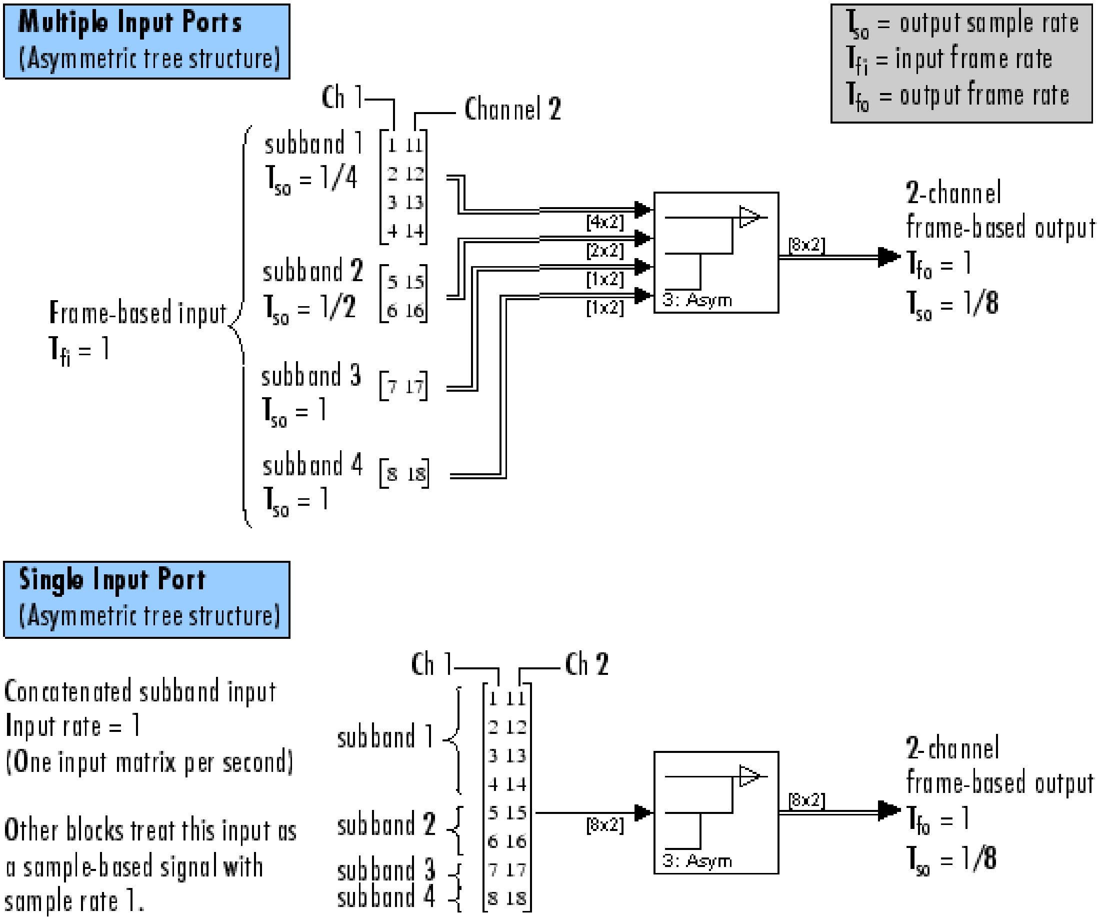
Valid Inputs for Input Set to Single Port

- Inputs must be sample-based vectors or sample-based matrices of concatenated subbands.

- Each input column contains the subbands for an independent signal.
- Upper input rows contain the high-frequency subbands, and the lower rows contain the low-frequency subbands.

Valid Inputs for Input Set to Multiple Ports

- Inputs must be a frame-based vector or frame-based matrix for each subband, each of which is input to a separate input port.
- The columns of each input contains a subband for an independent signal.
- The input to the topmost input port is the subband containing the highest frequencies, and the input to the bottommost port is the subband containing the lowest frequencies.



Valid Inputs to a 3-Level Asymmetric Dyadic Synthesis Filter Bank

For general information about the filter banks, see "Dyadic Synthesis Filter Banks".

Output Characteristics

The following table summarizes the output characteristics for both frame-based inputs, and concatenated subband inputs. For an illustration of why the output characteristics exist, see the figure Valid Inputs to a 3-Level Asymmetric Dyadic Synthesis Filter Bank on page 2-419.

	Frame-Based Inputs (Input = Multiple ports)	**Concatenated Subband Inputs (Input = Single port)**
Output Frame Status	Outputs are always frame based regardless of the input frame status. Each output column is an independent channel, reconstructed from the corresponding channel in the inputs.	
Output Frame Rate	Same as the input frame rate.	Same as the input rate (the rate of the concatenated subband inputs).
Output Frame Dimensions	• The output has the same number of columns as the inputs. • The number of output rows depends on the tree structure of the filter bank: - **Asymmetric** — The number of output rows is twice the number of rows in the input to the topmost input port. - **Symmetric** — The number of output rows is the product of the number of input ports and the number of rows in an input to any input port.	The output has the same number of rows and columns as the input.

For general information about the filter banks, see “Dyadic Synthesis Filter Banks”.

Filter Bank Filters

You must specify the highpass and lowpass filters in the filter bank by setting the **Filter** parameter to one of the following options:

- `User defined` — Allows you to explicitly specify the filters with two vectors of filter coefficients in the **Lowpass FIR filter coefficients** and **Highpass FIR filter coefficients** parameters. The block uses the same lowpass and highpass filters throughout the filter bank. The two filters should be halfband filters, where each filter passes the frequency band that the other filter stops. To use this block to perfectly reconstruct a signal decomposed by a Dyadic Analysis Filter Bank block, the filters in this block *must* be designed to perfectly reconstruct the outputs of the analysis filter bank. To learn how to design your own perfect reconstruction filters, see "References" on page 2-426.

- Wavelet such as `Biorthogonal` or `Daubechies` — The block uses the specified wavelet to construct the lowpass and highpass filters using the Wavelet Toolbox function `wfilters`. Depending on the wavelet, the block might enable either the **Wavelet order** or **Filter order [synthesis / analysis]** parameter. (The latter parameter allows you to specify different wavelet orders for the analysis and synthesis filter stages.) To use this block to reconstruct a signal decomposed by a Dyadic Analysis Filter Bank block, you must set both blocks to use the same wavelets with the same order. You must install the Wavelet Toolbox product to use wavelets.

Specifying Filters with the Filter Parameter and Related Parameters

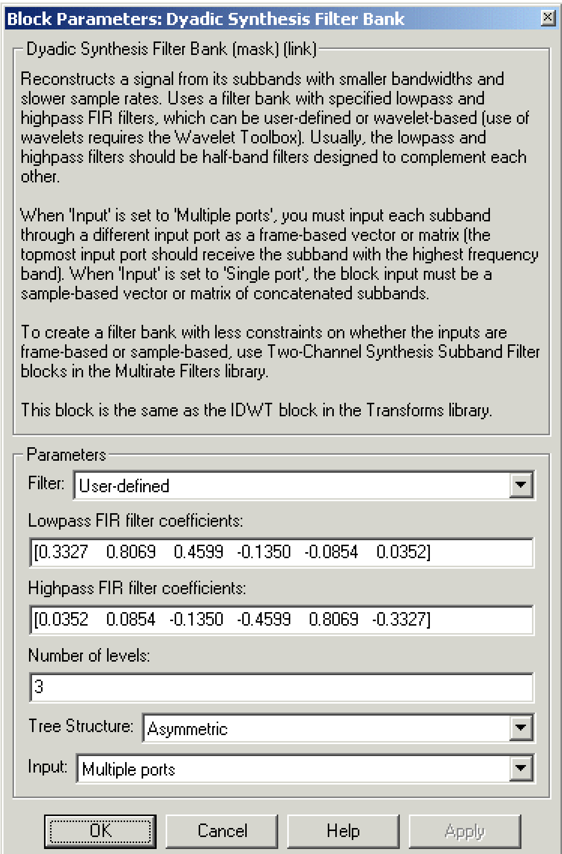
Filter	Sample Setting for Related Filter Specification Parameters	Corresponding Wavelet Function Syntax
User-defined	Filters based on Daubechies wavelets with wavelet order 3: • **Lowpass FIR filter coefficients =** `[0.0352 -0.0854 -0.1350 0.4599 0.8069 0.3327]` • **Highpass FIR filter coefficients =** `[-0.3327 0.8069 -0.4599 -0.1350 0.0854 0.0352]`	None
Haar	None	`wfilters('haar')`
Daubechies	**Wavelet order =** 4	`wfilters('db4')`
Symlets	**Wavelet order =** 3	`wfilters('sym3')`
Coiflets	**Wavelet order =** 1	`wfilters('coif1')`
Biorthogonal	**Filter order [synthesis / analysis] =** `[3/1]`	`wfilters('bior3.1')`
Reverse Biorthogonal	**Filter order [synthesis / analysis] =** `[3/1]`	`wfilters('rbio3.1')`
Discrete Meyer	None	`wfilters('dmey')`

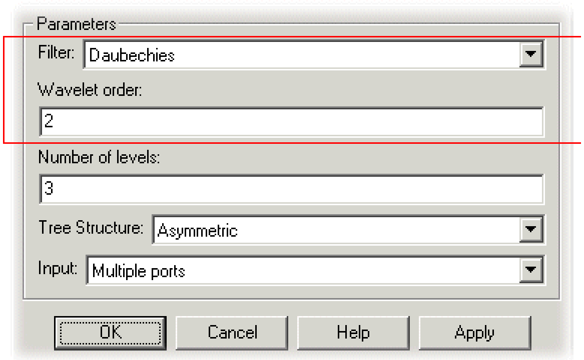
Examples

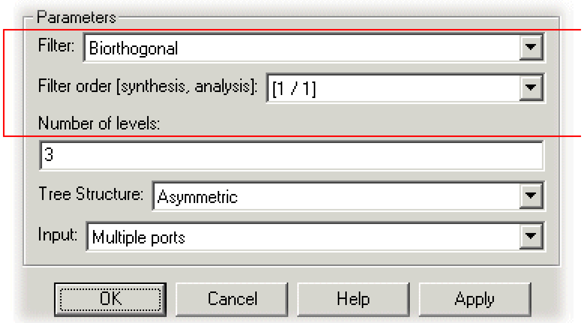
See "Examples" on page 2-410 in the Dyadic Analysis Filter Bank block reference.

Dialog Box

The parameters displayed in the block dialog vary depending on the setting of the **Filter** parameter. Only some of the parameters described below are visible in the dialog box at any one time.







Note To use this block to reconstruct a signal decomposed by a Dyadic Analysis Filter Bank block, all the parameters in this block must be the same as the corresponding parameters in the Dyadic Analysis Filter Bank block (except the **Lowpass FIR filter coefficients** and **Highpass FIR filter coefficients**; see the descriptions of these parameters).

Filter

The type of filter used to determine the high- and low-pass FIR filters in the dyadic synthesis filter bank:

- Select `User defined` to explicitly specify the filter coefficients in the **Lowpass FIR filter coefficients** and **Highpass FIR filter coefficients** parameters.
- Select a wavelet such as `Biorthogonal` or `Daubechies` to specify a wavelet-based filter. The block uses the Wavelet Toolbox function `wfilters` to construct the filters. Extra parameters such as **Wavelet order** or **Filter order [synthesis / analysis]** might become enabled. For a list of the supported wavelets, see the table Specifying Filters with the Filter Parameter and Related Parameters on page 2-422.

Lowpass FIR filter coefficients

A vector of filter coefficients (descending powers of z) that specifies coefficients used by all the lowpass filters in the filter bank. This parameter is enabled when you set **Filter** to `User defined`. The lowpass filter should be a half-band filter that passes the frequency band stopped by the filter specified in the **Highpass FIR filter coefficients** parameter. To perfectly reconstruct a signal decomposed by the Dyadic Analysis Filter Bank, the filters in this block *must* be designed to perfectly reconstruct the outputs of the analysis filter bank. Otherwise, the reconstruction is not perfect. The default values of this parameter specify a perfect reconstruction filter for the default settings of the Dyadic Analysis Filter Bank (based on a Daubechies wavelet with wavelet order 3).

Highpass FIR filter coefficients

A vector of filter coefficients (descending powers of *z*) that specifies coefficients used by all the highpass filters in the filter bank. This parameter is enabled when you set **Filter** to `User defined`. The highpass filter should be a half-band filter that passes the frequency band stopped by the filter specified in the **Lowpass FIR filter coefficients** parameter. To perfectly reconstruct a signal decomposed by the Dyadic Analysis Filter Bank, the filters in this block *must* be designed to perfectly reconstruct the outputs of the analysis filter bank. Otherwise, the reconstruction is not perfect. The default values of this parameter specify a perfect reconstruction filter for the default settings of the Dyadic Analysis Filter Bank (based on a Daubechies wavelet with wavelet order `3`).

Wavelet order

The order of the wavelet selected in the **Filter** parameter. This parameter is enabled only when you set **Filter** to certain types of wavelets, as shown in the table Specifying Filters with the Filter Parameter and Related Parameters on page 2-422.

Filter order [synthesis / analysis]

The order of the wavelet for the synthesis and analysis filter stages. For example, when you set the **Filter** parameter to **Biorthogonal** and set the **Filter order [synthesis / analysis]** parameter to `[2 / 6]`, the block calls the `wfilters` function with input argument `'bior2.6'`. This parameter is enabled only when you set **Filter** to certain types of wavelets, as shown in Specifying Filters with the Filter Parameter and Related Parameters on page 2-422.

Number of levels

The number of filter bank levels. An *n*-level asymmetric structure has n+1 outputs, and an *n*-level symmetric structure has 2^n outputs, as shown in "n-Level Asymmetric Dyadic Synthesis Filter Bank" and "n-Level Symmetric Dyadic Synthesis Filter Bank".

Tree structure

The structure of the filter bank: `Asymmetric`, or `Symmetric`. See the figures "n-Level Asymmetric Dyadic Synthesis Filter Bank" and "n-Level Symmetric Dyadic Synthesis Filter Bank".

Input

Set to `Multiple ports` to accept each input subband at a separate port (the topmost port accepts the subband with the highest frequency band). Set to `Single port` to accept one vector or matrix of concatenated subbands at a single port. For more information, see "Input Requirements" on page 2-417.

References

Fliege, N. J. *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets*. West Sussex, England: John Wiley & Sons, 1994.

Strang, G. and T. Nguyen. *Wavelets and Filter Banks*. Wellesley, MA: Wellesley-Cambridge Press, 1996.

Vaidyanathan, P. P. *Multirate Systems and Filter Banks*. Englewood Cliffs, NJ: Prentice Hall, 1993.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Dyadic Analysis Filter Bank	Signal Processing Blockset
Two-Channel Synthesis Subband Filter	Signal Processing Blockset

See "Multirate Filters" for related information.

Purpose

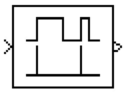
Detect transition from zero to nonzero value

Library

Signal Management / Switches and Counters

`dspswit3`

Description

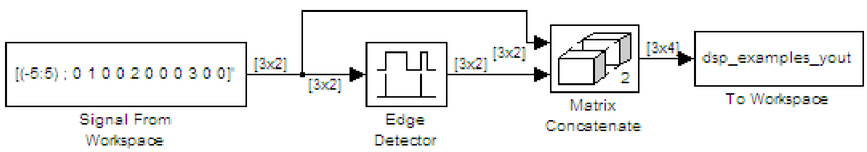


The Edge Detector block generates an impulse (the value 1) in a given output channel when the corresponding channel of the input transitions from zero to a nonzero value. Otherwise, the block generates zeros in each channel.

The output has the same dimension and sample rate as the input. When the input is frame based, the output is frame based; otherwise, the output is sample based. For frame-based input, an edge that is split across two consecutive frames (that is, a zero at the bottom of the first frame, and a nonzero value at the top of the following frame) is counted in the frame that contains the nonzero value.

Examples

In the model below, the Edge Detector block locates the edges (zero to nonzero transitions) in a two-channel frame-based input with frame size 3. The two input channels are horizontally concatenated with the two output channels to create the four-channel workspace variable `dsp_examples_yout`.



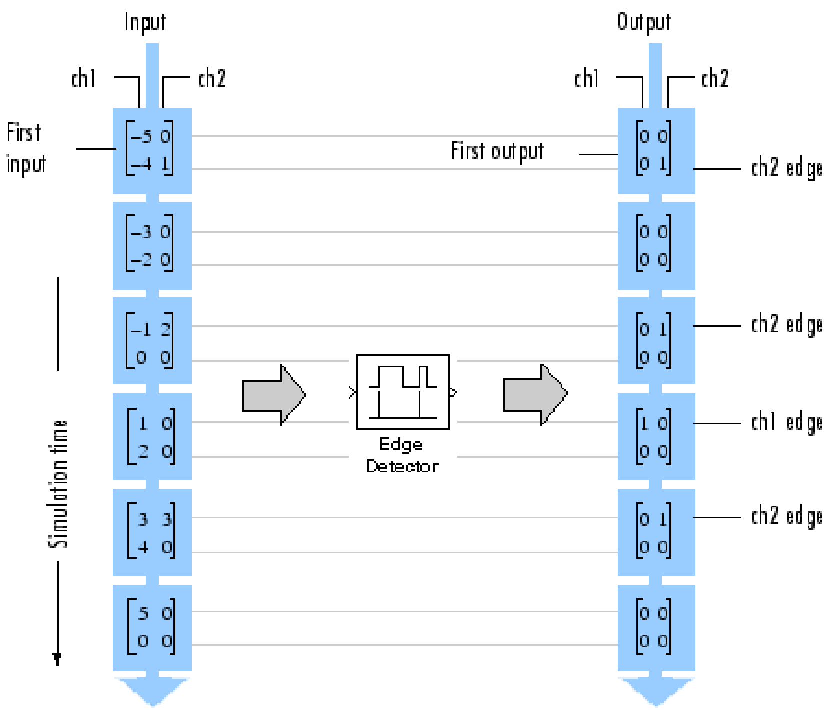
Adjust the block parameters as described below. (Use the default settings for the To Workspace block.)

- Set the Signal From Workspace block parameters as follows:
  - **Signal** = `[(-5:5) ; 0 1 0 0 2 0 0 0 3 0 0]'`
  - **Sample time** = `1`

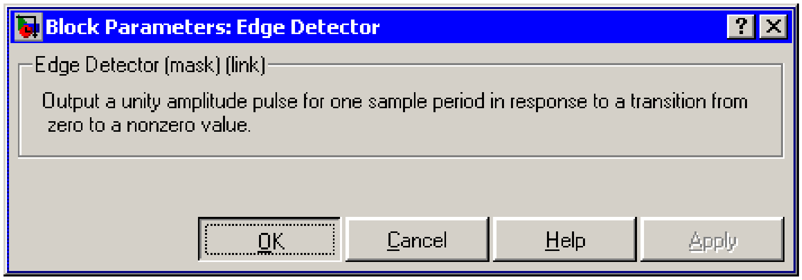
- **Samples per frame** = 3

- Set the Matrix Concatenate block parameters as follows:
  - **Number of inputs** = 2
  - **Mode** = `Multidimensional array`
  - **Concatenate dimension** = 2

As shown below, the block finds edges at sample 7 in channel 1, and at samples 2, 5, and 9 in channel 2.



Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- Boolean — The block might output Boolean values depending on the input data type, and whether Boolean support is enabled or disabled, as described in “Effects of Enabling and Disabling Boolean Support”. To learn how to disable Boolean output support, see “Steps to Disabling Boolean Support”.
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Counter	Signal Processing Blockset
Event-Count Comparator	Signal Processing Blockset

Event-Count Comparator

Purpose

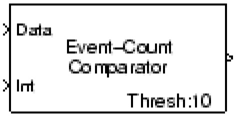
Detect threshold crossing of accumulated nonzero inputs

Library

Signal Management / Switches and Counters

`dspswit3`

Description



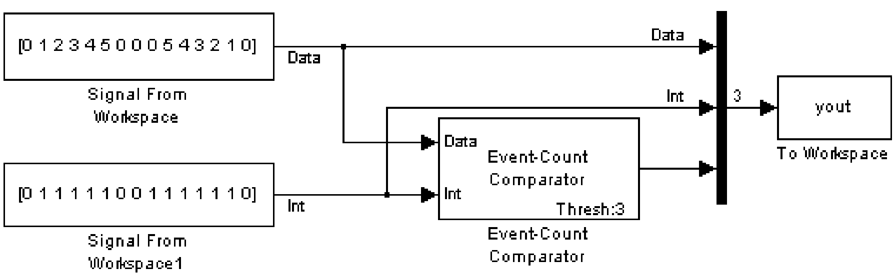
The Event-Count Comparator block records the number of nonzero inputs to the Data port during the period that the block is enabled by a high signal (the value 1) at the Int port. Both inputs must be scalars; the input to the Int port can be sample based or frame based. When the input to the Data port is frame based, the output is frame based; otherwise, the output is sample based.

When the number of accumulated nonzero inputs first equals the **Event threshold** setting, the block waits one additional sample interval, and then sets the output high (1). The block holds the output high until recording is restarted by a low-to-high (0-to-1) transition at the Int port.

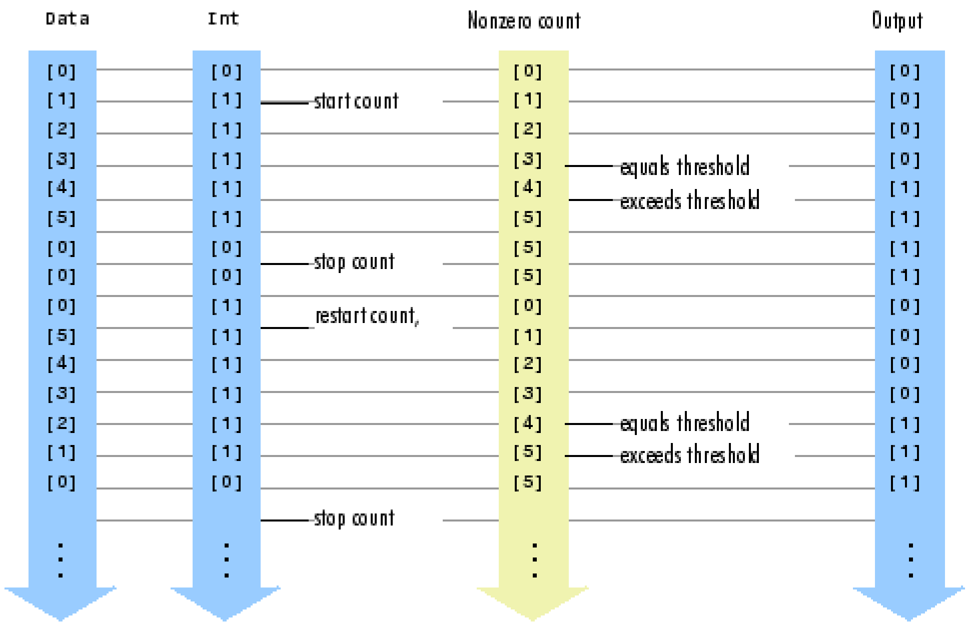
The Event-Count Comparator block accepts real and complex floating-point and fixed-point inputs. However, because the block has discrete state, it does not support constant or continuous sample times. Therefore, at least one input or output port of the Event-Count Comparator block must be connected to a block whose **Sample time** parameter is discrete. The Event-Count Comparator block inherits this non-infinite discrete sample time.

Examples

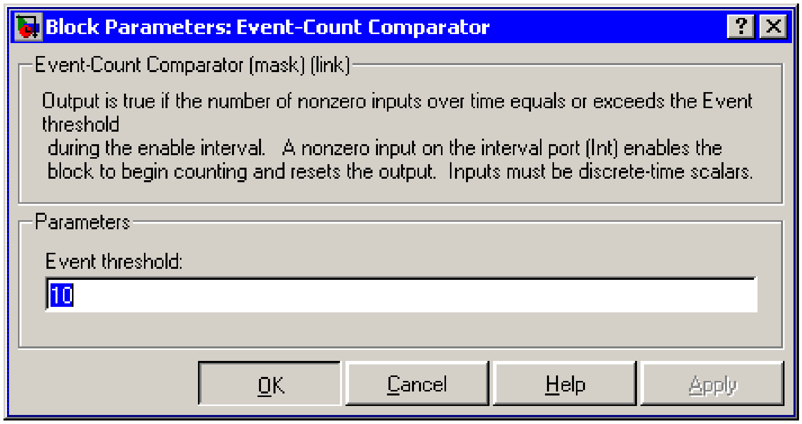
In the model below, the Event-Count Comparator block (**Event threshold** = 3) detects two threshold crossings in the input to the Data port, one at sample 4 and one at sample 12.



All inputs and outputs are multiplexed into the workspace variable `yout`, whose contents are shown in the figure below. The two left columns in the illustration show the inputs to the Data and Int ports, the center column shows the state of the block's internal counter, and the right column shows the block's output.



Dialog Box



Event threshold

Specify the value against which to compare the number of nonzero inputs. Tunable.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- Boolean
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Counter	Signal Processing Blockset
Edge Detector	Signal Processing Blockset

Purpose

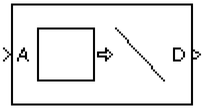
Extract main diagonal of input matrix

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description

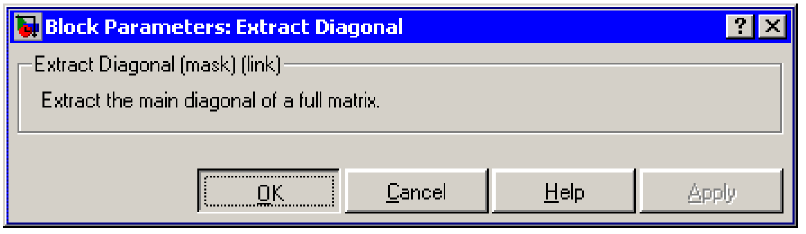


The Extract Diagonal block populates the 1-D output vector with the elements on the main diagonal of the *M*-by-*N* input matrix `A`.

```
D = diag(A)      Equivalent MATLAB code
```

The output vector has length `min(M,N)`, and is always sample based.

Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- Boolean — Block outputs are always Boolean. To learn how to disable Boolean support, see “Steps to Disabling Boolean Support”.
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Constant Diagonal Matrix	Signal Processing Blockset
Create Diagonal Matrix	Signal Processing Blockset

Extract Triangular Matrix	Signal Processing Blockset
`diag`	MATLAB

Purpose

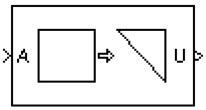
Extract lower or upper triangle from input matrices

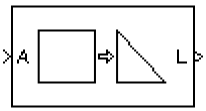
Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description





The Extract Triangular Matrix block creates a triangular matrix output from the upper or lower triangular elements of an M-by-N input matrix. A length-M 1-D vector input is treated as an M-by-1 matrix.

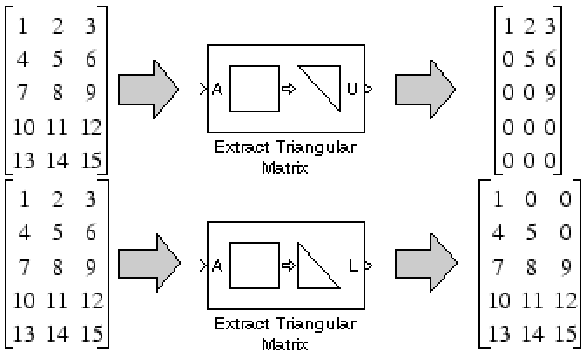
The **Extract** parameter selects between the two components of the input:

- `Upper` — Copies the elements on and above the main diagonal of the input matrix to an output matrix of the same size. The first *row* of the output matrix is therefore identical to the first *row* of the input matrix. The elements below the main diagonal of the output matrix are zero.
- `Lower` — Copies the elements on and below the main diagonal of the input matrix to an output matrix of the same size. The first *column* of the output matrix is therefore identical to the first *column* of the input matrix. The elements above the main diagonal of the output matrix are zero.

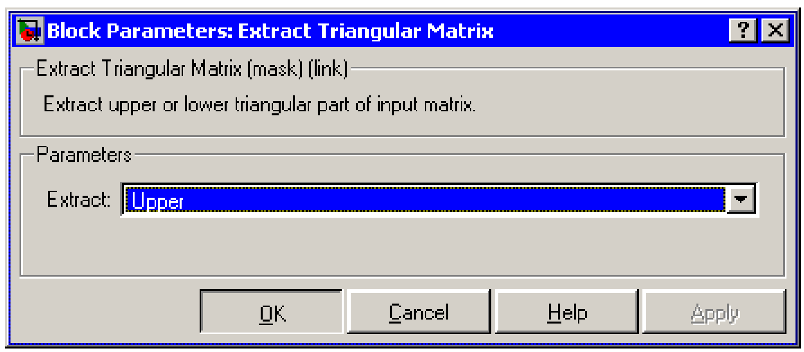
The output has the same frame status as the input.

Examples

The example below shows the extraction of upper and lower triangles from a 5-by-3 input matrix.



Dialog Box



Extract
: The component of the matrix to copy to the output, upper triangle or lower triangle.

Supported Data Types

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

See Also

Autocorrelation LPC	Signal Processing Blockset
Cholesky Factorization	Signal Processing Blockset
Constant Diagonal Matrix	Signal Processing Blockset
Extract Diagonal	Signal Processing Blockset
Forward Substitution	Signal Processing Blockset
LDL Factorization	Signal Processing Blockset
LU Factorization	Signal Processing Blockset
`tril`	MATLAB
`triu`	MATLAB

Purpose

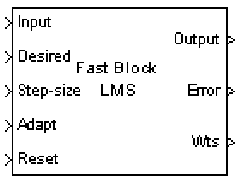
Compute filtered output, filter error, and filter weights for given input and desired signal using Fast Block LMS adaptive filter algorithm

Library

Filtering / Adaptive Filters

`dspadpt3`

Description



Fast Block LMS Filter

The Fast Block LMS Filter block implements an adaptive least mean-square (LMS) filter, where the adaptation of the filter weights occurs once for every block of data samples. The block estimates the filter weights, or coefficients, needed to convert the input signal into the desired signal. Connect the signal you want to filter to the Input port. This input signal can be a sample-based scalar or a single-channel frame-based signal. Connect the signal you want to model to the Desired port. The desired signal must have the same data type, frame status, complexity, and dimensions as the input signal. The Output port outputs the filtered input signal, which can be sample or frame based. The Error port outputs the result of subtracting the output signal from the desired signal.

The block calculates the filter weights using the Block LMS Filter equations. For more information, see Block LMS Filter. The Fast Block LMS Filter block implements the convolution operation involved in the calculations of the filtered output, y, and the weight update function in the frequency domain using the FFT algorithm used in the Overlap-Save FFT Filter block. See Overlap-Save FFT Filter for more information.

Use the **Filter length** parameter to specify the length of the filter weights vector.

The **Block size** parameter determines how many samples of the input signal are acquired before the filter weights are updated. The input frame length must be a multiple of the **Block size** parameter.

The **Step-size (mu)** parameter corresponds to µ in the equations. You can either specify a step-size using the input port, Step-size, or enter a value in the Block Parameters: Block LMS Filter dialog box.

Use the **Leakage factor (0 to 1)** parameter to specify the leakage factor, $0 < 1 - \mu\alpha \le 1$, in the leaky LMS algorithm shown below.

$$\mathbf{w}(k) = (1 - \mu\alpha)\mathbf{w}(k-1) - f(\mathbf{u}(n), e(n), \mu)$$

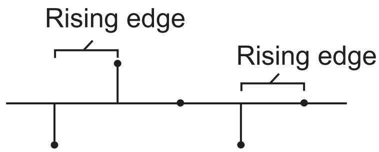
Enter the initial filter weights, $\mathbf{w}(0)$, as a vector or a scalar in the **Initial value of filter weights** text box. When you enter a scalar, the block uses the scalar value to create a vector of filter weights. This vector has length equal to the filter length and all of its values are equal to the scalar value.

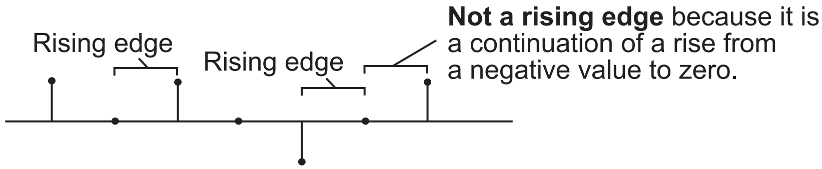
When you select the **Adapt port** check box, an Adapt port appears on the block. When the input to this port is nonzero, the block continuously updates the filter weights. When the input to this port is zero, the filter weights remain at their current values.

When you want to reset the value of the filter weights to their initial values, use the **Reset input** parameter. The block resets the filter weights whenever a reset event is detected at the Reset port. The reset signal rate must be the same rate as the data signal input.

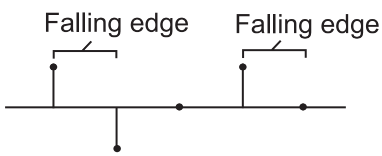
From the **Reset input** list, select `None` to disable the Reset port. To enable the Reset port, select one of the following from the **Reset input** list:

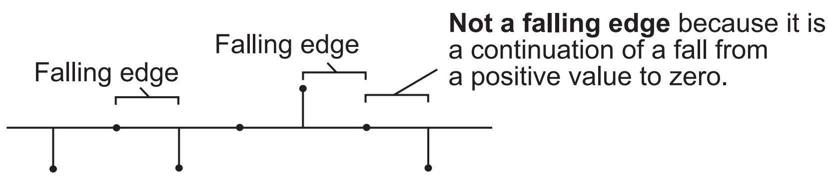
- `Rising edge` — Triggers a reset operation when the Reset input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)





- `Falling edge` — Triggers a reset operation when the Reset input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)





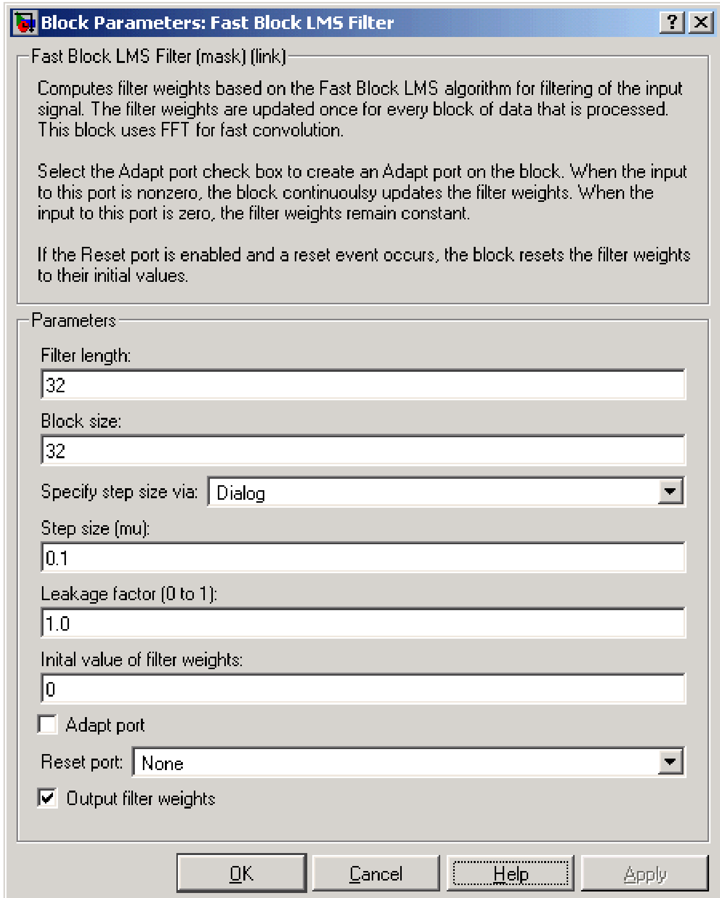
- `Either edge` — Triggers a reset operation when the Reset input is a `Rising edge` or `Falling edge` (as described above)
- `Non-zero sample` — Triggers a reset operation at each sample time that the Reset input is not zero

---

Note When running simulations in the Simulink® `MultiTasking` mode, sample-based reset signals have a one-sample latency, and frame-based reset signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a reset event, and when it applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Select the **Output filter weights** check box to create a Wts port on the block. For each iteration, the block outputs the current updated filter weights from this port.

Dialog Box



Filter length

Enter the length of the FIR filter weights vector. The sum of the block size and the filter length must be a power of 2.

Block size

Enter the number of samples to acquire before the filter weights are updated. The input frame length must be an integer multiple of the block size. The sum of the block size and the filter length must be a power of 2.

Specify step-size via
Select `Dialog` to enter a value for mu, or select `Input port` to specify mu using the Step-size input port.

Step-size (mu)
Enter the step-size. Tunable.

Leakage factor (0 to 1)
Enter the leakage factor, $0 < 1 - \mu\alpha \leq 1$. Tunable.

Initial value of filter weights
Specify the initial values of the FIR filter weights.

Adapt port
Select this check box to enable the Adapt input port.

Reset input
Select this check box to enable the Reset input port.

Output filter weights
Select this check box to export the filter weights from the Wts port.

References

Hayes, M.H. *Statistical Digital Signal Processing and Modeling.* New York: John Wiley & Sons, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Kalman Adaptive Filter	Signal Processing Blockset
LMS Filter	Signal Processing Blockset
RLS Filter	Signal Processing Blockset
Fast Block LMS Filter	Signal Processing Blockset
Overlap-Save FFT Filter	Signal Processing Blockset

See "Adaptive Filters" for related information.

Purpose

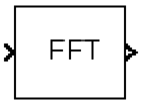
Compute fast Fourier transform (FFT) of input

Library

Transforms

`dspxfrm3`

Description



The FFT block computes the fast Fourier transform (FFT) of each channel of a P-by-N or length-P input, u. When the **Inherit FFT length from input dimensions** check box is selected, the input length P must be an integer power of two, and the FFT length M is equal to P. When the check box is not selected, P can be any length, and the value of the **FFT length** parameter must be a positive integer power of two. For user-specified FFT lengths, when M is not equal to P, zero padding or modulo-M data wrapping happens before the FFT operation, as per Orfanidis [1]:

```
y = fft(u,M)  % P ≤ M

y(:,l) = fft(datawrap(u(:,l),M))  % P > M; l = 1,...,N
```

To get zero padding or truncation rather than zero padding or wrapping, use a Pad block before the FFT block in your model to obtain a power-of-two input length.

The kth entry of the lth output channel, $y(k, l)$, is equal to the kth point of the M-point discrete Fourier transform (DFT) of the lth input channel:

$$y(k,l) = \sum_{p=1}^{P} u(p,l)e^{-j2\pi(p-1)(k-1)/M} \qquad k = 1,\ldots,M$$

This block supports real and complex floating-point and fixed-point inputs.

Input and Output Characteristics

The following table describes valid inputs to the FFT block, their corresponding outputs, and the dimension along which the block computes the DFT.

- Valid inputs to the FFT block. They can be real- or complex-valued, and they must be in linear order.
- The dimension along which the block computes the DFT.
- The corresponding block output characteristics. The output port rate must equal the input port rate.

Valid Block Inputs	Dimension Along Which Block Computes DFT	Corresponding Block Output Characteristics
Frame-based P-by-N matrix	Column	• Sample based • Complex valued • M-by-N matrix • Each output column contains the M-point DFT of the corresponding input channel in linear or bit-reversed order.

Valid Block Inputs	Dimension Along Which Block Computes DFT	Corresponding Block Output Characteristics
Sample-based P-by-N matrix, $P \neq 1$	Column	• Sample based • Complex valued • M-by-N matrix • Each output column contains the M-point DFT of the corresponding input channel in linear or bit-reversed order.
Sample-based 1-by-P row vector	Row	• Sample based • Complex valued • 1-by-M row vector • Each output row contains the M-point DFT of the corresponding input channel in linear or bit-reversed order.
Unoriented length-P 1-D vector	Vector	Unoriented, length-M, complex-valued 1-D output vector containing M-point DFT of input in linear or bit-reversed order

Selecting the Twiddle Factor Computation Method

The **Twiddle factor computation** parameter determines how the block computes the necessary sine and cosine terms to calculate the term $e^{-j2\pi(p-1)(k-1)/M}$, shown in the first equation of this block reference page. This parameter has two settings, each with its advantages and disadvantages, as described in the following table. Only `Table lookup` mode is supported for fixed-point signals.

Twiddle Factor Computation Parameter Setting	Sine and Cosine Computation Method	Effect on Block Performance
`Table lookup`	The block computes and stores the trigonometric values before the simulation starts, and retrieves them during the simulation. When you generate code from the block, the processor running the generated code stores the trigonometric values computed by the block, and retrieves the values during code execution.	The block usually runs much more quickly, but requires extra memory for storing the precomputed trigonometric values. You can optimize the table for memory consumption or speed, as described in "Optimizing the Table of Trigonometric Values" on page 2-450.
`Trigonometric fcn`	The block computes sine and cosine values during the simulation. When you generate code from the block, the processor running the generated code computes the sine and cosine values while the code runs.	The block usually runs more slowly, but does not need extra data memory. For code generation, the block requires a support library to emulate the trigonometric functions, increasing the size of the generated code.

Optimizing the Table of Trigonometric Values

When you set the **Twiddle factor computation** parameter to `Table lookup`, you also need to set the **Optimize table for** parameter.

This parameter optimizes the table of trigonometric values for speed or memory by varying the number of table entries as summarized in the following table.

Optimize Table for Parameter Setting	Number of Table Entries for N-Point FFT	Memory Required for Single-Precision 512-Point FFT
`Speed`	$3N/4$ — floating point N — fixed point	$\left(\frac{3 \times 512}{4} \text{table entries}\right) \times \left(4 \frac{\text{bytes}}{\text{table entry}}\right)$
`Memory`	$N/4$ — floating point Not supported for fixed point	$\left(\frac{512}{4} \text{table entries}\right) \times \left(4 \frac{\text{bytes}}{\text{table entry}}\right)$ $= 512 \text{ bytes}$

Ordering Output Column Entries

You can set the **Output in bit-reversed order** parameter to specify the ordering of the column elements of the block output. If you select the **Output in bit-reversed order** check box, the output is in bit-reversed order. If you clear the **Output in bit-reversed order** check box, the output is in linear order.

Note Linearly ordering the FFT block output requires a butterfly operation. Therefore, it might be better to output in bit-reversed order in some situations.

For more information ordering of the output, see “Linear and Bit-Reversed Output Order”.

Algorithms Used for FFT Computation

Depending on whether the block's input is real- or complex-valued and whether you want the output in linear or bit-reversed order, the block uses one or more of the following algorithms as summarized in the following table:

- Butterfly operation
- Double-signal algorithm
- Half-length algorithm
- Radix-2 decimation-in-time (DIT) algorithm
- Radix-2 decimation-in-frequency (DIF) algorithm

Complexity of Input	Output Ordering	Algorithms Used for FFT Computation
Complex	Linear	Butterfly operation and radix-2 DIT
Complex	Bit-reversed	Radix-2 DIF
Real	Linear	Butterfly operation and radix-2 DIT in conjunction with the half-length and double-signal algorithms
Real	Bit-reversed	Radix-2 DIF in conjunction with the half-length and double-signal algorithms

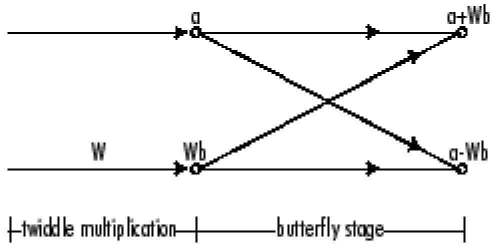
For more information on the double-signal and half-length algorithms, see Proakis [2]. "Efficient Computation of the DFT of Two Real Sequences" on page 475 describes the double signal algorithm. "Efficient Computation of the DFT of a 2N-Point Real Sequence" on page 476 describes the half-length algorithm.

Fixed-Point Data Types

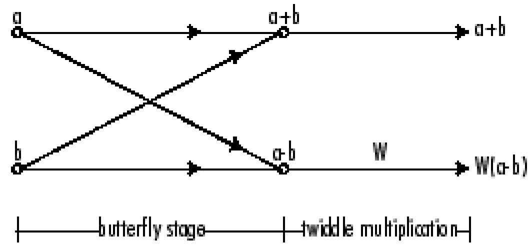
The diagrams below show the data types used within the FFT block for fixed-point signals. You can set the sine table, accumulator, product output, and output data types displayed in the diagrams in the FFT block dialog as discussed in "Dialog Box" on page 2-453.

Inputs to the FFT block are first cast to the output data type and stored in the output buffer. Each butterfly stage then processes signals in the accumulator data type, with the final output of the butterfly being cast back into the output data type. A twiddle factor is multiplied in before each butterfly stage in a decimation-in-time FFT, and after each butterfly stage in a decimation-in-frequency FFT.

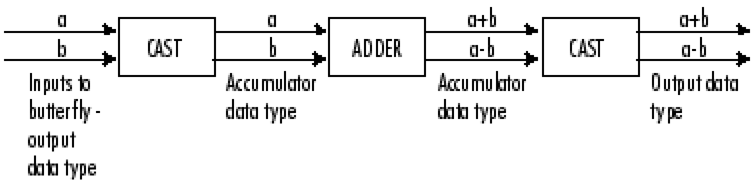
Decimation-in-Time FFT



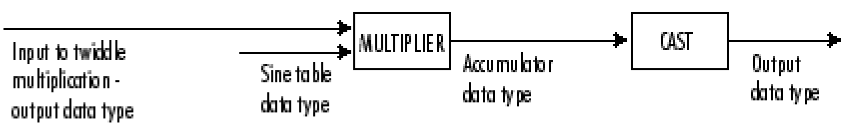
Decimation-in-Frequency FFT



Butterfly Stage Data Types



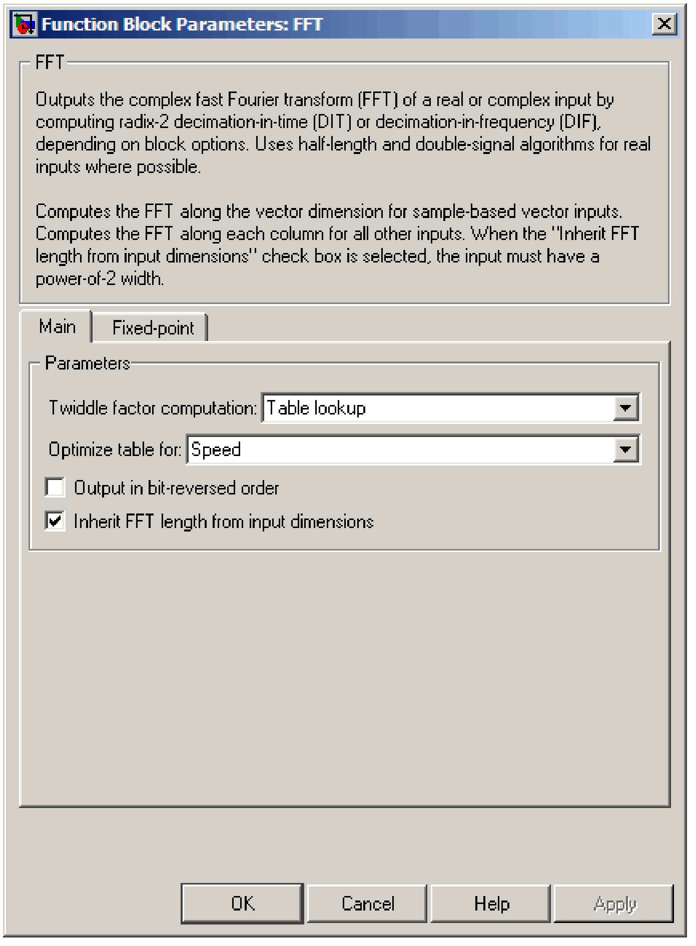
Twiddle Multiplication Data Types



The output of the multiplier is in the accumulator data type since both of the inputs to the multiplier are complex. For details on the complex multiplication performed, see “Multiplication Data Types”.

Dialog Box

The **Main** pane of the FFT block dialog appears as follows.



Twiddle factor computation

Specify the computation method of the term $e^{-j2\pi(p-1)(k-1)/M}$, shown in the first equation of this block reference page.

In `Table lookup` mode, the block computes and stores the sine and cosine values before the simulation starts.

In `Trigonometric fcn` mode, the block computes the sine and cosine values during the simulation. See “Selecting the Twiddle Factor Computation Method” on page 2-449.

This parameter must be set to `Table lookup` for fixed-point signals.

Optimize table for

Select the optimization of the table of sine and cosine values for `Speed` or `Memory`. This parameter is only available when the **Twiddle factor computation** parameter is set to `Table lookup`. See “Selecting the Twiddle Factor Computation Method” on page 2-449.

This parameter must be set to `Speed` for fixed-point signals.

Output in bit-reversed order

Designate the order of the output channel elements relative to the ordering of the input elements. When selected, the output channel elements are in bit-reversed order relative to the input ordering. Otherwise, the output column elements are linearly ordered relative to the input ordering.

Linearly ordering the output requires extra data sorting manipulation, so in some situations it might be better to output in bit-reversed order.

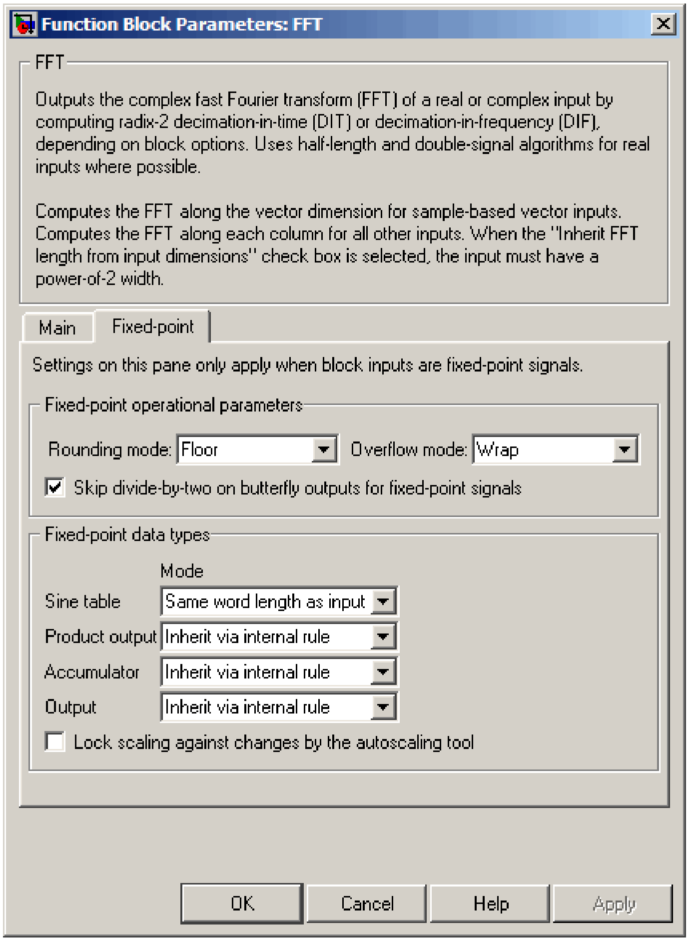
Inherit FFT length from input dimensions

Select to inherit the FFT length from the input dimensions. When this parameter is selected, the input length P must be a power of two. When this parameter is not selected, the **FFT length** parameter is available.

FFT length

Specify a power-of-two FFT length. This parameter is only available when the **Inherit FFT length from input dimensions** parameter is not selected.

The **Fixed-point** pane of the FFT block dialog appears as follows.



Rounding mode

Select the rounding mode for fixed-point operations. The sine table values do not obey this parameter; they always round to `Nearest`.

Overflow mode

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

Skip divide-by-two on butterfly outputs for fixed-point signals

When you select this parameter, no scaling occurs. When you do not select this parameter, the output of each butterfly of the FFT is divided by two for fixed-point signals.

Sine table

Choose how you specify the word length of the values of the sine table. The fraction length of the sine table values is always equal to the word length minus one:

- When you select `Same word length as input`, the word length of the sine table values match that of the input to the block.
- When you select `Specify word length`, you can enter the word length of the sine table values, in bits.

The sine table values do not obey the **Rounding mode** and **Overflow mode** parameters; they are always saturated and rounded to `Nearest`.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-44 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.

- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-44 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Inherit via internal rule`, the output word length and fraction length are calculated automatically.

The internal rule first calculates an ideal output word length and fraction length using the following equations:

$$WL_{ideal\ output} = WL_{input} + floor(\log_2(FFT\ length - 1)) + 1$$

$$FL_{ideal\ output} = FL_{input}$$

Using these ideal results, the internal rule then selects word lengths and fraction lengths that are appropriate for your hardware. For more information, see "Inherit via Internal Rule".

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

References

[1] Orfanidis, S. J. *Introduction to Signal Processing*. Upper Saddle River, NJ: Prentice Hall, 1996, p. 497.

[2] Proakis, John G. and Dimitris G. Manolakis. *Digital Signal Processing*, 3rd ed. Upper Saddle River, NJ: Prentice Hall, 1996.

Supported Data Types

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

See Also

DCT	Signal Processing Blockset
IFFT	Signal Processing Blockset
Pad	Signal Processing Blockset
`bitrevorder`	Signal Processing Toolbox
`fft`	Signal Processing Toolbox
`ifft`	Signal Processing Toolbox

Purpose

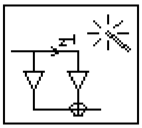
Construct filter realizations using Digital Filter block or Sum, Gain, and Delay blocks

Library

Filtering / Filter Designs

`dsparch4`

Description



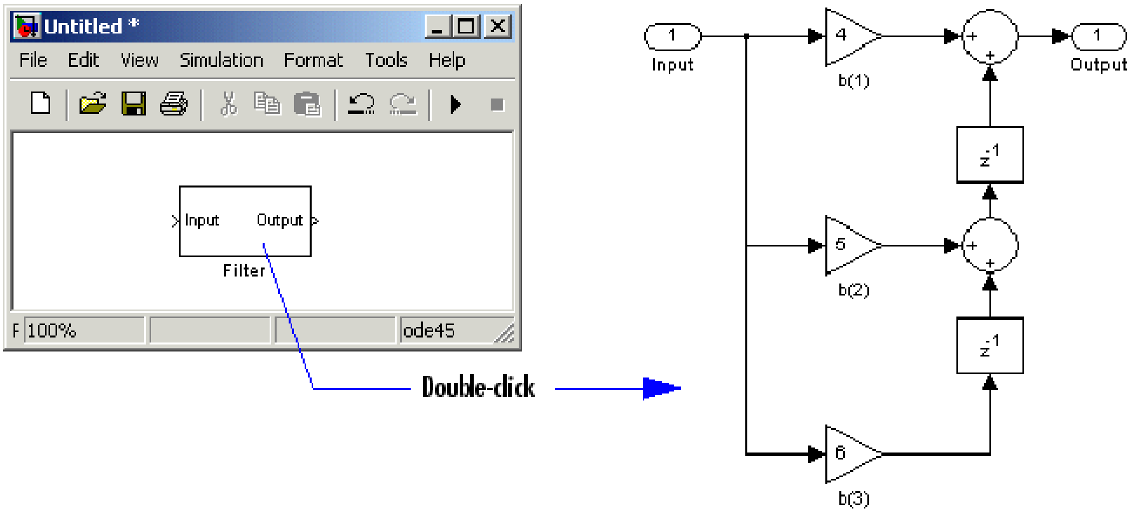
Note Use this block to implement fixed-point or floating-point digital filters using Sum, Gain, and Delay blocks or the Digital Filter block. You can either design a filter by using the block's filter design and analysis parameters, or import the coefficients of a filter you have designed elsewhere.

The following blocks also implement digital filters, but serve slightly different purposes:

- Digital Filter — Use to implement floating-point or fixed-point filters that you have already designed
- Digital Filter Design — Use to design, analyze, and then implement floating-point filters.

The Filter Realization Wizard is a tool for automatically implementing a digital filter. You must specify a filter, its structure, and the data types for the filter's inputs, outputs, and computations. The filter can support double-precision, single-precision, or fixed-point data types.

The Filter Realization Wizard can implement a digital filter in one of two ways. It can use a Digital Filter block, or it can create a subsystem block that implements the specified filter using Sum, Gain, and Delay blocks. If the Filter Realization Wizard creates a Digital Filter block, double-click the block to open the Block Parameters: Filter dialog box. If it creates a subsystem, double-click the subsystem block to see the filter implementation as shown in the figure below.



The subsystem block applies the specified filter to any sample-based input signal, or any frame-based row vector signal, and outputs the result. For more information about filter implementation, see "Specifying the Filter Implementation" on page 2-466.

The parameters of the Filter Realization Wizard are a part of a larger GUI, the Filter Design and Analysis Tool (`fdatool`), from the Signal Processing Toolbox™ product. You can use the tools in FDATool to design and analyze your filter, and then use the Filter Realization Wizard parameters to implement the filter in your models.

Sections of This Reference Page

- "Valid Inputs and Corresponding Outputs" on page 2-463
- "Specifying the Filter and Its Data Type Support" on page 2-464
- "Supported Filter Structures" on page 2-465
- "Specifying the Filter Implementation" on page 2-466
- "Corresponding Method for dfilt" on page 2-467
- "Dialog Box" on page 2-468
- "References" on page 2-470
- "Supported Data Types" on page 2-470
- "See Also" on page 2-471

Valid Inputs and Corresponding Outputs

When the Filter Realization Wizard implements the specified filter by creating a new subsystem block, the block applies the specified filter to an input signal and outputs the result.

Valid Inputs

The subsystem block accepts inputs that are

- Sample-based vectors and matrices
- Frame-based row vectors (nonrecursive structures only)

Corresponding Outputs

The output of the subsystem block has the same dimensions and frame status as the input.

What Is Considered an Independent Channel

The subsystem block treats each *element* of a vector or matrix as an independent channel.

Specifying the Filter and Its Data Type Support

To specify a purely double-precision filter, you can either design a filter using the **Design Filter** panel, or import a filter using the **Import Filter** panel. (You can import `dfilt` filter objects as well as vectors of filter coefficients designed using Signal Processing Toolbox functions and Filter Design Toolbox™ functions.)

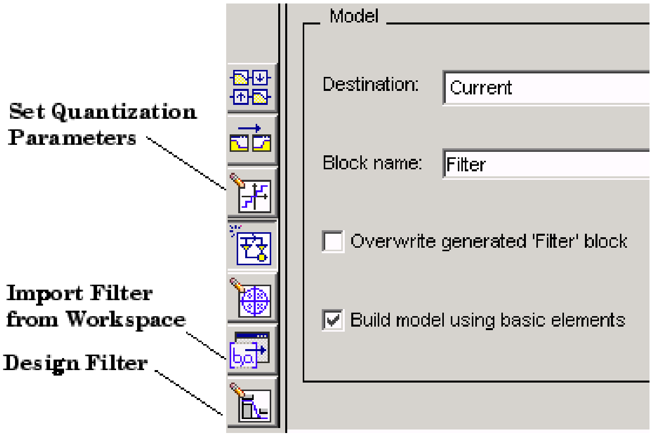
You can also specify a fixed-point filter or a single-precision filter. You can specify such filters by using the **Set Quantization Parameters** panel, which requires the Filter Design Toolbox product.

Note *Running* a model containing implementations of fixed-point filters requires the Simulink® Fixed Point™ product, but you can still edit models containing such filter implementations without it. See the Simulink Fixed Point documentation for more information.

See the following topics to learn how to use the panels to specify your filter:

- For more information on the **Design Filter** panel, see "FDATool: A Filter Design and Analysis GUI" in the Signal Processing Toolbox documentation.
- For more information on the **Import Filter** panel, see "Importing a Filter Design" in the Signal Processing Toolbox documentation.
- For more information on the **Set Quantization Parameters** panel, see "Switching FDATool to Quantization Mode" in the Filter Design Toolbox documentation.

To open a panel, click the appropriate button in the lower-left corner of FDATool.



Supported Filter Structures

The Filter Realization Wizard supports the following structures:

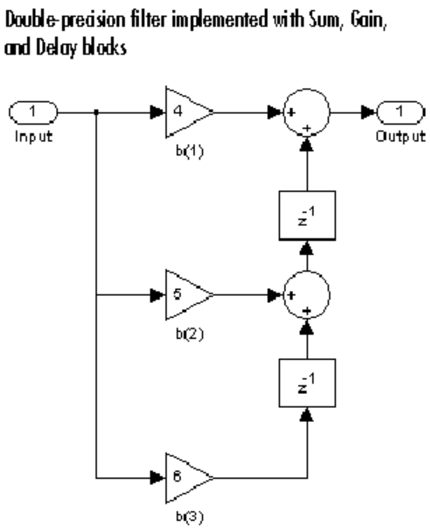
- Direct form I
- Direct form I, second-order sections
- Direct form I transposed
- Direct form I transposed, second-order sections
- Direct form II
- Direct form II, second-order sections
- Direct form II transposed
- Direct form II transposed, second-order sections
- Direct form FIR
- Direct form FIR transposed
- Direct form symmetric FIR

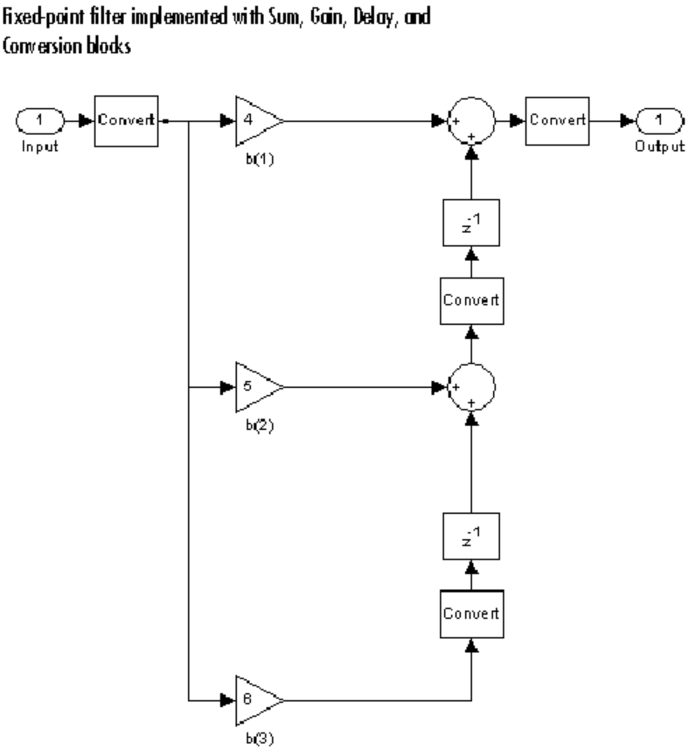
- Direct form antisymmetric FIR
- Lattice all-pass
- Lattice AR
- Lattice ARMA
- Lattice MA for maximum phase
- Lattice MA for minimum phase
- Cascade
- Parallel

Specifying the Filter Implementation

You can determine how the Filter Realization Wizard models the specified filter using the **Build model using basic elements** check box. When you select this check box, the Filter Realization Wizard creates a subsystem block that implements your filter using Sum, Gain, and Delay blocks. When you clear this check box, the Filter Realization Wizard uses a Digital Filter block to implement your filter. The **Build model using basic elements** check box is only available when your filter can be implemented using a Digital Filter block.

If you have Signal Processing Blockset™ software, Signal Processing Toolbox software, and Filter Design Toolbox software installed on your system, the Filter Realization Wizard can generate a subsystem that represents either a double-precision or fixed-point filter. You must install the Simulink Fixed Point product to simulate a fixed-point filter. You can still edit the blocks used to implement the filter without installing the Simulink Fixed Point product.





Implementations of Double-Precision and Fixed-Point Filters

Corresponding Method for dfilt

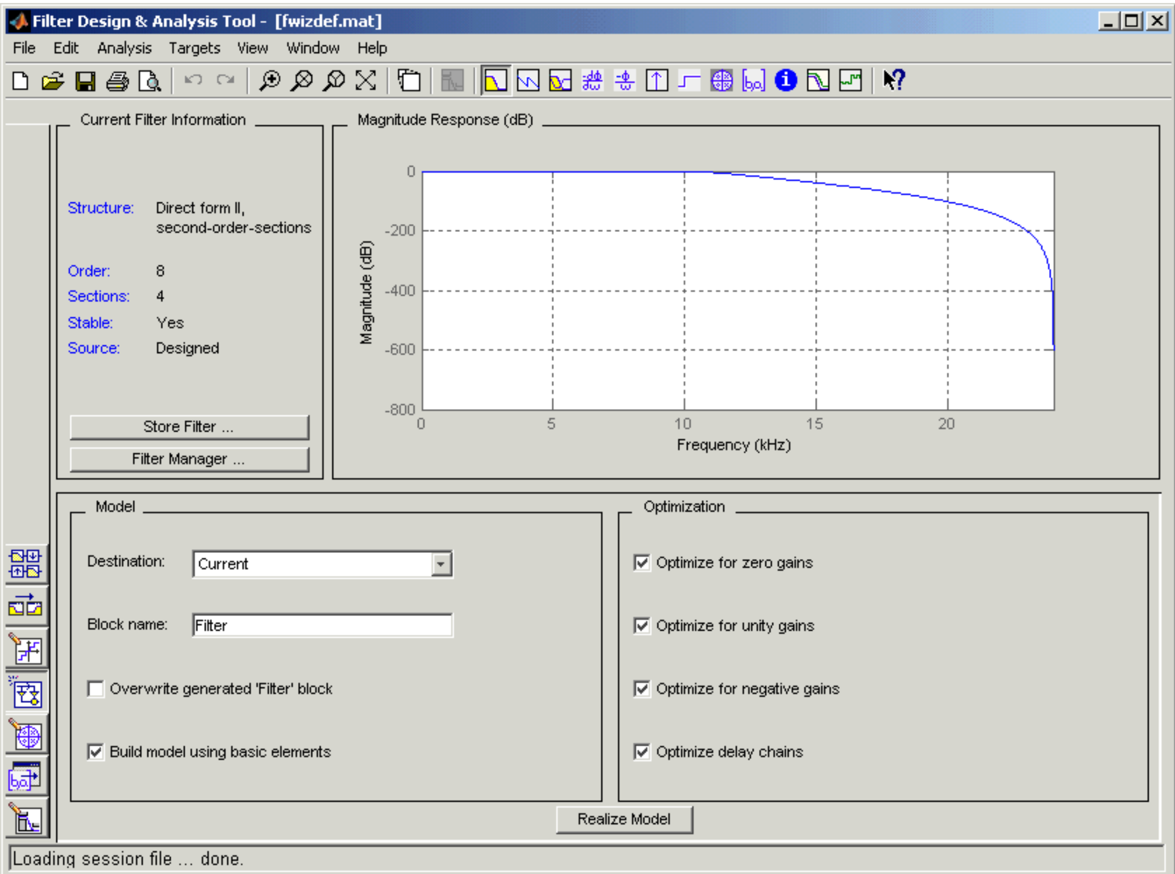
The Signal Processing Toolbox `dfilt` (digital filter) object in has a method, `realizemdl`, that allows you to access the capabilities of the Filter Realization Wizard from the command line.

For more information about the `realizemdl` method, see the following:

- The topic on "Methods" in the `dfilt` reference page in the Signal Processing Toolbox documentation
- The `realizemdl` reference page in the Filter Design Toolbox documentation

Dialog Box

Note The following parameters for the Filter Realization Wizard are in the **Realize Model** pane of the Filter Design and Analysis Tool (FDATool) GUI. To open different panels of FDATool, click the different buttons at the lower-left corner. For more information about relevant panels, see "Specifying the Filter and Its Data Type Support" on page 2-464.



Destination
Specify where the new filter block should be created. This can be in a new model or in the current (most recently selected) model.

Block Name
Enter the name of the new filter block.

Overwrite generated block "Filter" block
When selected, the block overwrites any filter block in the current model with the name specified in the **Block Name** parameter. This parameter is enabled when the **Destination** parameter is set to `Current`.

Build model using basic elements
Select this check box to implement your filter using Sum, Gain, and Delay blocks. Clear this check box to implement your filter using the Digital Filter block. This parameter is only available when your filter can be modeled using the Digital Filter block.

Note that when your filter is implemented using Sum, Gain, and Delay blocks, inputs to the filter must be sample based.

Optimize for zero gains
When selected, the block removes zero-gain paths from the filter structure. For an example, see "Optimizing the Filter Structure".

Optimize for unity gains
When selected, the block substitutes gains equal to 1 with a wire (short circuit). For an example, see "Optimizing the Filter Structure".

Optimize for negative gains
When selected, the block substitutes gains equal to -1 with a wire (short circuit), and changes the corresponding sums to subtractions. For an example, see "Optimizing the Filter Structure".

Optimize delay chains

When selected, the block substitutes any delay chains made up of n unit delays with a single delay by n. For an example, see "Optimizing the Filter Structure".

Realize Model

Click to create a subsystem block that implements the specified filter using Sum, Gain, and Delay blocks. To see the filter implementation, double-click the subsystem block. The subsystem block applies the specified filter to any sample-based input signal or frame-based row vector signal, and outputs the result.

Note For more information about relevant parameters in other panels of FDATool, see "Specifying the Filter and Its Data Type Support" on page 2-464.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point — Supported only when you install the following products: Filter Design Toolbox and Simulink Fixed Point
- Fixed point (signed and unsigned) — Supported only when you install the following products: Filter Design Toolbox, "Simulink Fixed Point", and Fixed-Point Toolbox™

See Also

Digital Filter	Signal Processing Blockset
Digital Filter Design	Signal Processing Blockset
`filter`	Filter Design Toolbox
`realizemdl`	Filter Design Toolbox
`dfilt`	Signal Processing Toolbox
`filter`	Signal Processing Toolbox

- "Filters" — Examples of when and how to use Signal Processing Blockset filtering blocks
- "Choosing Between Filter Design Blocks"

FIR Decimation

Purpose

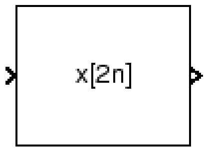
Filter and downsample input signals

Library

Filtering / Multirate Filters

`dspmlti4`

Description



The FIR Decimation block resamples the discrete-time input at a rate K times slower than the input sample rate, where the integer K is specified by the **Decimation factor** parameter. This process consists of two steps:

- The block filters the input data using a direct-form FIR filter.
- The block downsamples the filtered data to a lower rate by discarding K–1 consecutive samples following every sample retained.

The FIR Decimation block implements the above FIR filtering and downsampling steps together using a polyphase filter structure, which is more efficient than straightforward filter-then-decimate algorithms. See Fliege [1] for more information.

The **FIR filter coefficients** parameter specifies the numerator coefficients of the FIR filter transfer function $H(z)$.

$$H(z) = B(z) = b_1 + b_2 z^{-1} + \ldots + b_m z^{-(m-1)}$$

The length-m coefficient vector, `[b(1) b(2) ...  b(m)]`, can be generated by one of the filter design functions in Signal Processing Toolbox™ software, such as the `fir1` function used in Example 1 below. The filter should be lowpass with normalized cutoff frequency no greater than 1/K. All filter states are internally initialized to zero.

The FIR Decimation block supports real and complex floating-point and fixed-point inputs, except for complex unsigned fixed-point inputs. This block supports triggered subsystems when you select `Maintain input frame rate` for the **Framing** parameter.

Sample-Based Operation

An M-by-N sample-based matrix input is treated as $M*N$ independent channels, and the block decimates each channel over time. The output sample period is K times longer than the input sample period ($T_{so} = KT_{si}$), and the input and output sizes are identical.

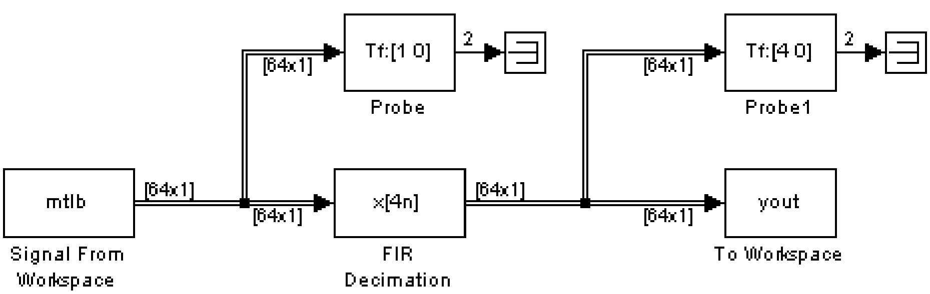
Frame-Based Operation

An M_i-by-N frame-based matrix input is treated as N independent channels, and the block decimates each channel over time. The **Framing** parameter determines how the block adjusts the rate at the output to accommodate the reduced number of samples. There are two available options:

- `Maintain input frame size`

  The block generates the output at the decimated rate by using a proportionally longer frame *period* at the output port than at the input port. For decimation by a factor of K, the output frame period is K times longer than the input frame period ($T_{fo} = KT_{fi}$), but the input and output frame sizes are equal.

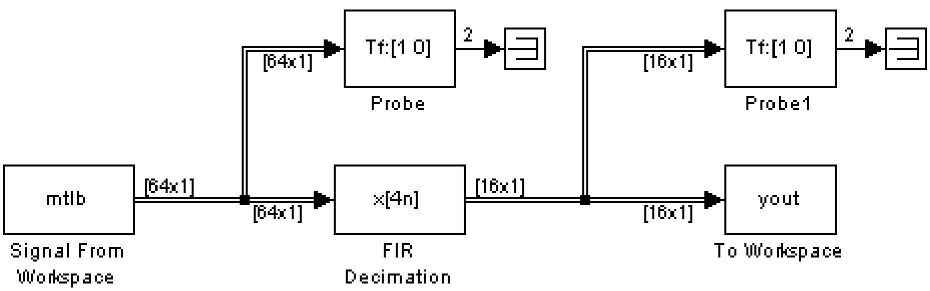
  The following model shows a single-channel input with a frame period of 1 second (**Sample time** = `1/64` and **Samples per frame** = `64` in the Signal From Workspace block) being decimated by a factor of 4 to a frame period of 4 seconds. The input and output frame sizes are identical.



- `Maintain input frame rate`

The block generates the output at the decimated rate by using a proportionally smaller frame size than the input. For decimation by a factor of K, the output frame size is K times smaller than the input frame size ($M_o = M_i/K$), but the input and output frame rates are equal. The input frame size, M_i, must be a multiple of the decimation factor, K.

The following model shows a single-channel input of frame size 64 being decimated by a factor of 4 to a frame size of 16. The block's input and output frame rates are identical.



Latency

The FIR Decimation block has *zero tasking latency* for all single-tasking and multitasking cases, except one:

Sample-Based Latency	Frame-Based Latency — Maintain input frame rate	Frame-Based Latency — Maintain input frame size
None	None	One frame (M_i samples)

Zero tasking latency means that the block propagates the first filtered input sample (received at t=0) as the first output sample, followed by filtered input samples K+1, 2K+1, and so on.

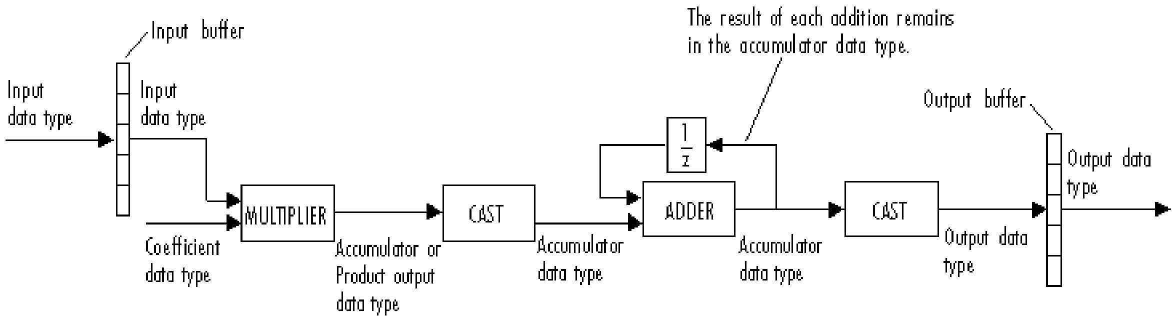
In cases of *one-frame latency*, the first M_i output rows contain zeros, where M_i is the input frame size. The first filtered input sample (first filtered row of the input matrix) appears in the output as sample M_i+1, followed by filtered input samples K+1, 2K+1, and so on.

When the block exhibits latency, enter a value in the **Output buffer initial conditions** text box to specify the value to output at the output port until the first filtered input sample is available. The default initial condition value is 0.

Note For more information on latency and the Simulink® tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Fixed-Point Data Types

The following diagram shows the data types used within the FIR Decimation block for fixed-point signals.



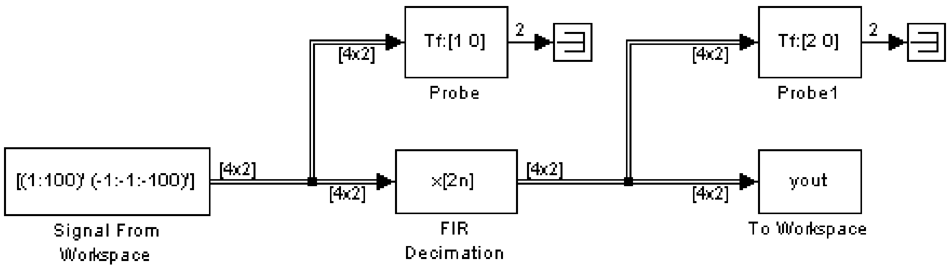
You can set the coefficient, product output, accumulator, and output data types in the block dialog as discussed in "Dialog Box" on page 2-478. This diagram shows that data is stored in the input buffer in the same data type and scaling as the input. Filtered data is stored in the output buffer in the output data type and scaling that you set in the block dialog. Any initial conditions are also stored in the output buffer in the output data type and scaling you set in the block dialog.

The output of the multiplier is in the product output data type when at least one of the inputs to the multiplier is real. When both of the inputs to the multiplier are complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see "Multiplication Data Types".

Examples

Example 1

Construct the following frame-based model.



Adjust the block parameters as follows:

- Configure the Signal From Workspace block to generate a two-channel signal with frame size of 4 and sample period of 0.25. This represents an output frame period of 1 (0.25*4). The first channel should contain the positive ramp signal 1, 2, ..., 100, and the second channel should contain the negative ramp signal -1, -2, ..., -100.
  - **Signal** = `[(1:100)' (-1:-1:-100)']`
  - **Sample time** = `0.25`
  - **Samples per frame** = `4`
- Configure the FIR Decimation block to decimate the two-channel input by decreasing the output frame rate by a factor of 2 relative to the input frame rate. Use a third-order filter with normalized cutoff frequency, f_{n0}, of 0.25. (Note that f_{n0} satisfies $f_{n0} \leq 1/K$.)
  - **FIR filter coefficients** = `fir1(3,0.25)`

- **Downsample factor** = 2
- **Framing** = `Maintain input frame size`

The filter coefficient vector generated by `fir1(3,0.25)` is

```
[0.0386 0.4614 0.4614 0.0386]
```

or, equivalently,

$$H(z) = B(z) = 0.0386 + 0.04614z^{-1} + 0.04614z^{-2} + 0.0386z^{-3}$$

- Configure the Probe blocks by clearing the **Probe width**, **Probe complex signal**, and **Probe signal dimensions** check boxes (if desired).

This model is multirate because there are at least two distinct sample rates, as shown by the two Probe blocks. To run this model in the Simulink multitasking mode, make the following settings in the **Solver** pane of the Configuration Parameters dialog box:

- From the **Type** list, select `Fixed-step`.
- From the **Solver** list, select `discrete (no continuous states)`.
- From the **Tasking mode for periodic sample times** list, select `MultiTasking`.
- Set the **Stop time** to `30`.

Run the model and look at the output, `yout`. The first few samples of each channel are shown below.

```
yout =
         0         0
         0         0
         0         0
         0         0
    0.0386   -0.0386
    1.5000   -1.5000
```

```
 3.5000   -3.5000
 5.5000   -5.5000
 7.5000   -7.5000
 9.5000   -9.5000
11.5000  -11.5000
```

Since this is a frame-based multirate model, the first four (M_i) output rows are zero. The first filtered input matrix row appears in the output as sample 5 (that is, sample M_i+1).

Example 2

The `doc_polyphasedec` model illustrates the underlying polyphase implementations of the FIR Decimation block. Run the model and view the results on the scope. The output of the FIR Decimation block is the same as the output of the Polyphase Decimation Filter block.

Example 3

The `doc_mrf_nlp` model illustrates the use of the FIR Decimation block in a number of multistage multirate filters.

Dialog Box

The FIR Decimation block can operate in two different modes. Select the mode in the **Coefficient source** group box. If you select

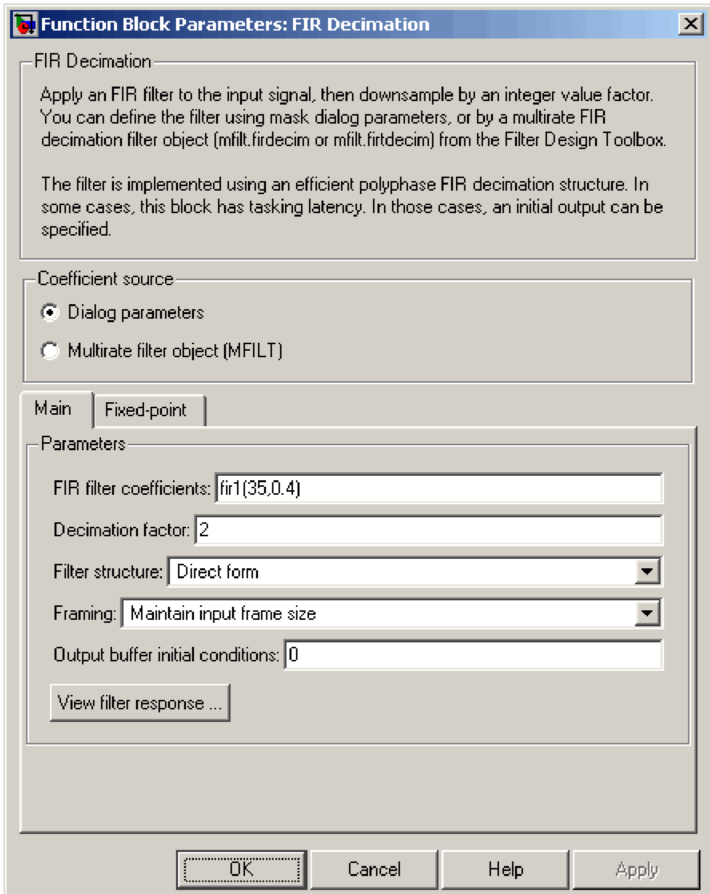
- **Dialog parameters**, you enter information about the filter such as structure and coefficients in the block mask.
- **Multirate filter object (MFILT)**, you specify the filter using a Filter Design Toolbox™ `mfilt` object.

Different items appear on the FIR Decimation block dialog depending on whether you select **Dialog parameters** or **Multirate filter object (MFILT)** in the **Coefficient source** group box. See the following sections for details:

- "Specify Filter Characteristics in Dialog" on page 2-479
- "Specify Multirate Filter Object" on page 2-486

Specify Filter Characteristics in Dialog

The **Main** pane of the FIR Decimation block dialog appears as follows when **Dialog parameters** is selected in the **Coefficient source** group box.



FIR filter coefficients
Specify the lowpass FIR filter coefficients, in descending powers of z.

Decimation factor
Specify the integer factor, K, by which to decrease the sample rate of the input sequence.

Filter Structure
Choose whether to implement a `Direct form` or `Direct form transposed` filter.

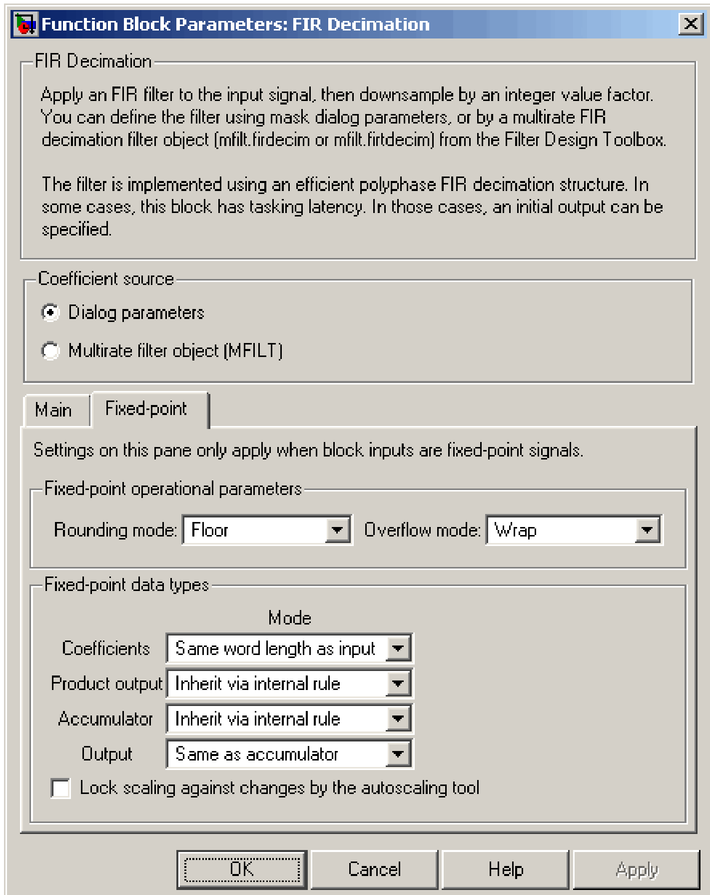
Framing
For frame-based operation, specify the method by which to implement the decimation; reduce the output frame rate, or reduce the output frame size. This parameter cannot be set to `Maintain input frame rate` for sample-based signals.

Output buffer initial conditions
When the block exhibits latency, enter a value in the **Output buffer initial conditions** text box to specify the value to output at the output port until the first filtered input sample is available. The default initial condition value is zero.

View filter response
This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the filter defined in the block. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

The **Fixed point** pane of the FIR Decimation block dialog appears as follows when **Dialog parameters** is specified in the **Coefficient source** group box.



Rounding mode

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

Overflow mode

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

Coefficients

Choose how you specify the word length and the fraction length of the filter coefficients:

- When you select `Same word length as input`, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select `Specify word length`, you can enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the coefficients, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the coefficients. This block requires power-of-two slope and a bias of zero.

The filter coefficients do not obey the **Rounding mode** and the **Overflow mode** parameters; they are always saturated and rounded to `Nearest`.

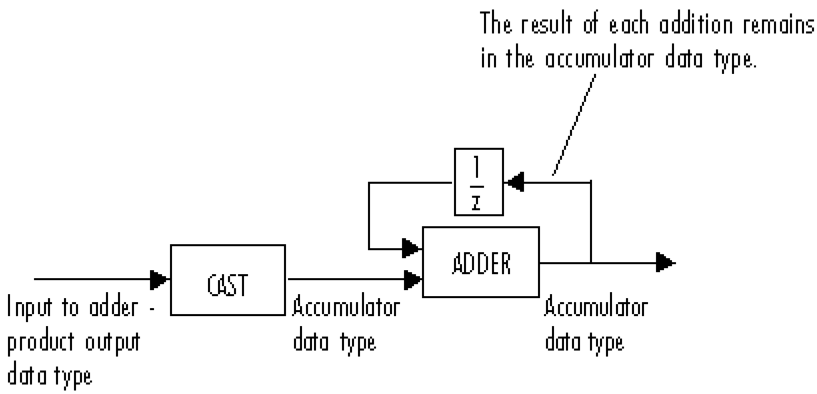
Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See “Fixed-Point

Data Types" on page 2-475 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator



As depicted in this graphic, inputs to the accumulator are cast to the accumulator data type. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you would like to designate this accumulator word and fraction lengths.

You also use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. See “Multiplication Data Types” for more information.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see “Inherit via Internal Rule”.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.

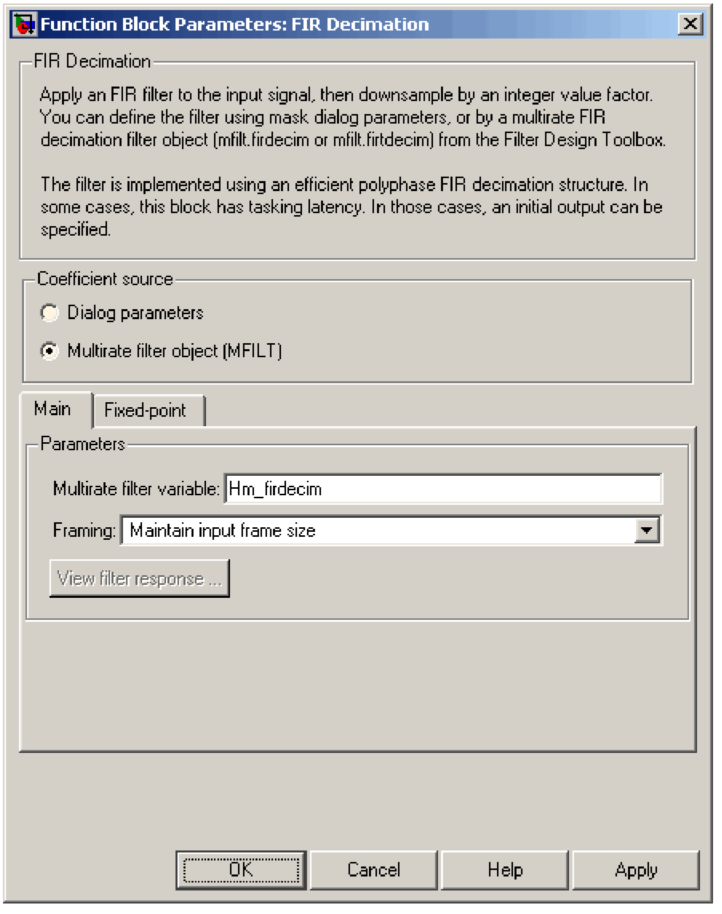
  A special case occurs when `Inherit via internal rule` is specified for **Accumulator**, and block inputs and coefficients are complex. In that case, the output word length is one less than the accumulator word length.
- When you select `Same as product output`, these characteristics match those of the product output.

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

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

Specify Multirate Filter Object

The **Main** pane of the FIR Decimation block dialog appears as follows when **Multirate filter object (MFILT)** is specified in the **Coefficient source** group box.



Multirate filter variable

Specify the multirate filter object (`mfilt`) that you would like the block to implement. You can do this in one of three ways:

- You can fully specify the `mfilt` object in the block mask.
- You can enter the variable name of a `mfilt` object that is defined in any workspace.
- You can enter a variable name for a `mfilt` object that is not yet defined, as shown in the default value.

For more information on creating `mfilt` objects, see the `mfilt` function reference page in the Filter Design Toolbox documentation.

Framing

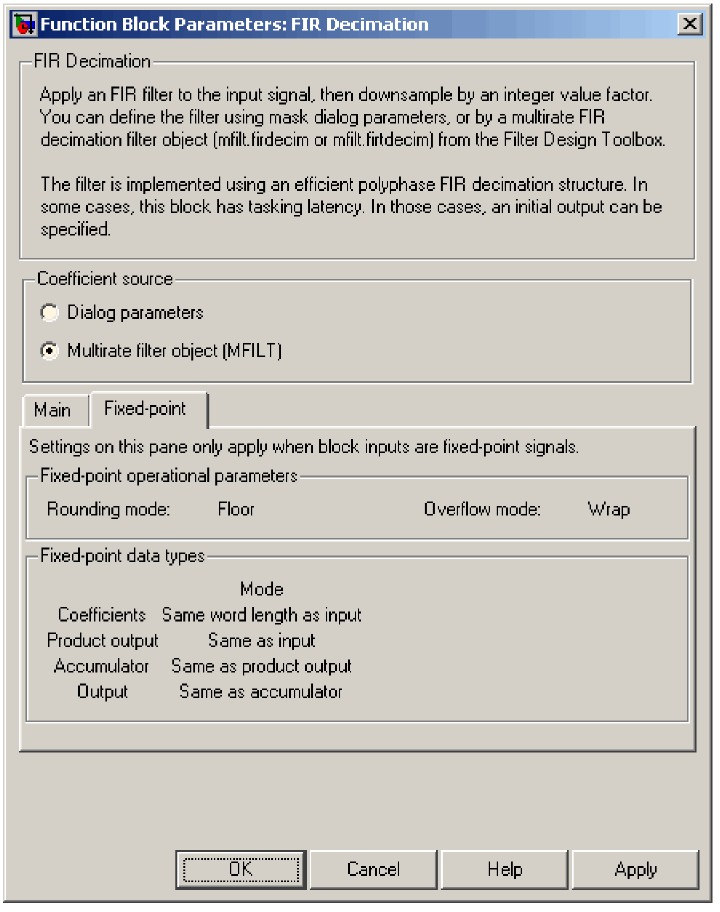
For frame-based operation, specify the method by which to implement the decimation; reduce the output frame rate, or reduce the output frame size. This parameter cannot be set to `Maintain input frame rate` for sample-based signals.

View filter response

This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the `mfilt` object specified in the **Multirate filter variable** parameter. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

The **Fixed-point** pane of the FIR Decimation block dialog appears as follows when **Multirate filter object (MFILT)** is specified in the **Coefficient source** group box.



The fixed-point settings of the filter object specified on the **Main** pane are displayed on the **Fixed-point** pane. You cannot change these settings directly on the block mask. To change the fixed-point settings you must edit the filter object directly.

For more information on multirate filter objects, see the `mfilt` function reference page in the Filter Design Toolbox documentation.

References

[1] Fliege, N. J. *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets*. West Sussex, England: John Wiley & Sons, 1994.

Supported Data Types

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

See Also

Downsample	Signal Processing Blockset
FIR Interpolation	Signal Processing Blockset
FIR Rate Conversion	Signal Processing Blockset
`decimate`	Signal Processing Toolbox
`fir1`	Signal Processing Toolbox
`fir2`	Signal Processing Toolbox
`firls`	Signal Processing Toolbox

FIR Interpolation

Purpose

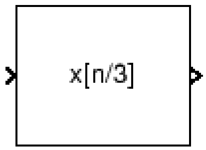
Upsample and filter input signals

Library

Filtering / Multirate Filters

`dspmlti4`

Description



The FIR Interpolation block resamples the discrete-time input at a rate L times faster than the input sample rate, where the integer L is specified by the **Interpolation factor** parameter. This process consists of two steps:

- The block upsamples the input to a higher rate by inserting L-1 zeros between samples.
- The block filters the upsampled data with a direct-form FIR filter.

The FIR Interpolation block implements the above upsampling and FIR filtering steps together using a polyphase filter structure, which is more efficient than straightforward upsample-then-filter algorithms. See N.J. Fliege, *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets* for more information.

The **FIR filter coefficients** parameter specifies the numerator coefficients of the FIR filter transfer function $H(z)$.

$$H(z) = B(z) = b_1 + b_2 z^{-1} + \ldots + b_m z^{-(m-1)}$$

The coefficient vector, `[b(1) b(2) ...  b(m)]`, can be generated by one of the Signal Processing Toolbox™ filter design functions (such as `fir1`), and should have a length greater than the interpolation factor (m>L). The filter should be lowpass with normalized cutoff frequency no greater than $1/L$. All filter states are internally initialized to zero.

The FIR Interpolation block supports real and complex floating-point and fixed-point inputs except for complex unsigned fixed-point inputs. This block supports triggered subsystems when you select `Maintain input frame rate` for the **Framing** parameter.

Sample-Based Operation

An M-by-N sample-based matrix input is treated as $M*N$ independent channels, and the block interpolates each channel over time. The output sample period is L times shorter than the input sample period (T_{so} = T_{si}/L), and the input and output sizes are identical.

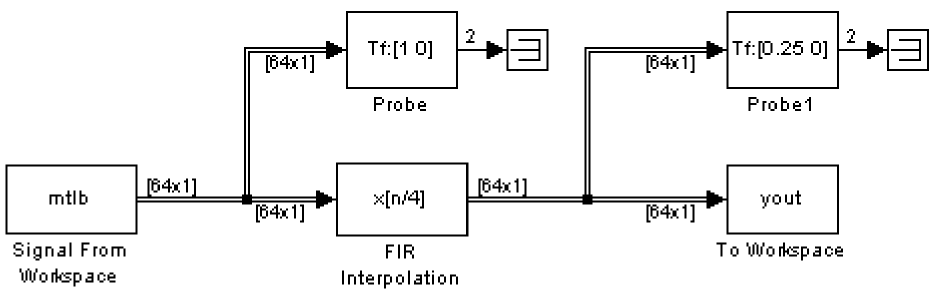
Frame-Based Operation

An M_i-by-N frame-based matrix input is treated as N independent channels, and the block interpolates each channel over time. The **Framing** parameter determines how the block adjusts the rate at the output to accommodate the added samples. There are two available options:

- `Maintain input frame size`

  The block generates the output at the interpolated rate by using a proportionally shorter frame *period* at the output port than at the input port. For interpolation by a factor of L, the output frame period is L times shorter than the input frame period (T_{fo} = T_{fi}/L), but the input and output frame sizes are equal.

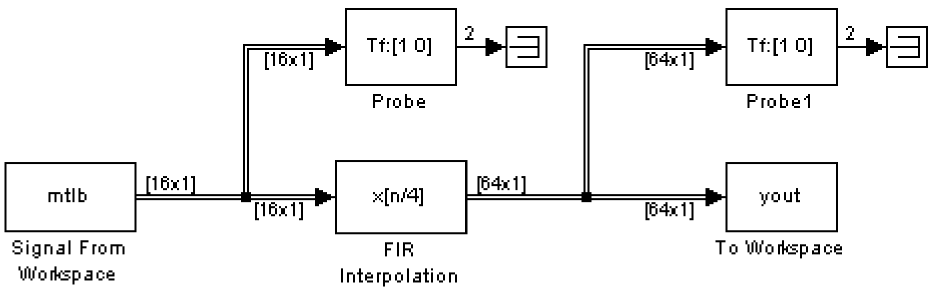
  The example below shows a single-channel input with a frame period of 1 second (**Sample time** = `1/64` and **Samples per frame** = `64` in the Signal From Workspace block) being interpolated by a factor of 4 to a frame period of 0.25 second. The input and output frame sizes are identical.



- `Maintain input frame rate`

The block generates the output at the interpolated rate by using a proportionally larger frame *size* than the input. For interpolation by a factor of L, the output frame size is L times larger than the input frame size ($M_o = M_i{*}L$), but the input and output frame rates are equal.

The example below shows a single-channel input of frame size 16 being interpolated by a factor of 4 to a frame size of 64. The block's input and output frame rates are identical.



Zero Latency

The FIR Interpolation block has *zero tasking latency* for all single-rate operations. The block is single rate for the particular combinations of sampling mode and parameter settings shown in the table below.

Sampling Mode	Parameter Settings
Sample based	**Interpolation factor** parameter, L, is `1`.
Frame based	**Interpolation factor** parameter, L, is `1`, *or* **Framing** parameter is `Maintain input frame rate`.

Note that in sample-based mode, single-rate operation occurs only in the trivial case of factor-of-1 interpolation.

The block also has zero latency for sample-based multirate operations in the Simulink® single-tasking mode. Zero tasking latency means that the block propagates the first filtered input (received at t=0) as the first input sample, followed by L-1 interpolated values, the second filtered input sample, and so on.

Nonzero Latency

The FIR Interpolation block is multirate for all settings other than those in the previous table. The amount of latency for multirate operation depends on the Simulink tasking mode and the block's sampling mode, as shown in the following table.

Multirate...	Sample-Based Latency	Frame-Based Latency
Single-tasking	None	None
Multitasking	L samples	L frames (M_i samples per frame)

When the block exhibits latency, the default initial condition is zero. Alternatively, you can enter a value in the **Output buffer initial conditions** text box. This value is divided by the **Interpolation factor** and output at the output port until the first filtered input sample is available.

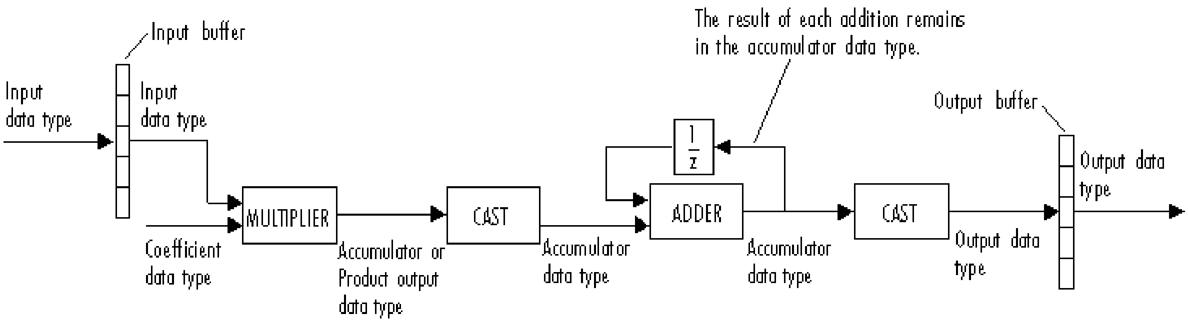
In sample-based cases, the scaled initial conditions appear at the start of each channel, followed immediately by the first filtered input sample, L-1 interpolated values, and so on.

In frame-based cases, with the default initial condition, the first M_iL output rows contain zeros, where M_i is the input frame size. The first filtered input sample (first filtered row of the input matrix) appears in the output as sample M_iL+1, followed by L-1 interpolated values, the second filtered input sample, and so on. See the following example for an illustration of this case.

Note For more information on latency and the Simulink tasking modes, see “Excess Algorithmic Delay (Tasking Latency)” and “Models with Multiple Sample Rates” in the *Real-Time Workshop® User’s Guide*.

Fixed-Point Data Types

The following diagram shows the data types used within the FIR Interpolation block for fixed-point signals.



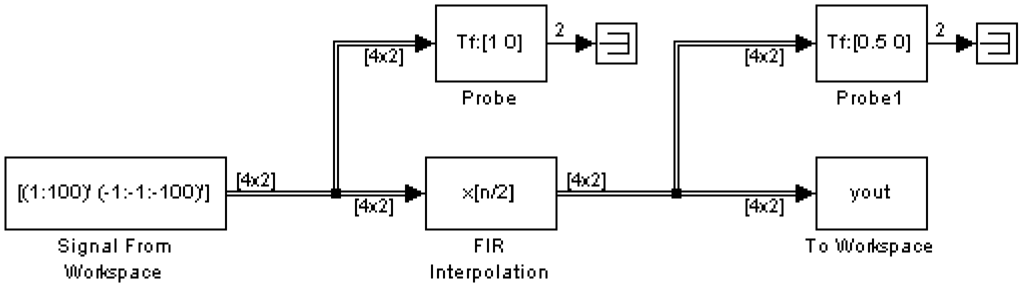
You can set the coefficient, product output, accumulator, and output data types in the block dialog as discussed in “Dialog Box” on page 2-497. The diagram shows that input data is stored in the input buffer in the same data type and scaling as the input. Filtered data is stored in the output buffer in the output data type and scaling that you set in the block dialog. Any initial conditions are also stored in the output buffer in the output data type and scaling you set in the block dialog.

The output of the multiplier is in the product output data type when at least one of the inputs to the multiplier is real. When both of the inputs to the multiplier are complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”.

Examples

Example 1

Construct the frame-based model shown below.



Adjust the block parameters as follows:

- Configure the Signal From Workspace block to generate a two-channel signal with frame size of 4 and sample period of 0.25. This represents an output frame period of 1 (0.25*4). The first channel should contain the positive ramp signal 1, 2, ..., 100, and the second channel should contain the negative ramp signal -1, -2, ..., -100.
  - **Signal** = `[(1:100)' (-1:-1:-100)']`
  - **Sample time** = `0.25`
  - **Samples per frame** = `4`
- Configure the FIR Interpolation block to interpolate the two-channel input by increasing the output frame rate by a factor of 2 relative to the input frame rate. Use a third-order filter (m=3) with normalized cutoff frequency, f_{n0}, of 0.25. (Note that f_{n0} and m satisfy $f_{n0} \leq 1/L$ and $m > L$.)
  - **FIR filter coefficients** = `fir1(3,0.25)`
  - **Interpolation factor** = `2`
  - **Framing** = `Maintain input frame size`

  The filter coefficient vector generated by `fir1(3,0.25)` is

  ```
  [0.0386 0.4614 0.4614 0.0386]
  ```

  or, equivalently,

$$H(z) = B(z) = 0.0386 + 0.04614z^{-1} + 0.04614z^{-2} + 0.0386z^{-3}$$

- Configure the Probe blocks by clearing the **Probe width**, **Probe complex signal**, and **Probe signal dimensions** check boxes (if desired).

This model is multirate because there are at least two distinct sample rates, as shown by the two Probe blocks. To run this model in the Simulink multitasking mode, open the Configuration Parameters dialog box. In the **Select** pane, click **Solver**. From the **Type** list, select `Fixed-step`, and from the **Solver** list, select `discrete (no continuous states)`. From the **Tasking mode for periodic sample times** list, select `MultiTasking`. Also set the **Stop time** to `30`.

Run the model and look at the output, `yout`. The first few samples of each channel are shown below.

```
dsp_examples_yout =
          0         0
          0         0
          0         0
          0         0
          0         0
          0         0
          0         0
          0         0
     0.0386   -0.0386
     0.4614   -0.4614
     0.5386   -0.5386
     0.9614   -0.9614
     1.0386   -1.0386
```

Since we ran this frame-based multirate model in multitasking mode, the first eight (M_iL) output rows are zero. The first filtered input matrix row appears in the output as sample 9 (that is, sample M_iL+1). Every other row is an interpolated value.

Example 2

The `doc_polyphaseinterp` model illustrates the underlying polyphase implementations of the FIR Interpolation block. Run the model and view the results on the scope. The output of the FIR Interpolation block is the same as the output of the Polyphase Interpolation Filter block.

Example 3

The `doc_mrf_nlp` model illustrates the use of the FIR Interpolation block in a number of multistage multirate filters.

Dialog Box

The FIR Interpolation block can operate in two different modes. Select the mode in the **Coefficient source** group box. If you select

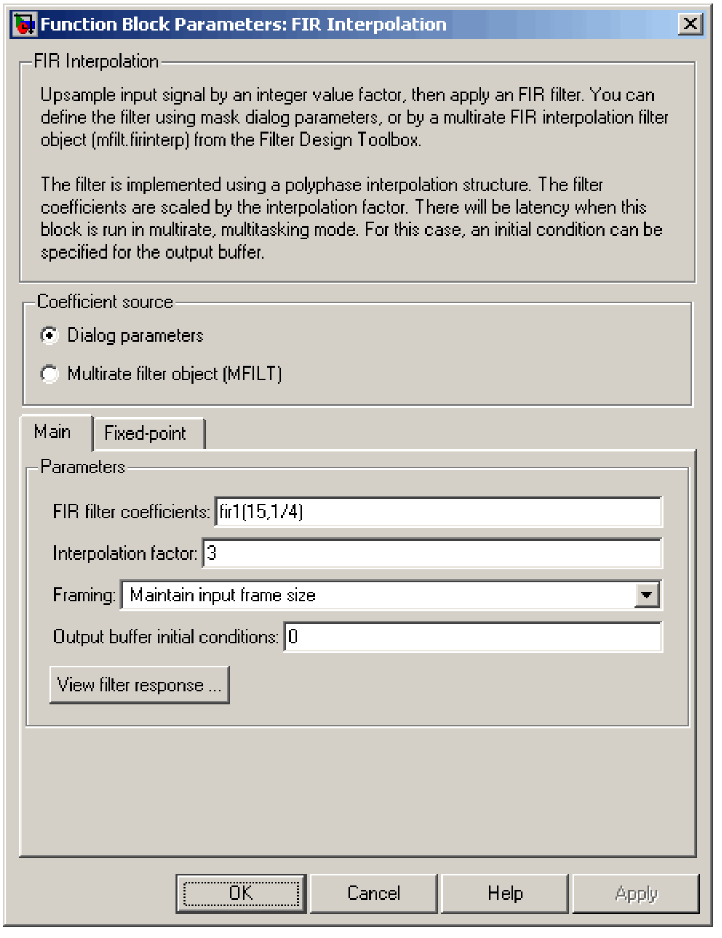
- **Dialog parameters**, you enter information about the filter such as structure and coefficients in the block mask.
- **Multirate filter object (MFILT)**, you specify the filter using a Filter Design Toolbox™ `mfilt` object.

Different items appear on the FIR Interpolation block dialog depending on whether you select **Dialog parameters** or **Multirate filter object (MFILT)** in the **Coefficient source** group box. See the following sections for details:

- "Specify Filter Characteristics in Dialog" on page 2-498
- "Specify Multirate Filter Object" on page 2-505

Specify Filter Characteristics in Dialog

The **Main** pane of the FIR Interpolation block dialog appears as follows when **Dialog parameters** is selected in the **Coefficient source** group box.



FIR filter coefficients
Specify the FIR filter coefficients, in descending powers of z.

Interpolation factor
Specify the integer factor, L, by which to increase the sample rate of the input sequence.

Framing
For frame-based operation, specify the method by which to implement the interpolation: increase the output frame rate, or increase the output frame size. This parameter cannot be set to `Maintain input frame rate` for sample-based signals.

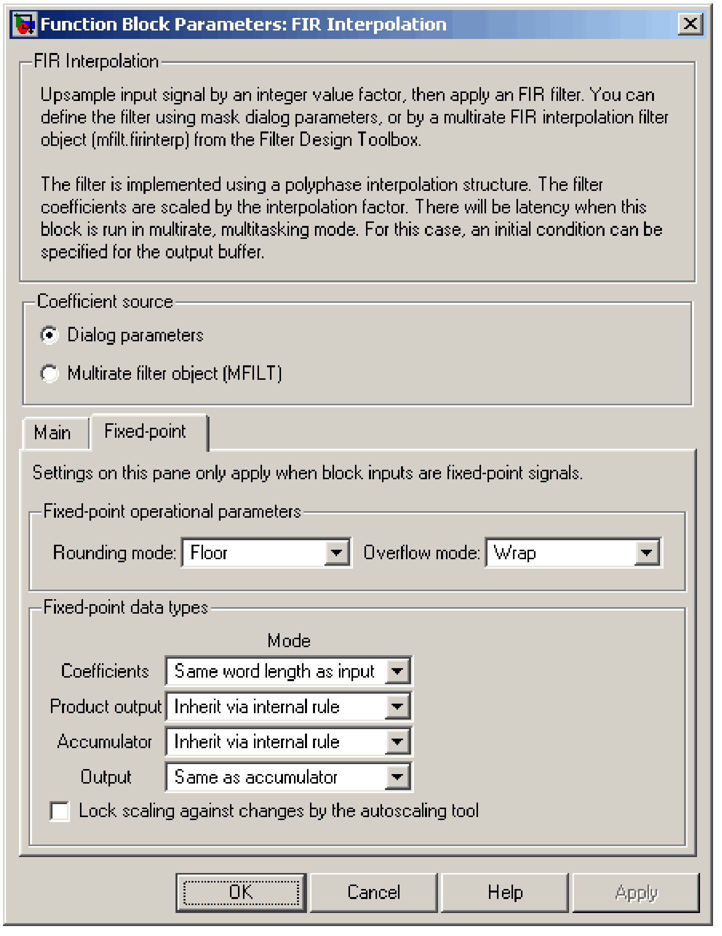
Output buffer initial conditions
When the block exhibits latency, enter a value in the **Output buffer initial conditions** text box to specify the value to output at the output port until the first filtered input sample is available. The default initial condition value is `0`.

Output buffer initial conditions are stored in the output data type and scaling.

View filter response
This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the filter defined in the block. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

The **Fixed point** pane of the FIR Interpolation block dialog appears as follows when **Dialog parameters** is specified in the **Coefficient source** group box.



Rounding mode

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

Overflow mode

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

Coefficients

Choose how you specify the word length and fraction length of the filter coefficients:

- When you select `Same word length as input`, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select `Specify word length`, you can enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the coefficients, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the coefficients. This block requires power-of-two slope and a bias of zero.

The filter coefficients do not obey the **Rounding mode** and the **Overflow mode** parameters; they are always saturated and rounded to `Nearest`.

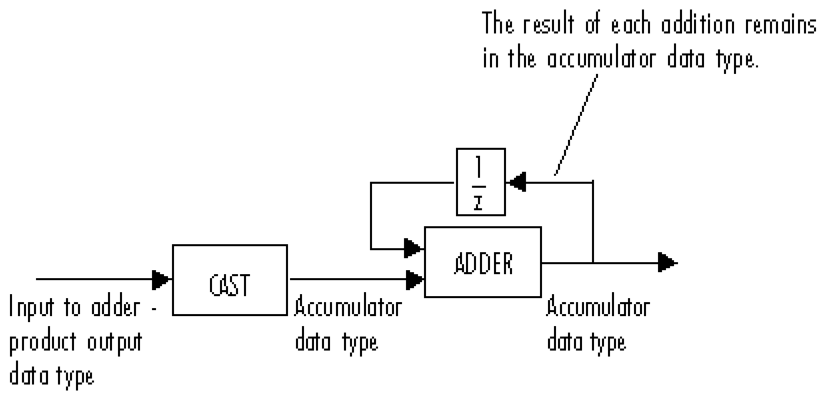
Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point

Data Types" on page 2-494 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator



As depicted above, inputs to the accumulator are cast to the accumulator data type. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you would like to designate this accumulator word and fraction lengths.

You also use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. See "Multiplication Data Types" for more information:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.

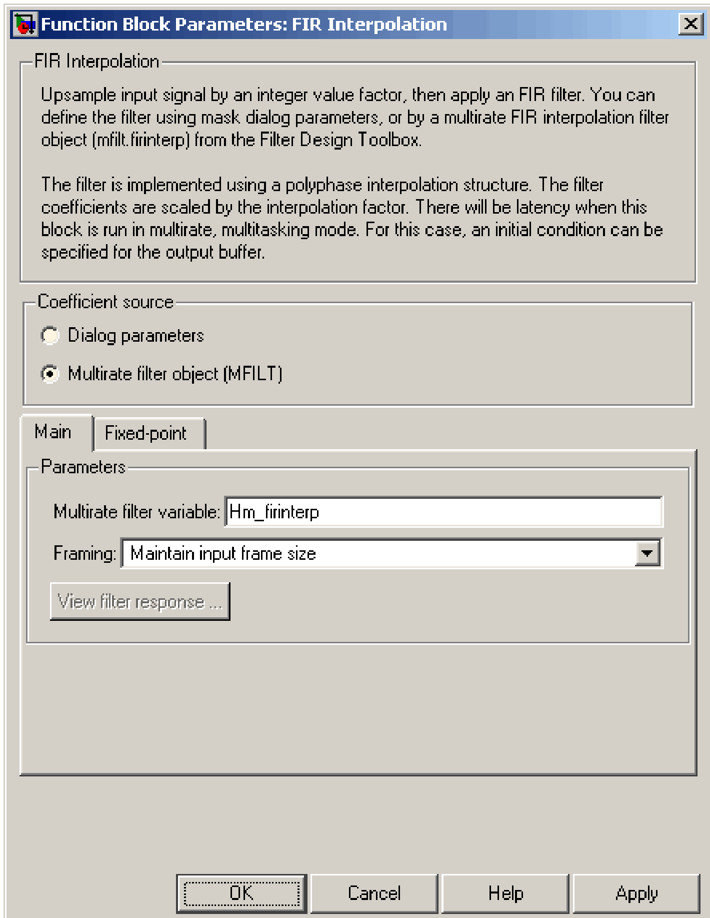
  A special case occurs when `Inherit via internal rule` is specified for **Accumulator**, and block inputs and coefficients are complex. In that case, the output word length be one less than the accumulator word length.
- When you select `Same as product output`, these characteristics match those of the product output.

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

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

Specify Multirate Filter Object

The **Main** pane of the FIR Interpolation block dialog appears as follows when **Multirate filter object (MFILT)** is specified in the **Coefficient source** group box.



Multirate filter variable

Specify the multirate filter object (`mfilt`) that you would like the block to implement. You can do this in one of three ways:

- You can fully specify the `mfilt` object in the block mask.
- You can enter the variable name of a `mfilt` object that is defined in any workspace.
- You can enter a variable name for a `mfilt` object that is not yet defined, as shown in the default value.

For more information on creating `mfilt` objects, see the `mfilt` function reference page in the Filter Design Toolbox documentation.

Framing

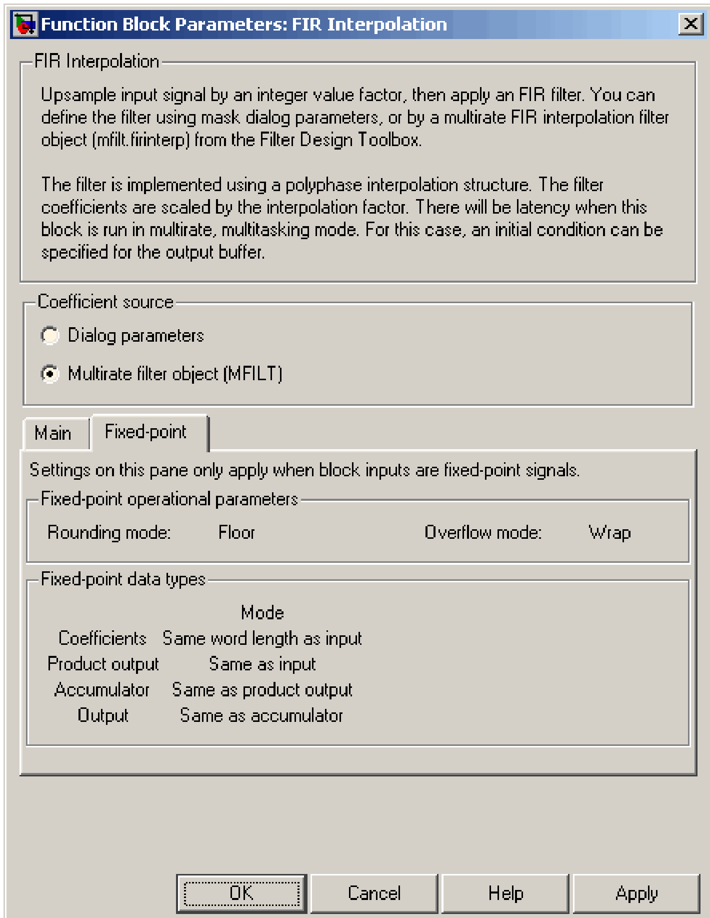
For frame-based operation, specify the method by which to implement the interpolation: increase the output frame rate, or increase the output frame size. This parameter cannot be set to `Maintain input frame rate` for sample-based signals.

View filter response

This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the `mfilt` object specified in the **Multirate filter variable** parameter. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

The **Fixed-point** pane of the FIR Interpolation block dialog appears as follows when **Multirate filter object (MFILT)** is specified in the **Coefficient source** group box.



The fixed-point settings of the filter object specified on the **Main** pane are displayed on the **Fixed-point** pane. You cannot change these settings directly on the block mask. To change the fixed-point settings you must edit the filter object directly.

For more information on multirate filter objects, see the `mfilt` function reference page in the Filter Design Toolbox documentation.

References

Fliege, N. J. *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets*. West Sussex, England: John Wiley & Sons, 1994.

Supported Data Types

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

See Also

FIR Decimation	Signal Processing Blockset
FIR Rate Conversion	Signal Processing Blockset
Upsample	Signal Processing Blockset
`fir1`	Signal Processing Toolbox
`fir2`	Signal Processing Toolbox

`firls`	Signal Processing Toolbox
`interp`	Signal Processing Toolbox

FIR Rate Conversion

Purpose

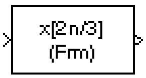
Upsample, filter, and downsample input signals

Library

Filtering / Multirate Filters

`dspmlti4`

Description



The FIR Rate Conversion block resamples the discrete-time input to a period *K*/*L* times the input sample period, where the integer *K* is specified by the **Decimation factor** parameter and the integer *L* is specified by the **Interpolation factor** parameter. The resampling process consists of the following steps:

1. The block upsamples the input to a higher rate by inserting *L*-1 zeros between input samples.
2. The upsampled data is passed through a direct-form II transpose FIR filter.
3. The block downsamples the filtered data to a lower rate by discarding *K*-1 consecutive samples following each sample retained.

K and *L* must be *relatively prime* integers; that is, the ratio *K*/*L* cannot be reducible to a ratio of smaller integers. The FIR Rate Conversion block implements the above three steps together using a polyphase filter structure, which is more efficient than straightforward upsample-filter-decimate algorithms. See Orfanidis [1] for more information.

The **FIR filter coefficients** parameter specifies the numerator coefficients of the FIR filter transfer function $H(z)$.

$$H(z) = B(z) = b_1 + b_2 z^{-1} + \ldots + b_m z^{-(m-1)}$$

The coefficient vector, `[b(1) b(2) ...  b(m)]`, can be generated by one of the Signal Processing Toolbox™ filter design functions (such as `fir1`), and should have a length greater than the interpolation factor ($m>L$). The filter should be lowpass with normalized cutoff frequency no greater than min(1/L,1/K). All filter states are internally initialized to zero.

The FIR Rate Conversion block supports real and complex floating-point and fixed-point inputs except for complex unsigned fixed-point inputs.

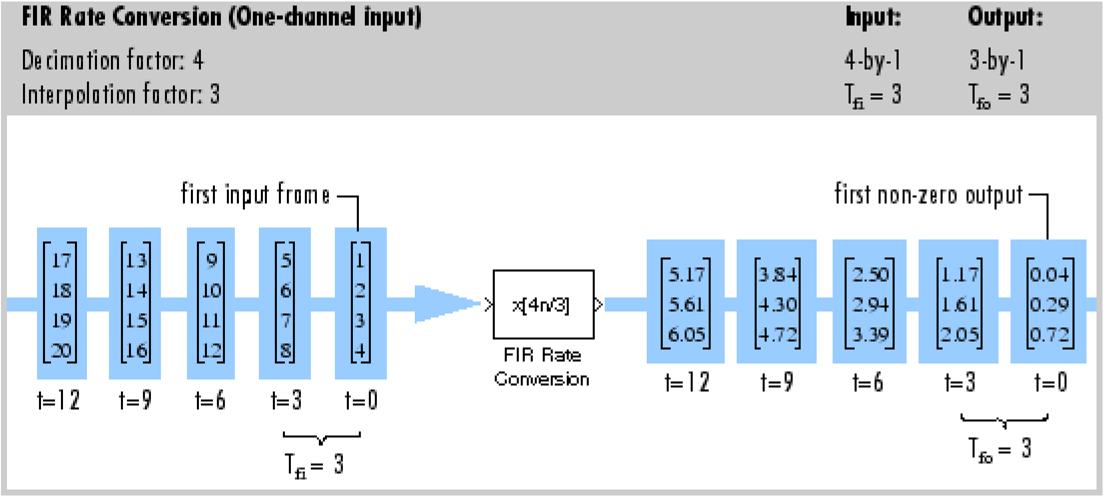
Frame-Based Operation

This block accepts *only* frame-based inputs. An M_i-by-N frame-based matrix input is treated as N independent channels, and the block resamples each channel independently over time.

The **Interpolation factor**, L, and **Decimation factor**, K, must satisfy the relation

$$\frac{K}{L} = \frac{M_i}{M_o}$$

for an *integer* output frame size M_o. The simplest way to satisfy this requirement is to let the **Decimation factor** equal the input frame size, M_i. The output frame size, M_o, is then equal to the **Interpolation factor**. This change in the frame size, from M_i to M_o, produces the desired rate conversion while leaving the output frame period the same as the input ($T_{fo} = T_{fi}$).

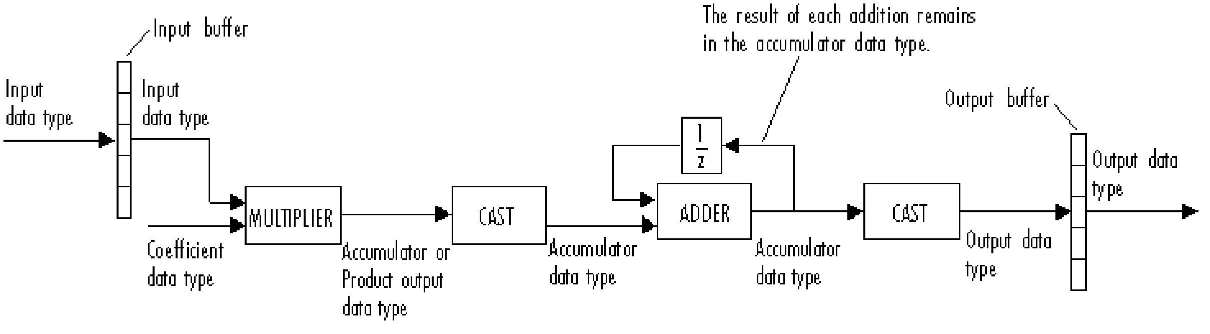


Latency

The FIR Rate Conversion block has no tasking latency. The block propagates the first filtered input (received at t=0) as the first output sample.

Fixed-Point Data Types

The following diagram shows the data types used within the FIR Rate Conversion block for fixed-point signals.



You can set the coefficient, product output, accumulator, and output data types in the block dialog as discussed in “Dialog Box” on page 2-514. The diagram shows that input data is stored in the input buffer in the same data type and scaling as the input. Filtered data is stored in the output buffer in the output data type and scaling that you set in the block dialog. Any initial conditions are also stored in the output buffer in the output data type and scaling you set in the block dialog.

The output of the multiplier is in the product output data type when at least one of the inputs to the multiplier is real. When both of the inputs to the multiplier are complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”.

Examples

The `doc_audio_src` model provides a simple illustration of one way to convert a speech signal from one sample rate to another. In this model, the data is first sampled at 22,050 Hz and then resampled at 8000 Hz. If you listen to the output, you can hear that the high frequency content has been removed from the signal, although the speech sounds basically the same.

Diagnostics

An error is generated when the relation between K and L shown above is not satisfied.

```
(Input port width)/(Output port width) must equal the
(Decimation factor)/(Interpolation factor).
```

A warning is generated when L and K are not relatively prime; that is, when the ratio L/K can be reduced to a ratio of smaller integers.

```
Warning: Integer conversion factors are not relatively prime
in block 'modelname/FIR Rate Conversion
(Frame)'. Converting ratio L/M to l/m.
```

The block scales the ratio to be relatively prime and continues the simulation.

Dialog Box

The FIR Rate Conversion block can operate in two different modes. Select the mode in the **Coefficient source** group box. If you select

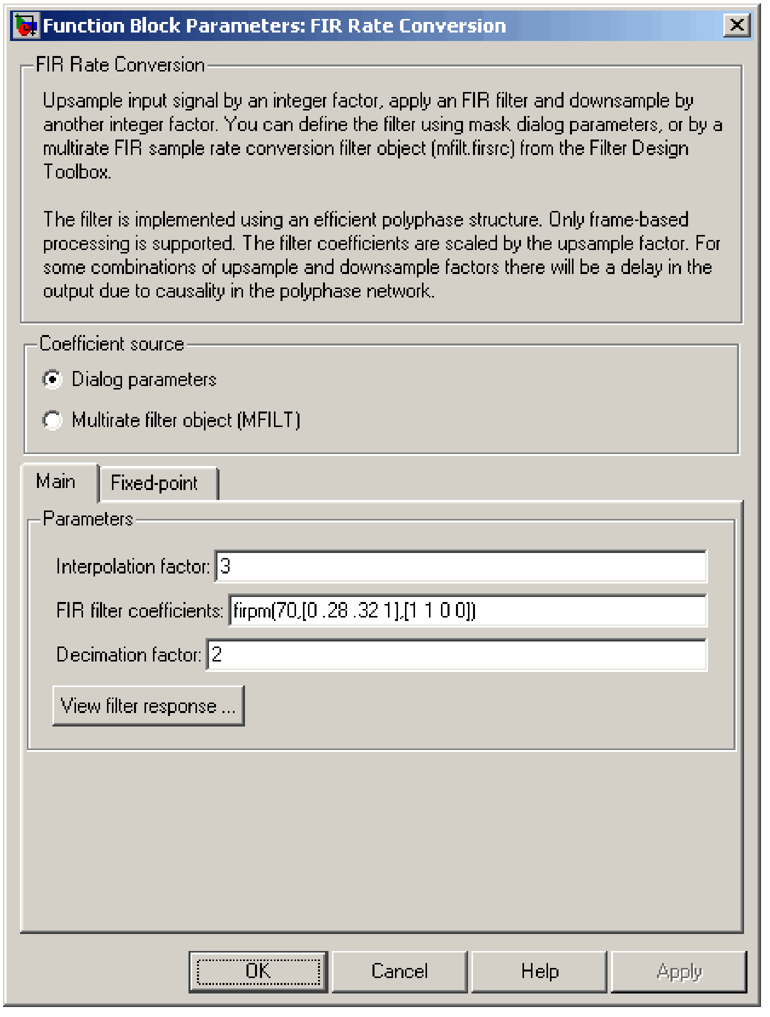
- **Dialog parameters**, you enter information about the filter such as structure and coefficients in the block mask.
- **Multirate filter object (MFILT)**, you specify the filter using a Filter Design Toolbox™ `mfilt` object.

Different items appear on the FIR Rate Conversion block dialog depending on whether you select **Dialog parameters** or **Multirate filter object (MFILT)** in the **Coefficient source** group box. See the following sections for details:

- "Specify Filter Characteristics in Dialog" on page 2-515
- "Specify Multirate Filter Object" on page 2-522

Specify Filter Characteristics in Dialog

The **Main** pane of the FIR Rate Conversion block dialog appears as follows when **Dialog parameters** is selected in the **Coefficient source** group box.



Interpolation factor
Specify the integer factor, *L*, by which to upsample the signal before filtering.

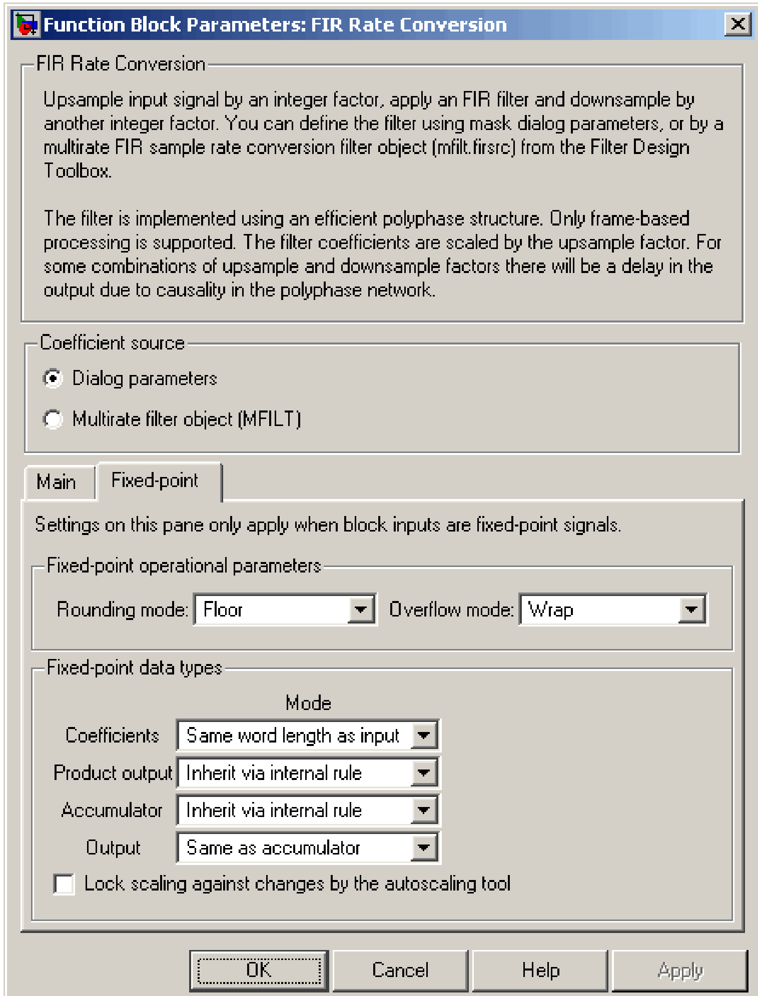
FIR filter coefficients
Specify the FIR filter coefficients in descending powers of *z*.

Decimation factor
Specify the integer factor, *K*, by which to downsample the signal after filtering.

View filter response
This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the filter defined in the block. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

The **Fixed point** pane of the FIR Rate Conversion block dialog appears as follows when **Dialog parameters** is specified in the **Coefficient source** group box.



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

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

Coefficients
Choose how you specify the word length and fraction length of the filter coefficients.

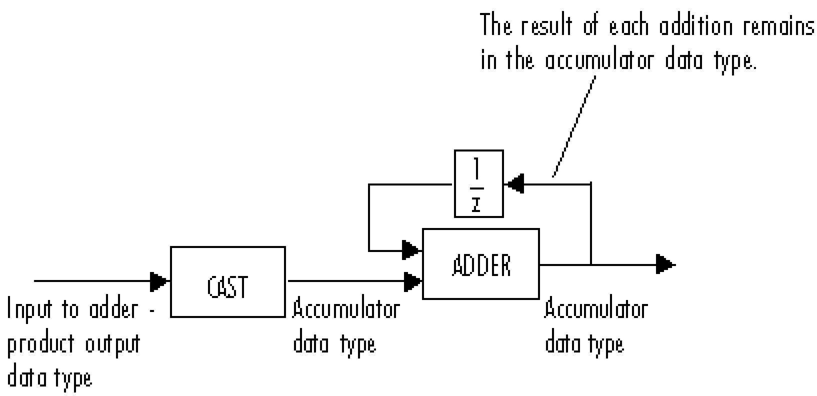
- When you select `Same word length as input`, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select `Specify word length`, you can enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the coefficients, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the coefficients. This block requires power-of-two slope and a bias of zero.
- The coefficients do not obey the **Round integer calculations toward** and the **Saturate on integer overflow** parameters; they are always saturated and rounded to `Nearest`.

Product output
Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-475 and "Multiplication Data Types" for

illustrations depicting the use of the product output data type in this block.

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see “Inherit via Internal Rule”.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator



As depicted above, inputs to the accumulator are cast to the accumulator data type. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you would like to designate this accumulator word and fraction lengths.

You also use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. See "Multiplication Data Types" for more information.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.

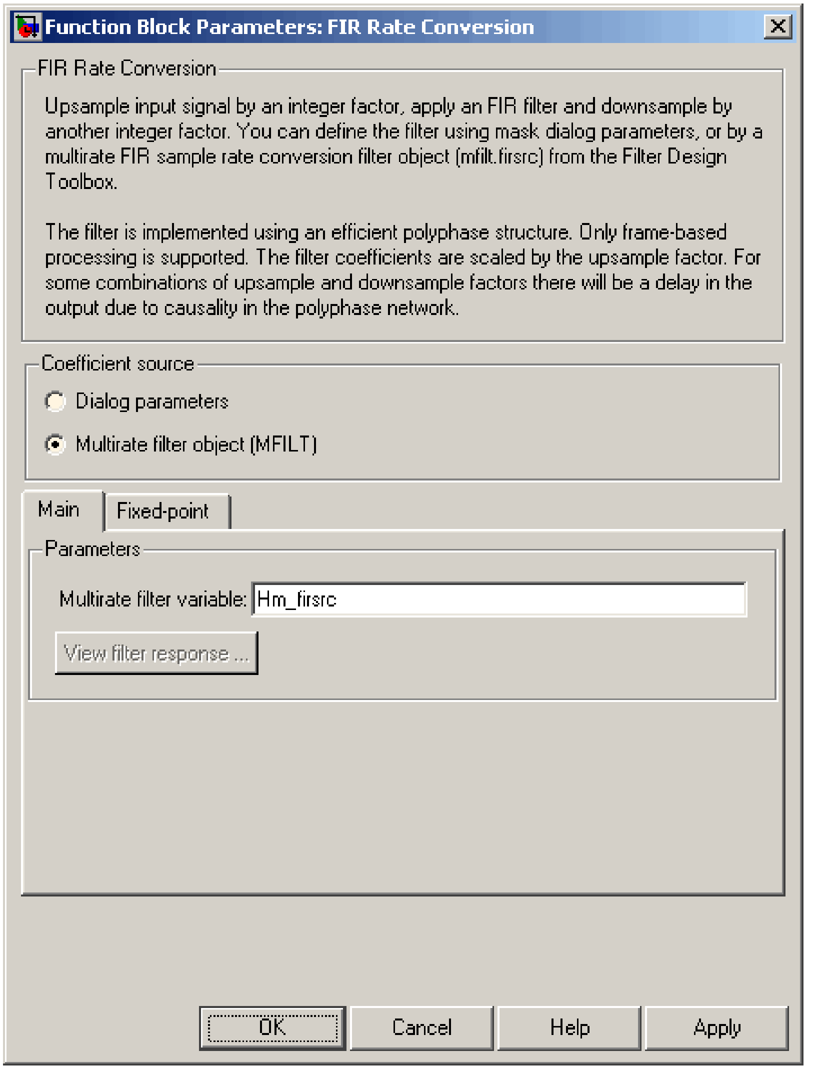
  A special case occurs when `Inherit via internal rule` is specified for **Accumulator**, and block inputs and coefficients are complex. In that case, the output word length is one less than the accumulator word length.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.

- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

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

Specify Multirate Filter Object

The **Main** pane of the FIR Rate Conversion block dialog appears as follows when **Multirate filter object (MFILT)** is specified in the **Coefficient source** group box.



Multirate filter variable

Specify the multirate filter object (`mfilt`) that you would like the block to implement. You can do this in one of three ways:

- You can fully specify the `mfilt` object in the block mask.
- You can enter the variable name of a `mfilt` object that is defined in any workspace.
- You can enter a variable name for a `mfilt` object that is not yet defined, as shown in the default value.

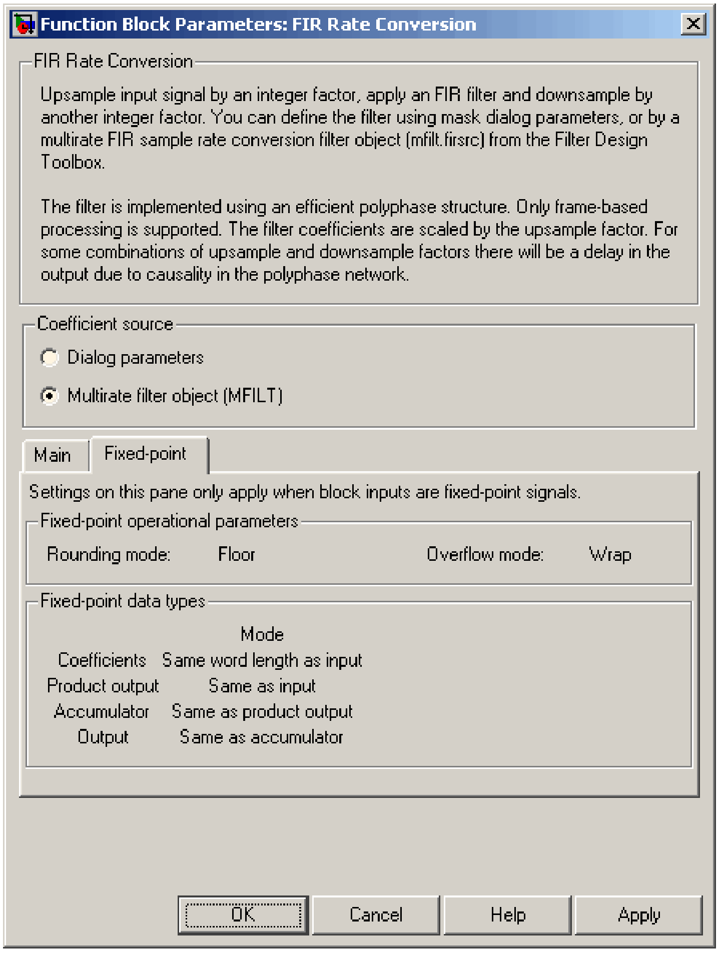
For more information on creating `mfilt` objects, see the `mfilt` function reference page in the Filter Design Toolbox documentation.

View filter response

This button opens the Filter Visualization Tool (`fvtool`) from the Signal Processing Toolbox product and displays the filter response of the `mfilt` object specified in the **Multirate filter variable** parameter. For more information on FVTool, see the Signal Processing Toolbox documentation.

Note This button is only available when the Filter Design Toolbox product is installed. If you specify a filter in the **Multirate filter variable** parameter, you must apply the filter by clicking the **Apply** button before using the **View filter response** button.

The **Fixed-point** pane of the FIR Rate Conversion block dialog appears as follows when **Multirate filter object (MFILT)** is specified in the **Coefficient source** group box.



The fixed-point settings of the filter object specified on the **Main** pane are displayed on the **Fixed-point** pane. You cannot change these settings directly on the block mask. To change the fixed-point settings you must edit the filter object directly.

For more information on multirate filter objects, see the `mfilt` function reference page in the Filter Design Toolbox documentation.

References

[1] Orfanidis, S. J. *Introduction to Signal Processing*. Prentice Hall, 1996.

Supported Data Types

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

See Also

Downsample	Signal Processing Blockset
FIR Decimation	Signal Processing Blockset
FIR Interpolation	Signal Processing Blockset
Upsample	Signal Processing Blockset
`fir1`	Signal Processing Toolbox
`fir2`	Signal Processing Toolbox

`firls`	Signal Processing Toolbox
`upfirdn`	Signal Processing Toolbox

See the following sections for related information:

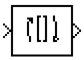
- "Converting Sample and Frame Rates"
- "Multirate Filters"

Purpose Flip input vertically or horizontally

Library Signal Management / Indexing

`dspindex`

Description



The Flip block vertically or horizontally reverses the *M*-by-*N* input matrix, *u*. The output always has the same dimension and frame status as the input.

When you select `Columns` from the **Flip along** menu, the block *vertically* flips the input so that the first row of the input is the last row of the output.

```
y = flipud(u)      % Equivalent MATLAB code
```

For convenience, length-*M* 1-D vector inputs are treated as *M*-by-1 column vectors for vertical flipping.

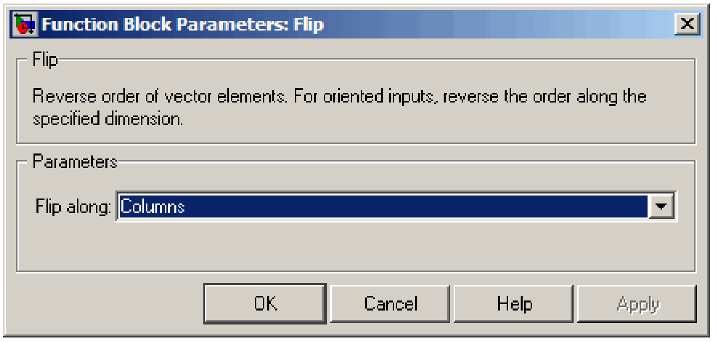
When you select `Rows` from the **Flip along** menu, the block *horizontally* flips the input so that the first column of the input is the last column of the output.

```
y = fliplr(u)      % Equivalent MATLAB code
```

For convenience, length-*N* 1-D vector inputs are treated as 1-by-*N* row vectors for horizontal flipping. The output always has the same dimension and frame status as the input.

This block supports Simulink® virtual buses.

Dialog Box



Flip along

The dimension along which to flip the input. `Columns` specifies vertical flipping, while `Rows` specifies horizontal flipping.

Supported Data Types

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

See Also

Selector	Simulink
Transpose	Signal Processing Blockset
Variable Selector	Signal Processing Blockset
`flipud`	MATLAB
`fliplr`	MATLAB

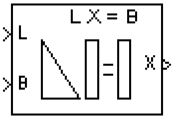
Forward Substitution

Purpose Solve $LX=B$ for X when L is lower triangular matrix

Library Math Functions / Matrices and Linear Algebra / Linear System Solvers

`dspsolvers`

Description



The Forward Substitution block solves the linear system $LX=B$ by simple forward substitution of variables, where L is the lower triangular M-by-M matrix input to the L port, and B is the M-by-N matrix input to the B port. The output is the solution of the equations, the M-by-N matrix X, and is always sample based. The block does not check the rank of the inputs.

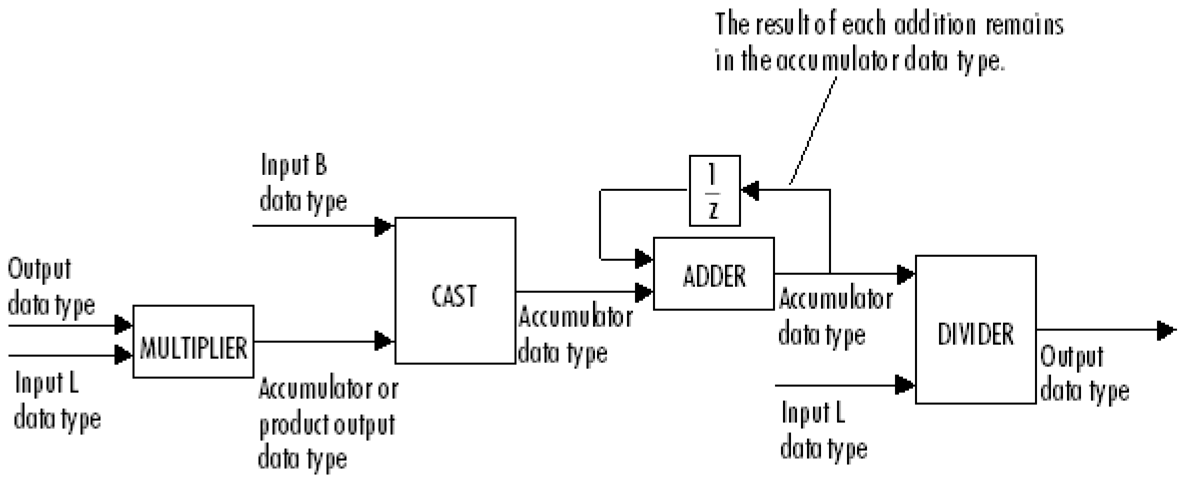
The block only uses the elements in the *lower triangle* of input L; the upper elements are ignored. When you select **Input L is unit-lower triangular**, the block replaces the elements on the diagonal of L with 1's. This is useful when matrix L is the result of another operation, such as an LDL decomposition, that uses the diagonal elements to represent the D matrix.

A length-M vector input at port B is treated as an M-by-1 matrix.

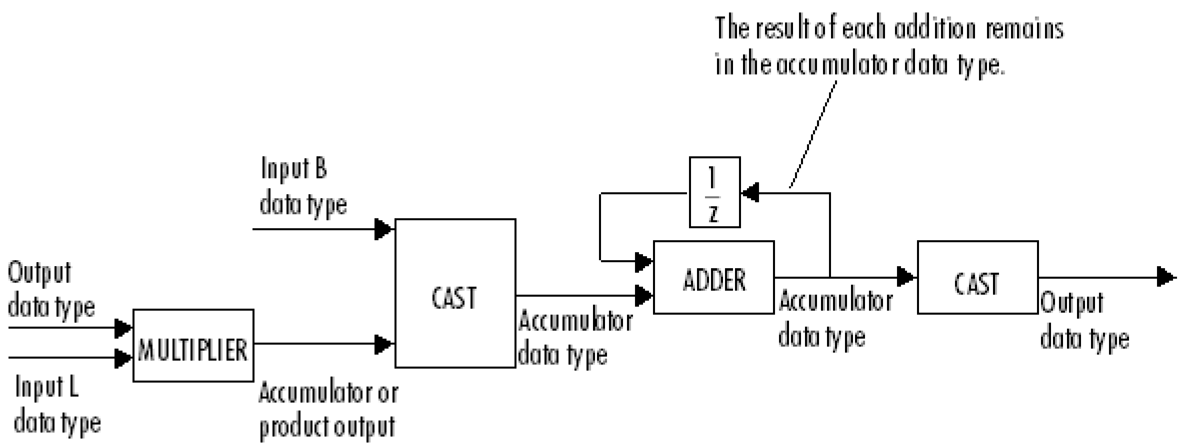
Fixed-Point Data Types

The following diagram shows the data types used within the Forward Substitution block for fixed-point signals.

When input L is not unit-lower triangular:



When input L is unit-lower triangular:

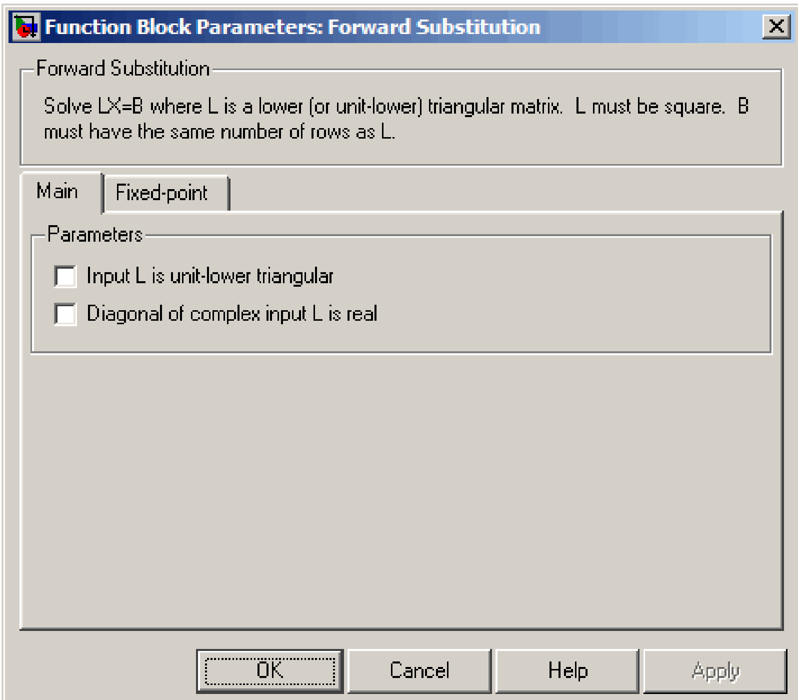


You can set the product output, accumulator, and output data types in the block dialog as discussed below.

The output of the multiplier is in the product output data type when the input is real. When the input is complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”.

Dialog Box

The **Main** pane of the Forward Substitution block dialog appears as follows.



Input L is unit-lower triangular

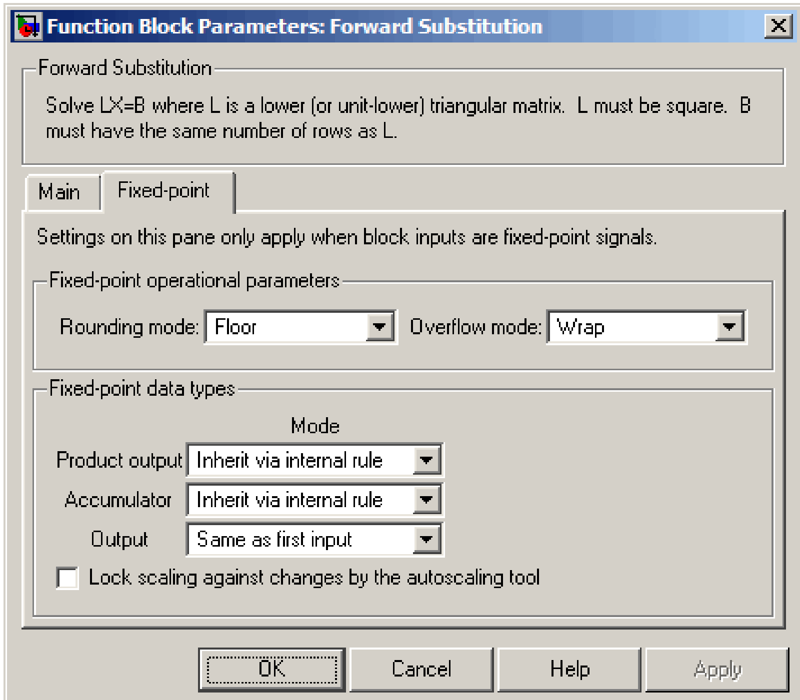
Select to replace the elements on the diagonal of L with 1’s.

Diagonal of complex input L is real

Select to optimize simulation speed when the diagonal elements of complex input *L* are real. This parameter is only visible when **Input L is unit-upper triangular** is not selected.

Note When L is a complex fixed-point signal, you must select either **Input L is unit-lower triangular** or **Diagonal of complex input L is real**. In such a case, any imaginary part of the diagonal of L is ignored.

The **Fixed-point** pane of the Forward Substitution block dialog appears as follows.



Rounding mode
: Select the rounding mode for fixed-point operations.

Overflow mode
: Select the overflow mode for fixed-point operations.

Product output
: Use this parameter to specify how you would like to designate the product output word and fraction lengths. See “Fixed-Point Data Types” on page 2-58 and “Multiplication Data Types” for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as first input`, these characteristics match those of the input L to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-58 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as first input`, these characteristics match those of the input L to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Use this parameter to specify how you would like to designate the output word and fraction lengths. See “Fixed-Point Data Types” on page 2-58 for an illustration depicting the use of the output data type in this block:

- When you select `Same as first input`, these characteristics match those of the input L to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

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

Supported Data Types

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

Port	Supported Data Types
B	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
X	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers

See Also

Backward Substitution	Signal Processing Blockset
Cholesky Solver	Signal Processing Blockset
LDL Solver	Signal Processing Blockset
Levinson-Durbin	Signal Processing Blockset
LU Solver	Signal Processing Blockset
QR Solver	Signal Processing Blockset

See “Linear System Solvers” for related information.

Purpose

Design fractional delay filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description

This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See "Fractional Delay Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Toolbox™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Port	Supported Data Types
D	Must be same as Input
Output	• Double-precision floating point • Single-precision floating point • Fixed point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers

Purpose

Specify sampling mode of output signal

Library

Signal Management / Signal Attributes

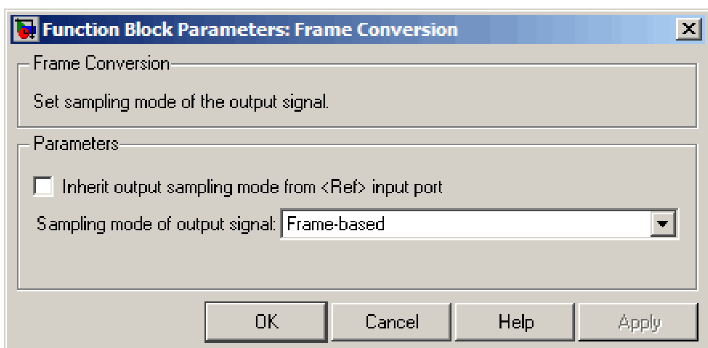
`dspsigattribs`

Description

The Frame Conversion block passes the input through to the output and sets the output sampling mode to the value of the **Sampling mode of output signal** parameter, which can be either `Frame-based` or `Sample-based`. The output sampling mode can also be inherited from the signal at the Ref (reference) input port, which you make visible by selecting the **Inherit output sampling mode from <Ref> input port** check box.

The Frame Conversion block does not make any changes to the input signal other than the sampling mode. In particular, the block does not rebuffer or resize 2-D inputs. Because 1-D vectors cannot be frame based, when the input is a length-M 1-D vector and the block is in `Frame-based` mode, the output is a frame-based M-by-1 matrix — that is, a single channel.

Dialog Box



Inherit output sampling mode from <Ref> input port
: Select to enable the Ref port from which the block inherits the output sampling mode.

Sampling mode of output signal

Specify the sampling mode of the output signal, `Frame-based` or `Sample-based`.

Supported Data Types

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

See Also

Buffer	Signal Processing Blockset
Check Signal Attributes	Signal Processing Blockset
Convert 1-D to 2-D	Signal Processing Blockset
Convert 2-D to 1-D	Signal Processing Blockset
Inherit Complexity	Signal Processing Blockset
Unbuffer	Signal Processing Blockset
Probe	Simulink
Reshape	Simulink
Signal Specification	Simulink

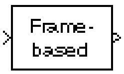
Purpose

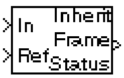
Specify frame status of output as sample based or frame based

Library

`dspobslib`

Description





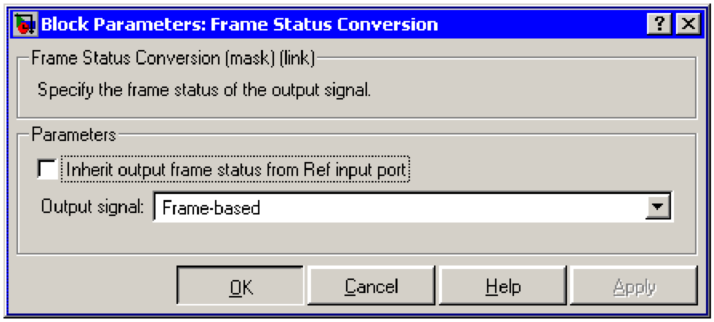
Note The Frame Status Conversion block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Frame Conversion block.

The Frame Status Conversion block passes the input through to the output, and sets the output frame status to the **Output signal** parameter, which can be either `Frame-based` or `Sample-based`. The output frame status can also be inherited from the signal at the `Ref` (reference) input port, which is made visible by selecting the **Inherit output frame status from Ref input port** check box.

When the **Output signal** parameter setting or the inherited signal's frame status differs from the input frame status, the block changes the input frame status accordingly, but does not otherwise alter the signal. In particular, the block does not rebuffer or resize 2-D inputs. Because 1-D vectors cannot be frame based, when the input is a length-M 1-D vector, and the **Output signal** parameter is set to `Frame-based`, the output is a frame-based M-by-1 matrix (that is, a single channel).

When the **Output signal** parameter or the inherited signal's frame status matches the input frame status, the block passes the input through to the output unaltered.

Dialog Box



Inherit output frame status from Ref input port

When selected, enables the `Ref` input port from which the block inherits the output frame status.

Output signal

The output frame status, `Frame-based` or `Sample-based`.

Supported Data Types

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

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

See Also

Check Signal Attributes	Signal Processing Blockset
Convert 1-D to 2-D	Signal Processing Blockset
Convert 2-D to 1-D	Signal Processing Blockset
Inherit Complexity	Signal Processing Blockset

From Audio Device

Purpose

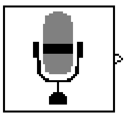
Read audio data from computer's audio device

Library

Signal Processing Sources

`dspsrcs4`

Description



The From Audio Device block reads audio data from an audio device in real time. This block has the following limitations:

- Not supported for use with the Simulink® Model block.
- Not currently supported on Solaris™ platforms.

Use the **Device** parameter to specify the device from which to acquire audio. This parameter is automatically populated based on the audio devices installed on your system. If you plug or unplug an audio device from your system, type `clear mex` at the MATLAB® command prompt to update this list.

Use the **Number of channels** parameter to specify the number of audio channels in the signal. For example:

- Enter `2` if the audio source is two channels (stereo).
- Enter `1` if the audio source is single channel (mono).
- Enter `6` if you are working with a 5.1 speaker system.

The block's output is an M-by-N matrix, where M is the number of consecutive samples and N is the number of audio channels.

Use the **Sample rate (Hz)** parameter to specify the number of samples per second in the signal. If the audio data is processed in uncompressed pulse code modulation (PCM) format, it should typically be sampled at one of the standard audio device rates: 8000, 11025, 22050, 44100, or 48000 Hz.

Use the **Device data type** parameter to specify the data type of the audio data that the device is placing in the buffer. You can choose:

- `8-bit integer`
- `16-bit integer`
- `24-bit integer`
- `32-bit float`
- `Determine from output data type`

If you choose `Determine from output data type`, the following table summarizes the block's behavior.

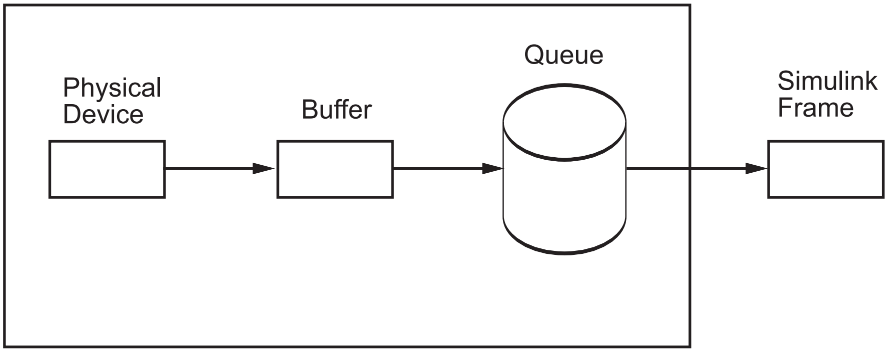
Output Data Type	Device Data Type
Double-precision floating point or single-precision floating point	32-bit floating point
32-bit integer	24-bit integer
16-bit integer	16-bit integer
8-bit integer	8-bit integer

If you choose `Determine from output data type` and the device does not support a data type, the block uses the next lowest precision data type supported by the device.

Use the **Frame size (samples)** parameter to specify the number of samples in the block's output. Use the **Output data type** parameter to specify the data type of audio data output by the block.

Buffering

The From Audio Device block buffers the data from the audio device using the process illustrated by the following figure.



From Audio Device Block

1. At the start of the simulation, the audio device begins writing the input data to a buffer. This data has the data type specified by the **Device data type** parameter.

2. When the buffer is full, the From Audio Device block writes the contents of the buffer to the queue. Specify the size of this queue using the **Queue duration (seconds)** parameter.

3. As the audio device appends audio data to the bottom of the queue, the From Audio Device block pulls data from the top of the queue to fill the Simulink frame. This data has the data type specified by the **Output data type** parameter.

Select the **Automatically determine buffer size** check box to allow the block to calculate a conservative buffer size using the following equation:

$$size = 2^{\left\lfloor \log_2 \frac{sr}{10} \right\rfloor}$$

In this equation, *size* is the buffer size, and *sr* is the sample rate. If you clear this check box, the **Buffer size (samples)** parameter appears on the block. Use this parameter to specify the buffer size in samples.

When the simulation throughput rate is lower than the hardware throughput rate, the queue, which is initially empty, fills up. If the queue is full, the block drops the incoming data from the audio device. When the simulation throughput rate is higher than the hardware throughput rate, the From Audio Device block waits for new samples to become available.

Troubleshooting

When Simulink cannot keep up with an audio device that is operating in real time, the queue fills up and the block begins to lose audio data. Here are several ways to deal with this situation:

- *Increase the queue duration*.

  The **Queue duration (seconds)** parameter specifies the duration of the signal, in seconds, that can be buffered during the simulation. This is the maximum length of time that the block's data demand can lag behind the hardware's data supply.

- *Increase the buffer size*.

  The size of the buffer processed in each interrupt from the audio device affects the performance of your model. If the buffer is too small, a large portion of hardware resources are used to write data to the queue. If the buffer is too big, Simulink must wait for the device to fill the buffer before it moves the data to the queue, which introduces latency.

- *Increase the simulation throughput rate*.

  Two useful methods for improving simulation throughput rates are increasing the signal frame size and compiling the simulation into native code:

  - Increase frame sizes and convert sample-based signals to frame-based signals throughout the model to reduce the amount of block-to-block communication overhead. This can increase throughput rates in many cases. However, larger frame sizes generally result in greater model latency due to initial buffering operations.

- Generate executable code with Real-Time Workshop® code generation software. Native code runs much faster than Simulink and should provide rates adequate for real-time audio processing.

Other ways to improve throughput rates include simplifying the model and running the simulation on a faster PC processor. For other ideas on improving simulation performance, see “Delay and Latency” and “Improving Simulation Performance and Accuracy” in the Simulink documentation.

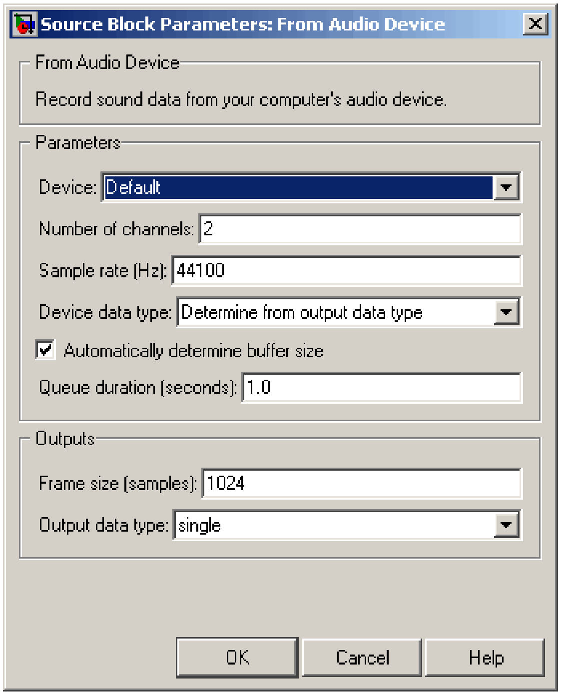
Audio Hardware API

The To Audio Device and From Audio Device blocks use the open-source PortAudio library in order to communicate with the audio hardware on a given computer. The PortAudio library supports a range of API's designed to communicate with the audio hardware on a given platform. The following API choices were made when building the PortAudio library for the Signal Processing Blockset™ product:

- Windows: DirectSound
- Linux: OSS
- Mac: CoreAudio

If you are interested in using a different audio API, such as ASIO (Windows®) or ALSA (Linux®) please search for PortAudio on the Matlab Central website.

Dialog Box



Device
Specify the device from which to acquire audio data.

Number of channels
Specify the number of audio channels.

Sample rate (Hz)
Specify the number of samples per second in the signal.

Device data type
Specify the data type used by the device to acquire audio data.

Automatically determine buffer size
Select this check box to enable the block to use a conservative buffer size.

Buffer size (samples)
: Specify the size of the buffer that the block uses to communicate with the audio device. This parameter is visible when the **Automatically determine buffer size** check box is cleared.

Queue duration (seconds)
: Specify the size of the queue in seconds.

Frame size (samples)
: Specify the number of samples in the block's output signal.

Output data type
: Select the data type of the block's output.

Supported Data Types

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

See Also

From Wave File	Signal Processing Blockset
To Audio Device	Signal Processing Blockset
`audiorecorder`	MATLAB

Purpose

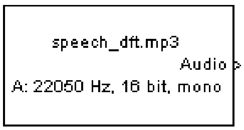
Read video frames and/or audio samples from compressed multimedia file

Library

Signal Processing Sources

`dspsrcs4`

Description



The From Multimedia File block reads video frames and/or audio samples from a multimedia file and imports them into a Simulink® model. Video processing requires the Video and Image Processing Blockset™ product.

You can view the video frames using a To Video Display block and listen to the audio using a To Audio Device block.

Notes

- This block supports code generation for the host computer that has file I/O available. This excludes RTWin (Real-Time Windows Target™ software, which does not support file I/O).
- On the UNIX® product platforms, this block supports only uncompressed AVI files. Also, it does not support OpenDML extensions to the AVI standard. For example, AVI files must be less than 4 gigabytes.
- This block performs best on platforms with Version 9.0 or later of DirectX® software and Version 9.0 or later of Windows Media® software.

The output ports of the From Multimedia File block change according to the content of the multimedia file. If the file contains video frames, the Image port appears on the block. If the file contains audio samples, the Audio port appears on the block.

Port	Output	Supported Data Types	Supports Complex Values?
Image	M-by-N matrix of intensity values or an M-by-N-by-P color video signal where P is the number of color planes	• Double-precision floating point • Single-precision floating point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers	
R, G, B	Matrix that represents one plane of the RGB video stream. Outputs from the R, G, or B port must have same dimensions.	Same as the Image port	No
Audio	Vector of audio data	• Double-precision floating point • Single-precision floating point • 16-bit signed integers • 8-bit unsigned integers	No

For sink blocks to display video data properly, double- and single-precision floating-point pixel values must be between 0 and 1. For other data types, the pixel values must be between the minimum and maximum values supported by their data type.

Use the **File name** parameter to specify the name of the multimedia file from which to read. If the location of this file is on your MATLAB® path, enter the filename. If the location of this file is not on your MATLAB path, use the **Browse** button to specify the full path to the file as well as the filename. On Windows® platforms, this parameter supports URLs that point to multimedia files. For more information, see “Supported File Formats – Windows® Platforms” on page 2-560.

Select the **Inherit sample time from file** check box if you want the sample time of the block to be the same as the sample time of the multimedia file. If you clear this check box, use the **Desired sample time** parameter to specify the block's sample time.

The default sample time for the From Multimedia File block is determined by the file that it references. You can also set the sample time for this block manually. Unless you already know the video's intended sample rate, set the sample rate from the file. If you have other sources in your model which need to operate at the same rate, you can set them to inherit the sample rate.

The table below provides the Sample time calculations that the From Multimedia File block uses for video and audio files:

For Video Files	For Audio Files	For Video and Audio Files
Sample time $= \frac{1}{FPS}$ where FPS is the Frames per Second.	Sample time = $\frac{1024}{SampleRate}$ where 1024 is the size of the audio frame, set by the block.	Sample time = $\frac{AudioSampleRate / FPS}{AudioSampleRate}$ When audio sample time is $\frac{AudioSampleRate}{FPS}$ is non-integer, the calculation does not equal $\frac{1}{FPS}$. In this case, to prevent synchronization problems, the corresponding video frame is dropped when the audio stream leads the video stream by more than $\frac{1}{FPS}$.

Use the **Number of times to play file** parameter to enter the number of times to play the file. The number you enter must be a positive integer or `inf`.

Use the **Output end-of-file indicator** parameter to determine when the last video frame or audio sample in the multimedia file is output from the block. When you select this check box, a Boolean output port labeled EOF appears on the block. The output from the EOF port is 1 when the last video frame or audio sample is output from the block. Otherwise, the output from the EOF port is 0.

Use the **Image color space** parameter to specify whether you want the block to output RGB or intensity video frames. If you select `RGB`, use the **Image signal** parameter to specify how to output a color signal. If you select `One multidimensional signal`, the block outputs an M-by-N-by-P color video signal, where P is the number of color planes, at one port. If you select `Separate color signals`, additional ports appear on the block. Each port outputs one M-by-N plane of an RGB video stream.

Use the **Audio output data type** parameter to set the data type of the audio samples output at the Audio port. You can choose `double`, `single`, `int16`, or `uint8`.

Use the **Video output data type** parameter to set the data type of the video frames output at the R, G, B, or Image ports. You can choose `double`, `single`, `int8`, `uint8`, `int16`, `uint16`, `int32`, `uint32`, or `Inherit from file`.

The block outputs one video frame at each Simulink time step. To calculate the number of audio samples to output at each time step, the block divides the audio sample rate by the video frame rate (fps). In the event that the audio sample rate is not evenly divisible by the number of video frames per second, two things occur:

- The block rounds the number of audio samples up to the nearest whole number.
- If necessary, the block will periodically drop a video frame to maintain synchronization for large audio and video files.

Troubleshooting

There are two issues that you need to be aware of when using the From Multimedia File block.

1. On Windows XP x64 platforms, the From Multimedia File block might not be able to read some compressed multimedia files because Windows XP x64 ships with a limited set of 64-bit video and audio codecs.

   There are two known workarounds:

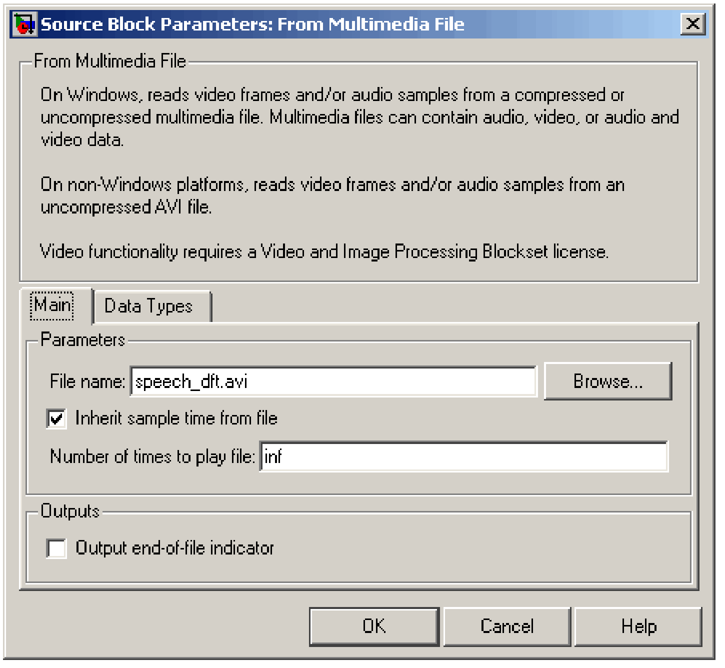
   a. Run the 32-bit version of MATLAB on your Windows XP x64 platform. Windows XP x64 ships with many 32-bit codecs that might work with your compressed multimedia file.

   b. Change the multimedia file to a different format using a codec that is supported on the Windows XP x64 platform.

2. When working with the From Multimedia File block, you might encounter this error: "ClassFactory cannot supply requested class". The From Multimedia File block uses the Microsoft® DirectX® product infrastructure, which does not allow you to play some AVI files from a remote network location.

   To workaround this issue, copy your AVI file to a local hard disk before using it with the From Multimedia File block.

Dialog Box



File name

Specify the name of the multimedia file from which to read.

Inherit sample time from file

Select this check box if you want the sample time of the block to be the same as the sample time of the multimedia file.

Desired sample time

Specify the block's sample time. This parameter is available if you clear the **Inherit sample time from file** check box.

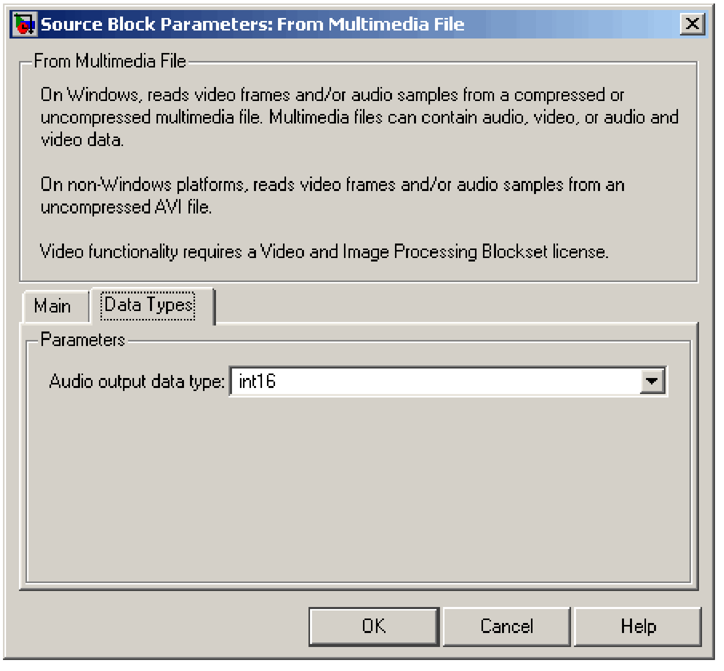
Number of times to play file

Enter a positive integer or `inf` to represent the number of times to play the file.

Output end-of-file indicator
Use this check box to determine whether the output is the last video frame or audio sample in the multimedia file.

Image color space
Specify whether you want the block to output RGB or intensity video frames.

Image signal
Specify how to output a color video signal. If you select `One multidimensional signal`, the block outputs an M-by-N-by-P color video signal, where P is the number of color planes, at one port. If you select `Separate color signals`, additional ports appear on the block. Each port outputs one M-by-N plane of an RGB video stream. This parameter is only available if you set the **Image color space** parameter to `RGB`.



Audio output data type

Set the data type of the audio samples output at the Audio port. This parameter is only available if the multimedia file contains audio.

Video output data type

Set the data type of the video data output from the block. This parameter is only available if the multimedia file contains video.

Supported File Formats – Windows Platforms

This block can support the following file formats provided you have the necessary codecs installed on your system.

Format	Filename Extensions
Apple® QuickTime, Macintosh® AIFF Resource	.qt, .aif, .aifc, .aiff, .mov
Microsoft® Windows Media formats	.avi, .asf, .asx, .rmi, .wav, .wma, .wax, .wmv
Moving Picture Experts Group (MPEG)	.mpg, .mpeg, .mlv, .mp2, .mp3, .mpa, .mpe
Audio formats	.au, .snd

Supported File Formats – UNIX Platforms

Format	Filename Extensions
Microsoft Windows Media formats (uncompressed)	.avi

See Also

To Multimedia File	Signal Processing Blockset
From Wave File	Signal Processing Blockset
Image From Workspace	Video and Image Processing Blockset

To Video Display	Video and Image Processing Blockset
Video From Workspace	Video and Image Processing Blockset
Video Viewer	Video and Image Processing Blockset
"Modeling and Simulating Discrete Systems"	Simulink

From Wave Device

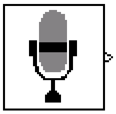
Purpose

Read audio data from standard audio device in real-time (32-bit Windows® operating systems only)

Library

`dspwin32`

Description



Note The From Wave Device block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the From Audio Device block.

The From Wave Device block reads audio data from a standard Windows audio device in real-time. It is compatible with most popular Windows hardware, including Sound Blaster cards. (Models that contain both this block and the To Wave Device block require a *duplex-capable* sound card.)

The **Use default audio device** parameter allows the block to detect and use the system's default audio hardware. This option should be selected on systems that have a single sound device installed, or when the default sound device on a multiple-device system is the desired source. In cases when the default sound device is not the desired input source, clear **Use default audio device**, and select the desired device in the **Audio device menu** parameter.

When the audio source contains two channels (stereo), the **Stereo** check box should be selected. When the audio source contains a single channel (mono), the **Stereo** check box should be cleared. For stereo input, the block's output is an M-by-2 matrix containing one frame (M consecutive samples) of audio data from each of the two channels. For mono input, the block's output is an M-by-1 matrix containing one frame (M consecutive samples) of audio data from the mono input. The frame size, M, is specified by the **Samples per frame** parameter. For M=1, the output is sample based; otherwise, the output is frame based.

The audio data is processed in uncompressed pulse code modulation (PCM) format, and should typically be sampled at one of the standard Windows audio device rates: 8000, 11025, 22050, or 44100 Hz. You can select one of these rates from the **Sample rate** parameter. To specify a different rate, select the **User-defined** option and enter a value in the **User-defined sample rate** parameter.

The **Sample Width (bits)** parameter specifies the number of bits used to represent the signal samples read by the audio device. The following settings are available:

- `8` — allocates 8 bits to each sample, allowing a resolution of 256 levels
- `16` — allocates 16 bits to each sample, allowing a resolution of 65536 levels
- `24` — allocates 24 bits to each sample, allowing a resolution of 16777216 levels (only for use with 24-bit audio devices)

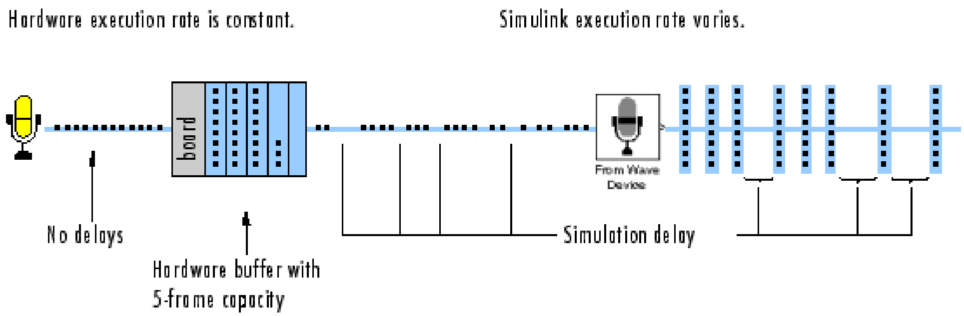
Higher sample width settings require more memory but yield better fidelity. The output from the block is independent of the **Sample width (bits)** setting. The output data type is determined by the **Data type** parameter setting.

Buffering

Since the audio device accepts real-time audio input, Simulink® software must read a continuous stream of data from the device throughout the simulation. Delays in reading data from the audio hardware can result in hardware errors or distortion of the signal. This means that the From Wave Device block must read data from the audio hardware as quickly as the hardware itself acquires the signal. However, the block often *cannot* match the throughput rate of the audio hardware, especially when the simulation is running from within Simulink rather than as generated code. (Simulink operations are generally slower than comparable hardware operations, and execution speed routinely varies during the simulation as the host operating system services other processes.) The block must therefore rely on a buffering strategy to ensure that signal data can be read on schedule without losing samples.

At the start of the simulation, the audio device begins writing the input data to a (hardware) buffer with a capacity of T_b seconds. The From Wave Device block immediately begins pulling the earliest samples off the buffer (first in, first out) and collecting them in length-M frames for output. As the audio device continues to append inputs to the bottom of the buffer, the From Wave Device block continues to pull inputs off the top of the buffer at the best possible rate.

The following figure shows an audio signal being acquired and output with a frame size of 8 samples. The buffer of the sound board is approaching its five-frame capacity at the instant shown, which means that the hardware is adding samples to the buffer more rapidly than the block is pulling them off. (If the signal sample rate was 8 kHz, this small buffer could hold approximately 0.005 second of data.



When the simulation throughput rate is higher than the hardware throughput rate, the buffer remains empty throughout the simulation. If necessary, the From Wave Device block simply waits for new samples to become available on the buffer (the block does not interpolate between samples). More typically, the simulation throughput rate is lower than the hardware throughput rate, and the buffer tends to fill over the duration of the simulation.

Troubleshooting

When the buffer size is too small in relation to the simulation throughput rate, the buffer might fill before the entire length of signal is processed. This usually results in a device error or undesired device output. When this problem occurs, you can choose to either increase the buffer size or the simulation throughput rate:

- *Increase the buffer size*

  The **Queue duration** parameter specifies the duration of signal, T_b (in real-time seconds), that can be buffered in hardware during the simulation. Equivalently, this is the maximum length of time that the block's data acquisition can lag the hardware's data acquisition. The number of frames buffered is approximately

  $$\frac{T_b F_s}{M}$$

  where F_s is the sample rate of the signal and M is the number of samples per frame. The required buffer size for a given signal depends on the signal length, the frame size, and the speed of the simulation. Note that increasing the buffer size might increase model latency.

- *Increase the simulation throughput rate*

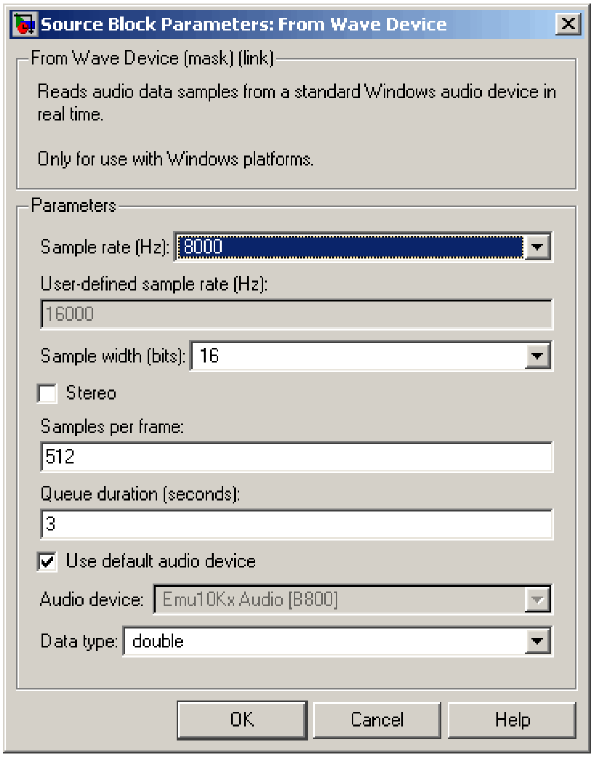
  Two useful methods for improving simulation throughput rates are increasing the signal frame size and compiling the simulation into native code:

  - Increase frame sizes (and convert sample-based signals to frame-based signals) throughout the model to reduce the amount of block-to-block communication overhead. This can drastically increase throughput rates in many cases. However, larger frame sizes generally result in greater model latency due to initial buffering operations.

  - Generate executable code with Real Time Workshop. Native code runs much faster than Simulink, and should provide rates adequate for real-time audio processing.

More general ways to improve throughput rates include simplifying the model, and running the simulation on a faster PC processor. See "Delay and Latency" and "Improving Simulation Performance and Accuracy" in the Simulink documentation for other ideas on improving simulation performance.

Dialog Box



Sample rate (Hz)
: The sample rate of the audio data to be acquired. Select one of the standard Windows rates or the `User-defined` option.

User-defined sample rate (Hz)
: The (nonstandard) sample rate of the audio data to be acquired.

Sample width (bits)
: The number of bits used to represent each signal sample.

Stereo
: Specifies stereo (two-channel) inputs when selected, mono (one-channel) inputs when cleared. Stereo output is M-by-2; mono output is M-by-1.

Samples per frame
The number of audio samples in each successive output frame, M. When the value of this parameter is `1`, the block outputs a sample-based signal.

Queue duration (seconds)
The length of signal (in seconds) to buffer to the hardware at the start of the simulation.

Use default audio device
Reads audio input from the system's default audio device when selected. Clear to enable the **Audio device ID** parameter and select a device.

Audio device
The name of the audio device from which to read the audio output (lists the names of the installed audio device drivers). Select **Use default audio device** when the system has only a single audio card installed.

Data type
The data type of the output: double-precision, single-precision, signed 16-bit integer, or unsigned 8-bit integer.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- 16-bit signed integer
- 8-bit unsigned integer

See Also

From Wave File	Signal Processing Blockset
To Wave Device	Signal Processing Blockset
`audiorecorder`	MATLAB

From Wave File

Purpose

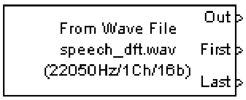
Read audio data from Microsoft® Wave (`.wav`) file

Library

Signal Processing Sources

`dspsrcs4`

Description



From Wave File

The From Wave File block streams audio data from a Microsoft Wave (`.wav`) file and generates a signal with one of the data types and amplitude ranges in the following table.

Output Data Type	Output Amplitude Range
`double`	±1
`single`	±1
`int16`	-32768 to 32767 (-2^{15} to $2^{15} - 1$)
`uint8`	0 to 255

The audio data must be in uncompressed pulse code modulation (PCM) format.

```
y = wavread('filename')       % Equivalent MATLAB code
```

The block supports 8-, 16-, 24-, and 32-bit Microsoft Wave (`.wav`) files.

The **File name** parameter can specify an absolute or relative path to the file. When the file is on the MATLAB® path or in the current directory (the directory returned by typing `pwd` at the MATLAB command line), you need only specify the file's name. You do not need to specify the`.wav` extension.

For an audio file containing C channels, the block's output is an M-by-C matrix containing one frame (M consecutive samples) of audio data from each channel. The frame size, M, is specified by the **Samples per output frame** parameter. For M=1, the output is sample based; otherwise, the output is frame based.

The output frame period, T_{fo}, is

$$T_{fo} = \frac{M}{F_s}$$

where F_s is the data sample rate in Hz.

To reduce the required number of file accesses, the block acquires L consecutive samples from the file during each access, where L is specified by the **Minimum number of samples for each read from file** parameter ($L \geq M$). For $L < M$, the block instead acquires M consecutive samples during each access. Larger values of L result in fewer file accesses, which reduces run-time overhead.

Use the **Data type** parameter to specify the data type of the block's output. Your choices are `double`, `single`, `uint8`, or `int16`.

Select the **Loop** check box if you want to play the file more than once. Then, enter the number of times to play the file. The number you enter must be a positive integer or `inf`.

Use the **Number of times to play file** parameter to enter the number of times to play the file. The number you enter must be a positive integer or `inf`, to play the file until you stop the simulation.

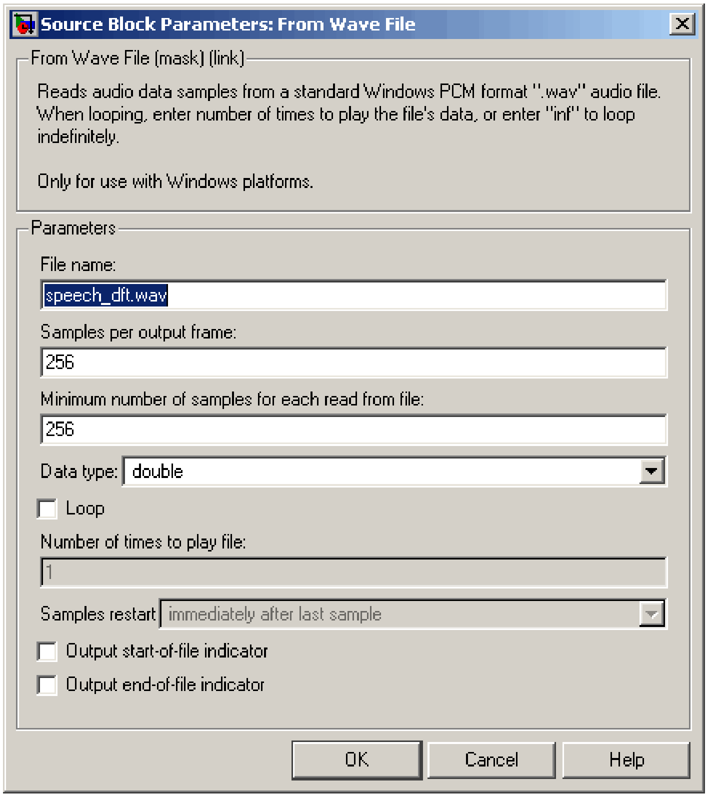
The **Samples restart** parameter determines whether the samples from the audio file repeat immediately or repeat at the beginning of the next frame output from the output port. When you select `immediately after last sample`, the samples repeat immediately. When you select `at beginning of next frame`, the frame containing the last sample value from the audio file is zero padded until the frame is filled. The block then places the first sample of the audio file in the first position of the next output frame.

Use the **Output start-of-file indicator** parameter to determine when the first audio sample in the file is output from the block. When you select this check box, a Boolean output port labeled SOF appears on the block. The output from the SOF port is 1 when the first audio sample in the file is output from the block. Otherwise, the output from the SOF port is 0.

Use the **Output end-of-file indicator** parameter to determine when the last audio sample in the file is output from the block. When you select this check box, a Boolean output port labeled EOF appears on the block. The output from the EOF port is 1 when the last audio sample in the file is output from the block. Otherwise, the output from the EOF port is 0.

The block icon shows the name, sample rate (in Hz), number of channels (1 or 2), and sample width (in bits) of the data in the specified audio file. All sample rates are supported; the sample width must be either 8, 16, 24, or 32 bits.

Dialog Box



File name

Enter the path and name of the file to read. Paths can be relative or absolute.

Samples per output frame

Enter the number of samples in each output frame, M. When the value of this parameter is `1`, the block outputs a sample-based signal.

Minimum number of samples for each read from file

Enter the number of consecutive samples to acquire from the file with each file access, L.

Data type

Select the output data type: `double`, `single`, `uint8`, or `int16`. The data type setting determines the output's amplitude range, as shown in the preceding table.

Loop

Select this check box if you want to play the file more than once.

Number of times to play file

Enter the number of times you want to play the file.

Samples restart

Select `immediately after last sample` to repeat the audio file immediately. Select `at beginning of next frame` to place the first sample of the audio file in the first position of the next output frame.

Output start-of-file indicator

Use this check box to determine whether the output contains the first audio sample in the file.

Output end-of-file indicator

Use this check box to determine whether the output contains the last audio sample in the file.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- 16-bit signed integer
- 8-bit unsigned integer

See Also

From Audio Device	Signal Processing Blockset
Signal From Workspace	Signal Processing Blockset
To Wave File	Signal Processing Blockset
`wavread`	MATLAB

Purpose

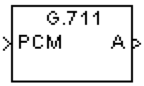
Quantize narrowband speech input signals

Library

Quantizers

`dspquant2`

Description



G.711 Codec

The G711 Codec block is a logarithmic scalar quantizer designed for narrowband speech. Narrowband speech is defined as a voice signal with an analog bandwidth of 4 kHz and a Nyquist sampling frequency of 8 kHz. The block quantizes a narrowband speech input signal so that it can be transmitted using only 8-bits. The G711 Codec block has three modes of operation: encoding, decoding, and conversion. You can choose the block's mode of operation by setting the **Mode** parameter.

If, for the **Mode** parameter, you choose `Encode PCM to A-law`, the block assumes that the linear PCM input signal has a dynamic range of 13 bits. Because the block always operates in saturation mode, it assigns any input value above $2^{12}-1$ to $2^{12}-1$ and any input value below -2^{12} to -2^{12}. The block implements an A-law quantizer on the input signal and outputs A-law index values. When you choose `Encode PCM to mu-law`, the block assumes that the linear PCM input signal has a dynamic range of 14 bits. Because the block always operates in saturation mode, it assigns any input value above $2^{13}-1$ to $2^{13}-1$ and any input value below -2^{13} to . The block implements a mu-law quantizer on the input signal and outputs mu-law index values.

If, for the **Mode** parameter, you choose `Decode A-law to PCM`, the block decodes the input A-law index values into quantized output values using an A-law lookup table. When you choose `Decode mu-law to PCM`, the block decodes the input mu-law index values into quantized output values using a mu-law lookup table.

If, for the **Mode** parameter, you choose `Convert A-law to mu-law`, the block converts the input A-law index values to mu-law index values. When you choose `Convert mu-law to A-law`, the block converts the input mu-law index values to A-law index values.

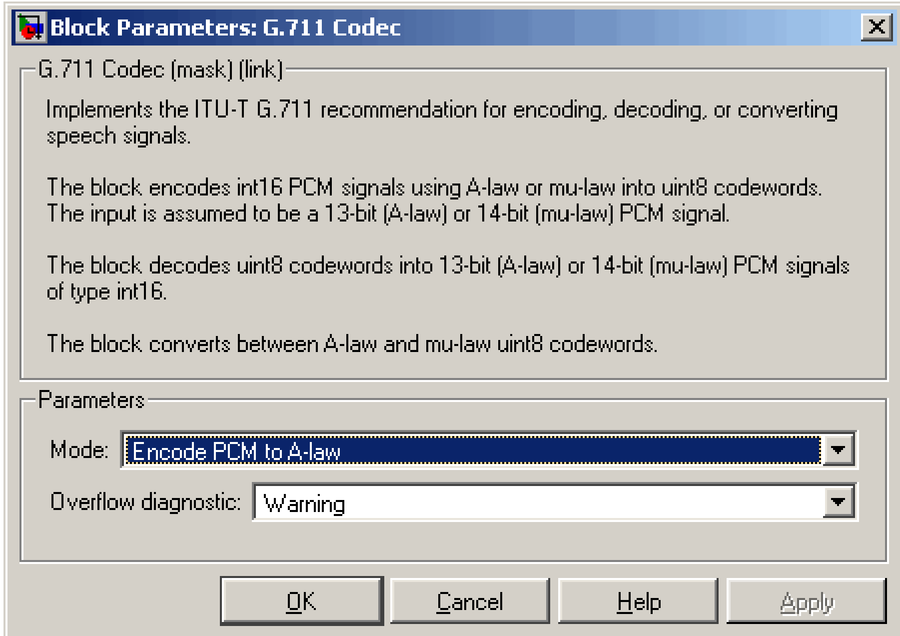
Note Set the **Mode** parameter to `Convert A-law to mu-law` or `Convert mu-law to A-law` only when the input to the block is A-law or mu-law index values.

If, for the **Mode** parameter, you choose `Encode PCM to A-law` or `Encode PCM to mu-law`, the **Overflow diagnostic** parameter appears on the block parameters dialog box. Use this parameter to determine the behavior of the block when overflow occurs. The following options are available:

- `Ignore` — Proceed with the computation and do not issue a warning message.
- `Warning` — Display a warning message in the MATLAB® Command Window, and continue the simulation.
- `Error` — Display an error dialog box and terminate the simulation.

Note Like all diagnostic parameters on the Configuration Parameters dialog box, **Overflow diagnostic** parameter is set to `Ignore` in the code generated for this block by Real-Time Workshop® code generation software.

Dialog Box



Mode

- When you choose `Encode PCM to A-law`, the block implements an A-law encoder.
- When you choose `Encode PCM to mu-law`, the block implements a mu-law encoder.
- When you choose `Decode A-law to PCM`, the block decodes the input index values into quantized output values using an A-law lookup table.
- When you choose `Decode mu-law to PCM`, the block decodes the input index values into quantized output values using a mu-law lookup table.
- When you choose `Convert A-law to mu-law`, the block converts the input A-law index values to mu-law index values.

- When you choose `Convert mu-law to A-law`, the block converts the input mu-law index values to A-law index values.

Overflow diagnostic

Use this parameter to determine the behavior of the block when overflow occurs.

- Select `Ignore` to proceed with the computation without a warning message.
- Select `Warning` to display a warning message in the MATLAB Command Window and continue the simulation.
- Select `Error` to display an error dialog box and terminate the simulation.

This parameter is only visible if, for the **Mode** parameter, you select `Encode PCM to A-law` or `Encode PCM to mu-law`.

References

ITU-T Recommendation G.711, "Pulse Code Modulation (PCM) of Voice Frequencies," *General Aspects of Digital Transmission Systems*; *Terminal Equipments*, International Telecommunication Union (ITU), 1993.

Supported Data Types

Port	Supported Data Types
PCM	• 16-bit signed integers
A	• 8-bit unsigned integers
mu	• 8-bit unsigned integers

See Also

Quantizer	Simulink
Scalar Quantizer Decoder	Signal Processing Blockset
Scalar Quantizer Design	Signal Processing Blockset
Uniform Decoder	Signal Processing Blockset
Uniform Encoder	Signal Processing Blockset
Vector Quantizer Decoder	Signal Processing Blockset
Vector Quantizer Design	Signal Processing Blockset
Vector Quantizer Encoder	Signal Processing Blockset

Halfband Filter

Purpose

Design halfband filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description

This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See "Halfband Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose Design highpass filter

Library Filtering / Filter Design Toolbox

`dspfdesign`

Description This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box See "Highpass Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Design Hilbert filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description

This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See "Hilbert Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Histogram

Purpose

Generate histogram of input or sequence of inputs

Library

Statistics

`dspstat3`

Description

The Histogram block computes the frequency distribution of the elements in a vector input, of the elements in each channel of a frame-based matrix input, or of the elements in a sample based N-D array. The **Running histogram** parameter selects between basic operation and running operation, described below. The Histogram block accepts real and complex fixed-point and floating-point inputs.

The block distributes the elements of the input into the number of discrete bins specified by the **Number of bins** parameter, `n`.

```
y = hist(u,n)     % Equivalent MATLAB code
```

Complex fixed-point inputs are distributed according to their magnitude squared values; complex floating-point inputs are distributed by their normalized values.

The histogram value for a given bin represents the frequency of occurrence of the input values bracketed by that bin. You specify the upper boundary of the highest-valued bin in the **Upper limit of histogram** parameter, B_M, and the lower boundary of the lowest-valued bin in the **Lower limit of histogram** parameter, B_m. The bins have equal width of

$$\Delta = \frac{B_M - B_m}{n}$$

and centers located at

$$B_m + \left(k + \frac{1}{2}\right)\Delta \qquad k = 0, 1, 2, \ldots, n-1$$

Input values that fall on the border between two bins are placed into the lower valued bin; that is, each bin includes its upper boundary. For example, a bin of width 4 centered on the value 5 contains the input value 7, but not the input value 3. Input values greater than the **Upper limit of histogram** parameter or less than **Lower limit of histogram** parameter are placed into the highest valued or lowest valued bin, respectively.

The values you enter for the **Upper limit of histogram** and **Lower limit of histogram** parameters must be real-valued scalars. `NaN` and `inf` are not valid values for the **Upper limit of histogram** and **Lower limit of histogram** parameters.

Basic Operation

When the **Running histogram** check box is not selected, the Histogram block computes the frequency distribution of the current input.

For frame-based M-by-N inputs, (including 1-by-N row vectors and M-by-1 column vectors), the Histogram block computes a histogram for each channel of the M-by-N matrix independently. The block outputs an n-by-N matrix, where n is the **Number of bins** specified in the Histogram block. The jth column of the output matrix contains the histogram for the data in the jth column of the M-by-N input matrix.

For all sample-based N-D input arrays, including length-M 1-D vectors and 1-by-N row vectors, the Histogram block computes the frequency distribution of the input data. The block outputs an n-by-1 vector, where n is the **Number of bins** specified in the Histogram block.

Running Operation

When you select the **Running histogram** check box, the Histogram block computes the frequency distribution of both the past and present data for successive inputs. The block resets the histogram (by emptying all of the bins) when it detects a reset event at the optional Rst port. See "Resetting the Running Histogram" on page 2-586 for more information on how to trigger a reset.

For frame-based M-by-N inputs (including 1-by-N row vectors and M-by-1 column vectors), the Histogram block computes a running histogram for each channel of the *M*-by-*N* matrix. The block outputs an *n*-by-*N* matrix, where *n* is the **Number of bins** specified in the Histogram block. The *j*th column of the output matrix contains the running histogram for the *j*th column of the *M*-by-*N* input matrix.

For all sample-based N-D input arrays, including length-*M* 1-D vectors, the Histogram block computes a running histogram for the data in the first dimension of the input. The block outputs an *n*-by-1 vector, where *n* is the **Number of bins** specified in the Histogram block.

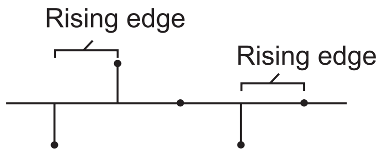
Note When the histogram block is used in running mode and the input data type is non-floating point, the output of the histogram is stored as a `uint32` data type. The largest number that can be represented by this data type is $2^{32}-1$. If the range of the `uint32` data type is exceeded, the output data will wrap back to 0.

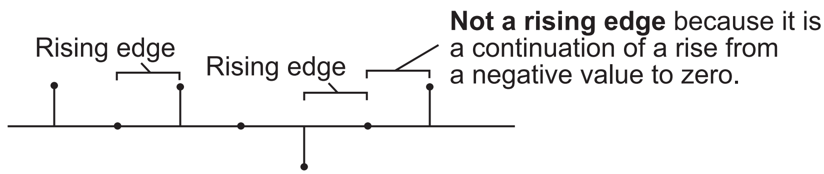
Resetting the Running Histogram

The block resets the running histogram whenever a reset event is detected at the optional Rst port. The reset signal and the input data signal must be the same rate.

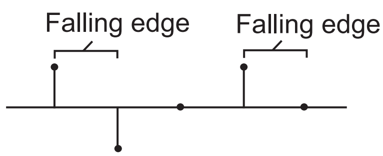
To enable the Rst port, select the **Reset port** parameter. You specify the reset event in the **Trigger type** parameter, and it can be one of the following:

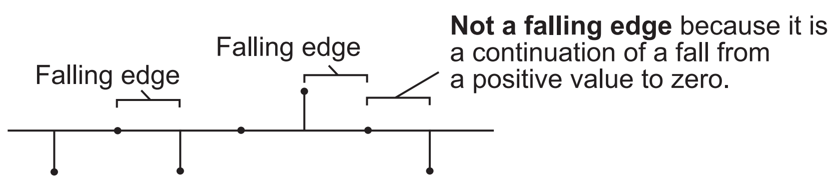
- `Rising edge` — Triggers a reset operation when the Rst input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)





- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)





- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described earlier)
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero

Examples

Real Input Data

The bin boundaries created by the Histogram block are determined by the data type of the input:

- Bin boundaries for real, double-precision input are cast into the data type `double`.
- Bin boundaries for real, fixed-point input are cast into the `int8` data type.

The following example shows the differences in the output of the Histogram block based on the data type of the input.

To create this model you need the following blocks.

Block	Library	Quantity
Constant	Simulink® / Sources library	2
Display	Signal Processing Sinks	2
Histogram	Statistics	2

The parameter settings for the Double Precision Input Constant block are:

- **Constant value** = `double([1 2 3 4 5]')`
- **Interpret parameters as 1-D** = Clear this check box.
- **Sampling mode** = `Sample based`
- **Sample time** = `inf`

The parameter settings for the Fixed-Point Input Constant block are:

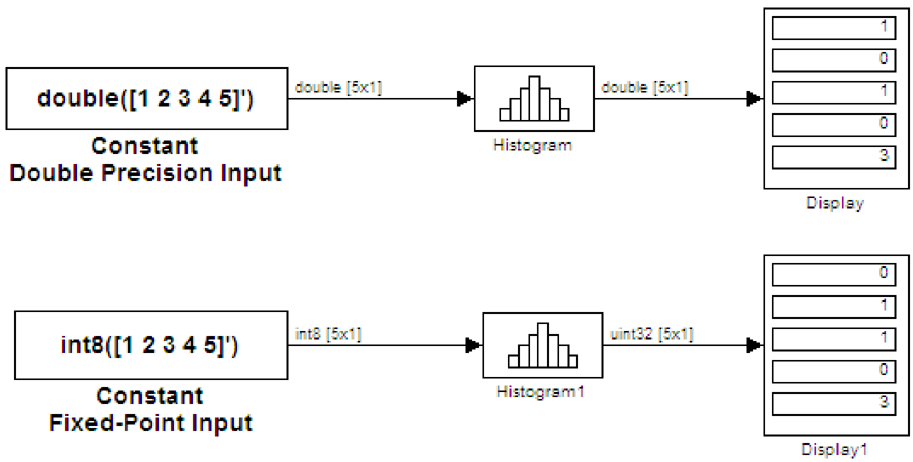
- **Constant value** = `int8([1 2 3 4 5]')`
- **Interpret parameters as 1-D** = Clear this check box.
- **Sampling mode** = `Sample based`

- **Sample time** = `inf`

The parameter settings for both Histogram blocks are:

- **Lower limit of histogram** = `1`
- **Upper limit of histogram** = `3`
- **Number of bins** = `5`
- **Normalized** = Clear this check box.
- **Running histogram** = Clear this check box.

Connect the blocks as shown in the following figure, and run your model.

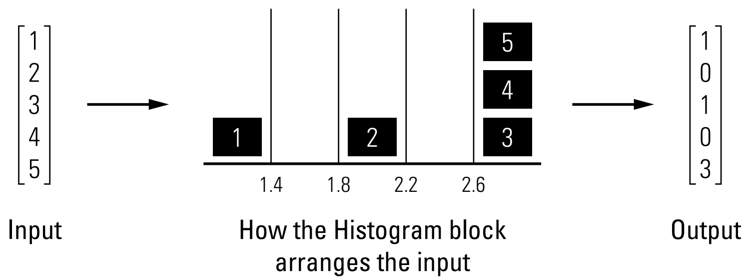


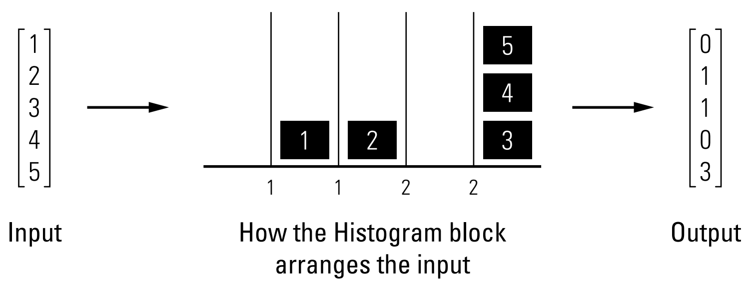
Running this model generates the following warning:

```
Warning:  The bin width resulting from the specified
parameters is less than the precision of the input data
type.  This might cause unexpected results.  Since bin
width is calculated by ((upper limit - lower limit)/number
of bins), you could increase upper limit or decrease lower
limit or number of bins.
```

This warning alerts you that it is not a good use case to have a histogram where 2 or more bin boundaries are the same. As the warning suggests, increasing the range of the limits of the histogram, or decreasing the number of bins, can correct this problem.

The following figures illustrate the different bins that are created by the Histogram block. The top figure shows the histogram for double-precision input, and the bottom figure shows the histogram for fixed-point input. The output of the histogram block differs based on the data type of the input, and the bin boundaries are duplicated in the histogram for the fixed-point input.





Complex Input Data

The bin boundaries created by the Histogram block are determined by the data type of the input.

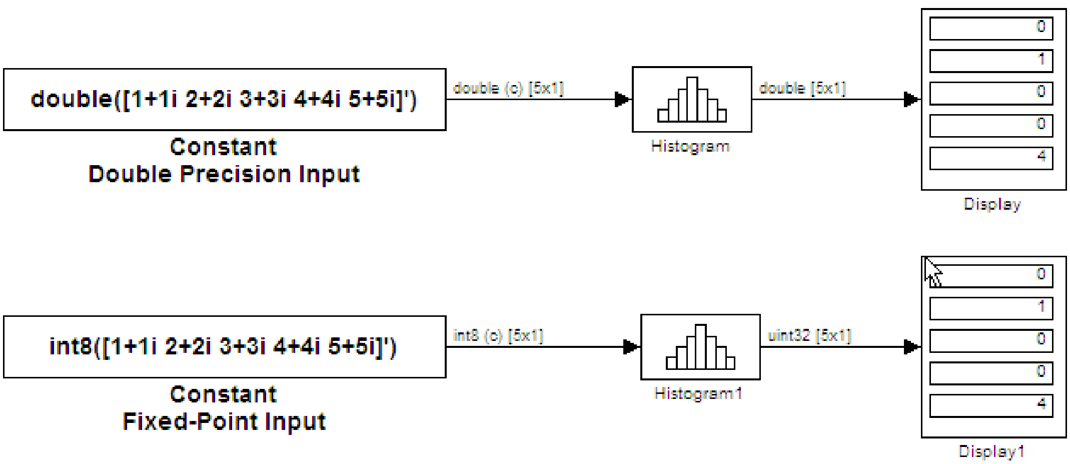
- Bin boundaries for complex, double-precision input are cast into the data type `double`. All complex, double-precision input values are placed in bins according to their normalized values.
- Bin boundaries for complex, fixed-point input are cast into the data type `double` and squared. All complex, fixed-point input values are placed in bins according to their magnitude-squared value.

The following example shows the differences in the bins created by the Histogram block based on the data type of the complex input.

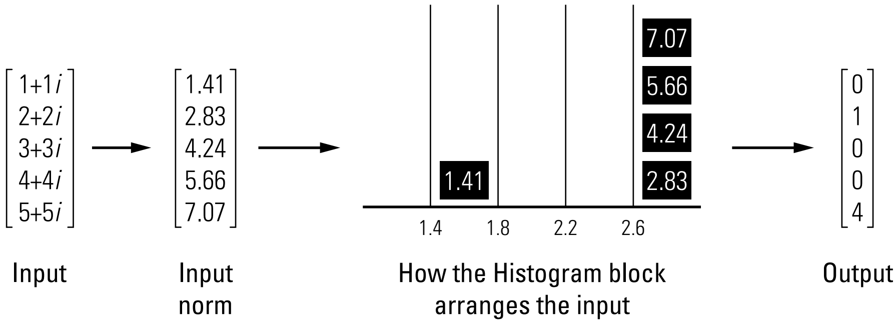
Using the same model you created for the example with real input data, modify the following parameters:

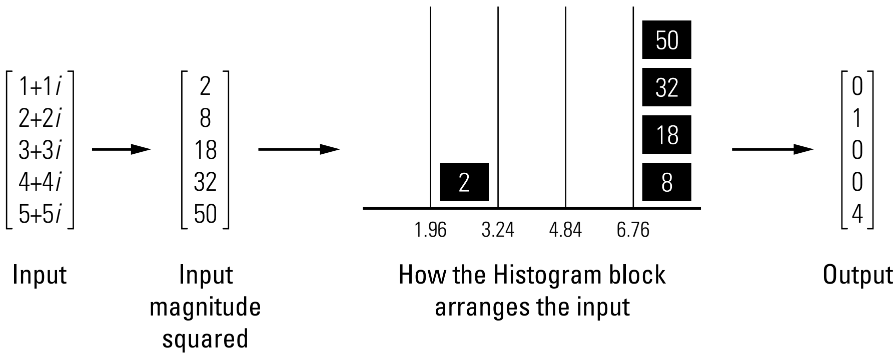
- In the Double Precision Input Constant block, set the **Constant value** parameter to `double([1+1i 2+2i 3+3i 4+4i 5+5i]')`
- In the Fixed-Point Input Constant block, set the **Constant value** parameter to `int8([1+1i 2+2i 3+3i 4+4i 5+5i]')`

Run your model. It should look similar to the following figure:



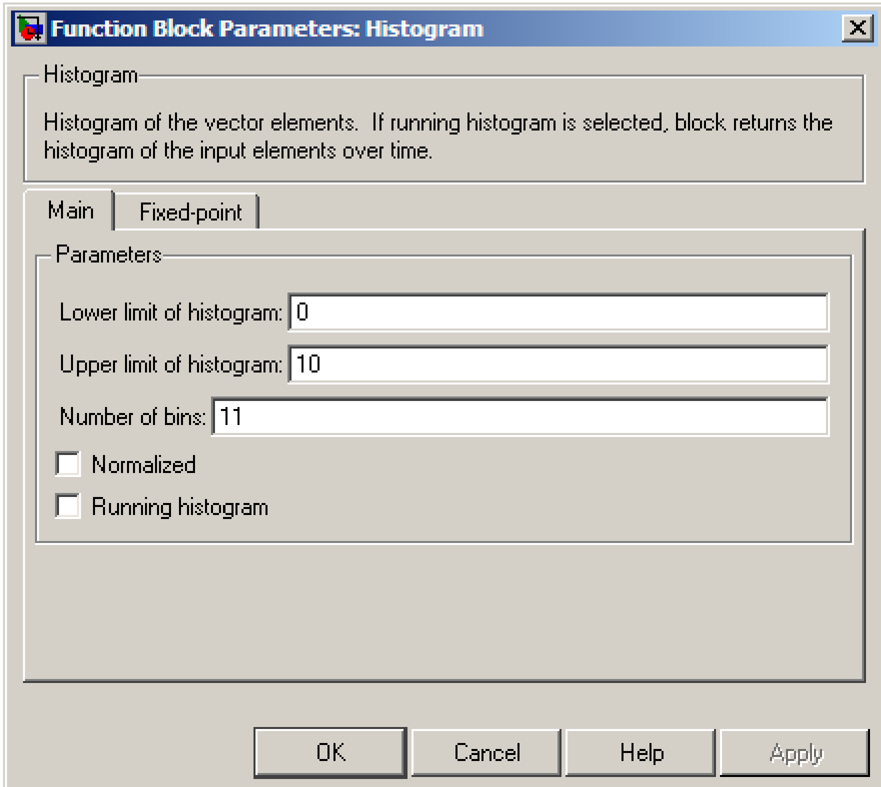
In this case, the Histogram block outputs the same result. The figures below illustrate how the Histogram block compares the input values to the bins it creates. The double-precision inputs are normalized for comparison, whereas the fixed-point inputs are placed using their magnitude squared value. The top figure shows the histogram for the double-precision input, and the bottom figure shows the histogram for the fixed-point input.





Dialog Box

The **Main** pane of the Histogram block dialog appears as follows.



Lower limit of histogram

Enter a real-valued scalar for the lower boundary, B_m, of the lowest-valued bin. `NaN` and `inf` are not valid values for B_m. Tunable.

Upper limit of histogram

Enter a real-valued scalar for the upper boundary, B_M, of the highest-valued bin. `NaN` and `inf` are not valid values for B_M. Tunable.

Number of bins

The number of bins, n, in the histogram.

Normalized

When selected, the output vector, v, is normalized such that `sum(v) = 1`.

Use of this parameter is not supported for fixed-point signals.

Running histogram

Set to enable the running histogram operation, and clear to enable basic histogram operation. For more information, see "Basic Operation" on page 2-585 and "Running Operation" on page 2-585.

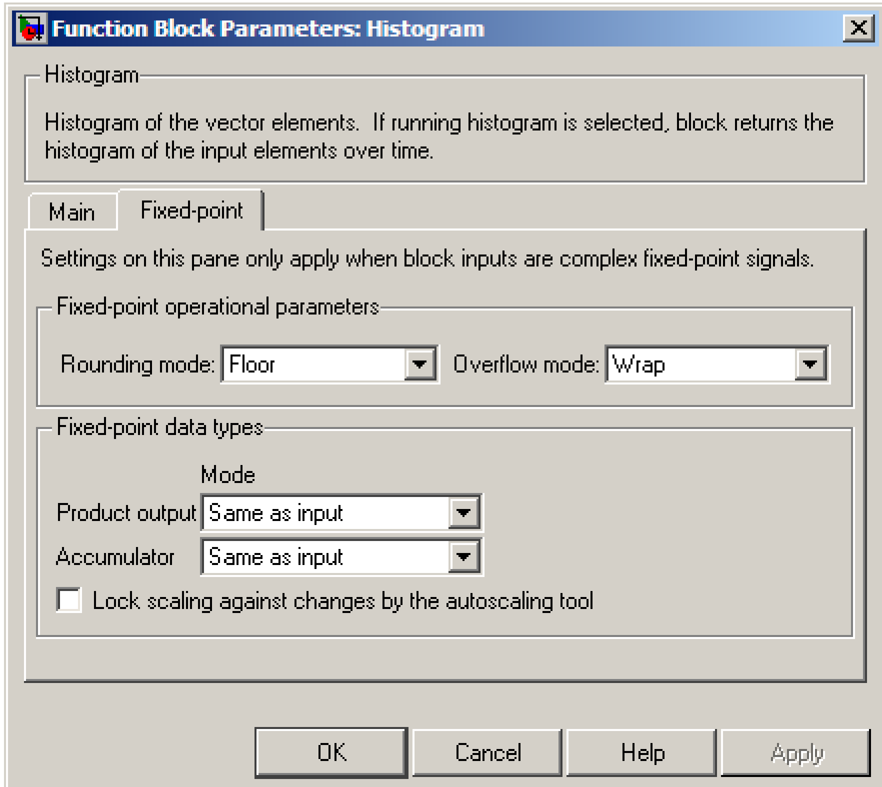
Reset port

Enables the Rst input port when selected. The reset signal and the input data signal must be the same rate. This parameter is enabled only when you set the **Running histogram** parameter. For more information, see "Running Operation" on page 2-585.

Trigger type

The type of event that resets the running histogram. For more information, see "Resetting the Running Histogram" on page 2-586. This parameter is enabled only when you set the **Reset port** parameter.

The **Fixed-point** pane of the Histogram block dialog appears as follows.



Note The fixed-point parameters listed are only used for fixed-point complex inputs, which are distributed by squared magnitude.

Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Product output
Use this parameter to specify how you would like to designate the product output word and fraction lengths:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator
Use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block:

- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

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

See Also

Sort	Signal Processing Blockset
`hist`	MATLAB

Purpose

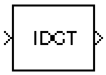
Compute inverse discrete cosine transform (IDCT) of input

Library

Transforms

`dspxfrm3`

Description



The IDCT block computes the inverse discrete cosine transform (IDCT) of each channel in the M-by-N input matrix, `u`.

```
y = idct(u)     % Equivalent MATLAB code
```

For both sample-based and frame-based inputs, the block assumes that each input column is a frame containing M consecutive samples from an independent channel. The frame size, M, must be a power of two. To work with other frame sizes, use the Pad block to pad or truncate the frame size to a power of two length.

The output is an M-by-N matrix whose lth column contains the length-M IDCT of the corresponding input column.

$$y(m,l) = \sum_{k=1}^{M} w(k)u(k,l)\cos\frac{\pi(2m-1)(k-1)}{2M}, \qquad m = 1,\ldots,M$$

where

$$w(k) = \begin{cases} \dfrac{1}{\sqrt{M}}, & k = 1 \\ \sqrt{\dfrac{2}{M}}, & 2 \le k \le M \end{cases}$$

The output is always frame based, and the output sample rate and data type (real/complex) are the same as those of the input.

For convenience, length-M 1-D vector inputs and sample-based length-M row vector inputs are processed as single channels (that is, as M-by-1 column vectors), and the output has the same dimension as the input.

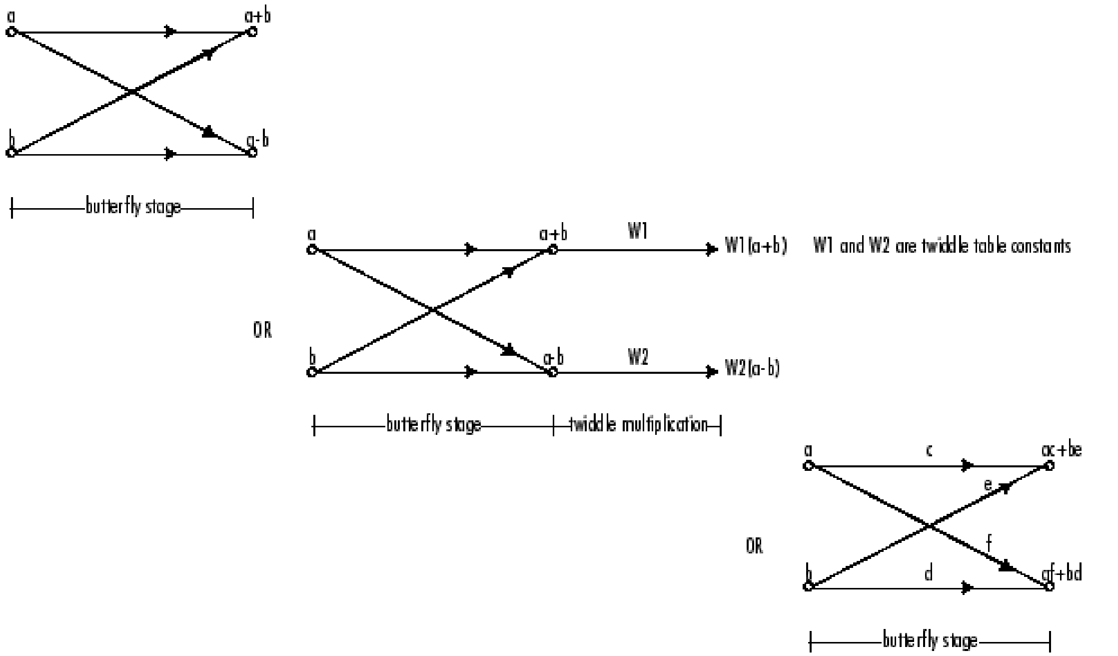
The **Sine and cosine computation** parameter determines how the block computes the necessary sine and cosine values. This parameter has two settings, each with its advantages and disadvantages, as described in the following table.

Sine and Cosine Computation Parameter Setting	Sine and Cosine Computation Method	Effect on Block Performance
`Table lookup`	The block computes and stores the trigonometric values before the simulation starts, and retrieves them during the simulation. When you generate code from the block, the processor running the generated code stores the trigonometric values computed by the block in a speed-optimized table, and retrieves the values during code execution.	The block usually runs much more quickly, but requires extra memory for storing the precomputed trigonometric values.
`Trigonometric fcn`	The block computes sine and cosine values during the simulation. When you generate code from the block, the processor running the generated code computes the sine and cosine values while the code runs.	The block usually runs more slowly, but does not need extra data memory. For code generation, the block requires a support library to emulate the trigonometric functions, increasing the size of the generated code.

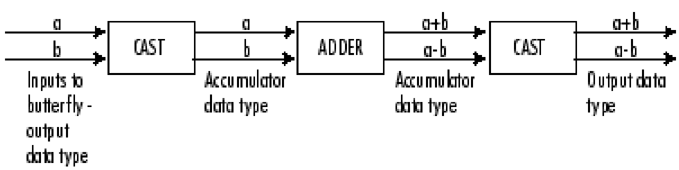
Fixed-Point Data Types

The diagrams below show the data types used within the IDCT block for fixed-point signals. You can set the sine table, accumulator, product output, and output data types displayed in the diagrams in the IDCT block dialog as discussed in "Dialog Box" on page 2-601.

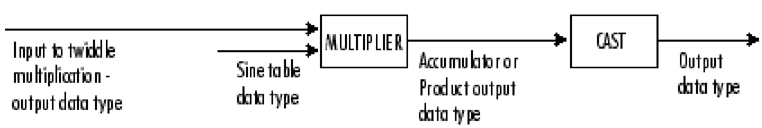
Inputs to the IDCT block are first cast to the output data type and stored in the output buffer. Each butterfly stage processes signals in the accumulator data type, with the final output of the butterfly being cast back into the output data type.



Butterfly Stage Data Types



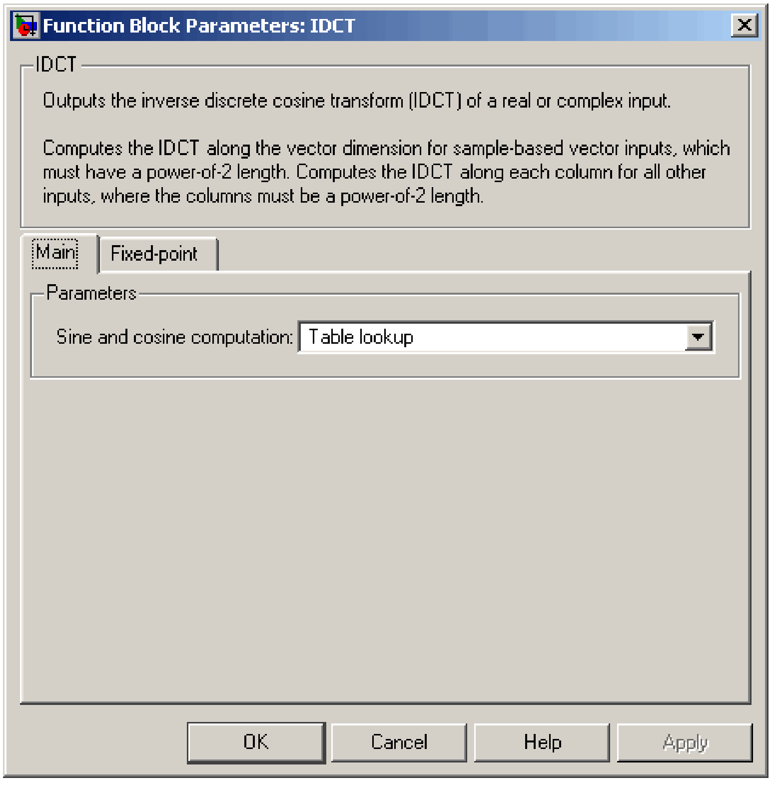
Twiddle Multiplication Data Types



The output of the multiplier is in the product output data type when at least one of the inputs to the multiplier is real. When both of the inputs to the multiplier are complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see "Multiplication Data Types".

Dialog Box

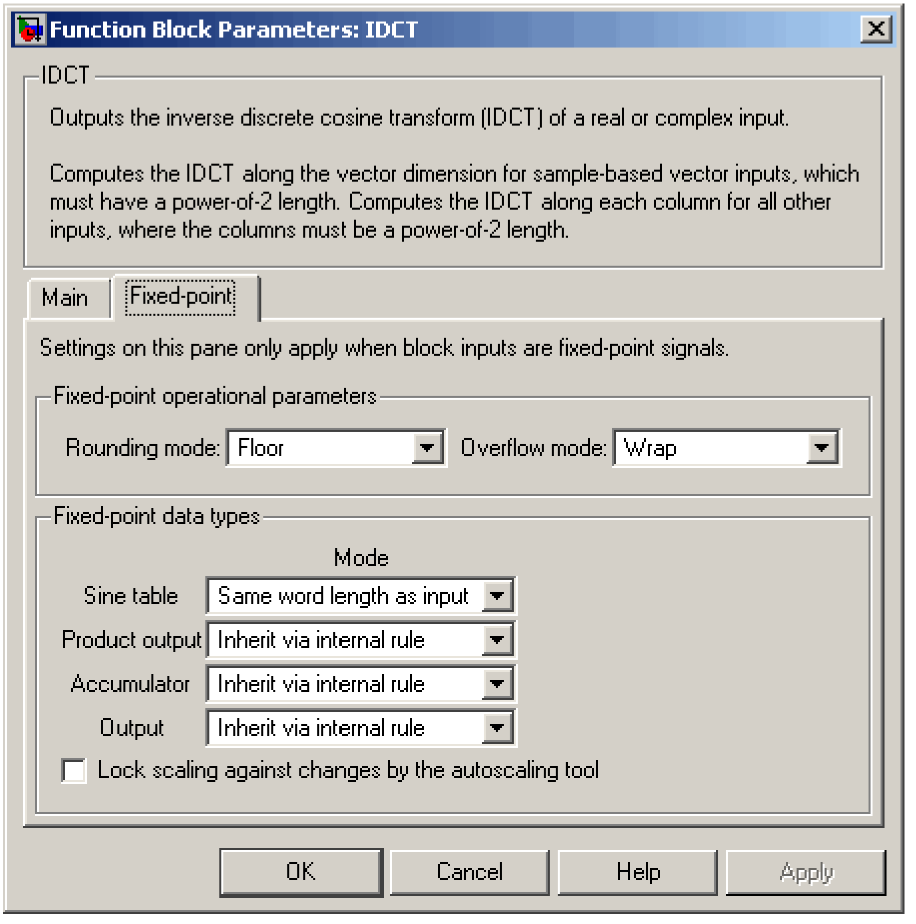
The **Main** pane of the IDCT block dialog appears as follows.



Sine and cosine computation

Sets the block to compute sines and cosines by either looking up sine and cosine values in a speed-optimized table (`Table lookup`), or by making sine and cosine function calls (`Trigonometric fcn`). See the table above.

The **Fixed-point** pane of the IDCT block dialog appears as follows.



Rounding mode

Select the rounding mode for fixed-point operations. The sine table values do not obey this parameter; they always round to `Nearest`.

Overflow mode

Select the overflow mode for fixed-point operations.

Sine table

Choose how you specify the word length of the values of the sine table. The fraction length of the sine table values is always equal to the word length minus one:

- When you select `Same word length as input`, the word length of the sine table values match that of the input to the block.
- When you select `Specify word length`, you can enter the word length of the sine table values, in bits.

The sine table values do not obey the **Rounding mode** and **Overflow mode** parameters; they are always saturated and rounded to `Nearest`.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-599 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-599 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Inherit via internal rule`, the output word length and fraction length are calculated automatically. The internal rule first calculates an ideal output word length and fraction length using the following equations:

$$WL_{ideal\ output} = WL_{input} + floor(\log_2(DCT\ length - 1)) + 1$$

$$FL_{ideal\ output} = FL_{input}$$

Using these ideal results, the internal rule then selects word lengths and fraction lengths that are appropriate for your hardware. For more information, see "Inherit via Internal Rule".

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

DCT	Signal Processing Blockset
IFFT	Signal Processing Blockset
`idct`	Signal Processing Toolbox

Identity Matrix

Purpose

Generate matrix with ones on main diagonal and zeros elsewhere

Library

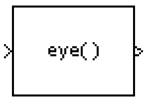
- Signal Processing Sources

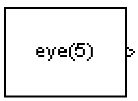
  `dspsrcs4`

- Math Functions / Matrices and Linear Algebra / Matrix Operations

  `dspmtrx3`

Description





The Identity Matrix block generates a rectangular matrix with ones on the main diagonal and zeros elsewhere.

When you select the **Inherit output port attributes from input port** check box, the input port is enabled, and an M-by-N matrix input generates a sample-based M-by-N matrix output with the same sample period. The values in the input matrix are ignored.

```
y = eye([M N])      % Equivalent MATLAB code
```

When you do not select the **Inherit output port attributes from input port** check box, the input port is disabled, and the dimensions of the output matrix are determined by the **Matrix size** parameter. A scalar value, M, specifies an M-by-M identity matrix, while a two-element vector, [M N], specifies an M-by-N unit-diagonal matrix. The output is sample based, and has the sample period specified by the **Sample time** parameter.

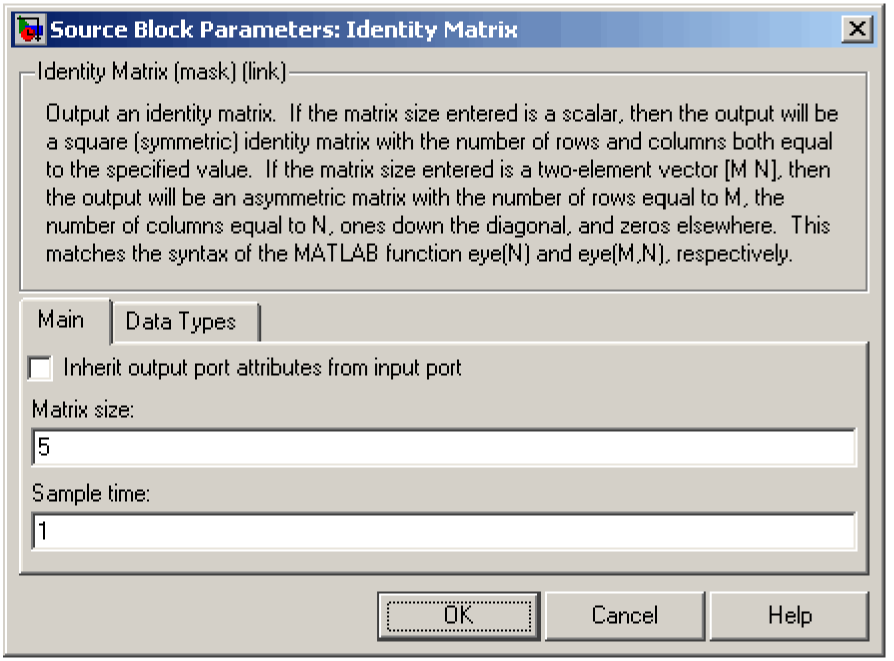
Examples

Set **Matrix size** to `[3 6]` to generate the 3-by-6 unit-diagonal matrix below.

$$\begin{bmatrix} 1 & 0 & 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 & 0 & 0 \\ 0 & 0 & 1 & 0 & 0 & 0 \end{bmatrix}$$

Dialog Box

The **Main** pane of the Identity Matrix block dialog appears as follows.



Inherit output port attributes from input port

Enables the input port when selected. In this mode, the output inherits its dimensions, sample period, and data type from the input. The output is always real.

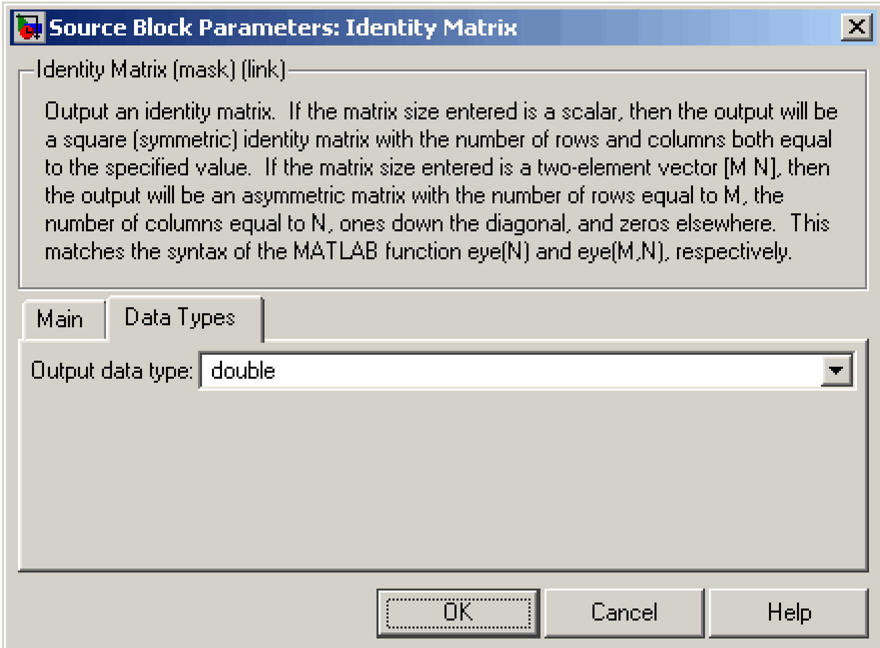
Matrix size

The number of rows and columns in the output matrix: a scalar *M* for a square *M*-by-*M* output, or a vector [*M N*] for an *M*-by-*N* output. This parameter is disabled when you select **Inherit input port attributes from input port**.

Sample time

The discrete sample period of the output. This parameter is disabled when you select **Inherit input port attributes from input port**.

The **Data Types** pane of the Identity Matrix block dialog appears as follows.



Output data type

Specify the output data type in one of the following ways:

- Choose one of the built-in data types from the list.
- Choose `Fixed-point` to specify the output data type and scaling in the **Signed**, **Word length**, **Set fraction length in output to,** and **Fraction length** parameters.
- Choose `User-defined` to specify the output data type and scaling in the **User-defined data type**, **Set fraction length in output to,** and **Fraction length** parameters.

- Choose `Inherit via back propagation` to set the output data type and scaling to match the following block

Signed
Select to output a signed fixed-point signal. Otherwise, the signal is unsigned. This parameter is visible only when you select `Fixed-point` for the **Output data type** parameter.

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

User-defined data type
Specify any built-in or fixed-point data type. You can specify fixed-point data types using the following Simulink® Fixed Point™ functions: `sfix`, `ufix`, `sint`, `uint`, `sfrac`, and `ufrac`. This parameter is visible only when you select `User-defined` for the **Output data type** parameter.

Set fraction length in output to
Specify the scaling of the fixed-point output by either of the following two methods:

- Choose `Best precision` to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose `User-defined` to specify the output scaling in the **Fraction length** parameter.

This parameter is visible only when you select `Fixed-point` for the **Output data type** parameter, or when you select `User-defined` and the specified output data type is a fixed-point data type.

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

the **Output data type** parameter and `User-defined` for the **Set fraction length in output to** parameter.

Supported Data Types

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

See Also

Constant Diagonal Matrix	Signal Processing Blockset
Constant	Simulink
`eye`	MATLAB

Purpose

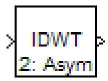
Compute inverse discrete wavelet transform (IDWT) of input

Library

Transforms

`dspxfrm3`

Description



Note The IDWT block is the same as the Dyadic Synthesis Filter Bank block in the Multirate Filters library, but with different default settings. See the Dyadic Synthesis Filter Bank for more information on how to use the block.

The IDWT block computes the inverse discrete wavelet transform (IDWT) of the input subbands. By default, the block accepts a single sample-based vector or matrix of concatenated subbands. The output is frame based, and has the same dimensions as the input. Each column of the output is the IDWT of the corresponding input column.

You must install the Wavelet Toolbox™ product for the block to automatically design wavelet-based filters to compute the IDWT. Otherwise, you must specify your own lowpass and highpass FIR filters by setting the **Filter** parameter to `User defined`.

For detailed information about how to use this block, see Dyadic Synthesis Filter Bank.

Examples

See the examples in the Dyadic Synthesis Filter Bank block reference.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Dyadic Synthesis Filter Bank	Signal Processing Blockset
DWT	Signal Processing Blockset

Purpose

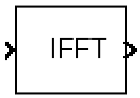
Compute inverse fast Fourier transform (IFFT) of input

Library

Transforms

`dspxfrm3`

Description



The IFFT block computes the inverse fast Fourier transform (IFFT) of each channel of a P-by-N or length-P input, u. When the **Inherit FFT length from input dimensions** check box is selected, the input length P must be an integer power of two, and the FFT length M is equal to P. When the check box is not selected, P can be any length, and the value of the **FFT length** parameter must be a positive integer power of two. For user-specified FFT lengths, when M is not equal to P, zero padding or modulo-M data wrapping happens before the IFFT operation, as per Orfanidis [1].

```
y = ifft(u,M)  % P ≤ M

y(:,l) = ifft(datawrap(u(:,l),M))  % P > M; l = 1,...,N
```

To get zero padding or truncation rather than zero padding or wrapping, use a Pad block before the FFT block in your model to obtain a power-of-two input length.

The kth entry of the lth output channel, $y(k, l)$, is equal to the kth point of the M-point inverse discrete Fourier transform (IDFT) of the lth input channel:

$$y(k,l) = \frac{1}{M}\sum_{p=1}^{P} u(p,l)e^{j2\pi(p-1)(k-1)/M} \qquad k = 1,\ldots,M$$

This block supports real and complex floating-point and fixed-point inputs.

Input and Output Characteristics

The following table describes valid inputs to the IFFT block, their corresponding outputs, and the dimension along which the block computes the IDFT.

- Valid inputs to the IFFT block. They can be real- or complex-valued, and they can be in linear or bit-reversed order.
- The dimension along which the block computes the IDFT.
- The corresponding block output characteristics. The output port rate must equal the input port rate.

<table>
<tr><th>Valid Block Inputs</th><th>Dimension Along Which Block Computes IDFT</th><th>Corresponding Block Output Characteristics</th></tr>
<tr><td>Frame-based P-by-N matrix</td><td>Column</td><td rowspan="4">The following output characteristics apply to all valid block inputs:
Frame based
Complex valued. If your input is conjugate symmetric and you select the Input is conjugate symmetric check box, the output is real valued.
Same dimension as input (for 1-D inputs, output is a length-M column).
Each output column (each row for sample-based row inputs) contains the M-point IDFT of the corresponding input channel in linear order. If you select the Skip scaling check box, the block computes a scaled version of the IDFT that does not include the multiplication factor of 1/M.</td></tr>
<tr><td>Sample-based P-by-N matrix</td><td>Column</td></tr>
<tr><td>Sample-based 1-by-P row vector</td><td>Row</td></tr>
<tr><td>1-D length-P vector</td><td>Vector</td></tr>
</table>

Selecting the Twiddle Factor Computation Method

The **Twiddle factor computation** parameter determines how the block computes the necessary sine and cosine terms to calculate the term $e^{j2\pi(p-1)(k-1)/M}$, shown in the first equation of this block reference page. This parameter has two settings, each with its advantages and disadvantages, as described in the following table. Only `Table lookup` mode is supported for fixed-point signals.

Twiddle Factor Computation Parameter Setting	Sine and Cosine Computation Method	Effect on Block Performance
`Table lookup`	The block computes and stores the trigonometric values before the simulation starts, and retrieves them during the simulation. When you generate code from the block, the processor running the generated code stores the trigonometric values computed by the block, and retrieves the values during code execution.	The block usually runs much more quickly, but requires extra memory for storing the precomputed trigonometric values. You can optimize the table for memory consumption or speed, as described in "Optimizing the Table of Trigonometric Values" on page 2-615.
`Trigonometric fcn`	The block computes sine and cosine values during the simulation. When you generate code from the block, the processor running the generated code computes the sine and cosine values while the code runs.	The block usually runs more slowly, but does not need extra data memory. For code generation, the block requires a support library to emulate the trigonometric functions, increasing the size of the generated code.

Optimizing the Table of Trigonometric Values

When you set the **Twiddle factor computation** parameter to `Table lookup`, you also need to set the **Optimize table for** parameter. This parameter optimizes the table of trigonometric values for speed or memory by varying the number of table entries as summarized in the following table.

Optimize Table for Parameter Setting	Number of Table Entries for N-Point IFFT	Memory Required for Single-Precision 512-Point IFFT
`Speed`	$3N/4$ — floating point N — fixed point	$\left(\frac{3\times 512}{4}\text{ table entries}\right)\times\left(4\frac{\text{bytes}}{\text{table entry}}\right)=1536\text{ bytes}$
`Memory`	$N/4$ — floating point Not supported for fixed point	$\left(\frac{512}{4}\text{ table entries}\right)\times\left(4\frac{\text{bytes}}{\text{table entry}}\right)=512\text{ bytes}$

Input Order

Select or clear the **Input is in bit-reversed order** check box to designate the ordering of the column elements of the block input. If you select the **Input is in bit-reversed order** check box, the block assumes the input is in bit-reversed order. If you clear the **Input is in bit-reversed order** check box, the block assumes the input is in linear order. For more information on ordering of the output, see “Linear and Bit-Reversed Output Order”.

Conjugate Symmetric Input

The FFT block yields conjugate symmetric output when its input is real valued. Taking the IFFT of a conjugate symmetric input matrix produces real-valued output. Therefore, if the input to the block is conjugate symmetric and you select the **Input is conjugate symmetric** check box, the block produces real-valued outputs. Selecting this check box optimizes the block’s computation method.

If the IFFT block input is conjugate symmetric and you do not select the **Input is conjugate symmetric** check box, the IFFT block outputs a complex-valued signal with small imaginary parts. The block output is invalid if you select this check box and the input is not conjugate symmetric.

Scaled Output

If you select the **Skip scaling** check box, the block's output is not scaled. If you clear the **Skip scaling** check box and your signal is a floating-point signal, the block computes a scaled version of the IDFT, $M \cdot y(k,l)$, which is defined by the following equation:

$$M \cdot y(k,l) = \sum_{p=1}^{P} u(p,l) e^{j2\pi(p-1)(k-1)/M} \qquad k = 1, \ldots, M$$

If you clear the **Skip scaling** check box and your signal is a fixed-point signal, the output of each butterfly of the IFFT is divided by two.

Algorithms Used for IFFT Computation

Depending on whether the block input is in bit-reversed order and/or conjugate symmetric, the block uses one or more of the following algorithms as summarized in the subsequent table:

- Butterfly operation
- Double-signal algorithm
- Half-length algorithm
- Radix-2 decimation-in-time (DIT) algorithm

Input Complexity	Other Parameter Settings	Algorithms Used for IFFT Computation
Real or complex	☐ Input is in bit-reversed order ☐ Input is conjugate symmetric	Butterfly operation and radix-2 DIT

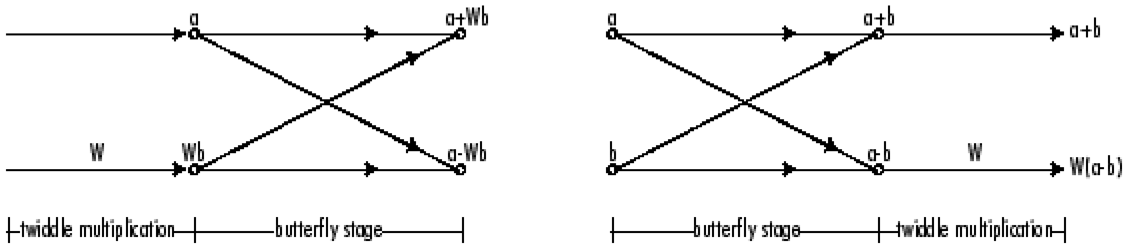
Input Complexity	Other Parameter Settings	Algorithms Used for IFFT Computation
Real or complex	☑ Input is in bit-reversed order ☐ Input is conjugate symmetric	Radix-2 DIF
Real or complex	☐ Input is in bit-reversed order ☑ Input is conjugate symmetric	Butterfly operation and radix-2 DIT in conjunction with the half-length and double-signal algorithms
Real or complex	☑ Input is in bit-reversed order ☑ Input is conjugate symmetric	Radix-2 DIF in conjunction with the half-length and double-signal algorithms

Fixed-Point Data Types

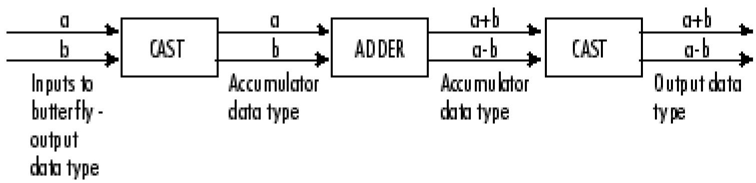
The following diagrams show the data types used within the IFFT block for fixed-point signals. You can set the sine table, accumulator, product output, and output data types displayed in the diagrams in the IFFT block dialog, as discussed in “Dialog Box” on page 2-620.

Inputs to the IFFT block are first cast to the output data type and stored in the output buffer. Each butterfly stage then processes signals in the accumulator data type, with the final output of the butterfly being cast back into the output data type. A twiddle factor is multiplied in before each butterfly stage in a decimation-in-time IFFT, and after each butterfly stage in a decimation-in-frequency IFFT.

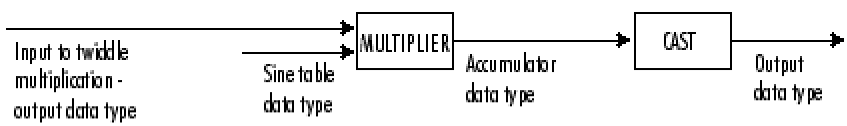
Decimation-in-time IFFT **Decimation-in-frequency IFFT**



Butterfly stage data types



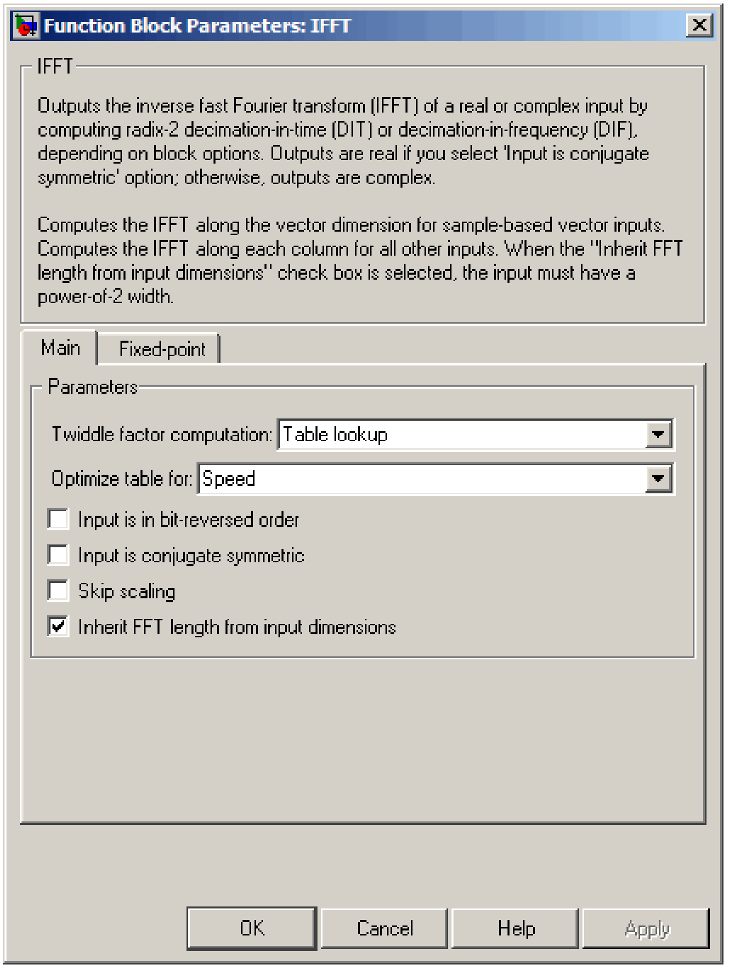
Twiddle multiplication data types



The output of the multiplier is in the accumulator data type since both of the inputs to the multiplier are complex. For details on the complex multiplication performed, see "Multiplication Data Types".

Dialog Box

The **Main** pane of the IFFT block dialog appears as follows.



Twiddle factor computation

Specify the computation method of the term $e^{j2\pi(p-1)(l-1)/M}$ shown in the first equation of this block reference page.

In `Table lookup` mode, the block computes and stores the sine and cosine values before the simulation starts.

In `Trigonometric fcn` mode, the block computes the sine and cosine values during the simulation. See “Selecting the Twiddle Factor Computation Method” on page 2-615.

This parameter must be set to `Table lookup` for fixed-point signals.

Optimize table for

Select the optimization of the table of sine and cosine values for `Speed` or `Memory`. This parameter is only available when the **Twiddle factor computation** parameter is set to `Table lookup`. See “Optimizing the Table of Trigonometric Values” on page 2-615.

This parameter must be set to `Speed` for fixed-point signals.

Input is in bit-reversed order

Designate the order of the input channel elements. Select when the input is in bit-reversed order, and clear when the input is in linear order. The block yields invalid outputs when you do not set this parameter correctly. See “Input Order” on page 2-616.

This parameter cannot be selected when the **Inherit FFT length from input dimensions** parameter is cleared and you are specifying the FFT length using the **FFT length** parameter.

Input is conjugate symmetric

Select when the input to the block is conjugate symmetric and you want real-valued outputs. The block output is invalid when you set this parameter when the input is not conjugate symmetric.

This parameter cannot be selected when the **Inherit FFT length from input dimensions** parameter is cleared and you are specifying the FFT length using the **FFT length** parameter.

Skip scaling

When you select this check box, no scaling occurs. When this parameter is unselected, scaling does occur:

- For floating-point signals, rather than computing the IDFT, the block computes a scaled version of the IDFT. This scaled version of the IDFT does not include the multiplication factor of $1/M$.
- For fixed-point signals, the output of each butterfly of the IFFT is divided by two.

Inherit FFT length from input dimensions

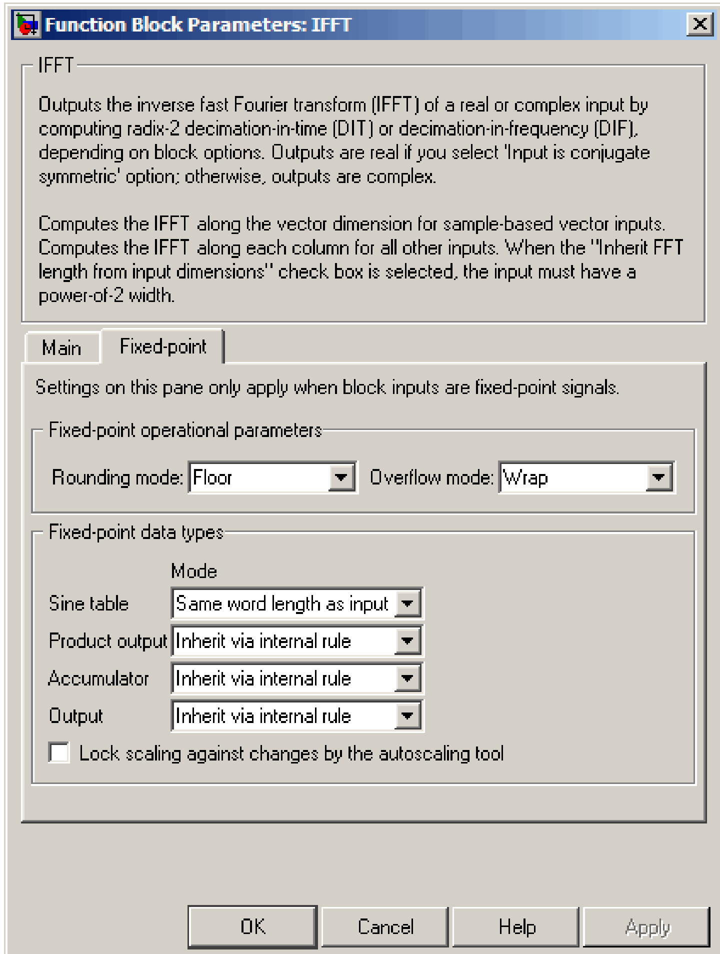
Select to inherit the FFT length from the input dimensions. When this parameter is selected, the input length P must be a power of two. When this parameter is not selected, the **FFT length** parameter is available.

This parameter cannot be cleared when either the **Input is in bit-reversed order** or the **Input is conjugate symmetric** parameter is selected.

FFT length

Specify a power-of-two FFT length. This parameter is only available when the **Inherit FFT length from input dimensions** parameter is not selected.

The **Fixed-point** pane of the IFFT block dialog appears as follows.



Rounding mode

Select the rounding mode for fixed-point operations. The sine table values do not obey this parameter; they always round to `Nearest`.

Overflow mode

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

Sine table

Choose how you specify the word length of the values of the sine table. The fraction length of the sine table values is always equal to the word length minus one:

- When you select `Same word length as input`, the word length of the sine table values match that of the input to the block.
- When you select `Specify word length`, you can enter the word length of the sine table values, in bits.

The sine table values do not obey the **Rounding mode** and **Overflow mode** parameters; they are always saturated and rounded to `Nearest`.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-618 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-618 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Inherit via internal rule`, the output word length and fraction length are calculated automatically. The internal rule first calculates an ideal output word length and fraction length using the following equations:

$$WL_{ideal\ output} = WL_{input} + floor(\log_2(FFT\ length - 1)) + 1$$

$$FL_{ideal\ output} = FL_{input}$$

Using these ideal results, the internal rule then selects word lengths and fraction lengths that are appropriate for your hardware. For more information, see "Inherit via Internal Rule".

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

References

[1] Orfanidis, S. J. *Introduction to Signal Processing*. Upper Saddle River, NJ: Prentice Hall, 1996, p. 497.

[2] Proakis, John G. and Dimitris G. Manolakis. *Digital Signal Processing*, 3rd ed. Upper Saddle River, NJ: Prentice Hall, 1996.

Supported Data Types

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

See Also

FFT	Signal Processing Blockset
IDCT	Signal Processing Blockset
Pad	Signal Processing Blockset
`bitrevorder`	Signal Processing Toolbox
`fft`	Signal Processing Toolbox
`ifft`	Signal Processing Toolbox

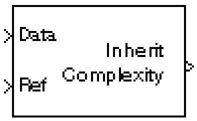
Inherit Complexity

Purpose Change complexity of input to match reference signal

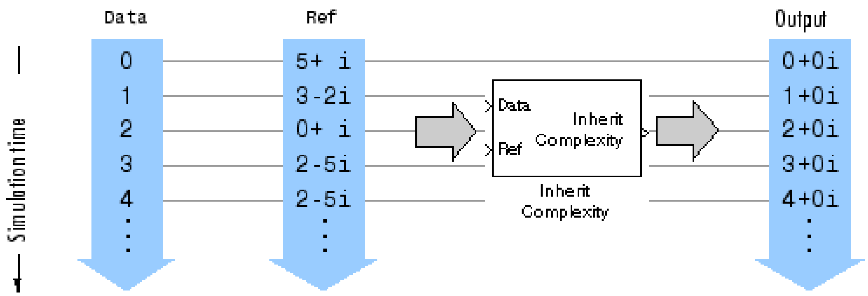
Library Signal Management / Signal Attributes

`dspsigattribs`

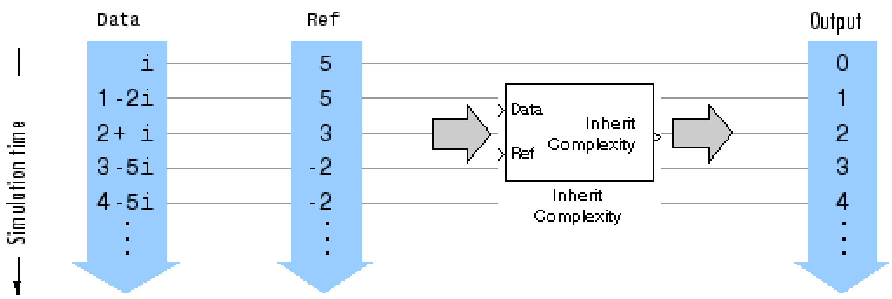
Description



The Inherit Complexity block alters the input data at the Data port to match the complexity of the reference input at the Ref port. When the Data input is real, and the Ref input is complex, the block appends a zero-valued imaginary component, `0i`, to each element of the Data input.

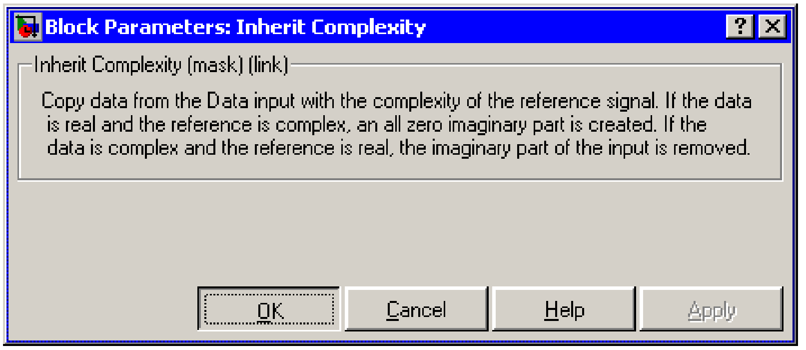


When the `Data` input is complex, and the `Ref` input is real, the block outputs the real component of the `Data` input.



When both the `Data` input and `Ref` input are real, or when both the `Data` input and `Ref` input are complex, the block propagates the `Data` input with no change.

Dialog Box



Supported Data Types

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

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

See Also

Check Signal Attributes	Signal Processing Blockset
Complex to Magnitude-Angle	Simulink
Complex to Real-Imag	Simulink
Magnitude-Angle to Complex	Simulink
Real-Imag to Complex	Simulink

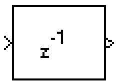
Purpose

Delay input by integer number of sample periods

Library

`dspobslib`

Description



Note The Integer Delay block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Delay block.

The Integer Delay block delays a discrete-time input by the number of sample intervals specified in the **Delay** parameter. Noninteger delay values are rounded to the nearest integer, and negative delays are clipped at 0.

Sample-Based Operation

When the input is a sample-based M-by-N matrix, the block treats each of the $M*N$ matrix elements as an independent channel. The **Delay** parameter, v, can be an M-by-N matrix of positive integers that specifies the number of sample intervals to delay each channel of the input, or a scalar integer by which to equally delay all channels.

For example, when the input is M-by-1 and v is the matrix `[v(1) v(2) ... v(M)]'`, the first channel is delayed by `v(1)` sample intervals, the second channel is delayed by `v(2)` sample intervals, and so on. Note that when a channel is delayed for Δ sample-time units, the output sample at time t is the input sample at time $t - \Delta$. When $t - \Delta$ is negative, then the output is the corresponding value specified by the **Initial conditions** parameter.

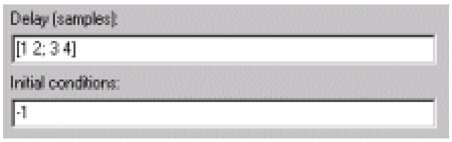
A 1-D vector of length M is treated as an M-by-1 matrix, and the output is 1-D.

The **Initial conditions** parameter specifies the output of the block during the initial delay in each channel. The initial delay for a particular channel is the time elapsed from the start of the simulation until the first input in that channel is propagated to the output. Both fixed and time-varying initial conditions can be specified in a variety of ways to suit the dimensions of the input.

Fixed Initial Conditions

A fixed initial condition in sample-based mode can be specified as one of the following:

- Scalar value to be repeated at each sample time of the initial delay (for every channel). For a 2-by-2 input with the parameter settings below,

  the block generates the following sequence of matrices at the start of the simulation,

  $$\begin{bmatrix} -1 & -1 \\ -1 & -1 \end{bmatrix}, \begin{bmatrix} u_{11}^1 & -1 \\ -1 & -1 \end{bmatrix}, \begin{bmatrix} u_{11}^2 & u_{12}^1 \\ -1 & -1 \end{bmatrix}, \begin{bmatrix} u_{11}^3 & u_{12}^2 \\ u_{21}^1 & -1 \end{bmatrix}, \begin{bmatrix} u_{11}^4 & u_{12}^3 \\ u_{21}^2 & u_{22}^1 \end{bmatrix}, \ldots$$

  where u_{ij}^k is the i,jth element of the kth matrix in the input sequence.
- Array of size M-by-N-by-d. In this case, you can set different fixed initial conditions for each element of a sample-based input. This setting is explained further in the Array bullet in “Time-Varying Initial Conditions” on page 2-632.

Initial conditions cannot be specified by full matrices.

Time-Varying Initial Conditions

A time-varying initial condition in sample-based mode can be specified in one of the following ways:

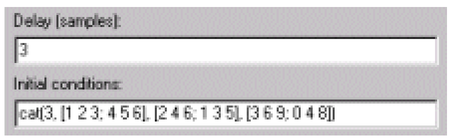
- Vector of length d, where d is the maximum value specified for any channel in the **Delay** parameter. The vector can be a L-by-d, 1-by-d, or 1-by-1-by-d. The d elements of the vector are output in sequence, one at each sample time of the initial delay.

  For a scalar input and the parameters shown below,

  the block outputs the sequence `-1, -1, -1, 0, 1,...` at the start of the simulation.

- Array of dimension M-by-N-by-d, where d is the value specified for the **Delay** parameter (the maximum value when the **Delay** is a vector) and M and N are the number of rows and columns, respectively, in the input matrix. The d pages of the array are output in sequence, one at each sample time of the initial delay. For a 2-by-3 input, and the parameters below,

  the block outputs the matrix sequence

  $$\begin{bmatrix} 1 & 2 & 3 \\ 4 & 5 & 6 \end{bmatrix}, \begin{bmatrix} 2 & 4 & 6 \\ 1 & 3 & 5 \end{bmatrix}, \begin{bmatrix} 3 & 6 & 9 \\ 0 & 4 & 8 \end{bmatrix}$$

  at the start of the simulation. Note that setting **Initial conditions** to an array with the same matrix for each entry implements constant initial conditions; a different constant initial condition for each input matrix element (channel).

Initial conditions cannot be specified by full matrices.

Frame-Based Operation

When the input is a frame-based M-by-N matrix, the block treats each of the N columns as an independent channel, and delays each channel as specified by the **Delay** parameter.

For frame-based inputs, the **Delay** parameter can be a scalar integer by which to equally delay all channels. It can also be a 1-by-N row vector, each element of which serves as the delay for the corresponding channel of the N-channel input. Likewise, it can also be an M-by-1 column vector, each element of which serves as the delay for one of the corresponding M samples for each channel. The **Delay** parameter can be an M-by-N matrix of positive integers as well; in this case, each element of each channel is delayed by the corresponding element in the delay matrix. For instance, if the fifth element of the third column of the delay matrix was 3, then the fifth element of the third channel of the input matrix is always delayed by three sample-time units.

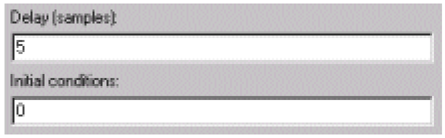
When a channel is delayed for Δ sample-time units, the output sample at time t is the input sample at time $t - \Delta$. When $t - \Delta$ is negative, then the output is the corresponding value specified in the **Initial conditions** parameter.

The **Initial conditions** parameter specifies the output during the initial delay. Both fixed and time-varying initial conditions can be specified. The initial delay for a particular channel is the time elapsed from the start of the simulation until the first input in that channel is propagated to the output.

Fixed Initial Conditions

The settings shown below specify fixed initial conditions. The value entered in the **Initial conditions** parameter is repeated at the output for each sample time of the initial delay. A fixed initial condition in frame-based mode can be one of the following:

- Scalar value to be repeated for all channels of the output at each sample time of the initial delay. For a general M-by-N input with the parameter settings below,

  the first five samples in each of the N channels are zero. Notice that when the frame size is larger than the delay, all of these zeros are all included in the first output from the block.

- Array of size 1-by-N-by-D. In this case, you can also specify different fixed initial conditions for each channel. See "Time-Varying Initial Conditions" on page 2-635 for details.

Initial conditions cannot be specified by full matrices.

Time-Varying Initial Conditions

The following settings specify time-varying initial conditions. For time-varying initial conditions, the values specified in the **Initial conditions** parameter are output in sequence during the initial delay. A time-varying initial condition in frame-based mode can be specified in the following ways:

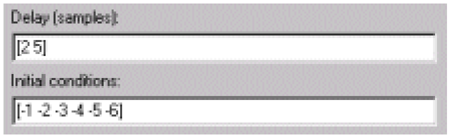
- Vector of length D, where each of the N channels have the same initial conditions sequence specified in the vector. D is defined as follows:
  - When an element of the delay entry is less than the frame size,

    $D = d + 1$

    where d is the maximum delay.
  - When the all elements of the delay entry are greater than the input frame size,

    $D = d + \texttt{input frame size} - 1$

Only the first d entries of the initial condition vector are used; the rest of the values are ignored, but you must include them nonetheless. For a two-channel ramp input `[1:100; 1:100]'` with a frame size of 4 and the parameter settings below,

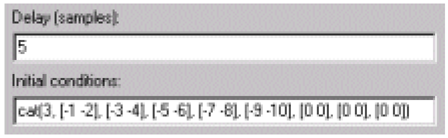


the block outputs the following sequence of frames at the start of the simulation.

$$\begin{bmatrix} -4 & -1 \\ -5 & -2 \\ 1 & -3 \\ 2 & -4 \end{bmatrix}, \begin{bmatrix} 3 & -5 \\ 4 & 1 \\ 5 & 2 \\ 6 & 3 \end{bmatrix}, \begin{bmatrix} 7 & 4 \\ 8 & 5 \\ 9 & 6 \\ 10 & 7 \end{bmatrix}, \ldots$$

Note that since one of the delays, 2, is less than the frame size of the input, 4, the length of the **Initial conditions** vector is the sum of the maximum delay and 1 (5+1), which is 6. The first five entries of the initial conditions vector are used by the channel with the maximum delay, and the rest of the entries are ignored. Since the first channel is delayed for less than the maximum delay (2 sample time units), it only makes use of two of the initial condition entries.

- Array of size 1-by-N-by-D, where D is defined in “Time-Varying Initial Conditions” on page 2-635. In this case, the kth entry of each 1-by-N entry in the array corresponds to an initial condition for the kth channel of the input matrix. Thus, a 1-by-N-by-D initial conditions input allows you to specify different initial conditions for each channel. For instance, for a two-channel ramp input `[1:100; 1:100]'` with a frame size of 4 and the parameter settings below,



the block outputs the following sequence of frames at the start of the simulation.

$$\begin{bmatrix} -1 & -2 \\ -3 & -4 \\ -5 & -6 \\ -7 & -8 \end{bmatrix}, \begin{bmatrix} -9 & -10 \\ 1 & 1 \\ 2 & 2 \\ 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 \\ 5 & 5 \\ 6 & 6 \\ 7 & 7 \end{bmatrix}, \ldots$$

Note that the channels have distinct time varying initial conditions; the initial conditions for channel 1 correspond to the first entry of each length-2 row vector in the initial conditions array, and the initial conditions for channel 2 correspond to the second entry of each row vector in the initial conditions array. Only the first five entries in the initial conditions array are used; the rest are ignored.

The 1-by-N-by-D array entry can also specify different fixed initial conditions for every channel; in this case, every 1-by-N entry in the array would be identical, so that the initial conditions for each column are fixed over time.

Initial conditions cannot be specified by full matrices.

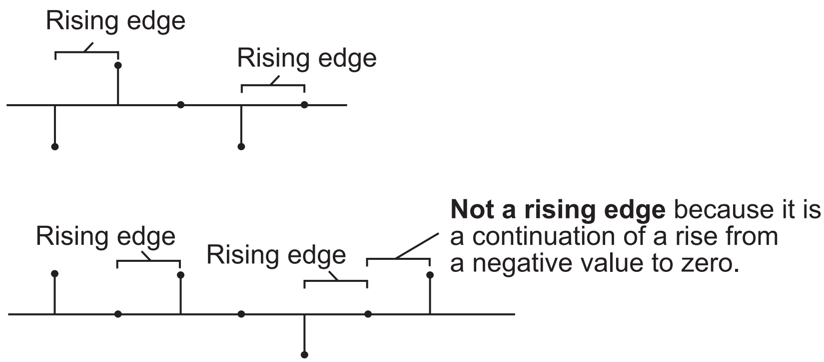
Resetting the Delay

The block resets the delay whenever it detects a reset event at the optional Rst port. The reset sample time must be a positive integer multiple of the input sample time.

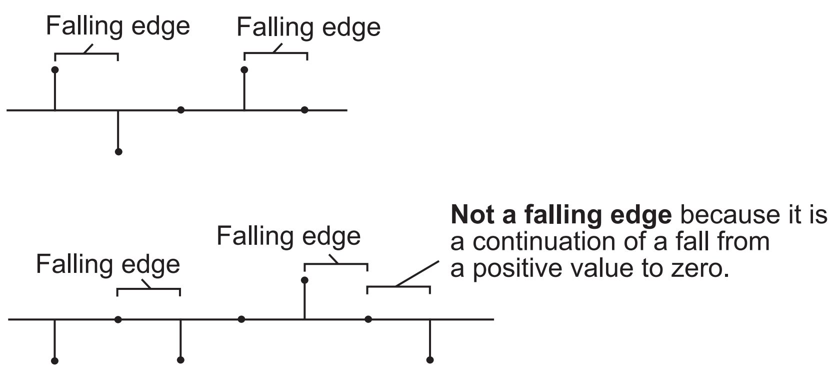
You specify the reset event in the **Reset port** parameter:

- `None` disables the Rst port.
- `Rising edge` — Triggers a reset operation when the Rst input does one of the following:

- Rises from a negative value to a positive value or zero
- Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)



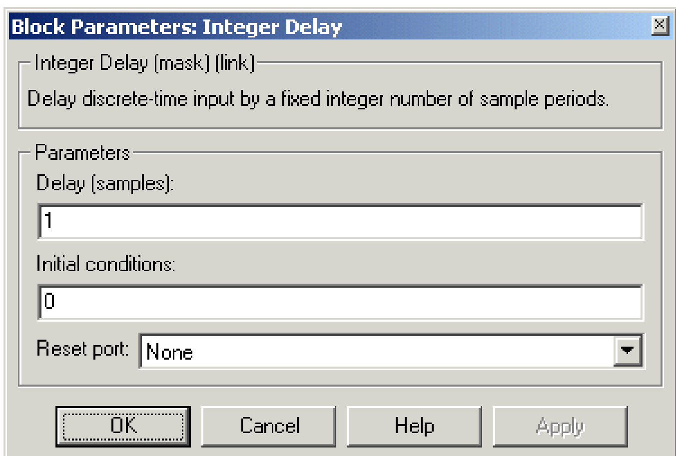
- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described above).
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero.

Note When running simulations in the Simulink® `MultiTasking` mode, sample-based reset signals have a one-sample latency, and frame-based reset signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a reset event, and when it applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Examples

The `dspafxr` demo illustrates an audio reverberation system built around the Integer Delay block.

Dialog Box



Delay

The number of sample periods to delay the input signal.

Initial conditions

The value of the block's output during the initial delay.

Reset port

Determines the reset event that causes the block to reset the delay. For more information, see "Resetting the Delay" on page 2-637.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)
- Boolean — The block accepts Boolean inputs to the Rst port, which is enabled by the **Reset port** parameter.
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

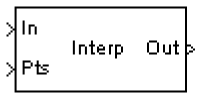
Unit Delay	Simulink
Variable Fractional Delay	Signal Processing Blockset
Variable Integer Delay	Signal Processing Blockset

Purpose Interpolate values of real input samples

Library Signal Operations

`dspsigops`

Description The Interpolation block interpolates each channel of discrete, real, inputs using linear or FIR interpolation. The input can be a sample or frame based vector or matrix. The output is a vector or matrix of the interpolated values, and has the same frame status and frame rate as the input.



Interpolation

You must specify the interpolation points (times at which to interpolate values) in an interpolation vector, I_n. An entry of 1 in I_n refers to the first sample of the input, an entry of 2.5 refers to the sample half-way between the second and third input sample, and so on. I_n must have the same frame status and frame rate as the input, and can be a length-P row or column vector, where P is usually any positive integer.

Usually, the block applies the vector I_n to each column of an input matrix, or to each input vector. You can set the block to either apply the same interpolation vector for all input vectors or matrices (static interpolation points), or use a different interpolation vector for each input vector or matrix (time-varying interpolation points).

For more information, see other sections of this reference page.

Sections of This Reference Page

- “Specifying Static Interpolation Points” on page 2-642
- “Specifying Time-Varying Interpolation Points” on page 2-642
- “How the Block Applies Interpolation Vectors to Inputs” on page 2-642
- “Handling Out-of-Range Interpolation Points” on page 2-645
- “Linear Interpolation Mode” on page 2-646
- “FIR Interpolation Mode” on page 2-647
- “Dialog Box” on page 2-648

- “Supported Data Types” on page 2-650

Specifying Static Interpolation Points

To supply the block with a static interpolation vector (an interpolation vector applied to every input vector or matrix), do the following:

- Set the **Source of interpolation points** parameter to `Specify via dialog`.
- Enter the interpolation vector in the **Interpolation points** parameter. To learn about interpolation vectors, see “How the Block Applies Interpolation Vectors to Inputs” on page 2-642.

Specifying Time-Varying Interpolation Points

To supply the block with time-varying interpolation vectors (where the block uses a different interpolation vector for each input vector or matrix), do the following:

1. Set the **Source of interpolation points** parameter to `Input port`, the Pts port appears on the block.

2. Generate a signal of interpolation vectors with the same frame status and same frame rate as the input signal, and supply it to the Pts port. The block uses the input to this port as the interpolation points. To learn about interpolation vectors, see “How the Block Applies Interpolation Vectors to Inputs” on page 2-642.

How the Block Applies Interpolation Vectors to Inputs

The interpolation vector I_n represents the points in time at which to interpolate values of the input signal. An entry of 1 in I_n refers to the first sample of the input, an entry of 2.5 refers to the sample half-way between the second and third input sample, and so on. In most cases, the vector I_n can be of any length.

Depending on the dimension and frame status of the input and the dimension of I_n, the block usually applies I_n to the input in one of the following ways:

- Applies the vector I_n to each channel of a matrix input, resulting in a matrix output.
- Applies the vector I_n to each input vector (as if the input vector were a single channel), resulting in a vector output with the same orientation as the input (row or column).

The following tables summarize how the block applies the vector I_n to all the possible types of sample- and frame-based inputs, and show the resulting output dimensions. (The block applies both static and time-varying interpolation vectors to the input signal in the same way.)

How Block Applies Interpolation Vectors to Frame-Based Inputs

Frame-Based Input Dimensions	Dimensions of Interpolation Vector I_n (P is a positive integer)	How Block Applies I_n to Input	Frame-Based Output Dimensions
M-by-*N* matrix	*P*-by-1 column	Applies I_n to each input column	*P*-by-*N* matrix
	1-by-*N* row	Applies each column of I_n (each element of I_n) to the corresponding columns of the input	1-by-*N* row
M-by-1 column	*P*-by-1 column	Applies I_n to the input column	*P*-by-1 column
	1-by-*P* row (block treats as a column)	Applies I_n to the input column	*P*-by-1 column

How Block Applies Interpolation Vectors to Frame-Based Inputs (Continued)

Frame-Based Input Dimensions	Dimensions of Interpolation Vector I_n (P is a positive integer)	How Block Applies I_n to Input	Frame-Based Output Dimensions
1-by-N row (not recommended)	P-by-1 column	not applicable	P-by-N matrix where each row is a copy of the input vector
	1-by-P row	not applicable	1-by-N row, a copy of the input vector

How Block Applies Interpolation Vectors to Sample-Based Inputs

Sample-Based Input Dimensions	Dimensions of Interpolation Vector I_n(P is any positive integer)	How Block Applies I_n to Input	Sample-Based Output Dimensions
M-by-N matrix	P-by-1 column	Applies I_n to each input column	P-by-N matrix
	1-by-P row (block treats as a column)	Applies I_n to each input column	P-by-N matrix
M-by-1 column	P-by-1 column	Applies I_n to the input column	P-by-1 column
	1-by-P row (block treats as a column)	Applies I_n to the input column	P-by-1 column

How Block Applies Interpolation Vectors to Sample-Based Inputs (Continued)

Sample-Based Input Dimensions	Dimensions of Interpolation Vector I_n(P is any positive integer)	How Block Applies I_n to Input	Sample-Based Output Dimensions
1-by-N row	P-by-1 column (block treats as a row)	Applies I_n to the input row	1-by-P row
	1-by-P row	Applies I_n to the input row	1-by-P row

Handling Out-of-Range Interpolation Points

The valid range of the values in the interpolation vector I_n is from 1 to the number of samples in each channel of the input. For instance, given a length-5 input vector `D`, all entries of I_n must range from 1 to 5. I_n cannot contain entries such as 7 or -9, since there is no 7th or -9th entry in `D`.

The **Out of range interpolation points** parameter sets how the block handles interpolation points that are not within the valid range, and has the following settings:

- `Clip` — The block replaces any out-of-range values in I_n with the closest value in the valid range (from 1 to the number of input samples), and then proceeds with computations using the clipped version of I_n.
- `Clip and warn` — In addition to `Clip`, the block issues a warning at the MATLAB® command line every time clipping occurs.
- `Error` — When the block encounters an out-of-range value in I_n, the simulation stops and the block issues an error at the MATLAB command line.

Example of Clipping

Suppose the block is set to clip out-of-range interpolation points, and gets the following input vector and interpolation points:

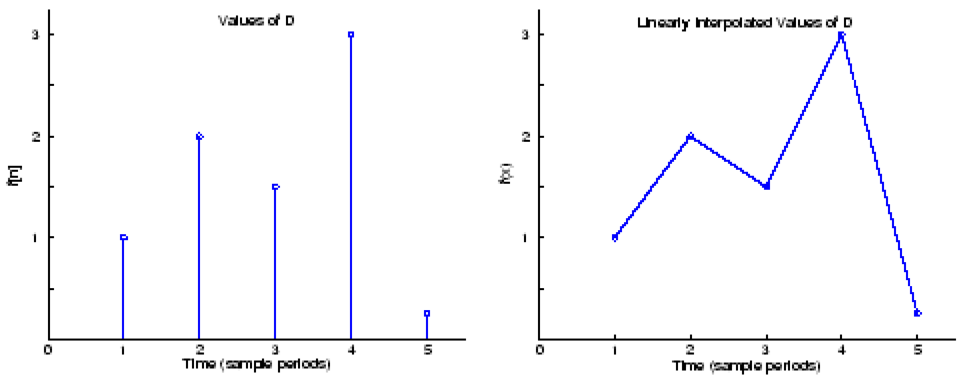
- `D = [11, 22, 33, 44]'`
- I_n `= [10, 2.6, -3]'`

Since *D* has four samples, valid interpolation points range from 1 to 4. The block clips the interpolation point 10 to 4 and the point -3 to 1, resulting in the clipped interpolation vector $I_{nclipped}$ `= [4, 2.6, 1]'`.

Linear Interpolation Mode

When **Interpolation Mode** is set to `Linear`, the block interpolates data values by assuming that the data varies linearly between samples taken at adjacent sample times.

For instance, if the input signal *D* = [1, 2, 1.5, 3, 0.25]', the following left-hand plot shows the samples in *D*, and the right-hand plot shows the linearly interpolated values between the samples in *D*.

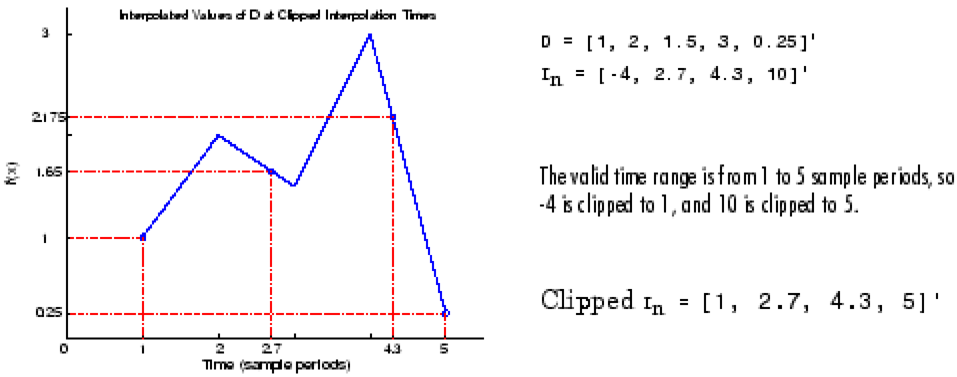


As illustrated below, if the block is in linear interpolation mode and is set to clip out-of-range interpolation points, where

- `D = [1, 2, 1.5, 3, 0.25]'`

- I_n = [-4, 2.7, 4.3, 10]'

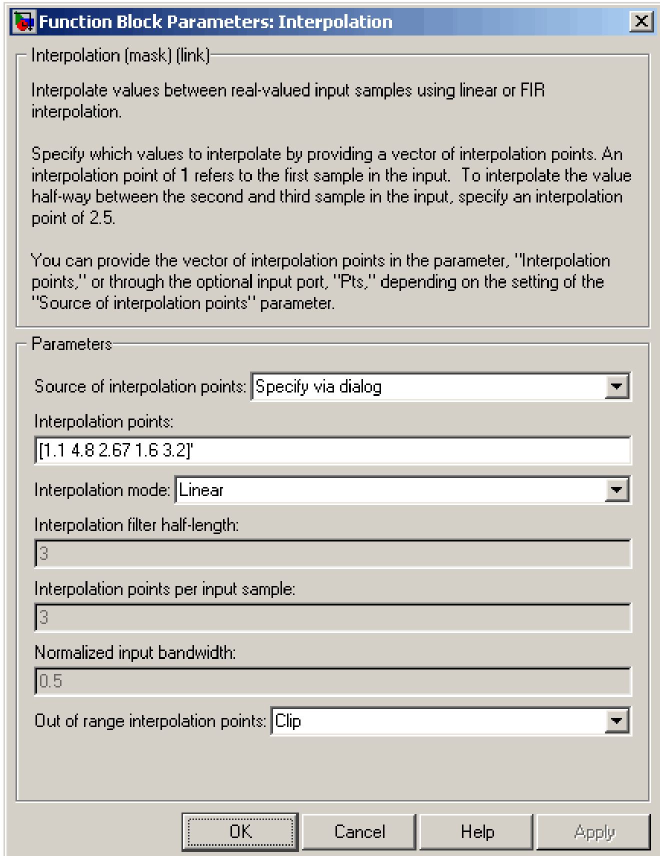
then the block clips the invalid interpolation points, and outputs the linearly interpolated values in a vector, `[1, 1.65, 2.175, 0.25]'`.



FIR Interpolation Mode

When **Interpolation Mode** is set to `FIR`, the block interpolates data values using an FIR interpolation filter, specified by various block parameters. See "FIR Interpolation Mode" on page 2-1284 in the Variable Fractional Delay block reference for more information.

Dialog Box



Source of interpolation points

Choose how you want to specify the interpolation points. If you select `Specify via dialog`, the **Interpolation points** parameter become available. Use this option for static interpolation points. If you select `Input port`, the Pts port appears on the block. The block uses the input to this port as the interpolation points. Use this option for time-varying interpolation points. For more information, see "Specifying Static

Interpolation Points" on page 2-642 and "Specifying Time-Varying Interpolation Points" on page 2-642.

Interpolation points

The vector I_n of points in time at which to interpolate the input signal. An entry of 1 in I_n refers to the first sample of the input, an entry of 2.5 refers to the sample half-way between the second and third input sample, and so on. See "How the Block Applies Interpolation Vectors to Inputs" on page 2-642. Tunable.

Interpolation mode

Sets the block to interpolate by either linear or FIR interpolation. For more information, see "Linear Interpolation Mode" on page 2-646 and "FIR Interpolation Mode" on page 2-647.

Interpolator filter half-length

Half the length of the FIR interpolation filter. For more information, see "FIR Interpolation Mode" on page 2-647.

Interpolation points per input sample

The number Q, where the FIR interpolation filter uses the nearest 2*Q points in the signal to interpolate the value at an interpolation point. When there are less than 2*Q neighboring points, the block uses linear interpolation in place of FIR interpolation. For more information, see "FIR Interpolation Mode" on page 2-647. and "Linear Interpolation Mode" on page 2-646.

Normalized input bandwidth

The bandwidth of the input divided by Fs/2 (half the input sample frequency). For more information, see "FIR Interpolation Mode" on page 2-647.

Out of range interpolation points

When an interpolation point is out of range, this parameter sets the block to either clip the interpolation point, clip the value and issue a warning at the MATLAB command line, or stop the simulation and issue an error at the MATLAB command line. For more information, see "Handling Out-of-Range Interpolation Points" on page 2-645.

Supported Data Types

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

Purpose

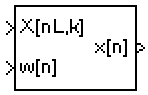
Recover time-domain signals by performing inverse short-time, fast Fourier transform (FFT)

Library

Transforms

`dspxfrm3`

Description



Inverse
Short-Time FFT

The Inverse Short-Time FFT block reconstructs the time-domain signal from the frequency-domain output of the Short-Time FFT block using a two-step process. First, the block performs the overlap add algorithm shown below.

$$x[n] = \frac{L}{W(0)} \sum_{p=-\infty}^{\infty} \left[\frac{1}{N} \sum_{k=0}^{N-1} X[pL,k] e^{j2\pi kn/N} \right]$$

Then, the block rebuffers the signal in order to reconstruct the frame-based time-domain signal. Depending on the analysis window used by the Short-Time FFT block, the Inverse Short-Time FFT block might or might not achieve perfect reconstruction of the time domain signal.

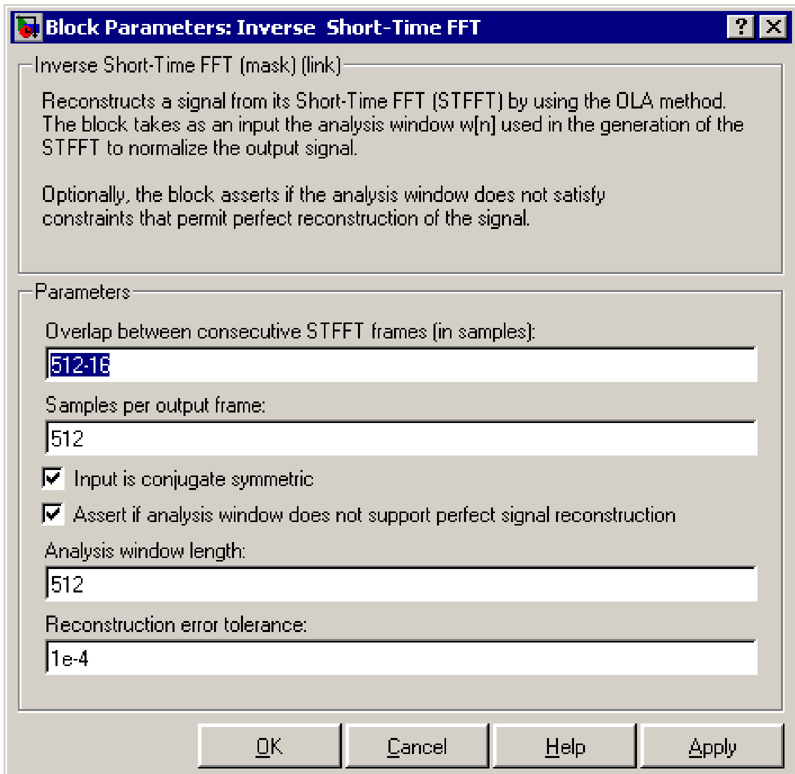
Connect your complex-valued, sample-based, single-channel or multichannel input signal to the X(n,k) port. The block uses the **Overlap between consecutive STFFT frames (in samples)** and **Samples per output frame** parameters as well as the **Input is conjugate symmetric** check box to reconstruct the original time-domain signal. The real or complex-valued, frame-based, single-channel or multichannel inverse short-time FFT is output at port x(n).

Connect your complex-valued, sample-based or frame-based, single-channel analysis window to the w(n) port. When you select the **Assert if analysis window does not support perfect signal reconstruction** check box, the block displays an error when the input signal cannot be perfectly reconstructed. The block uses the values you enter for the **Analysis window length (W)** and **Reconstruction error tolerance**, or maximum amount of allowable error in the reconstruction process, to determine if the signal can be perfectly reconstructed.

Examples

The `dspstsa` demo illustrates how to use the Short-Time FFT and Inverse Short-Time FFT blocks to remove the background noise from a speech signal.

Dialog Box



Overlap between consecutive STFFT frames (in samples)

Enter the number of samples of overlap for each frame of the Short-Time FFT block's input signal. This value should be the same as the **Overlap between consecutive windows (in**

samples) parameter in the Short-Time FFT block parameters dialog.

Samples per output frame
Enter the desired frame length of the frame-based output signal.

Input is conjugate symmetric
Select this check box when the input to the block is both floating point and conjugate symmetric, and you want real-valued outputs. When you select this check box when the input is not conjugate symmetric, the output of the block is invalid. This parameter cannot be used for fixed-point signals.

Assert if analysis window does not support perfect signal reconstruction
Select this check box to display an error when the analysis window used by the Short-Time FFT block does not support perfect signal reconstruction.

Analysis window length
Enter the length of the analysis window. This parameter is visible when you select the **Assert if analysis window does not support perfect signal reconstruction** check box.

Reconstruction error tolerance
Enter the amount of acceptable error in the reconstruction of the original signal. This parameter is visible when you select the **Assert if analysis window does not support perfect signal reconstruction** check box.

References

Quatieri, Thomas E. *Discrete-Time Speech Signal Processing.* Englewood Cliffs, NJ: Prentice-Hall, 2001.

Supported Data Types

Port	Supported Data Types
X(n,k)	• Double-precision floating point • Single-precision floating point

Port	Supported Data Types
w(n)	• Double-precision floating point • Single-precision floating point
x(n)	• Double-precision floating point • Single-precision floating point

See Also

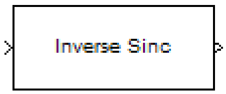
Burg Method	Signal Processing Blockset
Magnitude FFT	Signal Processing Blockset
Periodogram	Signal Processing Blockset
Short-Time FFT	Signal Processing Blockset
Spectrum Scope	Signal Processing Blockset
Window Function	Signal Processing Blockset
Yule-Walker Method	Signal Processing Blockset
`pwelch`	Signal Processing Toolbox

Purpose Design inverse sinc filter

Library Filtering / Filter Design Toolbox

`dspfdesign`

Description



This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See "Inverse Sinc Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Compute filter estimates for inputs using Kalman adaptive filter algorithm

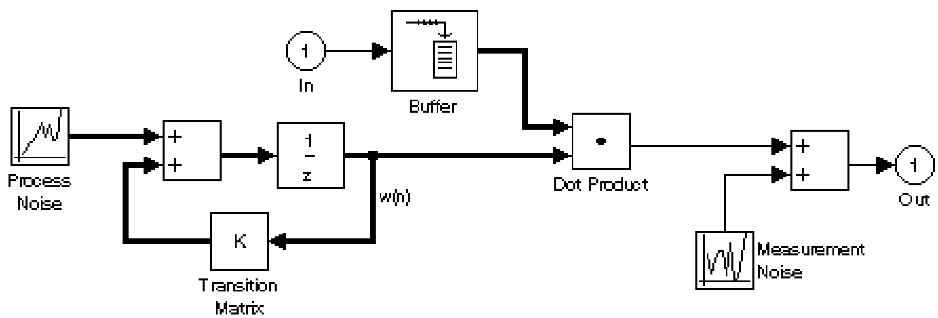
Library

`dspobslib`

Description

Note The Kalman Adaptive Filter block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Kalman Filter block.

The Kalman Adaptive Filter block computes the optimal linear minimum mean-square estimate (MMSE) of the FIR filter coefficients using a one-step predictor algorithm. This Kalman filter algorithm is based on the following physical realization of a dynamic system.



The Kalman filter assumes that there are no deterministic changes to the filter taps over time (that is, the transition matrix is identity), and that the only observable output from the system is the filter output with additive noise. The corresponding Kalman filter is expressed in matrix form as

$$g(n) = \frac{K(n-1)u(n)}{u^H(n)K(n-1)u(n) + Q_M}$$
$$y(n) = u^H(n)\widehat{w}(n)$$
$$e(n) = d(n) - y(n)$$
$$\widehat{w}(n+1) = \widehat{w}(n) + e(n)g(n)$$
$$K(n) = K(n-1) - g(n)u^H(n)K(n-1) + Q_p$$

The variables are as follows

Variable	Description
n	The current algorithm iteration
$u(n)$	The buffered input samples at step n
$K(n)$	The correlation matrix of the state estimation error
$g(n)$	The vector of Kalman gains at step n
$\widehat{w}(n)$	The vector of filter-tap estimates at step n
$y(n)$	The filtered output at step n
$e(n)$	The estimation error at step n
$d(n)$	The desired response at step n
Q_M	The correlation matrix of the measurement noise
Q_P	The correlation matrix of the process noise

The correlation matrices, Q_M and Q_P, are specified in the parameter dialog by scalar variance terms to be placed along the matrix diagonals, thus ensuring that these matrices are symmetric. The filter algorithm based on this constraint is also known as the random-walk Kalman filter.

The implementation of the algorithm in the block is optimized by exploiting the symmetry of the input covariance matrix $K(n)$. This decreases the total number of computations by a factor of two.

The block icon has port labels corresponding to the inputs and outputs of the Kalman algorithm. Note that inputs to the In and Err ports must be sample-based scalars with the same complexity. The signal at the Out port is a scalar, while the signal at the Taps port is a sample-based vector.

Block Ports	Corresponding Variables
In	u, the scalar input, which is internally buffered into the vector $u(n)$
Out	$y(n)$, the filtered scalar output
Err	$e(n)$, the scalar estimation error
Taps	$\hat{w}(n)$, the vector of filter-tap estimates

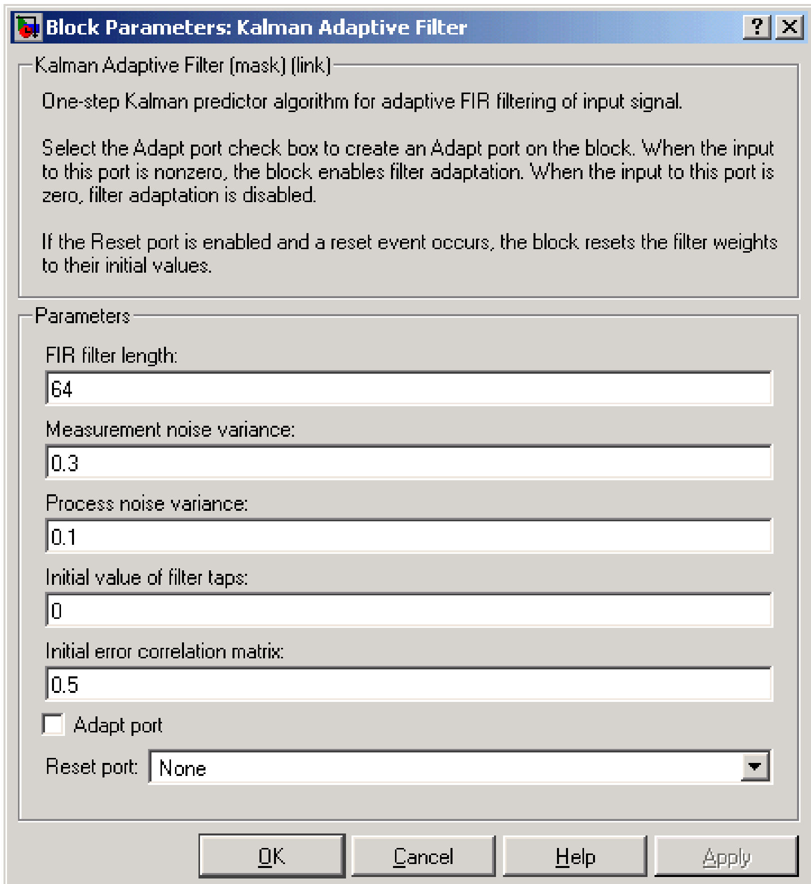
An optional Adapt input port is added when you select the **Adapt port** check box in the dialog. When this port is enabled, the block continuously adapts the filter coefficients while the Adapt input is nonzero. A zero-valued input to the `Adapt` port causes the block to stop adapting, and to hold the filter coefficients at their current values until the next nonzero Adapt input.

The **FIR filter length** parameter specifies the length of the filter that the Kalman algorithm estimates. The **Measurement noise variance** and the **Process noise variance** parameters specify the correlation matrices of the measurement and process noise, respectively. The **Measurement noise variance** must be a scalar, while the **Process noise variance** can be a vector of values to be placed along the diagonal, or a scalar to be repeated for the diagonal elements.

The **Initial value of filter taps** specifies the initial value $\hat{w}(0)$ as a vector, or as a scalar to be repeated for all vector elements. The **Initial**

error correlation matrix specifies the initial value *K(0)*, and can be a diagonal matrix, a vector of values to be placed along the diagonal, or a scalar to be repeated for the diagonal elements.

Dialog Box



FIR filter length

The length of the FIR filter.

Measurement noise variance
The value to appear along the diagonal of the measurement noise correlation matrix. Tunable.

Process noise variance
The value to appear along the diagonal of the process noise correlation matrix. Tunable.

Initial value of filter taps
The initial FIR filter coefficients.

Initial error correlation matrix
The initial value of the error correlation matrix.

Adapt port
Enables the `Adapt` port.

References

Haykin, S. *Adaptive Filter Theory*. 3rd ed. Englewood Cliffs, NJ: Prentice Hall, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

LMS Adaptive Filter	Signal Processing Blockset
RLS Adaptive Filter	Signal Processing Blockset

See “Adaptive Filters” for related information.

Kalman Filter

Purpose

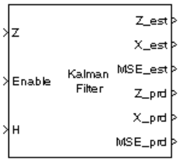
Predict or estimate states of dynamic systems

Library

Filtering/Adaptive Filters

`dspadpt3`

Description



Kalman Filter

Use the Kalman Filter block to predict or estimate the state of a dynamic system from a series of incomplete and/or noisy measurements. Suppose you have a noisy linear system that is defined by the following equations:

$$
\begin{aligned}
x_k &= Ax_{k-1} + w_{k-1} \\
z_k &= Hx_k + v_k
\end{aligned}
$$

This block can use the previously estimated state, $\hat{x}_{k-1}$, to predict the current state at time k, x_k^-, as shown by the following equation:

$$
\begin{aligned}
x_k^- &= A\hat{x}_{k-1} \\
P_k^- &= A\hat{P}_{k-1}A^T + Q
\end{aligned}
$$

The block can also use the current measurement, z_k, and the predicted state, x_k^-, to estimate the current state value at time k, $\hat{x}_k$, so that it is a more accurate approximation:

$$
\begin{aligned}
K_k &= P_k^- H^T \left(HP_k^- H^T + R\right)^{-1} \\
\hat{x}_k &= x_k^- + K_k\left(z_k - Hx_k^-\right) \\
\hat{P}_k &= (I - K_k H)P_k^-
\end{aligned}
$$

The variables in the previous equations are defined in the following table.

Variable	Definition	Default Value or Initial Condition
x	State	N/A
$\hat{x}$	Estimated state	`zeros([6, 1])`
x^{-}	Predicted state	N/A
A	State transition matrix	$\begin{bmatrix} 1 & 0 & 1 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 & 0 & 0 \\ 0 & 0 & 1 & 0 & 0 & 0 \\ 0 & 0 & 0 & 1 & 0 & 0 \\ 0 & 0 & 0 & 0 & 1 & 0 \\ 0 & 0 & 0 & 0 & 0 & 1 \end{bmatrix}$
w	Process noise	N/A
z	Measurement	N/A
H	Measurement matrix	$\begin{bmatrix} 1 & 0 & 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 0 & 1 & 0 \\ 0 & 0 & 0 & 0 & 0 & 1 \end{bmatrix}$
v	Measurement noise	N/A
$\hat{P}$	Estimated error covariance	`10*eye(6)`
P^{-}	Predicted error covariance	N/A
Q	Process noise covariance	`0.05*eye(6)`
K	Kalman gain	N/A

Variable	Definition	Default Value or Initial Condition
R	Measurement noise covariance	`eye(4)`
I	Identity matrix	N/A

In the previous equations, z is a vector of measurement values. Most of the time, the block processes Z, an M-by-N matrix, where M is the number of measurement values and N is the number of filters.

Port	Input/Output	Supported Data Types	Complex Values Supported
Z	M-by-N measurement where M is the length of the measurement vector and N is the number of filters.	• Double-precision floating point • Single-precision floating point	No
Enable	1-by-N vector of 1s and 0s where N is the number of filters.	• Double-precision floating point • Single-precision floating point • Boolean	No
H	M-by-P measurement matrix where M is the length of the measurement vector and P is the length of the filter state vectors.	Same as Z port	No

Port	Input/Output	Supported Data Types	Complex Values Supported
Z_est	M-by-N estimated measurement matrix where M is the length of the measurement vector and N is the number of filters.	Same as Z port	No
X_est	P-by-N estimated state matrix where P is the length of the filter state vectors and N is the number of filters.	Same as Z port	No
MSE_est	1-by-N vector that represents the mean-squared-error of the estimated state. N is the number of filters.	Same as Z port	No
Z_prd	M-by-N predicted measurement matrix where M is the length of the measurement vector and N is the number of filters.	Same as Z port	No

Port	Input/Output	Supported Data Types	Complex Values Supported
X_prd	P-by-N predicted state matrix where P is the length of the filter state vectors and N is the number of filters.	Same as Z port	No
MSE_prd	1-by-N vector that represents the mean-squared-error of the predicted state. N is the number of filters.	Same as Z port	No

Use the **Number of filters** parameter to specify the number of filters to use to predict or estimate the current value.

Use the **Enable filters** parameter to specify which filters are enabled or disabled at each time step. If you select `Always`, the filters are always enabled. If you choose `Specify via input port <Enable>`, the Enable port appears on the block. The input to this port must be a row vector of 1s and 0s whose length is equal to the number of filters. For example, if there are 3 filters and the input to the Enable port is [1 0 1], only the first and third filter are enabled at this time step. If you select the **Reset the estimated state and estimated error covariance when filters are disabled** check box, the estimated and predicted states as well as the estimated error covariance that correspond to the disabled filters are reset to their initial values.

Note All filters have the same state transition matrix, measurement matrix, initial conditions, and noise covariance, but their state, measurement, enable, and MSE signals are unique. Within the state, measurement, enable, and MSE signals, each column corresponds to a filter.

Use the **Measurement matrix source** parameter to specify how to enter the measurement matrix values. If you select `Specify via dialog`, the **Measurement matrix** parameter appears in the dialog box. If you select `Input port <H>`, the H port appears on the block. Use this port to specify your measurement matrix.

See the Radar Tracking demo for an example of how to use this block. You can open this demo by typing

```
aero_radmod_dsp
```

at the MATLAB® command prompt.

If you have the Video and Image Processing Blockset™ product installed, you can also explore the People Tracking demo. Open this demo by typing

```
viptrackpeople
```

at the MATLAB command prompt.

Fixed-Point Data Types

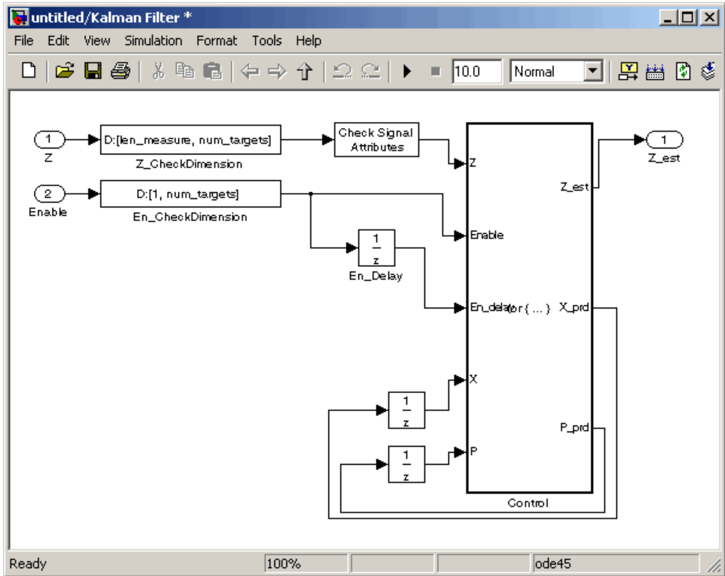
This block can process fixed-point data types if you make several modifications to its subsystem:

1. Add a Kalman Filter block to a Simulink® model.

2. Right-click the block and select **Link Options > Disable Link**.

3. Right-click the block and select **Look Under Mask**.

   The subsystem that implements the Kalman filter opens.

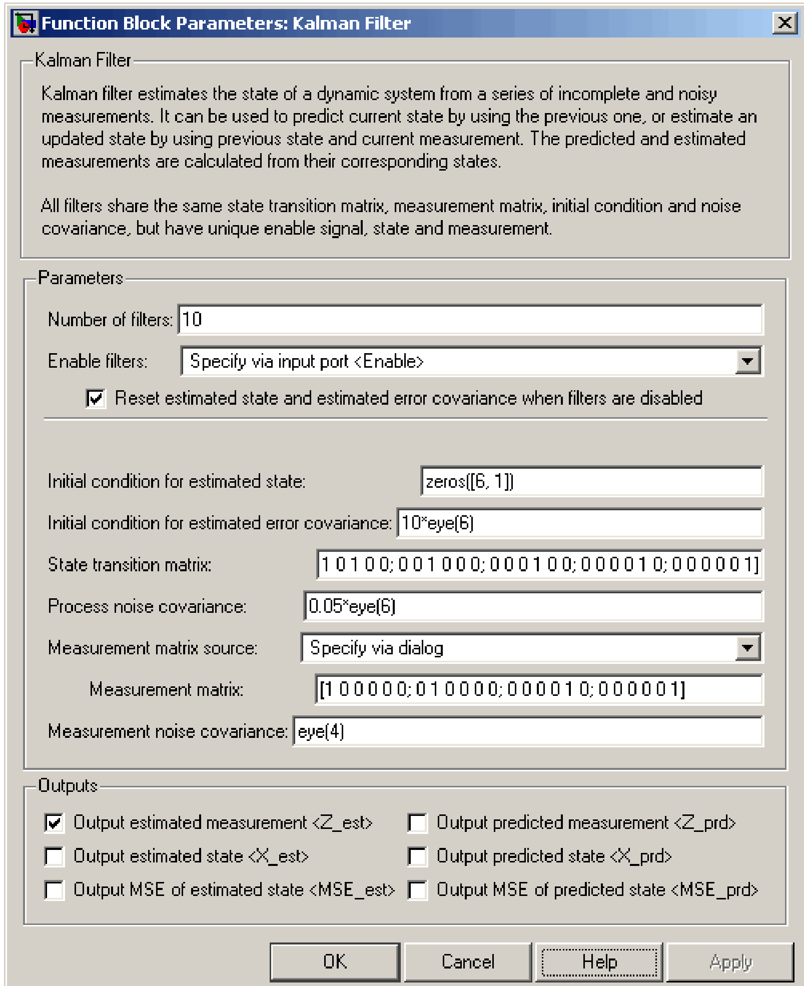


4 Double-click the Check Signal Attributes block and change the **Data type** parameter to `Ignore`.

If you have the Video and Image Processing Blockset product installed, you can see an example by typing `viptrackpeople_fixpt_all` at the MATLAB command prompt. This opens the fixed-point version of the People Tracking demo. The Kalman Filter block is located inside the Enabled Subsystem block called Tracking.

Dialog Box

The Kalman Filter dialog box appears as shown in the following figure.



Number of filters

Specify the number of filters to use to predict or estimate the current value.

Enable filters

Specify which filters are enabled or disabled at each time step. If you select `Always`, the filters are always enabled. If you choose `Specify via input port <Enable>`, the Enable port appears on the block.

Reset the estimated state and estimated error covariance when filters are disabled

If you select this check box, the estimated and predicted states as well as the estimated error covariance that correspond to the disabled filters are reset to their initial values. This parameter is visible if, for the **Enable filters** parameter, you select `Specify via input port <Enable>`.

Initial condition for estimated state

Enter the initial condition for the estimated state.

Initial condition for estimated error covariance

Enter the initial condition for the estimated error covariance.

State transition matrix

Enter the state transition matrix.

Process noise covariance

Enter the process noise covariance.

Measurement matrix source

Specify how to enter the measurement matrix values. If you select `Specify via dialog`, the **Measurement matrix** parameter appears in the dialog box. If you select `Input port <H>`, the H port appears on the block.

Measurement matrix

Enter the measurement matrix values. This parameter is visible if you select `Specify via dialog` for the **Measurement matrix source** parameter.

Measurement noise covariance

Enter the measurement noise covariance.

Output estimated measurement <Z_est>
Select this check box if you want the block to output the estimated measurement.

Output estimated state <X_est>
Select this check box if you want the block to output the estimated state.

Output MSE of estimated state <MSE_est>
Select this check box if you want the block to output the mean-squared error of the estimated state.

Output predicted measurement <Z_prd>
Select this check box if you want the block to output the predicted measurement.

Output predicted state <X_prd>
Select this check box if you want the block to output the predicted state.

Output MSE of predicted state <MSE_prb>
Select this check box if you want the block to output the mean-squared error of the predicted state.

References

[1] Haykin, Simon. *Adaptive Filter Theory*. Upper Saddle River, NJ: Prentice Hall, 1996.

[2] Welch, Greg and Gary Bishop, "An Introduction to the Kalman Filter," TR 95–041, Department of Computer Science, University of North Carolina.

See Also

LDL Solver	Signal Processing Blockset

LDL Factorization

Purpose

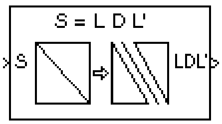
Factor square Hermitian positive definite matrices into lower, upper, and diagonal components

Library

Math Functions / Matrices and Linear Algebra / Matrix Factorizations

`dspfactors`

Description



The LDL Factorization block uniquely factors the square Hermitian positive definite input matrix S as

$$S = LDL^*$$

where L is a lower triangular square matrix with unity diagonal elements, D is a diagonal matrix, and L^* is the Hermitian (complex conjugate) transpose of L. Only the diagonal and lower triangle of the input matrix are used, and any imaginary component of the diagonal entries is disregarded.

The block's output is a composite matrix with lower triangle elements l_{ij} from L, diagonal elements d_{ij} from D, and upper triangle elements u_{ij} from L^*. It is always sample based. The output format is shown below for a 5-by-5 matrix.

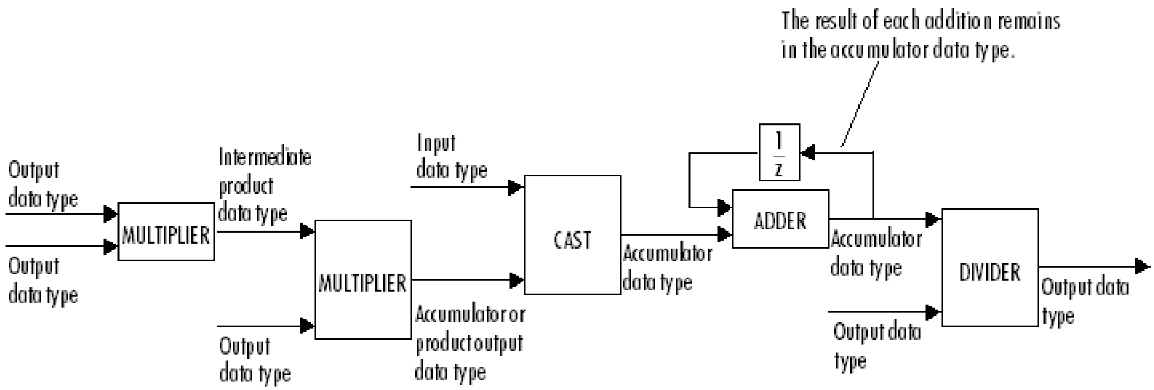
$$\begin{bmatrix} d_{11} & u_{12} & u_{13} & u_{14} & u_{15} \\ l_{21} & d_{22} & u_{23} & u_{24} & u_{25} \\ l_{31} & l_{32} & d_{33} & u_{34} & u_{35} \\ l_{41} & l_{42} & l_{43} & d_{44} & u_{45} \\ l_{51} & l_{52} & l_{53} & l_{54} & d_{55} \end{bmatrix} \qquad u_{ij} = l_{ji}^*$$

LDL factorization requires half the computation of Gaussian elimination (LU decomposition), and is always stable. It is more efficient than Cholesky factorization because it avoids computing the square roots of the diagonal elements.

The algorithm requires that the input be square and Hermitian positive definite. When the input is not positive definite, the block reacts with the behavior specified by the **Non-positive definite input** parameter.

Fixed-Point Data Types

The following diagram shows the data types used within the LDL Factorization block for fixed-point signals.



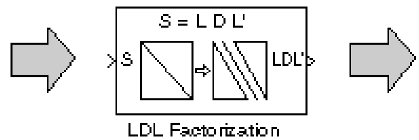
You can set the intermediate product, product output, accumulator, and output data types in the block dialog as discussed below.

The output of the second multiplier is in the product output data type when the input is real. When the input is complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”.

Examples

LDL decomposition of a 3-by-3 Hermitian positive definite matrix:

$$\begin{bmatrix} 9 & -1 & 2 \\ -1 & 8 & -5 \\ 2 & -5 & 7 \end{bmatrix}$$

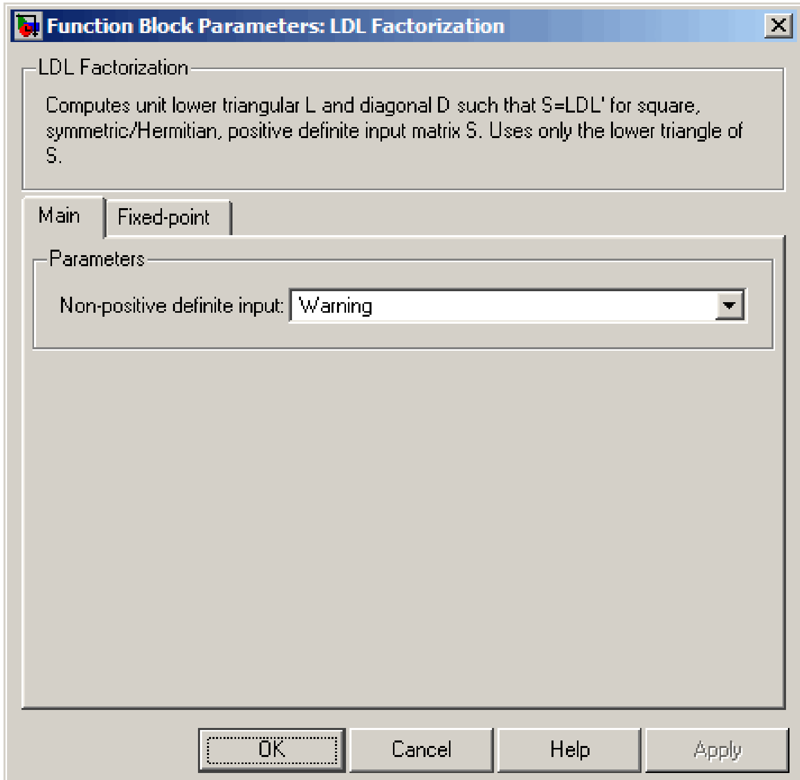


$$\begin{bmatrix} 9.00 & -0.11 & 0.22 \\ -0.11 & 7.89 & -0.61 \\ 0.22 & -0.61 & 3.66 \end{bmatrix}$$

$$L = \begin{bmatrix} 1 & 0 & 0 \\ -0.11 & 1 & 0 \\ 0.22 & -0.61 & 1 \end{bmatrix} \qquad D = \begin{bmatrix} 9.00 & 0 & 0 \\ 0 & 7.89 & 0 \\ 0 & 0 & 3.66 \end{bmatrix} \qquad L' = \begin{bmatrix} 1 & -0.11 & 0.22 \\ 0 & 1 & -0.61 \\ 0 & 0 & 1 \end{bmatrix}$$

Dialog Box

The **Main** pane of the LDL Factorization block dialog appears as follows.



Non-positive definite input

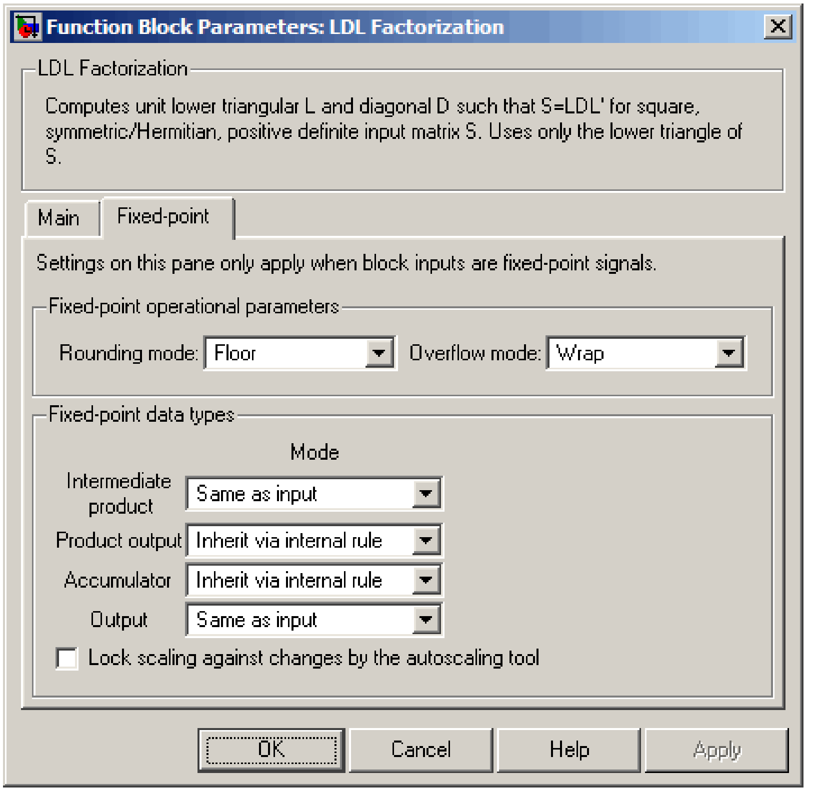
Specify the action when nonpositive definite matrix inputs occur:

- `Ignore` — Proceed with the computation and do not issue an alert. The output is not a valid factorization. A partial factorization is present in the upper left corner of the output.
- `Warning` — Display a warning message in the MATLAB® Command Window, and continue the simulation. The output is

not a valid factorization. A partial factorization is present in the upper left corner of the output.

- `Error` — Display an error dialog and terminate the simulation.

The **Fixed-point** pane of the LDL Factorization block dialog appears as follows.



Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

Select the overflow mode for fixed-point operations.

Intermediate product

Use this parameter to specify how you would like to designate the intermediate product word and fraction lengths. See “Fixed-Point Data Types” on page 2-673 for an illustration depicting the use of the intermediate product data type in this block:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the intermediate product, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the intermediate product. This block requires power-of-two slope and a bias of zero.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See “Fixed-Point Data Types” on page 2-673 and “Multiplication Data Types” for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see “Inherit via Internal Rule”.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-673 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Use this parameter to specify how you would like to designate the output word and fraction lengths. See "Fixed-Point Data Types" on page 2-673 for an illustration depicting the use of the output data type in this block:

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

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

Port	Supported Data Types
S	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
LDL'	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers

See Also

Cholesky Factorization	Signal Processing Blockset
LDL Inverse	Signal Processing Blockset
LDL Solver	Signal Processing Blockset
LU Factorization	Signal Processing Blockset
QR Factorization	Signal Processing Blockset

See “Matrix Factorizations” for related information.

Purpose

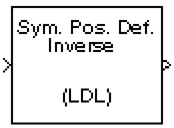
Compute inverse of Hermitian positive definite matrix using LDL factorization

Library

Math Functions / Matrices and Linear Algebra / Matrix Inverses

`dspinverses`

Description



The LDL Inverse block computes the inverse of the Hermitian positive definite input matrix S by performing an LDL factorization.

$$S^{-1} = \left(LDL^{*}\right)^{-1}$$

L is a lower triangular square matrix with unity diagonal elements, D is a diagonal matrix, and L^{*} is the Hermitian (complex conjugate) transpose of L. Only the diagonal and lower triangle of the input matrix are used, and any imaginary component of the diagonal entries is disregarded. The output is always sample based.

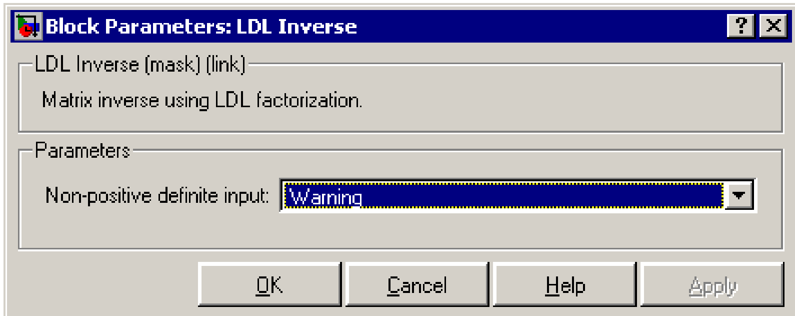
LDL factorization requires half the computation of Gaussian elimination (LU decomposition), and is always stable. It is more efficient than Cholesky factorization because it avoids computing the square roots of the diagonal elements.

The algorithm requires that the input be Hermitian positive definite. When the input is not positive definite, the block reacts with the behavior specified by the **Non-positive definite input** parameter. The following options are available:

- `Ignore` — Proceed with the computation and do not issue an alert. The output is not a valid inverse.
- `Warning` — Display a warning message in the MATLAB® command window, and continue the simulation. The output is not a valid inverse.
- `Error` — Display an error dialog and terminate the simulation.

Note The **Non-positive definite input** parameter is a diagnostic parameter. Like all diagnostic parameters on the Configuration Parameters dialog, it is set to `Ignore` in the code generated for this block by Real-Time Workshop® code generation software.

Dialog Box



Non-positive definite input
Response to nonpositive definite matrix inputs.

References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Cholesky Inverse	Signal Processing Blockset
LDL Factorization	Signal Processing Blockset
LDL Solver	Signal Processing Blockset
LU Inverse	Signal Processing Blockset
Pseudoinverse	Signal Processing Blockset
`inv`	MATLAB

See “Matrix Inverses” for related information.

Purpose

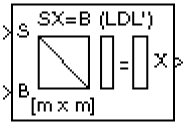
Solve $SX=B$ for X when S is square Hermitian positive definite matrix

Library

Math Functions / Matrices and Linear Algebra / Linear System Solvers

`dspsolvers`

Description



The LDL Solver block solves the linear system SX=B by applying LDL factorization to the matrix at the S port, which must be square (M-by-M) and Hermitian positive definite. Only the diagonal and lower triangle of the matrix are used, and any imaginary component of the diagonal entries is disregarded. The input to the B port is the right side M-by-N matrix, B. The output is the unique solution of the equations, M-by-N matrix X, and is always sample based.

A length-M 1-D vector input for right side B is treated as an M-by-1 matrix.

When the input is not positive definite, the block reacts with the behavior specified by the **Non-positive definite input** parameter. The following options are available:

- `Ignore` — Proceed with the computation and do not issue an alert. The output is not a valid solution.
- `Warning` — Proceed with the computation and display a warning message in the MATLAB® Command Window. The output is not a valid solution.
- `Error` — Display an error dialog and terminate the simulation.

Note The **Non-positive definite input** parameter is a diagnostic parameter. Like all diagnostic parameters on the Configuration Parameters dialog, it is set to `Ignore` in the code generated for this block by Real-Time Workshop® code generation software.

Algorithm

The LDL algorithm uniquely factors the Hermitian positive definite input matrix S as

$$S = LDL^*$$

where L is a lower triangular square matrix with unity diagonal elements, D is a diagonal matrix, and L^* is the Hermitian (complex conjugate) transpose of L.

The equation

$$LDL^*X = B$$

is solved for X by the following steps:

1. Substitute

   $$Y = DL^*X$$

2. Substitute

   $$Z = L^*X$$

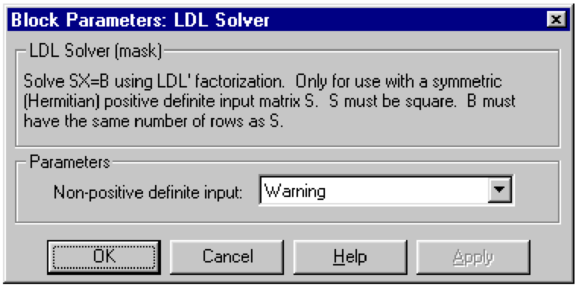
3. Solve one diagonal and two triangular systems.

   $$LY = B$$

   $$DZ = Y$$

   $$L^*X = Z$$

Dialog Box



Non-positive definite input
Response to nonpositive definite matrix inputs.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Autocorrelation LPC	Signal Processing Blockset
Cholesky Solver	Signal Processing Blockset
LDL Factorization	Signal Processing Blockset
LDL Inverse	Signal Processing Blockset
Levinson-Durbin	Signal Processing Blockset
LU Solver	Signal Processing Blockset
QR Solver	Signal Processing Blockset

See “Linear System Solvers” for related information.

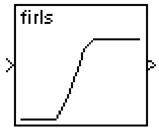
Purpose

Design and implement least-squares FIR filter

Library

`dspobslib`

Description



Note The Least Squares FIR Filter Design block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Least Squares FIR Filter Design block designs an FIR filter and applies it to a discrete-time input using the Direct Form II Transpose Filter block. The filter design uses the Signal Processing Toolbox™ `firls` function to minimize the integral of the squared error between the desired frequency response and the actual frequency response.

An M-by-N sample-based matrix input is treated as $M*N$ independent channels, and an M-by-N frame-based matrix input is treated as N independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

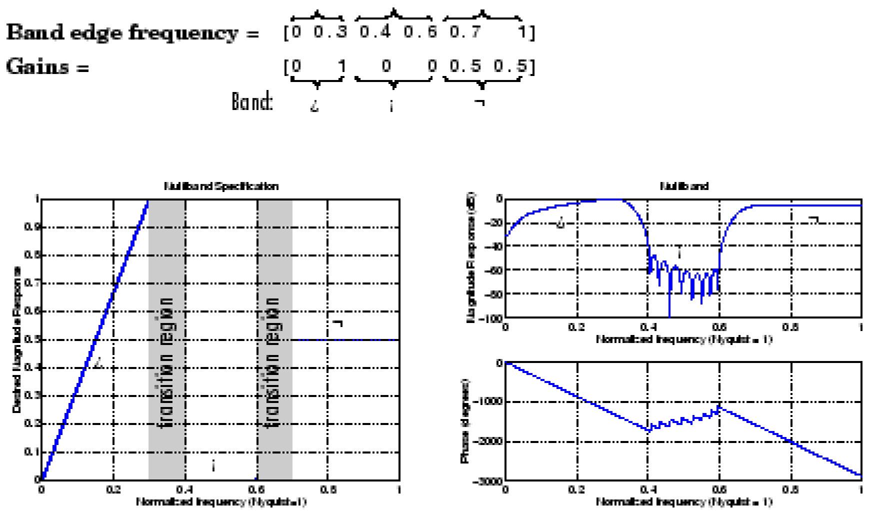
The **Filter type** parameter allows you to specify one of the following filters:

- `Multiband` — The `Multiband` filter designs a linear-phase filter with an arbitrary magnitude response.
- `Differentiator` — The `Differentiator` filter approximates the ideal differentiator. Differentiators are antisymmetric FIR filters with approximately linear magnitude responses. To obtain the correct derivative, scale the **Gains at these frequencies** vector by πF_s rad/s, where F_s is the sample frequency in Hertz.

- `Hilbert Transformer` — The `Hilbert Transformer` filter approximates the ideal Hilbert transformer. Hilbert transformers are antisymmetric FIR filters with approximately constant magnitude.

The **Band-edge frequency vector** parameter is a vector of frequency points in the range `0` to `1`, where `1` corresponds to half the sample frequency. This vector must have even length, and intermediate points must appear in ascending order. The **Gains at these frequencies** parameter is a vector containing the desired magnitude response at the corresponding points in the **Band-edge frequency vector**.

Each odd-indexed frequency-amplitude pair defines the left endpoint of a line segment representing the desired magnitude response in that frequency band. The corresponding even-indexed frequency-amplitude pair defines the right endpoint. Between the frequency bands specified by these end-points, there may be undefined sections of the specified frequency response. These are called “don’t care” or “transition” regions, and the magnitude response in these areas is a result of the optimization in the other (specified) frequency ranges.



The **Weights** parameter is a vector that specifies the emphasis to be placed on minimizing the error in certain frequency bands relative to others. This vector specifies one weight per band, so it is half the length of the **Band-edge frequency vector** and **Gains at these frequencies** vectors.

In most cases, differentiators and Hilbert transformers have only a single band, so the weight is a scalar value that does not affect the final filter. However, the **Weights** parameter is useful when using the block to design an antisymmetric multiband filter, such as a Hilbert transformer with stopbands.

For more information on the **Band-edge frequency vector**, **Gains at these frequencies**, and **Weights** parameters, see "Filter Designs and Implementation" in the Signal Processing Toolbox documentation. For more on the FIR filter algorithm, see the description of the `firls` function in the Signal Processing Toolbox documentation.

Examples

Example 1: Multiband

Consider a lowpass filter with a transition band in the normalized frequency range 0.4 to 0.5, and 10 times more error minimization in the stopband than the passband. In this case,

- **Filter type** = `Multiband`
- **Band-edge frequency vector** = `[0 0.4 0.5 1]`
- **Gains at these frequencies** = `[1 1 0 0]`
- **Weights** = `[1 10]`

Example 2: Differentiator

Assume the specifications for a differentiator filter require it to have order 21. The "ramp" response extends over the entire frequency range. In this case, specify:

- **Filter type** = `Differentiator`
- **Filter order** = `21`

- **Band-edge frequency vector** = `[0 1]`
- **Gains at these frequencies** = `[0 pi*Fs]`

For a type III (even order) filter, the differentiation band should stop short of half the sample frequency. For example, if the filter order is 20, you could specify the block parameters as follows:

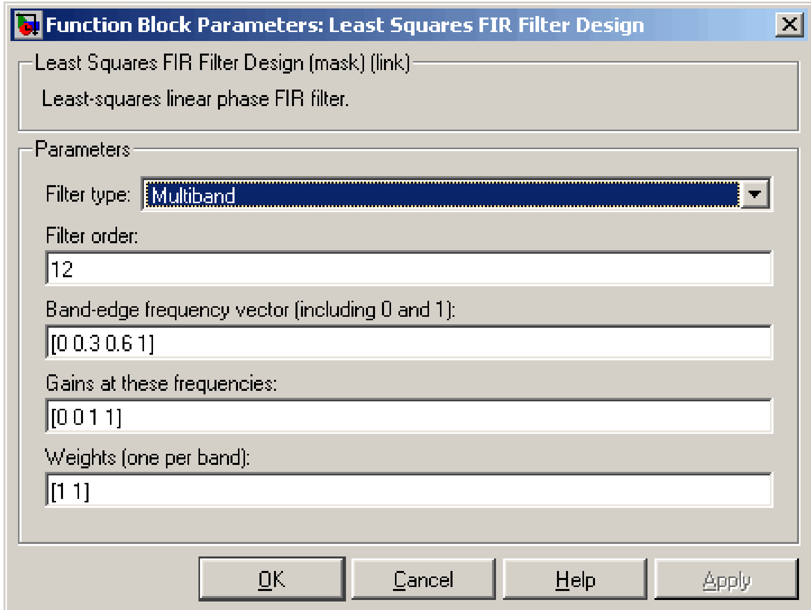
- **Filter type** = `Differentiator`
- **Filter order** = `20`
- **Band-edge frequency vector** = `[0 0.9]`
- **Gains at these frequencies** = `[0 0.9*pi*Fs]`

Example 3: Hilbert Transformer

Assume the specifications for a Hilbert transformer filter require it to have order 21. The passband extends over approximately the entire frequency range. In this case, specify:

- **Filter type** = `Hilbert Transform`
- **Filter order** = `21`
- **Band-edge frequency vector** = `[0.1 1]`
- **Gains at these frequencies** = `[1 1]`

Dialog Box



Filter type
The filter type. Tunable.

Filter order
The filter order.

Band-edge frequency vector
A vector of frequency points, in ascending order, in the range 0 to 1. The value 1 corresponds to half the sample frequency. This vector must have even length. Tunable.

Gains at these frequencies
A vector of frequency-response amplitudes corresponding to the points in the **Band-edge frequency vector**. This vector must be the same length as the **Band-edge frequency vector**. Tunable.

Weights

A vector containing one weight for each frequency band. This vector must be half the length of the **Band-edge frequency vector** and **Gains at these frequencies** vectors. Tunable.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Purpose

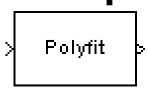
Compute polynomial coefficients that best fit input data in least-squares sense

Library

Math Functions / Polynomial Functions

`dsppolyfun`

Description



The Least Squares Polynomial Fit block computes the coefficients of the nth order polynomial that best fits the input data in the least-squares sense, where you specify n in the **Polynomial order** parameter. A distinct set of n+1 coefficients is computed for each column of the M-by-N input, u.

For a given input column, the block computes the set of coefficients, c_1, c_2, ..., c_{n+1}, that minimizes the quantity

$$\sum_{i=1}^{M}(u_i - \hat{u}_i)^2$$

where u_i is the ith element in the input column, and

$$\hat{u}_i = f(x_i) = c_1 x_i^n + c_2 x_i^{n-1} + \ldots + c_{n+1}$$

The values of the independent variable, x_1, x_2, ..., x_M, are specified as a length-M vector by the **Control points** parameter. The same M control points are used for all N polynomial fits, and can be equally or unequally spaced. The equivalent MATLAB® code is shown below.

```
c = polyfit(x,u,n)      % Equivalent MATLAB code
```

Inputs can be frame based or sample based. For convenience, a length-M 1-D vector input is treated as an M-by-1 matrix.

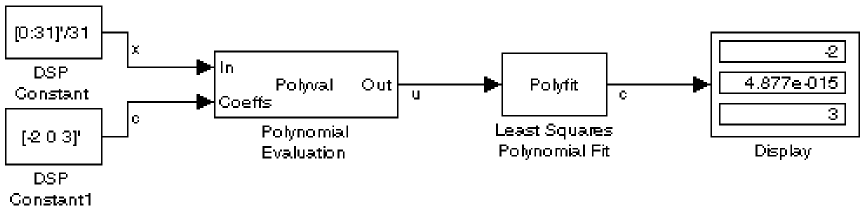
Each column of the (n+1)-by-N output matrix, c, represents a set of n+1 coefficients describing the best-fit polynomial for the corresponding column of the input. The coefficients in each column are arranged in order of descending exponents, c_1, c_2, ..., c_{n+1}. The output is always sample based.

Examples

In the model below, the Polynomial Evaluation block uses the second-order polynomial

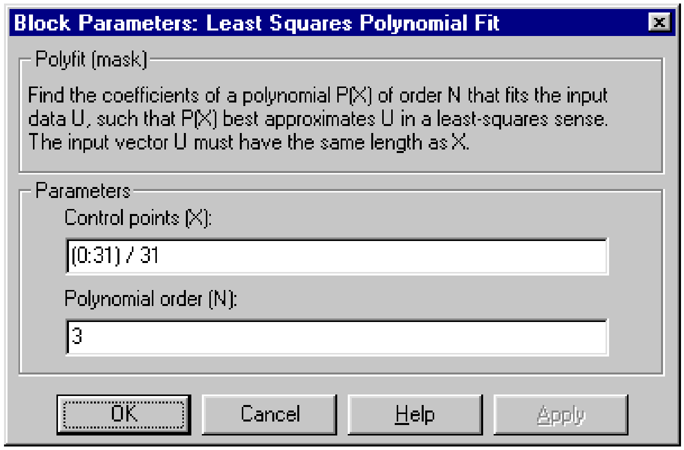
$$y = -2u^2 + 3$$

to generate four values of dependent variable y from four values of independent variable u, received at the top port. The polynomial coefficients are supplied in the vector `[-2 0 3]` at the bottom port. Note that the coefficient of the first-order term is zero.



The **Control points** parameter of the Least Squares Polynomial Fit block is configured with the same four values of independent variable u that are used as input to the Polynomial Evaluation block, `[1 2 3 4]`. The Least Squares Polynomial Fit block uses these values together with the input values of dependent variable y to reconstruct the original polynomial coefficients.

Dialog Box



Control points

The values of the independent variable to which the data in each input column correspond. For an M-by-N input, this parameter must be a length-M vector. Tunable.

Polynomial order

The order, n, of the polynomial to be used in constructing the best fit. The number of coefficients is n+1.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Detrend	Signal Processing Blockset
Polynomial Evaluation	Signal Processing Blockset
Polynomial Stability Test	Signal Processing Blockset
`polyfit`	MATLAB

Levinson-Durbin

Purpose Solve linear system of equations using Levinson-Durbin recursion

Library Math Functions / Matrices and Linear Algebra / Linear System Solvers

`dspsolvers`

Description

The Levinson-Durbin block solves the nth-order system of linear equations

$$Ra = b$$

for the particular case where R is a Hermitian, positive-definite, Toeplitz matrix and b is identical to the first column of R shifted by one element and with the opposite sign.

$$\begin{bmatrix} r(1) & r^*(2) & \cdots & r^*(n) \\ r(2) & r(1) & \cdots & r^*(n-1) \\ \vdots & \vdots & \ddots & \vdots \\ r(n) & r(n-1) & \cdots & r(1) \end{bmatrix} \begin{bmatrix} a(2) \\ a(3) \\ \vdots \\ a(n+1) \end{bmatrix} = \begin{bmatrix} -r(2) \\ -r(3) \\ \vdots \\ -r(n+1) \end{bmatrix}$$

The input to the block, r `= [`r`(1)` r`(2) ...` r`(`n`+1)]`, can be a 1-D or 2-D row or column vector or a sample- or frame-based matrix. If the input is a matrix, each column is treated as an independent channel and is solved separately. Each channel of the input contains lags *0* through n of an autocorrelation sequence, which appear in the matrix R.

The block can output the polynomial coefficients, A, the reflection coefficients, K, and the prediction error power, P, in various combinations. The **Output(s)** parameter allows you to enable the A and K outputs by selecting one of the following settings:

- `A` — For each channel, port A outputs A`=[1` a`(2)` a`(3) ...` a`(`n`+1)]`, the solution to the Levinson-Durbin equation. A has the same dimension as the input. The elements of each output channel can also be viewed as the coefficients of an nth-order autoregressive (AR) process (see below).

- `K` — For each channel, port K outputs K=`[`k`(1)` k`(2)` `...` k`(`n`)]`, which contains n reflection coefficients, and has the same dimension as the input, less one element. A scalar input channel causes an error when you select `K`. Reflection coefficients are useful for realizing a lattice representation of the AR process described below.
- `A and K` — The block outputs both representations at their respective ports. A scalar input channel causes an error when you select `A and K`.

A and K are matrices if the input is a matrix. Otherwise, A and K are 1-D vectors.

The prediction error power for each channel, P, is output when you select the **Output prediction error power (P)** check box. For each channel, P represents the power of the output of an FIR filter with taps A and input autocorrelation described by r, where A represents a prediction error filter and r is the input to the block. In this case, A is a whitening filter. P has one element per input channel.

When you select the **If the value of lag 0 is zero, A=[1 zeros], K=[zeros], P=0** check box (default), an input channel whose r`(1)` element is zero generates a zero-valued output. When you do not select this check box, an input with r`(1) = 0` generates `NaN`s in the output. In general, an input with r`(1) = 0` is invalid because it does not construct a positive-definite matrix R; however, it is common for blocks to receive zero-valued inputs at the start of a simulation. The check box allows you to avoid propagating `NaN`s during this period.

Applications

One application of the Levinson-Durbin formulation above is in the Yule-Walker AR problem, which concerns modeling an unknown system as an autoregressive process. Such a process would be modeled as the output of an all-pole IIR filter with white Gaussian noise input. In the Yule-Walker problem, the use of the signal's autocorrelation sequence to obtain an optimal estimate leads to an $Ra = b$ equation of the type shown above, which is most efficiently solved by Levinson-Durbin recursion. In this case, the input to the block represents the autocorrelation sequence, with `r(1)` being the zero-lag value. The output at the block's A port then contains the coefficients of the autoregressive process that optimally models the system. The coefficients are ordered in descending powers of z, and the AR process is minimum phase. The prediction error, G, defines the gain for the unknown system, where $G = \sqrt{P}$.

$$H(z) = \frac{G}{A(z)} = \frac{G}{1 + a(2)z^{-1} + \ \dots \ + a(n+1)z^{-n}}$$

The output at the block's K port contains the corresponding reflection coefficients, `[`k`(1)` k`(2) ...` k`(n)]`, for the lattice realization of this IIR filter. The Yule-Walker AR Estimator block implements this autocorrelation-based method for AR model estimation, while the Yule-Walker Method block extends the method to spectral estimation.

Another common application of the Levinson-Durbin algorithm is in linear predictive coding, which is concerned with finding the coefficients of a moving average (MA) process (or FIR filter) that predicts the next value of a signal from the current signal sample and a finite number of past samples. In this case, the input to the block represents the signal's autocorrelation sequence, with r`(1)` being the zero-lag value, and the output at the block's A port contains the coefficients of the predictive MA process (in descending powers of z).

$$H(z) = A(z) = 1 + a(2)z^{-1} + \ \dots \ a(n+1)z^{-n}$$

These coefficients solve the optimization problem below.

$$\min_{\{a_i\}}$$

$$E\left[\left|x_n - \sum_{i=1}^{N} a_i x_{n-i}\right|^2\right]$$

Again, the output at the block's K port contains the corresponding reflection coefficients, `[k(1) k(2) ... k(n)]`, for the lattice realization of this FIR filter. The Autocorrelation LPC block in the Linear Prediction library implements this autocorrelation-based prediction method.

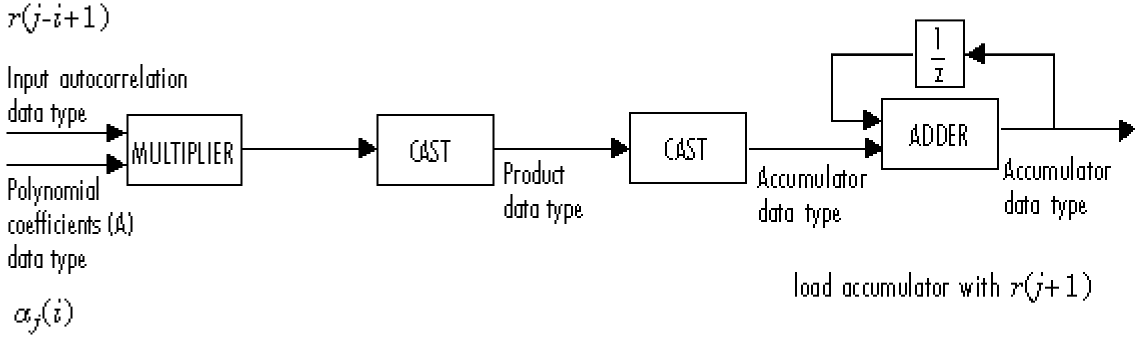
Fixed-Point Data Types

The diagrams in this section show the data types used within the Levinson-Durbin block for fixed-point signals.

After initialization, n updates are performed. At the (j+1) update,

$$\text{value in accumulator } = r(j+1) + \sum a_j(i) \times r(j-i+1)$$

The diagram below displays the fixed-point data types used in this calculation:



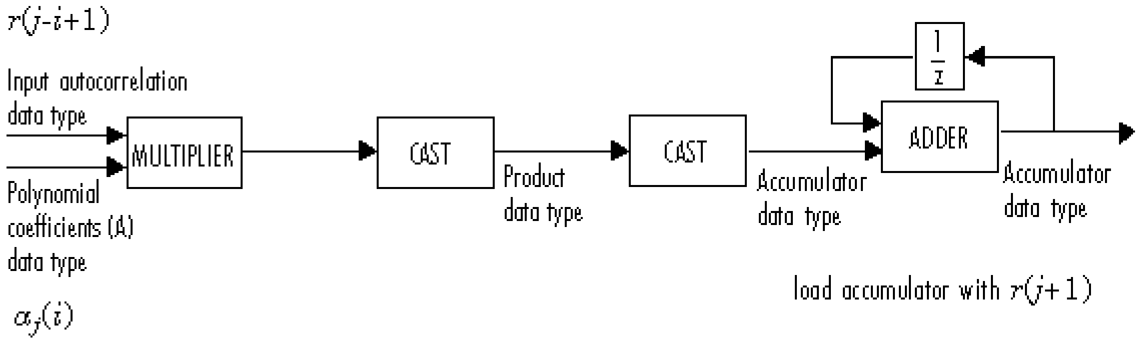
The reflection coefficients K are then updated according to

$$K_{j+1} = \text{ value in accumulator} / P_j$$

The prediction error power P is then updated according to

$$P_{j+1} = P_j - P_j \times K_{j+1} \times \text{conj}\left(K_{j+1}\right)$$

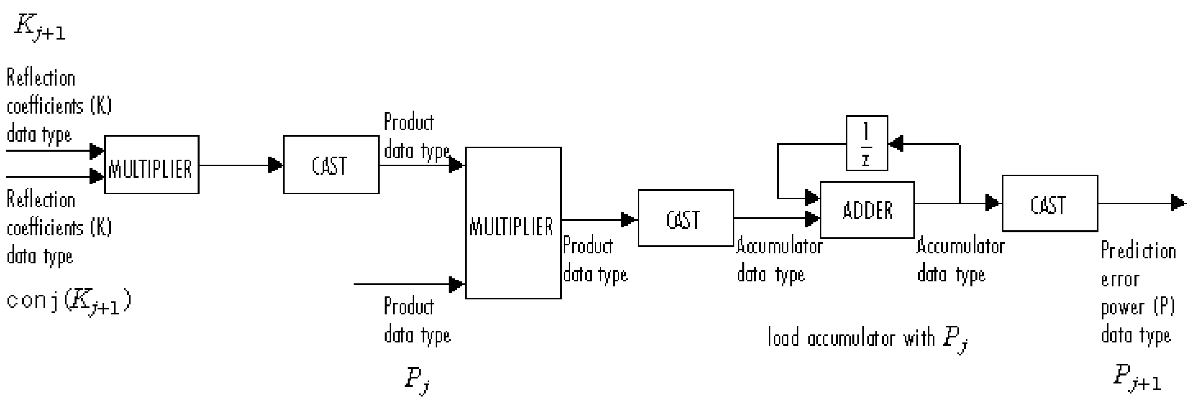
The diagram below displays the fixed-point data types used in this calculation:



The polynomial coefficients A are then updated according to

$$a_{j+1}(i) = a_j(i) + K_{j+1} \times \mathrm{conj}\left(\mathrm{a_j}(j-1+i)\right)$$

The diagram below displays the fixed-point data types used in this calculation:

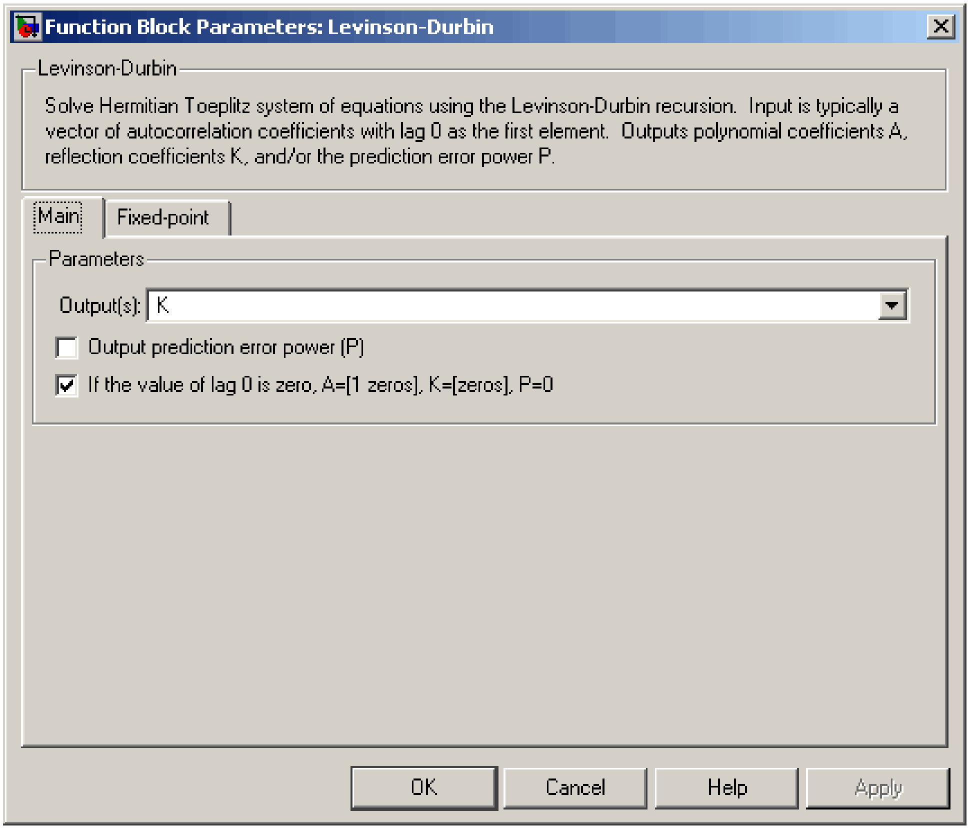


Algorithm

The algorithm requires $O(n^2)$ operations for each input channel, and is therefore much more efficient for large n than standard Gaussian elimination, which requires $O(n^3)$ operations per channel.

Dialog Box

The **Main** pane of the Levinson-Durbin block dialog appears as follows.



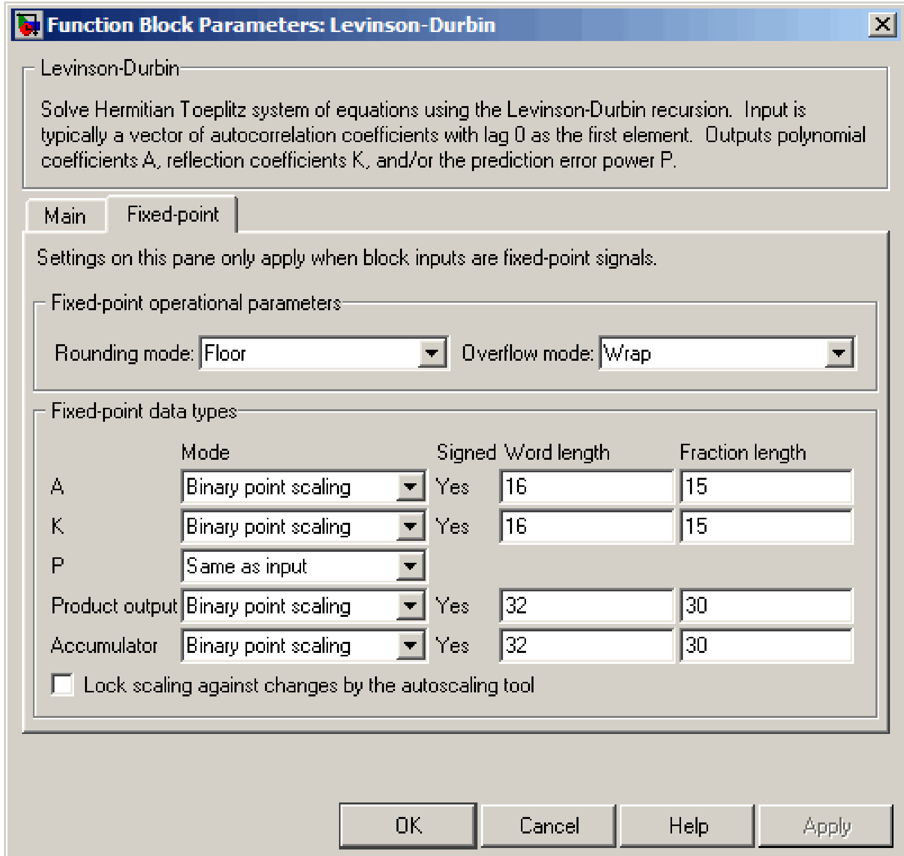
Output(s)

Specify the solution representation of $Ra = b$ to output: model coefficients (`A`), reflection coefficients (`K`), or both (`A and K`). For scalar and frame-based row vector inputs, this parameter must be set to `A`.

Output prediction error power (P)
Select to output the prediction error at port P.

If the value of lag 0 is zero, A=[1 zeros], K=[zeros], P=0
Set to output a zero-vector for inputs having `r(1) = 0`. Otherwise, the block outputs `NaN`s for these inputs.

The **Fixed-point** pane of the Levinson-Durbin block dialog appears as follows.



Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

Select the overflow mode for fixed-point operations.

A

Use this parameter to designate how you would like to specify the word and fraction lengths of the polynomial coefficients (A). See

"Fixed-Point Data Types" on page 2-699 for illustrations depicting the use of the polynomial coefficients data type in this block.

- When you select `Binary point scaling`, you can enter the word length and fraction length of *A*, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of *A*. This block requires power-of-two slope and a bias of zero.

K

Use this parameter to designate how you would like to specify the word and fraction lengths of the reflection coefficients (*K*). See "Fixed-Point Data Types" on page 2-699 for illustrations depicting the use of the reflection coefficients data type in this block.

- When you select `Binary point scaling`, you can enter the word length and fraction length of *K*, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of *K*. This block requires power-of-two slope and a bias of zero.

P

Use this parameter to designate how you would like to specify the word and fraction lengths of the prediction error power (*P*). See "Fixed-Point Data Types" on page 2-699 for illustrations depicting the use of the prediction error power data type in this block.

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and fraction length of *P*, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of *P*. This block requires power-of-two slope and a bias of zero.

Product output

Use this parameter to designate how you would like to specify the product output word and fraction lengths. See "Fixed-Point

Data Types” on page 2-699 for illustrations depicting the use of the product output data type in this block.

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to designate how you would like to specify the accumulator word and fraction lengths. See “Fixed-Point Data Types” on page 2-699 for illustrations depicting the use of the accumulator data type in this block.

- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

References

Golub, G. H., and C. F. Van Loan. Sect. 4.7 in *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Ljung, L. *System Identification: Theory for the User*. Englewood Cliffs, NJ: Prentice Hall, 1987. Pgs. 278-280.

Kay, Steven M., *Modern Spectral Estimation: Theory and Application*. Englewood Cliffs, NJ: Prentice Hall, 1988.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)
- 8-, 16-, and 32-bit signed integers

See Also

Cholesky Solver	Signal Processing Blockset
LDL Solver	Signal Processing Blockset
Autocorrelation LPC	Signal Processing Blockset
LU Solver	Signal Processing Blockset
QR Solver	Signal Processing Blockset
Yule-Walker AR Estimator	Signal Processing Blockset
Yule-Walker Method	Signal Processing Blockset
`levinson`	Signal Processing Toolbox

See “Linear System Solvers” for related information.

Purpose

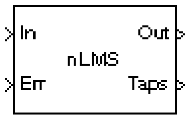
Compute filter estimates for input using LMS adaptive filter algorithm

Library

Filtering / Adaptive Filters

`dspadpt3`

Description



Note The LMS Adaptive Filter block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the LMS Filter block.

The LMS Adaptive Filter block implements an adaptive FIR filter using the stochastic gradient algorithm known as the normalized least mean-square (LMS) algorithm.

$$y(n) = \hat{w}^H(n-1)u(n)$$
$$e(n) = d(n) - y(n)$$
$$\hat{w}(n) = \hat{w}(n-1) + \frac{u(n)}{a + u^H(n)u(n)}\mu e^*(n)$$

The variables are as follows.

Variable	Description
n	The current algorithm iteration
$u(n)$	The buffered input samples at step n
$\hat{w}(n)$	The vector of filter-tap estimates at step n
$y(n)$	The filtered output at step n
$e(n)$	The estimation error at step n
$d(n)$	The desired response at step n
μ	The adaptation step size

To overcome potential numerical instability in the tap-weight update, a small positive constant (a = 1e-10) has been added in the denominator.

To turn off normalization, clear the **Use normalization** check box in the parameter dialog. The block then computes the filter-tap estimate as

$$\hat{w}(n) = \hat{w}(n-1) + u(n)\mu e^{*}(n)$$

The block icon has port labels corresponding to the inputs and outputs of the LMS algorithm. Note that inputs to the In and Err ports must be sample-based scalars. The signal at the Out port is a scalar, while the signal at the Taps port is a sample-based vector.

Block Ports	Corresponding Variables
`In`	u, the scalar input, which is internally buffered into the vector $u(n)$
`Out`	$y(n)$, the filtered scalar output
`Err`	$e(n)$, the scalar estimation error
`Taps`	$\hat{w}(n)$, the vector of filter-tap estimates

An optional `Adapt` input port is added when you select the **Adapt input** check box in the dialog. When this port is enabled, the block continuously adapts the filter coefficients while the `Adapt` input is nonzero. A zero-valued input to the Adapt port causes the block to stop adapting, and to hold the filter coefficients at their current values until the next nonzero `Adapt` input.

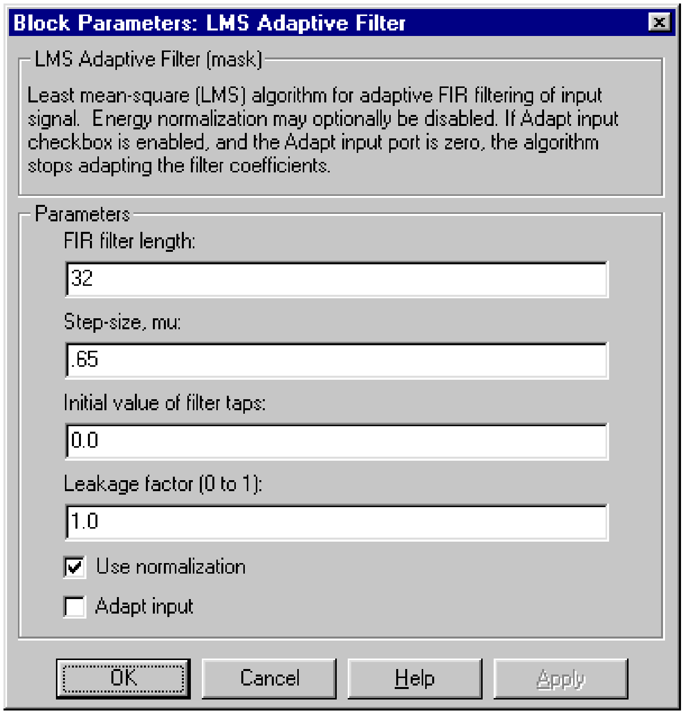
The **FIR filter length** parameter specifies the length of the filter that the LMS algorithm estimates. The **Step size** parameter corresponds to μ in the equations. Typically, for convergence in the mean square, μ must be greater than 0 and less than 2. The **Initial value of filter taps** specifies the initial value $\hat{w}(0)$ as a vector, or as a scalar to be repeated for all vector elements. The **Leakage factor** specifies the value of the leakage factor, 1 –μ α, in the leaky LMS algorithm below. This parameter must be between 0 and 1.

$$\hat{w}(n+1) = (1-\mu\alpha)\hat{w}(n) + \frac{u(n)}{u^H(n)u(n)}\mu e^*(n)$$

Examples

See the `lmsadeq` and `lmsadtde` demos.

Dialog Box



FIR filter length
The length of the FIR filter.

Step-size
The step-size, usually in the range (0, 2). Tunable.

Initial value of filter taps
The initial FIR filter coefficients.

Leakage factor
The leakage factor, in the range [0, 1]. Tunable.

Use normalization
Select this check box to compute the filter-tap estimate using the normalized equations.

Adapt input
Enables the Adapt port when selected.

References

Haykin, S. *Adaptive Filter Theory*. 3rd ed. Englewood Cliffs, NJ: Prentice Hall, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Kalman Adaptive Filter	Signal Processing Blockset
RLS Adaptive Filter	Signal Processing Blockset

See "Adaptive Filters" for related information.

LMS Filter

Purpose

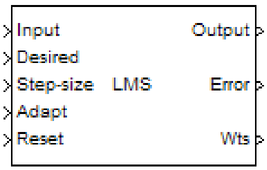
Compute filtered output, filter error, and filter weights for given input and desired signal using LMS adaptive filter algorithm

Library

Filtering / Adaptive Filters

`dspadpt3`

Description



The LMS Filter block can implement an adaptive FIR filter using five different algorithms. The block estimates the filter weights, or coefficients, needed to minimize the error, *e(n)*, between the output signal *y(n)* and the desired signal, *d(n)*. Connect the signal you want to filter to the Input port. This input signal can be a sample-based scalar or a single-channel frame-based signal. Connect the desired signal to the Desired port. The desired signal must have the same data type, frame status, complexity, and dimensions as the input signal. The Output port outputs the filtered input signal, which is the estimate of the desired signal. The output of the Output port has the same frame status as the input signal. The Error port outputs the result of subtracting the output signal from the desired signal.

When you select `LMS` for the **Algorithm** parameter, the block calculates the filter weights using the least mean-square (LMS) algorithm. This algorithm is defined by the following equations.

$$
\begin{aligned}
y(n) &= \mathbf{w}^T(n-1)\mathbf{u}(n) \\
e(n) &= d(n) - y(n) \\
\mathbf{w}(n) &= \mathbf{w}(n-1) + f(\mathbf{u}(n), e(n), \mu)
\end{aligned}
$$

The weight update function for the LMS adaptive filter algorithm is defined as

$$f(\mathbf{u}(n), e(n), \mu) = \mu e(n)\mathbf{u}^*(n)$$

The variables are as follows.

Variable	Description
n	The current time index
$\mathbf{u}(n)$	The vector of buffered input samples at step n
$\mathbf{u}^*(n)$	The complex conjugate of the vector of buffered input samples at step n
$\mathbf{w}(n)$	The vector of filter weight estimates at step n
$y(n)$	The filtered output at step n
$e(n)$	The estimation error at step n
$d(n)$	The desired response at step n
μ	The adaptation step size

When you select `Normalized LMS` for the **Algorithm** parameter, the block calculates the filter weights using the normalized LMS algorithm. The weight update function for the normalized LMS algorithm is defined as

$$f(\mathbf{u}(n), e(n), \mu) = \mu e(n)\frac{\mathbf{u}^*(n)}{\varepsilon + \mathbf{u}^H(n)\mathbf{u}(n)}$$

To overcome potential numerical instability in the update of the weights, a small positive constant, epsilon, has been added in the denominator. For double-precision floating-point input, epsilon is 2.2204460492503131e-016. For single-precision floating-point input, epsilon is 1.192092896e-07. For fixed-point input, epsilon is 0.

When you select `Sign-Error LMS` for the **Algorithm** parameter, the block calculates the filter weights using the LMS algorithm equations.

However, each time the block updates the weights, it replaces the error term *e(n)* with `+1` when the error term is positive, `-1` when it is negative, or `0` when it is zero.

When you select `Sign-Data LMS` for the **Algorithm** parameter, the block calculates the filter weights using the LMS algorithm equations. However, each time the block updates the weights, it replaces each sample of the input vector $\boldsymbol{u}(n)$ with `+1` when the input sample is positive, `-1` when it is negative, or `0` when it is zero.

When you select `Sign-Sign LMS` for the **Algorithm** parameter, the block calculates the filter weights using the LMS algorithm equations. However, each time the block updates the weights, it replaces the error term *e(n)* with `+1` when the error term is positive, `-1` when it is negative, or `0` when it is zero. It also replaces each sample of the input vector $\boldsymbol{u}(n)$ with `+1` when the input sample is positive, `-1` when it is negative, or `0` when it is zero.

Use the **Filter length** parameter to specify the length of the filter weights vector.

The **Step size (mu)** parameter corresponds to μ in the equations. For convergence of the normalized LMS equations, $0<\mu<2$. You can either specify a step size using the input port, Step-size, or by entering a value in the Block Parameters: LMS Filter dialog.

Use the **Leakage factor (0 to 1)** parameter to specify the leakage factor $1 - \mu\alpha$ where $0 < 1 - \mu\alpha \le 1$ in the leaky LMS algorithm shown below.

$$\mathbf{w}(n) = (1-\mu\alpha)\mathbf{w}(n-1) + f\left(\mathbf{u}(n), e(n), \mu\right)$$

When you select `LMS` from the **Algorithm** list, the weight update function in the above equation is the LMS weight update function. When you select `Normalized LMS` from the **Algorithm** list, the weight update function in the above equation is the normalized LMS weight update function.

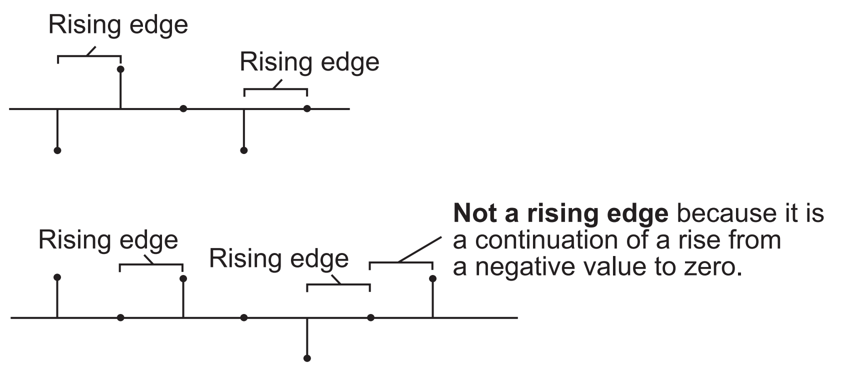
Enter the initial filter weights $\mathbf{w}(0)$ as a vector or a scalar in the **Initial value of filter weights** text box. When you enter a scalar, the block uses the scalar value to create a vector of filter weights. This vector has length equal to the filter length and all of its values are equal to the scalar value.

When you select the **Adapt port** check box, an Adapt port appears on the block. When the input to this port is greater than zero, the block continuously updates the filter weights. When the input to this port is less than or equal to zero, the filter weights remain at their current values.

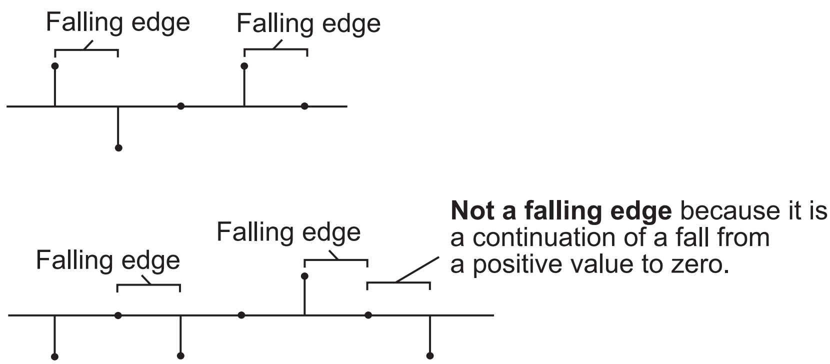
When you want to reset the value of the filter weights to their initial values, use the **Reset port** parameter. The block resets the filter weights whenever a reset event is detected at the Reset port. The reset signal rate must be the same rate as the data signal input.

From the **Reset port** list, select `None` to disable the Reset port. To enable the Reset port, select one of the following from the **Reset port** list:

- `Rising edge` — Triggers a reset operation when the Reset input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers a reset operation when the Reset input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)



- `Either edge` — Triggers a reset operation when the Reset input is a `Rising edge` or `Falling edge` (as described above)
- `Non-zero sample` — Triggers a reset operation at each sample time that the Reset input is not zero

---

Note When running simulations in the Simulink® `MultiTasking` mode, sample-based reset signals have a one-sample latency, and frame-based reset signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a reset event, and when it applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Select the **Output filter weights** check box to create a Wts port on the block. For each iteration, the block outputs the current updated filter weights from this port.

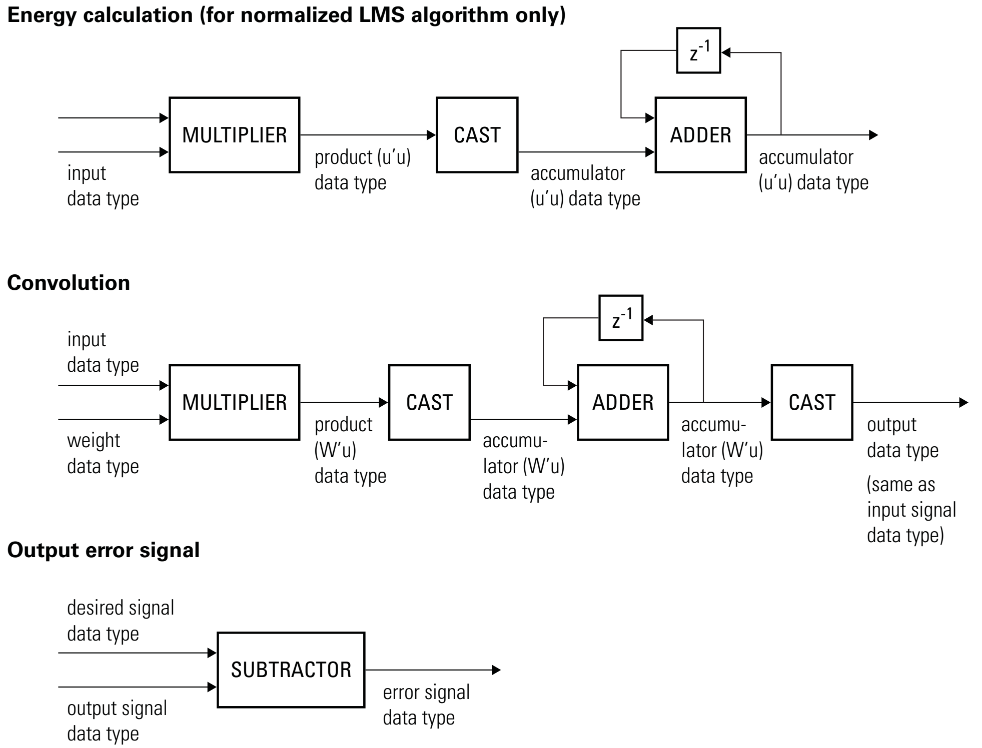
Fixed-Point Data Types

The following diagrams show the data types used within the LMS Filter block for fixed-point signals; the table summarizes the definitions of variables used in the diagrams:

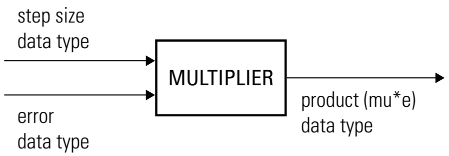
Variable	Definition
$\mathbf{u}$	Input vector
$\mathbf{W}$	Vector of filter weights
μ	Step size
e	Error
Q	Quotient, $Q = \frac{\mu \cdot e}{\mathbf{u'u}}$
Product **u'u**	Product data type in Energy calculation diagram
Accumulator **u'u**	Accumulator data type in Energy calculation diagram
Product **W'u**	Product data type in Convolution diagram
Accumulator **W'u**	Accumulator data type in Convolution diagram
Product $\mu \cdot e$	Product data type in Product of step size and error diagram
Product $Q \cdot \mathbf{u}$	Product and accumulator data type in Weight update diagram. [1]

[1]The accumulator data type for this quantity is automatically set to be the same as the product data type. The minimum, maximum, and

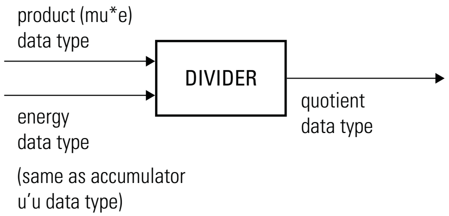
overflow information for this accumulator is logged as part of the product information. Autoscaling treats this product and accumulator as one data type.



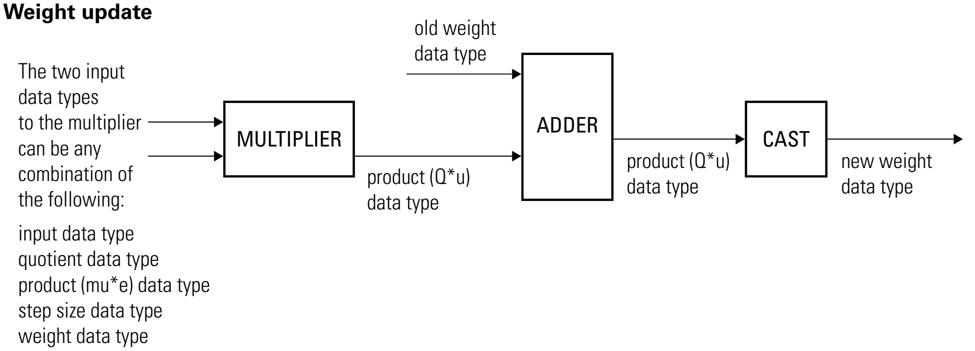
Product of step size and error (for LMS and Sign-Data LMS algorithms only)



Quotient (for normalized LMS only)



Weight update



You can set the data type of the parameters, weights, products, quotient, and accumulators in the block mask. Fixed-point inputs, outputs, and mask parameters of this block must have the following characteristics:

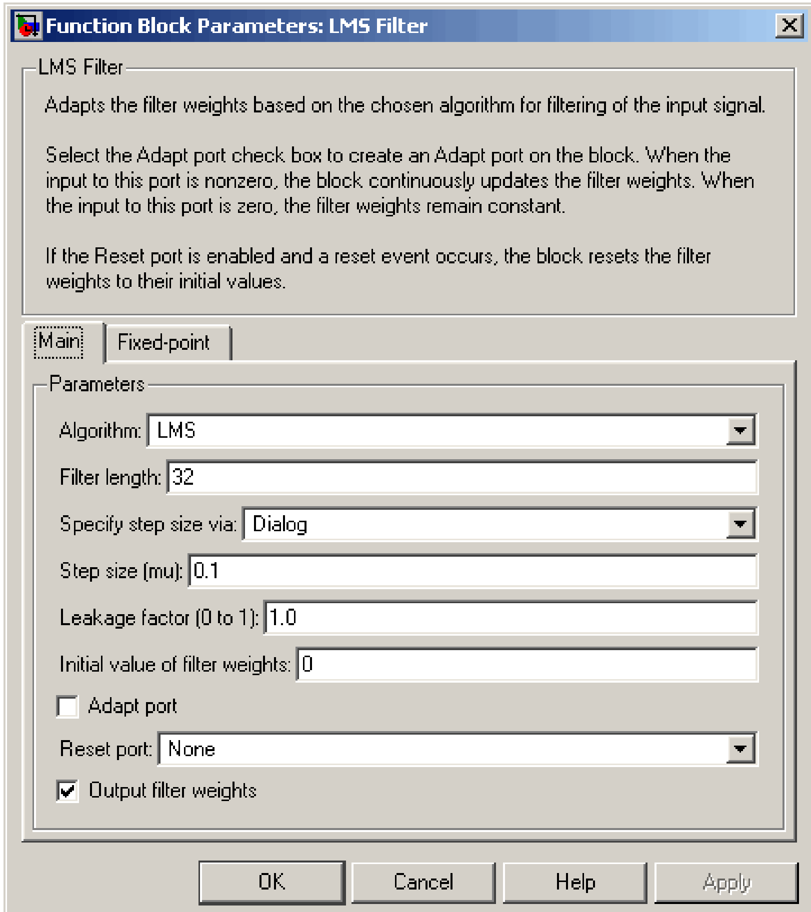
- The input signal and the desired signal must have the same word length, but their fraction lengths can differ.

- The step size and leakage factor must have the same word length, but their fraction lengths can differ.
- The output signal and the error signal have the same word length and the same fraction length as the desired signal.
- The quotient and the product output of the **u'u**, **W'u**, $\mu \cdot e$, and $Q \cdot \mathbf{u}$ operations must have the same word length, but their fraction lengths can differ.
- The accumulator data type of the **u'u** and **W'u** operations must have the same word length, but their fraction lengths can differ.

The output of the multiplier is in the product output data type if at least one of the inputs to the multiplier is real. If both of the inputs to the multiplier are complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”.

Dialog Box

The **Main** pane of the LMS Filter block dialog appears as follows.



Algorithm

Choose the algorithm used to calculate the filter weights.

Filter length

Enter the length of the FIR filter weights vector.

Specify step size via
: Select `Dialog` to enter a value for step size in the Block parameters: LMS Filter dialog. Select `Input port` to specify step size using the Step-size input port.

Step size (mu)
: Enter the step size μ. Tunable.

Leakage factor (0 to 1)
: Enter the leakage factor, $0 < 1 - \mu\alpha \leq 1$. Tunable.

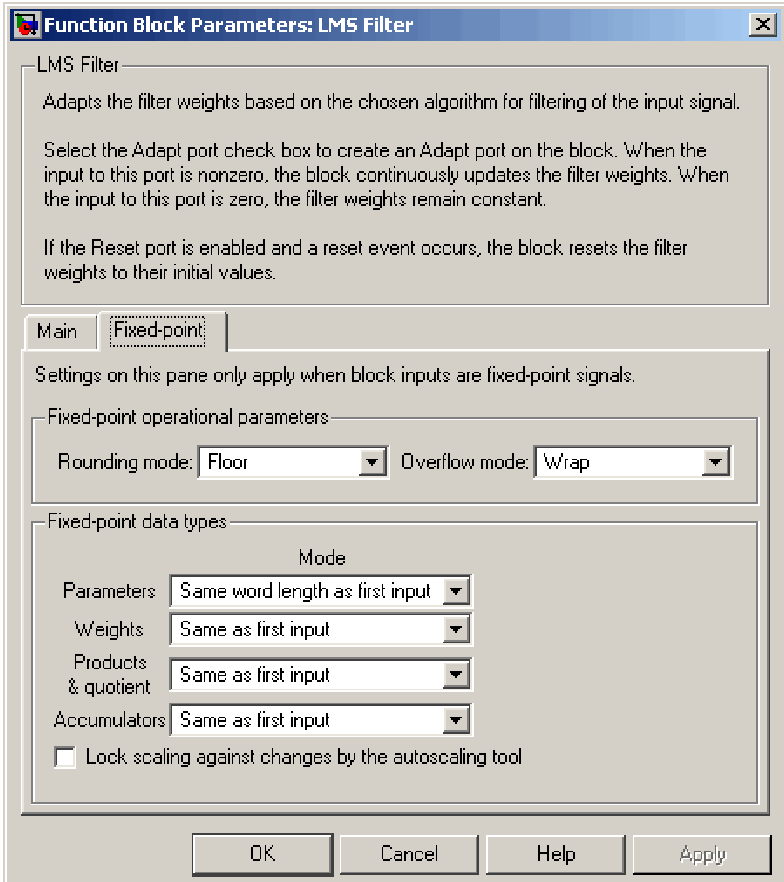
Initial value of filter weights
: Specify the initial values of the FIR filter weights.

Adapt port
: Select this check box to enable the Adapt input port.

Reset port
: Select this check box to enable the Reset input port.

Output filter weights
: Select this check box to export the filter weights from the Wts port.

The **Fixed-point** pane of the LMS Filter block dialog appears as follows.



Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

Select the overflow mode for fixed-point operations.

Parameters

This parameter is visible if, for the **Specify step size via** parameter, you choose `Dialog`. Choose how you specify the word length and the fraction length of the leakage factor and step size:

- When you select `Same word length as first input`, the word length of the leakage factor and step size match that of the first input to the block. In this mode, the fraction length of the leakage factor and step size is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select `Specify word length`, you can enter the word length of the leakage factor and step size, in bits. In this mode, the fraction length of the leakage factor and step size is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the leakage factor and step size, in bits. The leakage factor and the step size must have the same word length, but the fraction lengths can differ.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the leakage factor and step size. The leakage factor and the step size must have the same word length, but the slopes can differ. This block requires a power-of-two slope and a bias of zero.

If, for the **Specify step size via** parameter, you choose `Input port`, the word length of the leakage factor is the same as the word length of the step size input at the Step size port. The fraction length of the leakage factor is automatically set to the best precision possible based on the word length of the leakage factor.

Weights

Choose how you specify the word length and fraction length of the filter weights of the block:

- When you select `Same as first input`, the word length and fraction length of the filter weights match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the filter weights, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the filter weights. This block requires a power-of-two slope and a bias of zero.

Products & quotient

Choose how you specify the word length and fraction length of **u'u**, **W'u**, $\mu \cdot e$, $Q \cdot \mathbf{u}$, and the quotient, Q. Here, **u** is the input vector, **W** is the vector of filter weights, μ is the step size, *e* is the error, and *Q* is the quotient, which is defined as $Q = \dfrac{\mu \cdot e}{\mathbf{u'u}}$

- When you select `Same as first input`, the word length and fraction length of these quantities match those of the first input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of these quantities, in bits. The word length of the quantities must be the same, but the fraction lengths can differ.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of these quantities. The word length of the quantities must be the same, but the slopes can differ. This block requires a power-of-two slope and a bias of zero.

Accumulators

Use this parameter to specify how you would like to designate the word and fraction lengths of the accumulators for the **u'u** and **W'u** operations.

Note This parameter is *not* used to designate the word and fraction lengths of the accumulator for the $Q \cdot \mathbf{u}$ operation. The accumulator data type for this quantity is automatically set to be the same as the product data type. The minimum, maximum, and overflow information for this accumulator is logged as part of the product information. Autoscaling treats this product and accumulator as one data type.

See "Fixed-Point Data Types" on page 2-717 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block:

- When you select `Same as first input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulators, in bits. The word length of both the accumulators must be the same, but the fraction lengths can differ.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulators. The word length of both the accumulators must be the same, but the slopes can differ. This block requires a power-of-two slope and a bias of zero.

References

Hayes, M.H. *Statistical Digital Signal Processing and Modeling.* New York: John Wiley & Sons, 1996.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point • Fixed point
Desired	• Double-precision floating point • Single-precision floating point • Fixed point
Step-size	• Double-precision floating point • Single-precision floating point • Fixed point
Adapt	• Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Reset	• Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Output	• Double-precision floating point • Single-precision floating point • Fixed point

Port	Supported Data Types
Error	• Double-precision floating point • Single-precision floating point • Fixed point
Wts	• Double-precision floating point • Single-precision floating point • Fixed point

See Also

Kalman Adaptive Filter	Signal Processing Blockset
RLS Filter	Signal Processing Blockset
Block LMS Filter	Signal Processing Blockset
Fast Block LMS Filter	Signal Processing Blockset

See “Adaptive Filters” for related information.

Purpose

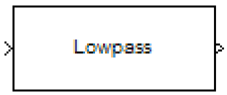
Design lowpass Filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description



This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See “Lowpass Filter Design Dialog Box — Main Pane” in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Convert linear prediction coefficients to line spectral pairs or line spectral frequencies

Library

Estimation / Linear Prediction

`dsplp`

Description

The LPC to LSF/LSP Conversion block takes a vector or matrix of linear prediction coefficients (LPCs) and converts it to a vector or matrix of line spectral pairs (LSPs) or line spectral frequencies (LSFs). When converting LPCs to LSFs, the block outputs match those of the `poly2lsf` function.

The block input must be a sample-based row vector, which is treated as a single channel, or a matrix, which is treated as a single channel per column.

The input LPCs for each channel, 1, a_1, a_2, ..., a_m, must be the denominator of the transfer function of a stable all-pole filter with the form given in the first equation of "Requirements for Valid Outputs" on page 2-731. A length-M+1 input channel yields a length-M output channel. Inputs can be sample based or frame based, but outputs are always sample based.

See other sections of this reference page to learn about how to ensure that you get valid outputs, how to detect invalid outputs, how the block computes the LSF/LSP values, and more.

Requirements for Valid Outputs

To get valid outputs, your inputs and the **Root finding coarse grid points** parameter value must meet these requirements:

- The input LPCs for each channel, 1, a_1, a_2, ..., a_m, must come from the denominator of the following transfer function, *H(z),* of a stable all-pole filter (all roots of *H(z)* must be inside the unit circle). Note that the first term in *H(z)*'s denominator must be 1. When the input LPCs do not come from a transfer function of the following form, the block outputs are invalid.

$$H(z) = \frac{1}{1 + a_1 z^{-1} + a_2 z^{-2} + \ \ldots\ + a_m z^{-m}}$$

- The **Root finding coarse grid points** parameter value must be large enough so that the block can find all the LSP or LSF values. (The output LSFs and LSPs are roots of polynomials related to the input LPC polynomial; the block looks for these roots to produce the output. For details, see "LSF and LSP Computation Method: Chebyshev Polynomial Method for Root Finding" on page 2-740.) When you do not set **Root finding coarse grid points** to a high enough value relative to the number of LPCs, the block might not find all the LSPs or LSFs and yield invalid outputs as described in "Root Finding Method Limitations: Failure to Find Roots" on page 2-743.

To learn about recognizing invalid inputs and outputs and parameters for dealing with them, see "Handling and Recognizing Invalid Inputs and Outputs" on page 2-734.

Setting Outputs to LSFs or LSPs

Set the **Output** parameter to one of the following settings to determine whether the block outputs LSFs or LSPs:

- `LSF in radians (0 pi)` — Block outputs the LSF values between 0 and π radians in increasing order. The block does not output the guaranteed LSF values, 0 and π.
- `LSF normalized in range (0 0.5)` — Block outputs normalized LSF values in increasing order, computed by dividing the LSF values between 0 and π radians by 2π. The block does not output the guaranteed normalized LSF values, 0 and 0.5.
- `LSP in range (-1 1)` — Block outputs LSP values in decreasing order, equal to the cosine of the LSF values between 0 and π radians. The block does not output the guaranteed LSP values, -1 and 1.

Adjusting Output Computation Time and Accuracy with Root Finding Parameters

The values n and k determine the block's output computation time and accuracy, where

- n is the value of the **Root finding coarse grid points** parameter (choose this value with care; see the note below).
- k is the value of the **Root finding bisection refinement** parameter.
- Decreasing the values of n and k decreases the output computation time, but also decreases output accuracy:
  - The upper bound of block's computation time is proportional to $k \cdot (n-1)$.
  - Each LSP output is within $1/(n \cdot 2^k)$ of the actual LSP value.
  - Each LSF output is within ΔLSF of the actual LSF value, LSF_{act}, where

$$\Delta LSF = \left| a\cos(LSF_{act}) - a\cos\left(LSF_{act} + 1/\left(n \cdot 2^k\right)\right) \right|$$

Note When the value of the **Root finding coarse grid points** parameter is too small relative to the number of LPCs, the block might output invalid data as described in "Requirements for Valid Outputs" on page 2-731. Also see "Handling and Recognizing Invalid Inputs and Outputs" on page 2-734.

Notable Input and Output Properties

- To get valid outputs, your input LPCs and the value of the **Root finding coarse grid points** parameter must meet the requirements described in "Requirements for Valid Outputs" on page 2-731.
- Length-L+1 input channel yields length-L output channel

- Output is always sample based
- **Output** parameter determines the output type (see "Setting Outputs to LSFs or LSPs" on page 2-732):
  - LSFs — frequencies, w_k, where $0 < w_k < \pi$ and $w_k < w_{k+1}$
  - Normalized LSFs — $w_k / 2\pi$
  - LSPs — $\cos(w_k)$

Handling and Recognizing Invalid Inputs and Outputs

The block outputs invalid data when your input LPCs and the value of the **Root finding coarse grid points** parameter do not meet the requirements described in "Requirements for Valid Outputs" on page 2-731. The following topics describe what invalid outputs look like, and how to set the block parameters provided for handling invalid inputs and outputs:

- "What Invalid Outputs Look Like" on page 2-734
- "Parameters for Handling Invalid Inputs and Outputs" on page 2-735

What Invalid Outputs Look Like

The channels of an invalid output have the same dimensions, sizes, and frame statues as the channels of a valid output. However, invalid output channels do not contain all the LSP or LSF values. Instead, they contain none or some of the LSP and LSF values and the rest of the output is filled with place holder values (-1, 0.5, or π) depending on the **Output** parameter setting).

In short, all invalid outputs in a channel end in one of the place holder values (-1, 0.5, or π) as illustrated in the following table. To learn how to use the block's parameters for handling invalid inputs and outputs, see the next section.

Output Parameter Setting	Place Holder	Sample Invalid Outputs
`LSF in radians (0 pi)`	π	$[w_1 \quad w_2 \quad w_3 \quad \pi \quad \pi \quad \pi \quad \pi \quad \pi\ ]$

Output Parameter Setting	Place Holder	Sample Invalid Outputs
`LSF normalized in range (0 0.5)`	`0.5`	$\begin{bmatrix} w_1 \\ w_2 \\ 0.5 \end{bmatrix}$
`LSP in range (-1 1)`	`-1`	$\begin{bmatrix} \cos(w_{13}) \\ \cos(w_{23}) \\ -1 \\ -1 \\ -1 \end{bmatrix}$

Parameters for Handling Invalid Inputs and Outputs

You must set how the block handles invalid inputs and outputs by setting these parameters:

- **Show output validity status (1=valid, 0=invalid)** — Set this parameter to activate a second output port that outputs a vector with one Boolean element per channel; `1` when the output of the corresponding channel is valid, and `0` when the output is invalid. The LSF and LSP outputs are invalid when the block fails to find all the LSF or LSP values or when the input LPCs are unstable (for details, see "Requirements for Valid Outputs" on page 2-731). See the previous section to learn how to recognize invalid outputs.
- **If current output is invalid, overwrite with previous output** — Select this check box to cause the block to overwrite invalid outputs with the previous output. When you set this parameter you also need to consider these parameters:
  - **When first output is invalid, overwrite with user-defined values** — When the first input is unstable, you can overwrite the invalid first output with either

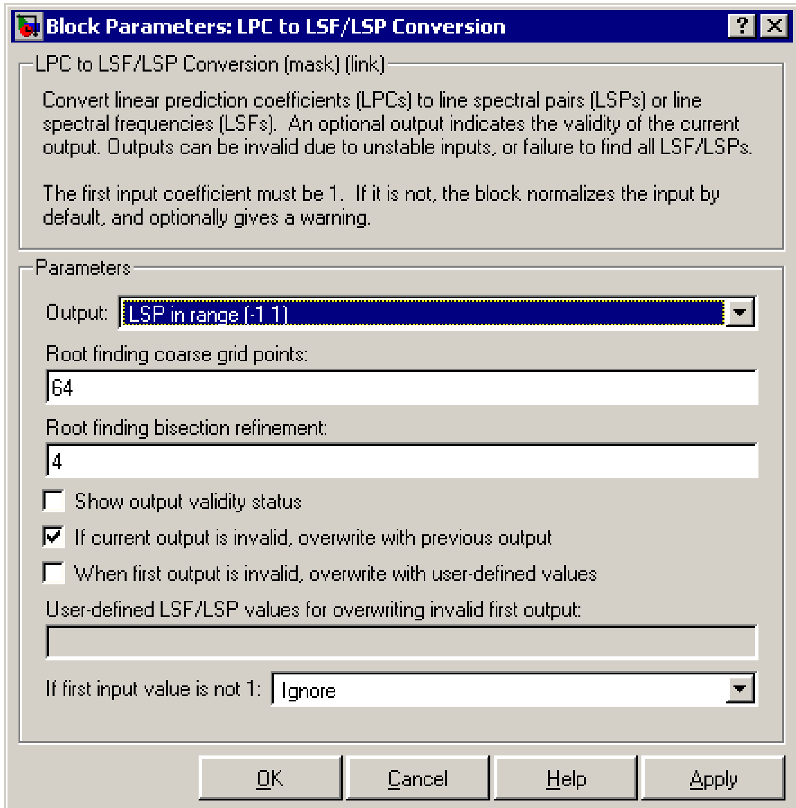
- The default values, by clearing this check box
- Values you specify, by selecting this check box

  The default initial overwrite values are the LSF or LSP representations of an all-pass filter. The vector that is used to overwrite invalid output is stored as an internal state.

  - **User-defined LSP/LSF values for overwriting invalid first output** — Specify a vector of values for overwriting an invalid first output if you selected the **When first output is invalid, overwrite with user-defined values** parameter. For multichannel inputs, provide a matrix with the same number of channels as the input, or one vector that will be applied to every channel. The vector or matrix of LSP/LSF values you specify should have the same dimension, size, and frame status as the other outputs.

- **If first input value is not 1** — The block output in any channel is invalid when the first coefficient in an LPC vector is not 1; this parameter determines what the block does when given such inputs:
  - `Ignore` — Proceed with computations as if the first coefficient is 1.
  - `Normalize` — Divide the input LPCs by the value of the first coefficient before computing the output.
  - `Normalize and warn` — In addition to `Normalize`, display a warning message at the MATLAB® command line.
  - `Error` — Stop the simulation and display an error message at the MATLAB command line.

Dialog Box



Output

Specifies whether to convert the input linear prediction polynomial coefficients (LPCs) to `LSP in range (-1 1)`, `LSF in radians (0 pi)`, or `LSF normalized in range (0 0.5)`. See "Setting Outputs to LSFs or LSPs" on page 2-732 for descriptions of the three settings.

Root finding coarse grid points

The value *n*, where the block divides the interval (-1, 1) into *n* subintervals of equal length, and looks for roots (LSP values) in each subinterval. You must pick *n* large enough or the block output might be invalid as described in "Requirements for Valid Outputs" on page 2-731. To learn how the block uses this parameter to compute the output, see "LSF and LSP Computation Method: Chebyshev Polynomial Method for Root Finding" on page 2-740. Also see "Adjusting Output Computation Time and Accuracy with Root Finding Parameters" on page 2-733. Tunable.

Root finding bisection refinement

The value k, where each LSP output is within $1/(n \cdot 2^k)$ of the actual LSP value, where n is the value of the **Root finding coarse grid points** parameter. To learn how the block uses this parameter to compute the output, see "LSF and LSP Computation Method: Chebyshev Polynomial Method for Root Finding" on page 2-740. Also see "Adjusting Output Computation Time and Accuracy with Root Finding Parameters" on page 2-733. Tunable.

Show output validity status

Set this parameter to activate a second output port that outputs a vector with one Boolean element per channel; 1 when the output of the corresponding channel is valid, and 0 when the output is invalid. The LSF and LSP outputs are invalid when the block fails to find all the LSF or LSP values or when the input LPCs are unstable (for details, see "Requirements for Valid Outputs" on page 2-731).

If current output is invalid, overwrite with previous output

Selecting this check box causes the block to overwrite invalid outputs with the previous output. Setting this parameter activates other parameters for taking care of initial overwrite values (when the very first output of the block is invalid). For more information, see "Parameters for Handling Invalid Inputs and Outputs" on page 2-735.

When first output is invalid, overwrite with user-defined values

When the first input is unstable, you can overwrite the invalid first output with either

- The default values, by clearing this check box

- Values you specify, by selecting this check box

The default initial overwrite values are the LSF or LSP representations of an all-pass filter. The vector that is used to overwrite invalid output is stored as an internal state. For more information, see "Parameters for Handling Invalid Inputs and Outputs" on page 2-735.

User-defined LSP/LSF values for overwriting invalid first output

Specify a vector of values for overwriting an invalid first output if you selected the **When first output is invalid, overwrite with user-defined values** parameter. For multichannel inputs, provide a matrix with the same number of channels as the input, or one vector that will be applied to every channel. The vector or matrix of LSP/LSF values you specify should have the same dimension, size, and frame status as the other outputs.

If first input value is not 1

Determines what the block does when the first coefficient of an input is not `1`. The block can either proceed with computations as when the first coefficient is 1 (`Ignore`); divide the input LPCs by the value of the first coefficient before computing the output (`Normalize`); in addition to `Normalize`, display a warning message at the MATLAB command line (`Normalize and warn`); stop the simulation and display an error message at the MATLAB command line (`Error`). For more information, see "Parameters for Handling Invalid Inputs and Outputs" on page 2-735.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Boolean — Supported only by the optional output port that appears when you set the parameter, **Show output validity status (1=valid, 0=invalid)**

References

Kabal, P. and Ramachandran, R. "The Computation of Line Spectral Frequencies Using Chebyshev Polynomials."*IEEE Transactions* *on Acoustics, Speech, and Signal Processing*, Vol. ASSP-34 No. 6, December 1986. pp. 1419-1426.

Theory

LSF and LSP Computation Method: Chebyshev Polynomial Method for Root Finding

Note To learn the principles on which the block's LSP and LSF computation method is based, see the reference listed in "References" on page 2-739.

To compute LSP outputs for each channel, the block relies on the fact that LSP values are the roots of two particular polynomials related to the input LPC polynomial; the block finds these roots using the Chebyshev polynomial root finding method, described next. To compute LSF outputs, the block computes the arc cosine of the LSPs, outputting values ranging from 0 to π radians.

Root Finding Method

LSPs, which are the roots of two particular polynomials, always lie in the range (-1, 1). (The guaranteed roots at 1 and -1 are factored out.) The block finds the LSPs by looking for a sign change of the two polynomials' values between points in the range (-1, 1). The block searches a maximum of $k(n-1)$ points, where

- n is the value of the **Root finding coarse grid points** parameter.
- k is the value of the **Root finding bisection refinement** parameter.

The block's method for choosing which points to check consists of the following two steps:

1. **Coarse Root Finding** —- The block divides the interval [-1, 1] into n intervals, each of length $2/n$, and checks the signs of both polynomials' values at the endpoints of the intervals. The block starts checking signs at 1, and continues checking signs at $1 - 4/n$, $1 - 6/n$,

and so on at steps of length $2/n$, outputting any point if it is a root. The block stops searching in these situations:

a The block finds a sign change of a polynomial's values between two adjacent points. An interval containing a sign change is guaranteed to contain a root, so the block further searches the interval as described in Step 2, Root Finding Refinement.

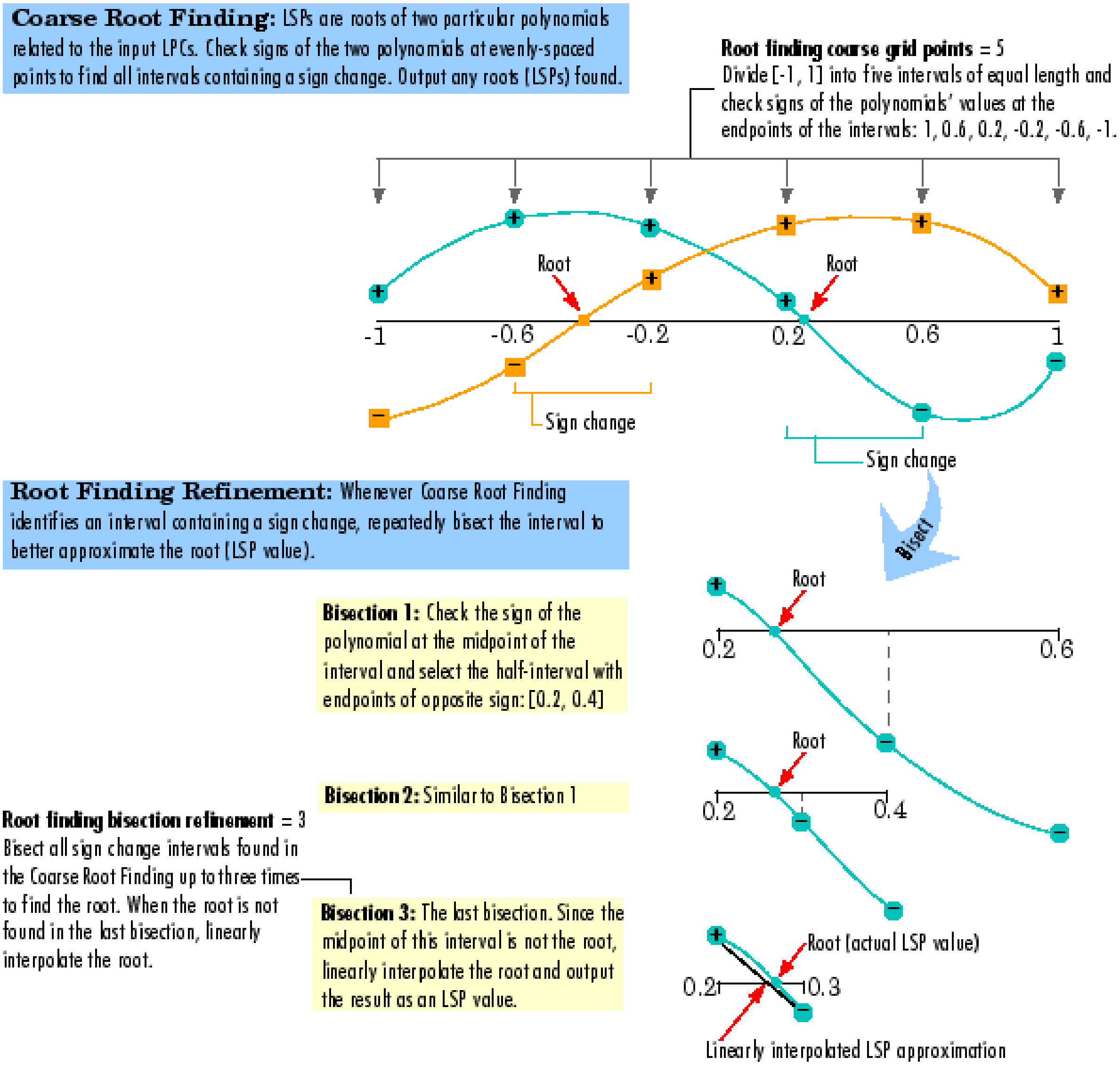
b The block finds and outputs all M roots (given a length-M+1 LPC input).

c The block fails to find all M roots and yields invalid outputs as described in "Handling and Recognizing Invalid Inputs and Outputs" on page 2-734.

2 **Root Finding Refinement** — When the block finds a sign change in an interval, $[a, b]$, it searches for the root guaranteed to lie in the interval by following these steps:

a **Check if Midpoint Is a Root** — The block checks the sign of the midpoint of the interval $[a, b]$. The block outputs the midpoint if it is a root, and continues Step 1, Coarse Root Finding, at the next point, $a - 2/n$. Otherwise, the block selects the half-interval with endpoints of opposite sign (either $[a, (a + b)/2]$ or $[(a + b)/2, b]$) and executes Step 2b, Stop or Continue Root Finding Refinement.

b **Stop or Continue Root Finding Refinement** — When the block has repeated Step 2a k times (k is the value of the **Root finding bisection refinement** parameter), the block linearly interpolates the root by using the half-interval's endpoints, outputs the result as an LSP value, and returns to Step 1, Coarse Root Finding. Otherwise, the block repeats Step 2a using the half-interval.



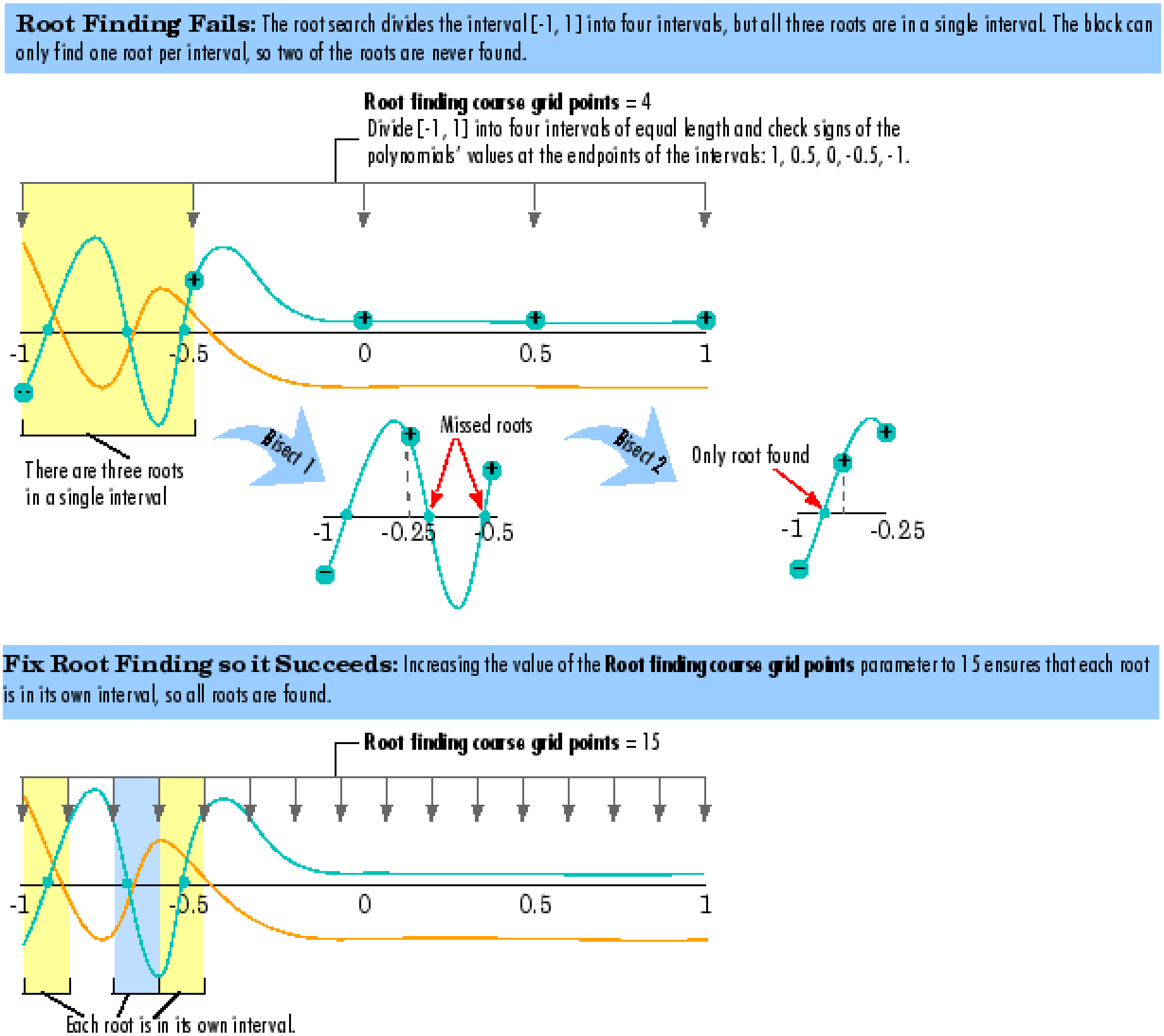
Coarse Root Finding and Root Finding Refinement

Root Finding Method Limitations: Failure to Find Roots

The block root finding method described above can fail, causing the block to produce invalid outputs (for details on invalid outputs, see "Handling and Recognizing Invalid Inputs and Outputs" on page 2-734).

In particular, the block can fail to find some roots if the value of the **Root finding coarse grid points** parameter, n, is too small. If the polynomials oscillate quickly and have roots that are very close together, the root finding might be too coarse to identify roots that are very close to each other, as illustrated in Fixing a Failed Root Finding on page 2-744.

For higher-order input LPC polynomials, you should increase the **Root finding coarse grid points** value to ensure the block finds all the roots and produces valid outputs.



Fixing a Failed Root Finding

See Also

LSF/LSP to LPC Conversion	Signal Processing Blockset
LPC to/from RC	Signal Processing Blockset
LPC/RC to Autocorrelation	Signal Processing Blockset
`poly2lsf`	Signal Processing Toolbox

LSF/LSP to LPC Conversion

Purpose

Convert line spectral frequencies or line spectral pairs to linear prediction coefficients

Library

Estimation / Linear Prediction

`dsplp`

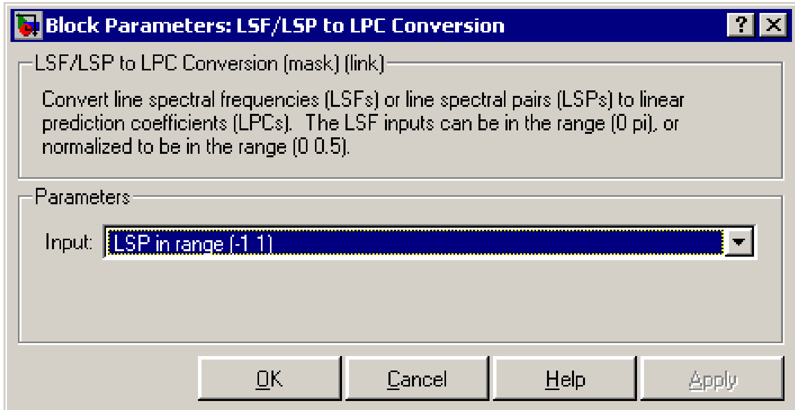
Description

The LSF/LSP to LPC Conversion block takes a vector or matrix of line spectral pairs (LSPs) or line spectral frequencies (LSFs) and converts it to a vector or matrix of linear prediction polynomial coefficients (LPCs). When converting LSFs to LPCs, the block outputs match those of the `lsf2poly` function.

The block input must be a sample-based row vector, which is treated as a single channel, or a matrix, which is treated as a single channel per column. Each input channel must be in the same format, which you specify in the **Input** parameter:

- `LSF in range (0 pi)` — Vector of LSF values between 0 and π radians in increasing order. Do not include the guaranteed LSF values, 0 and π.
- `LSF normalized in range (0 0.5)` — Vector of normalized LSF values in increasing order, (compute by dividing the LSF values between 0 and π radians by 2π). Do not include the guaranteed normalized LSF values, 0 and 0.5.
- `LSP in range (-1 1)` — Vector of LSP values in decreasing order, equal to the cosine of the LSF values between 0 and π radians. Do not include the guaranteed LSP values, -1 and 1.

Dialog Box



Input

Specifies whether to convert `LSP in range (-1 1)`, `LSF in range (0 pi)`, or `LSF normalized in range (0 0.5)` to linear prediction coefficients (LPCs).

Supported Data Types

- Double-precision floating point
- Single-precision floating point

References

Kabal, P. and Ramachandran, R. "The Computation of Line Spectral Frequencies Using Chebyshev Polynomials." *IEEE Transactions on Acoustics, Speech, and Signal Processing*, Vol. ASSP-34 No. 6, December 1986. pp. 1419-1426.

See Also

LPC to LSF/LSP Conversion	Signal Processing Blockset
LPC to/from RC	Signal Processing Blockset
LPC/RC to Autocorrelation	Signal Processing Blockset
`lsf2poly`	Signal Processing Toolbox

Purpose

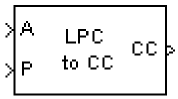
Convert linear prediction coefficients to cepstral coefficients or cepstral coefficients to linear prediction coefficients

Library

Estimation / Linear Prediction

`dsplp`

Description



LPC to/from Cepstral Coefficients

The LPC to/from Cepstral Coefficients block either converts linear prediction coefficients (LPCs) to cepstral coefficients (CCs) or cepstral coefficients to linear prediction coefficients. Set the **Type of conversion** parameter to `LPCs to cepstral coefficients` or `Cepstral coefficients to LPCs` to select the domain into which you want to convert your coefficients. The LPC port corresponds to LPCs, and the CC port corresponds to the CCs. For more information, see "Algorithm" on page 2-749.

The block input must be a sample-based row vector, which is treated as a single channel, or a matrix, which is treated as a single channel per column.

Consider a signal $x(n)$ as the input to an FIR analysis filter represented by LPCs. The output of this analysis filter, $e(n)$, is known as the prediction error signal. The power of this error signal is denoted by P, the prediction error power.

When you select `LPCs to cepstral coefficients` from the **Type of conversion** list, you can specify the prediction error power in two ways. From the **Specify P** list, choose `via input port` to input the prediction error power using input port P. The input to the port must be a vector with length equal to the number of input channels. Select `assume P equals 1` to set the prediction error power equal to 1 for all channels.

When you select `LPCs to cepstral coefficients` from the **Type of conversion** list, the **Output size same as input size** check box appears. When you select this check box, the length of the input vector of LPCs is equal to the output vector of CCs. When you do not select this check box, enter a positive scalar for the **Length of output cepstral coefficients** parameter.

When you select `LPCs to cepstral coefficients` from the **Type of conversion** list, you can use the **If first input value is not 1** parameter to specify the behavior of the block when the first coefficient of the LPC vector is not 1. The following options are available:

- `Replace it with 1` —- Changes the first value of the coefficient vector to 1. The other coefficient values are unchanged.
- `Normalize` — Divides the entire vector of coefficients by the first coefficient so that the first coefficient of the LPC vector is 1.
- `Normalize and Warn` — Divides the entire vector of coefficients by the first coefficient so that the first coefficient of the LPC vector is 1. The block displays a warning message telling you that your vector of coefficients has been normalized.
- `Error` — Displays an error telling you that the first coefficient of the LPC vector is not 1.

When you select `Cepstral coefficients to LPCs` from the **Type of conversion** list, the **Output P** check box appears on the block. Select this check box when you want to output the prediction error power from output port P.

Algorithm

The cepstral coefficients are the coefficients of the Fourier transform representation of the logarithm magnitude spectrum. Consider a sequence, $x(n)$, having a Fourier transform $X(\omega)$. The cepstrum, $c_x(n)$, is defined by the inverse Fourier transform of $C_x(\omega)$, where $C_x(\omega) = \log_e X(\omega)$. See the Real Cepstrum block reference page for information on computing cepstrum coefficients from time-domain signals.

LPC to CC

When in this mode, this block uses a recursion technique to convert LPCs to CCs. The LPC vector is defined by $\begin{bmatrix} a_0 & a_1 & a_2 & \dots & a_p \end{bmatrix}$ and the CC vector is defined by $\begin{bmatrix} c_0 & c_1 & c_2 & \dots & c_p & \dots & c_{n-1} \end{bmatrix}$. The recursion is defined by the following equations:

$$c_0 = \log_e E^2$$

$$c_m = -a_m + \frac{1}{m}\sum_{k=1}^{m-1}\left[-(m-k)\cdot a_k \cdot c_{(m-k)}\right], 1 \le m \le p$$

$$c_m = \sum_{k=1}^{p}\left[\frac{-(m-k)}{m}\cdot a_k \cdot c_{(m-k)}\right], p < m < n$$

CC to LPC

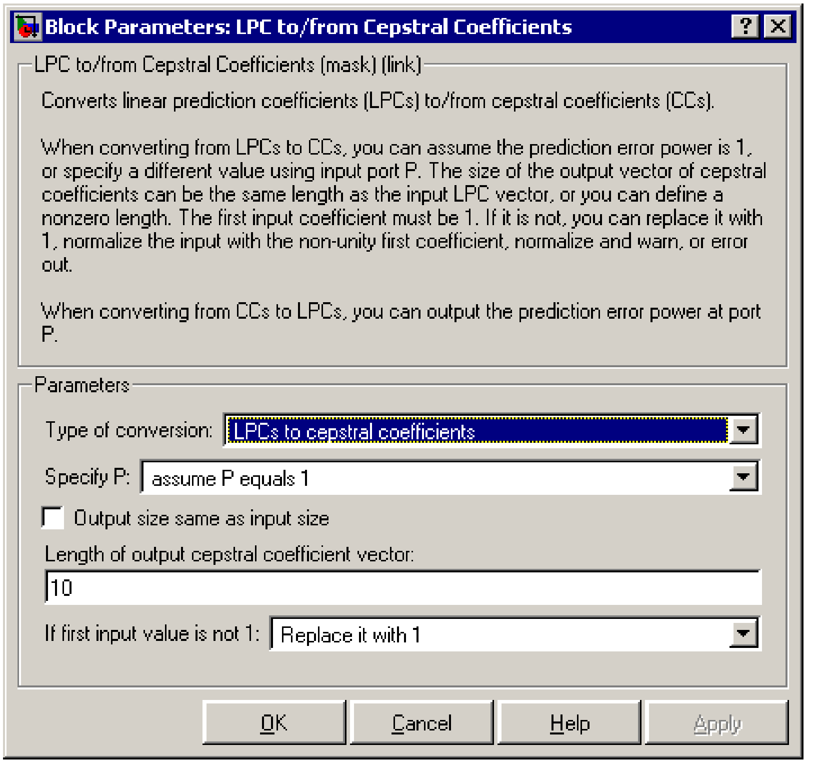
When in this mode, this block uses a recursion technique to convert CCs to LPCs. The CC vector is defined by $\left[c_0 \quad c_1 \quad c_2 \quad \ldots \quad c_p \quad \ldots \quad c_n\right]$ and the LPC vector is defined by $\left[a_0 \quad a_1 \quad a_2 \quad \ldots \quad a_p\right]$. The recursion is defined by the following equations

$$a_m = -c_m - \frac{1}{m}\sum_{k=1}^{m-1}\left[(m-k)\cdot c_{(m-k)}\cdot a_k\right]$$

$$P = \exp(C_0)$$

where $m = 1, 2, \ldots, p$.

Dialog Box



Type of conversion

Choose `LPCs to cepstral coefficients` or `Cepstral coefficients to LPCs` to specify the domain into which you want to convert your coefficients.

Specify P

Choose `via input port` to input the values of prediction error power using input port P. Select `assume P equals 1` to set the prediction error power equal to 1.

Output size same as input size

When you select this check box, the length of the input vector of LPCs is equal to the output vector of CCs.

Length of output cepstral coefficients
Enter a positive scalar that is the length of each output channel of CCs.

If first input value is not 1
Select what you would like the block to do when the first coefficient of the LPC vector is not 1. You can choose `Replace it with 1`, `Normalize`, `Normalize and Warn`, and `Error`.

Output P
Select this check box to output the prediction error power for each channel from output port P.

References

Papamichalis, Panos E. *Practical Approaches to Speech Coding*. Englewood Cliffs, NJ: Prentice Hall, 1987.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Levinson-Durbin	Signal Processing Blockset
LPC to LSF/LSP Conversion	Signal Processing Blockset
LSF/LSP to LPC Conversion	Signal Processing Blockset
LPC to/from RC	Signal Processing Blockset
LPC/RC to Autocorrelation	Signal Processing Blockset
Real Cepstrum	Signal Processing Blockset
Complex Cepstrum	Signal Processing Blockset

Purpose

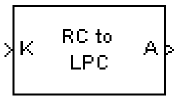
Convert linear prediction coefficients to reflection coefficients or reflection coefficients to linear prediction coefficients

Library

Estimation / Linear Prediction

`dsplp`

Description



LPC to/from RC

The LPC to/from RC block either converts linear prediction coefficients (LPCs) to reflection coefficients (RCs) or reflection coefficients to linear prediction coefficients. Set the **Type of conversion** parameter to `LPC to RC` or `RC to LPC` to select the domain into which you want to convert your coefficients. The A port corresponds to LPC coefficients, and the K port corresponds to the RC coefficients. For more information, see "Algorithm" on page 2-754.

The block input must be a sample-based row vector, which is treated as a single channel, or a matrix, which is treated as a single channel per column.

Consider a signal $x\,(n)$ as the input to an FIR analysis filter represented by LPC coefficients. The output of the this analysis filter, $e\,(n)$, is known as the prediction error signal. The power of this error signal is denoted by P. When the zero lag autocorrelation coefficient of $x\,(n)$ is one, the autocorrelation sequence and prediction error power are said to be normalized.

Select the **Output normalized prediction error power** check box to enable port P. The normalized prediction error power output at `P` is a vector with one element per input channel. Each element varies between zero and one.

Select the **Output LPC filter stability** check box to output the stability of the filter represented by the LPCs or RCs. The synthesis filter represented by the LPCs is stable when the absolute value of each of the roots of the LPC polynomial is less than one. The lattice filter represented by the RCs is stable when the absolute value of each reflection coefficient is less than 1. When the filter is stable, the block outputs a Boolean value of `1` for each input channel at the S port. When the filter is unstable, the block outputs a Boolean value of `0` for each input channel at the S port.

If first input value is not 1 parameter specifies the behavior of the block when the first coefficient of the LPC coefficient vector in any channel is not 1. The following options are available:

- `Replace it with 1` — Changes the first value of the coefficient channel to 1. The other coefficient values are unchanged.
- `Normalize` — Divides the entire channel of coefficients by the first coefficient so that the first coefficient of the LPC coefficient vector is 1.
- `Normalize and Warn` — Divides the entire channel of coefficients by the first coefficient so that the first coefficient of the LPC coefficient vector is 1. The block displays a warning message telling you that your vector of coefficients has been normalized.
- `Error` — Displays an error telling you that the first coefficient of the LPC coefficient channel is not 1.

Algorithm

LPC to RC

When in this mode, this block uses backward Levinson recursion to convert linear prediction coefficients (LPCs) to reflection coefficients (RCs). For a given Nth order LPC vector

$LPC_N = [\ 1 \quad a_{N1} \quad a_{N2} \quad \ldots \quad a_{NN}\ ]$, the block calculates the Nth reflection coefficient value using the formula $\gamma_N = -a_{NN}$. The block then finds the lower order LPC vectors , LPC_{N-1}, LPC_{N-2}, ..., LPC_1, using the following recursion.

for $p = N, N-1, ..., 2,$

$$\gamma_p = a_{pp}$$
$$F = 1 - \gamma_p{}^2$$
$$a_{p-1,m} = \frac{a_{p,m}}{F} - \frac{\gamma_p a_{p,p-m}}{F}, \quad 1 \le m < p$$

end

Finally, $\gamma_1 = -a_{11}$. The reflection coefficient vector is $[\gamma_1, \ \gamma_2, \ \ldots, \ \gamma_N]$.

RC to LPC

When in this mode, this block uses Levinson recursion to convert reflection coefficients (RCs) to linear prediction coefficients (LPCs).

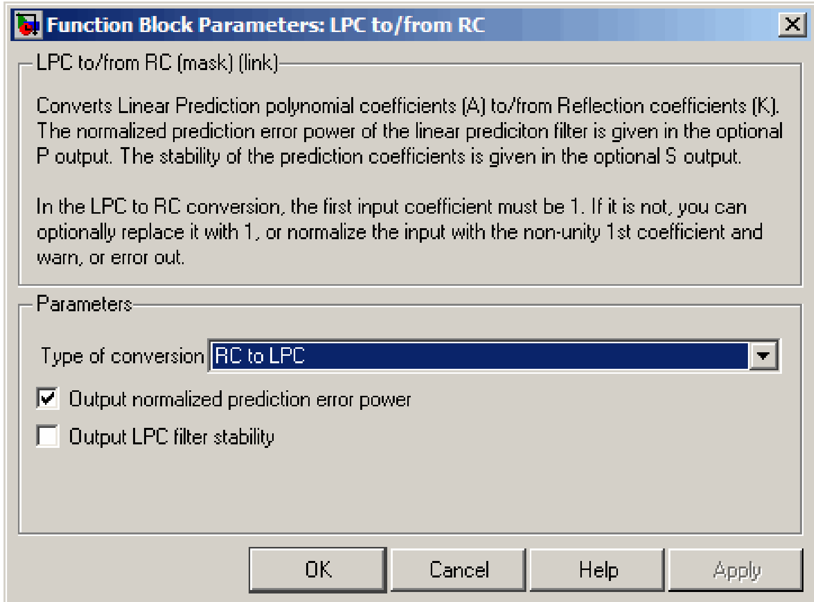
In this case, the input to the block is $RC = [\gamma_1 \quad \gamma_2 \quad \ldots \quad \gamma_N]$. The zeroth order LPC vector term is 1. Starting with this term, the block uses recursion to calculate the higher order LPC vectors, $LPC_2, LPC_3, \ldots LPC_N$, until it has calculated the entire LPC matrix.

$$LPC_{matrix} = \begin{bmatrix} LPC_0 \\ LPC_1 \\ LPC_2 \\ \cdots \\ LPC_N \end{bmatrix} = \begin{bmatrix} 1 & 0 & 0 & 0 & \cdots & 0 \\ 1 & a_{11} & 0 & 0 & \cdots & 0 \\ 1 & a_{21} & a_{22} & 0 & \cdots & 0 \\ 1 & a_{31} & a_{32} & a_{33} & \cdots & 0 \\ \cdots & \cdots & \cdots & \cdots & \cdots & \cdots \\ 1 & a_{N1} & a_{N2} & a_{N3} & \cdots & a_{NN} \end{bmatrix}$$

This LPC matrix consists of LPC vectors of order 0 through N found by using the Levinson recursion. The following are the formulas for the recursion steps, for $p = 0, 1, \ldots, N - 1$.

$$a_{p+1,m} = a_{p,m} + \gamma_{p+1} a_{p,p+1-m}, \quad 1 \le m \le p$$
$$a_{p+1,p+1} = \gamma_{p+1}$$

Dialog Box



Type of conversion

Select `LPC to RC` or `RC to LPC` to select the domain into which you want to convert your coefficients.

Output normalized prediction error power

Select this check box to output the normalized prediction error power at port P.

Output LPC filter stability

Select this check box to output the stability of the filter. When the filter represented by the LPCs or RCs is stable, the block outputs a Boolean value of `1` for each input channel at the S port. When the filter represented by the LPCs or RCs is unstable, the block outputs a Boolean value of `0` for each input channel at the S port.

If first input value is not 1

Select what you would like the block to do when the first coefficient of the LPC coefficient vector is not 1. You can choose `Replace it with 1`, `Normalize`, `Normalize and Warn`, and `Error`.

References

Makhoul, J *Linear Prediction: A tutorial review.* Proc. IEEE. 63, 63, 56 (1975).

Markel, J.D. and A. H. Gray, Jr., *Linear Prediction of Speech.* New York, Springer-Verlag, 1976.

Supported Data Types

- Double-precision floating-point
- Single-precision floating-point

See Also

Levinson-Durbin	Signal Processing Blockset
LPC to LSF/LSP Conversion	Signal Processing Blockset
LSF/LSP to LPC Conversion	Signal Processing Blockset
LPC/RC to Autocorrelation	Signal Processing Blockset

Purpose

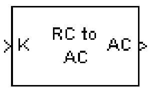
Convert linear prediction coefficients or reflection coefficients to autocorrelation coefficients

Library

Estimation / Linear Prediction

`dsplp`

Description



LPC/RC to Autocorrelation

The LPC/RC to Autocorrelation block either converts linear prediction coefficients (LPCs) to autocorrelation coefficients (ACs) or reflection coefficients (RCs) to autocorrelation coefficients (ACs). Set the **Type of conversion** parameter to `LPC to autocorrelation` or `RC to autocorrelation` to select the domain from which you want to convert your coefficients. The A port corresponds to LPC coefficients, and the K port corresponds to the RC coefficients.

The block input must be a sample-based row vector, which is treated as a single channel, or a matrix, which is treated as a single channel per column.

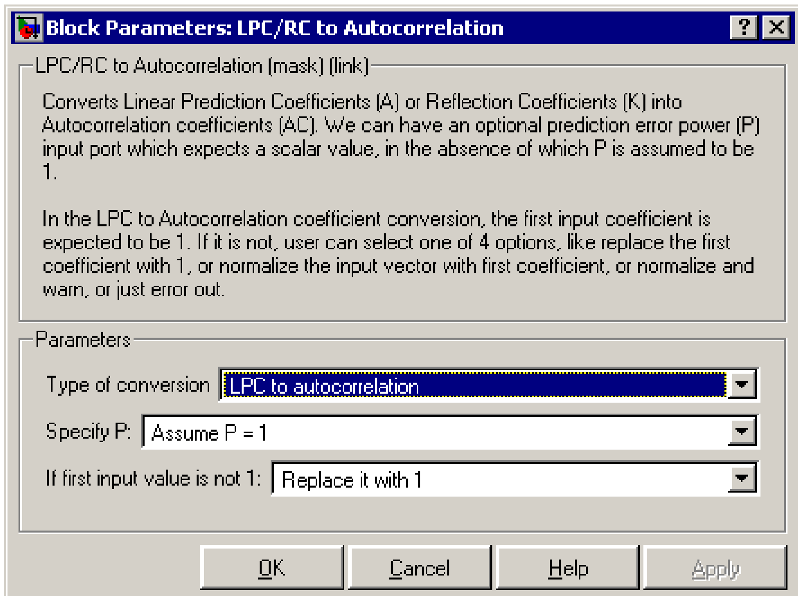
Use the **Specify P** parameter to set the value of the prediction error power. You can set this parameter to 1 by selecting `Assume P=1`. When you select `Via input port`, a P port appears on the block. You can use this port to input the value of the actual, non-unity prediction error power for each channel. The length of this vector must equal the number of channels in the input.

The **If first input value is not 1** parameter specifies the behavior of the block when the first coefficient of the LPC coefficient vector is not 1. The following options are available:

- `Replace it with 1` — The block changes the first value of the coefficient vector to 1. The rest of the coefficient values are unchanged.
- `Normalize` — The block divides the entire vector of coefficients by the first coefficient so that the first coefficient of the LPC coefficient vector is 1.

- `Normalize and Warn` — The block divides the entire vector of coefficients by the first coefficient so that the first coefficient of the LPC coefficient vector is 1. The block displays a warning message telling you that your vector of coefficients has been normalized.
- `Error` — The block displays an error telling you that the first coefficient of the LPC coefficient vector is not 1.

Dialog Box



Type of conversion

From the list select `LPC to autocorrelation` or `RC to autocorrelation` to specify the domain from which you want to convert your coefficients.

Specify P

From the list select `Assume P=1` or `Via input port` to specify the value of prediction error power.

If first input value is not 1

Select what you would like the block to do when the first coefficient of the LPC coefficient vector is not 1. You can choose `Replace it with 1`, `Normalize`, `Normalize and Warn`, and `Error`.

References

Orfanidis, S.J. *Optimum Signal Processing*. New York, McGraw-Hill, 1988.

Makhoul, J. *Linear Prediction: A tutorial review*. Proc. IEEE. 63, 63, 56 (1975).

Markel, J.D. and A. H. Gray, Jr., *Linear Prediction of Speech*. New York, Springer-Verlag, 1976.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

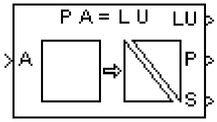
Levinson-Durbin	Signal Processing Blockset
LPC to LSF/LSP Conversion	Signal Processing Blockset
LSF/LSP to LPC Conversion	Signal Processing Blockset
LPC to/from RC	Signal Processing Blockset

Purpose Factor square matrix into lower and upper triangular components

Library Math Functions / Matrices and Linear Algebra / Matrix Factorizations

`dspfactors`

Description



LU Factorization

The LU Factorization block factors a row permutation of the square input matrix A as $A_p = L*U$, where L is the unit-lower triangular matrix, and U is the upper triangular matrix. For more information, see the `lu` function reference page in the MATLAB® documentation. The row-pivoted matrix A_p contains the rows of A permuted as indicated by the permutation index vector P.

```
Ap = A(P,:)     % Equivalent
MATLAB code
```

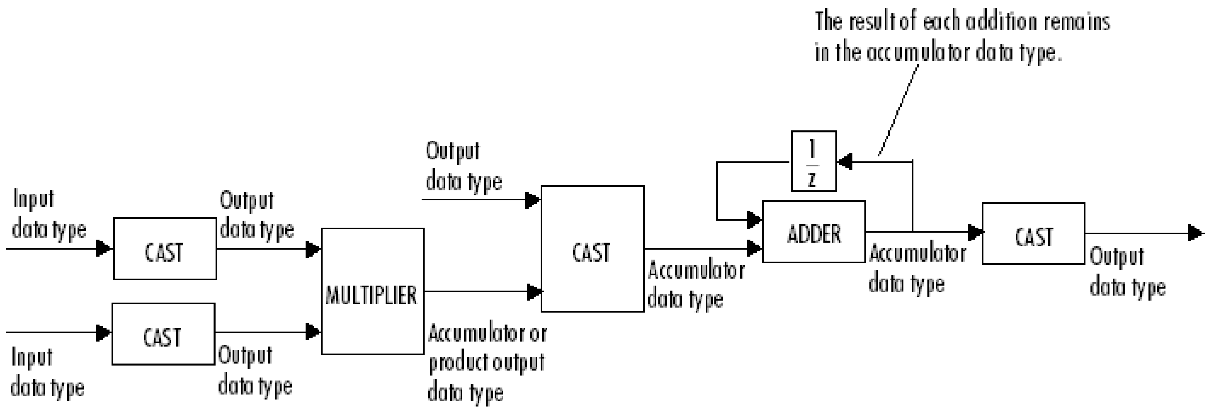
The output of the LU Factorization block at port LU is a composite matrix with lower subtriangle elements from L and upper triangle elements from U. It is always sample based. The output is not in the same form as the output of the MATLAB `lu` function. In order to convert the output of the LU Factorization block to the MATLAB form, use the following equations:

```
L = tril(LU,-1)+diag(ones(size(LU,1),1));
U = triu(LU);
```

Here, `LU` is the output of the LU Factorization block. Due to roundoff error, these equations do not produce a result that is exactly the same as the MATLAB result.

Fixed-Point Data Types

The following diagram shows the data types used within the LU Factorization block for fixed-point signals.



You can set the product output, accumulator, and output data types in the block dialog as discussed below.

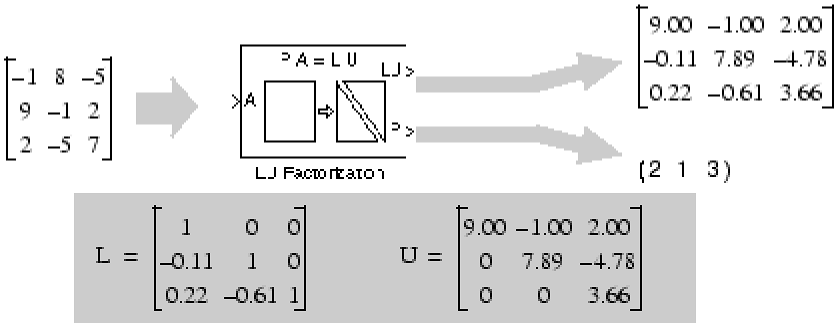
The output of the multiplier is in the product output data type when the input is real. When the input is complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see "Multiplication Data Types".

Examples

The row-pivoted matrix A_p and permutation index vector P computed by the block are shown below for 3-by-3 input matrix A.

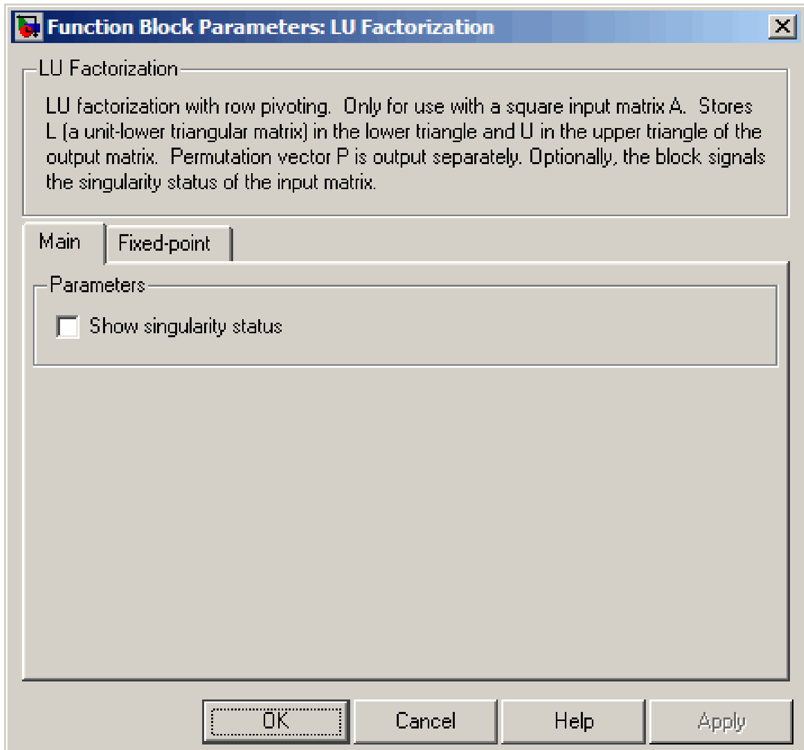
$$A = \begin{bmatrix} -1 & 8 & -5 \\ 9 & -1 & 2 \\ 2 & -5 & 7 \end{bmatrix} \qquad P = (2 \quad 1 \quad 3) \qquad A_P = \begin{bmatrix} 9 & -1 & 2 \\ -1 & 8 & -5 \\ 2 & -5 & 7 \end{bmatrix}$$

The `LU` output is a composite matrix whose lower subtriangle forms L and whose upper triangle forms U.



Dialog Box

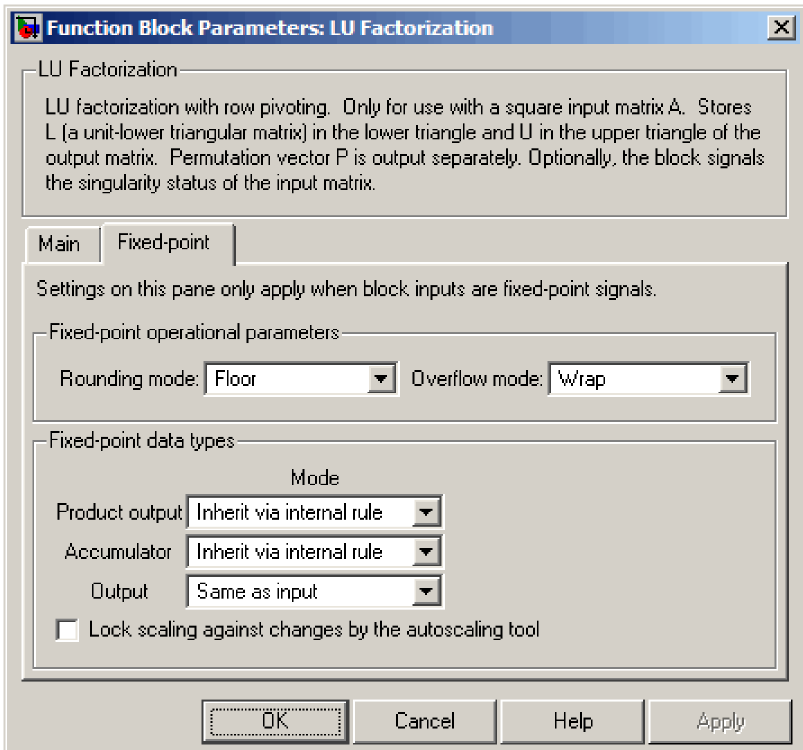
The **Main** pane of the LU Factorization block dialog appears as follows.



Show singularity status

Select to output the singularity of the input at port S, which outputs Boolean data type values of 1 or 0. An output of 1 indicates that the current input is singular, and an output of 0 indicates the current input is nonsingular.

The **Fixed-point** pane of the LU Factorization block dialog appears as follows.



Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

Select the overflow mode for fixed-point operations.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-762 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-762 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block.

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".

- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Use this parameter to specify how you would like to designate the output word and fraction lengths. See "Fixed-Point Data Types" on page 2-762 for an illustration depicting the use of the output data type in this block:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

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

See Also

Autocorrelation LPC	Signal Processing Blockset
Cholesky Factorization	Signal Processing Blockset
LDL Factorization	Signal Processing Blockset
LU Inverse	Signal Processing Blockset
LU Solver	Signal Processing Blockset
Permute Matrix	Signal Processing Blockset
QR Factorization	Signal Processing Blockset
`lu`	MATLAB

See “Matrix Factorizations” for related information.

LU Inverse

Purpose

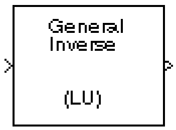
Compute inverse of square matrix using LU factorization

Library

Math Functions / Matrices and Linear Algebra / Matrix Inverses

`dspinverses`

Description

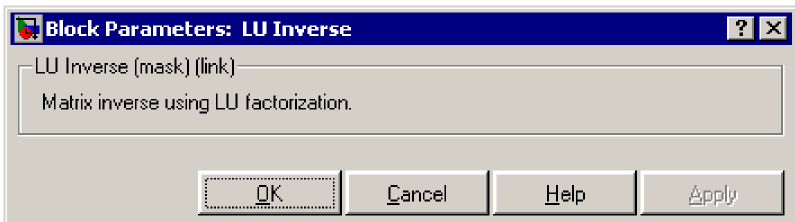


The LU Inverse block computes the inverse of the square input matrix A by factoring and inverting row-pivoted variant A_p.

$$A_p^{-1} = (LU)^{-1}$$

L is a lower triangular square matrix with unity diagonal elements, and U is an upper triangular square matrix. The block's output is A^{-1}, and is always sample based.

Dialog Box



References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Cholesky Inverse	Signal Processing Blockset
LDL Inverse	Signal Processing Blockset
LU Factorization	Signal Processing Blockset
LU Solver	Signal Processing Blockset
`inv`	MATLAB

See "Matrix Inverses" for related information.

LU Solver

Purpose

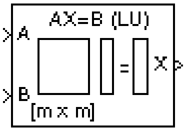
Solve AX=B for X when A is square matrix

Library

Math Functions / Matrices and Linear Algebra / Linear System Solvers

`dspsolvers`

Description



The LU Solver block solves the linear system AX=B by applying LU factorization to the M-by-M matrix at the A port. The input to the B port is the right side M-by-N matrix, B. The output is the unique solution of the equations, M-by-N matrix X, and is always sample based.

A length-M 1-D vector input for right side B is treated as an M-by-1 matrix.

Algorithm

The LU algorithm factors a row-permuted variant (A_p) of the square input matrix A as

$$A_p = LU$$

where L is a lower triangular square matrix with unity diagonal elements, and U is an upper triangular square matrix.

The matrix factors are substituted for A_p in

$$A_p X = B_p$$

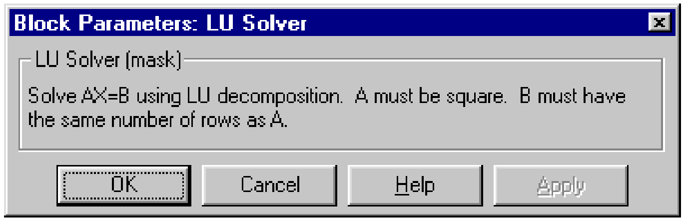
where B_p is the row-permuted variant of B, and the resulting equation

$$LUX = B_p$$

is solved for X by making the substitution $Y = UX$, and solving two triangular systems.

$$LY = B_p$$
$$UX = Y$$

Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Autocorrelation LPC	Signal Processing Blockset
Cholesky Solver	Signal Processing Blockset
LDL Solver	Signal Processing Blockset
Levinson-Durbin	Signal Processing Blockset
LU Factorization	Signal Processing Blockset
LU Inverse	Signal Processing Blockset
QR Solver	Signal Processing Blockset

See “Linear System Solvers” for related information.

Purpose

Compute nonparametric estimate of spectrum using periodogram method

Library

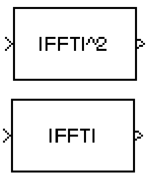
- Estimation / Power Spectrum Estimation

  `dspspect3`

- Transforms

  `dspxfrm3`

Description



The Magnitude FFT block computes a nonparametric estimate of the spectrum using the periodogram method.

When the **Output** parameter is set to `Magnitude squared`, the block output for an M-by-N input u is equivalent to

```
y = abs(fft(u,nfft)).^2       % M ≤ nfft
```

When the **Output** parameter is set to `Magnitude`, the block output for an input u is equivalent to

```
y = abs(fft(u,nfft))         % M ≤ nfft
```

When $M > N_{fft}$, the block wraps the input to N_{fft} before computing the FFT using one of the above equations:

```
y(:,k)=datawrap(u(:,k),nfft)     % 1 ≤ k ≤ N
```

Both an M-by-N frame-based matrix input and an M-by-N sample-based matrix input are treated as M sequential time samples from N independent channels. The block computes a separate estimate for each of the N independent channels and generates an N_{fft}-by-N matrix output. When you select **Inherit FFT length from input dimensions**, N_{fft} is specified by the frame size of the input, which must be a power of 2. When you do not select **Inherit FFT length from input dimensions**, N_{fft} is specified as a power of 2 by the **FFT length** parameter, and the block zero pads or wraps the input to N_{fft} before computing the FFT.

Each column of the output matrix contains the estimate of the corresponding input column's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where F_s is the signal's sample frequency. The output is always sample based.

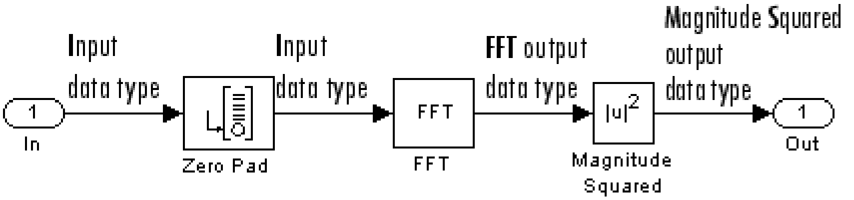
The block does not accept sample-based 1-by-N row vector inputs.

The Magnitude FFT block supports real and complex floating-point inputs. The block also supports real fixed-point inputs in both `Magnitude` and `Magnitude squared` modes, and complex fixed-point inputs in the `Magnitude squared` mode.

The Magnitude FFT block supports real and complex floating-point inputs. The block also supports real fixed-point inputs in both `Magnitude` and `Magnitude squared` modes, and complex fixed-point inputs in the `Magnitude squared` mode.

Fixed-Point Data Types

The following diagram shows the data types used within the Magnitude FFT subsystem block for fixed-point signals.



The settings for the fixed-point parameters of the FFT block in the diagram above are as follows:

- Sine table — `Same word length as input`
- Round integer calculations toward — `Floor`
- Saturate on integer overflow — unchecked
- Product output — `Inherit via internal rule`
- Accumulator — `Inherit via internal rule`
- Output — `Inherit via internal rule`

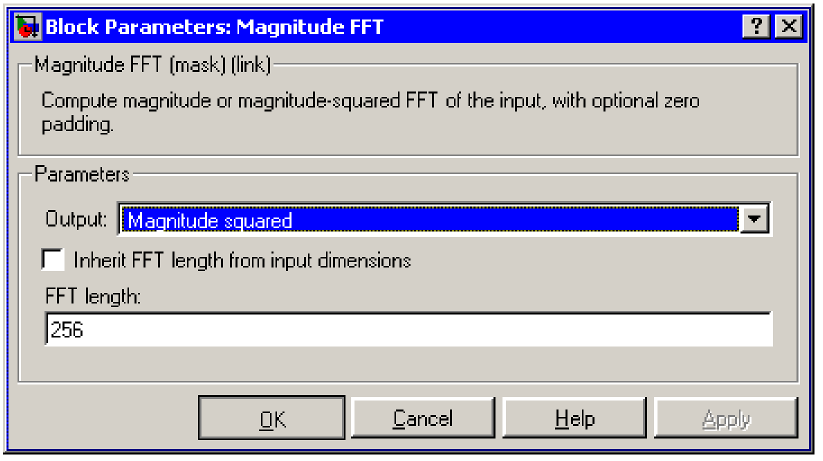
The settings for the fixed-point parameters of the Magnitude Squared block in the diagram above are as follows:

- Round integer calculations toward — `Floor`
- Saturate on integer overflow — checked
- Output — `Inherit via internal rule`

Examples

The `dspsacomp` demo compares the periodogram method with several other spectral estimation methods.

Dialog Box



Output

Specify whether the block computes the magnitude FFT or magnitude-squared FFT of the input.

Inherit FFT length from input dimensions

Select to use the input frame size as the number of data points, N_{fft}, on which to perform the FFT.

FFT length

Enter the number of data points on which to perform the FFT, N_{fft}. When N_{fft} is larger than the input frame size, each frame is zero-padded as needed. When N_{fft} is smaller than the input frame size, each frame is wrapped as needed. This parameter is enabled when you clear the **Inherit FFT length from input dimensions** check box.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1989.

Orfanidis, S. J. *Introduction to Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1995.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Supported Data Types

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

See Also

Burg Method	Signal Processing Blockset
Short-Time FFT	Signal Processing Blockset
Spectrum Scope	Signal Processing Blockset
Yule-Walker Method	Signal Processing Blockset
`pwelch`	Signal Processing Toolbox

See "Power Spectrum Estimation" for related information.

Purpose

Compute 1-norm of matrix

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description

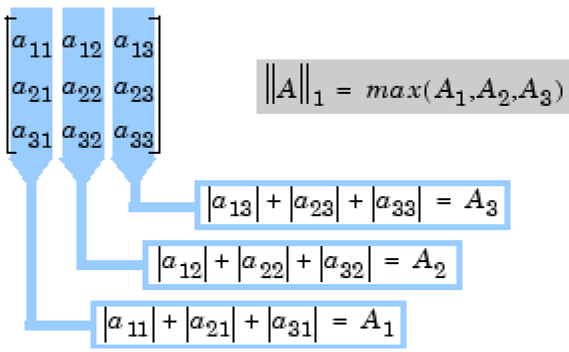
Matrix
1-Norm

The Matrix 1-Norm block computes the 1-norm, or maximum column-sum, of an M-by-N input matrix, A.

$$y = \|A\|_1 = \max_{1 \le j \le N} \sum_{i=1}^{M} |a_{ij}|$$

This is equivalent to

```
y = max(sum(abs(A)))      % Equivalent
MATLAB code
```

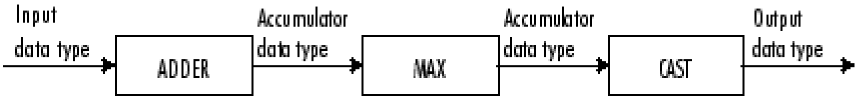


A length-M 1-D vector input is treated as an M-by-1 matrix. The output, y, is always a scalar.

The Matrix 1-Norm block supports real and complex floating-point inputs, and real fixed-point inputs.

Fixed-Point Data Types

The following diagram shows the data types used within the Matrix 1-Norm block for fixed-point signals.

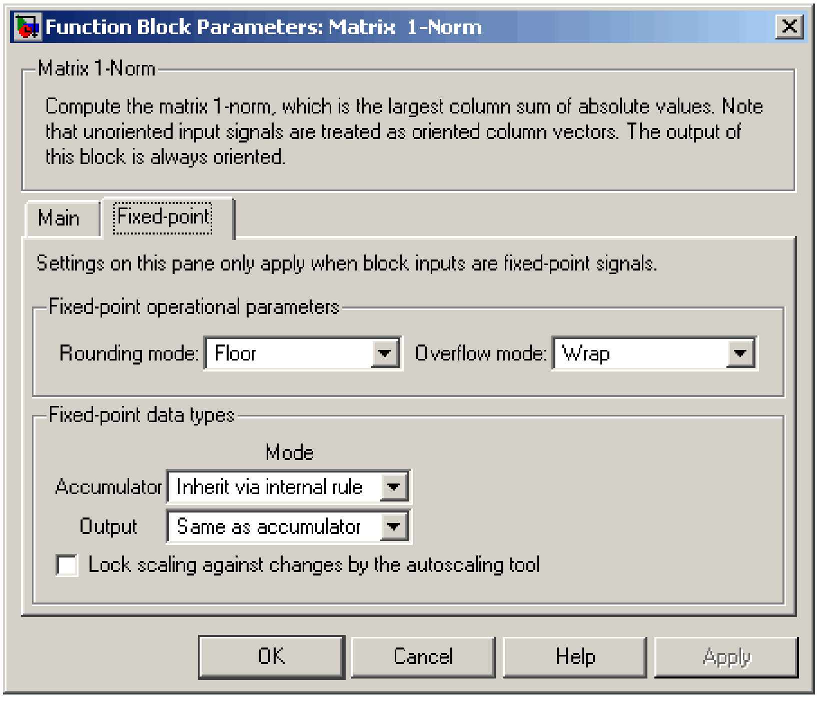


The block calculations are all done in the accumulator data type until the `max` is performed. The result is then cast to the output data type. You can set the accumulator and output data types in the block dialog as discussed in "Dialog Box" on page 2-778 below.

Dialog Box

There are no parameters on the **Main** pane of this dialog.

The **Fixed-point** pane of the Matrix 1-Norm block dialog appears as follows.



Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

Select the overflow mode for fixed-point operations.

Accumulator

Choose how you specify the word length and fraction length of the accumulator:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the word length and fraction length of the output of the block:

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

Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)
- 8-, 16-, and 32-bit signed integers

See Also

Normalization	Signal Processing Blockset
Reciprocal Condition	Signal Processing Blockset
`norm`	MATLAB

Purpose Concatenate input signals of same data type to create contiguous output signal

Library Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description Refer to the Simulink® Matrix Concatenate reference page for more information.

Matrix Exponential

Purpose

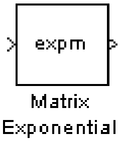
Compute matrix exponential

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

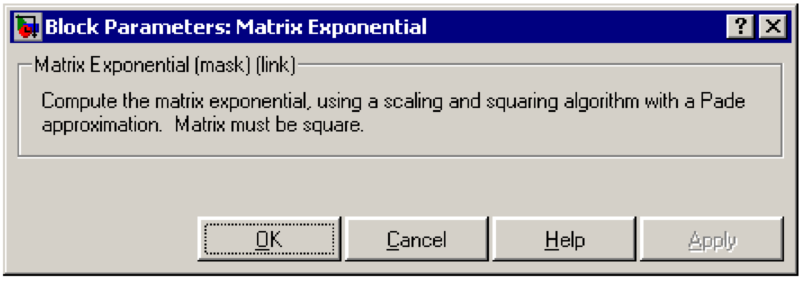
`dspmtrx3`

Description



The Matrix Exponential block computes the matrix exponential using a scaling and squaring algorithm with a Pade approximation. The input matrix must be square.

Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Array-Vector Multiply	Signal Processing Blockset
`expm`	MATLAB
Dot Product	Simulink
Matrix Product	Signal Processing Blockset
Product	Simulink

Matrix Multiply

Purpose

Multiply or divide inputs

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description

The Matrix Multiply block is an implementation of the Simulink® Product block. See Product for more information.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point
- Boolean
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Array-Vector Multiply	Signal Processing Blockset
Dot Product	Simulink
Matrix Product	Signal Processing Blockset
Product	Simulink

Purpose

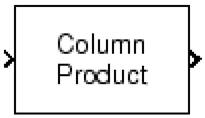
Multiply matrix elements along rows, columns, or entire input

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description



The Matrix Product block multiplies the elements of an M-by-N input matrix u along its rows, its columns, or over all its elements.

When the **Multiply over** parameter is set to `Rows`, the block multiplies across the elements of each row and outputs the resulting M-by-1 matrix. A length-N 1-D vector input is treated as a 1-by-N matrix.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix} \Rightarrow \begin{bmatrix} y_1 \\ y_2 \\ y_3 \end{bmatrix} = \begin{bmatrix} \left(\prod_{j=1}^{3} u_{1j} \right) \\ \left(\prod_{j=1}^{3} u_{2j} \right) \\ \left(\prod_{j=1}^{3} u_{3j} \right) \end{bmatrix}$$

When the **Multiply over** parameter is set to `Columns`, the block multiplies down the elements of each column and outputs the resulting 1-by-N matrix. A length-M 1-D vector input is treated as a M-by-1 matrix.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix}$$
$$\Downarrow$$
$$\begin{bmatrix} y_1 & y_2 & y_3 \end{bmatrix} = \begin{bmatrix} \left(\prod_{i=1}^{3} u_{i1} \right) & \left(\prod_{i=1}^{3} u_{i2} \right) & \left(\prod_{i=1}^{3} u_{i3} \right) \end{bmatrix}$$

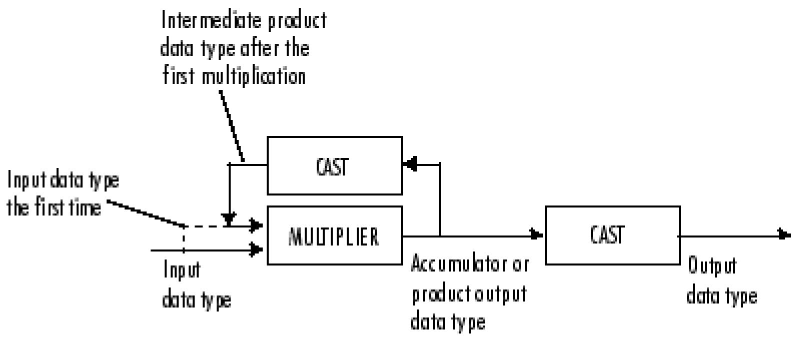
When the **Multiply over** parameter is set to `Entire input`, the block multiplies all the elements of the input together and outputs the resulting scalar.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix} \Rightarrow y = \left(\prod_{i=1}^{3}\prod_{j=1}^{3} u_{ij}\right)$$

The output of the Matrix Product block has the same frame status as the input. This block accepts real and complex fixed-point and floating-point inputs except for complex unsigned fixed-point inputs.

Fixed-Point Data Types

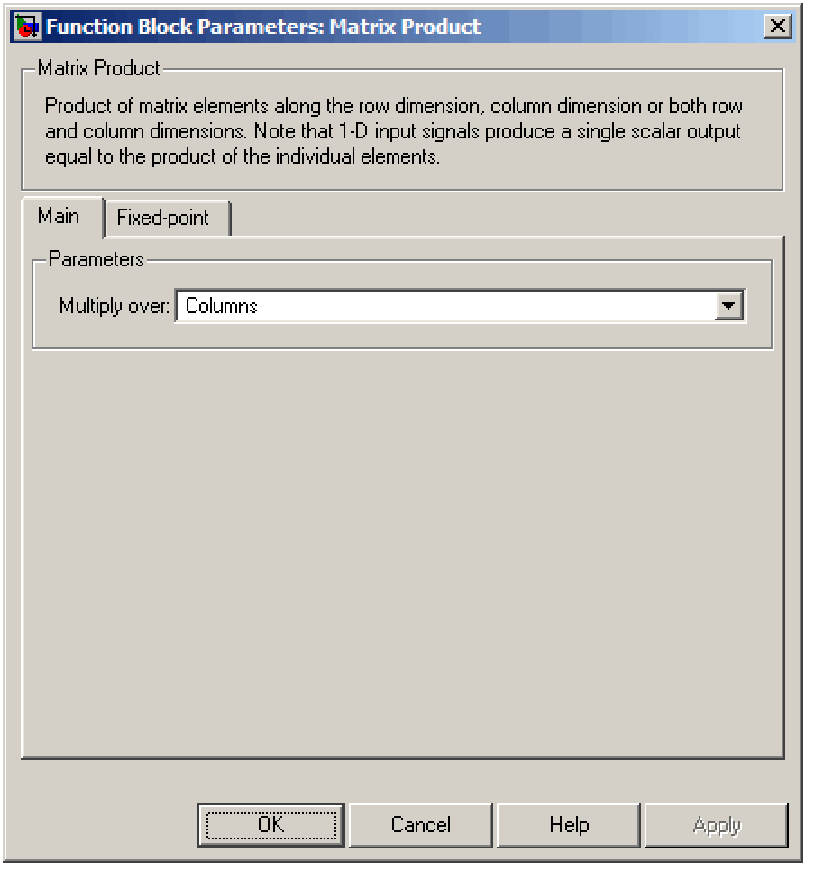
The following diagram shows the data types used within the Matrix Product block for fixed-point signals.



The output of the multiplier is in the product output data type when at least one of the inputs to the multiplier is real. When both of the inputs to the multiplier are complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”. You can set the accumulator, product output, intermediate product, and output data types in the block dialog as discussed in “Dialog Box” on page 2-787 below.

Dialog Box

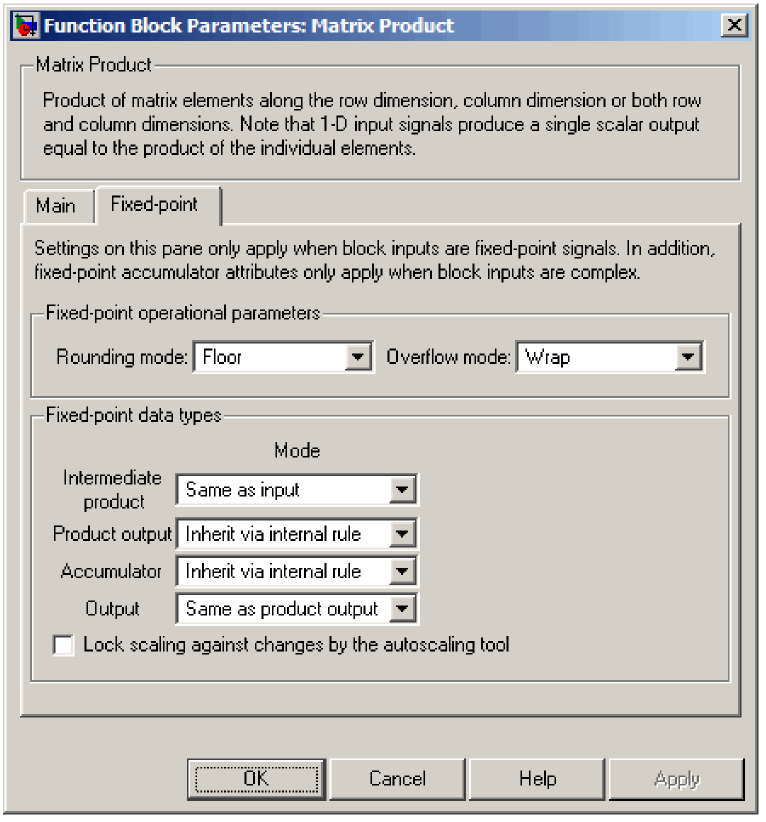
The **Main** pane of the Matrix Product block dialog appears as follows.



Multiply over

Indicate whether to multiply together the elements of each row, each column, or the entire input.

The **Fixed-point** pane of the Matrix Product block dialog appears as follows.



Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Intermediate product

As shown in "Fixed-Point Data Types" on page 2-786, the output of the multiplier is cast to the intermediate product data type before the next element of the input is multiplied into it. Use this parameter to specify how you would like to designate the intermediate product word and fraction lengths:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the intermediate product, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the intermediate product. This block requires power-of-two slope and a bias of zero.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-786 and "Multiplication Data Types" for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths. See "Fixed-Point Data Types" on page 2-786 and "Multiplication Data Types" for illustrations depicting the use of the accumulator data type in this block. Note that the accumulator data type is only used when both inputs to the multiplier are complex:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the word length and fraction length of the output of the block:

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

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

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

See Also

Array-Vector Multiply	Signal Processing Blockset
Matrix Square	Signal Processing Blockset
Matrix Sum	Signal Processing Blockset
`prod`	MATLAB

Matrix Square

Purpose

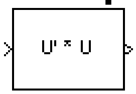
Compute square of input matrix

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description



The Matrix Square block computes the square of an M-by-N input matrix, u, by premultiplying with the Hermitian transpose.

```
y = u' * u     % Equivalent MATLAB code
```

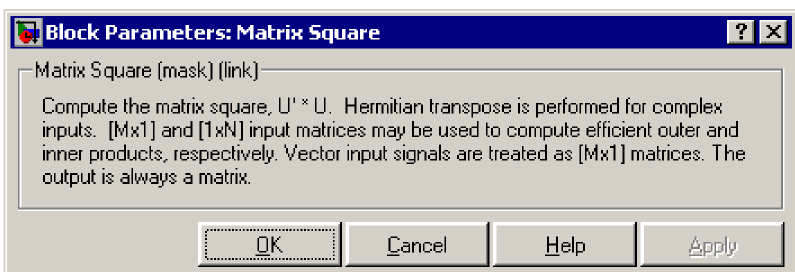
A length-M 1-D vector input is treated as an M-by-1 matrix. For both sample-based and frame-based inputs, output y is sample based with dimension N-by-N.

Applications

The Matrix Square block is useful in a variety of applications:

- General matrix squares — The Matrix Square block computes the output matrix, y, without explicitly forming u'. It is therefore more efficient than other methods for computing the matrix square.
- Sum of squares — When the input is a column vector (N=1), the block's operation is equivalent to a multiply-accumulate (MAC) process, or inner product. The output is the sum of the squares of the input, and is always a real scalar.
- Correlation matrix — When the input is a row vector (M=1), the output, y, is the symmetric autocorrelation matrix, or outer product.

Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Matrix Multiply	Signal Processing Blockset
Matrix Product	Signal Processing Blockset
Matrix Sum	Signal Processing Blockset
Transpose	Signal Processing Blockset

Matrix Sum

Purpose

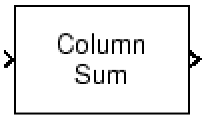
Sum matrix elements along rows, columns, or entire input

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description



The Matrix Sum block sums the elements of an M-by-N input matrix u along its rows, its columns, or over all its elements.

When the **Sum over** parameter is set to `Rows`, the block sums across the elements of each row and outputs the resulting M-by-1 matrix. A length-N 1-D vector input is treated as a 1-by-N matrix.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix} \Rightarrow \begin{bmatrix} y_1 \\ y_2 \\ y_3 \end{bmatrix} = \begin{bmatrix} \left(\sum_{j=1}^{3} u_{1j} \right) \\ \left(\sum_{j=1}^{3} u_{2j} \right) \\ \left(\sum_{j=1}^{3} u_{3j} \right) \end{bmatrix}$$

When the **Sum over** parameter is set to `Columns`, the block sums down the elements of each column and outputs the resulting 1-by-N matrix. A length-M 1-D vector input is treated as a M-by-1 matrix.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix}$$

$$\Downarrow$$

$$\begin{bmatrix} y_1 & y_2 & y_3 \end{bmatrix} = \begin{bmatrix} \left(\sum_{i=1}^{3} u_{i1} \right) & \left(\sum_{i=1}^{3} u_{i2} \right) & \left(\sum_{i=1}^{3} u_{i3} \right) \end{bmatrix}$$

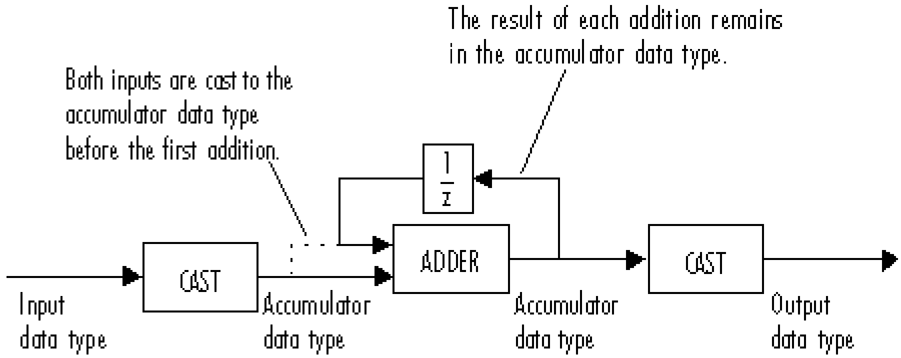
When the **Sum over** parameter is set to `Entire input`, the block sums all the elements of the input together and outputs the resulting scalar.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix} \Rightarrow y = \left(\sum_{i=1}^{3}\sum_{j=1}^{3} u_{ij}\right)$$

The output of the Matrix Sum block has the same frame status as the input. This block accepts real and complex fixed-point and floating-point inputs except for complex unsigned fixed-point inputs.

Fixed-Point Data Types

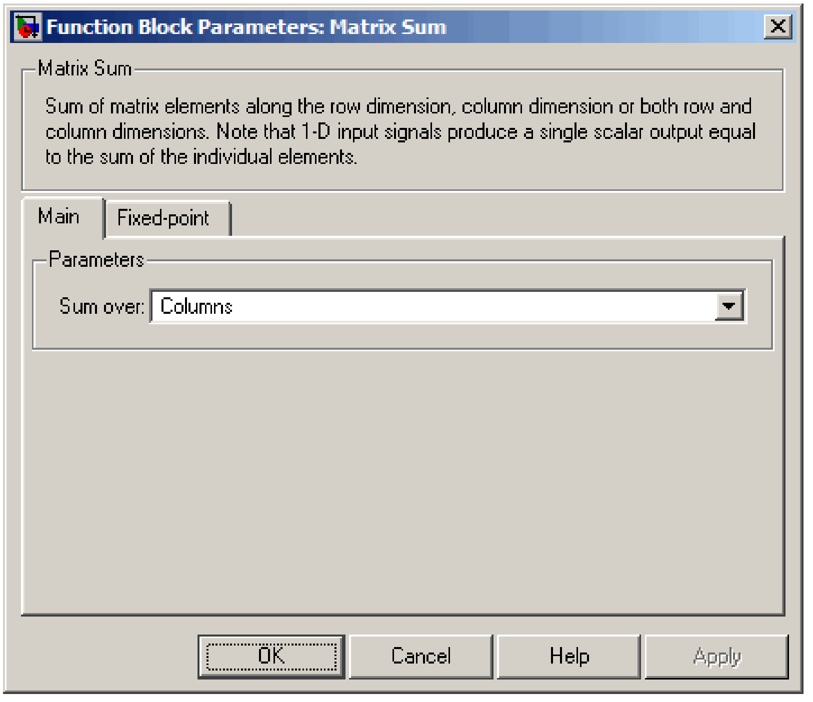
The following diagram shows the data types used within the Matrix Sum block for fixed-point signals.



You can set the accumulator and output data types in the block dialog as discussed in “Dialog Box” on page 2-796 below.

Dialog Box

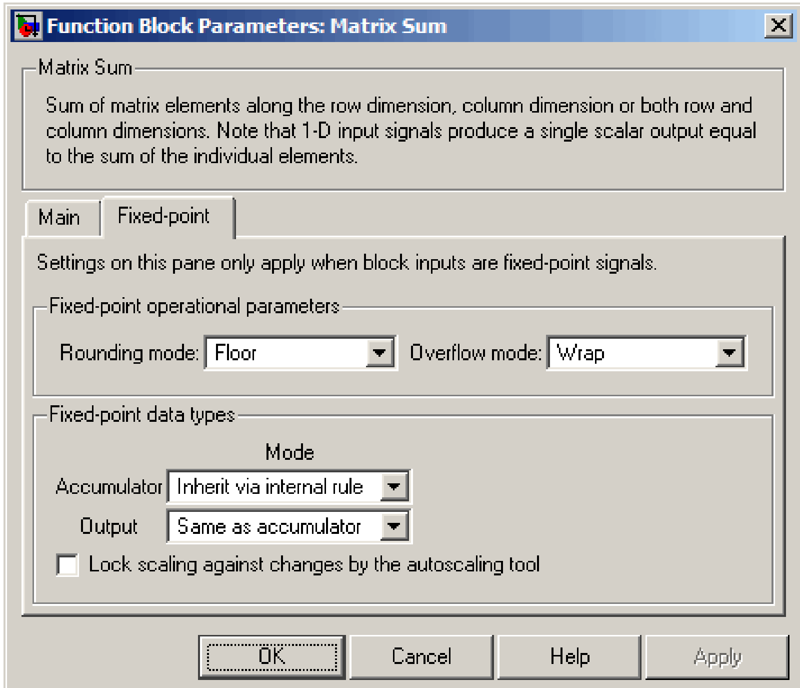
The **Main** pane of the Matrix Sum block dialog appears as follows.



Sum over

Indicate whether to sum the elements of each row, each column, or of the entire input.

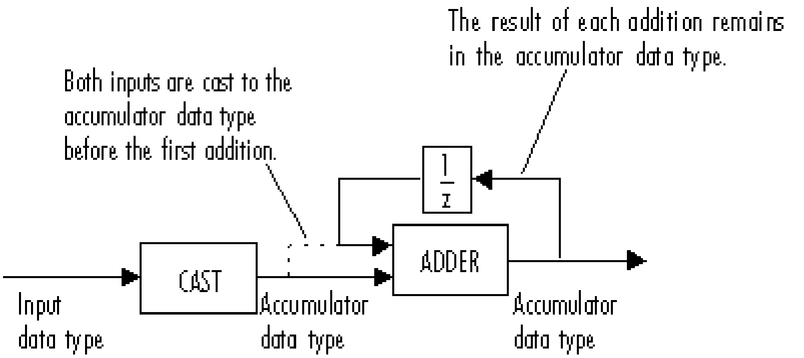
The **Fixed-point** pane of the Matrix Sum block dialog appears as follows.



Rounding mode
: Select the rounding mode for fixed-point operations.

Overflow mode
: Select the overflow mode for fixed-point operations.

Accumulator



As depicted above, the elements of the block input are cast to the accumulator data type before they are added together. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you would like to designate this accumulator word and fraction lengths:

- When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

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

Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

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

See Also

Matrix Product	Signal Processing Blockset
Matrix Multiply	Signal Processing Blockset
`sum`	MATLAB

Purpose

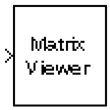
Display matrices as color images

Library

Signal Processing Sinks

`dspsnks4`

Description



The Matrix Viewer block displays an M-by-N matrix input by mapping the matrix element values to a specified range of colors. The display is updated as each new input is received. This block treats a length M 1-D vector input as an M-by-1 matrix.

Image Properties

Select the **Image Properties** tab to show the image property parameters, which control the colormap and display.

You specify the mapping of matrix element values to colors in the **Colormap matrix**, **Minimum input value**, and **Maximum input value** parameters. For a colormap with L colors, the colormap matrix has dimension L-by-3, with one row for each color and one column for each element of the RGB triple that defines the color. Examples of RGB triples are

```
[ 1   0   0 ]   (red)
[ 0   0   1 ]   (blue)
[0.8 0.8 0.8]   (light gray)
```

See the `ColorSpec` property in the MATLAB® documentation for complete information about defining RGB triples.

MATLAB provides a number of functions for generating predefined colormaps, such as `hot`, `cool`, `bone`, and `autumn`. Each of these functions accepts the colormap size as an argument, and can be used in the **Colormap matrix** parameter. For example, when you specify `gray(128)` for the **Colormap matrix** parameter, the matrix is displayed in 128 shades of gray. The color in the first row of the colormap matrix represents the value specified by the **Minimum input value** parameter, and the color in the last row represents the value specified by the **Maximum input value** parameter. Values between the minimum and maximum are quantized and mapped to the intermediate rows of the colormap matrix.

The documentation for the MATLAB `colormap` function provides complete information about specifying colormap matrices, and includes a complete list of the available colormap functions.

Axis Properties

Select the **Axis Properties** tab to show the axis property parameters, which control labeling and positioning.

The **Axis origin** parameter determines where the first element of the input matrix, $U(1,1)$, is displayed. When you specify `Upper left corner`, the matrix is displayed in matrix orientation, with $U(1,1)$ in the upper-left corner.

$$\begin{bmatrix} U_{11} & U_{12} & U_{13} & U_{14} \\ U_{21} & U_{22} & U_{23} & U_{24} \\ U_{31} & U_{32} & U_{33} & U_{34} \\ U_{41} & U_{42} & U_{43} & U_{44} \end{bmatrix}$$

When you specify `Lower left corner`, the matrix is flipped vertically to image orientation, with $U(1,1)$ in the lower-left corner.

$$\begin{bmatrix} U_{41} & U_{42} & U_{43} & U_{44} \\ U_{31} & U_{32} & U_{33} & U_{34} \\ U_{21} & U_{22} & U_{23} & U_{24} \\ U_{11} & U_{12} & U_{13} & U_{14} \end{bmatrix}$$

Axis zoom, when selected, causes the image display to completely fill the figure window. Axis titles are not displayed. This option can also be selected from the pop-up menu that is displayed when you right-click in the figure window. When **Axis zoom** is cleared, the axis labels and titles are displayed in a gray border surrounding the image axes.

Figure Window

The image title in the figure title bar is the same as the block title. The axis tick marks reflect the size of the input matrix; the *x*-axis is numbered from 1 to N (number of columns), and the *y*-axis is numbered from 1 to M (number of rows).

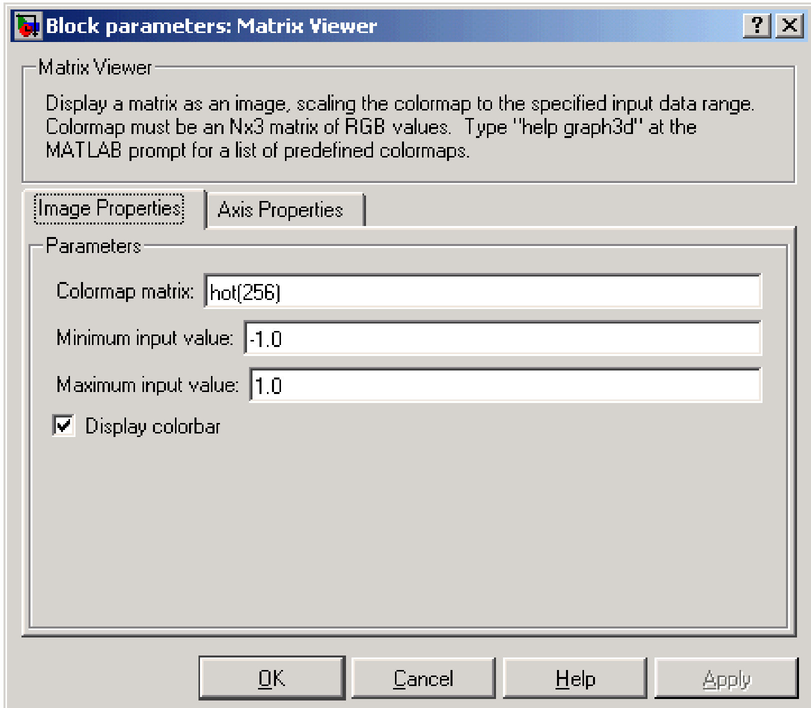
Right-click the image in the figure window to access the following menu items:

- **Refresh** erases all data on the scope display except for the most recent image.
- **Autoscale** recomputes the minimum and maximum input values to fit the range of values observed in a series of 10 consecutive inputs. The numerical limits selected by the autoscale feature are shown in the **Minimum input value** and **Maximum input value** parameters, where you can make further adjustments to them manually.
- **Axis zoom**, when selected, causes the image to completely fill the figure window. Axis titles are not displayed. When **Axis zoom** is cleared, the axis labels and titles are displayed in a gray border surrounding the scope axes. This option can also be set in the Axis Properties pane of the parameter dialog.
- **Colorbar**, when selected, displays a bar with the specified colormap to the right of the image axes.
- **Save Position** automatically updates the **Figure position** parameter in the **Axis Properties** pane to reflect the figure window's current position and size on the screen. To make the scope window open at a particular location on the screen when the simulation runs, drag the window to the desired location, resize it, and select **Save Position**. The parameter dialog must be closed when you select **Save Position** for the **Figure position** parameter to be updated.

Examples

See the demo `dspstfft.mdl` for an example of using the Matrix Viewer block to create a moving spectrogram, or time-frequency plot, of a speech signal by updating just one column of the input matrix at each sample time.

Dialog Box



Colormap matrix

A 3-column matrix defining the colormap as a set of RGB triples, or a call to a colormap-generating function such as `hot` or `spring`. See the `ColorSpec` property for complete information about defining RGB triples, and the MATLAB `colormap` function for a list of colormap-generating functions. Tunable.

Minimum input value

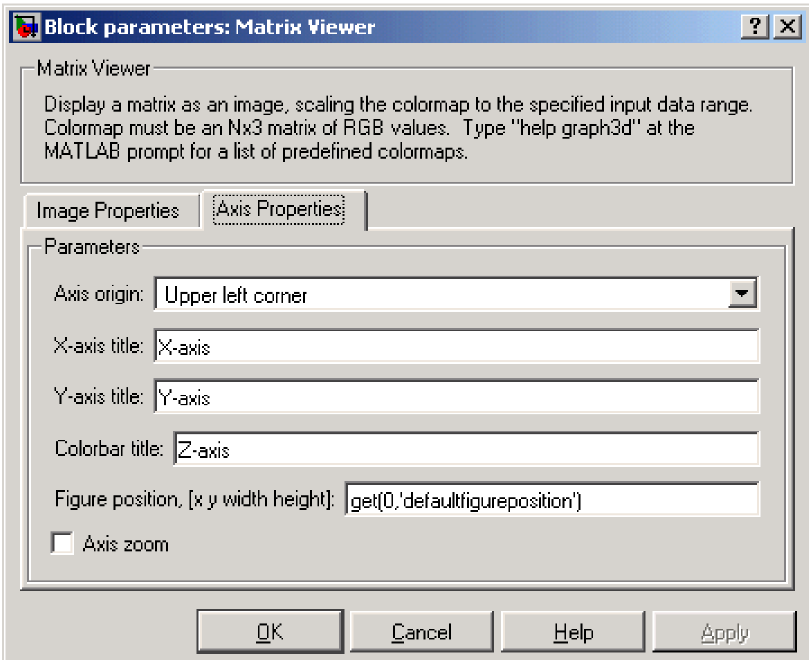
The input value to be mapped to the color defined in the first row of the colormap matrix. Right-click in the figure window and select `Autoscale` from pop-up menu to set this parameter to the minimum value observed in a series of 10 consecutive matrix inputs. Tunable.

Maximum input value

The input value to be mapped to the color defined in the last row of the colormap matrix. Right-click in the figure window and select `Autoscale` from the pop-up menu to set this parameter to the maximum value observed in a series of 10 consecutive matrix inputs. Tunable.

Display colorbar

Select to display a bar with the selected colormap to the right of the image axes. Tunable.



Axis origin
: The position within the axes where the first element of the input matrix, $U(1,1)$, is plotted; bottom left or top left. Tunable.

X-axis title
: The text to be displayed below the x-axis. Tunable.

Y-axis title
: The text to be displayed to the left of the y-axis. Tunable.

Colorbar title
: The text to be displayed to the right of the color bar, when **Display colorbar** is currently selected. Tunable.

Figure position, [x y width height]
: A 4-element vector of the form `[x y width height]` specifying the position of the figure window, where `(0,0)` is the lower-left corner of the display. Tunable.

Axis zoom
: Resizes the image to fill the figure window. Tunable.

Supported Data Types

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

See Also

Spectrum Scope	Signal Processing Blockset
Vector Scope	Signal Processing Blockset
`colormap`	MATLAB

`ColorSpec`	MATLAB
`image`	MATLAB

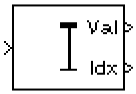
Maximum

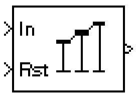
Purpose Find maximum values in input or sequence of inputs

Library Statistics

`dspstat3`

Description





The Maximum block identifies the value and/or position of the largest element in each row or column of the input, along vectors of a specified dimension of the input, or of the entire input. The Maximum block can also track the maximum values in a sequence of inputs over a period of time. The **Mode** parameter specifies the block's mode of operation and can be set to `Value`, `Index`, `Value and Index`, or `Running`.

The Maximum block supports real and complex floating-point, fixed-point, and Boolean inputs. Real fixed-point inputs can be either signed or unsigned, while complex fixed-point inputs must be signed. The data type of the maximum values output by the block match the data type of the input. The index values output by the block are `double` when the input is `double`, and `uint32` otherwise.

The frame status of the block output is the same as that of the input, except when the **Find the maximum value of** parameter is set to `Entire input`. The output is always sample based when `Entire input` is selected.

For the `Value`, `Index`, and `Value and Index` modes, the Maximum block produces identical results as the MATLAB® `max` function when it is called as `[y I] = max(u,[],D)`, where *u* and *y* are the input and output, respectively, *D* is the dimension, and *I* is the index.

Value Mode

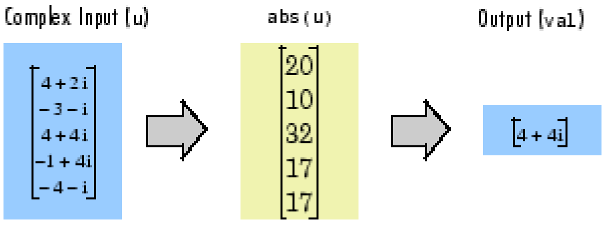
When the **Mode** parameter is set to `Value`, the block computes the maximum value in each row or column of the input, along vectors of a specified dimension of the input, or of the entire input at each sample time, and outputs the array *y*. Each element in *y* is the maximum value in the corresponding column, row, vector, or entire input. The output *y* depends on the setting of the **Find the maximum value**

over parameter. For example, consider a 3-dimensional input signal of size *M*-by-*N*-by-*P*:

- `Each row` — The output at each sample time consists of an *M*-by-1-by-*P* array, where each element contains the maximum value of each vector over the second dimension of the input. For an input that is an *M*-by-*N* matrix, the output at each sample time is an *M*-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.
- `Each column` — The output at each sample time consists of a 1-by-*N*-by-*P* array, where each element contains the maximum value of each vector over the first dimension of the input. For an input that is an *M*-by-*N* matrix, the output at each sample time is a 1-by-*N* row vector. In this mode, the frame status of the output is the same as that of the input.

  For convenience, length-*M* 1-D vector inputs are treated as *M*-by-1 column vectors when the block is in this mode. Sample-based length-*M* row vector inputs are also treated as *M*-by-1 column vectors when the **Treat sample-based row input as a column** check box is selected.
- `Entire input` — The output at each sample time is a scalar that contains the maximum value in the *M*-by-*N*-by-*P* input matrix. In this mode, the block output is always sample based.
- `Specified dimension` — The output at each sample time depends on **Dimension**. If **Dimension** is set to `1`, the output is the same as that when you select `Each column`. If **Dimension** is set to `2`, the output is the same as when you select `Each row`. If **Dimension** is set to `3`, the output at each sample time is an *M*-by-*N* matrix containing the maximum value of each vector over the third dimension of the input. In this mode, the frame status of the output is the same as that of the input.

For complex inputs, the block selects the value in each row or column of the input, along vectors of a specified dimension of the input, or of the entire input that has the maximum magnitude squared as shown below. For complex value $u = a + bi$, the magnitude squared is $a^2 + b^2$.



Index Mode

When **Mode** is set to `Index`, the block computes the maximum value in each row or column of the input, along vectors of a specified dimension of the input, or of the entire input, and outputs the index array I. Each element in I is an integer indexing the maximum value in the corresponding column, row, vector, or entire input. The output I depends on the setting of the **Find the maximum value over** parameter. For example, consider a 3-dimensional input signal of size M-by-N-by-P:

- `Each row` — The output at each sample time consists of an M-by-1-by-P array, where each element contains the index of the maximum value of each vector over the second dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is an M-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.
- `Each column` — The output at each sample time consists of a 1-by-N-by-P array, where each element contains the index of the maximum value of each vector over the first dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is an M-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.

  For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors when the block is in this mode. Sample-based length-M row vector inputs are also treated as M-by-1 column vectors when the **Treat sample-based row input as a column** check box is selected.

For an input that is an *M*-by-*N* matrix, the output at each sample time is an *M*-by-1 column vector.

- `Entire input` — The output at each sample time is a 1-by-3 vector that contains the location of the maximum value in the *M*-by-*N*-by-*P* input matrix. In this mode, the block output is always sample based. For an input that is an *M*-by-*N* matrix, the output will be a 1-by-2 vector.
- `Specified dimension` — The output at each sample time depends on **Dimension**. If **Dimension** is set to 1, the output is the same as when you select `Each column`. If **Dimension** is set to 2, the output is the same as when you select `Each row`. If **Dimension** is set to 3, the output at each sample time is an *M*-by-*N* matrix containing the indices of the maximum values of each vector over the third dimension of the input. In this mode, the frame status of the output is the same as that of the input.

When a maximum value occurs more than once, the computed index corresponds to the first occurrence. For example, when the input is the column vector `[3 2 1 2 3]'`, the computed one-based index of the maximum value is `1` rather than `5` when `Each column` is selected.

When inputs to the block are double-precision values, the index values are double-precision values. Otherwise, the index values are 32-bit unsigned integer values.

Value and Index Mode

When **Mode** is set to `Value and Index`, the block outputs both the maxima and the indices.

Running Mode

When **Mode** is set to `Running`, the block tracks the maximum value of each channel in a time sequence of *M*-by-*N* inputs. For sample-based inputs, the output is a sample-based *M*-by-*N* array with each element y_{ij} containing the maximum value observed in element u_{ij} for all inputs since the last reset. For frame-based inputs, the output is a frame-based M-by-N matrix with each element y_{ij} containing the maximum value observed in the jth column of all inputs since the last reset, up to and including element u_{ij} of the current input.

N-D signals cannot be frame based. When the block is set to `Running` mode, each element of the N-D signal is treated as a separate channel.

There are $\prod d_i$ channels, where d_i is the size of the ith dimension.

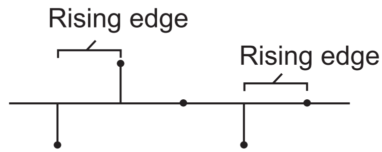
Resetting the Running Maximum

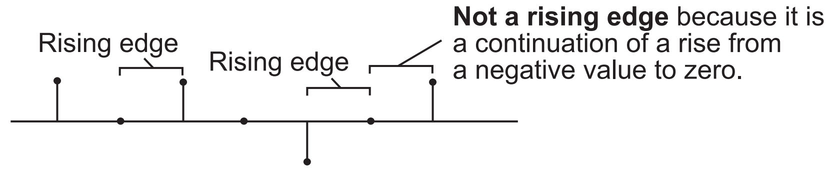
The block resets the running maximum whenever a reset event is detected at the optional Rst port. The reset sample time must be a positive integer multiple of the input sample time.

For sample-based inputs, a reset event causes the running maximum for each channel to be initialized to the value in the corresponding channel of the current input. For frame-based inputs, a reset event causes the running maximum for each channel to be initialized to the earliest value in each channel of the current input.

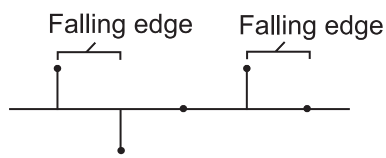
You specify the reset event in the **Reset port** menu:

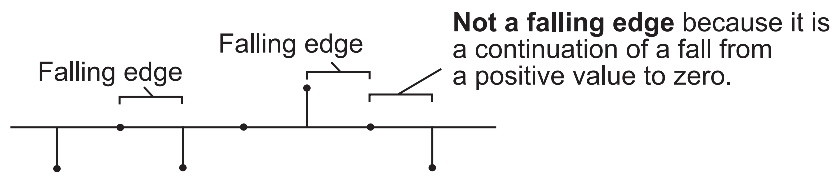
- `None` — Disables the Rst port.
- `Rising edge` — Triggers a reset operation when the Rst input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)





- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)





- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described above)
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero

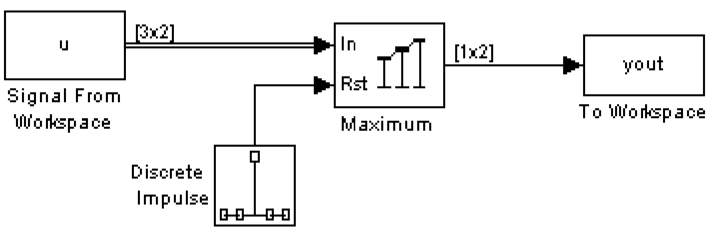
Note When running simulations in the Simulink® `MultiTasking` mode, reset signals have a one-sample latency. Therefore, when the block detects a reset event, there is a one-sample delay at the reset port rate before the block applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Fixed-Point Data Types

The parameters on the **Fixed-point** pane of the block dialog are only used for complex fixed-point inputs. The sum of the squares of the real and imaginary parts of such an input are formed before a comparison is made, as described in "Value Mode" on page 2-808. The results of the squares of the real and imaginary parts are placed into the product output data type. The result of the sum of the squares is placed into the accumulator data type. These parameters are ignored for other types of inputs.

Examples

The Maximum block in the following model calculates the running maximum of a frame-based 3-by-2 (two-channel) matrix input, u. The running maximum is reset at t=2 by an impulse to the block's Rst port.



The Maximum block has the following settings:

- **Mode** = `Running`
- **Reset port** = `Non-zero signal`

The Signal From Workspace block has the following settings:

- **Signal** = u
- **Sample time** = 1/3
- **Samples per frame** = 3

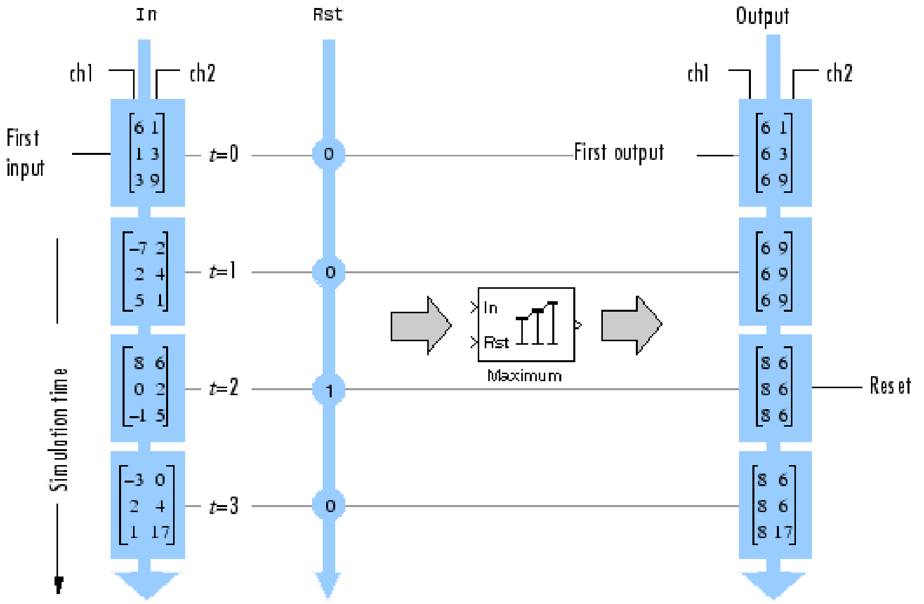
where

```
u
= [6 1 3 -7 2 5 8 0 -1 -3 2 1;1 3 9 2 4 1 6 2 5 0 4 17]'
```

The Discrete Impulse block has the following settings:

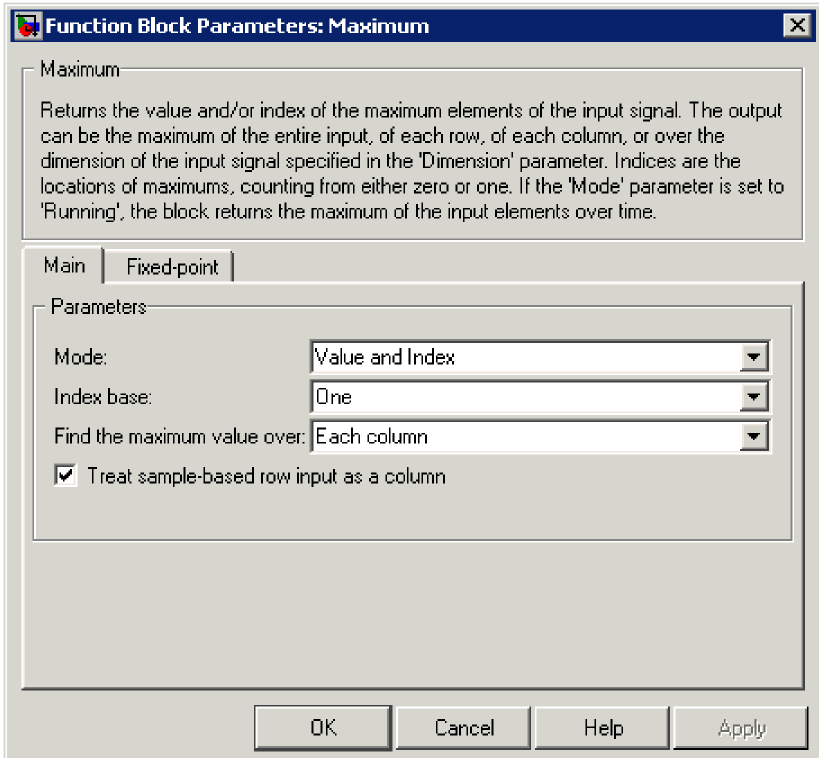
- **Delay (samples)** = 2
- **Sample time** = 1
- **Samples per frame** = 1

The block's operation is shown in the figure below.



Dialog Box

The **Main** pane of the Maximum block dialog appears as follows.



Mode

Specify the block's mode of operation:

- `Value` — Output the maximum value of each input
- `Index` — Output the index of the maximum value
- `Value and index` — Output both the value and the index
- `Running` — Track the maximum value of the input sequence over time

For more information, see Description.

Index base

Specify whether the index of the maximum value is reported using one-based or zero-based numbering. This parameter is only visible when the **Mode** parameter is set to `Index` or `Value and index`.

Find the maximum value over

Specify whether to find the maximum value along rows, columns, entire input, or the dimension specified in the **Dimension** parameter. For more information, see Description.

Treat sample-based row input as a column

Select to treat sample-based length-*M* row vector inputs as *M*-by-1 column vectors. This parameter is only visible when the **Find the maximum value of** parameter is set to `Each column`.

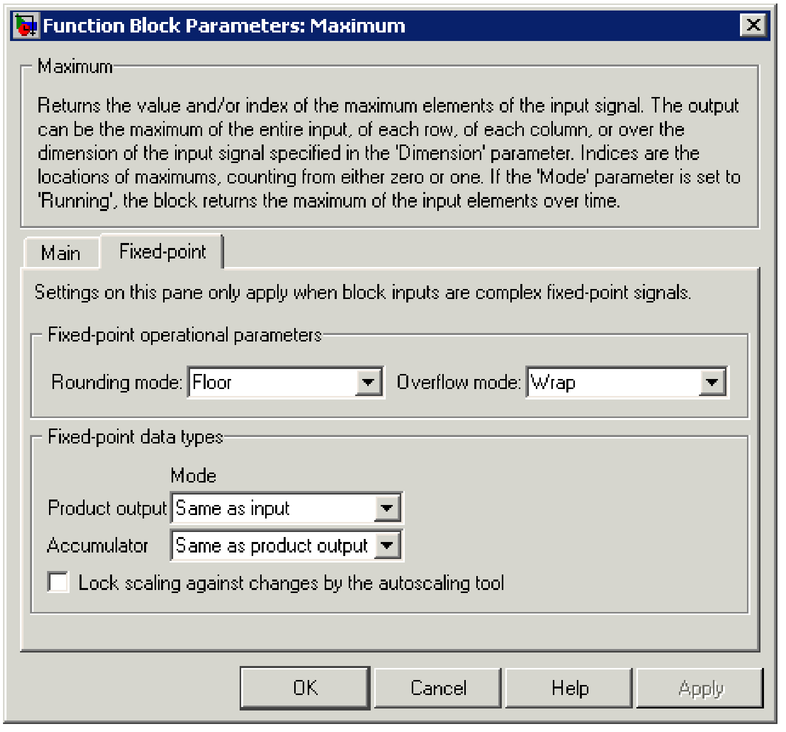
Reset port

Specify the reset event detected at the Rst input port when you select `Running` for the **Mode** parameter. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input. For information about the possible values of this parameter, see "Resetting the Running Maximum" on page 2-812.

Dimension

Specify the dimension (one-based value) of the input signal, over which the maximum is computed. The value of this parameter cannot exceed the number of dimensions in the input signal. This parameter is only visible when the **Find the maximum value over** parameter is set to `Specified dimension`.

The **Fixed-point** pane of the Maximum block dialog appears as follows.



Note The parameters on the **Fixed-point** pane are only used for complex fixed-point inputs. The sum of the squares of the real and imaginary parts of such an input are formed before a comparison is made, as described in "Value Mode" on page 2-808. The results of the squares of the real and imaginary parts are placed into the product output data type. The result of the sum of the squares is placed into the accumulator data type. These parameters are ignored for other types of inputs.

Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Product output
Use this parameter to specify how you would like to designate the product output word and fraction lengths resulting from a complex-complex multiplication in the block. See “Multiplication Data Types” for more information:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator
Use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. See “Multiplication Data Types” for more information:

- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the

autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

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

See Also

Mean	Signal Processing Blockset
Minimum	Signal Processing Blockset
MinMax	Simulink
`max`	MATLAB

Purpose

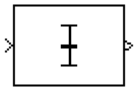
Find mean value of input or sequence of inputs

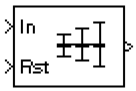
Library

Statistics

`dspstat3`

Description





The Mean block computes the mean of each row or column of the input, along vectors of a specified dimension of the input, or of the entire input. The Mean block can also track the mean value in a sequence of inputs over a period of time. The **Running mean** parameter selects between basic operation and running operation.

The Mean block accepts real and complex fixed-point and floating-point inputs.

Basic Operation

When you do not select the **Running mean** check box, the block computes the mean value in each row or column of the input, along vectors of a specified dimension of the input, or of the entire input at each individual sample time. Each element in the output array `y` is the mean value of the corresponding column, row, vector, or entire input. The output array `y` depends on the setting of the **Find the mean value over** parameter. For example, consider a 3-dimensional input signal of size M-by-N-by-P:

- `Entire input` — The output at each sample time is a scalar that contains the mean value of the M-by-N-by-P input matrix. In this mode, the block output is always sample based.

  ```
  y = mean(u(:))     % Equivalent MATLAB code
  ```

- `Each row` — The output at each sample time consists of an M-by-1-by-P array, where each element contains the mean value of each vector over the second dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is an M-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.

```
y = mean(u,2)     % Equivalent MATLAB code
```

- `Each column` — The output at each sample time consists of a 1-by-N-by-P array, where each element contains the mean value of each vector over the first dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is a 1-by-N row vector. In this mode, the frame status of the output is the same as that of the input.

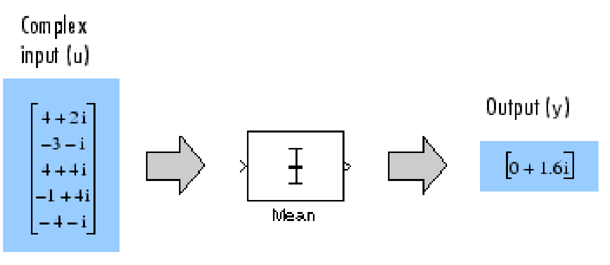
  ```
  y = mean(u)     % Equivalent MATLAB code
  ```

  For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors when the block is in this mode. Sample-based length-M row vector inputs are also treated as M-by-1 column vectors when the **Treat sample-based row input as a column** check box is selected.

- `Specified dimension` — The output at each sample time depends on **Dimension**. If **Dimension** is set to `1`, the output is the same as when you select `Each column`. If **Dimension** is set to `2`, the output is the same as when you select `Each row`. If **Dimension** is set to `3`, the output at each sample time is an M-by-N matrix containing the mean value of each vector over the third dimension of the input. In this mode, the frame status of the output is the same as that of the input.

  ```
  y = mean(u,Dimension)     % Equivalent MATLAB code
  ```

The mean of a complex input is computed independently for the real and imaginary components, as shown in the next figure.



Running Operation

When the **Running mean** check box is selected, the block tracks the mean value of each channel in a time sequence of inputs. For sample-based M-by-N inputs, the output is a sample-based M-by-N array with each element y_{ij} containing the mean value of the elements u_{ij} for all inputs since the last reset. For frame-based M-by-N inputs, the output is a frame-based M-by-N matrix with each element y_{ij} containing the mean of the values in the jth column of all inputs since the last reset, up to and including element u_{ij} of the current input.

N-D signals cannot be frame based. When the block is set to `Running` mode, each element of the N-D signal is treated as a separate channel.

There are $\prod d_i$ channels, where d_i is the size of the ith dimension.

Resetting the Running Mean

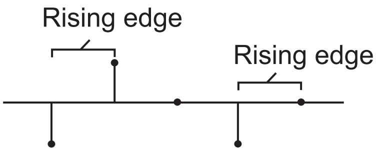
The block resets the running mean whenever a reset event is detected at the optional Rst port. The reset sample time must be a positive integer multiple of the input sample time.

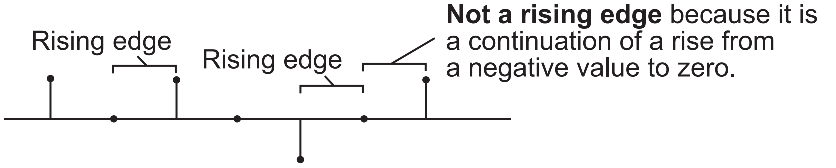
When the block is reset for sample-based inputs, the running mean for each channel is initialized to the value in the corresponding channel of the current input. For frame-based inputs, the running mean for each channel is initialized to the earliest value in each channel of the current input.

You specify the reset event by the **Reset port** parameter:

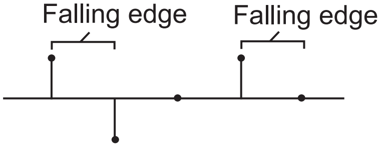
- `None` disables the Rst port.
- `Rising edge` — Triggers a reset operation when the Rst input does one of the following:
  - Rises from a negative value to a positive value or zero

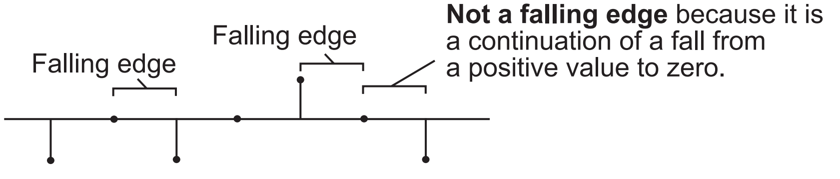
- Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)





- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)





- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described earlier)
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero

---

Note When running simulations in the Simulink® MultiTasking mode, reset signals have a one-sample latency. Therefore, when the block detects a reset event, there is a one-sample delay at the reset port rate before the block applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

ROI Processing

To calculate the statistical value within a particular region of interest (ROI) of the input, select the **Enable ROI processing** check box. This option is only available when the **Find the mean value over** parameter is set to `Entire input` and the **Running mean** check box is not selected. ROI processing is only supported for 2-D inputs.

Note Full ROI processing is only available to users who have a Video and Image Processing Blockset™ license. If you only have a Signal Processing Blockset™ license, you can still use ROI processing, but are limited to the **ROI type** `Rectangles`.

Use the **ROI type** parameter to specify whether the ROI is a rectangle, line, label matrix, or binary mask. A binary mask is a binary image that enables you to specify which pixels to highlight, or select. In a label matrix, pixels equal to 0 represent the background, pixels equal to 1 represent the first object, pixels equal to 2 represent the second object, and so on. When the **ROI type** parameter is set to `Label matrix`, the Label and Label Numbers ports appear on the block. Use the Label

Numbers port to specify the objects in the label matrix for which the block calculates statistics. The input to this port must be a vector of scalar values that correspond to the labeled regions in the label matrix. For more information about the format of the input to the ROI port when the ROI is a rectangle or a line, see the Draw Shapes block reference page.

For rectangular ROIs, use the **ROI portion to process** parameter to specify whether to calculate the statistical value for the entire ROI or just the ROI perimeter.

Use the **Output** parameter to specify the block output. The block can output separate statistical values for each ROI or the statistical value for all specified ROIs. This parameter is not available if, for the **ROI type** parameter, you select `Binary mask`.

If, for the **ROI type** parameter, you select `Rectangles` or `Lines`, the **Output flag indicating if ROI is within image bounds** check box appears in the dialog box. If you select this check box, the Flag port appears on the block. The following tables describe the Flag port output based on the block parameters.

Output = Individual statistics for each ROI

Flag Port Output	Description
0	ROI is completely outside the input image.
1	ROI is completely or partially inside the input image.

Output = Single statistic for all ROIs

Flag Port Output	Description
0	All ROIs are completely outside the input image.
1	At least one ROI is completely or partially inside the input image.

If the ROI is partially outside the image, the block only computes the statistical values for the portion of the ROI that is within the image.

If, for the **ROI type** parameter, you select `Label matrix`, the **Output flag indicating if input label numbers are valid** check box appears in the dialog box. If you select this check box, the Flag port appears on the block. The following tables describe the Flag port output based on the block parameters.

Output = Individual statistics for each ROI

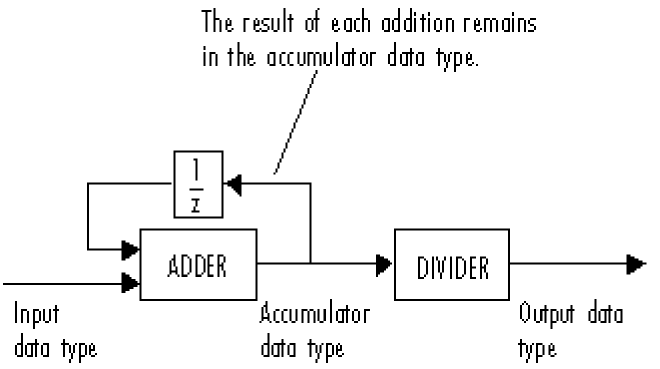
Flag Port Output	Description
0	Label number is not in the label matrix.
1	Label number is in the label matrix.

Output = Single statistic for all ROIs

Flag Port Output	Description
0	None of the label numbers are in the label matrix.
1	At least one of the label numbers is in the label matrix.

Fixed-Point Data Types

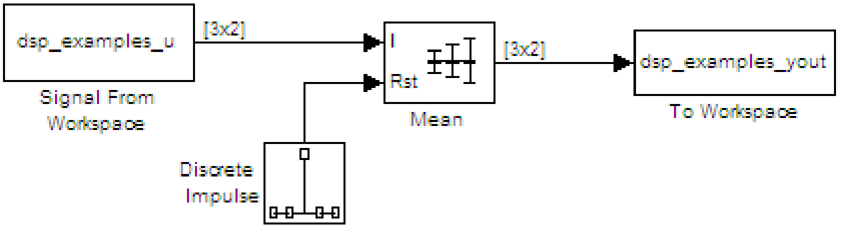
The following diagram shows the data types used within the Mean block for fixed-point signals.



You can set the accumulator and output data types in the block dialog, as discussed in "Dialog Box" on page 2-833.

Examples

The Mean block in the following model calculates the running mean of a frame-based 3-by-2 (two-channel) matrix input, u. The running mean is reset at t=2 by an impulse to the block's Rst port.



The Mean block has the following settings:

- **Running mean** = Select this check box
- **Reset port** = `Non-zero sample`

The Signal From Workspace block has the following settings:

- **Signal** = dsp_examples_u
- **Sample time** = 1/3
- **Samples per frame** = 3

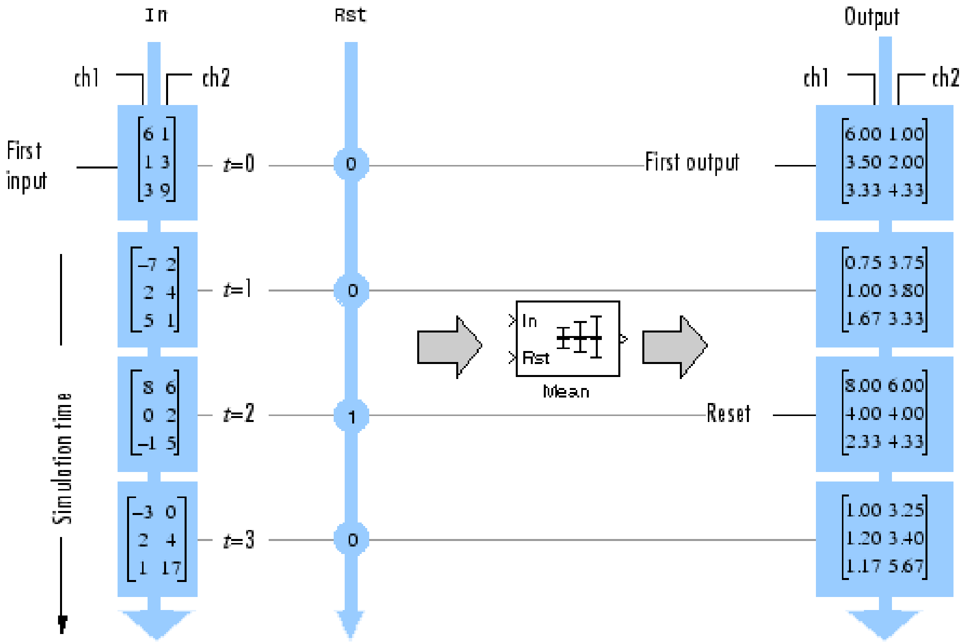
where

```
dsp_examples_u = [6 1 3 -7 2 5 8 0 -1 -3 2 1;1 3 9 2 4 1 6 2 5 0 4 17]'
```

The Discrete Impulse block has the following settings:

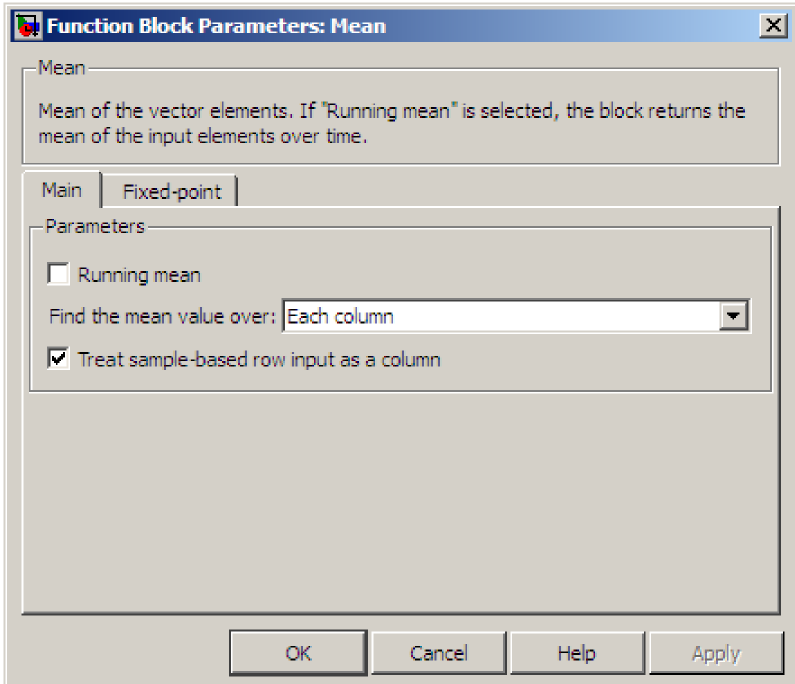
- **Delay (samples)** = 2
- **Sample time** = 1
- **Samples per frame** = 1

The block's operation is shown in the next figure.



Dialog Box

The **Main** pane of the Mean block dialog appears as follows.



Running mean
Enables running operation when selected.

Reset port
Determines the reset event that causes the block to reset the running mean. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input. This parameter is enabled only when you set the **Running mean** parameter. For more information, see "Resetting the Running Mean" on page 2-825.

Find the mean value over
Specify whether to find the mean value along rows, columns, entire input, or the dimension specified in the **Dimension** parameter. For more information, see "Basic Operation" on page 2-823.

Treat sample-based row input as a column
Select to treat sample-based length-M row vector inputs as M-by-1 column vectors. This parameter is only visible when the **Find the mean value over** parameter is set to `Each column`.

Dimension
Specify the dimension (one-based value) of the input signal, over which the mean is computed. The value of this parameter cannot exceed the number of dimensions in the input signal. This parameter is only visible when the **Find the mean value over** parameter is set to `Specified dimension`.

Enable ROI Processing
Select this check box to calculate the statistical value within a particular region of each image. This parameter is only available when the **Find the mean value over** parameter is set to `Entire input`, and the block is not in running mode.

Note Full ROI processing is only available to users who have a Video and Image Processing Blockset license. If you only have a Signal Processing Blockset license, you can still use ROI processing, but are limited to the **ROI type** `Rectangles`.

ROI type
Specify the type of ROI you want to use. Your choices are `Rectangles`, `Lines`, `Label matrix`, or `Binary mask`.

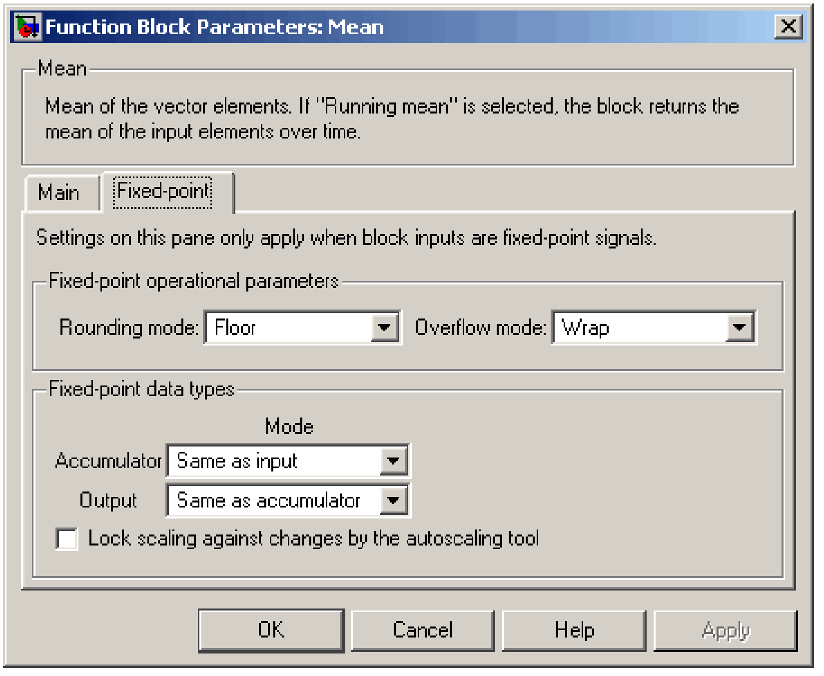
ROI portion to process
Specify whether you want to calculate the statistical value for the entire ROI or just the ROI perimeter. This parameter is only visible if, for the **ROI type** parameter, you specify `Rectangles`.

Output
Specify the block output. The block can output a vector of separate statistical values for each ROI or a scalar value that represents the statistical value for all the specified ROIs. This parameter is not available if, for the **ROI type** parameter, you select `Binary mask`.

Output flag indicating if ROI is within image bounds
If you select this check box, the Flag port appears on the block. For a description of the Flag port output, see the tables in "ROI Processing" on page 2-827. This parameter is visible if, for the **ROI type** parameter, you select `Rectangles` or `Lines`.

Output flag indicating if label numbers are valid
If you select this check box, the Flag port appears on the block. For a description of the Flag port output, see the tables in "ROI Processing" on page 2-827. This parameter is visible if, for the **ROI type** parameter, you select `Label matrix`.

The **Fixed-point** pane of the Mean block dialog appears as follows.



Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Accumulator
Use this parameter to specify the accumulator word and fraction lengths:

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

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

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

Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

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

Port	Supported Data Types
ROI	Rectangles and lines: • Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers Binary Mask: • Boolean
Label	• 8-, 16-, and 32-bit unsigned integers
Label Numbers	• 8-, 16-, and 32-bit unsigned integers
Output	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Flag	• Boolean

See Also

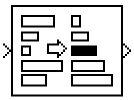
Maximum	Signal Processing Blockset
Median	Signal Processing Blockset
Minimum	Signal Processing Blockset
Standard Deviation	Signal Processing Blockset
`mean`	MATLAB

Purpose Find median value of input

Library Statistics

`dspstat3`

Description



The Median block computes the median value of each row or column of the input, along vectors of a specified dimension of the input, or of the entire input. The median of a set of input values is calculated as follows:

1 The values are sorted.

2 If the number of values is odd, the median is the middle value.

3 If the number of values is even, the median is the average of the two middle values.

For a given input u, the size of the output array `y` depends on the setting of the **Find the median value over** parameter. For example, consider a 3-dimensional input signal of size M-by-N-by-P:

- `Entire input` — The output at each sample time is a scalar that contains the median value of the M-by-N-by-P input matrix. In this mode, the block output is always sample based.

```
y = median(u(:))     % Equivalent MATLAB code
```

- `Each row` — The output at each sample time consists of an M-by-1-by-P array, where each element contains the median value of each vector over the second dimension of the input. For an input that is an M-by-N matrix, the output is an M-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.

```
y = median(u,2)     % Equivalent MATLAB code
```

- `Each column` — The output at each sample time consists of a 1-by-N-by-P array, where each element contains the median value of each vector over the first dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is a 1-by-N row vector. In this mode, the frame status of the output is the same as that of the input.

```
y = median(u)     % Equivalent MATLAB code
```

For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors when the block is in this mode. Sample-based length-M row vector inputs are also treated as M-by-1 column vectors when the **Treat sample-based row input as a column** check box is selected.

- `Specified dimension` — The output at each sample time depends on **Dimension**. If **Dimension** is set to `1`, the output is the same as when you select `Each column`. If **Dimension** is set to `2`, the output is the same as when you select `Each row`. If **Dimension** is set to `3`, the output at each sample time is an M-by-N matrix containing the median value of each vector over the third dimension of the input. In this mode, the frame status of the output is the same as that of the input.

  ```
  y = median(u,Dimension)     % Equivalent MATLAB code
  ```

Complex inputs are sorted by magnitude squared. For complex value $u = a + bi$, the magnitude squared is a^2+b^2.

The Median block accepts real and complex fixed-point and floating-point inputs.

Fixed-Point Data Types

For fixed-point inputs, you can specify accumulator, product output, and output data types as discussed in “Dialog Box” on page 2-842. Not all these fixed-point parameters are applicable for all types of fixed-point inputs. The following table shows when each kind of data type and scaling is used.

	Output data type	Accumulator data type	Product output data type
Even *M*	X	X	
Odd *M*	X		
Odd *M* and complex	X	X	X
Even *M* and complex	X	X	X

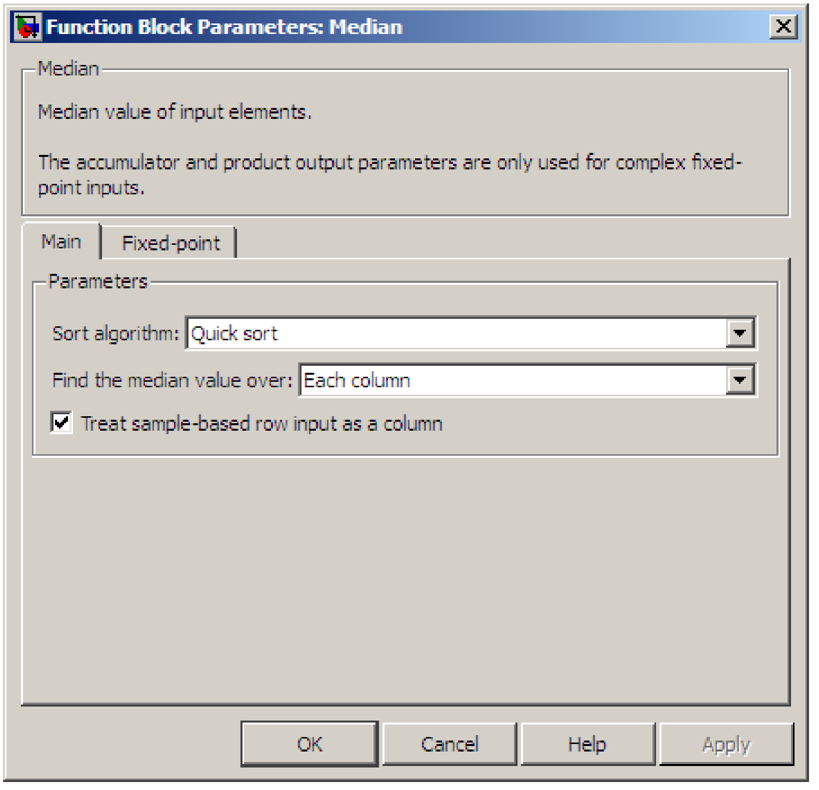
The accumulator and output data types and scalings are used for fixed-point signals when *M* is even. The result of the sum performed while calculating the average of the two central rows of the input matrix is stored in the accumulator data type and scaling. The total result of the average is then put into the output data type and scaling.

The accumulator and product output parameters are used for complex fixed-point inputs. The sum of the squares of the real and imaginary parts of such an input are formed before the input elements are sorted, as described in Description. The results of the squares of the real and imaginary parts are placed into the product output data type and scaling. The result of the sum of the squares is placed into the accumulator data type and scaling.

For fixed-point inputs that are both complex and have even *M*, the data types are used in all of the ways described. Therefore, in such cases, the accumulator type is used in two different ways.

Dialog Box

The **Main** pane of the Median block dialog appears as follows.



Sort algorithm

Specify whether to sort the elements of the input using a `Quick sort` or an `Insertion sort` algorithm.

Find the median value over

Specify whether to find the median value along rows, columns, entire input, or the dimension specified in the **Dimension** parameter. For more information, see Description.

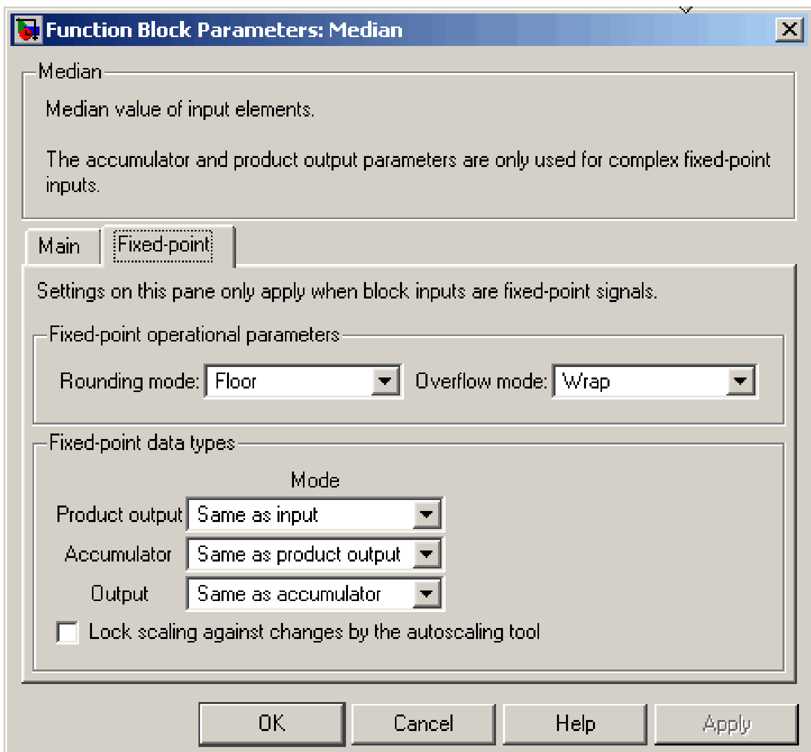
Treat sample-based row input as a column

Select to treat sample-based length-M row vector inputs as M-by-1 column vectors. This parameter is only visible when the **Find the median value over** parameter is set to `Each column`.

Dimension

Specify the dimension (one-based value) of the input signal, over which the median is computed. The value of this parameter cannot exceed the number of dimensions in the input signal. This parameter is only visible when the **Find the median value over** parameter is set to `Specified dimension`.

The **Fixed-point** pane of the Median block dialog appears as follows.



Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

Select the overflow mode for fixed-point operations.

Note The product output, accumulator, and output parameters listed are only used in certain cases. See "Fixed-Point Data Types" on page 2-840 for more information.

Product output

Use this parameter to specify how you want to designate the product output word and fraction lengths:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block:

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

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • 8-, 16-, 32-, and 128-bit signed integers • 8-, 16-, 32-, and 128-bit unsigned integers
Output	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • 8-, 16-, 32-, and 128-bit signed integers • 8-, 16-, 32-, and 128-bit unsigned integers

See Also

Maximum	Signal Processing Blockset
Mean	Signal Processing Blockset
Minimum	Signal Processing Blockset
Sort	Signal Processing Blockset
Standard Deviation	Signal Processing Blockset
Variance	Signal Processing Blockset
`median`	MATLAB

Purpose

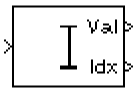
Find minimum values in input or sequence of inputs

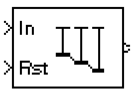
Library

Statistics

`dspstat3`

Description





The Minimum block identifies the value and/or position of the smallest element in each row or column of the input, along vectors of a specified dimension of the input, or of the entire input. The Minimum block can also track the minimum values in a sequence of inputs over a period of time. The **Mode** parameter specifies the block's mode of operation, and can be set to `Value`, `Index`, `Value and Index`, or `Running`.

The Minimum block supports real and complex floating-point, fixed-point, and Boolean inputs. Real fixed-point inputs can be either signed or unsigned, while complex fixed-point inputs must be signed. The data type of the minimum values output by the block match the data type of the input. The index values output by the block are `double` when the input is `double`, and `uint32` otherwise.

The frame status of the block output is the same as that of the input, except when the **Find the minimum value of** parameter is set to `Entire input`. The output is always sample-based when `Entire input` is selected.

For the `Value`, `Index`, and `Value and Index` modes, the Minimum block produces identical results as the MATLAB® `min` function when it is called as `[y I] = min(u,[],D)`, where u and y are the input and output, respectively, D is the dimension, and I is the index.

Value Mode

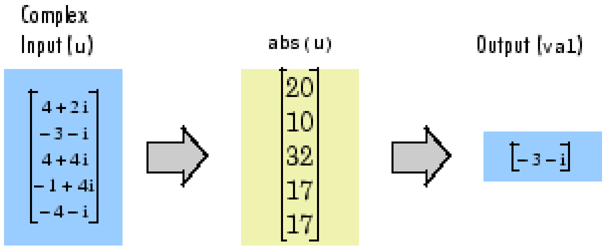
When the **Mode** parameter is set to `Value`, the block computes the minimum value in each row or column of the input, along vectors of a specified dimension of the input, or of the entire input at each sample time, and outputs the array y. Each element in y is the minimum value in the corresponding column, row, vector, or entire input. The output y depends on the setting of the **Find the minimum value**

over parameter. For example, consider a 3-dimensional input signal of size M-by-N-by-P:

- `Each row` — The output at each sample time consists of an M-by-1-by-P array, where each element contains the minimum value of each vector over the second dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is an M-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.
- `Each column` — The output at each sample time consists of a 1-by-N-by-P array, where each element contains the minimum value of each vector over the first dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is a 1-by-N row vector. In this mode, the frame status of the output is the same as that of the input.

  For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors when the block is in this mode. Sample-based length-M row vector inputs are also treated as M-by-1 column vectors when the **Treat sample-based row input as a column** check box is selected.
- `Entire input` — The output at each sample time is a scalar that contains the minimum value in the M-by-N-by-P input matrix. In this mode, the block output is always sample based.
- `Specified dimension` — The output at each sample time depends on **Dimension**. If **Dimension** is set to `1`, the output is the same as when you select `Each column`. If **Dimension** is set to `2`, the output is the same as when you select `Each row`. If **Dimension** is set to `3`, the output at each sample time is an M-by-N matrix containing the minimum value of each vector over the third dimension of the input. In this mode, the frame status of the output is the same as that of the input.

For complex inputs, the block selects the value in each row or column of the input, along vectors of a specified dimension of the input, or of the entire input that has the minimum magnitude squared as shown below. For complex value $u = a + bi$, the magnitude squared is $a^2 + b^2$.



Index Mode

When **Mode** is set to `Index`, the block computes the minimum value in each row or column of the input, along vectors of a specified dimension of the input, or of the entire input, and outputs the index array I. Each element in I is an integer indexing the minimum value in the corresponding column, row, vector, or entire input. The output I depends on the setting of the **Find the minimum value over** parameter. For example, consider a 3-dimensional input signal of size M-by-N-by-P:

- `Each row` — The output at each sample time consists of an M-by-1-by-P array, where each element contains the index of the minimum value of each vector over the second dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is an M-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.
- `Each column` — The output at each sample time consists of a 1-by-N-by-P array, where each element contains the index of the minimum value of each vector over the first dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is an Mx1 column vector. In this mode, the frame status of the output is the same as that of the input.

  For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors when the block is in this mode. Sample-based length-M row vector inputs are also treated as M-by-1 column vectors when the **Treat sample-based row input as a column** check box is selected.

For an input that is an M-by-N matrix, the output at each sample time is an M-by-1 column vector.

- `Entire input` — The output at each sample time is a 1-by-3 vector that contains the location of the minimum value in the M-by-N-by-P input matrix. In this mode, the block output is always sample based. For an input that is an M-by-N matrix, the output will be a 1-by-2 vector.
- `Specified dimension` — The output at each sample time depends on **Dimension**. If **Dimension** is set to 1, the output is the same as when you select `Each column`. If **Dimension** is set to 2, the output is the same as when you select `Each row`. If **Dimension** is set to 3, the output at each sample time is an M-by-N matrix containing the indices of the minimum values of each vector over the third dimension of the input. In this mode, the frame status of the output is the same as that of the input.

When a minimum value occurs more than once, the computed index corresponds to the first occurrence. For example, when the input is the column vector `[-1 2 3 2 -1]'`, the computed one-based index of the minimum value is `1` rather than `5` when `Each column` is selected.

Value and Index Mode

When **Mode** is set to `Value and Index`, the block outputs both the minima and the indices.

Running Mode

When **Mode** is set to `Running`, the block tracks the minimum value of each channel in a time sequence of M-by-N inputs. For sample-based inputs, the output is a sample-based M-by-N array with each element y_{ij} containing the minimum value observed in element u_{ij} for all inputs since the last reset. For frame-based inputs, the output is a frame-based M-by-N array with each element y_{ij} containing the minimum value observed in the jth column of all inputs since the last reset, up to and including element u_{ij} of the current input.

N-D signals cannot be frame based. When the block is set to `Running` mode, each element of the N-D signal is considered as a separate channel. There are $\prod d_i$ channels, where d_i is the size of the ith dimension.

Resetting the Running Minimum

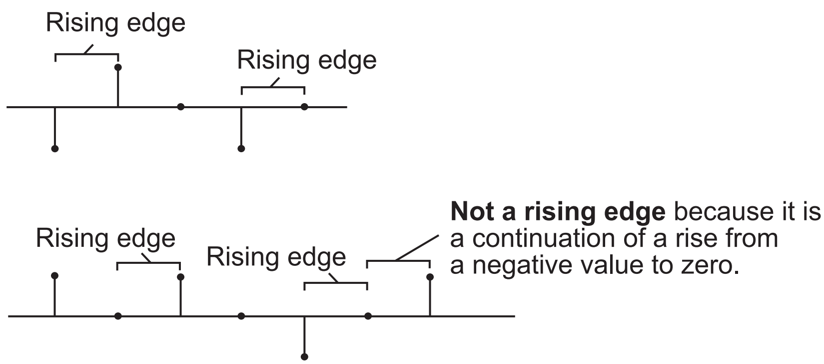
The block resets the running minimum whenever a reset event is detected at the optional Rst port. The reset sample time must be a positive integer multiple of the input sample time.

When the block is reset for sample-based inputs, the running minimum for each channel is initialized to the value in the corresponding channel of the current input. For frame-based inputs, the running minimum for each channel is initialized to the earliest value in each channel of the current input.

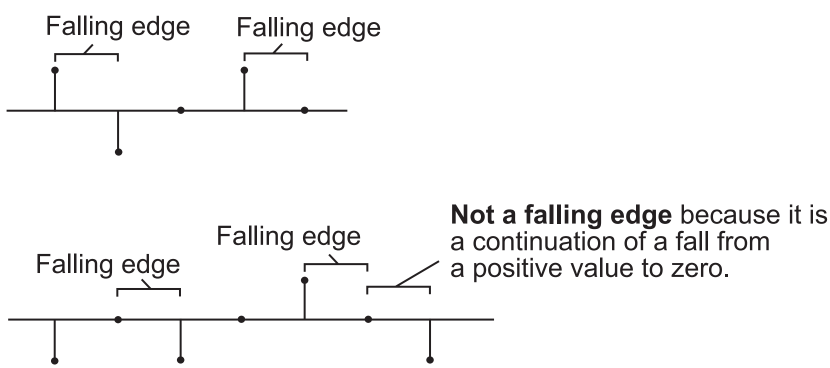
You specify the reset event by the **Reset port** parameter:

- `None` — Disables the Rst port
- `Rising edge` — Triggers a reset operation when the Rst input does one of the following:
  - Rises from a negative value to a positive value or zero

- Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)



- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described above)
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero

---

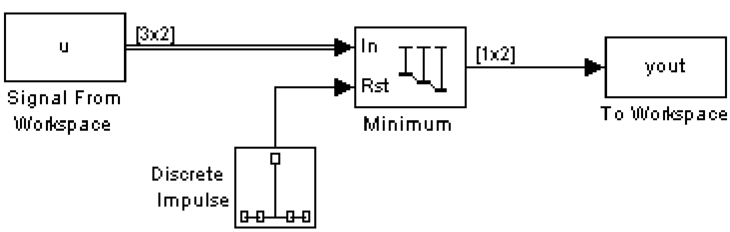
Note When running simulations in the Simulink® `MultiTasking` mode, reset signals have a one-sample latency. Therefore, when the block detects a reset event, there is a one-sample delay at the reset port rate before the block applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Fixed-Point Data Types

The parameters on the **Fixed-point** pane of the block dialog are only used for complex fixed-point inputs. The sum of the squares of the real and imaginary parts of such an input are formed before a comparison is made, as described in "Value Mode" on page 2-847. The results of the squares of the real and imaginary parts are placed into the product output data type. The result of the sum of the squares is placed into the accumulator data type. These parameters are ignored for other types of inputs.

Examples

The Minimum block in the following model calculates the running minimum of a frame-based 3-by-2 (two-channel) matrix input. The running minimum is reset at t=2 by an impulse to the block's Rst port.



The Minimum block has the following settings:

- **Mode** = `Running`
- **Reset port** = `Non-zero sample`

The Signal From Workspace block has the following settings:

- **Signal** = `u`
- **Sample time** = `1/3`
- **Samples per frame** = `3`

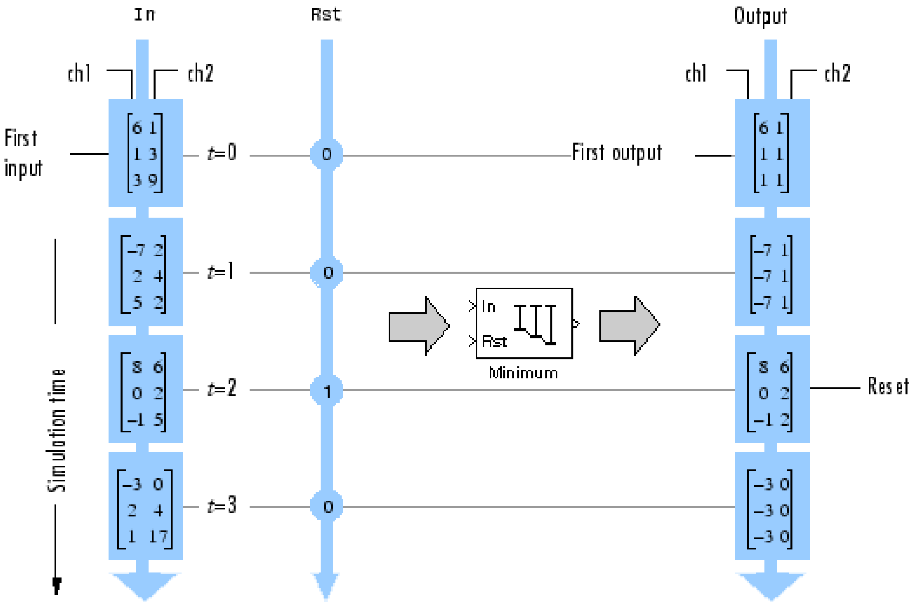
where

```
u = [6 1 3 -7 2 5 8 0 -1 -3 2 1;1 3 9 2 4 2 6 2 5 0 4 17]'
```

The Discrete Impulse block has the following settings:

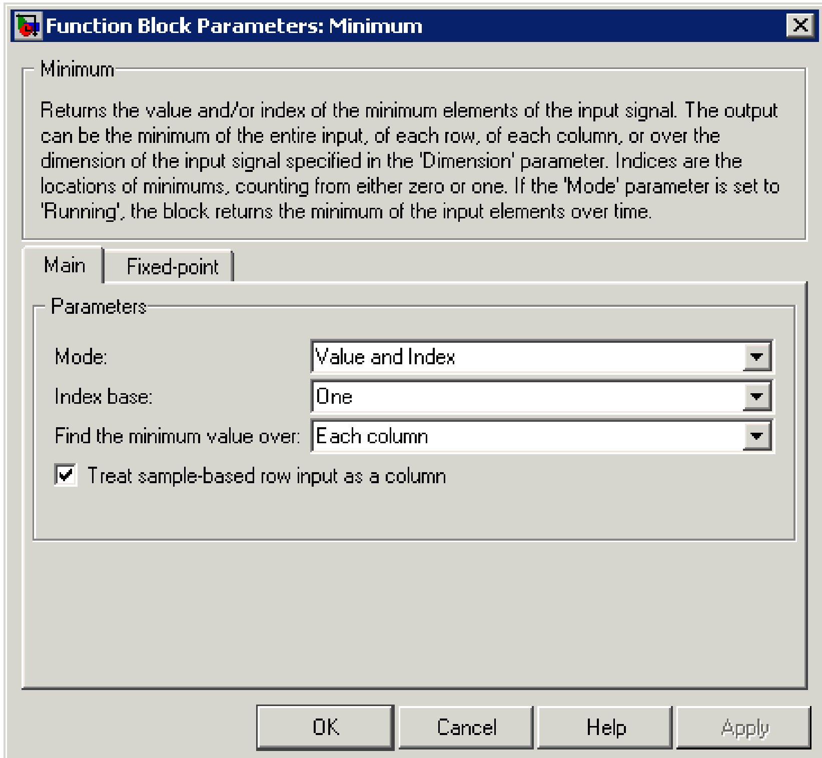
- **Delay (samples)** = `2`
- **Sample time** = `1`
- **Samples per frame** = `1`

The block's operation is shown in the figure below.



Dialog Box

The **Main** pane of the Minimum block dialog appears as follows.



Mode

Specify the block's mode of operation:

- `Value` — Output the minimum value of each input
- `Index` — Output the index of the minimum value
- `Value and index` — Output both the value and the index
- `Running` — Track the minimum value of the input sequence over time

For more information, see Description.

Index base
Specify whether the index of the minimum value is reported using one-based or zero-based numbering. This parameter is only visible when the **Mode** parameter is set to `Index` or `Value and index`.

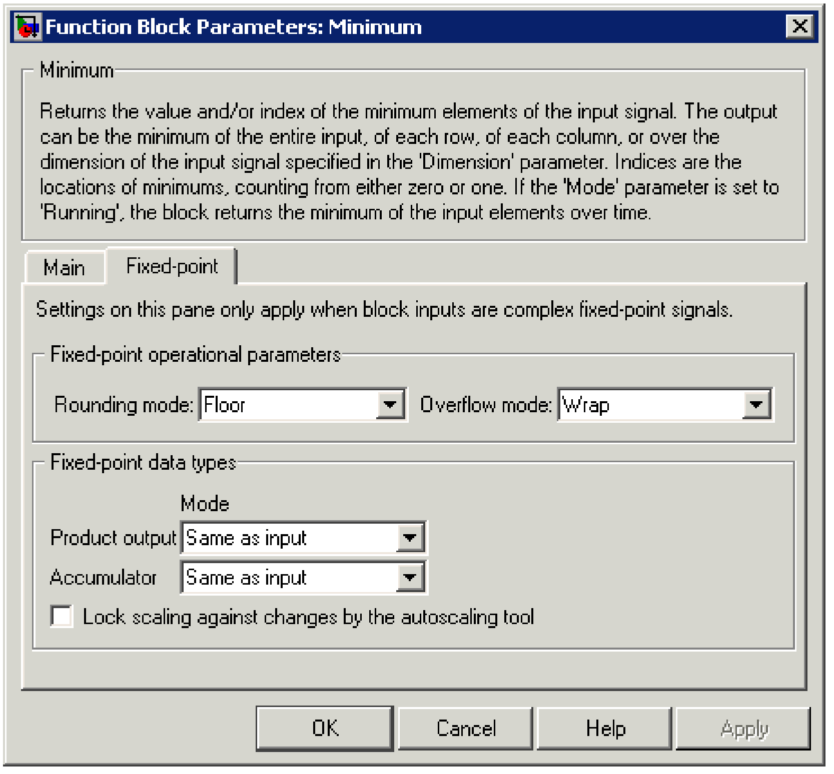
Find the minimum value over
Specify whether to find the minimum value along rows, columns, entire input, or the dimension specified in the **Dimension** parameter. For more information, see Description.

Treat sample-based row input as a column
Select to treat sample-based length-M row vector inputs as M-by-1 column vectors. This parameter is only visible when the **Find the minimum value of** parameter is set to `Each column`.

Reset port
Specify the reset event detected at the `RST` input port when you select `Running` for the **Mode** parameter. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input. This parameter is enabled only when you set the **Mode** parameter to `Running`. For information about the possible values of this parameter, see "Resetting the Running Minimum" on page 2-851.

Dimension
Specify the dimension (one-based value) of the input signal, over which the minimum is computed. The value of this parameter cannot exceed the number of dimensions in the input signal. This parameter is only visible when the **Find the minimum value over** parameter is set to `Specified dimension`.

The **Fixed-point** pane of the Minimum block dialog appears as follows.



Note The parameters on the **Fixed-point** pane are only used for complex fixed-point inputs. The sum of the squares of the real and imaginary parts of such an input are formed before a comparison is made, as described in "Value Mode" on page 2-847. The results of the squares of the real and imaginary parts are placed into the product output data type. The result of the sum of the squares is placed into the accumulator data type. These parameters are ignored for other types of inputs.

Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

Select the overflow mode for fixed-point operations.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths resulting from a complex-complex multiplication in the block. See "Multiplication Data Types" for more information:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. See "Multiplication Data Types" for more information:

- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

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

See Also

Maximum	Signal Processing Blockset
Mean	Signal Processing Blockset
MinMax	Simulink
Histogram	Signal Processing Blockset
`min`	MATLAB

Modified Covariance AR Estimator

Purpose

Compute estimate of autoregressive (AR) model parameters using modified covariance method

Library

Estimation / Parametric Estimation

`dspparest3`

Description

The Modified Covariance AR Estimator block uses the modified covariance method to fit an autoregressive (AR) model to the input data. This method minimizes the forward and backward prediction errors in the least squares sense. The input is a frame of consecutive time samples, which is assumed to be the output of an AR system driven by white noise. The block computes the normalized estimate of the AR system parameters, *A(z)*, independently for each successive input.

$$H(z) = \frac{G}{A(z)} = \frac{G}{1 + a(2)z^{-1} + \ldots + a(p+1)z^{-p}}$$

You specify the order, p, of the all-pole model in the **Estimation order** parameter. To guarantee a valid output, you must set the **Estimation order** parameter to be less than or equal to two thirds the input vector length.

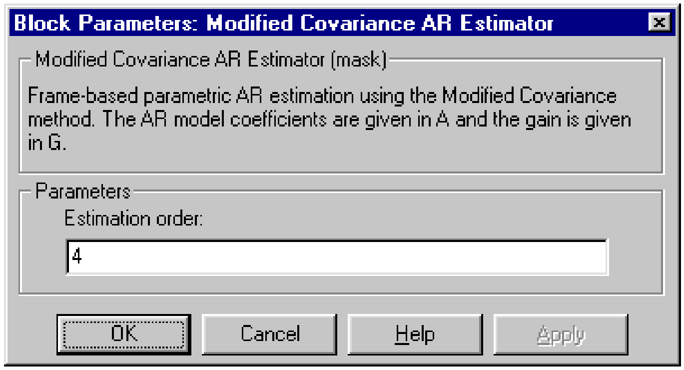
The output port labeled A outputs the normalized estimate of the AR model coefficients in descending powers of z.

```
[1 a(2) ... a(p+1)]
```

The scalar gain, G, is output from the output port labeled G.

See the Burg AR Estimator block reference page for a comparison of the Burg AR Estimator, Covariance AR Estimator, Modified Covariance AR Estimator, and Yule-Walker AR Estimator blocks.

Dialog Box



Estimation order

The order of the AR model, p.

References

Kay, S. M. *Modern Spectral Estimation: Theory and Application.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

Marple, S. L., Jr., *Digital Spectral Analysis with Applications.* Englewood Cliffs, NJ: Prentice-Hall, 1987.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
A	• Double-precision floating point • Single-precision floating point
G	• Double-precision floating point • Single-precision floating point

The output data type is the same as the input data type.

See Also

Burg AR Estimator	Signal Processing Blockset
Covariance AR Estimator	Signal Processing Blockset
Modified Covariance Method	Signal Processing Blockset
Yule-Walker AR Estimator	Signal Processing Blockset
`armcov`	Signal Processing Toolbox

Purpose

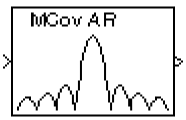
Compute parametric spectral estimate using modified covariance method

Library

Estimation / Power Spectrum Estimation

`dspspect3`

Description



The Modified Covariance Method block estimates the power spectral density (PSD) of the input using the modified covariance method. This method fits an autoregressive (AR) model to the signal by minimizing the forward and backward prediction errors in the least squares sense. The order of the all-pole model is the value specified by the **Estimation order** parameter. To guarantee a valid output, you must set the **Estimation order** parameter to be less than or equal to two thirds the input vector length. The spectrum is computed from the FFT of the estimated AR model parameters.

The input is a sample-based vector (row, column, or 1-D) or frame-based vector (column only) representing a frame of consecutive time samples from a single-channel signal. The block's output (a column vector) is the estimate of the signal's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where F_s is the signal's sample frequency.

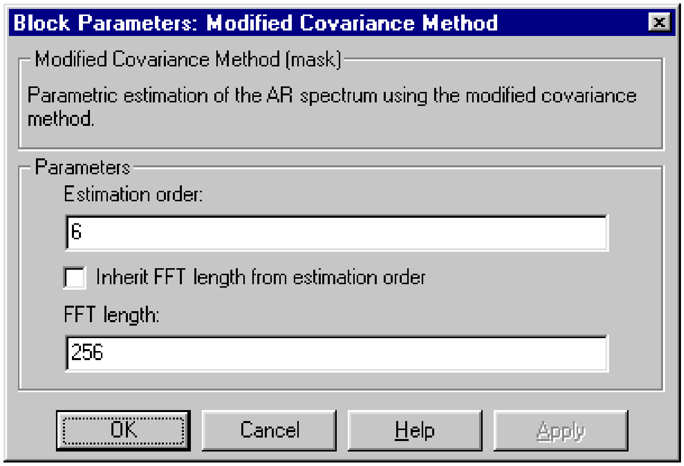
When you select **Inherit FFT length from input dimensions**, N_{fft} is specified by the frame size of the input, which must be a power of 2. When you do not select **Inherit FFT length from input dimensions**, N_{fft} is specified as a power of 2 by the **FFT length** parameter, and the block zero pads or wraps the input to N_{fft} before computing the FFT. The output is always sample based.

See the Burg Method block reference for a comparison of the Burg Method, Covariance Method, Modified Covariance Method, and Yule-Walker Method blocks.

Examples

The `dspsacomp` demo compares the modified covariance method with several other spectral estimation methods.

Dialog Box



Estimation order

The order of the AR model.

Inherit FFT length from input dimensions

When selected, uses the input frame size as the number of data points, N_{fft}, on which to perform the FFT. Tunable.

FFT length

Enter the number of data points on which to perform the FFT, N_{fft}. When N_{fft} is larger than the input frame size, each frame is zero-padded as needed. When N_{fft} is smaller than the input frame size, each frame is wrapped as needed. This parameter is enabled when you clear the **Inherit FFT length from input dimensions** check box.

References

Kay, S. M. *Modern Spectral Estimation: Theory and Application.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

Marple, S. L., Jr., *Digital Spectral Analysis with Applications.* Englewood Cliffs, NJ: Prentice-Hall, 1987.

Orfanidis, S. J. *Introduction to Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1995.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
Output	• Double-precision floating point • Single-precision floating point

The output data type is the same as the input data type.

See Also

Burg Method	Signal Processing Blockset
Covariance Method	Signal Processing Blockset
Modified Covariance AR Estimator	Signal Processing Blockset
Short-Time FFT	Signal Processing Blockset
Yule-Walker Method	Signal Processing Blockset
`pmcov`	Signal Processing Toolbox

See "Power Spectrum Estimation" for related information.

Purpose

Generate multiple binary clock signals

Library

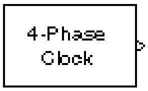
- Signal Processing Sources

  `dspsrcs4`

- Signal Management / Switches and Counters

  `dspswit3`

Description



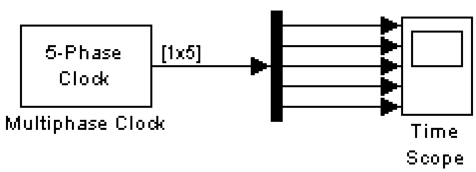
The Multiphase Clock block generates a sample-based 1-by-*N* vector of clock signals, where you specify the integer *N* in the **Number of phases** parameter. Each of the *N* phases has the same frequency, *f*, specified in hertz by the **Clock frequency** parameter.

The clock signal indexed by the **Starting phase** parameter is the first to become active, at *t*=0. The other signals in the output vector become active in turn, each one lagging the preceding signal's activation by 1/(*N**f*) seconds, the phase interval. The period of the sample-based output is therefore 1/(*N**f*) seconds.

The active level can be either high (`1`) or low (`0`), as specified by the **Active level (polarity)** parameter. The duration of the active level, *D*, is set by the **Number of phase intervals over which the clock is active**. This value, which can be an integer value between 1 and *N*-1, specifies the number of phase intervals that each signal should remain in the active state after becoming active. The active duty cycle of the signal is *D*/*N*.

Examples

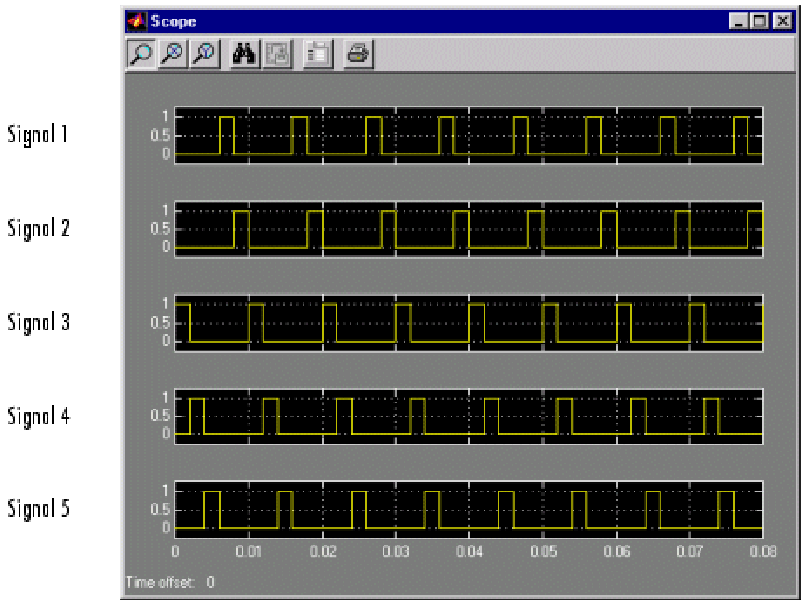
Configure the Multiphase Clock block in the model below to generate a 100 Hz five-phase output in which the third signal is first to become active. Use a high active level with a duration of one interval.



The corresponding settings are as follows:

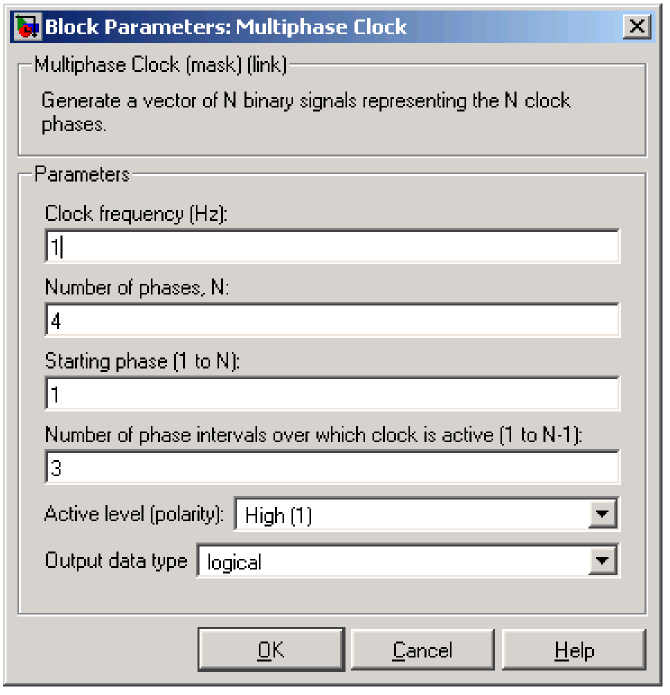
- **Clock frequency** = `100`
- **Number of phases** = `5`
- **Starting phase** = `3`
- **Number of phase intervals over which the clock is active** = `1`
- **Active level (polarity)** = `High (1)`

The Scope window below shows the Multiphase Clock block's output for these settings. Note that the first active level appears at *t*=0 on `y(3)`, the second active level appears at *t*=0.002 on `y(4)`, the third active level appears at *t*=0.004 on `y(5)`, the fourth active level appears at *t*=0.006 on `y(1)`, and the fifth active level appears at *t*=0.008 on `y(2)`. Each signal becomes active 1/(5*100) seconds after the previous signal.



To experiment further, try changing the **Number of phase intervals over which clock is active** setting to 3 so that the active-level duration is three phase intervals (60% duty cycle).

Dialog Box



Opening this dialog causes a running simulation to pause. See "Changing Source Block Parameters During Simulation" in the online Simulink® documentation for details.

Clock frequency
: The frequency of all output clock signals.

Number of phases
: The number of different phases, *N*, in the output vector.

Starting phase
: The vector index of the output signal to first become active.

Number of phase intervals over which clock is active
The duration of the active level for every output signal.

Active level
The active level, `High (1)` or `Low (0)`.

Output data type
The output data type. For information on the `Logical` and `Boolean` options of this parameter, see "Effects of Enabling and Disabling Boolean Support".

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Boolean — The block might output Boolean values depending on the **Output data type** parameter setting, as described in "Effects of Enabling and Disabling Boolean Support". To learn how to disable Boolean output support, see "Steps to Disabling Boolean Support".

See Also

Clock	Simulink
Counter	Signal Processing Blockset
Pulse Generator	Simulink
Event-Count Comparator	Signal Processing Blockset

Multiport Selector

Purpose

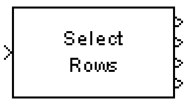
Distribute arbitrary subsets of input rows or columns to multiple output ports

Library

Signal Management / Indexing

`dspindex`

Description



The Multiport Selector block extracts multiple subsets of rows or columns from M-by-N input matrix u, and propagates each new submatrix to a distinct output port. A length-M 1-D vector input is treated as an M-by-1 matrix.

The **Indices to output** parameter is a cell array whose kth cell contains a one-dimensional indexing expression specifying the subset of input rows or columns to be propagated to the kth output port. The total number of cells in the array determines the number of output ports on the block.

When the **Select** parameter is set to `Rows`, the specified one-dimensional indices are used to select matrix rows, and all elements on the chosen rows are included. When the **Select** parameter is set to `Columns`, the specified one-dimensional indices are used to select matrix columns, and all elements on the chosen columns are included. A given input row or column can appear any number of times in any of the outputs, or not at all.

When an index references a nonexistent row or column of the input, the block reacts with the behavior specified by the **Invalid index** parameter. The following options are available:

- `Clip index` — Clip the index to the nearest valid value, and do not issue an alert.

  Example: For a 64-by-4 input with **Select** = `Rows`, an index of 72 is clipped to 64; with **Select** = `Columns`, an index of 72 is clipped to 4. In both cases, an index of -2 is clipped to 1.

- `Clip and warn` — Display a warning message in the MATLAB® Command Window, and clip as above.

- `Generate error` — Display an error dialog and terminate the simulation.

Examples

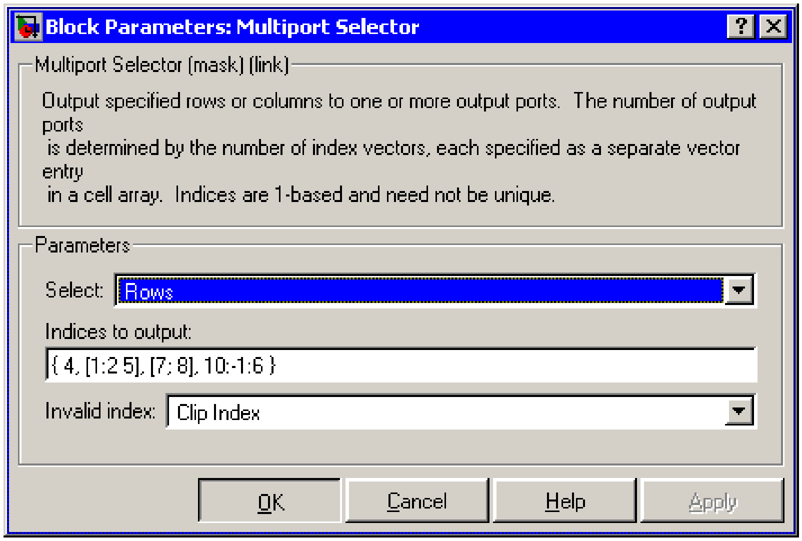
Consider the following **Indices to output** cell array:

```
{4,[1:2
5],[7;8],10:-1:6}
```

This is a four-cell array, which requires the block to generate four independent outputs (each at a distinct port). The table below shows the dimensions of these outputs when **Select** = `Rows` and the input dimension is M-by-N.

Cell	Expression	Description	Output Size
1	`4`	Row 4 of input	1-by-N
2	`[1:2 5]`	Rows 1, 2, and 5 of input	3-by-N
3	`[7;8]`	Rows 7 and 8 of input	2-by-N
4	`10:-1:6`	Rows 10, 9, 8, 7, and 6 of input	5-by-N

Dialog Box



Select

The dimension of the input to select, `Rows` or `Columns`.

Indices to output

A cell array specifying the row- or column-subsets to propagate to each of the output ports. The number of cells in the array determines the number of output ports on the block.

Invalid index

Response to an invalid index value.

Supported Data Types

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

See Also

Permute Matrix	Signal Processing Blockset
Selector	Simulink
Submatrix	Signal Processing Blockset
Variable Selector	Signal Processing Blockset

N-Sample Enable

Purpose

Output ones or zeros for specified number of sample times

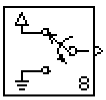
Library

- Signal Processing Sources

  `dspsrcs4`
- Signal Management / Switches and Counters

  `dspswit3`

Description



The N-Sample Enable block outputs the inactive value (`0` or `1`, whichever is not selected in the **Active level** parameter) during the first *N* sample times, where *N* is the **Trigger count** value. Beginning with output sample *N*+1, the block outputs the active value (`1` or `0`, whichever you select in the **Active level** parameter) until a reset event occurs or the simulation terminates.

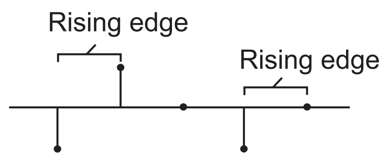
The output is always sample based.

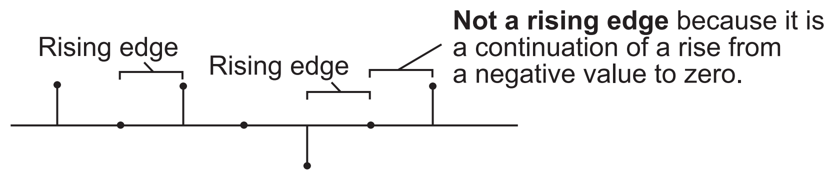
The **Reset input** check box enables the Rst input port. At any time during the count, a trigger event at the input port resets the counter to its initial state. The reset sample time must be a positive integer multiple of the input sample time. This block supports triggered subsystems when you select the **Reset input** check box.

You specify the triggering event in the **Trigger type** pop-up menu:

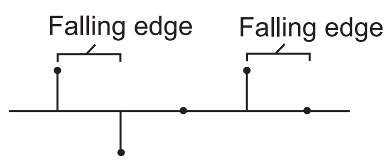
- `Rising edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Rises from a negative value to a positive value or zero

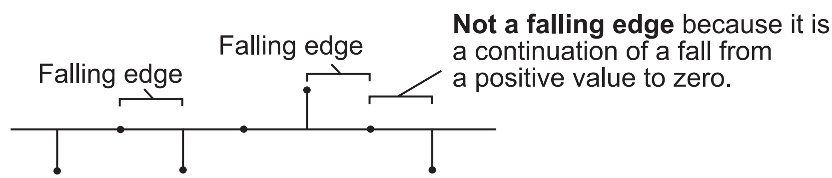
- Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)





- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)

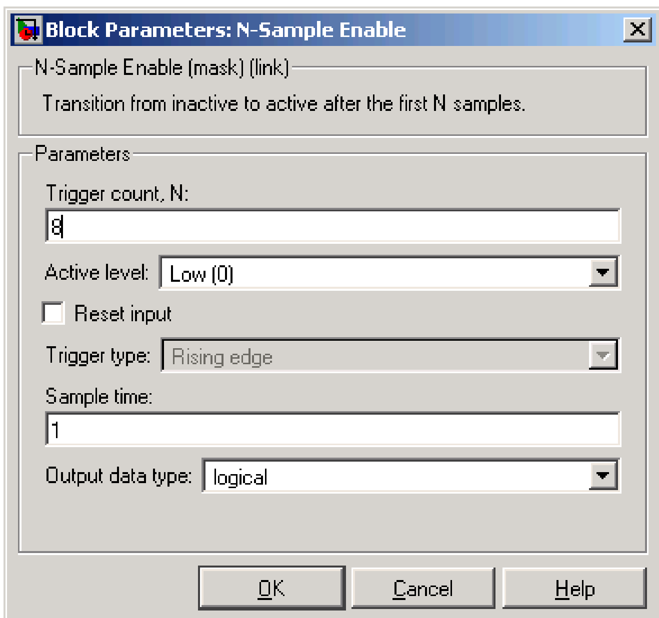




- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described above).
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero.

Note When running simulations in the Simulink® `MultiTasking` mode, sample-based reset signals have a one-sample latency, and frame-based reset signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a reset event, and when it applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Dialog Box



Opening this dialog causes a running simulation to pause. See "Changing Source Block Parameters During Simulation" in the online Simulink documentation for details.

Trigger count
Specify the number of samples for which the block outputs the active value. Tunable.

Active level
Specify the value to output after the first *N* sample times, `0` or `1`. Tunable.

Reset input
Select to enable the Rst input port. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input.

Trigger type
Select type of event that triggers a reset when the Rst port is enabled.

Sample time
Specify the sample period, T_s, for the block's counter. The block switches from the active value to the inactive value at $t=T_s*(N+1)$.

Output data type
Select the output data type. For information on the `Logical` and `Boolean` options of this parameter, see "Effects of Enabling and Disabling Boolean Support".

Supported Data Types

- Double-precision floating point
- Boolean — The block accepts Boolean inputs to the Rst port, which is enabled when you set the **Reset input** parameter. The block might output Boolean values depending on the **Output data type** parameter setting, as described in "Effects of Enabling and Disabling Boolean Support". To learn how to disable Boolean output support, see "Steps to Disabling Boolean Support".

See Also

Counter	Signal Processing Blockset
N-Sample Switch	Signal Processing Blockset

Purpose

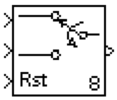
Switch between two inputs after specified number of sample periods

Library

Signal Management / Switches and Counters

`dspswit3`

Description



The N-Sample Switch block outputs the signal connected to the top input port during the first N sample times after the simulation begins or the block is reset, where you specify N in the **Switch count** parameter. Beginning with output sample N+1, the block outputs the signal connected to the bottom input until the next reset event or the end of the simulation.

You specify the sample period of the output in the **Sample time** parameter (that is, the output sample period is not inherited from the sample period of either input). The block applies a zero-order hold at the input ports, so the value the block reads from a given port between input sample times is the value of the most recent input to that port.

Both inputs must have the same dimension, except in the following two cases:

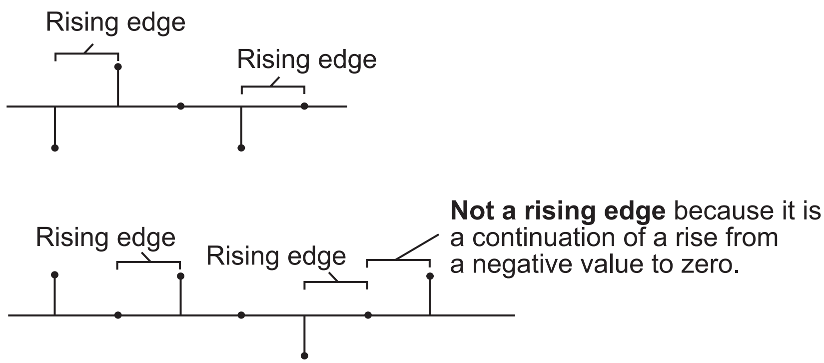
- When one input is a scalar, the block expands the scalar input to match the size of the other input.
- When one input is a 1-D vector and the other input is a row or column vector with the same number of elements, the block reshapes the 1-D vector to match the dimension of the other input.

The inputs must either both be frame based or both be sample based.

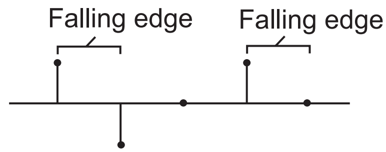
The **Reset input** check box enables the Rst input port. At any time during the count, a trigger event at the Rst port resets the counter to zero. The reset sample time must be a positive integer multiple of the input sample time. This block supports triggered subsystems when you select the **Reset input** check box.

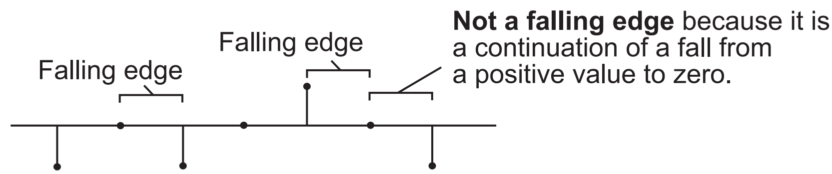
You specify the triggering event in the **Trigger type** pop-up menu, and can be one of the following:

- `Rising edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)

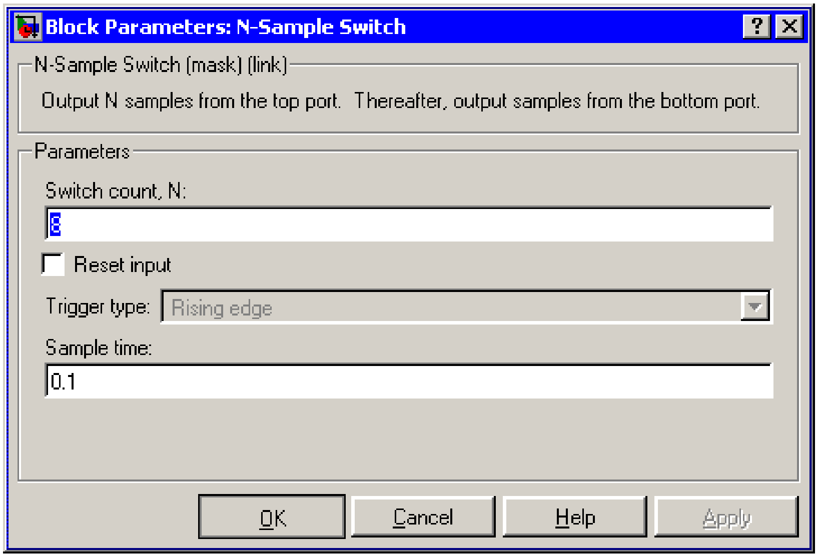




- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described above).
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero.

Note When running simulations in the Simulink® `MultiTasking` mode, sample-based reset signals have a one-sample latency, and frame-based reset signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a reset event, and when it applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Dialog Box



Switch count

The number of sample periods, N, for which the output is connected to the top input before switching to the bottom input. Tunable.

Reset input

Enables the Rst input port when selected. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input.

Trigger type

The type of event at the Rst port that resets the block's counter. This parameter is enabled when you select **Reset input**. Tunable.

Sample time

The sample period, T_s, for the block's counter. The block switches inputs at $t=T_s*(N+1)$.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- Boolean — The block accepts Boolean inputs to the Rst port, which is enabled when you set the **Reset input** parameter.
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Counter	Signal Processing Blockset
N-Sample Enable	Signal Processing Blockset

Purpose

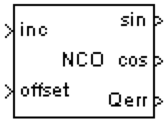
Generate real or complex sinusoidal signals

Library

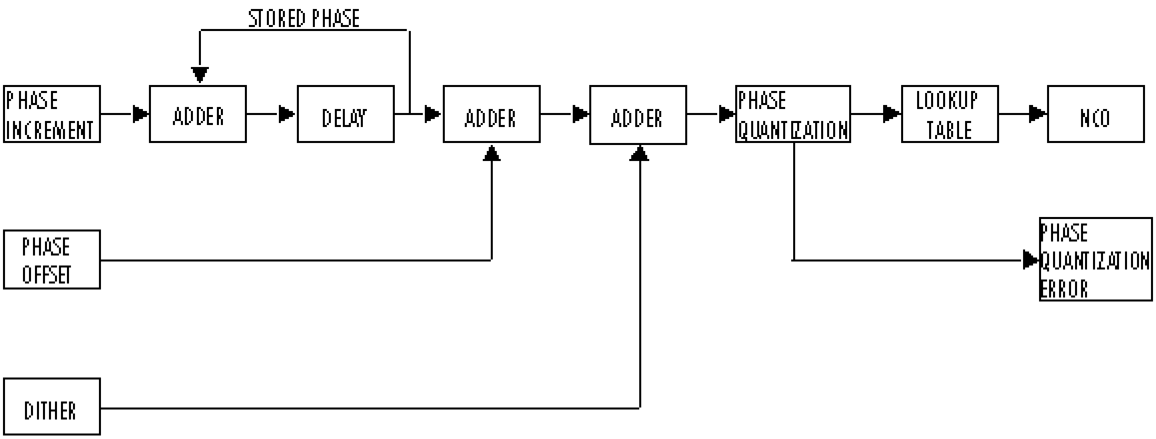
Signal Operations

`dspsigops`

Description



The NCO block generates a multichannel real or complex sinusoidal signal, with independent frequency and phase in each output channel. The amplitude of the created signal is always 1. The block implements the algorithm as shown in the following diagram:



The implementation of a numerically controlled oscillator (NCO) has two distinct parts. First, a phase accumulator accumulates the phase increment and adds in the phase offset. In this stage, an optional internal dither signal can also be added. The NCO output is then calculated by quantizing the results of the phase accumulator section and using them to select values from a lookup table.

Given a desired output frequency F_0, calculate the value of the **Phase increment** block parameter with

$$phase\ increment = \left(\frac{F_0 \cdot 2^N}{F_s}\right)$$

where N is the accumulator word length and

$$F_s = \frac{1}{T_s} = \frac{1}{sample\ time}$$

The frequency resolution of an NCO is defined by

$$\Delta f = \frac{1}{T_s \cdot 2^N}\ \mathrm{Hz}$$

Given a desired phase offset (in radians), calculate the **Phase offset** block parameter with

$$phase\ offset = \frac{2^N \cdot desired\ phase\ offset}{2\pi}$$

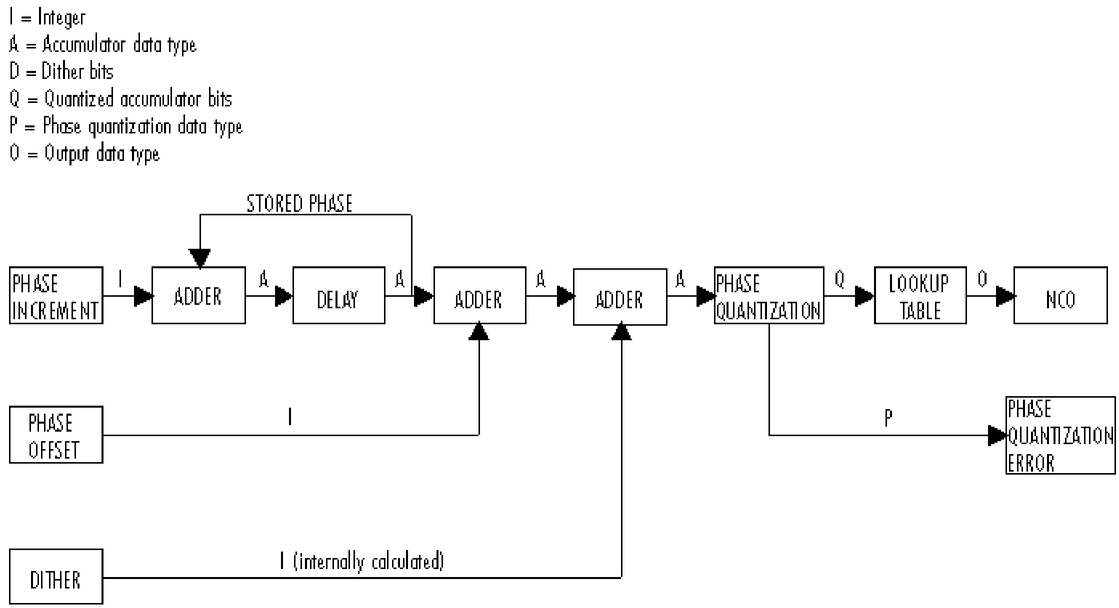
The spurious free dynamic range (SFDR) is estimated as follows for a lookup table with 2^P entries, where P is the number of quantized accumulator bits:

$$SFDR = (6P)\ \mathrm{dB} \qquad \text{without dither}$$
$$SFDR = (6P + 12)\ \mathrm{dB} \qquad \text{with dither}$$

The NCO block supports real inputs only. All outputs are real except for the output signal in `Complex exponential` mode.

Fixed-Point Data Types

The following diagram shows the data types used within the NCO block.



- You can set the accumulator and output data types in the block dialog as discussed in “Dialog Box” on page 2-895 below.
- The phase increment and phase offset inputs must be integers or fixed-point data types with zero fraction length.
- You specify the number of quantized accumulator bits in the **Number of quantized accumulator bits** parameter.
- The phase quantization error word length is equal to the accumulator word length minus the number of quantized accumulator bits, and the fraction length is zero.

Examples

Design an NCO source with the following specifications:

- Desired output frequency $F_0 = 510$ Hz
- Frequency resolution $\Delta f = 0.05$ Hz
- Spurious free dynamic range $SFDR \geq 90$ dB
- Sample period $T_s = 1/8000$ s
- Desired phase offset $\pi/2$

1

Calculate the number of required accumulator bits from the equation for frequency resolution:

$$\Delta f = \frac{1}{T_s \cdot 2^N}\,\text{Hz}$$

$$0.05 = \frac{1}{\frac{1}{8000} \cdot 2^N}\,\text{Hz}$$

$$N = 18$$

Note that N must be an integer value. The value of N is rounded up to the nearest integer; 18 accumulator bits are needed to accommodate the value of the frequency resolution.

2

Using this best value of N, calculate the frequency resolution that will be achieved by the NCO block:

$$\Delta f = \frac{1}{T_s \cdot 2^N}\,\text{Hz}$$

$$\Delta f = \frac{1}{\frac{1}{8000} \cdot 2^{18}}\ \text{Hz}$$

$$\Delta f = 0.0305$$

3

Calculate the number of quantized accumulator bits from the equation for spurious free dynamic range and the fact that for a lookup table with 2^P entries, P is the number of quantized accumulator bits:

$$SFDR = (6P + 12)\,\text{dB}$$

$$96\ \text{dB} = (6P + 12)\ \text{dB}$$

$$P = 14$$

4

Select the number of dither bits. In general, a good choice for the number of dither bits is the accumulator word length minus the output word length; in this case 4.

5

Calculate the phase increment:

$$phase\ increment = \text{round}\left(\frac{F_0 \cdot 2^N}{F_s}\right)$$

$$phase\ increment = \text{round}\left(\frac{501 \cdot 2^{18}}{8000}\right)$$

$$phase\ increment = 16417$$

6

Calculate the phase offset:

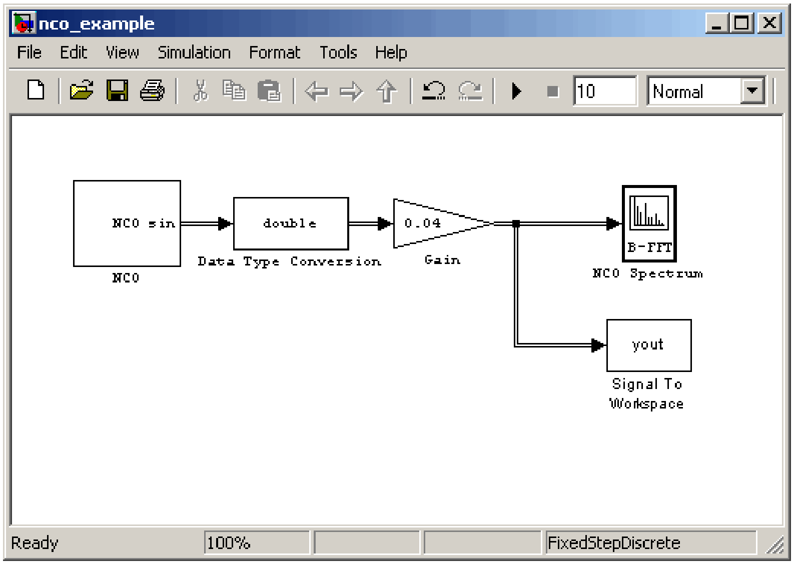
$$phase\ offset = \frac{2^{accumulator\ word\ length} \cdot desired\ phase\ offset}{2\pi}$$

$$phase\ offset = \frac{2^{18} \cdot \frac{\pi}{2}}{2\pi}$$

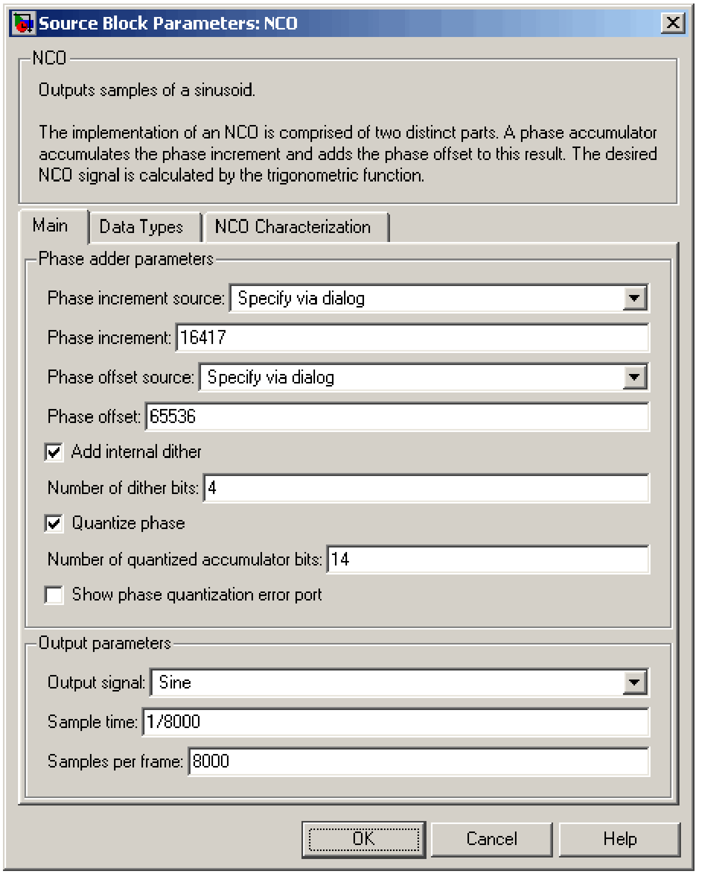
$$phase\ offset = 65536$$

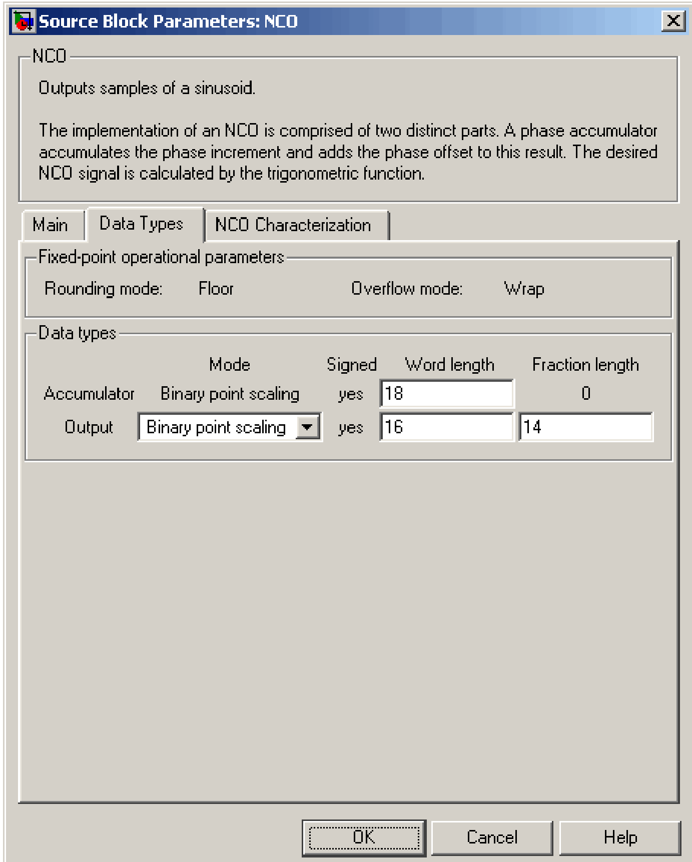
7

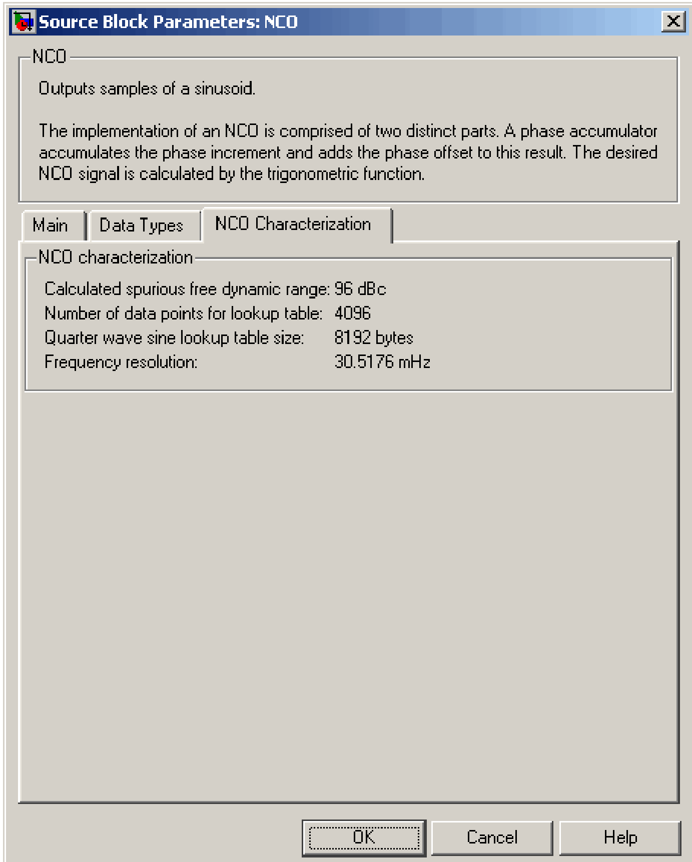
Type `doc_nco_example` at the MATLAB® command line to open the following model:



The NCO block in the model is populated with the specifications and quantities you just calculated. The output word length and fraction length depend on the constraints of your hardware; this example uses a word length of 16 and a fraction length of 14. The three panes of the block mask appear as follows.




Source Block Parameters: NCO
NCO
Outputs samples of a sinusoid.
The implementation of an NCO is comprised of two distinct parts. A phase accumulator accumulates the phase increment and adds the phase offset to this result. The desired NCO signal is calculated by the trigonometric function.
Main
Data Types
NCO Characterization
Fixed-point operational parameters
Rounding mode: Floor
Overflow mode: Wrap
Data types
Mode
Signed
Word length
Fraction length
Accumulator Binary point scaling yes 18 0
Output Binary point scaling yes 16 14
OK
Cancel
Help



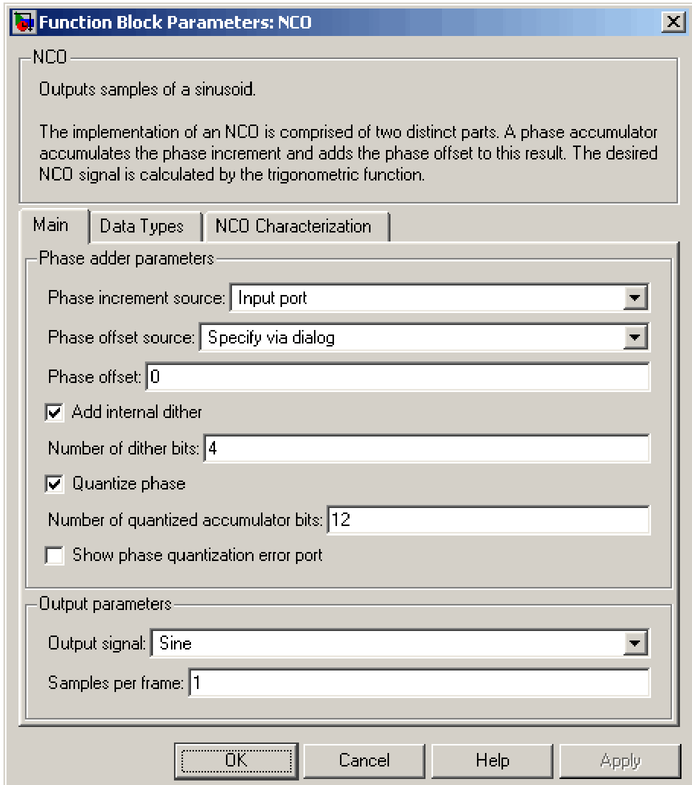
Looking at the **NCO Characterization** pane, you can verify that the specifications of this problem have been met.

8

Experiment with the model to observe the effects on the output shown on the Spectrum Scope. For example, try turning dithering on and off, and try changing the number of dither bits.

Dialog Box

The **Main** pane of the NCO dialog appears as follows.



Phase increment source

Choose how you specify the phase increment. The phase increment can come from an input port or from the dialog.

- If you select `Input port`, the inc port appears on the block icon.
- If you select `Specify via dialog`, the **Phase increment** parameter appears.

Phase increment
Specify the phase increment. Only integer data types, including fixed-point data types with zero fraction length, are allowed.

This parameter is visible only if `Specify via dialog` is selected for the **Phase increment source** parameter.

Phase offset source
Choose how you specify the phase offset. The phase offset can come from an input port or from the dialog.

- If you select `Input port`, the offset port appears on the block icon.
- If you select `Specify via dialog`, the **Phase offset** parameter appears.

Phase offset
Specify the phase offset. Only integer data types, including fixed-point data types with fraction length, are allowed.

This parameter is visible only if `Specify via dialog` is selected for the **Phase offset source** parameter.

Add internal dither
Select to add internal dithering to the NCO algorithm. Dithering is added using the PN Sequence Generator from the Communications Blockset™ product.

Number of dither bits
Specify the number of dither bits.

This parameter is visible only if **Add internal dither** is selected.

Quantize phase
Select to enable quantization of the accumulated phase.

Number of quantized accumulator bits
Specify the number of quantized accumulator bits. This determines the number of entries in the lookup table. The number

of quantized accumulator bits must be less than the accumulator word length.

This parameter is visible only if **Quantize phase** is selected.

Show phase quantization error port
Select to output the phase quantization error. When you select this, the Qerr port appears on the block icon.

This parameter is visible only if **Quantize phase** is selected.

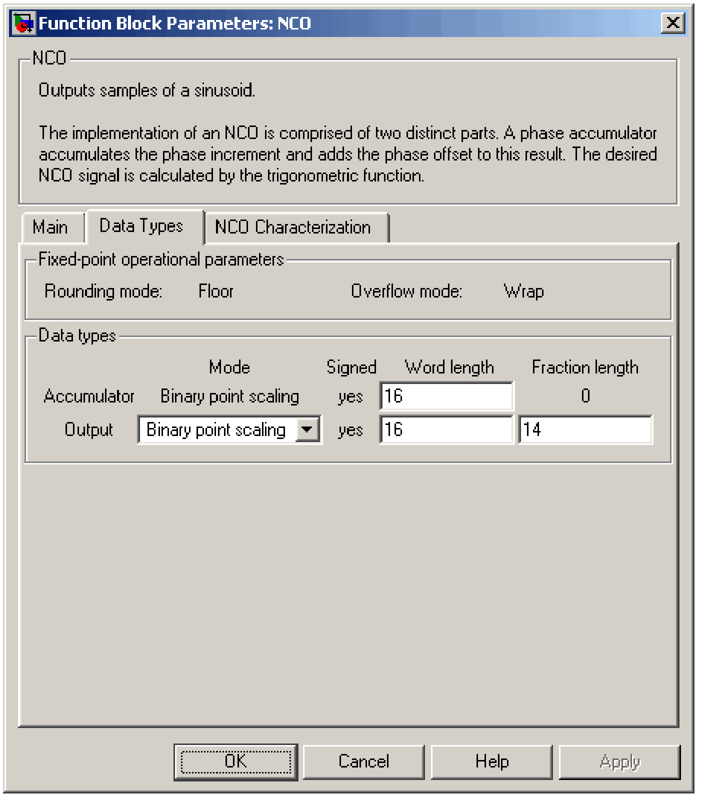
Output signal
Choose whether the block should output a `Sine`, `Cosine`, `Complex exponential`, or `Sine and cosine` signals. If you select `Sine and cosine`, the two signals output on different ports.

Sample time
Specify the sample time in seconds when the block is acting as a source. When either the phase increment or phase offset come in via block input ports, the sample time is inherited and this parameter is not visible.

Samples per frame
Specify the number of samples per frame. When the value of this parameter is `1`, the block outputs a sample-based signal.

When it exists, the phase offset input port has the same frame status as the output port(s). The phase increment input port, when it exists, does not support frames.

The **Data Types** pane of the NCO dialog appears as follows.



Rounding mode

The rounding mode used for this block when inputs are fixed point is always `Floor`.

Overflow mode

The overflow mode used for this block when inputs are fixed point is always `Wrap`.

Accumulator

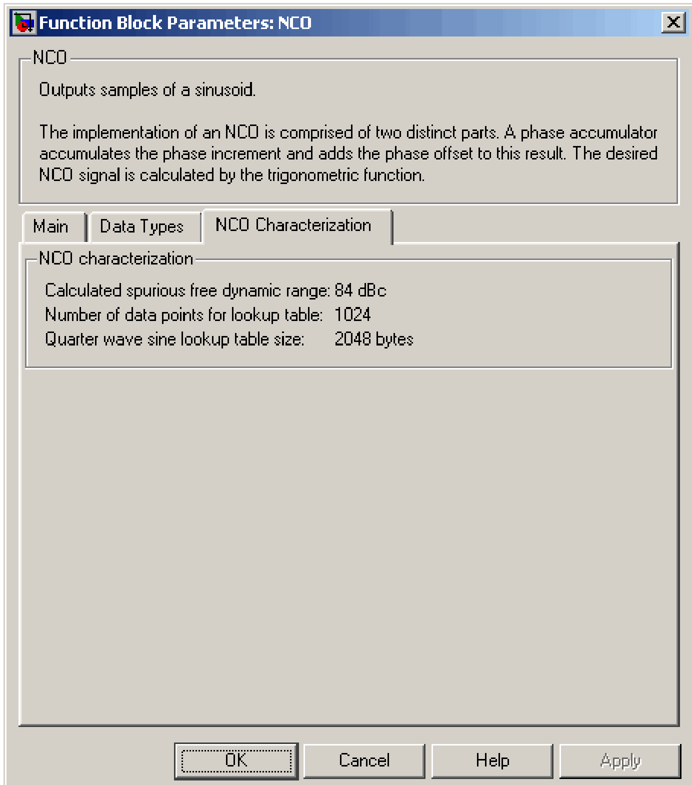
Specify the word length of the accumulator data type. The fraction length is always zero; this is an integer data type.

Output

Specify the output data type.

- Choose `double` or `single` for a floating-point implementation.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.

The **NCO Characterization** pane of the NCO dialog appears as follows.



The **NCO Characterization** pane does not have any parameters. Instead, it provides you with details on the NCO signal currently being implemented by the block:

- `Calculated spurious free dynamic range` — The spurious free dynamic range (SFDR) is calculated as follows for a lookup table with 2^P entries:

$$SFDR = (6P) \text{ dB} \qquad \text{without dither}$$
$$SFDR = (6P + 12) \text{ dB} \qquad \text{with dither}$$

- `Number of data points for lookup table` — The lookup table is implemented as a quarter-wave sine table. The number of lookup table data points is defined by

$$2^{\text{number of quantized accumulator bits}-2}$$

- `Quarter wave sine lookup table size` — The quarter wave sine lookup table size is defined by

$$\frac{(\text{number of lookup table data points}) \cdot (\text{output word length})}{8} \text{ bytes}$$

Supported Data Types

Port	Supported Data Types
inc	• Fixed point (signed) with zero fraction length • 8-, 16-, and 32-bit signed integers
offset	• Fixed point (signed) with zero fraction length • 8-, 16-, and 32-bit signed integers
sin	• Double-precision floating point • Single-precision floating point • Fixed point (signed)
Qerr	• 8-, 16-, and 32-bit signed integers

See Also

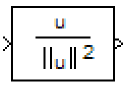
PN Sequence Generator	Communications Blockset
Sine Wave	Signal Processing Blockset

Purpose Perform vector normalization along rows, columns, or specified dimension

Library Math Functions / Math Operations

`dspmathops`

Description



The Normalization block independently normalizes each row, column, or vector of the specified dimension of the input. The Normalization block accepts real and complex inputs. The block only accepts floating-point signals for the 2-norm mode, and both fixed-point and floating-point signals for the squared 2-norm mode. The output always has the same dimensions and frame status as the input.

This block treats an arbitrarily dimensioned input *U* as a collection of vectors oriented along the specified dimension. The block normalizes these vectors by either their norm or the square of their norm.

For example, consider a 3-dimensional input *U(i,j,k)* and assume that you want to normalize along the second dimension. First, define the 2-dimensional intermediate quantity *V(i,k)* such that each element of *V* is the norm of one of the vectors in *U*:

$$V(i,k) = \left(\sum_{j=1}^{J} U^2(i,j,k) \right)^{1/2}$$

Given *V*, the output of the block *Y(i, j,k)* in 2-norm mode is

$$Y(i,j,k)\frac{U(i,j,k)}{V(i,k)+b}$$

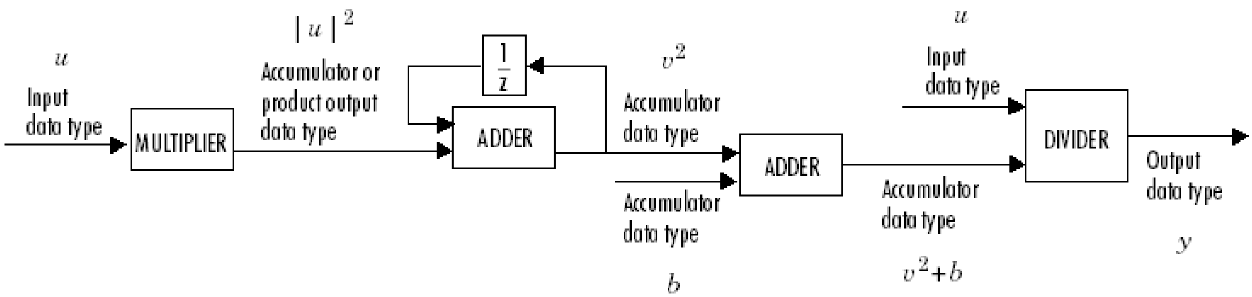
In squared 2-norm mode, the block output is

$$Y(i,j,k) = \frac{U(i,j,k)}{V(i,k)^2 + b}$$

The normalization bias, b, is typically chosen to be a small positive constant (for example, 1e-10) that prevents potential division by zero.

Fixed-Point Data Types

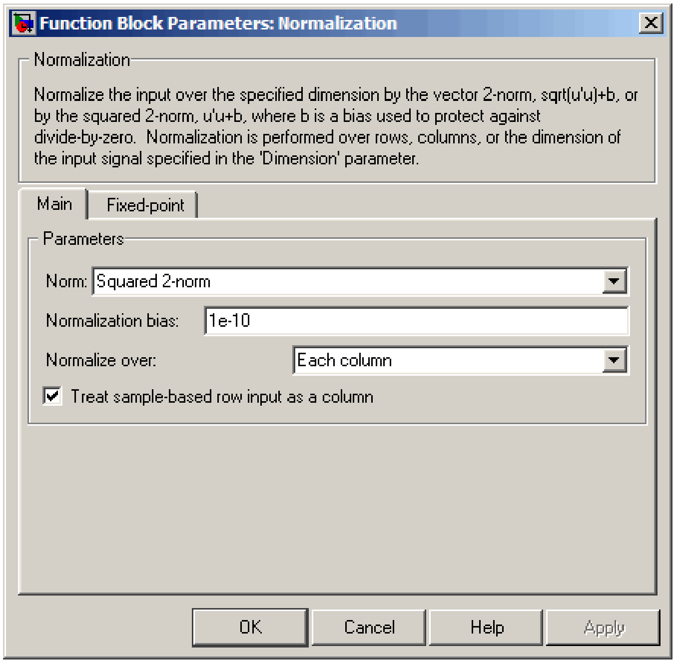
The following diagram shows the data types used within the Normalization block for fixed-point signals (squared 2-norm mode only).



The output of the multiplier is in the product output data type when the input is real. When the input is complex, the result of the multiplication is in the accumulator data type. For details on the complex multiplication performed, see “Multiplication Data Types”. You can set the accumulator, product output, and output data types in the block dialog as discussed in “Dialog Box” on page 2-904.

Dialog Box

The **Main** pane of the Normalization dialog appears as follows.



Norm

Specify the type of normalization to perform, `2-norm` or `Squared 2-norm`. `2-norm` mode supports floating-point signals only. `Squared 2-norm` supports both fixed-point and floating-point signals.

Normalization bias

Specify the real value *b* to be added in the denominator to avoid division by zero. Tunable.

Normalize over

Specify whether to normalize along rows, columns, or the dimension specified in the **Dimension** parameter.

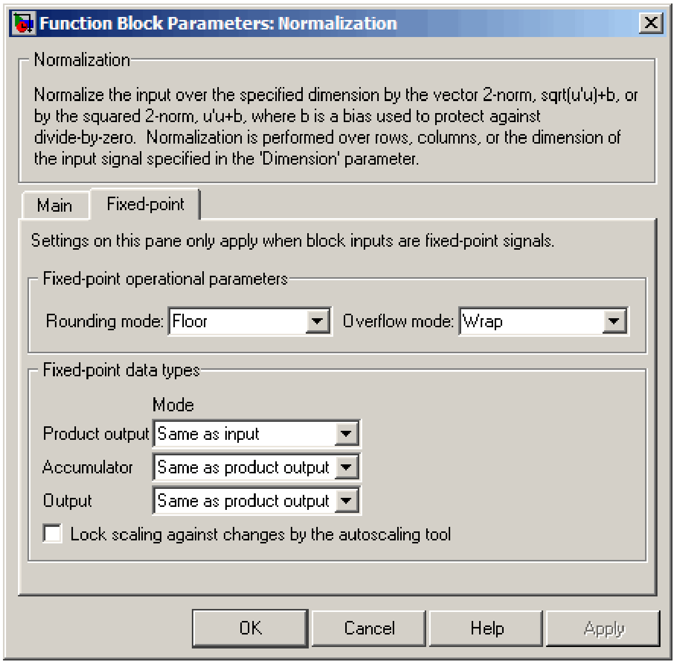
Dimension

Specify the one-based value of the dimension over which to normalize. The value of this parameter cannot exceed the number of dimensions in the input signal. This parameter is only visible if `Specified dimension` is selected for the **Normalize over** parameter.

Treat sample-based row input as column

Select to treat a sample-based row input as a column.

The **Fixed-Point** pane of the Normalization dialog appears as follows.



Note The parameters on this pane are only applicable to fixed-point signals when the block is in squared 2-norm mode. See "Fixed-Point Data Types" on page 2-903 for a diagram of how the product output, accumulator, and output data types are used in this case.

Rounding mode
: Select the rounding mode for fixed-point operations.

Overflow mode
: Select the overflow mode for fixed-point operations.

Product output
: Use this parameter to specify how you would like to designate the product output word and fraction lengths:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator
: Use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. The bias b is also quantized into the accumulator data type:

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

- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.
- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

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

See Also

Array-Vector Multiply	Signal Processing Blockset
Reciprocal Condition	Signal Processing Blockset
`norm`	MATLAB

Purpose

Design Nyquist filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description

This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See "Nyquist Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Design octave filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description

This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See "Octave Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Truncate vectors by removing or keeping beginning or ending values

Library

Signal Operations

`dspsigops`

Description

The Offset block removes or keeps values from the beginning or end of a vector and outputs the result in a vector of user-specified length. The inputs to the In ports (In1, In2, ...) can be scalars or vectors, but they must be the same size and data type. The offset values are the inputs to the O ports (O1, O2, ...); they must be scalar values with the same data type. These offset values should be integer values because they determine the number of values the block discards or retains from each input vector. The block rounds any offset value that is a noninteger value to the nearest integer value. There is one output port for each pair of In and O ports. This block supports sample-based and frame-based signals.

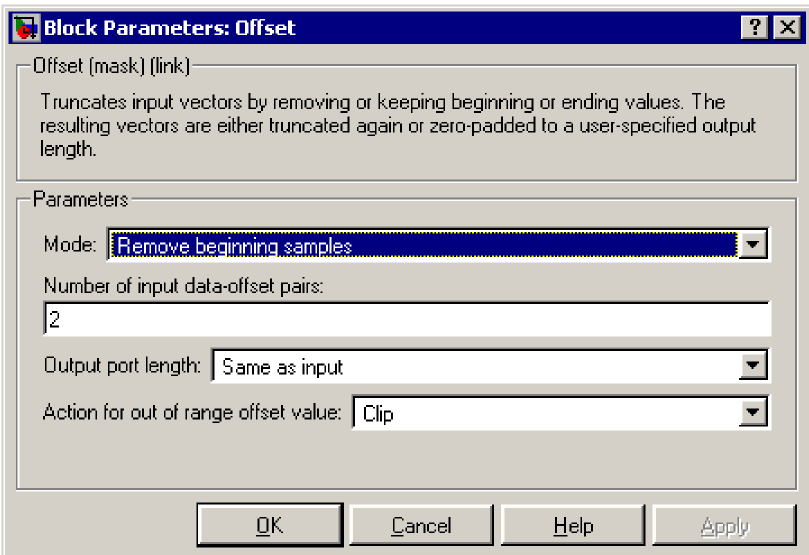
Use the **Mode** parameter to determine which values the block discards or retains from the input vector. To discard the initial values of the vector, select `Remove beginning samples`. To discard the final values of the vector, select `Remove ending samples`. To retain the initial values of the vector, select `Keep beginning samples`. To retain the final values of a vector, select `Keep ending` samples.

Use the **Number of input data-offset pairs** parameter to specify the number of inputs to the block. The number of input ports is twice the scalar value you enter. For example, if you enter `3`, ports In1, O1, In2, O2, In3, and O3 appear on the block.

The block uses the **Output port length** parameter to determine the length of the output vectors. If you select `Same as input`, the block outputs vectors that are the same length as the input to the In ports. If you select `User-defined`, the **Output length** parameter appears. Enter a scalar that represents the desired length of the output vectors. If your desired output length is greater than the number of values you extracted from your input vector, the block zero-pads the end of the vector to reach the length you specified.

Use the **Action for out of range offset value** parameter to determine how the block behaves when an offset value is not in the range 0 ≤ offset value ≤ N, where N is the input vector length. Select `Clip` if you want any offset values less than 0 to be set to 0 and any offset values greater than N to be set to N. Select `Clip and warn` if you want to be warned when any offset values less than 0 are set to 0 and any offset values greater than N are set to N. Select `Error` if you want the simulation to stop and display an error when the offset values are out of range.

Dialog Box



Mode

Use this parameter to determine which values the block discards or retains from the input vector. Your choices are `Remove beginning samples`, `Remove ending samples`, `Keep beginning samples`, and `Keep ending samples`.

Number of input data-offset pairs

Specify the number of inputs to the block. The number of input ports is twice the scalar value you enter.

Output port length

Use this parameter to specify the length of the output vectors. If you select `Same as input`, the output vectors are the same length as the input vectors. If you select `User-defined`, you can enter the desired length of the output vectors.

Output length

Enter a scalar that represents the desired length of the output vectors. This parameter is visible if, for the **Output port length** parameter, you select `User-defined`.

Action for out of range offset value

Use this parameter to determine how the block behaves when an offset value is not in the range such that 0 ≤ offset value ≤ N, where N is the input vector length. When you want any offset values less than 0 to be set to 0 and any offset values greater than N to be set to N, select `Clip`. When you want to be warned when any offset values less than 0 are set to 0 and any offset values greater than N are set to N, select `Clip and warn`. When you want the simulation to stop and display an error when the offset values are out of range, select `Error`.

Supported Data Types

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

Purpose

Implement overlap-add method of frequency-domain filtering

Library

Filtering / Filter Designs

`dsparch4`

Description

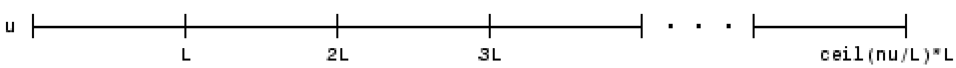
The Overlap-Add FFT Filter block uses an FFT to implement the *overlap-add method*, a technique that combines successive frequency-domain filtered sections of an input sequence.

Valid inputs to this block are 1-D vectors, sample-based vectors, frame-based vectors, and frame-based full matrices. All outputs are unbuffered into sample-based row vectors. The length of the output vector is equal to the number of channels in the input vector. An M-by-1 *sample-based* input has M channels, so it would result in a length-M sample-based output vector. An M-by-1 *frame-based* input has only one channel, so would result in a 1-by-1 (scalar) output.

The block's data output rate is M times faster than its data input rate, where M is the input frame-size. Thus, the block's data input and output rates are the same when the inputs are 1-D vectors, sample-based vectors, or frame-based row vectors. For frame-based column and frame-based full-matrix inputs, the block's data output rate is M times greater than the block's data input rate.

1-D vectors are treated as length-N sample-based vectors, and result in sample-based length-N row vectors.

The block breaks the scalar input sequence `u`, of length `nu`, into length-`L` nonoverlapping data sections,



which it linearly convolves with the filter's FIR coefficients,

$$H(z) = B(z) = b_1 + b_2 z^{-1} + \ldots + b_{n+1} z^{-n}$$

The numerator coefficients for H(z) are specified as a vector by the **FIR coefficients** parameter. The coefficient vector, `b = [b(1) b(2) ...`

b(n+1)], can be generated by one of the filter design functions in the Signal Processing Toolbox™ product, such as fir1. All filter states are internally initialized to zero.

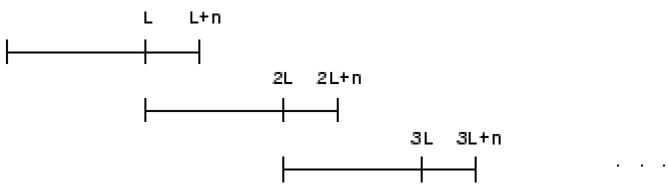
When either the filter coefficients or the inputs to the block are complex, the **Output** parameter should be set to Complex. Otherwise, the default **Output** setting, Real, instructs the block to take only the real part of the solution.

The block's overlap-add operation is equivalent to

```
y = ifft(fft(u(i:i+L-1),nfft) .* fft(b,nfft))
```

where you specify nfft in the **FFT size** parameter as a power-of-two value greater (typically *much* greater) than n+1. Values for **FFT size** that are not powers of two are rounded upwards to the nearest power-of-two value to obtain nfft.

The block overlaps successive output sections by n points and sums them.



The first L samples of each summation are output in sequence. The block chooses the parameter L based on the filter order and the FFT size.

```
L = nfft - n
```

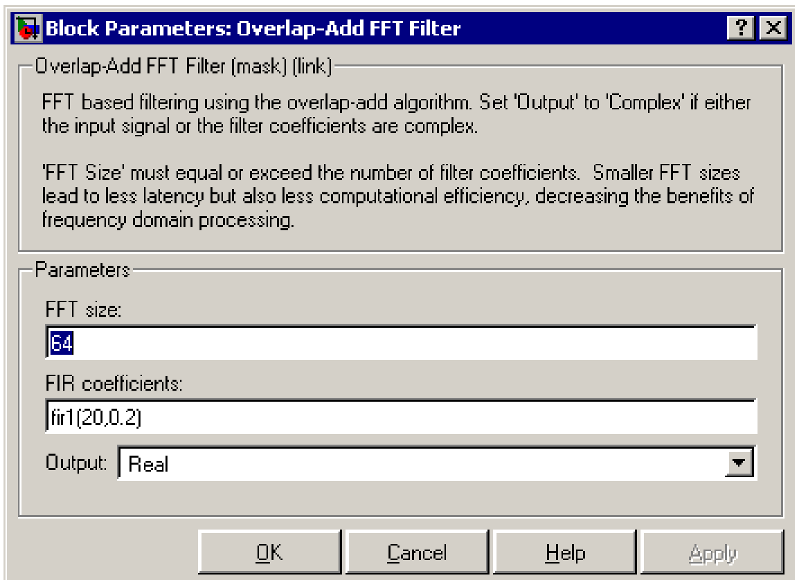
Latency

In *single-tasking* operation, the Overlap-Add FFT Filter block has a latency of nfft-n+1 samples. The first nfft-n+1 consecutive outputs from the block are zero; the first filtered input value appears at the output as sample nfft-n+2.

In *multitasking* operation, the Overlap-Add FFT Filter block has a latency of `2*(nfft-n+1)` samples. The first `2*(nfft-n+1)` consecutive outputs from the block are zero; the first filtered input value appears at the output as sample `2*(nfft-n)+3`.

Note For more information on latency and the Simulink® software tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Dialog Box



FFT size

The size of the FFT, which should be a power-of-two value greater than the length of the specified FIR filter.

FIR coefficients

The filter numerator coefficients.

Output

The complexity of the output; `Real` or `Complex`. When the input signal or the filter coefficients are complex, this should be set to `Complex`.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Overlap-Save FFT Filter	Signal Processing Blockset™ product

Purpose

Implement overlap-save method of frequency-domain filtering

Library

Filtering / Filter Designs

`dsparch4`

Description

The Overlap-Save FFT Filter block uses an FFT to implement the *overlap-save method*, a technique that combines successive frequency-domain filtered sections of an input sequence.

Valid inputs to this block are 1-D vectors, sample-based vectors, frame-based vectors, and frame-based full matrices. All outputs are unbuffered into sample-based row vectors. The length of the output vector is equal to the number of channels in the input vector. An M-by-1 sample-based input has M channels, so it would result in a length-M sample-based output vector. An M-by-1 frame-based input has only one channel, so would result in a 1-by-1 (scalar) output.

The block's data output rate is M times faster than its data input rate, where M is the input frame-size. Thus, the block's data input and output rates are the same when the inputs are 1-D vectors, sample-based vectors, or frame-based row vectors. For frame-based column and frame-based full-matrix inputs, the block's data output rate is M times greater than the block's data input rate.

1-D vectors are treated as length-N sample-based vectors, and result in sample-based length-N row vectors.

Overlapping sections of input `u` are circularly convolved with the FIR filter coefficients

$$H(z) = B(z) = b_1 + b_2 z^{-1} + \ldots + b_{n+1} z^{-n}$$

The numerator coefficients for $H(z)$ are specified as a vector by the **FIR coefficients** parameter. The coefficient vector, `b = [b(1) b(2) ... b(n+1)]`, can be generated by one of the filter design functions in the Signal Processing Toolbox™ product, such as `fir1`. All filter states are internally initialized to zero.

When either the filter coefficients or the inputs to the block are complex, the **Output** parameter should be set to `Complex`. Otherwise, the default **Output** setting, `Real`, instructs the block to take only the real part of the solution.

The circular convolution of each section is computed by multiplying the FFTs of the input section and filter coefficients, and computing the inverse FFT of the product.

```
y = ifft(fft(u(i:i+(L-1)),nfft) .* fft(b,nfft))
```

where you specify `nfft` in the **FFT size** parameter as a power of two value greater (typically *much* greater) than `n+1`. Values for **FFT size** that are not powers of two are rounded upwards to the nearest power-of-two value to obtain `nfft`.

The first `n` points of the circular convolution are invalid and are discarded. The Overlap-Save FFT Filter block outputs the remaining `nfft-n` points, which are equivalent to the linear convolution.

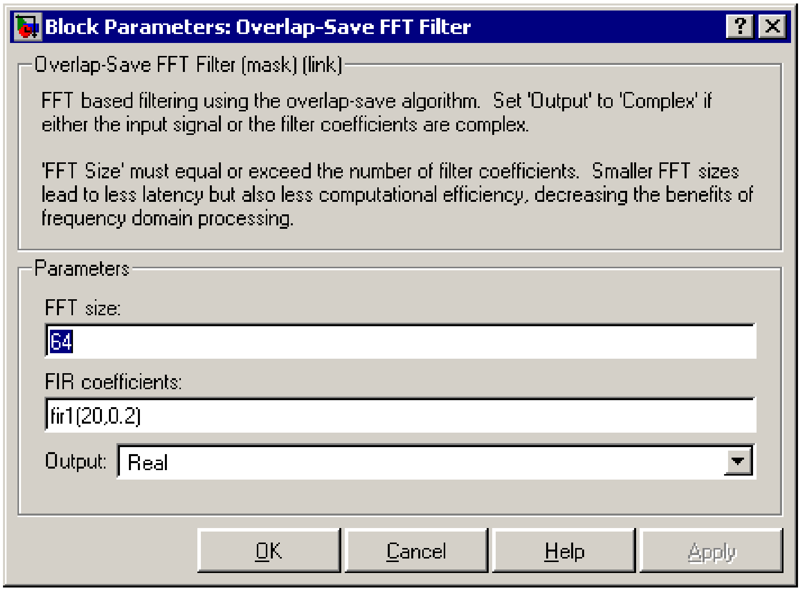
Latency

In *single-tasking* operation, the Overlap-Save FFT Filter block has a latency of `nfft-n+1` samples. The first `nfft-n+1` consecutive outputs from the block are zero; the first filtered input value appears at the output as sample `nfft-n+2`.

In *multitasking* operation, the Overlap-Save FFT Filter block has a latency of `2*(nfft-n+1)` samples. The first `2*(nfft-n+1)` consecutive outputs from the block are zero; the first filtered input value appears at the output as sample `2*(nfft-n)+3`.

Note For more information on latency and the Simulink® environment tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Dialog Box



FFT size
: The size of the FFT, which should be a power of two value greater than the length of the specified FIR filter.

FIR coefficients
: The filter numerator coefficients.

Output
: The complexity of the output; `Real` or `Complex`. When the input signal or the filter coefficients are complex, this should be set to `Complex`.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Overlap-Add FFT Filter	Signal Processing Blockset

Purpose

Overwrite submatrix or subdiagonal of input

Library

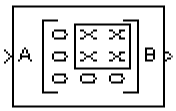
- Math Functions / Matrices and Linear Algebra / Matrix Operations

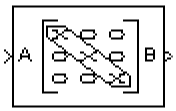
  `dspmtrx3`

- Signal Management / Indexing

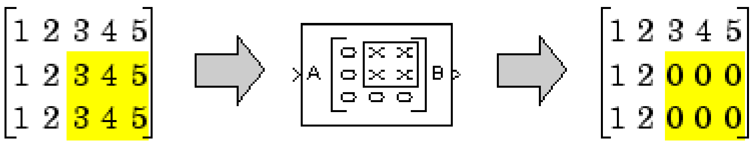
  `dspindex`

Description





The Overwrite Values block overwrites a contiguous submatrix or subdiagonal of an input matrix. You can provide the overwriting values by typing them in a block parameter, or through an additional input port, which is useful for providing overwriting values that change at each time step.



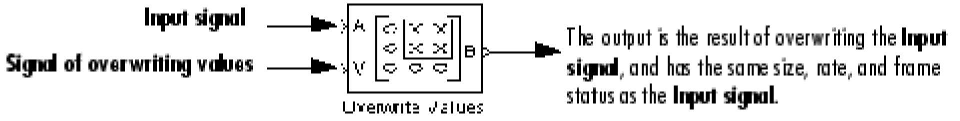
The block accepts both sample- and frame-based vectors and matrices. The output has the same size and frame status as the original input signal, not necessarily the same size and frame status as the signal containing the overwriting values. The input(s) and output of this block must have the same data type.

Specifying the Overwriting Values

The **Source of overwriting value(s)** parameter determines how you must provide the overwriting values, and has the following settings.

- `Specify via dialog` — You must provide the overwriting value(s) in the **Overwrite with** parameter. The block uses the same overwriting values to overwrite the specified portion of the input at each time step. To learn how to specify valid overwriting values, see “Valid Overwriting Values” on page 2-926.
- `Second input port` — You must provide overwriting values through a second block input port, `V`. Use this setting to provide different

overwriting values at each time step. The output inherits its size, rate, and frame status from the input signal, *not* the overwriting values.



The rate at which you provide the overwriting values through input port V must match the rate at which the block receives each input matrix at input port A. The rate requirements depend on whether the input signal and overwriting values signal have the same frame status:

- When both signals are sample based, their sample rates must be the same.
- When both signals are frame based, their frame rates must be the same.
- When one signal is sample based and one signal is frame based, the sample rate of the sample-based signal must be the same as the frame rate of the frame-based signal.

Valid Overwriting Values

The overwriting values can be a single constant, vector, or matrix, depending on the portion of the input you are overwriting, regardless of whether you provide the overwriting values through an input port or by providing them in the **Overwrite with** parameter.

Valid Overwriting Values

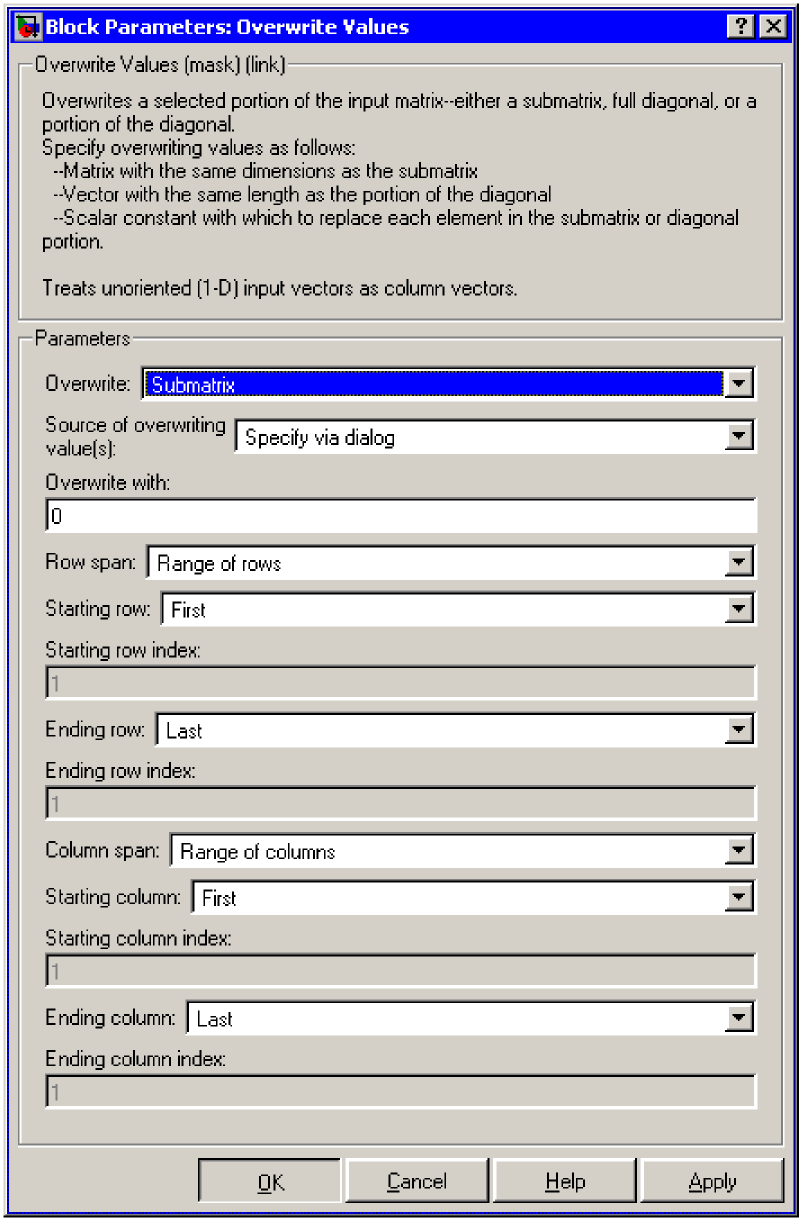
Portion of Input to Overwrite	Valid Overwriting Values	Example
A single element in the input $\begin{bmatrix} x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \end{bmatrix}$	Any constant value, v	$v = 9$ $\begin{bmatrix} x & x & x & x & x \\ x & x & x & 9 & x \\ x & x & x & x & x \\ x & x & x & x & x \end{bmatrix}$
A length-k portion of the diagonal $\begin{bmatrix} x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \end{bmatrix}$	Any length-k column or row vector, v	$k = 3 \quad v = [2 \quad 4 \quad 6] \quad \text{or} \quad \begin{bmatrix} 2 \\ 4 \\ 6 \end{bmatrix}$ $\begin{bmatrix} 2 & x & x & x & x \\ x & 4 & x & x & x \\ x & x & 6 & x & x \\ x & x & x & x & x \end{bmatrix}$
A length-k portion of a row $\begin{bmatrix} x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \end{bmatrix}$	Any length-k row vector, v	$k = 3 \quad v = [2 \quad 4 \quad 6]$ $\begin{bmatrix} x & x & x & x & x \\ x & 2 & 4 & 6 & x \\ x & x & x & x & x \\ x & x & x & x & x \end{bmatrix}$

Valid Overwriting Values (Continued)

Portion of Input to Overwrite	Valid Overwriting Values	Example
A length-k portion of a column $\begin{bmatrix} x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \end{bmatrix}$	Any length-k column vector, v	$k = 2 \quad v = \begin{bmatrix} 4 \\ 6 \end{bmatrix}$ $\begin{bmatrix} x & x & x & x & x \\ x & x & x & 4 & x \\ x & x & x & 6 & x \\ x & x & x & x & x \end{bmatrix}$
An m-by-n submatrix $\begin{bmatrix} x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \\ x & x & x & x & x \end{bmatrix}$	Any m-by-n matrix, v	$m = 2$ $n = 3 \quad v = \begin{bmatrix} 4 & 5 & 6 \\ 7 & 8 & 9 \end{bmatrix}$ $\begin{bmatrix} x & x & x & x & x \\ x & x & 4 & 5 & 6 \\ x & x & 7 & 8 & 9 \\ x & x & x & x & x \end{bmatrix}$

This block supports Simulink® virtual buses.

Dialog Box


Block Parameters: Overwrite Values
Overwrite Values (mask) (link)
Overwrites a selected portion of the input matrix--either a submatrix, full diagonal, or a portion of the diagonal.
Specify overwriting values as follows:
--Matrix with the same dimensions as the submatrix
--Vector with the same length as the portion of the diagonal
--Scalar constant with which to replace each element in the submatrix or diagonal portion.
Treats unoriented (1-D) input vectors as column vectors.
Parameters
Overwrite:
Submatrix
Source of overwriting value(s):
Specify via dialog
Overwrite with:
0
Row span:
Range of rows
Starting row:
First
Starting row index:
1
Ending row:
Last
Ending row index:
1
Column span:
Range of columns
Starting column:
First
Starting column index:
1
Ending column:
Last
Ending column index:
1
OK
Cancel
Help
Apply

Note Only some of the following parameters are visible in the dialog box at any one time.

Overwrite
: Determines whether to overwrite a specified submatrix or a specified portion of the diagonal.

Source of overwriting value(s)
: Determines where you must provide the overwriting values: either through an input port, or by providing them in the **Overwrite with** parameter. For more information, see "Specifying the Overwriting Values" on page 2-925.

Overwrite with
: The value(s) with which to overwrite the specified portion of the input matrix. Enabled only when **Source of overwriting value(s)** is set to `Specify via dialog`. To learn how to specify valid overwriting values, see "Valid Overwriting Values" on page 2-926.

Row span
: The range of input rows to be overwritten. Options are `All rows`, `One row`, or `Range of rows`. For descriptions of these options, see "Dialog Box" on page 2-929.

Row/Starting row
: The input row that is the first row of the submatrix that the block overwrites. For a description of the options for the **Row** and **Starting row** parameters, see Settings for Row, Column, Starting Row, and Starting Column Parameters on page 2-935. **Row** is enabled when **Row span** is set to `One row`, and **Starting row** when **Row span** is set to `Range of rows`.

Row index/Starting row index
: Index of the input row that is the first row of the submatrix that the block overwrites. See how to use these parameters in Settings for Row, Column, Starting Row, and Starting Column Parameters

on page 2-935. **Row index** is enabled when **Row** is set to `Index`, and **Starting row index** when **Starting row** is set to `Index`.

Row offset/Starting row offset
The offset of the input row that is the first row of the submatrix that the block overwrites. See how to use these parameters in Settings for Row, Column, Starting Row, and Starting Column Parameters on page 2-935. **Row offset** is enabled when **Row** is set to `Offset from middle` or `Offset from last`, and **Starting row offset** is enabled when **Starting row** is set to `Offset from middle` or `Offset from last`.

Ending row
The input row that is the last row of the submatrix that the block overwrites. For a description of this parameter's options, see Settings for Ending Row and Ending Column Parameters on page 2-936. This parameter is enabled when **Row span** is set to `Range of rows`, and **Starting row** is set to any option but `Last`.

Ending row index
Index of the input row that is the last row of the submatrix that the block overwrites. See how to use this parameter in Settings for Ending Row and Ending Column Parameters on page 2-936. Enabled when **Ending row** is set to `Index`.

Ending row offset
The offset of the input row that is the last row of the submatrix that the block overwrites. See how to use this parameter in Settings for Ending Row and Ending Column Parameters on page 2-936. Enabled when **Ending row** is set to `Offset from middle` or `Offset from last`.

Column span
The range of input columns to be overwritten. Options are `All columns`, `One column`, or `Range of columns`. For descriptions of the analogous row options, see "Dialog Box" on page 2-929.

Column/Starting column
The input column that is the first column of the submatrix that the block overwrites. For a description of the options for the

Column and **Starting column** parameters, see Settings for Row, Column, Starting Row, and Starting Column Parameters on page 2-935. **Column** is enabled when **Column span** is set to `One column`, and **Starting column** when **Column span** is set to `Range of columns`.

Column index/Starting column index

Index of the input column that is the first column of the submatrix that the block overwrites. See how to use these parameters in Settings for Row, Column, Starting Row, and Starting Column Parameters on page 2-935. **Column index** is enabled when **Column** is set to `Index`, and **Starting column index** when **Starting column** is set to `Index`.

Column offset/Starting column offset

The offset of the input column that is the first column of the submatrix that the block overwrites. See how to use these parameters in Settings for Row, Column, Starting Row, and Starting Column Parameters on page 2-935. **Column offset** is enabled when **Column** is set to `Offset from middle` or `Offset from last`, and **Starting column offset** is enabled when **Starting column** is set to `Offset from middle` or `Offset from last`.

Ending column

The input column that is the last column of the submatrix that the block overwrites. For a description of this parameter's options, see Settings for Ending Row and Ending Column Parameters on page 2-936. This parameter is enabled when **Column span** is set to `Range of columns`, and **Starting column** is set to any option but `Last`.

Ending column index

Index of the input column that is the last column of the submatrix that the block overwrites. See how to use this parameter in Settings for Ending Row and Ending Column Parameters on page 2-936. This parameter is enabled when **Ending column** is set to `Index`.

Ending column offset

The offset of the input column that is the last column of the submatrix that the block overwrites. See how to use this parameter in Settings for Ending Row and Ending Column Parameters on page 2-936. This parameter is enabled when **Ending column** is set to `Offset from middle` or `Offset from last`.

Diagonal span

The range of diagonal elements to be overwritten. Options are `All elements`, `One element`, or `Range of elements`. For descriptions of these options, see "Overwriting a Subdiagonal" on page 2-939.

Element/Starting element

The input diagonal element that is the first element in the subdiagonal that the block overwrites. For a description of the options for the **Element** and **Starting element** parameters, see Element and Starting Element Parameters on page 2-939. **Element** is enabled when **Element span** is set to `One element`, and **Starting element** when **Element span** is set to `Range of elements`.

Element index/Starting element index

Index of the input diagonal element that is the first element of the subdiagonal that the block overwrites. See how to use these parameters in Element and Starting Element Parameters on page 2-939. **Element index** is enabled when **Element** is set to `Index`, and **Starting element index** when **Starting element** is set to `Index`.

Element offset/Starting element offset

The offset of the input diagonal element that is the first element of the subdiagonal that the block overwrites. See how to use these parameters in Element and Starting Element Parameters on page 2-939. **Element offset** is enabled when **Element** is set to `Offset from middle` or `Offset from last`, and **Starting element offset** is enabled when **Starting element** is set to `Offset from middle` or `Offset from last`.

Ending element

The input diagonal element that is the last element of the subdiagonal that the block overwrites. For a description of this parameter's options, see Ending Element Parameters on page 2-940. This parameter is enabled when **Element span** is set to `Range of elements`, and **Starting element** is set to any option but `Last`.

Ending element index

Index of the input diagonal element that is the last element of the subdiagonal that the block overwrites. See how to use this parameter in Ending Element Parameters on page 2-940. This parameter is enabled when **Ending element** is set to `Index`.

Ending element offset

The offset of the input diagonal element that is the last element of the subdiagonal that the block overwrites. See how to use this parameter in Ending Element Parameters on page 2-940. This parameter is enabled when **Ending element** is set to `Offset from middle` or `Offset from last`.

Examples

Overwriting a Submatrix

To overwrite a submatrix, follow these steps:

1 Set the **Overwrite** parameter to `Submatrix`.

2 Specify the overwriting values as described in "Specifying the Overwriting Values" on page 2-925.

3 Specify which rows and columns of the input matrix are contained in the submatrix that you want to overwrite by setting the **Row span** parameter to one of the following options and the **Column span** to the analogous column-related options:

- `All rows` — The submatrix contains all rows of the input matrix.
- `One row` — The submatrix contains only one row of the input matrix, which you must specify in the **Row** parameter, as described in the following table.

- `Range of rows` — The submatrix contains one or more rows of the input, which you must specify in the **Starting Row** and **Ending row** parameters, as described in the following tables.

4. When you set **Row span** to `One row` or `Range of rows`, you need to further specify the row(s) contained in the submatrix by setting the **Row** or **Starting row** and **Ending row** parameters. Likewise, when you set **Column span** to `One column` or `Range of columns`, you must further specify the column(s) contained in the submatrix by setting the **Column** or **Starting column** and **Ending column** parameters. For descriptions of the settings for these parameters, see the following tables.

Settings for Row, Column, Starting Row, and Starting Column Parameters

Settings for Specifying the Submatrix's First Row or Column	First Row of Submatrix (Only row for Row span = One row)	First Column of Submatrix (Only row for Row span = One row)
`First`	First row of the input	First column of the input
`Index`	Input row specified in the **Row index** parameter	Input column specified in the **Column index** parameter
`Offset from last`	Input row with the index `M - rowOffset` where `M` is the number of input rows, and `rowOffset` is the value of the **Row offset** or **Starting row offset** parameter	Input column with the index `N - colOffset` where `N` is the number of input columns, and `colOffset` is the value of the **Column offset** or **Starting column offset** parameter
`Last`	Last row of the input	Last column of the input

Settings for Row, Column, Starting Row, and Starting Column Parameters (Continued)

Settings for Specifying the Submatrix's First Row or Column	First Row of Submatrix (Only row for Row span = One row)	First Column of Submatrix (Only row for Row span = One row)
`Offset from middle`	Input row with the index `floor(M/2 + 1 - rowOffset)` where `M` is the number of input rows, and `rowOffset` is the value of the **Row offset** or **Starting row offset** parameter	Input column with the index `floor(N/2 + 1 - rowOffset)` where `N` is the number of input columns, and `colOffset` is the value of the or **Column offset** or **Starting column offset** parameter
`Middle`	Input row with the index `floor(M/2 + 1)` where `M` is the number of input rows	Input columns with the index `floor(N/2 + 1)` where `N` is the number of input columns

Settings for Ending Row and Ending Column Parameters

Settings for Specifying the Submatrix's Last Row or Column	Last Row of Submatrix	Last Column of Submatrix
`Index`	Input row specified in the **Ending row index** parameter	Input column specified in the **Ending column index** parameter

Settings for Ending Row and Ending Column Parameters (Continued)

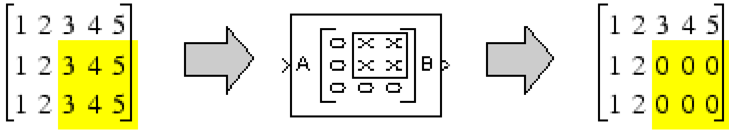
Settings for Specifying the Submatrix's Last Row or Column	Last Row of Submatrix	Last Column of Submatrix
`Offset from last`	Input row with the index `M - rowOffset` where `M` is the number of input rows, and `rowOffset` is the value of the **Ending row offset** parameter	Input column with the index `N - colOffset` where `N` is the number of input columns, and `colOffset` is the value of the **Ending column offset** parameter
`Last`	Last row of the input	Last column of the input
`Offset from middle`	Input row with the index `floor(M/2 + 1 - rowOffset)` where `M` is the number of input rows, and `rowOffset` is the value of the **Ending row offset** parameter	Input column with the index `floor(N/2 + 1 - rowOffset)` where `N` is the number of input columns, and `colOffset` is the value of the **Ending column offset** parameter
`Middle`	Input row with the index `floor(M/2 + 1)` where `M` is the number of input rows	Input columns with the index `floor(N/2 + 1)` where `N` is the number of input columns

For example, to overwrite the lower-right 2-by-3 submatrix of a 3-by-5 input matrix with all zeros, enter the following set of parameters:

- **Overwrite** = `Submatrix`
- **Source of overwriting value(s)** = `Specify via dialog`
- **Overwrite with** = `0`
- **Row span** = `Range of rows`
- **Starting row** = `Index`

- **Starting row index** = `2`
- **Ending row** = `Last`
- **Column span** = `Range of columns`
- **Starting column** = `Offset from last`
- **Starting column offset** = `2`
- **Ending column** = `Last`

The following figure shows the block with the above settings overwriting a portion of a 3-by-5 input matrix.



There are often several possible parameter combinations that select the *same* submatrix from the input. For example, instead of specifying `Last` for **Ending column**, you could select the same submatrix by specifying

- **Ending column** = `Index`
- **Ending column index** = `5`

Overwriting a Subdiagonal

To overwrite a subdiagonal, follow these steps:

1. Set the **Overwrite** parameter to `Diagonal`.

2. Specify the overwriting values as described in "Specifying the Overwriting Values" on page 2-925.

3. Specify the subdiagonal that you want to overwrite by setting the **Diagonal span** parameter to one of the following options:

   - `All elements` — Overwrite the entire input diagonal.
   - `One element` — Overwrite one element in the diagonal, which you must specify in the **Element** parameter (described below).
   - `Range of elements` — Overwrite a portion of the input diagonal, which you must specify in the **Starting element** and **Ending element** parameters, as described in the following table.

4. When you set **Diagonal span** to `One element` or `Range of elements`, you need to further specify which diagonal element(s) to overwrite by setting the **Element** or **Starting element** and **Ending element** parameters. See the following tables.

Element and Starting Element Parameters

Settings for Element and Starting Element Parameters	First Element in Subdiagonal (Only element when Diagonal span = One element)
`First`	Diagonal element in first row of the input
`Index`	*k*th diagonal element, where *k* is the value of the **Element index** or **Starting element index** parameter

Element and Starting Element Parameters (Continued)

Settings for Element and Starting Element Parameters	First Element in Subdiagonal (Only element when Diagonal span = One element)
`Offset from last`	Diagonal element in the row with the index `M - offset` where `M` is the number of input rows, and `offset` is the value of the **Element offset** or **Starting element offset** parameter
`Last`	Diagonal element in the last row of the input
`Offset from middle`	Diagonal element in the input row with the index `floor(M/2 + 1 - offset)` where `M` is the number of input rows, and `offset` is the value of the **Element offset** or **Starting element offset** parameter
`Middle`	Diagonal element in the input row with the index `floor(M/2 + 1)` where `M` is the number of input rows

Ending Element Parameters

Settings for Ending Element Parameter	Last Element in Subdiagonal
`Index`	kth diagonal element, where k is the value of the **Ending element index** parameter
`Offset from last`	Diagonal element in the row with the index `M - offset` where `M` is the number of input rows, and `offset` is the value of the **Ending element offset** parameter
`Last`	Diagonal element in the last row of the input

Ending Element Parameters (Continued)

Settings for Ending Element Parameter	Last Element in Subdiagonal
`Offset from middle`	Diagonal element in the input row with the index `floor(M/2 + 1 - offset)` where `M` is the number of input rows, and `offset` is the value of the **Ending element offset** parameter
`Middle`	Diagonal element in the input row with the index `floor(M/2 + 1)` where `M` is the number of input rows

Supported Data Types

The input(s) and output of this block must have the same data type.

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

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

See Also

Reshape	Simulink
Selector	Simulink
Submatrix	Signal Processing Blockset
Variable Selector	Signal Processing Blockset
`reshape`	MATLAB

Purpose Pad or truncate specified dimension(s)

Library Signal Operations

`dspsigops`

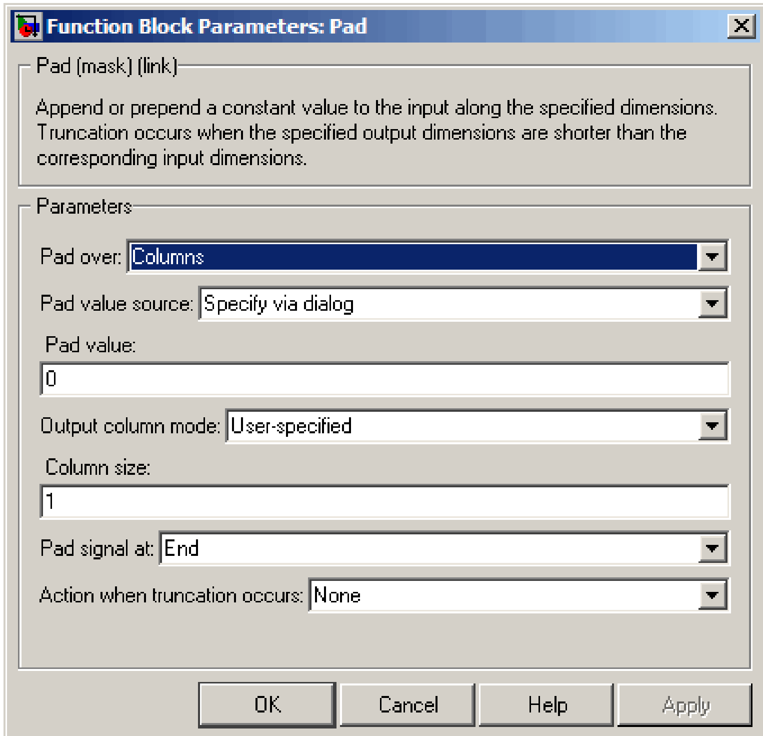
Description

The Pad block extends or crops the dimensions of the input by padding or truncating along its columns, rows, columns and rows, or any dimension(s) you specify. Truncation occurs when you specify output dimensions that are shorter than the corresponding input dimensions. If the input and output lengths are the same, the block is a pass-through.

You can enter the pad value in the block mask or via an input port. You can enter output sizes in the block mask, or have the block pad the specified dimensions until their length is the next highest power of two. The **Pad signal at** parameter controls whether the specified input dimensions are padded or truncated at their beginning, end, or both. For odd pad or truncation lengths, the extra pad value or truncation is applied to the end of the signal. When the block is in `Specified dimensions` mode, you can specify either the output size or the pad size.

You can have the block warn or error when an input signal is truncated using the **Action when truncation occurs** parameter.

Dialog Box



Pad over

Specify the dimensions over which to pad or truncate: `Columns`, `Rows`, `Columns and rows`, `None`, or `Specified dimensions`.

Dimensions to pad

Specify the one-based dimension(s) over which to pad or truncate. The value for this parameter can be a scalar or a vector. For example, specify `1` to pad columns. Specify `[1 2]` to pad columns and rows. Specify `[1 3 5]` to pad the first, third, and fifth dimensions.

This parameter is only visible when `Specified dimensions` is selected for the **Pad over** parameter.

Pad value source

Choose how you specify the pad value. The pad value can come from an input port or from the dialog:

- If you select `Input port`, the `PVal` port appears on the block icon.
- If you select `Specify via dialog`, the **Pad value** parameter appears.

Pad value

Specify the constant scalar value with which to pad the input. Tunable.

This parameter is only visible when `Specify via dialog` is selected for the **Pad value source** parameter.

Output column mode

Choose how you specify the column length of the output:

- If you select `User-specified`, the **Column size** parameter appears.
- If you select `Next power of two`, the block pads the output columns until their length is the next highest power of two. If the column length is already a power of two, the columns are not padded.

This parameter is only visible when `Columns` or `Columns and rows` is selected for the **Pad over** parameter.

Column size

Specify the column length of the output. If the specified column length is longer than the input column length, the columns are padded. If the specified column length is shorter than the input column length, the columns are truncated. This parameter is only visible when `User-specified` is selected for the **Output column mode** parameter.

Output row mode

Choose how you specify the output row length of the output:

- If you select `User-specified`, the **Row size** parameter appears.
- If you select `Next power of two`, the block pads the output rows until their length is the next highest power of two. If the row length is already a power of two, the rows are not padded.

This parameter is only visible when `Rows` or `Columns and rows` is selected for the **Pad over** parameter.

Row size

Specify the row length of the output. If the specified row length is longer than the input row length, the rows are padded. If the specified row length is shorter than the input row length, the rows are truncated. This parameter is only visible when `User-specified` is selected for the **Output row mode** parameter.

Specify

Choose whether you want to control the output length of the specified dimensions by specifying the pad size or the output size.

This parameter is only visible when `Specified dimensions` is selected for the **Pad over** parameter.

Pad size at beginning

Specify how many values to add to the beginning of the input signal along the specified dimension(s). This parameter must be a scalar or a vector with the same number of elements as the **Dimensions to pad** parameter. Each element in the **Pad size at beginning** parameter gives the pad length for the beginning of the corresponding dimension in the **Dimensions to pad** parameter. Values of this parameter must be zero or a positive integer.

This parameter is only visible if `Pad size` is selected for the **Specify** parameter.

Pad size at end

Specify how many values to add to the end of the input signal along the specified dimension(s). This parameter must be a scalar or a vector with the same number of elements as the **Dimensions to pad** parameter. Each element in the **Pad size at end** parameter gives the pad length for the end of the corresponding dimension in the **Dimensions to pad** parameter. Values of this parameter must be zero or a positive integer.

This parameter is only visible if `Pad size` is selected for the **Specify** parameter.

Output size mode

Choose how you specify the output length of the specified dimensions:

- If you select `User-specified`, the **Output size** parameter appears.
- If you select `Next power of two`, the block pads the specified dimensions until their length is the next highest power of two. If the dimension length is already a power of two, no padding occurs in that dimension.

This parameter is only visible if `Output size` is selected for the **Specify** parameter.

Output size

Specify the output length of the specified dimension(s). This parameter must be a scalar or a vector with the same number of elements as the **Dimensions to pad** parameter. Each element in the **Output size** vector gives the output length for the corresponding dimension in the **Dimensions to pad** vector. If the specified length is longer than the input length for a given dimensions, that dimension is padded. If the specified length is shorter than the input length for a given dimension, that dimension is truncated.

This parameter is only visible if `Output size` is selected for the **Specify** parameter.

Pad signal at

Specify whether to pad or truncate the signal at the `Beginning`, `End`, or `Beginning and end` of the specified dimension(s). When you select `Beginning and end`, half the pad length is added to the beginning of the signal, and half is added to the end of the signal. For an odd pad length, the extra value is added to the end of the signal. This also applies to truncation. In this mode, an equal number of values are truncated from the beginning and the end of the signal. In the case of an odd truncation length, the extra value is removed from the end of the signal.

Action when truncation occurs

Choose `None` when you do not want to be notified that the input is truncated. Select `Warning` to display a warning when the input is truncated. Choose `Error` when to display an error and terminate the simulation when the input is truncated.

Supported Data Types

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

See Also

Concatenate	Simulink
Repeat	Signal Processing Blockset
Submatrix	Signal Processing Blockset
Upsample	Signal Processing Blockset
Variable Selector	Signal Processing Blockset

Parametric Equalizer

Purpose

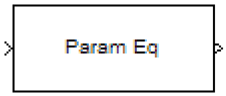
Design parametric equalizer

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description



This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have Filter Design Toolbox product.

Dialog Box

See “Parametric Equalizer Filter Design Dialog Box — Main Pane” in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Determine whether each value of input signal is local minimum or maximum

Library

Signal Operations

`dspsigops`

Description

The Peak Finder block outputs the number of local extrema in the input signal at the Cnt port. Optionally, it can also output the extrema indices, the extrema values, and a binary indicator of whether or not the extrema are maxima or minima.

The Peak Finder block compares the current signal value to the previous and next values to determine if the current value is an extremum. Use the **Peak type(s)** parameter to specify whether you are looking for maxima, minima, or both.

If you select the **Output peak indices** check box, the Idx port appears on the block. The block outputs the zero-based extrema indices at the Idx port. If you select the **Output peak values** check box, the Val port appears on the block. The block outputs the extrema values at the Val port. If you select either of these check boxes and `Maxima and Minima` is selected for the **Peak type(s)**, the Pol port also appears on the block. If the signal value is a maximum, the block outputs a 1 at the Pol ("Polarity") port. If the signal value is a minimum, the block outputs a 0 at the Pol port.

Note that nothing is output at the Idx, Val, and Pol ports for an input signal value that is not an extremum.

Use the **Maximum number of peaks to find** parameter to specify how many extrema to look for in each input signal. The block stops searching the input signal once this maximum number of extrema has been found.

If you select the **Ignore peaks within threshold of neighboring values** check box, the block no longer detects low-amplitude peaks. This feature allows the block to ignore noise within a threshold value that you define. Enter a threshold value for the **Threshold** parameter. Now, the current value is a maximum if (current – previous) > threshold and (current – next) > threshold. The current value is a minimum if (current – previous) < –threshold and (current – next) < –threshold.

This block supports single-channel, multichannel, sample-based, and frame-based inputs. These input signals must be real-valued fixed-point or floating-point scalars or vectors.

Examples

Example 1

Consider the input vector

```
[9 6 10 3 4 5 0 12]
```

The table below shows the analysis made by the Peak Finder block. Note that the first and last input signal values are not considered:

Previous, current, and next values	9 6 10	6 10 3	10 3 4	3 4 5	4 5 0	5 0 12
Current value if it is an extremum	6	10	3	—	5	0
Index of current value if it is an extremum	1	2	3	—	5	6
Polarity of current value if it is an extremum	0	1	0	—	1	0

Therefore, for this example the outputs at the block ports are

Cnt: `5`

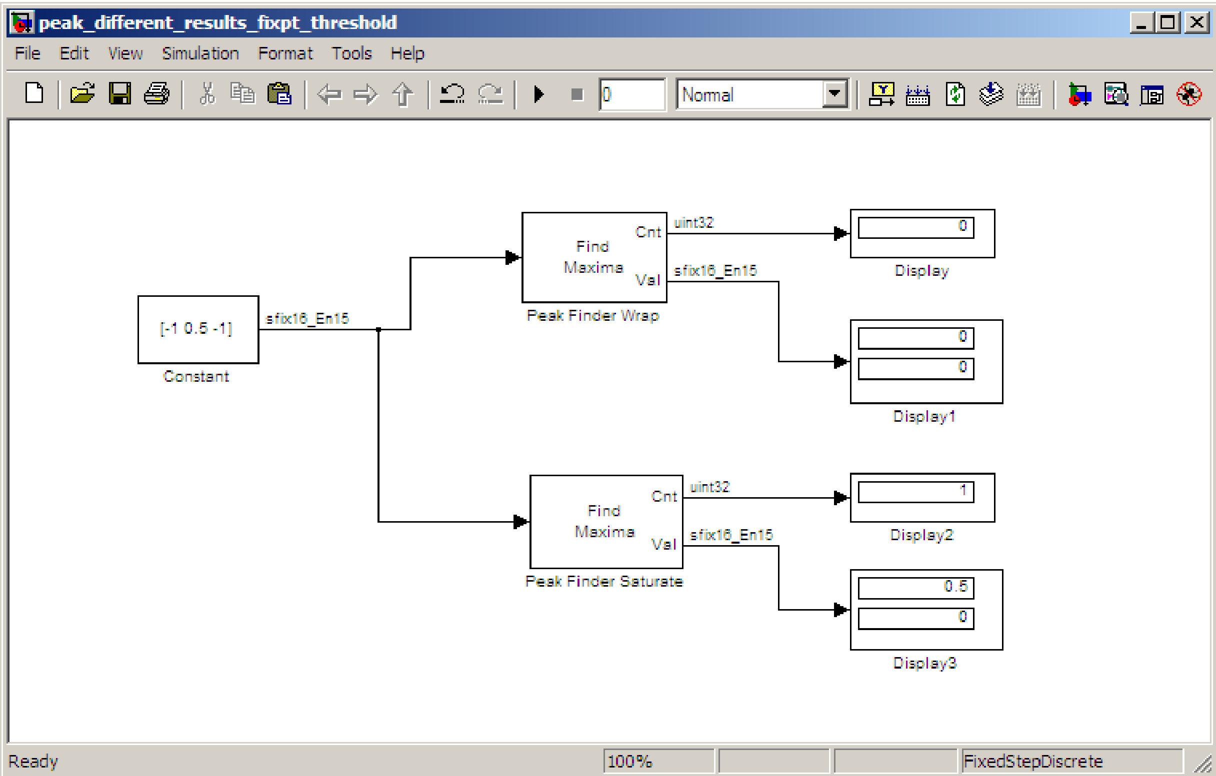
Idx: `[1 2 3 5 6]`

Val: `[6 10 3 5 0]`

Pol: `[0 1 0 1 0]`

Example 2

Note that the **Overflow mode** parameter can affect the output of the block when the input is fixed point. Consider the following model:



In this model, the settings in the Constant block are

- **Constant value** — `[-1 0.5 -1]`
- **Interpret vector parameters as 1-D** — not selected
- **Sampling mode** — `Sample based`
- **Sample time** — `1`

- **Output data type** — `<data type expression>`
- **Mode** — `Fixed point`
- **Sign** — `Signed`
- **Scaling** — `Binary point`
- **Word length** — `16`
- **Fraction length** — `15`

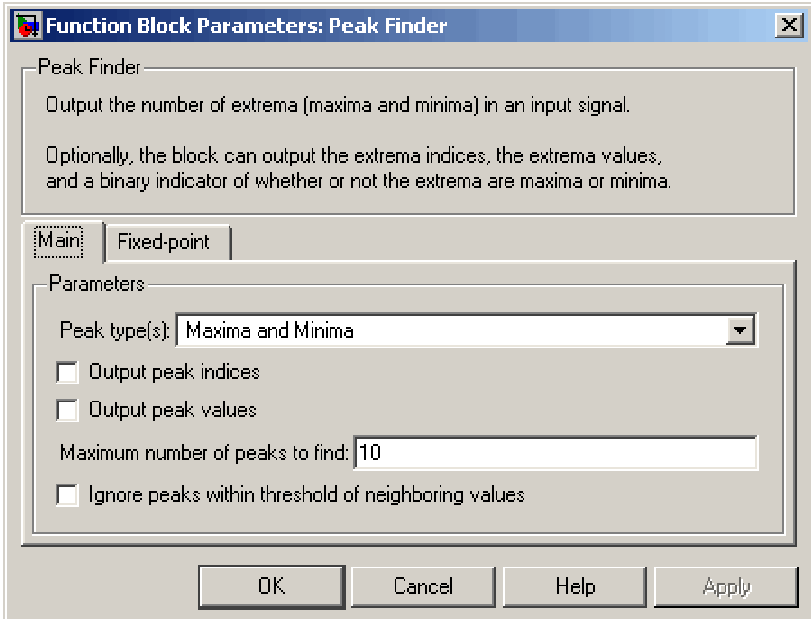
The settings in the Peak Finder blocks are

- **Peak type(s)** — `Maxima`
- **Output peak indices** — not selected
- **Output peak values** — selected
- **Maximum number of peaks to find** — `2`
- **Ignore peaks within threshold of neighboring values** — selected
- **Threshold** — `0.25`
- **Rounding mode** — `Floor`
- **Overflow mode** — `Wrap` for Peak Finder Wrap, `Saturate` for Peak Finder Saturate

Setting the **Overflow mode** parameter of the Peak Finder Wrap block to `Wrap` causes the calculations (current – previous) > threshold and (current – next) > threshold to wrap on overflow, thereby causing the maximum to be missed.

Dialog Box

The **Main** pane of the Peak Finder block dialog appears as follows.



Peak type(s)
: Specify whether you are looking for maxima, minima, or both.

Output peak indices
: Select this check box if you want the block to output the extrema indices at the Idx port.

Output peak values
: Select this check box if you want the block to output the extrema values at the Val port.

Maximum number of peaks to find
: Enter the number of extrema to look for in each input signal. The block stops searching the input signal for extrema once the maximum number of extrema has been found. The value of this parameter must be an integer greater than or equal to one.

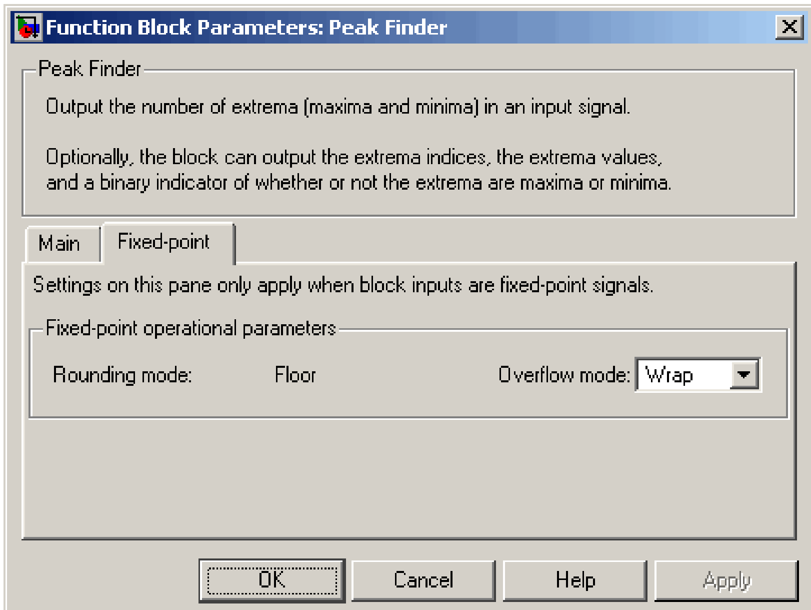
Ignore peaks within threshold of neighboring values

Select this check box if you want to eliminate the detection of peaks whose amplitudes are within a specified threshold of neighboring values.

Threshold

Enter your threshold value. This parameter appears if you select the **Ignore peaks within threshold of neighboring values** check box.

The **Fixed-point** pane of the Peak Finder block dialog appears as follows.



Rounding mode

The rounding mode of this block is always `Floor`.

Overflow mode

Select the overflow mode to be used when block inputs are fixed point.

Supported Data Types

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

See Also

Maximum	Signal Processing Blockset
Minimum	Signal Processing Blockset

Purpose

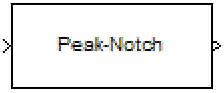
Design peak or notch filter

Library

Filtering / Filter Design Toolbox

`dspfdesign`

Description



This block brings the functionality of the Filter Design Toolbox™ `filterbuilder` function to the Simulink® environment. You must have a Filter Design Toolbox product license to design filters with this block. However, you can run models containing this block without a license. This allows you to run a model sent to you by a colleague who has designed a filter using this block, even if you do not have the Filter Design Toolbox product.

Dialog Box

See "Notch/Peak Filter Design Dialog Box — Main Pane" in the Filter Design Toolbox documentation for more information about the parameters of this block. The **Data Types** pane is not available for Filter Design Toolbox blocks in the Signal Processing Blockset™ product.

Parameters of this block that do not change filter order or structure are tunable. Tunable parameters are enabled during simulation; nontunable parameters are not.

Supported Data Types

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

Purpose

Compute nonparametric estimate of spectrum

Library

Estimation / Power Spectrum Estimation

`dspspect3`

Description

The Periodogram block computes a nonparametric estimate of the spectrum. The block averages the squared magnitude of the FFT computed over windowed sections of the input and normalizes the spectral average by the square of the sum of the window samples.

Both an M-by-N frame-based matrix input and an M-by-N sample-based matrix input are treated as M sequential time samples from N independent channels. The block computes a separate estimate for each of the N independent channels and generates an N_{fft}-by-N matrix output. When you select the **Inherit FFT length from input dimensions** check box, N_{fft} is specified by the frame size of the input, which must be a power of 2. When you clear the **Inherit FFT length from input dimensions** check box, N_{fft} is specified as a power of 2 by the **FFT length** parameter, and the block zero pads or wraps the input to N_{fft} before computing the FFT.

Each column of the output matrix contains the estimate of the corresponding input column's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where F_s is the signal's sample frequency. The output is always sample based.

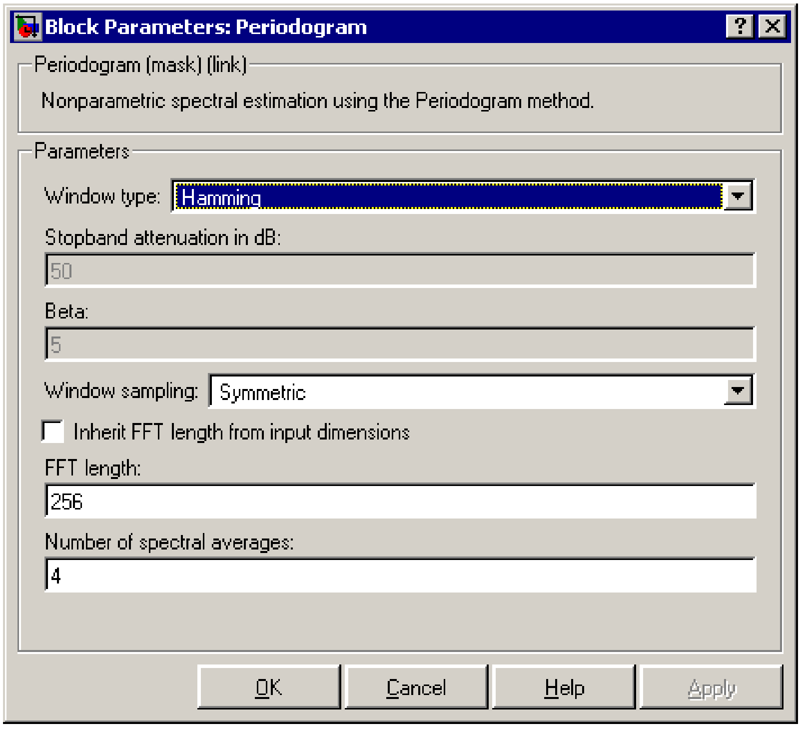
The **Number of spectral averages** specifies the number of spectra to average. Setting this parameter to `1` effectively disables averaging.

The **Window type**, **Stopband ripple**, **Beta**, and **Window sampling** parameters all apply to the specification of the window function; see the Window Function block reference page for more details on these four parameters.

Example

The `dspstfft` demo provides an illustration of using the Periodogram and Matrix Viewer blocks to create a spectrogram. The `dspsacomp` demo compares the Periodogram block with several other spectral estimation methods.

Dialog Box



Window type

Enter the type of window to apply. See the Window Function block reference page for more details. Tunable.

Stopband attenuation in dB

Enter the level, in dB, of stopband attenuation, R_s, for the Chebyshev window. This parameter is enabled if, for the **Window type** parameter, you choose `Chebyshev`. Tunable.

Beta

Enter the β parameter for the Kaiser window. This parameter is enabled if, for the **Window type** parameter, you chose `Kaiser`. Increasing **Beta** widens the mainlobe and decreases the amplitude of the window sidelobes in the window's frequency magnitude response. Tunable.

Window sampling
From the list, choose `Symmetric` or `Periodic`. Tunable.

Inherit FFT length from input dimensions
When you select this check box, the block uses the input frame size as the number of data points, N_{fft}, on which to perform the FFT.

FFT length
Enter the number of data points on which to perform the FFT, N_{fft}. When N_{fft} is larger than the input frame size, each frame is zero-padded as needed. When N_{fft} is smaller than the input frame size, each frame is wrapped as needed. This parameter is enabled when you clear the **Inherit FFT length from input dimensions** check box.

Number of spectral averages
Enter the number of spectra to average; setting this parameter to `1` disables averaging.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Orfanidis, S. J. *Introduction to Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1995.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
Output	• Double-precision floating point • Single-precision floating point

See Also

Burg Method	Signal Processing Blockset
Inverse Short-Time FFT	Signal Processing Blockset
Magnitude FFT	Signal Processing Blockset
Short-Time FFT	Signal Processing Blockset
Spectrum Scope	Signal Processing Blockset
Window Function	Signal Processing Blockset
Yule-Walker Method	Signal Processing Blockset
`pwelch`	Signal Processing Toolbox

See “Power Spectrum Estimation” for related information.

Purpose

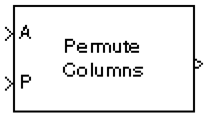
Reorder matrix rows or columns

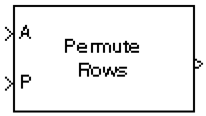
Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description





The Permute Matrix block reorders the rows or columns of M-by-N input matrix `A` as specified by indexing input `P`.

When the **Permute** parameter is set to `Rows`, the block uses the rows of `A` to create a new matrix with the same column dimension. Input `P` is a length-L vector whose elements determine where each row from `A` should be placed in the L-by-N output matrix.

```
% Equivalent MATLAB code
y = [A(P(1),:) ; A(P(2),:) ; A(P(3),:) ; ... ; A(P(end),:)]
```

For row permutation, a length-M 1-D vector input at the `A` port is treated as a M-by-1 matrix.

When the **Permute** parameter is set to `Columns`, the block uses the columns of `A` to create a new matrix with the same row dimension. Input `P` is a length-L vector whose elements determine where each column from `A` should be placed in the M-by-L output matrix.

```
% Equivalent MATLAB code
y = [A(:,P(1)) A(:,P(2)) A(:,P(3)) ... A(:,P(end))]
```

For column permutation, a length-N 1-D vector input at the `A` port is treated as a 1-by-N matrix.

When an index value in input `P` references a nonexistent row or column of matrix `A`, the block reacts with the behavior specified by the **Invalid permutation index** parameter. The following options are available:

- `Clip index` — Clip the index to the nearest valid value (1 or M for row permutation, and 1 or N for column permutation), and *do not* issue an alert. Example: For a 3-by-7 input matrix, a column index of 9 is clipped to 7, and a row index of -2 is clipped to 1.

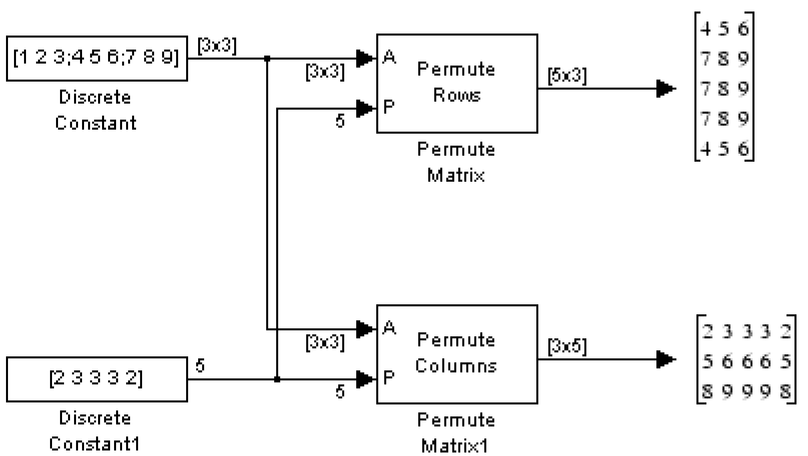
- `Clip and warn` — Display a warning message in the MATLAB® command window, and clip the index as described above.
- `Generate error` — Display an error dialog box and terminate the simulation.

When length of the permutation vector `P` is not equal to the number of rows or columns of the input matrix `A`, you can choose to get an error dialog box and terminate the simulation by selecting **Error when length of P is not equal to Permute dimension size**.

When input `A` is frame based, the output is frame based; otherwise, the output is sample based.

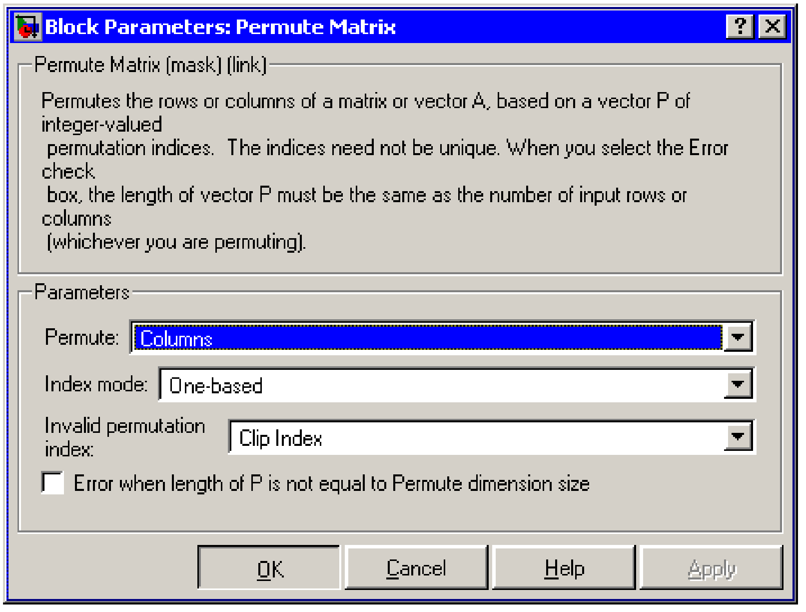
Examples

In the model below, the top Permute Matrix block places the second row of the input matrix in both the first and fifth rows of the output matrix, and places the third row of the input matrix in the three middle rows of the output matrix. The bottom Permute Matrix block places the second column of the input matrix in both the first and fifth columns of the output matrix, and places the third column of the input matrix in the three middle columns of the output matrix.



As shown in the example above, rows and columns of `A` can appear any number of times in the output, or not at all.

Dialog Box



Permute

Method of constructing the output matrix; by permuting rows or columns of the input.

Index mode

When set to `One-based`, a value of `1` in the permutation vector `P` refers to the first row or column of the input matrix `A`. When set to `Zero-based`, a value of `0` in `P` refers to the first row or column of `A`.

Invalid permutation index

Response to an invalid index value. Tunable.

Error when length of P is not equal to Permute dimension size
Option to display an error dialog box and terminate the simulation when the length of the permutation vector `P` is not equal to the number of rows or columns of the input matrix `A`.

Supported Data Types

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

See Also

Submatrix	Signal Processing Blockset
Transpose	Signal Processing Blockset

Variable Selector	Signal Processing Blockset
`permute`	MATLAB

See "Reordering Channels in Multichannel Frame-Based Signals" for related information.

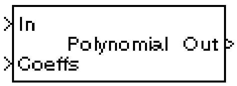
Polynomial Evaluation

Purpose Evaluate polynomial expression

Library Math Functions / Polynomial Functions

`dsppolyfun`

Description



The Polynomial Evaluation block applies a polynomial function to the real or complex input at the `In` port.

```
y = polyval(u)      % Equivalent MATLAB code
```

The Polynomial Evaluation block performs these types of operation more efficiently than the equivalent construction using Simulink® Sum and Math Function blocks.

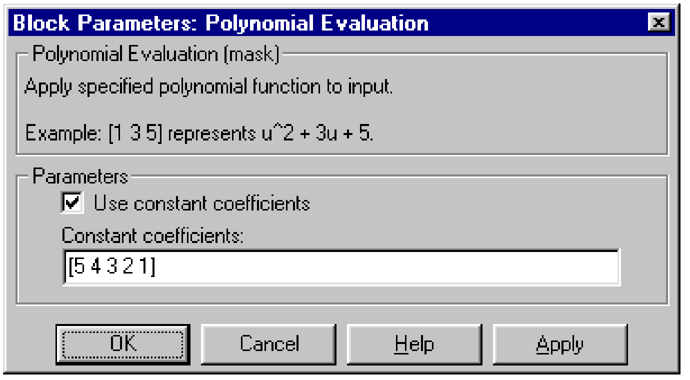
When you select the **Use constant coefficients** check box, you specify the polynomial expression in the **Constant coefficients** parameter. When you do not select **Use constant coefficients**, a variable polynomial expression is specified by the input to the `Coeffs` port. In both cases, the polynomial is specified as a vector of real or complex coefficients in order of descending exponents.

The table below shows some examples of the block's operation for various coefficient vectors.

Coefficient Vector	Equivalent Polynomial Expression
`[1 2 3 4 5]`	$y = u^4 + 2u^3 + 3u^2 + 4u + 5$
`[1 0 3 0 5]`	$y = u^4 + 3u^2 + 5$
`[1 2+i 3 4-3i 5i]`	$y = u^4 + (2+i)u^3 + 3u^2 + (4-3i)u + 5i$

Each element of a vector or matrix input to the `In` port is processed independently, and the output size and frame status are the same as the input.

Dialog Box



Use constant coefficients

Select to enable the **Constant coefficients** parameter and disable the `Coeffs` input port.

Constant coefficients

Specify the vector of polynomial coefficients to apply to the input, in order of descending exponents. This parameter is enabled when you select the **Use constant coefficients** check box.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Least Squares Polynomial Fit	Signal Processing Blockset
Math Function	Simulink
Sum	Simulink
`polyval`	MATLAB

Polynomial Stability Test

Purpose

Use Schur-Cohn algorithm to determine whether all roots of input polynomial are inside unit circle

Library

Math Functions / Polynomial Functions

`dsppolyfun`

Description

The Polynomial Stability Test block uses the Schur-Cohn algorithm to determine whether all roots of a polynomial are within the unit circle.

```
y = all(abs(roots(u)) < 1)        % Equivalent MATLAB code
```

Each column of the M-by-N input matrix u contains M coefficients from a distinct polynomial,

$$f(x) = u_1 x^{M-1} + u_2 x^{M-2} + \ldots + u_M$$

arranged in order of descending exponents, u_1, u_2, ..., u_M. The polynomial has order M-1 and positive integer exponents.

Inputs can be frame based or sample based, and both represent the polynomial coefficients as shown above. For convenience, a length-M 1-D vector input is treated as an M-by-1 matrix.

The output is a 1-by-N matrix with each column containing the value `1` or `0`. The value `1` indicates that the polynomial in the corresponding column of the input is stable; that is, the magnitudes of all solutions to $f(x) = 0$ are less than 1. The value `0` indicates that the polynomial in the corresponding column of the input might be unstable; that is, the magnitude of at least one solution to $f(x) = 0$ is greater than or equal to 1.

The output is always sample based.

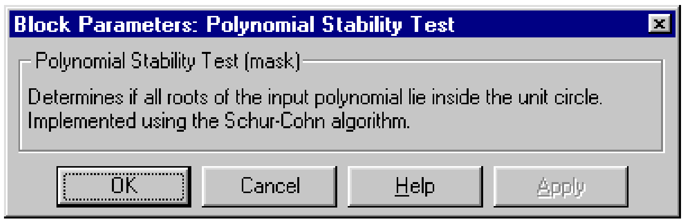
Applications

This block is most commonly used to check the pole locations of the denominator polynomial, $A(z)$, of a transfer function, $H(z)$.

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2 z^{-1} + \ldots + b_m z^{-(m-1)}}{a_1 + a_2 z^{-1} + \ldots + a_n z^{-(n-1)}}$$

The poles are the n-1 roots of the denominator polynomial, $A(z)$. When any poles are located outside the unit circle, the transfer function $H(z)$ is unstable. As is typical in DSP applications, the transfer function above is specified in descending powers of z^{-1} rather than z.

Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Boolean — Block outputs are always Boolean.

See Also

Least Squares Polynomial Fit	Signal Processing Blockset
Polynomial Evaluation	Signal Processing Blockset
`polyfit`	MATLAB

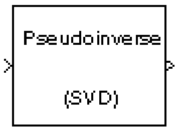
Pseudoinverse

Purpose Compute Moore-Penrose pseudoinverse of matrix

Library Math Functions / Matrices and Linear Algebra / Matrix Inverses

`dspinverses`

Description The Pseudoinverse block computes the Moore-Penrose pseudoinverse of input matrix A.




```
[U,S,V] = svd(A,0)       % Equivalent MATLAB code
```

The pseudoinverse of A is the matrix $A^{\dagger}$ such that

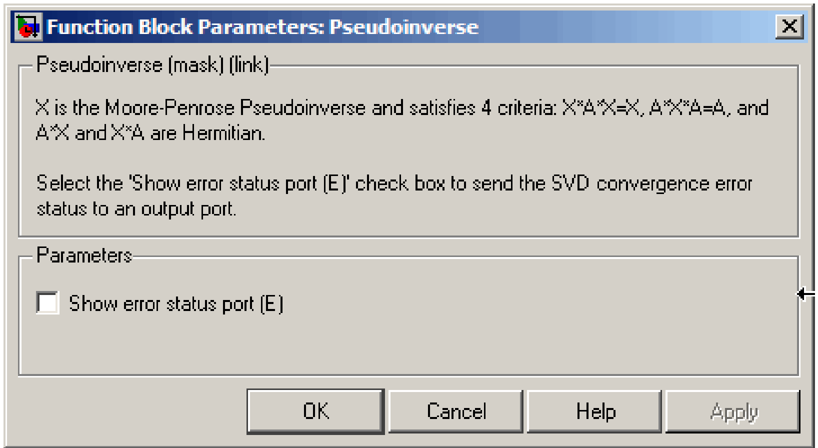
$$A^{\dagger} = VS^{\dagger}U^{*}$$

where U and V are orthogonal matrices, and S is a diagonal matrix. The pseudoinverse has the following properties:

- $AA^{\dagger} = (AA^{\dagger})^{*}$
- $A^{\dagger}A = (A^{\dagger}A)^{*}$
- $AA^{\dagger}A = A$
- $A^{\dagger}AA^{\dagger} = A^{\dagger}$

The output is always sample based.

Dialog Box



Show error status port

Select to enable the E output port, which reports a failure to converge. The possible values you can receive on the port are:

- 0 — The pseudoinverse calculation converges.
- 1 — The pseudoinverse calculation does not converge.

If the pseudoinverse calculation fails to converge, the output at port X is an undefined matrix of the correct size.

References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

Port	Supported Data Types
A	• Double-precision floating point • Single-precision floating point

Port	Supported Data Types
X	• Double-precision floating point • Single-precision floating point
E	• Boolean

See Also

Cholesky Inverse	Signal Processing Blockset
LDL Inverse	Signal Processing Blockset
LU Inverse	Signal Processing Blockset
Singular Value Decomposition	Signal Processing Blockset
`inv`	MATLAB

See “Matrix Inverses” for related information.

Purpose

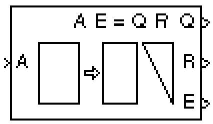
Factor rectangular matrix into unitary and upper triangular components

Library

Math Functions / Matrices and Linear Algebra / Matrix Factorizations

`dspfactors`

Description



The QR Factorization block uses a modified Gram-Schmidt iteration to factor a column permutation of the M-by-N input matrix A as

$$A_e = QR$$

where Q is an M-by-min(M,N) unitary matrix, and R is a min(M,N)-by-N upper-triangular matrix. A length-M vector input is treated as an M-by-1 matrix, and is always sample based.

The column-pivoted matrix A_e contains the columns of A permuted as indicated by the contents of length-N permutation vector E.

```
Ae = A(:,E)     % Equivalent MATLAB code
```

The block selects a column permutation vector E, which ensures that the diagonal elements of matrix R are arranged in order of decreasing magnitude.

$$|r_{i+1,j+1}| > |r_{i,j}| \qquad i = j$$

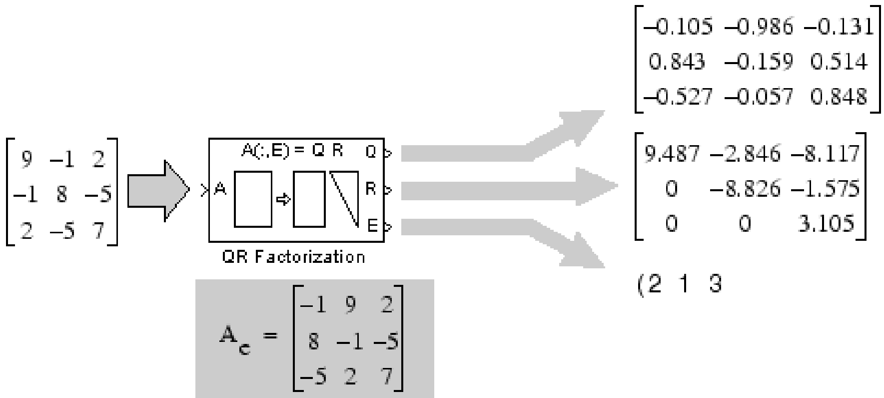
QR factorization is an important tool for solving linear systems of equations because of good error propagation properties and the invertability of unitary matrices.

$$Q^{-1} = Q^*$$

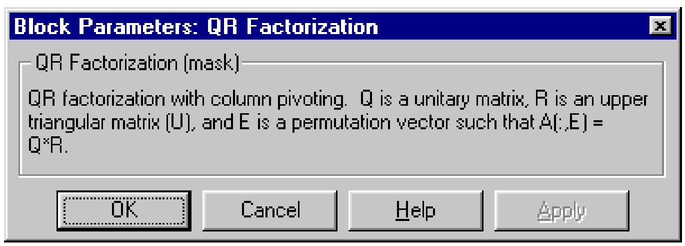
Unlike LU and Cholesky factorizations, the matrix A does not need to be square for QR factorization. Note, however, that QR factorization requires twice as many operations as Gaussian elimination.

Examples

A sample factorization is shown below. The input to the block is matrix A, which is permuted according to vector E to produce matrix A_e. Matrix A_e is factored to produce the Q and R output matrices.



Dialog Box



References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Cholesky Factorization	Signal Processing Blockset
LU Factorization	Signal Processing Blockset
QR Solver	Signal Processing Blockset
Singular Value Decomposition	Signal Processing Blockset
`qr`	MATLAB

See “Matrix Factorizations” for related information.

QR Solver

Purpose

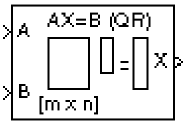
Find minimum-norm-residual solution to A*X*=B

Library

Math Functions / Matrices and Linear Algebra / Linear System Solvers

`dspsolvers`

Description



The QR Solver block solves the linear system A*X*=B, which can be overdetermined, underdetermined, or exactly determined. The system is solved by applying QR factorization to the M-by-N matrix, A, at the `A` port. The input to the `B` port is the right side M-by-L matrix, B. A length-M 1-D vector input at either port is treated as an M-by-1 matrix.

The output at the `x` port is the N-by-L matrix, X. X is always sample based, and is chosen to minimize the sum of the squares of the elements of B-AX. When B is a vector, this solution minimizes the vector 2-norm of the residual (B-AX is the residual). When B is a matrix, this solution minimizes the matrix Frobenius norm of the residual. In this case, the columns of X are the solutions to the L corresponding systems $AX_k=B_k$, where B_k is the kth column of B, and X_k is the kth column of *X*.

X is known as the minimum-norm-residual solution to AX=B. The minimum-norm-residual solution is unique for overdetermined and exactly determined linear systems, but it is not unique for underdetermined linear systems. Thus when the QR Solver is applied to an underdetermined system, the output *X* is chosen such that the number of nonzero entries in X is minimized.

Algorithm

QR factorization factors a column-permuted variant (A_e) of the M-by-N input matrix A as

$$A_e = QR$$

where Q is a M-by-min(M,N) unitary matrix, and R is a min(M,N)-by-N upper-triangular matrix.

The factored matrix is substituted for A_e in

$$A_e X = B_e$$

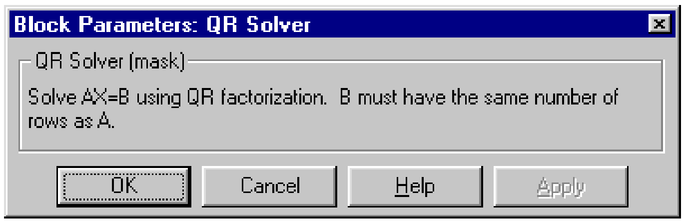
and

$$QRX = B_e$$

is solved for X by noting that $Q^{-1} = Q^{*}$ and substituting $Y = Q^{*}B_e$. This requires computing a matrix multiplication for Y and solving a triangular system for X.

$$RX = Y$$

Dialog Box



Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Levinson-Durbin	Signal Processing Blockset
LDL Solver	Signal Processing Blockset
LU Solver	Signal Processing Blockset
QR Factorization	Signal Processing Blockset
SVD Solver	Signal Processing Blockset

See “Linear System Solvers” for related information.

Quantizer

Purpose Discretize input at specified interval

Library Quantizers

`dspquant2`

Description Refer to the Simulink® Quantizer reference page for more information.

Purpose

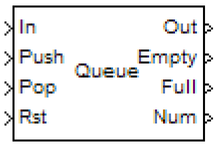
Store inputs in FIFO register

Library

Signal Management / Buffers

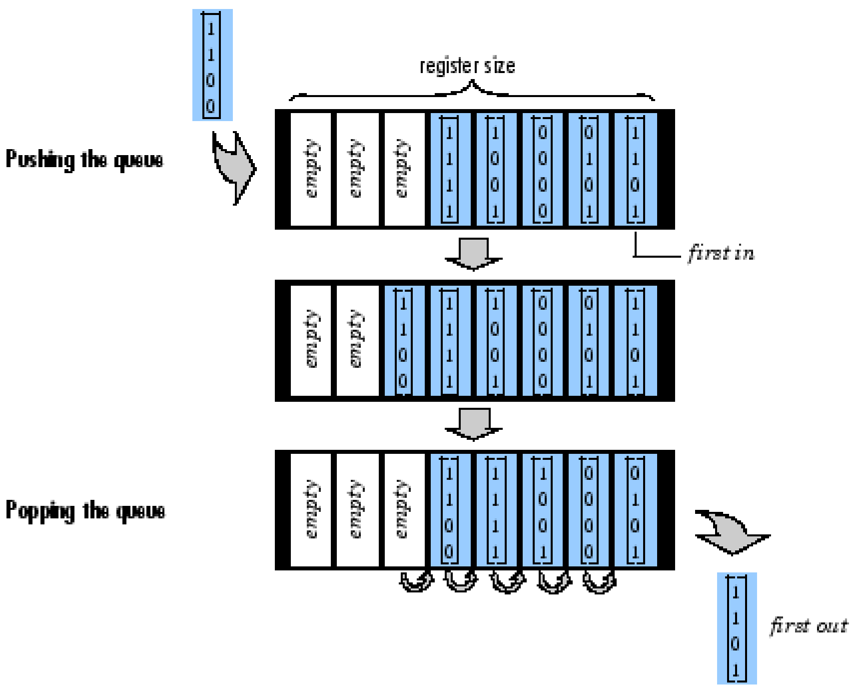
`dspbuff3`

Description



The Queue block stores a sequence of input samples in a first in, first out (FIFO) register. The register capacity is set by the **Register size** parameter, and inputs can be scalars, vectors, or matrices.

The block *pushes* the input at the `In` port onto the end of the queue when a trigger event is received at the `Push` port. When a trigger event is received at the `Pop` port, the block *pops* the first element off the queue and holds the `Out` port at that value. The first input to be pushed onto the queue is always the first to be popped off.



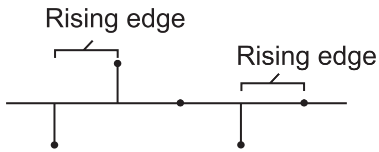
A trigger event at the optional Rst port empties the queue contents. When you select **Clear output port on reset**, then a trigger event at the Rst port empties the queue *and* sets the value at the Out port to zero. This setting also applies when a disabled subsystem containing the Queue block is reenabled; the Out port value is only reset to zero in this case when you select **Clear output port on reset**.

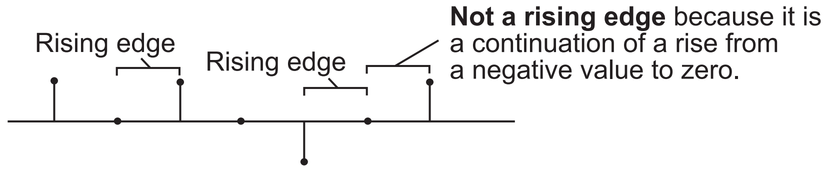
When two or more of the control input ports are triggered at the same time step, the operations are executed in the following order:

1. Rst
2. Push
3. Pop

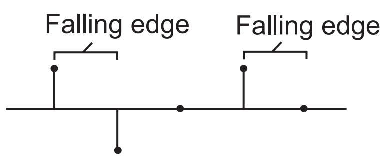
The rate of the trigger signal must be the same as the rate of the data signal input. You specify the triggering event for the Push, Pop, and Rst ports by the **Trigger type** pop-up menu:

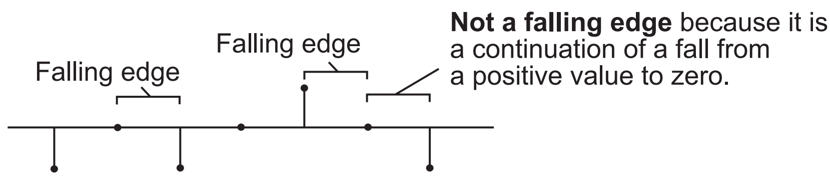
- Rising edge — Triggers execution of the block when the trigger input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero; see the following figure





- `Falling edge` — Triggers execution of the block when the trigger input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero; see the following figure





- `Either edge` — Triggers execution of the block when the trigger input is a `Rising edge` or `Falling edge` (as described above).
- `Non-zero sample` — Triggers execution of the block at each sample time that the trigger input is not zero.

Note When running simulations in the Simulink® `MultiTasking` mode, sample-based trigger signals have a one-sample latency, and frame-based trigger signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a trigger event, and when it applies the trigger. For more information on latency and the Simulink tasking modes, see “Excess Algorithmic Delay (Tasking Latency)” and “Models with Multiple Sample Rates” in the *Real-Time Workshop® User’s Guide*.

The **Push onto full register** parameter specifies the block’s behavior when a trigger is received at the `Push` port but the register is full. The **Pop empty register** parameter specifies the block’s behavior when a trigger is received at the `Pop` port but the register is empty. The following options are available for both cases:

- `Ignore` — Ignore the trigger event, and continue the simulation.
- `Warning` — Ignore the trigger event, but display a warning message in the MATLAB® Command Window.
- `Error` — Display an error dialog box and terminate the simulation.

Note The **Push onto full register** and **Pop empty register** parameters are diagnostic parameters. Like all diagnostic parameters on the Configuration Parameters dialog box, they are set to `Ignore` in the code generated for this block by Real-Time Workshop code generation software.

The **Push onto full register** parameter additionally offers the **Dynamic reallocation** option, which dynamically resizes the register to accept as many additional inputs as memory permits. To find out how many elements are on the queue at a given time, enable the `Num` output port by selecting the **Show number of register entries port** parameter.

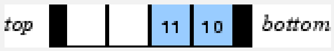
Note When `Dynamic reallocation` is selected, the **System target file** parameter on the **Real-Time Workshop** pane of the Configuration Parameters dialog box must be set to `grt_malloc.tlc   Generic Real-Time Target with dynamic memory allocation`.

Examples

Example 1

The table below illustrates the Queue block's operation for a **Register size** of `4`, **Trigger type** of `Either edge`, and **Clear output port on reset** enabled. Because the block triggers on both rising and falling edges in this example, each transition from `1` to `0` or `0` to `1` in the `Push`, `Pop`, and `Rst` columns below represents a distinct trigger event. A `1` in the `Empty` column indicates an empty queue, while a `1` in the `Full` column indicates a full queue.

In	Push	Pop	Rst	Queue	Out	Empty	Full	Num
1	0	0	0	*top* [ ][ ][ ][ ] *bottom*	0	1	0	0
2	1	0	0	*top* [ ][ ][ ][2] *bottom*	0	0	0	1
3	0	0	0	*top* [ ][ ][3][2] *bottom*	0	0	0	2
4	1	0	0	*top* [ ][4][3][2] *bottom*	0	0	0	3
5	0	0	0	*top* [5][4][3][2] *bottom*	0	0	1	4
6	0	1	0	*top* [ ][5][4][3] *bottom*	2	0	0	3
7	0	0	0	*top* [ ][ ][5][4] *bottom*	3	0	0	2
8	0	1	0	*top* [ ][ ][ ][5] *bottom*	4	0	0	1

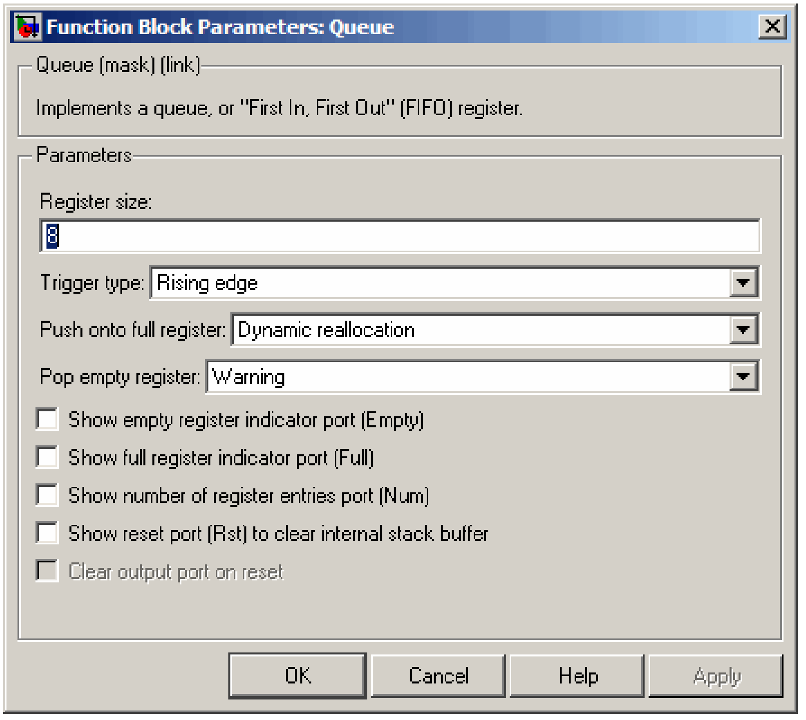
In	Push	Pop	Rst	Queue	Out	Empty	Full	Num
9	0	0	0		5	1	0	0
10	1	0	0		5	0	0	1
11	0	0	0		5	0	0	2
12	1	0	1		0	0	0	1

Note that at the last step shown, the `Push` and `Rst` ports are triggered simultaneously. The `Rst` trigger takes precedence, and the queue is first cleared and then pushed.

Example 2

The `dspqdemo` demo provides another example of the operation of the Queue block.

Dialog Box



Register size

The number of entries that the FIFO register can hold.

Trigger type

The type of event that triggers the block's execution. The rate of the trigger signal must be the same as the rate of the data signal input.

Push onto full register

Response to a trigger received at the `Push` port when the register is full. Inputs to this port must have the same built-in data type as inputs to the `Pop` and `Rst` input ports.

When `Dynamic reallocation` is selected, the **System target file** parameter on the **Real-Time Workshop** pane of the Configuration Parameters dialog box must be set to `grt_malloc.tlc   Generic Real-Time Target with dynamic memory allocation`.

Pop empty register

Response to a trigger received at the `Pop` port when the register is empty. Inputs to this port must have the same built-in data type as inputs to the `Push` and `Rst` input ports.

Show empty register indicator port

Enable the `Empty` output port, which is high (`1`) when the queue is empty, and low (`0`) otherwise.

Show full register indicator port

Enable the `Full` output port, which is high (`1`) when the queue is full, and low (`0`) otherwise. The `Full` port remains low when you select `Dynamic reallocation` from the **Push onto full register** parameter.

Show number of register entries port

Enable the `Num` output port, which tracks the number of entries currently on the queue. When inputs to the `In` port are double-precision values, the outputs from the `Num` port are double-precision values. Otherwise, the outputs from the `Num` port are 32-bit unsigned integer values.

Show reset port to clear internal stack buffer

Enable the `Rst` input port, which empties the queue when the trigger specified by the **Trigger type** is received. Inputs to this port must have the same built-in data type as inputs to the `Push` and `Pop` input ports.

Clear output port on reset

Reset the `Out` port to zero, in addition to clearing the queue, when a trigger is received at the `Rst` input port.

Supported Data Types

Port	Supported Data Types
In	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Push	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers Inputs to this port must have the same built-in data type as inputs to the Pop and Rst input ports
Pop	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers Inputs to this port must have the same built-in data type as inputs to the Push and Rst input ports.

Port	Supported Data Types
Rst	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers Inputs to this port must have the same built-in data type as inputs to the Push and Pop input ports.
Out	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Empty	• Double-precision floating point • Boolean The block outputs Boolean values at this port when Boolean support is enabled, as described in "Effects of Enabling and Disabling Boolean Support". To learn how to disable Boolean output support, see "Steps to Disabling Boolean Support"

Port	Supported Data Types
Full	• Double-precision floating point • Boolean The block outputs Boolean values at this port when Boolean support is enabled, as described in “Effects of Enabling and Disabling Boolean Support”. To learn how to disable Boolean output support, see “Steps to Disabling Boolean Support”
Num	• Double-precision floating point The block outputs a double-precision floating-point value at this port when the data type of the In port is double-precision floating-point. • 32-bit unsigned integers The block outputs a 32-bit unsigned integer value at this port when the data type of the In port is anything other than double-precision floating-point.

See Also

Buffer	Signal Processing Blockset
Delay Line	Signal Processing Blockset
Stack	Signal Processing Blockset

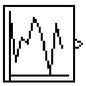
Random Source

Purpose Generate randomly distributed values

Library Signal Processing Sources

`dspsrcs4`

Description



The Random Source block generates a frame of M values drawn from a uniform or Gaussian pseudorandom distribution, where you specify M in the **Samples per frame** parameter.

This reference page contains a detailed discussion of the following Random Source block topics:

- “Distribution Type” on page 2-994
- “Output Complexity” on page 2-995
- “Output Repeatability” on page 2-997
- “Specifying the Initial Seed” on page 2-997
- “Sample Period” on page 2-998
- “Dialog Box” on page 2-999
- “Supported Data Types” on page 2-1002
- “See Also” on page 2-1003

Distribution Type

When the **Source type** parameter is set to `Uniform`, the output samples are drawn from a uniform distribution whose minimum and maximum values are specified by the **Minimum** and **Maximum** parameters, respectively. All values in this range are equally likely to be selected. A length-N vector specified for one or both of these parameters generates an N-channel output (M-by-N matrix) containing a unique random distribution in each channel.

For example, specify

- **Minimum** = `[ 0 0 -3 -3]`

- **Maximum** = `[10 10 20 20]`

to generate a four-channel output whose first and second columns contain random values in the range [0, 10], and whose third and fourth columns contain random values in the range [-3, 20]. When you specify only one of the **Minimum** and **Maximum** parameters as a vector, the block scalar expands the other parameter so it is the same length as the vector.

When the **Source type** parameter is set to `Gaussian`, you must also set the **Method** parameter, which determines the method by which the block computes the output, and has the following settings:

- `Ziggurat` — Produces Gaussian random values by using the Ziggurat method, which is the same method used by the MATLAB® `randn` function.
- `Sum of uniform values` — Produces Gaussian random values by adding and scaling uniformly distributed random signals based on the central limit theorem. This theorem states that the probability distribution of the sum of a sufficiently high number of random variables approaches the Gaussian distribution. You must set the **Number of uniform values to sum** parameter, which determines the number of uniformly distributed random numbers to sum to produce a single Gaussian random value.

For both settings of the **Method** parameter, the output samples are drawn from the normal distribution defined by the **Mean** and **Variance** parameters. A length-N vector specified for one or both of the **Mean** and **Variance** parameters generates an N-channel output (M-by-N frame matrix) containing a distinct random distribution in each column. When you specify only one of these parameters as a vector, the block scalar expands the other parameter so it is the same length as the vector.

Output Complexity

The block's output can be either real or complex, as determined by the `Real` and `Complex` options in the **Complexity** parameter. These settings control all channels of the output, so real and complex data cannot be combined in the same output. For complex output with a `Uniform` distribution, the real and imaginary components in each channel are both drawn from the same uniform random distribution, defined by the **Minimum** and **Maximum** parameters for that channel.

For complex output with a `Gaussian` distribution, the real and imaginary components in each channel are drawn from normal distributions with different means. In this case, the **Mean** parameter for each channel should specify a complex value; the real component of the **Mean** parameter specifies the mean of the real components in the channel, while the imaginary component specifies the mean of the imaginary components in the channel. When either the real or imaginary component is omitted from the **Mean** parameter, a default value of 0 is used for the mean of that component.

For example, a **Mean** parameter setting of `[5+2i 0.5 3i]` generates a three-channel output with the following means.

Channel 1 mean	*real* = 5	*imaginary* = 2
Channel 2 mean	*real* = 0.5	*imaginary* = 0
Channel 3 mean	*real* = 0	*imaginary* = 3

For complex output, the **Variance** parameter, σ^2, specifies the *total variance* for each output channel. This is the sum of the variances of the real and imaginary components in that channel.

$$\sigma^2 = \sigma_{\mathrm{Re}}^2 + \sigma_{\mathrm{Im}}^2$$

The specified variance is equally divided between the real and imaginary components, so that

$$\sigma_{\mathrm{Re}}^2 = \frac{\sigma^2}{2}$$

$$\sigma_{\mathrm{Im}}^2 = \frac{\sigma^2}{2}$$

Output Repeatability

The **Repeatability** parameter determines whether or not the block outputs the same signal each time you run the simulation. You can set the parameter to one of the following options:

- `Repeatable` — Outputs the same signal each time you run the simulation. The first time you run the simulation, the block randomly selects an initial seed. The block reuses these same initial seeds every time you rerun the simulation.
- `Specify seed` — Outputs the same signal each time you run the simulation. Every time you run the simulation, the block uses the initial seed(s) specified in the **Initial seed** parameter. Also see "Specifying the Initial Seed" on page 2-997.
- `Not repeatable` — Does not output the same signal each time you run the simulation. Every time you run the simulation, the block randomly selects an initial seed.

Specifying the Initial Seed

When you set the **Repeatability** parameter to `Specify seed`, you must set the **Initial seed** parameter. The **Initial seed** parameter specifies the initial seed for the pseudorandom number generator. The generator produces an identical sequence of pseudorandom numbers each time it is executed with a particular initial seed.

Specifying Initial Seeds for Real Outputs

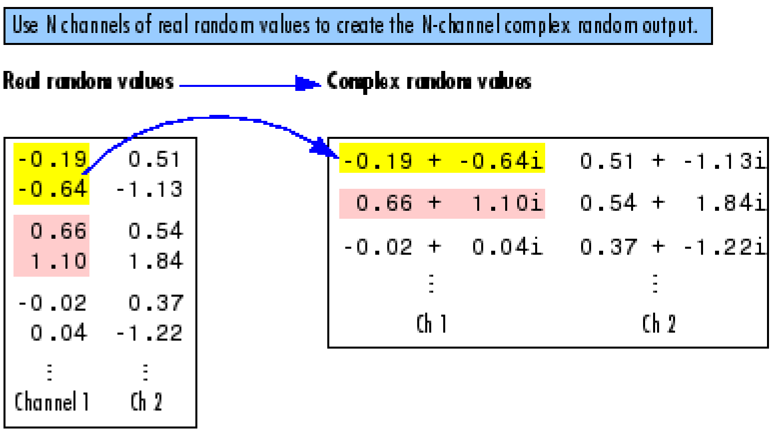
To specify the N initial seeds for an N-channel real-valued output, **Complexity** parameter set to `Real`, provide one of the following in the **Initial seed** parameter:

- Length-N vector of initial seeds — Uses each vector element as an initial seed for the corresponding channel in the N-channel output.
- Single scalar — Uses the scalar to generate N random values, which it uses as the seeds for the N-channel output.

Specifying Initial Seeds for Complex Outputs

To specify the initial seeds for an N-channel complex-valued output, **Complexity** parameter set to `Complex`, provide one of the following in the **Initial seed** parameter:

- Length-N vector of initial seeds — Uses each vector element as an initial seed for generating N channels of *real* random values. The block uses pairs of adjacent values in each of these channels as the real and imaginary components of the final output, as illustrated in the following figure.
- Single scalar — Uses the scalar to generate N random values, which it uses as the seeds for generating N channels of *real* random values. The block uses pairs of adjacent values in each of these channels as the real and imaginary components of the final output, as illustrated in the following figure.



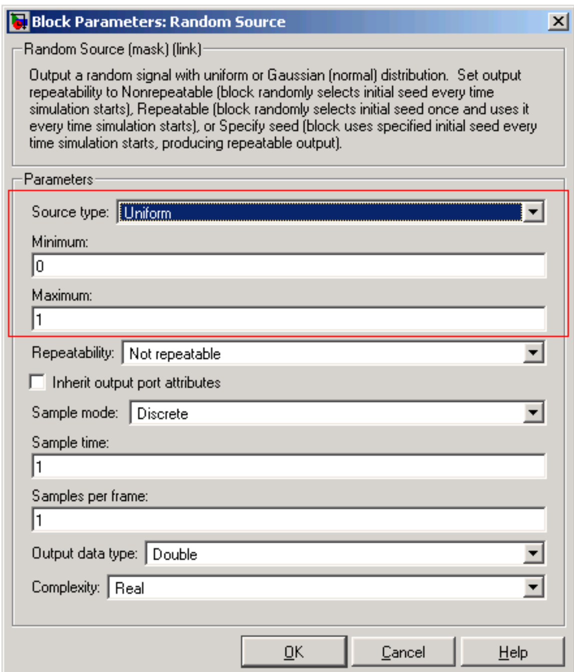
Sample Period

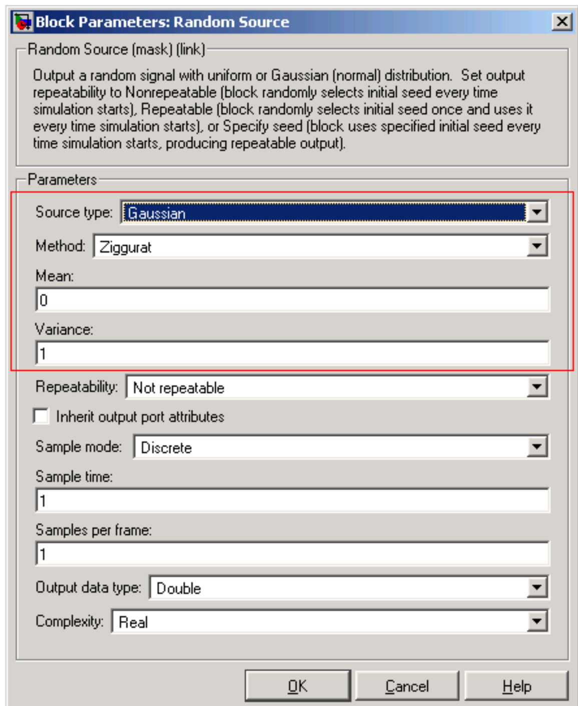
The **Sample time** parameter value, T_s, specifies the random sequence sample period when the **Sample mode** parameter is set to `Discrete`. In this mode, the block generates the number of samples specified by the **Samples per frame** parameter value, M, and outputs this frame with a period of $M*T_s$. For M=1, the output is sample based; otherwise, the output is frame based.

When **Sample mode** is set to `Continuous`, the block is configured for continuous-time operation, and the **Sample time** and **Samples per frame** parameters are disabled. Note that many Signal Processing Blockset™ blocks do not accept continuous-time inputs.

Dialog Box

Only some of the parameters described below are visible in the dialog box at any one time.





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

Source type
: The distribution from which to draw the random values, `Uniform` or `Gaussian`. For more information, see "Distribution Type" on page 2-994.

Method
: The method by which the block computes the Gaussian random values, `Ziggurat` or `Sum of uniform values`. This parameter is enabled when **Source type** is set to `Gaussian`. For more information, see "Distribution Type" on page 2-994.

Minimum
: The minimum value in the uniform distribution. This parameter is enabled when you select `Uniform` from the **Source type** parameter. Tunable.

Maximum
: The maximum value in the uniform distribution. This parameter is enabled when you select you select `Uniform` from the **Source type** parameter. Tunable.

Number of uniform values to sum
: The number of uniformly distributed random values to sum to compute a single number in a Gaussian random distribution. This parameter is enabled when the **Source type** parameter is set to `Gaussian`, and the **Method** parameter is set to `Sum of uniform values`. For more information, see "Distribution Type" on page 2-994.

Mean
: The mean of the Gaussian (normal) distribution. This parameter is enabled when you select `Gaussian` from the **Source type** parameter. Tunable.

Variance
: The variance of the Gaussian (normal) distribution. This parameter is enabled when you select `Gaussian` from the **Source type** parameter. Tunable.

Repeatability

The repeatability of the block output: `Not repeatable`, `Repeatable`, or `Specify seed`. In the `Repeatable` and `Specify seed` settings, the block outputs the same signal every time you run the simulation. For details, see "Output Repeatability" on page 2-997.

Initial seed

The initial seed(s) to use for the random number generator when you set the **Repeatability** parameter to `Specify seed`. For details, see "Specifying the Initial Seed" on page 2-997. Tunable.

Inherit output port attributes

When you select this check box, block inherits the sample mode, sample time, output data type, complexity, and signal dimensions of a sample-based signal from a downstream block. When you select this check box, the **Sample mode**, **Sample time**, **Samples per frame**, **Output data type**, and **Complexity** parameters are disabled.

Suppose you want to back propagate a 1-D vector. The output of the Random Source block is a length M sample-based 1-D vector, where length M is inherited from the downstream block. When the **Minimum**, **Maximum**, **Mean**, or **Variance** parameter specifies N channels, the 1-D vector output contains M/N samples from each channel. An error occurs in this case when M is not an integer multiple of N.

Suppose you want to back propagate a M-by-N signal. When N>1, your signal has N channels. When N = 1, your signal has M channels. The value of the **Minimum**, **Maximum**, **Mean**, or **Variance** parameter can be a scalar or a vector of length equal to the number of channels. You can specify these parameters as either row or column vectors, except when the signal is a row vector. In this case, the **Minimum**, **Maximum**, **Mean**, or **Variance** parameter must also be specified as a row vector.

Sample mode
: The sample mode, `Continuous` or `Discrete`. This parameter is enabled when the **Inherit output port attributes** check box is cleared.

Sample time
: The sample period, T_s, of the random output sequence. The output frame period is $M*T_s$. This parameter is enabled when the **Inherit output port attributes** check box is cleared.

Samples per frame
: The number of samples, M, in each output frame. When the value of this parameter is `1`, the block outputs a sample-based signal.

  This parameter is enabled when the **Inherit output port attributes** check box is cleared.

Output data type
: The data type of the output, single-precision or double-precision. This parameter is enabled when the **Inherit output port attributes** check box is cleared.

Output complexity
: The complexity of the output, `Real` or `Complex`. This parameter is enabled when the **Inherit output port attributes** check box is cleared.

Supported Data Types

- Double-precision floating-point
- Single-precision floating-point

See Also

Discrete Impulse	Signal Processing Blockset
Maximum	Signal Processing Blockset
Minimum	Signal Processing Blockset
Signal From Workspace	Signal Processing Blockset
Standard Deviation	Signal Processing Blockset
Variance	Signal Processing Blockset
Constant	Simulink
Random Number	Simulink
Signal Generator	Simulink
`rand`	MATLAB
`randn`	MATLAB

Real Cepstrum

Purpose

Compute real cepstrum of input

Library

Transforms

`dspxfrm3`

Description

The Real Cepstrum block computes the real cepstrum of each channel in the real-valued M-by-N input matrix, `u`. For both sample-based and frame-based inputs, the block assumes that each input column is a frame containing M consecutive samples from an independent channel. The block does not accept complex-valued inputs.

The output is a real M_o-by-N matrix, where you specify M_o in the **FFT length** parameter. Each output column contains the length-M_o real cepstrum of the corresponding input column.

```
y = real(ifft(log(abs(fft(u,Mo)))))
```

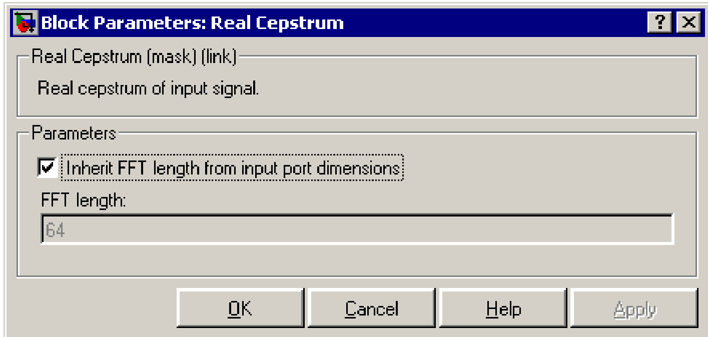
or, more compactly,

```
y = rceps(u,Mo)
```

When you select the **Inherit FFT length from input port dimensions** check box, the output frame size matches the input frame size (M_o=M). In this case, a *sample-based* length-M row vector input is processed as a single channel, that is, as an M-by-1 column vector, and the output is a length-M row vector. A 1-D vector input is *always* processed as a single channel, and the output is a 1-D vector.

The output is always sample based, and the output port rate is the same as the input port rate.

Dialog Box



Inherit FFT length from input port dimensions

When selected, matches the output frame size to the input frame size.

FFT length

The number of frequency points at which to compute the FFT, which is also the output frame size, M_o. This parameter is available when you do not select **Inherit FFT length from input port dimensions**.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Complex Cepstrum	Signal Processing Blockset
DCT	Signal Processing Blockset
FFT	Signal Processing Blockset
`rceps`	Signal Processing Toolbox

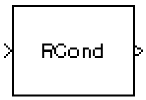
Reciprocal Condition

Purpose Compute reciprocal condition of square matrix in 1-norm

Library Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description The Reciprocal Condition block computes the reciprocal of the condition number for a square input matrix A.




```
y = rcond(A)     % Equivalent MATLAB code
```

or

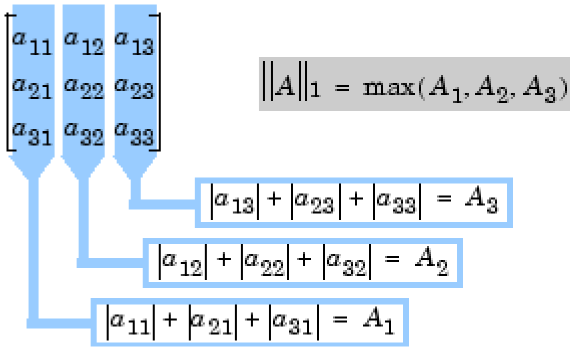
$$y = \frac{1}{\kappa} = \frac{1}{\|A^{-1}\|_1 \|A\|_1}$$

where κ is the condition number ($\kappa \geq 1$), and y is the scalar sample-based output ($0 \leq y < 1$).

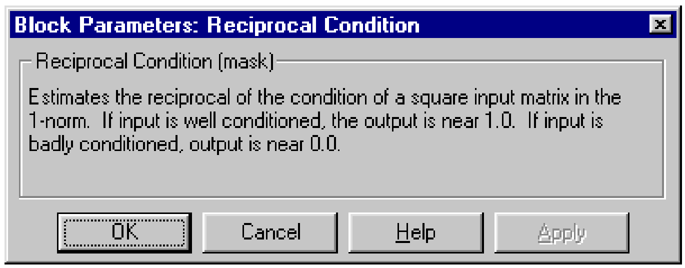
The matrix 1-norm, $\|A\|_1$, is the maximum column-sum in the M-by-M matrix A.

$$\|A\|_1 = \max_{1 \leq j \leq M} \sum_{i=1}^{M} |a_{ij}|$$

For a 3-by-3 matrix:



Dialog Box



References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Matrix 1-Norm	Signal Processing Blockset
Normalization	Signal Processing Blockset
`rcond`	MATLAB

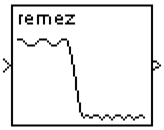
Purpose

Design and apply equiripple FIR filter

Library

dspobslib

Description



Note The Remez FIR Filter Design block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Remez FIR Filter Design block implements the Parks-McClellan algorithm to design and apply a linear-phase filter with an arbitrary multiband magnitude response. The filter design, which uses the Signal Processing Toolbox™ `firpm` function, minimizes the maximum error between the desired frequency response and the actual frequency response. Such filters are called *equiripple* due to the equiripple behavior of their approximation error. The block applies the filter to a discrete-time input using the Direct-Form II Transpose Filter block.

An M-by-N sample-based matrix input is treated as M*N independent channels, and an M-by-N frame-based matrix input is treated as N independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

The **Filter type** parameter allows you to specify one of the following filters:

- `Multiband`

  The multiband filter has an arbitrary magnitude response and linear phase.

- `Differentiator`

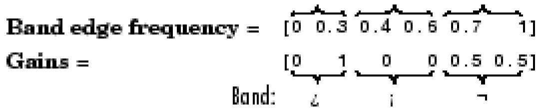
  The differentiator filter approximates the ideal differentiator. Differentiators are antisymmetric FIR filters with approximately linear magnitude responses. To obtain the correct derivative, scale the **Gains at these frequencies** vector by πF_s rad/s, where F_s is the sample frequency in Hertz.

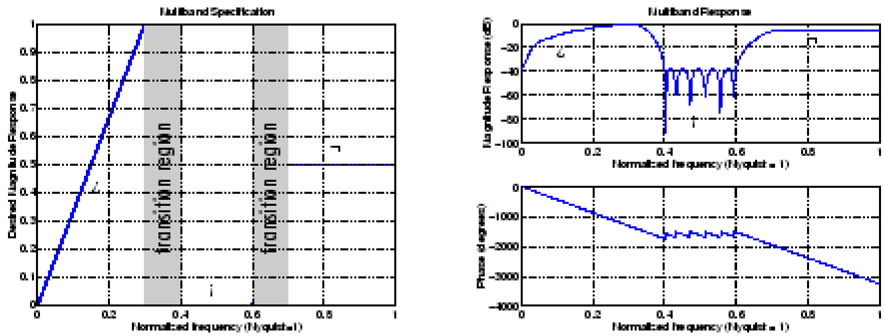
- `Hilbert Transformer`

  The Hilbert transformer filter approximates the ideal Hilbert transformer. Hilbert transformers are antisymmetric FIR filters with approximately constant magnitude.

The **Band-edge frequency vector** parameter is a vector of frequency points in the range `0` to `1`, where `1` corresponds to half the sample frequency. Each band is defined by the two bounding frequencies, so this vector must have even length. Frequency points must appear in ascending order. The **Gains at these frequencies** parameter is a vector of the same size containing the desired magnitude response at the corresponding points in the **Band-edge frequency vector**.

Each odd-indexed frequency-amplitude pair defines the left endpoint of a line segment representing the desired magnitude response in that frequency band. The corresponding even-indexed frequency-amplitude pair defines the right endpoint. Between the frequency bands specified by these end-points, there may be undefined sections of the specified frequency response. These are called “don’t care” or “transition” regions, and the magnitude response in these areas is a by-product of the optimization in the other specified frequency ranges.





The **Weights** parameter is a vector that specifies the emphasis to be placed on minimizing the error in certain frequency bands relative to others. This vector specifies one weight per band, so it is half the length of the **Band-edge frequency vector** and **Gains at these frequencies** vectors.

In most cases, differentiators and Hilbert transformers have only a single band, so the weight is a scalar value that does not affect the final filter. However, the **Weights** parameter is useful when using the block to design an antisymmetric multiband filter, such as a Hilbert transformer with stopbands.

Examples

Example 1: Multiband

Consider a lowpass filter with a transition band in the normalized frequency range 0.4 to 0.5, and 10 times greater error minimization in the stopband than in the passband.

In this case:

- **Filter type** = `Multiband`

- **Band-edge frequency vector** = `[0 0.4 0.5 1]`
- **Gains at these frequencies** = `[1 1 0 0]`
- **Weights** = `[1 10]`

Example 2: Differentiator

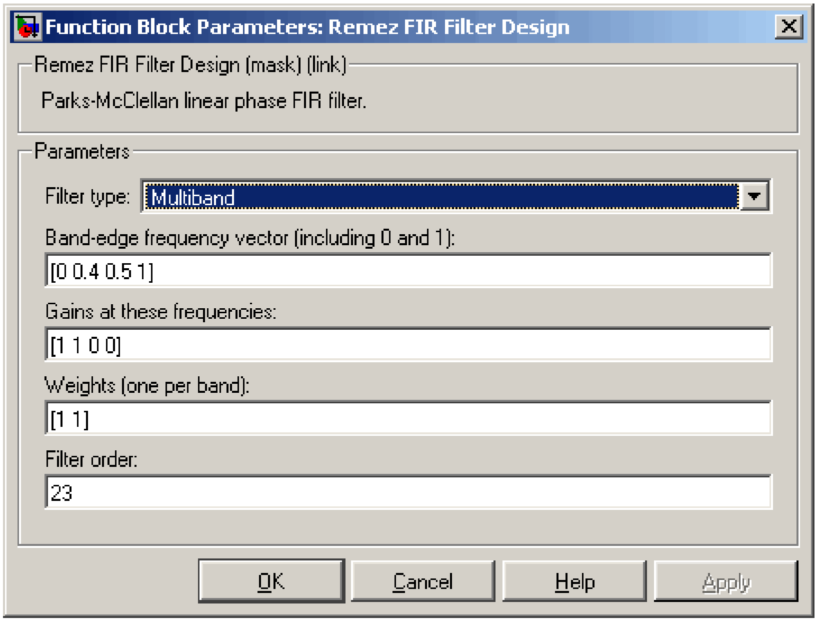
Assume the specifications for a differentiator filter require it to have order 21. The "ramp" response extends over the entire frequency range. In this case, specify:

- **Filter type** = `Differentiator`
- **Band-edge frequency vector** = `[0 1]`
- **Gains at these frequencies** = `[0 pi*Fs]`
- **Filter order** = `21`

For a type III even order filter, the differentiation band should stop short of half the sample frequency. For example, if the filter order is 20, you could specify the block parameters as follows:

- **Filter type** = `Differentiator`
- **Band-edge frequency vector** = `[0 0.9]`
- **Gains at these frequencies** = `[0 0.9*pi*Fs]`
- **Filter order** = `20`

Dialog Box



Filter type
The filter type. Tunable.

Band-edge frequency vector
A vector of frequency points, in ascending order, in the range `0` to `1`. The value `1` corresponds to half the sample frequency. This vector must have even length. Tunable.

Gains at these frequencies
A vector of frequency-response magnitudes corresponding to the points in the **Band-edge frequency** vector. This vector must be the same length as the **Band-edge frequency** vector. Tunable.

Weights
A vector containing one weight for each frequency band. This vector must be half the length of the **Band-edge frequency** and **Gains at these frequencies** vectors. Tunable.

Filter order

The filter order.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Purpose

Resample input at higher rate by repeating values

Library

Signal Operations

`dspsigops`

Description

The Repeat block upsamples each channel of the M_i-by-N input to a rate L times higher than the input sample rate by repeating each consecutive input sample L times at the output. You specify the integer L in the **Repetition count** parameter.

This block supports triggered subsystems if, for **Frame-based mode,** you select `Maintain input frame rate`.

Sample-Based Operation

When the input is sample based, the block treats each of the M*N matrix elements as an independent channel, and upsamples each channel over time. The **Frame-based mode** parameter must be set to `Maintain input frame size`. The output sample rate is L times higher than the input sample rate ($T_{so} = T_{si}/L$), and the input and output sizes are identical.

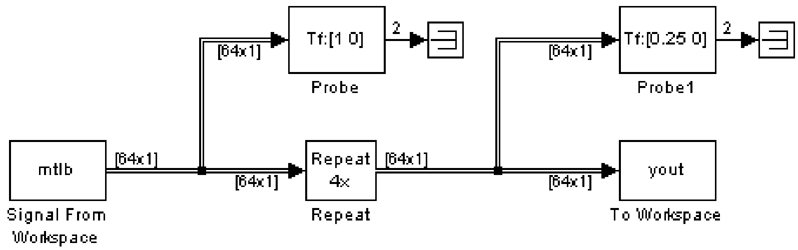
Frame-Based Operation

When the input is frame based, the block treats each of the N input columns as a frame containing M_i sequential time samples from an independent channel. The block upsamples each channel independently by repeating each row of the input matrix L times at the output. The **Frame-based mode** parameter determines how the block adjusts the rate at the output to accommodate the repeated rows. There are two available options:

- `Maintain input frame size`

  The block generates the output at the faster (upsampled) rate by using a proportionally shorter frame *period* at the output port than at the input port. For L repetitions of the input, the output frame period is L times shorter than the input frame period ($T_{fo} = T_{fi}/L$), but the input and output frame sizes are equal.

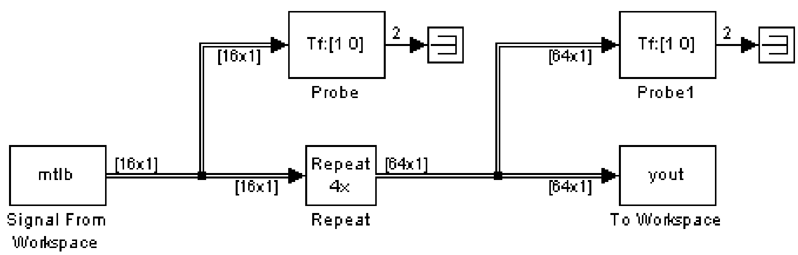
The model below shows a single-channel input with a frame period of 1 second being upsampled through 4-times repetition to a frame period of 0.25 second. The input and output frame sizes are identical.



- `Maintain input frame rate`

  The block generates the output at the faster (upsampled) rate by using a proportionally larger frame *size* than the input. For L repetitions of the input, the output frame size is L times larger than the input frame size ($M_o = M_i$*L), but the input and output frame rates are equal.

  The model below shows a single-channel input of frame size 16 being upsampled through 4-times repetition to a frame size of 64. The input and output frame rates are identical.



Zero Latency

The Repeat block has *zero-tasking latency* for all single-rate operations. The block is single-rate for the particular combinations of sampling mode and parameter settings shown in the table below.

Sampling Mode	Parameter Settings
Sample based	**Repetition count** parameter, L, is `1`.
Frame based	**Repetition count** parameter, L, is `1`, *or* **Frame-based mode** parameter is `Maintain input frame rate`.

The block also has zero latency for all multirate operations in the Simulink® single-tasking mode.

Zero tasking latency means that the block repeats the first input (received at *t*=0) for the first L output samples, the second input for the next L output samples, and so on. The **Initial condition** parameter value is not used.

Nonzero Latency

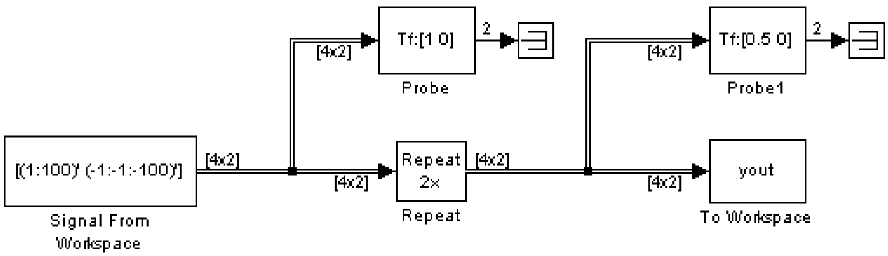
The Repeat block has tasking latency only for multirate operation in the Simulink multitasking mode:

- In sample-based mode, the initial condition for each channel is repeated for the first L output samples. The channel's first input appears as output sample L+1. The **Initial condition** value can be an M_i-by-N matrix containing one value for each channel, or a scalar to be applied to all signal channels.
- In frame-based mode, the first row of the initial condition matrix is repeated for the first L output samples, the second row of the initial condition matrix is repeated for the next L output samples, and so on. The first row of the first input matrix appears in the output as sample M_iL+1. The **Initial condition** value can be an M_i-by-N matrix, or a scalar to be repeated across all elements of the M_i-by-N matrix. See the example below for an illustration of this case.

Note For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Examples

Construct the frame-based model shown below.



Adjust the block parameters as follows.

- Configure the Signal From Workspace block to generate a two-channel signal with frame size of 4 and sample period of 0.25. This represents an output frame period of 1 (0.25*4). The first channel should contain the positive ramp signal 1, 2, ..., 100, and the second channel should contain the negative ramp signal -1, -2, ..., -100.
  - **Signal** = `[(1:100)' (-1:-1:-100)']`
  - **Sample time** = `0.25`
  - **Samples per frame** = `4`
- Configure the Repeat block to upsample the two-channel input by increasing the output frame rate by a factor of 2 relative to the input frame rate. Set an initial condition matrix of

$$\begin{bmatrix} 11 & -11 \\ 12 & -12 \\ 13 & -13 \\ 14 & -14 \end{bmatrix}$$

- **Repetition count** = `2`
- **Initial condition** = `[11 -11;12 -12;13 -13;14 -14]`
- **Frame-based mode** = `Maintain input frame size`

- Configure the Probe blocks by clearing the **Probe width** and **Probe complex signal** check boxes (if desired).

This model is multirate because there are at least two distinct sample rates, as shown by the two Probe blocks. To run this model in the Simulink multitasking mode, in the **Solver** pane of the Configuration Parameters dialog box, set the **Type** list to `Fixed-step` and set the **Solver** list to `discrete (no continuous states)`. For the **Tasking mode for periodic sample times** parameter, select `MultiTasking`. Also set the **Stop time** to `30`.

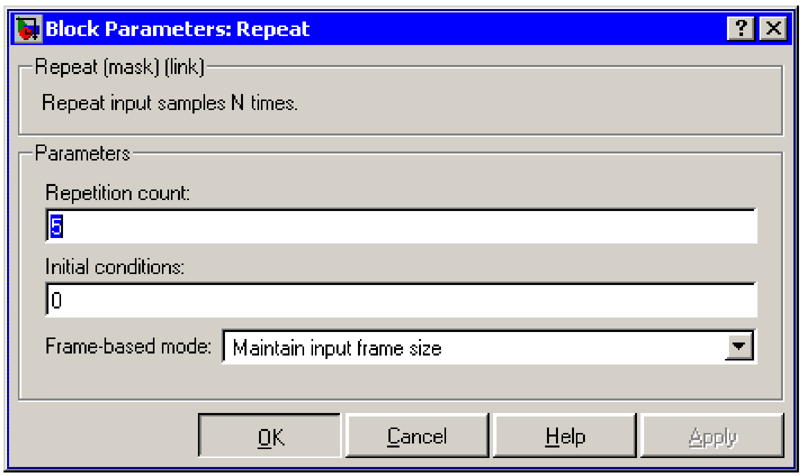
Run the model and look at the output, `yout`. The first few samples of each channel are shown below.

```
yout =
    11   -11
    11   -11
    12   -12
    12   -12
    13   -13
    13   -13
    14   -14
    14   -14
     1    -1
     1    -1
     2    -2
     2    -2
```

```
3    -3
3    -3
4    -4
4    -4
5    -5
5    -5
```

Since we ran this frame-based multirate model in multitasking mode, the block repeats each row of the initial condition matrix for L output samples, where L is the **Repetition count** of 2. The first row of the first input matrix appears in the output as sample 9 ,that is, sample M_iL+1, where M_i is the input frame size.

Dialog Box



Repetition count
: The integer number of times, L, that the input value is repeated at the output. This is the factor by which the output frame size or sample rate is increased.

Initial conditions
: The value with which the block is initialized for cases of nonzero latency; a scalar or matrix.

Frame-based mode

For frame-based operation, the method by which to implement the repetition (upsampling): `Maintain input frame size` that is, increase the frame rate, or `Maintain input frame rate`, that is, increase the frame size. The **Frame-based mode** parameter must be set to `Maintain input frame size` for sample-base inputs.

Supported Data Types

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

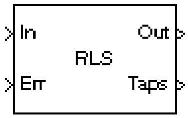
See Also

FIR Interpolation	Signal Processing Blockset
Upsample	Signal Processing Blockset

Purpose Compute filter estimates for input using RLS adaptive filter algorithm

Library `dspobslib`

Description



Note The RLS Adaptive Filter block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the RLS Filter block.

The RLS Adaptive Filter block recursively computes the recursive least squares (RLS) estimate of the FIR filter coefficients.

The corresponding RLS filter is expressed in matrix form as

$$
\begin{aligned}
k(n) &= \frac{\lambda^{-1}P(n-1)u(n)}{1+\lambda^{-1}u^{H}(n)P(n-1)u(n)} \\
y(n) &= \hat{w}^{H}(n-1)u(n) \\
e(n) &= d(n)-y(n) \\
\hat{w}(n) &= \hat{w}(n-1)+k(n)e^{*}(n) \\
P(n) &= \lambda^{-1}P(n-1)-\lambda^{-1}k(n)u^{H}(n)P(n-1)
\end{aligned}
$$

where λ^{-1} denotes the reciprocal of the exponential weighting factor. The variables are as follows

Variable	Description
n	The current algorithm iteration
$u(n)$	The buffered input samples at step n
$P(n)$	The inverse correlation matrix at step n
$k(n)$	The gain vector at step n
$\hat{w}(n)$	The vector of filter-tap estimates at step n

Variable	Description
$y(n)$	The filtered output at step n
$e(n)$	The estimation error at step n
$d(n)$	The desired response at step n
λ	The exponential memory weighting factor

The block icon has port labels corresponding to the inputs and outputs of the RLS algorithm. Note that inputs to the `In` and `Err` ports must be sample-based scalars. The signal at the `Out` port is a scalar, while the signal at the `Taps` port is a sample-based vector.

Block Ports	Corresponding Variables
`In`	u, the scalar input, which is internally buffered into the vector $u(n)$
`Out`	$y(n)$, the filtered scalar output
`Err`	$e(n)$, the scalar estimation error
`Taps`	$\hat{w}(0)$, the vector of filter-tap estimates

An optional `Adapt` input port is added when you select the **Adapt input** check box in the dialog box. When this port is enabled, the block continuously adapts the filter coefficients while the `Adapt` input is nonzero. A zero-valued input to the `Adapt` port causes the block to stop adapting, and to hold the filter coefficients at their current values until the next nonzero `Adapt` input.

The implementation of the algorithm in the block is optimized by exploiting the symmetry of the inverse correlation matrix $P(n)$. This decreases the total number of computations by a factor of two.

The **FIR filter length** parameter specifies the length of the filter that the RLS algorithm estimates. The **Memory weighting factor** corresponds to λ in the equations, and specifies how quickly the filter "forgets" past sample information. Setting `λ=1` specifies an infinite memory; typically, `0.95≤λ≤1`.

The **Initial value of filter taps** specifies the initial value $\hat{w}(0)$ as a vector, or as a scalar to be repeated for all vector elements. The initial value of $P(n)$ is

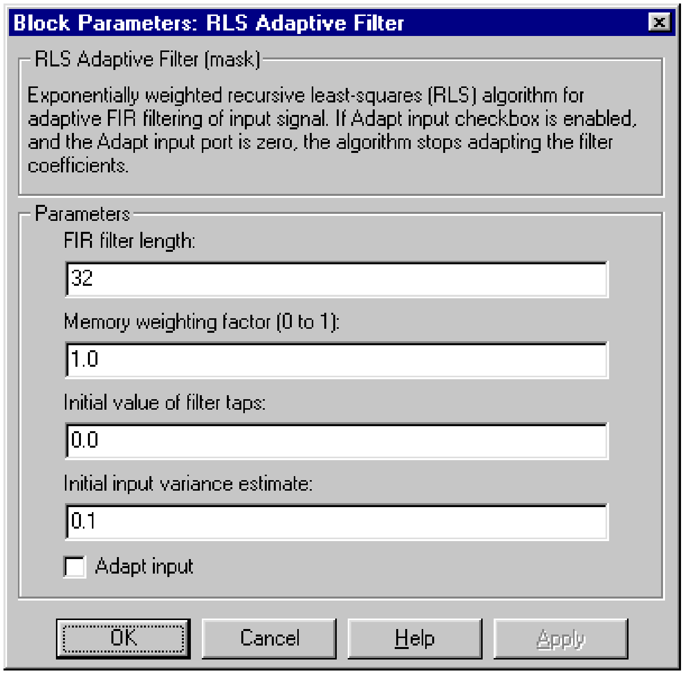
$$I\frac{1}{\hat{\sigma}^2}$$

where you specify $\hat{\sigma}^2$ in the **Initial input variance estimate** parameter.

Examples

The `rlsdemo` demo illustrates a noise cancellation system built around the RLS Adaptive Filter block.

Dialog Box



FIR filter length
: The length of the FIR filter.

Memory weighting factor
: The exponential weighting factor, in the range `[0,1]`. A value of `1` specifies an infinite memory. Tunable.

Initial value of filter taps
: The initial FIR filter coefficients.

Initial input variance estimate
: The initial value of 1/P(n).

Adapt input
: Enables the `Adapt` port.

References

Haykin, S. *Adaptive Filter Theory*. 3rd ed. Englewood Cliffs, NJ: Prentice Hall, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Kalman Adaptive Filter	Signal Processing Blockset
LMS Adaptive Filter	Signal Processing Blockset

See “Adaptive Filters” for related information.

RLS Filter

Purpose

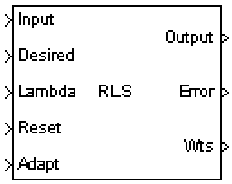
Compute filtered output, filter error, and filter weights for given input and desired signal using RLS adaptive filter algorithm

Library

Filtering / Adaptive Filters

`dspadpt3`

Description



RLS Filter

The RLS Filter block recursively computes the least squares estimate (RLS) of the FIR filter weights. The block estimates the filter weights, or coefficients, needed to convert the input signal into the desired signal. Connect the signal you want to filter to the Input port. This input signal can be a sample-based scalar or a single-channel frame-based signal. Connect the signal you want to model to the Desired port. The desired signal must have the same data type, frame status, complexity, and dimensions as the input signal. The Output port outputs the filtered input signal, which can be sample or frame based. The Error port outputs the result of subtracting the output signal from the desired signal.

The corresponding RLS filter is expressed in matrix form as

$$\mathbf{k}(n) = \frac{\lambda^{-1}\mathbf{P}(n-1)\mathbf{u}(n)}{1+\lambda^{-1}\mathbf{u}^{H}(n)\mathbf{P}(n-1)\mathbf{u}(n)}$$

$$y(n) = \mathbf{w}(n-1)\mathbf{u}(n)$$

$$e(n) = d(n) - y(n)$$

$$\mathbf{w}(n) = \mathbf{w}(n-1) + \mathbf{k}^{H}(n)e(n)$$

$$\mathbf{P}(n) = \lambda^{-1}\mathbf{P}(n-1) - \lambda^{-1}\mathbf{k}(n)\mathbf{u}^{H}(n)\mathbf{P}(n-1)$$

where λ^{-1} denotes the reciprocal of the exponential weighting factor. The variables are as follows

Variable	Description
n	The current time index
$\mathbf{u}(n)$	The vector of buffered input samples at step n

Variable	Description
$\mathbf{P}(n)$	The inverse correlation matrix at step n
$\mathbf{k}(n)$	The gain vector at step n
$\mathbf{w}(n)$	The vector of filter-tap estimates at step n
$y(n)$	The filtered output at step n
$e(n)$	The estimation error at step n
$d(n)$	The desired response at step n
λ	The forgetting factor

The implementation of the algorithm in the block is optimized by exploiting the symmetry of the inverse correlation matrix $P(n)$. This decreases the total number of computations by a factor of two.

Use the **Filter length** parameter to specify the length of the filter weights vector.

The **Forgetting factor (0 to 1)** parameter corresponds to λ in the equations. It specifies how quickly the filter "forgets" past sample information. Setting λ=1 specifies an infinite memory. Typically,

$1 - \frac{1}{2L} < \lambda < 1$, where L is the filter length. You can specify a forgetting factor using the input port, Lambda, or enter a value in the **Forgetting factor (0 to 1)** parameter in the Block Parameters: RLS Filter dialog box.

Enter the initial filter weights, $\widehat{w}(0)$, as a vector or a scalar for the **Initial value of filter weights** parameter. When you enter a scalar, the block uses the scalar value to create a vector of filter weights. This vector has length equal to the filter length and all of its values are equal to the scalar value.

The initial value of $P(n)$ is

$$\frac{1}{\sigma^2} I$$

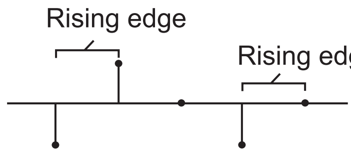
where you specify σ^2 in the **Initial input variance estimate** parameter.

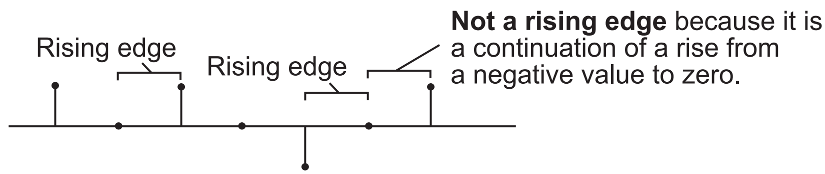
When you select the **Adapt port** check box, an Adapt port appears on the block. When the input to this port is nonzero, the block continuously updates the filter weights. When the input to this port is zero, the filter weights remain at their current values.

When you want to reset the value of the filter weights to their initial values, use the **Reset input** parameter. The block resets the filter weights whenever a reset event is detected at the Reset port. The reset signal rate must be the same rate as the data signal input.

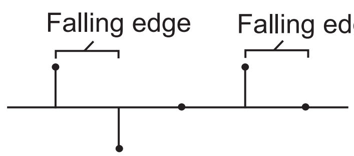
From the **Reset input** list, select `None` to disable the Reset port. To enable the Reset port, select one of the following from the **Reset input** list:

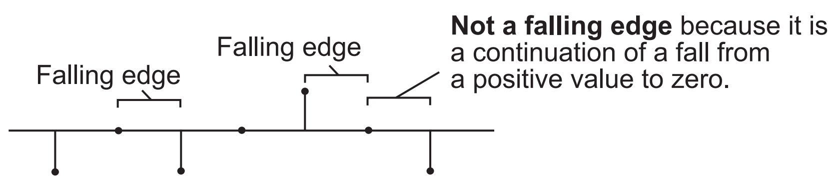
- `Rising edge` — Triggers a reset operation when the Reset input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero; see the following figure





- `Falling edge` — Triggers a reset operation when the Reset input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero; see the following figure





- `Either edge` — Triggers a reset operation when the Reset input is a `Rising edge` or `Falling edge`, as described above
- `Non-zero sample` — Triggers a reset operation at each sample time that the Reset input is not zero

---

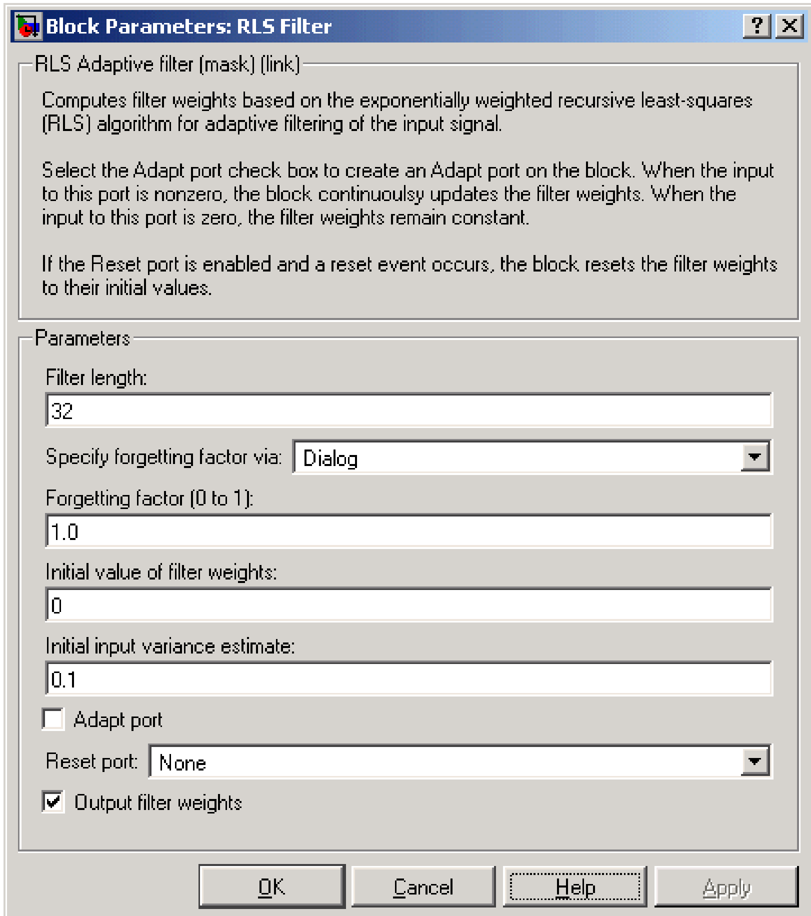
Note When running simulations in the Simulink® `MultiTasking` mode, sample-based reset signals have a one-sample latency, and frame-based reset signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a reset event, and when it applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Select the **Output filter weights** check box to create a Wts port on the block. For each iteration, the block outputs the current updated filter weights from this port.

Examples

The `rlsdemo` demo illustrates a noise cancellation system built around the RLS Filter block.

Dialog Box



Filter length

Enter the length of the FIR filter weights vector.

Specify forgetting factor via

Select `Dialog` to enter a value for the forgetting factor in the Block parameters: RLS Filter dialog box. Select `Input port` to specify the forgetting factor using the Lambda input port.

Forgetting factor (0 to 1)
Enter the exponential weighting factor in the range $0 \leq \lambda \leq 1$. A value of 1 specifies an infinite memory. Tunable.

Initial value of filter weights
Specify the initial values of the FIR filter weights.

Initial input variance estimate
The initial value of 1/P(*n*).

Adapt port
Select this check box to enable the Adapt input port.

Reset input
Select this check box to enable the Reset input port.

Output filter weights
Select this check box to export the filter weights from the Wts port.

References

Hayes, M.H. *Statistical Digital Signal Processing and Modeling.* New York: John Wiley & Sons, 1996.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Kalman Adaptive Filter	Signal Processing Blockset
LMS Filter	Signal Processing Blockset
Block LMS Filter	Signal Processing Blockset
Fast Block LMS Filter	Signal Processing Blockset

See "Adaptive Filters" for related information.

Purpose

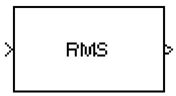
Compute root-mean-square value of input or sequence of inputs

Library

Statistics

`dspstat3`

Description



The RMS block computes the RMS value of each row or column of the input, along vectors of a specified dimension of the input, or of the entire input. The RMS block can also track the RMS value in a sequence of inputs over a period of time. The **Running RMS** parameter selects between basic operation and running operation.

Basic Operation

When you do not select the **Running RMS** check box, the block computes the RMS value of each row or column of the input, along vectors of a specified dimension of the input, or of the entire input at each individual sample time, and outputs the array y. Each element in y is the RMS value of the corresponding column, row, vector, or entire input. The output y depends on the setting of the **Find the RMS value over** parameter. For example, consider a 3-dimensional input signal of size M-by-N-by-P:

- `Entire input` — The output at each sample time is a scalar that contains the RMS value of the entire input. In this mode, the output is always sample based.
- `Each row` — The output at each sample time consists of an M-by-1-by-P array, where each element contains the RMS value of each vector over the second dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is an M-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.
- `Each column` — The output at each sample time consists of a 1-by-N-by-P array, where each element contains the RMS value of each vector over the first dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is a 1-by-N

row vector. In this mode, the frame status of the output is the same as that of the input.

For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors when the block is in this mode. Sample-based length-M row vector inputs are also treated as M-by-1 column vectors when the **Treat sample-based row input as a column** check box is selected.

- `Specified dimension` — The output at each sample time depends on **Dimension**. If **Dimension** is set to 1, the output is the same as that when you select `Each column`. If **Dimension** is set to 2, the output is the same as when you select `Each row`. If **Dimension** is set to 3, the output at each sample time is an M-by-N matrix containing the RMS value of each vector over the third dimension of the input. In this mode, the frame status of the output is the same as that of the input.

The RMS value of the jth column of an M-by-N input matrix u is given by

$$y_j = \sqrt{\frac{\sum_{i=1}^{M} \left|u_{ij}\right|^2}{M}} \qquad 1 \le j \le N$$

```
y = sqrt(sum(u.*conj(u))/size(u,1))     % Equivalent MATLAB code
```

Running Operation

When you select the **Running RMS** check box, the block tracks the RMS value of successive inputs to the block. For sample-based M-by-N inputs, the output is a sample-based M-by-N matrix, with each element y_{ij} containing the RMS value of element u_{ij} over all inputs since the last reset. For frame-based M-by-N inputs, the output is a frame-based M-by-N matrix with each element y_{ij} containing the RMS value of the jth column over all inputs since the last reset, up to and including element u_{ij} of the current input.

N-D signals cannot be frame based. When the **Running RMS** check box is selected, each element of the N-D signal is treated as a separate channel.

There are $\prod d_i$ channels, where d_i is the size of the ith dimension.

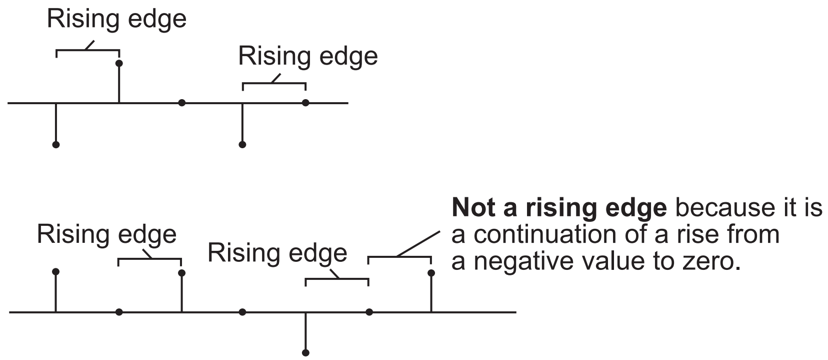
Resetting the Running RMS

The block resets the running RMS whenever a reset event is detected at the optional `Rst` port. The reset sample time must be a positive integer multiple of the input sample time.

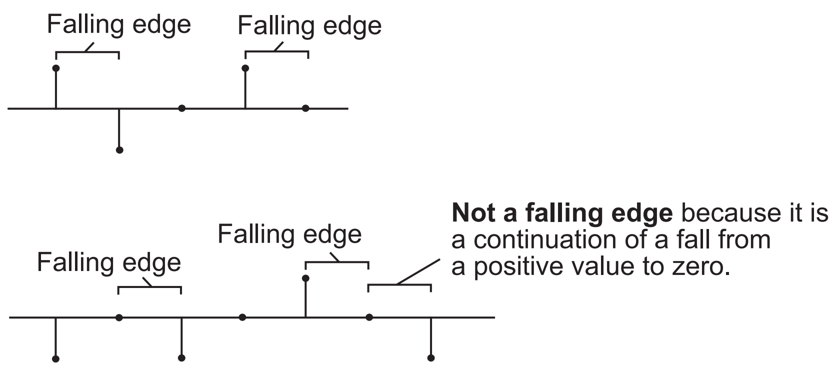
When the block is reset for sample-based inputs, the running RMS for each channel is initialized to the value in the corresponding channel of the current input. For frame-based inputs, the running RMS for each channel is initialized to the earliest value in each channel of the current input.

You specify the reset event in the **Reset port** parameter:

- `None` disables the `Rst` port.
- `Rising edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)



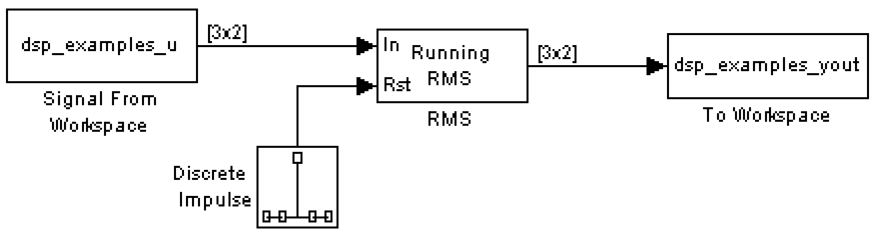
- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge`, as described earlier
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero

---

Note When running simulations in the Simulink® MultiTasking mode, reset signals have a one-sample latency. Therefore, when the block detects a reset event, there is a one-sample delay at the reset port rate before the block applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Examples

The RMS block in the next model calculates the running RMS of a frame-based 3-by-2 (two-channel) matrix input, `u`. The running RMS is reset at t=2 by an impulse to the block's `Rst` port.



The RMS block has the following settings:

- **Running RMS** = Select this check box.
- **Reset port** = `Non-zero sample`

The Signal From Workspace block has the following settings:

- **Signal** = `dsp_examples_u`
- **Sample time** = `1/3`
- **Samples per frame** = `3`

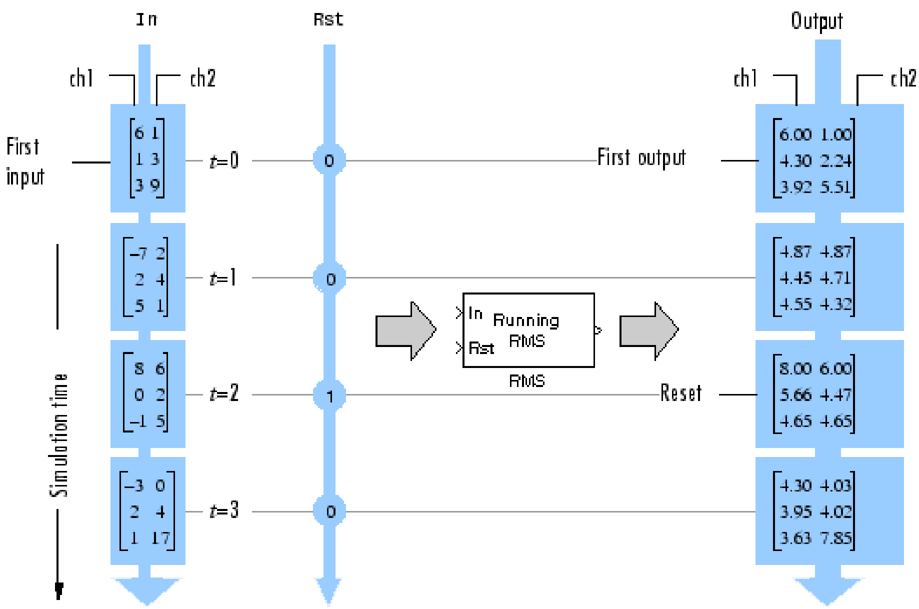
where

```
dsp_examples_u = [6 1 3 -7 2 5 8 0 -1 -3 2 1;1 3 9 2 4 1 6 2 5 0 4 17]'
```

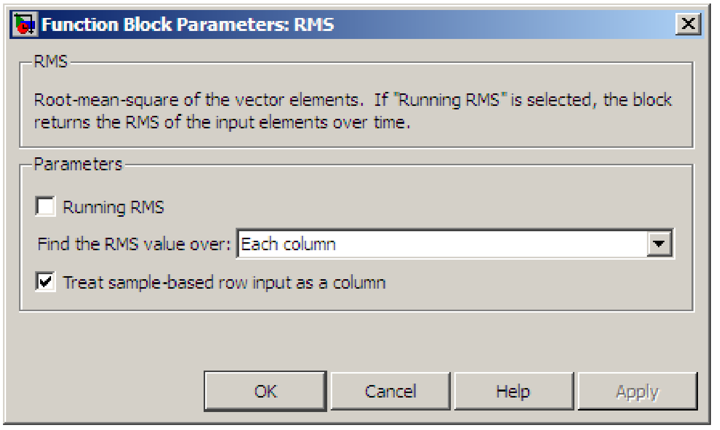
The Discrete Impulse block has the following settings:

- **Delay (samples)** = 2
- **Sample time** = 1
- **Samples per frame** = 1

The block's operation is shown in the next figure.



Dialog Box



Running RMS
Enables running operation when selected.

Reset port
Determines the reset event that causes the block to reset the running RMS. The reset signal rate must be a positive integer multiple of the rate of the data signal input. This parameter is enabled only when you set the **Running RMS** parameter. For more information, see "Resetting the Running RMS" on page 2-1035.

Find the RMS value over
Specify whether to find the RMS value along rows, columns, entire input, or the dimension specified in the **Dimension** parameter. For more information, see "Basic Operation" on page 2-1033.

Treat sample-based row input as a column
Select to treat sample-based length-M row vector inputs as M-by-1 column vectors. This parameter is only visible when the **Find the RMS value over** parameter is set to `Each column`.

Dimension
Specify the dimension (one-based value) of the input signal, over which the RMS value is computed. The value of this parameter

cannot exceed the number of dimensions in the input signal. This parameter is only visible when the **Find the RMS value over** parameter is set to `Specified dimension`.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Boolean — The block accepts Boolean inputs to the `Rst` port.

See Also

Mean	Signal Processing Blockset
Variance	Signal Processing Blockset

Purpose

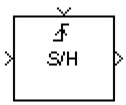
Sample and hold input signal

Library

Signal Operations

`dspsigops`

Description



The Sample and Hold block acquires the input at the signal port whenever it receives a trigger event at the trigger port (marked by). The block then holds the output at the acquired input value until the next triggering event occurs. When the acquired input is frame based, the output is frame based; otherwise, the output is sample based.

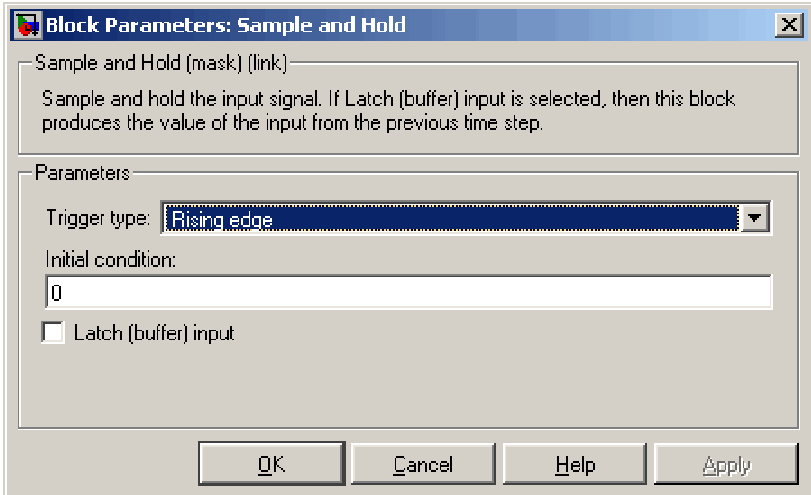
The trigger input must be a sample-based scalar with sample rate equal to the input frame rate at the signal port. You specify the trigger event in the **Trigger type** pop-up menu:

- `Rising edge` triggers the block to acquire the signal input when the trigger input rises from a negative value or zero to a positive value.
- `Falling edge` triggers the block to acquire the signal input when the trigger input falls from a positive value or zero to a negative value.
- `Either edge` triggers the block to acquire the signal input when the trigger input either rises from a negative value or zero to a positive value or falls from a positive value or zero to a negative value.

You specify the block's output prior to the first trigger event using the **Initial condition** parameter. When the acquired input is an M-by-N matrix, the **Initial condition** can be an M-by-N matrix, or a scalar to be repeated across all elements of the matrix. When the input is a length-M 1-D vector, the **Initial condition** can be a length-M row or column vector, or a scalar to be repeated across all elements of the vector.

If you select the **Latch (buffer) input** check box, the block outputs the value of the input from the previous time step until the next triggering event occurs. To use this block in a loop, select this check box.

Dialog Box



Trigger type
: The type of event that triggers the block to acquire the input signal.

Initial condition
: The block's output prior to the first trigger event.

Latch (buffer) input
: If you select this check box, the block outputs the value of the input from the previous time step until the next triggering event occurs.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Trigger	• Any data type supported by the Trigger block
Outputs	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers

See Also

Downsample	Signal Processing Blockset
N-Sample Switch	Signal Processing Blockset

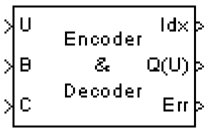
Purpose

Convert input signal into set of quantized output values or index values, or convert set of index values into quantized output signal

Library

`dspobslib`

Description



Scalar Quantizer

Note The Scalar Quantizer block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Scalar Quantizer Encoder block or the Scalar Quantizer Decoder block.

The Scalar Quantizer block has three modes of operation. In `Encoder` mode, the block maps each input value to a quantization region by comparing the input value to the quantizer boundary points defined in the **Boundary points** parameter. The block outputs the index of the associated region. In `Decoder` mode, the block transforms the input index values into quantized output values, defined in the **Codebook** parameter. In the `Encoder and Decoder` mode, the block performs both the encoding and decoding operations. The block outputs the index values and the quantized output values.

You can select how you want to enter the **Boundary points** and/or **Codebook** values using the **Source of quantizer** parameters. When you select `Specify via dialog`, type the parameters into the block parameters dialog box. Select `Input ports`, and port B and/or C appears on the block. In `Encoder` and `Encoder and decoder` mode, the input to port B is used as the **Boundary points**. In `Decoder` and `Encoder and decoder` mode, the input to port C is used as the **Codebook**.

In `Encoder` and `Encoder and decoder` mode, the **Boundary points** are the values used to break up the input signal into regions. Each region is specified by an index number. When your first boundary point is `-inf` and your last boundary point is `inf`, your quantizer is unbounded. When your first and last boundary point is finite, your quantizer is bounded. When only your first or last boundary point is `-inf` or `inf`, your quantizer is semi-bounded.

For instance, when your input signal ranges from 0 to 11, you can create a bounded quantizer using the following boundary points:

```
[0 0.5 3.7 5.8 6.0 11]
```

The boundary points can have equal or varied spacing. Any input values between 0 and 0.5 would correspond to index 0. Input values between 0.5 and 3.7 would correspond to index 1, and so on.

Suppose you wanted to create an unbounded quantizer with the following boundary points:

```
[-inf 0 2 5.5 7.1 10 inf]
```

When your input signal has values less than 0, these values would be assigned to index 0. When your input signal has values greater than 10, these values would be assigned to index 6.

When an input value is the same as a boundary point, the **Tie-breaking rule** parameter defines the index to which the value is assigned. When you want the input value to be assigned to the lower index value, select `Choose the lower index`. To assign the input value with the higher index, select `Choose the higher index`.

In `Decoder` and `Encoder and decoder` mode, the **Codebook** is a vector of quantized output values that correspond to each index value.

In `Encoder` and `Encoder and decoder` mode, the **Searching method** determines how the appropriate quantizer index is found. Select `Linear` and the Scalar Quantizer block compares the input value to the first region defined by the first two boundary points. When the input value does not fall within this region, the block then compares the input value to the next region. This process continues until the input value is determined to be within a region and is associated with the appropriate index value. The computational cost of this process is of the order P, where P is the number of boundary points.

Select Binary for the **Searching method** and the block compares the input value to the middle value of the boundary points vector. When the input value is larger than this boundary point, the block discards the boundary points that are lower than this middle value. The block then compares the input value to the middle boundary point of the new range, defined by the remaining boundary points. This process continues until the input value is associated with the appropriate index value. The computational cost of this process is of the order $\log_2 P$, where P is the number of boundary points. In most cases, the Binary option is faster than the Linear option.

In Decoder mode, the input to this block is a vector of index values, where $0 \le \text{index} < N$ and N is the length of the codebook vector. Use the **Action for out of range input** parameter to determine what happens when an input index value is out of this range. When you want any index values less than 0 to be set to 0 and any index values greater than or equal to N to be set to N -1, select Clip. When you want to be warned when any index values less than 0 are set to 0 and any index values greater than or equal to N are set to N -1, select Clip and warn. When you want the simulation to stop and display an error when the index values are out of range, select Error.

In Encoder and decoder mode, you can select the **Output the quantization error** check box. The quantization error is the difference between the input value and the quantized output value. Select this check box to output the quantization error for each input value from the Err port on this block.

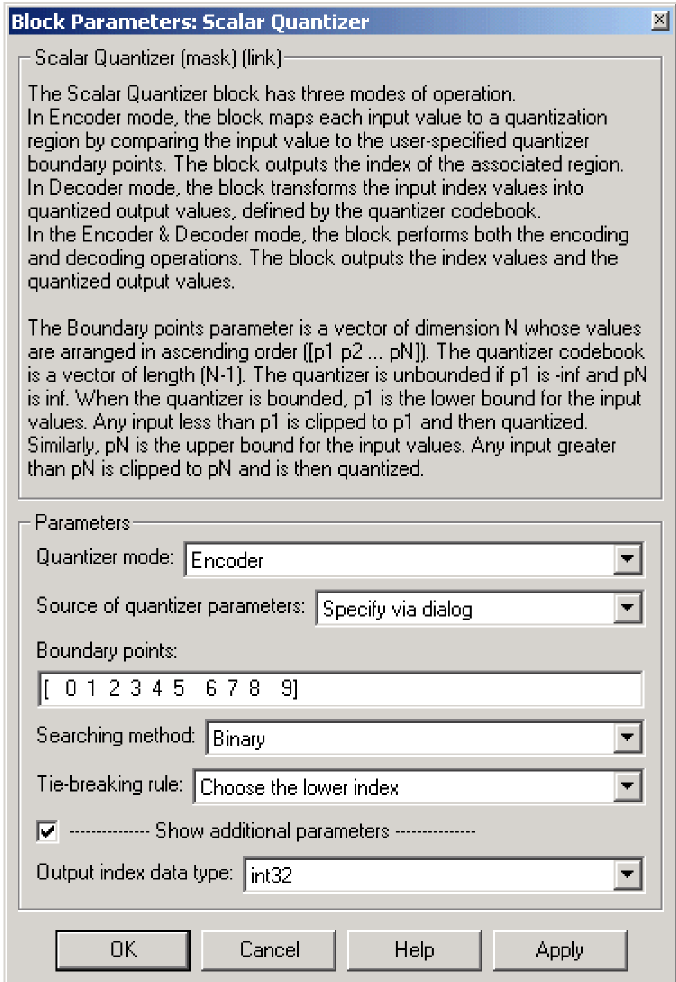
Data Type Support

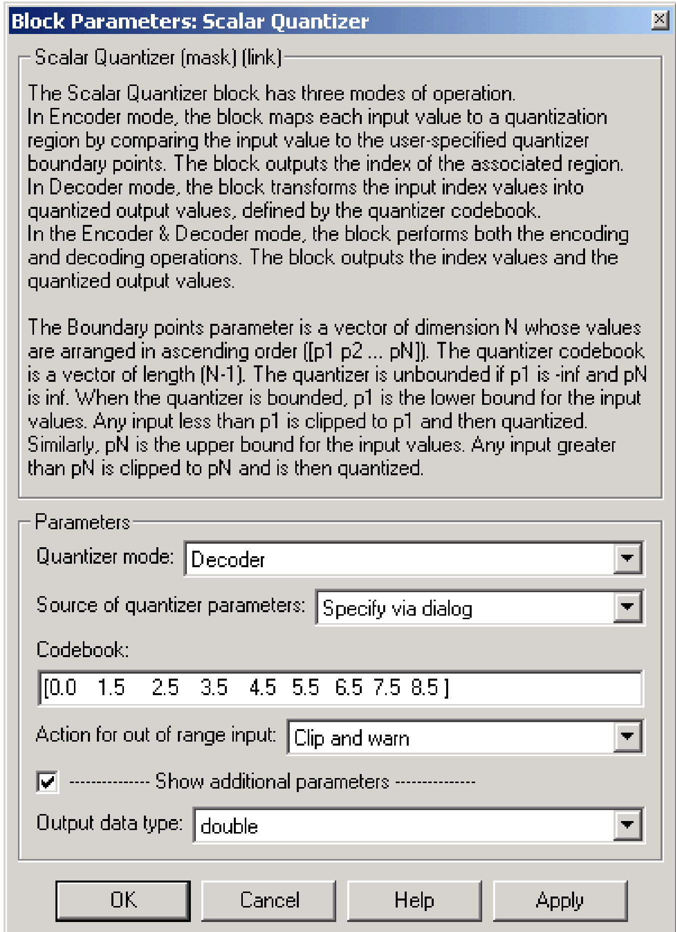
In Encoder mode, the input data values and the boundary points can be the input to the block at ports U and B. Similarly, in Encoder and decoder mode, the codebook values can also be the input to the block at port C. The data type of the input data values, boundary points, and codebook values can be double, single, uint8, uint16, uint32, int8, int16, or int32. In Decoder mode, the input to the block can be the index values and the codebook values. The data type of the index input to the block at port Idx can be uint8, uint16, uint32, int8, int16, or int32. The data type of the codebook values can be double, single, uint8, uint16, uint32, int8, int16, or int32.

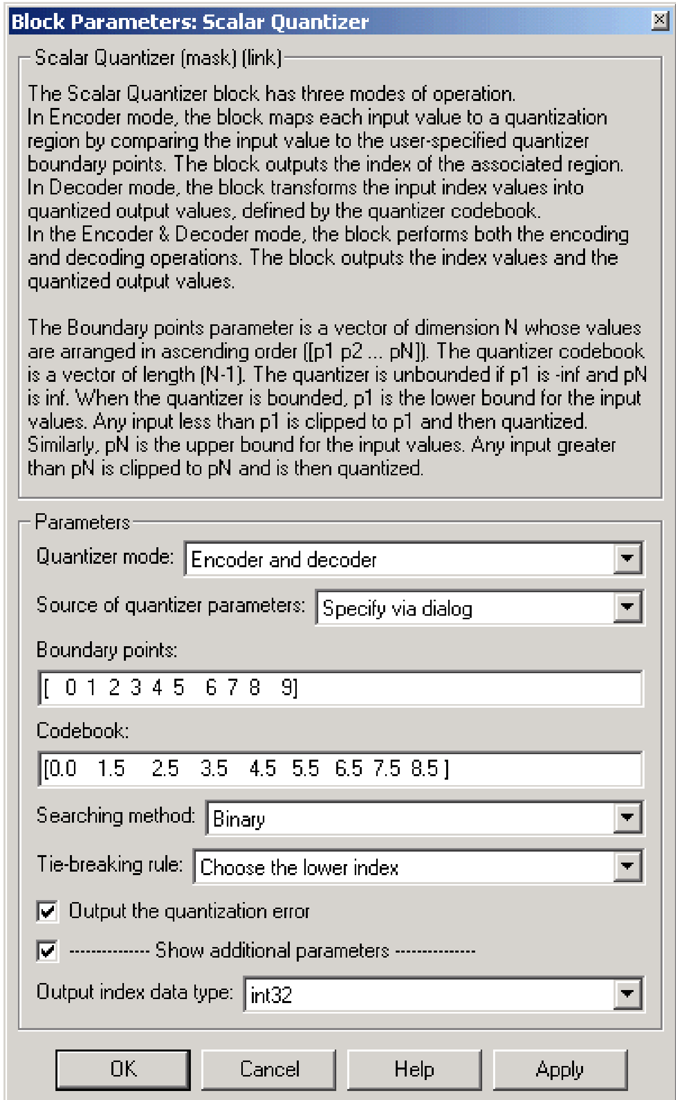
In Encoder mode, the output of the block is the index values. In Encoder and decoder mode, the output can also include the quantized output values and the quantization error. In Encoder and Encoder and decoder mode, use the **Output index data type** parameter to specify the data type of the index output from the block at port Idx. The data type of the index output can be uint8, uint16, uint32, int8, int16, or int32. The data type of the quantized output and the quantization error can be double, single, uint8, uint16, uint32, int8, int16, or int32. In Decoder mode, the output of the block is the quantized output values. Use the **Output data type** parameter to specify the data type of the quantized output values. The data type can be double, single, uint8, uint16, uint32, int8, int16, int32.

Note The input data, codebook values, boundary points, quantization error, and the quantized output values must have the same data type whenever they are present in any of the quantizer modes.

Dialog Box


Block Parameters: Scalar Quantizer
Scalar Quantizer (mask) (link)
The Scalar Quantizer block has three modes of operation.
In Encoder mode, the block maps each input value to a quantization region by comparing the input value to the user-specified quantizer boundary points. The block outputs the index of the associated region.
In Decoder mode, the block transforms the input index values into quantized output values, defined by the quantizer codebook.
In the Encoder & Decoder mode, the block performs both the encoding and decoding operations. The block outputs the index values and the quantized output values.
The Boundary points parameter is a vector of dimension N whose values are arranged in ascending order ([p1 p2 ... pN]). The quantizer codebook is a vector of length (N-1). The quantizer is unbounded if p1 is -inf and pN is inf. When the quantizer is bounded, p1 is the lower bound for the input values. Any input less than p1 is clipped to p1 and then quantized. Similarly, pN is the upper bound for the input values. Any input greater than pN is clipped to pN and is then quantized.
Parameters
Quantizer mode:
Encoder
Source of quantizer parameters:
Specify via dialog
Boundary points:
[0 1 2 3 4 5 6 7 8 9]
Searching method:
Binary
Tie-breaking rule:
Choose the lower index
-------------- Show additional parameters --------------
Output index data type:
int32
OK
Cancel
Help
Apply


Block Parameters: Scalar Quantizer
Scalar Quantizer (mask) (link)
The Scalar Quantizer block has three modes of operation.
In Encoder mode, the block maps each input value to a quantization region by comparing the input value to the user-specified quantizer boundary points. The block outputs the index of the associated region.
In Decoder mode, the block transforms the input index values into quantized output values, defined by the quantizer codebook.
In the Encoder & Decoder mode, the block performs both the encoding and decoding operations. The block outputs the index values and the quantized output values.
The Boundary points parameter is a vector of dimension N whose values are arranged in ascending order ([p1 p2 ... pN]). The quantizer codebook is a vector of length (N-1). The quantizer is unbounded if p1 is -inf and pN is inf. When the quantizer is bounded, p1 is the lower bound for the input values. Any input less than p1 is clipped to p1 and then quantized. Similarly, pN is the upper bound for the input values. Any input greater than pN is clipped to pN and is then quantized.
Parameters
Quantizer mode:
Decoder
Source of quantizer parameters:
Specify via dialog
Codebook:
[0.0 1.5 2.5 3.5 4.5 5.5 6.5 7.5 8.5]
Action for out of range input:
Clip and warn
-------------- Show additional parameters --------------
Output data type:
double
OK
Cancel
Help
Apply


Block Parameters: Scalar Quantizer
Scalar Quantizer (mask) (link)
The Scalar Quantizer block has three modes of operation.
In Encoder mode, the block maps each input value to a quantization region by comparing the input value to the user-specified quantizer boundary points. The block outputs the index of the associated region.
In Decoder mode, the block transforms the input index values into quantized output values, defined by the quantizer codebook.
In the Encoder & Decoder mode, the block performs both the encoding and decoding operations. The block outputs the index values and the quantized output values.
The Boundary points parameter is a vector of dimension N whose values are arranged in ascending order ([p1 p2 ... pN]). The quantizer codebook is a vector of length (N-1). The quantizer is unbounded if p1 is -inf and pN is inf. When the quantizer is bounded, p1 is the lower bound for the input values. Any input less than p1 is clipped to p1 and then quantized. Similarly, pN is the upper bound for the input values. Any input greater than pN is clipped to pN and is then quantized.
Parameters
Quantizer mode:
Encoder and decoder
Source of quantizer parameters:
Specify via dialog
Boundary points:
[0 1 2 3 4 5 6 7 8 9]
Codebook:
[0.0 1.5 2.5 3.5 4.5 5.5 6.5 7.5 8.5]
Searching method:
Binary
Tie-breaking rule:
Choose the lower index
Output the quantization error
-------------- Show additional parameters --------------
Output index data type:
int32
OK
Cancel
Help
Apply

Quantizer mode

Specify `Encoder`, `Decoder`, or `Encoder and decoder` as a mode of operation.

Source of quantizer parameters

Choose `Specify via dialog` to type the parameters into the block parameters dialog box. Select `Input ports` to specify the parameters using the block's input ports. In `Encoder` and `Encoder and decoder` mode, input the **Boundary points** using port B. In `Decoder` and `Encoder and decoder` mode, input the **Codebook** values using port C.

Boundary points

Enter a vector of values that represent the boundary points of the quantizer regions. Tunable.

Codebook

Enter a vector of quantized output values that correspond to each index value. Tunable.

Searching method

Select `Linear` and the block finds the region in which the input value is located using a linear search. Select `Binary` and the block finds the region in which the input value is located using a binary search.

Tie-breaking rule

Set this parameter to determine the behavior of the block when the input value is the same as the boundary point. When you select `Choose the lower index`, the input value is assigned to lower index value. When you select `Choose the higher index`, the value is assigned to the higher index.

Action for out of range input

Choose the block's behavior when an input index value is out of range, where $0 \leq \text{index} < N$ and N is the length of the codebook vector. Select `Clip`, when you want any index values less than 0 to be set to 0 and any index values greater than or equal to N to be set to N -1. Select `Clip and warn`, when you want to be warned when any index values less than 0 are set to 0 and any index values greater than or equal to N are set to N -1. Select `Error`, when you want the simulation to stop and display an error when the index values are out of range.

Output the quantization error

In `Encoder and decoder` mode, select this check box to output the quantization error from the Err port on this block.

Output index data type

In `Encoder` and `Encoder and decoder` mode, specify the data type of the index output from the block at port Idx. The data type can be `uint8`, `uint16`, `uint32`, `int8`, `int16`, or `int32`. This parameter becomes visible when you select the **Show additional parameters** check box.

Output data type

In `Decoder` mode, specify the data type of the quantized output. The data type can be `uint8`, `uint16`, `uint32`, `int8`, `int16`, `int32`, `single`, or `double`. This parameter becomes visible when you select `Specify via dialog` for the **Source of quantizer parameters** and you select the **Show additional parameters** check box.

References

Gersho, A. and R. Gray. *Vector Quantization and Signal Compression*. Boston: Kluwer Academic Publishers, 1992.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

For more information on what data types are supported for each quantizer mode, see “Data Type Support” on page 2-1046.

See Also

Quantizer	Simulink
Scalar Quantizer Decoder	Signal Processing Blockset
Scalar Quantizer Design	Signal Processing Blockset
Scalar Quantizer Encoder	Signal Processing Blockset
Uniform Encoder	Signal Processing Blockset
Uniform Decoder	Signal Processing Blockset

Scalar Quantizer Decoder

Purpose

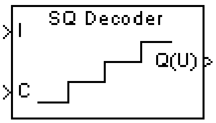
Convert each index value into quantized output value

Library

Quantizers

`dspquant2`

Description



Scalar Quantizer Decoder

The Scalar Quantizer Decoder block transforms the zero-based input index values into quantized output values. The set of all possible quantized output values is defined by the **Codebook values** parameter.

Use the **Codebook values** parameter to specify a matrix containing all possible quantized output values. You can select how you want to enter the codebook values using the **Source of codebook** parameter. When you select `Specify via dialog`, type the codebook values into the block parameters dialog box. When you select `Input port`, port C appears on the block. The block uses the input to port C as the **Codebook values** parameter.

The input to this block is a vector of integer index values, where $0 \leq \text{index} < N$ and N is the number of distinct codeword vectors in the codebook matrix. Use the **Action for out of range index value** parameter to determine what happens when an input index value is outside this range. When you want any index value less than 0 to be set to 0 and any index value greater than or equal to N to be set to N -1, select `Clip`. When you want to be warned when clipping occurs, select `Clip and warn`. When you want the simulation to stop and the block to display an error when the index values are out of range, select `Error`.

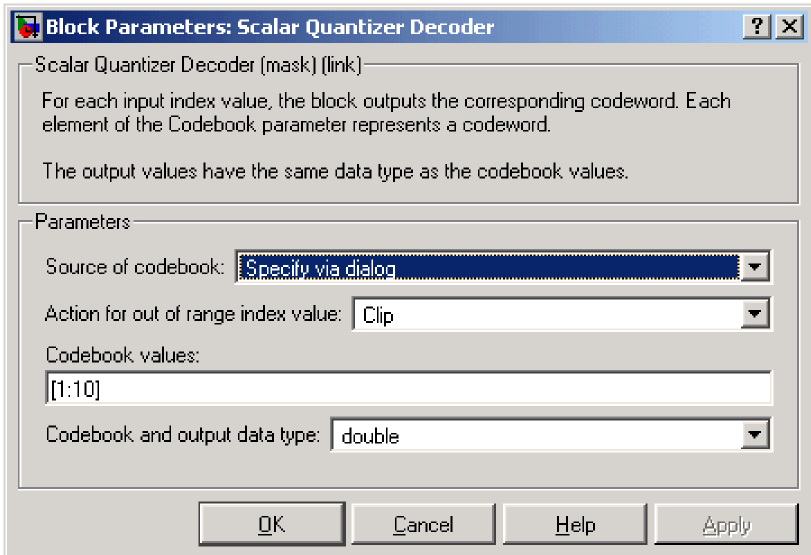
Data Type Support

The data type of the index values input at port I can be `uint8`, `uint16`, `uint32`, `int8`, `int16`, or `int32`. The data type of the codebook values input at port C can be `double`, `single`, or Fixed-point.

The output of the block is the quantized output values. If, for the **Source of codebook** parameter, you select `Specify via dialog`, the **Codebook and output data type** parameter appears. You can use this parameter to specify the data type of the codebook and quantized output values. In this case, the data type of the output values can be `Same as input`, `double`, `single`, `Fixed-point`, or `User-defined`.

If, for the **Source of codebook** parameter you select `Input port`, the quantized output values have the same data type as the codebook values input at port C.

Dialog Box



Source of codebook

Choose `Specify via dialog` to type the codebook values into the block parameters dialog box. Select `Input port` to specify the codebook using input port C.

Action for out of range index value

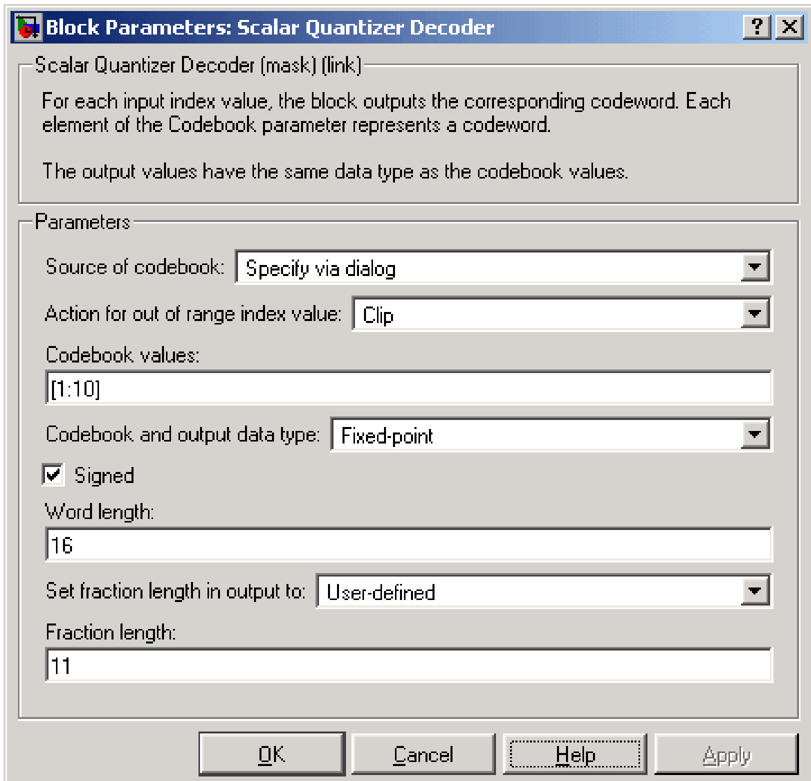
Use this parameter to determine the block's behavior when an input index value is out of range, where $0 \leq$ index $<N$ and N is the length of the codebook vector. Select `Clip`, when you want any index values less than 0 to be set to 0 and any index values greater than or equal to N to be set to N -1. Select `Clip and warn`, when you want to be warned when clipping occurs. Select `Error`, when you want the simulation to stop and the block to display an error when the index values are outside the range.

Codebook values

Enter a vector of quantized output values that correspond to each index value. Tunable.

Codebook and output data type

Use this parameter to specify the data type of the codebook and quantized output values. The data type can be `Same as input`, `double`, `single`, `Fixed-point`, or `User-defined`. This parameter becomes visible when you select `Specify via dialog` for the **Source of codebook** parameter.



Signed

Select to output a signed fixed-point signal. Otherwise, the signal is unsigned. This parameter is only visible if, from the **Codebook and output data type** list, you select `Fixed-point`.

Word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible if, from the **Codebook and output data type** list, you select `Fixed-point`.

Set fraction length in output to

Specify the scaling of the fixed-point output by either of the following two methods:

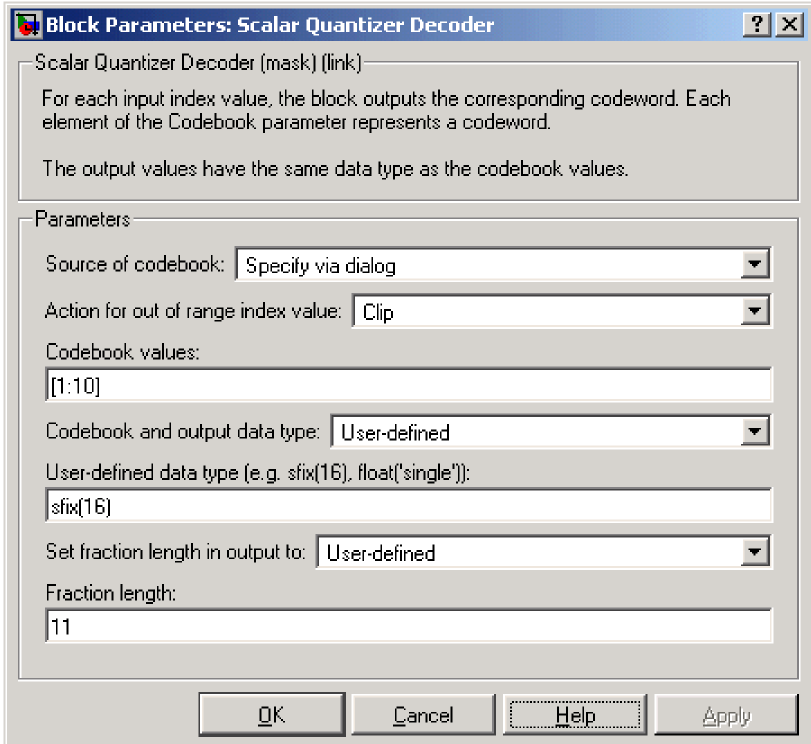
Choose `Best precision` to have the output scaling automatically set such that the output signal has the best possible precision.

Choose `User-defined` to specify the output scaling in the **Fraction length** parameter.

This parameter is only visible if, from the **Codebook and output data type** list, you select `Fixed-point` or when you select `User-defined` and the specified output data type is a fixed-point data type.

Fraction length

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



User-defined data type

Specify any built-in or fixed-point data type. You can specify fixed-point data types using the following Simulink® Fixed Point™ functions: `sfix`, `ufix`, `sint`, `uint`, `sfrac`, and `ufrac`. This parameter is only visible when you select `User-defined` for the **Codebook and output data type** parameter.

References

Gersho, A. and R. Gray. *Vector Quantization and Signal Compression*. Boston: Kluwer Academic Publishers, 1992.

Supported Data Types

Port	Supported Data Types
I	• 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
C	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers
Q(U)	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers

For more information on what data types are supported for each quantizer mode, see "Data Type Support" on page 2-1054.

See Also

Quantizer	Simulink
Scalar Quantizer Design	Signal Processing Blockset
Scalar Quantizer Encoder	Signal Processing Blockset
Uniform Encoder	Signal Processing Blockset
Uniform Decoder	Signal Processing Blockset

Purpose

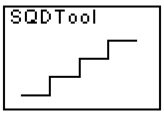
Start Scalar Quantizer Design Tool (SQDTool) to design scalar quantizer using Lloyd algorithm

Library

Quantizers

`dspquant2`

Description



Scalar Quantizer Design

Double-click on the Scalar Quantizer Design block to start SQDTool, a GUI that allows you to design and implement a scalar quantizer. Based on your input values, SQDTool iteratively calculates the codebook values that minimize the mean squared error until the stopping criteria for the design process is satisfied. The block uses the resulting quantizer codebook values and boundary points to implement your scalar quantizer encoder and/or decoder.

For the **Training Set** parameter, enter a set of observations, or samples, of the signal you want to quantize. This data can be any variable defined in the MATLAB® workspace including a variable created using a MATLAB function, such as the default value `randn(10000,1)`.

You have two choices for the **Source of initial codebook** parameter. Select `Auto-generate` to have the block choose the values of the initial codebook vector. In this case, the minimum training set value becomes the first codeword, and the maximum training set value becomes the last codeword. Then, the remaining initial codewords are equally spaced between these two values to form a codebook vector of length N, where N is the **Number of levels** parameter. When you select `User defined`, enter the initial codebook values in the **Initial codebook** field. Then, set the **Source of initial boundary points** parameter. You can select `Mid-points` to locate the boundary points at the midpoint between the codewords. To calculate the mid-points, the block internally arranges the initial codebook values in ascending order. You can also choose `User defined` and enter your own boundary points in the **Initial boundary points (unbounded)** field. Only one boundary point can be located between two codewords. When you select `User defined` for the **Source of initial boundary points** parameter, the values you enter in the **Initial codebook** and **Initial boundary points (unbounded)** fields must be arranged in ascending order.

Note This block assumes that you are designing an unbounded quantizer. Therefore, the first and last boundary points are always `-inf` and `inf` regardless of any other boundary point values you might enter.

After you have specified the quantization parameters, the block performs an iterative process to design the optimal scalar quantizer. Each step of the design process involves using the Lloyd algorithm to calculate codebook values and quantizer boundary points. Then, the block calculates the squared quantization error and checks whether the stopping criteria has been satisfied.

The two possible options for the **Stopping criteria** parameter are `Relative threshold` and `Maximum iteration`. When you want the design process to stop when the fractional drop in the squared quantization error is below a certain value, select `Relative threshold`. Then, for **Relative threshold**, type the maximum acceptable fractional drop. When you want the design process to stop after a certain number of iterations, choose `Maximum iteration`. Then, enter the maximum number of iterations you want the block to perform in the **Maximum iteration** field. For **Stopping criteria**, you can also choose `Whichever comes first` and enter a **Relative threshold** and **Maximum iteration** value. The block stops iterating as soon as one of these conditions is satisfied.

With each iteration, the block quantizes the training set values based on the newly calculated codebook values and boundary points. When the training point lies on a boundary point, the algorithm uses the **Tie-breaking rules** parameter to determine which region the value is associated with. When you want the training point to be assigned to the lower indexed region, select `Lower indexed codeword`. To assign the training point with the higher indexed region, select `Higher indexed codeword`.

The **Searching methods** parameter determines how the block compares the training points to the boundary points. Select `Linear search` and SQDTool compares each training point to each quantization region sequentially. This process continues until all the training points are associated with the appropriate regions.

Select `Binary search` for the **Searching methods** parameter and the block compares the training point to the middle value of the boundary points vector. When the training point is larger than this boundary point, the block discards the lower boundary points. The block then compares the training point to the middle boundary point of the new range, defined by the remaining boundary points. This process continues until the training point is associated with the appropriate region.

Click **Design and Plot** to design the quantizer with the parameter values specified on the left side of the GUI. The performance curve and the staircase character of the quantizer are updated and displayed in the figures on the right side of the GUI.

Note You must click **Design and Plot** to apply any changes you make to the parameter values in the SQDTool dialog box.

SQDTool can export parameter values that correspond to the figures displayed in the GUI. Click the **Export Outputs** button, or press **Ctrl+E**, to export the **Final Codebook**, **Final Boundary Points**, and **Error** values to the workspace, a text file, or a MAT-file. The **Error** values represent the mean squared error for each iteration.

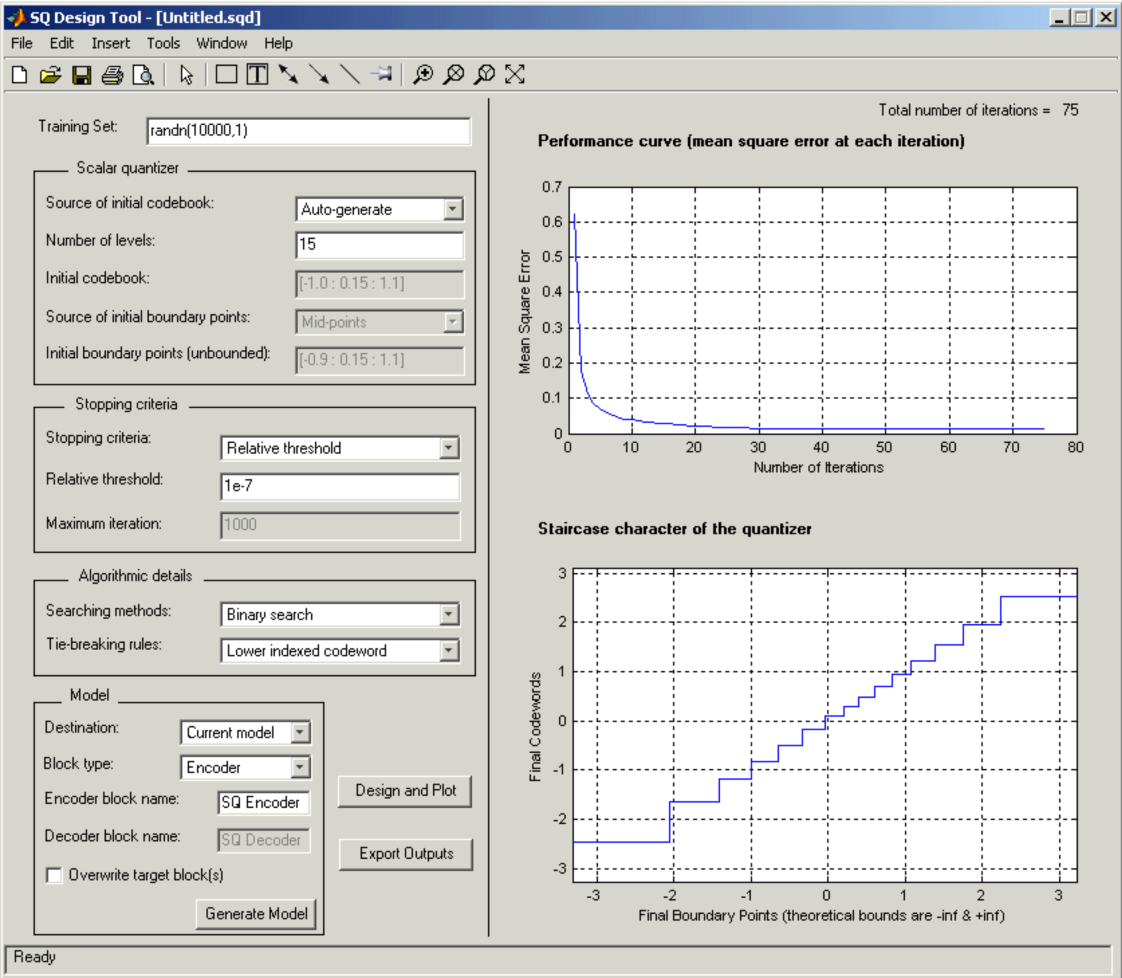
In the **Model** section of the GUI, specify the destination of the block that will contain the parameters of your quantizer. For **Destination**, select `Current model` to create a block with your parameters in the model you most recently selected. Type `gcs` in the MATLAB Command Window to display the name of your current model. Select `New model` to create a block in a new model file.

From the **Block type** list, select `Encoder` to design a Scalar Quantizer Encoder block. Select `Decoder` to design a Scalar Quantizer Decoder block. Select `Both` to design a Scalar Quantizer Encoder block and a Scalar Quantizer Decoder block.

In the **Encoder block name** field, enter a name for the Scalar Quantizer Encoder block. In the **Decoder block name** field, enter a name for the Scalar Quantizer Decoder block. When you have a Scalar Quantizer Encoder and/or Decoder block in your destination model with the same name, select the **Overwrite target block(s)** check box to replace the block's parameters with the current parameters. When you do not select this check box, a new Scalar Quantizer Encoder and/or Decoder block is created in your destination model.

Click **Generate Model**. SQDTool uses the parameters that correspond to the current plots to set the parameters of the Scalar Quantizer Encoder and/or Decoder blocks.

Dialog Box


SQ Design Tool - [Untitled.sqd]
File Edit Insert Tools Window Help
Training Set: randn(10000,1)
Scalar quantizer
Source of initial codebook: Auto-generate
Number of levels: 15
Initial codebook: [-1.0 : 0.15 : 1.1]
Source of initial boundary points: Mid-points
Initial boundary points (unbounded): [-0.9 : 0.15 : 1.1]
Stopping criteria
Stopping criteria: Relative threshold
Relative threshold: 1e-7
Maximum iteration: 1000
Algorithmic details
Searching methods: Binary search
Tie-breaking rules: Lower indexed codeword
Model
Destination: Current model
Block type: Encoder
Encoder block name: SQ Encoder
Decoder block name: SQ Decoder
Overwrite target block(s)
Generate Model
Design and Plot
Export Outputs
Total number of iterations = 75
Performance curve (mean square error at each iteration)
Mean Square Error
Number of Iterations
Staircase character of the quantizer
Final Codewords
Final Boundary Points (theoretical bounds are -inf & +inf)
Ready

Training Set

Enter the samples of the signal you would like to quantize. This data set can be a MATLAB function or a variable defined in the MATLAB workspace. The typical length of this data vector is `1e6`.

Source of initial codebook

Select `Auto-generate` to have the block choose the initial codebook values. Select `User defined` to enter your own initial codebook values.

Number of levels

Enter the length of the codebook vector. For a b-bit quantizer, the length should be $N = 2^b$.

Initial codebook

Enter your initial codebook values. From the **Source of initial codebook** list, select `User defined` in order to activate this parameter.

Source of initial boundary points

Select `Mid-points` to locate the boundary points at the midpoint between the codebook values. Choose `User defined` to enter your own boundary points. From the **Source of initial codebook** list, select `User defined` in order to activate this parameter.

Initial boundary points (unbounded)

Enter your initial boundary points. This block assumes that you are designing an unbounded quantizer. Therefore, the first and last boundary point are `-inf` and `inf`, regardless of any other boundary point values you might enter. From the **Source of initial boundary points** list, select `User defined` in order to activate this parameter.

Stopping criteria

Choose `Relative threshold` to enter the maximum acceptable fractional drop in the squared quantization error. Choose `Maximum iteration` to specify the number of iterations at which to stop. Choose `Whichever comes first` and the block stops the iteration process as soon as the relative threshold or maximum iteration value is attained.

Relative threshold
Type the value that is the maximum acceptable fractional drop in the squared quantization error.

Maximum iteration
Enter the maximum number of iterations you want the block to perform. From the **Stopping criteria** list, select `Maximum iteration` in order to activate this parameter.

Searching methods
Choose `Linear search` to use a linear search method when comparing the training points to the boundary points. Choose `Binary search` to use a binary search method when comparing the training points to the boundary points.

Tie-breaking rules
When a training point lies on a boundary point, choose `Lower indexed codeword` to assign the training point to the lower indexed quantization region. Choose `Higher indexed codeword` to assign the training point to the higher indexed region.

Design and Plot
Click this button to display the performance curve and the staircase character of the quantizer in the figures on the right side of the GUI. These plots are based on the current parameter settings.

You must click **Design and Plot** to apply any changes you make to the parameter values in the SQDTool GUI.

Export Outputs
Click this button, or press **Ctrl+E**, to export the **Final Codebook**, **Final Boundary Points**, and **Error** values to the workspace, a text file, or a MAT-file.

Destination
Choose `Current model` to create a Scalar Quantizer block in the model you most recently selected. Type `gcs` in the MATLAB Command Window to display the name of your current model. Choose `New model` to create a block in a new model file.

Block type
Select `Encoder` to design a Scalar Quantizer Encoder block. Select `Decoder` to design a Scalar Quantizer Decoder block. Select `Both` to design a Scalar Quantizer Encoder block and a Scalar Quantizer Decoder block.

Encoder block name
Enter a name for the Scalar Quantizer Encoder block.

Decoder block name
Enter a name for the Scalar Quantizer Decoder block.

Overwrite target block(s)
When you do not select this check box and a Scalar Quantizer Encoder and/or Decoder block with the same block name exists in the destination model, a new Scalar Quantizer Encoder and/or Decoder block is created in the destination model. When you select this check box and a Scalar Quantizer Encoder and/or Decoder block with the same block name exists in the destination model, the parameters of these blocks are overwritten by new parameters.

Generate Model
Click this button and SQDTool uses the parameters that correspond to the current plots to set the parameters of the Scalar Quantizer Encoder and/or Decoder blocks.

References

Gersho, A. and R. Gray. *Vector Quantization and Signal Compression*. Boston: Kluwer Academic Publishers, 1992.

Supported Data Types

- Double-precision floating point

See Also

Quantizer	Simulink
Scalar Quantizer Decoder	Signal Processing Blockset

Scalar Quantizer Design

Scalar Quantizer Encoder	Signal Processing Blockset
Uniform Encoder	Signal Processing Blockset
Uniform Decoder	Signal Processing Blockset

Purpose

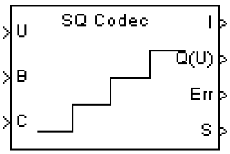
Encode each input value by associating it with index value of quantization region

Library

Quantizers

`dspquant2`

Description



Scalar Quantizer Encoder

The Scalar Quantizer Encoder block maps each input value to a quantization region by comparing the input value to the quantizer boundary points defined in the **Boundary points** parameter. The block outputs the zero-based index of the associated region.

You can select how you want to enter the **Boundary points** using the **Source of quantizer parameters**. When you select `Specify via dialog`, type the boundary points into the block parameters dialog box. When you select `Input port`, port B appears on the block. The block uses the input to port B as the **Boundary points** parameter.

Use the **Boundary points** parameter to specify the boundary points for your quantizer. These values are used to break up the set of input numbers into regions. Each region is specified by an index number.

Let `N` be the number of quantization regions. When the codebook is defined as `[c1 c2 c3 ...  cN]`, and the **Boundary points** parameter is defined as `[p0 p1 p2 p3 ...  pN]`, then `p0<c1<p1<c2 ...  p(N-1)<cN<pN` for a regular quantizer. When your quantizer is bounded, from the **Partitioning** list, select `Bounded`. You need to specify N+1 boundary points, or `[p0 p1 p2 p3 ...  pN]`. When your quantizer is unbounded, from the **Partitioning** list, select `Unbounded`. You need to specify N-1 boundary points, or`[p1 p2 p3 ...  p(N-1)]`; the block sets `p0` equal to `-inf` and `pN` equal to `inf`.

The block uses the **Partitioning** parameter to interpret the boundary points you enter. For instance, to create a bounded quantizer, from the **Partitioning** list, select `Bounded` and enter the following boundary points:

```
[0 0.5 3.7 5.8 6.0 11]
```

The block assigns any input values between 0 and 0.5 to index 0, input values between 0.5 and 3.7 to index 1, and so on. The block assigns any values that are less than 0 to index 0, the lowest index value. The block assigns any values that are greater than 11 to index 4, the highest index value.

To create an unbounded quantizer, from the **Partitioning** list, select `Unbounded` and enter the following boundary points:

```
[0 0.5 3.7 5.8 6.0 11]
```

The block assigns any input values between 0 and 0.5 to index 1, input values between 0.5 and 3.7 to index 2, and so on. The block assigns any input values less than 0 to index 0 and any values greater than 11 to index 6.

The **Searching method** parameter determines how the appropriate quantizer index is found. When you select `Linear`, the Scalar Quantizer Encoder block compares the input value to the first region defined by the first two boundary points. When the input value does not fall within this region, the block then compares the input value to the next region. This process continues until the input value is determined to be within a region and is associated with the appropriate index value. The computational cost of this process is of the order P, where P is the number of boundary points.

When you select `Binary` for the **Searching method**, the block compares the input value to the middle value of the boundary points vector. When the input value is larger than this boundary point, the block discards the boundary points that are lower than this middle value. The block then compares the input value to the middle boundary point of the new range, defined by the remaining boundary points. This process continues until the input value is associated with the appropriate index value. The computational cost of this process is of the order $\log_2 P$, where P is the number of boundary points. In most cases, the `Binary` option is faster than the `Linear` option.

When an input value is the same as a boundary point, the **Tie-breaking rule** parameter determines the region to which the value is assigned.

When you want the input value to be assigned to the lower indexed region, select `Choose the lower index`. To assign the input value with the higher indexed region, select `Choose the higher index`.

Select the **Output codeword** check box to output the codeword values that correspond to each index value at port Q(U).

Select the **Output the quantization error** check box to output the quantization error for each input value from the Err port on this block. The quantization error is the difference between the input value and the quantized output value.

When you select either the **Output codeword** check box or the **Output quantization error** check box, you must also enter your codebook values. If, from the **Source of quantizer parameters** list, you choose `Specify via dialog`, use the **Codebook** parameter to enter a vector of quantized output values that correspond to each region. If, from the **Source of quantizer parameters** list, you choose `Input port`, use input port C to specify your codebook values.

If, for the **Partitioning** parameter, you select `Bounded`, the **Output clipping status** check box and the **Action for out of range input** parameter appear. When you select the **Output clipping status** check box, port S appears on the block. Any time an input value is outside the range defined by the **Boundary points** parameter, the block outputs a 1 at the S port. When the value is inside the range, the blocks outputs a 0.

You can use the **Action for out of range input** parameter to determine the block's behavior when an input value is outside the range defined by the **Boundary points** parameter. Suppose the boundary points for a bounded quantizer are defined as `[p0 p1 p2 p3 ...  pN]` and the possible index values are defined as `[i0 i1 i2 ...  i(N-1)]`, where `i0=0` and `i0<i1<i2<...<i(N-1)`. When you want any input value less than `p0` to be assigned to index value `i0` and any input values greater than `pN` to be assigned to index value `i(N-1)`, select `Clip`. When you want to be warned when clipping occurs, select `Clip and warn`. When you want the simulation to stop and the block to display an error when the index values are out of range, select `Error`.

The Scalar Quantizer Encoder block accepts real floating-point and fixed-point inputs. For more information on the data types accepted by each port, see “Data Type Support” on page 2-1072 or “Supported Data Types” on page 2-1077.

Data Type Support

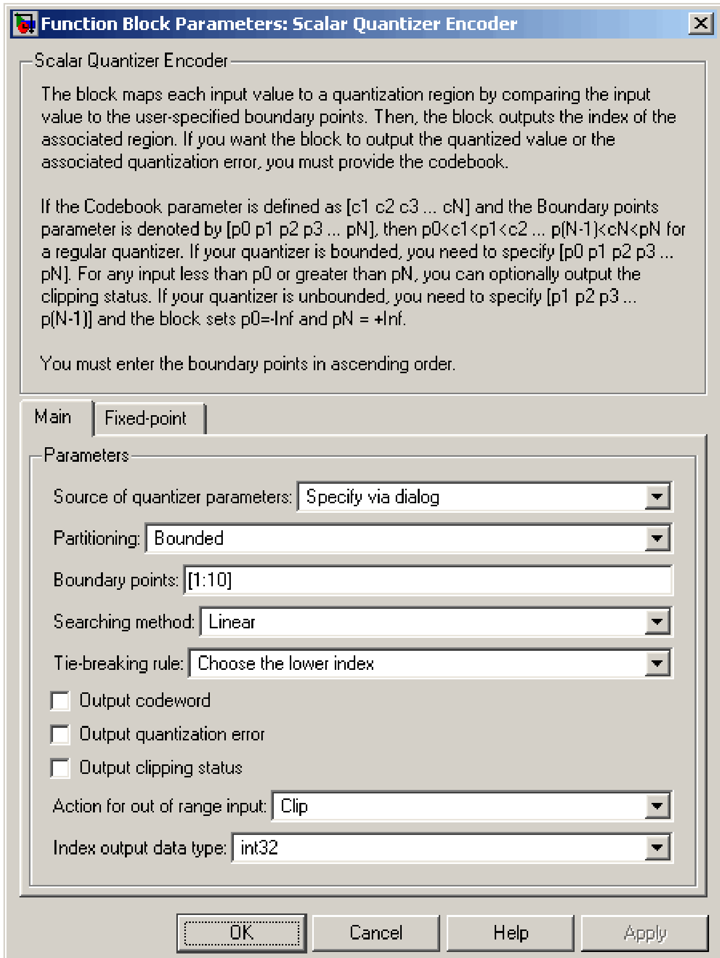
The input data values, boundary points, and codebook values can be input to the block at ports U, B, and C, respectively. The data type of the inputs can be `double`, `single`, or Fixed-point.

The outputs of the block can be the index values, the quantized output values, the quantization error, and the clipping status. Use the **Index output data type** parameter to specify the data type of the index output from the block at port I. You can choose `int8`, `uint8`, `int16`, `uint16`, `int32`, or `uint32`. The data type of the quantized output and the quantization error can be `double`, `single`, or Fixed-point. The clipping status values output at port S are Boolean values.

Note The input data, boundary points, codebook values, quantized output values, and the quantization error must have the same data type whenever they are present.

Dialog Box

The **Main** pane of the Scalar Quantizer Encoder block dialog appears as follows.



Source of quantizer parameters

Choose `Specify via dialog` to enter the boundary points and codebook values using the block parameters dialog box. Select

`Input port` to specify the parameters using the block's input ports. Input the boundary points and codebook values using ports B and C, respectively.

Partitioning
When your quantizer is bounded, select `Bounded`. When your quantizer is unbounded, select `Unbounded`.

Boundary points
Enter a vector of values that represent the boundary points of the quantizer regions. This parameter is visible when you select `Specify via dialog` from the **Source of quantizer parameters** list. Tunable.

Searching method
When you select `Linear`, the block finds the region in which the input value is located using a linear search. When you select `Binary`, the block finds the region in which the input value is located using a binary search.

Tie-breaking rule
Set this parameter to determine the behavior of the block when the input value is the same as the boundary point. When you select `Choose the lower index`, the input value is assigned to lower indexed region. When you select `Choose the higher index`, the value is assigned to the higher indexed region.

Output codeword
Select this check box to output the codeword values that correspond to each index value at port Q(U).

Output quantization error
Select this check box to output the quantization error for each input value at port Err.

Codebook
Enter a vector of quantized output values that correspond to each index value. If, for the **Partitioning** parameter, you select `Bounded` and your boundary points vector has length N, then you must specify a codebook of length N-1. If, for the **Partitioning**

parameter, you select `Unbounded` and your boundary points vector has length N, then you must specify a codebook of length N+1.

This parameter is visible when you select `Specify via dialog` from the **Source of quantizer parameters** list and you select either the **Output codeword** or **Output quantization error** check box. Tunable.

Output clipping status

When you select this check box, port S appears on the block. Any time an input value is outside the range defined by the **Boundary points** parameter, the block outputs a 1 at this port. When the value is inside the range, the block outputs a 0. This parameter is visible when you select `Bounded` from the **Partitioning** list.

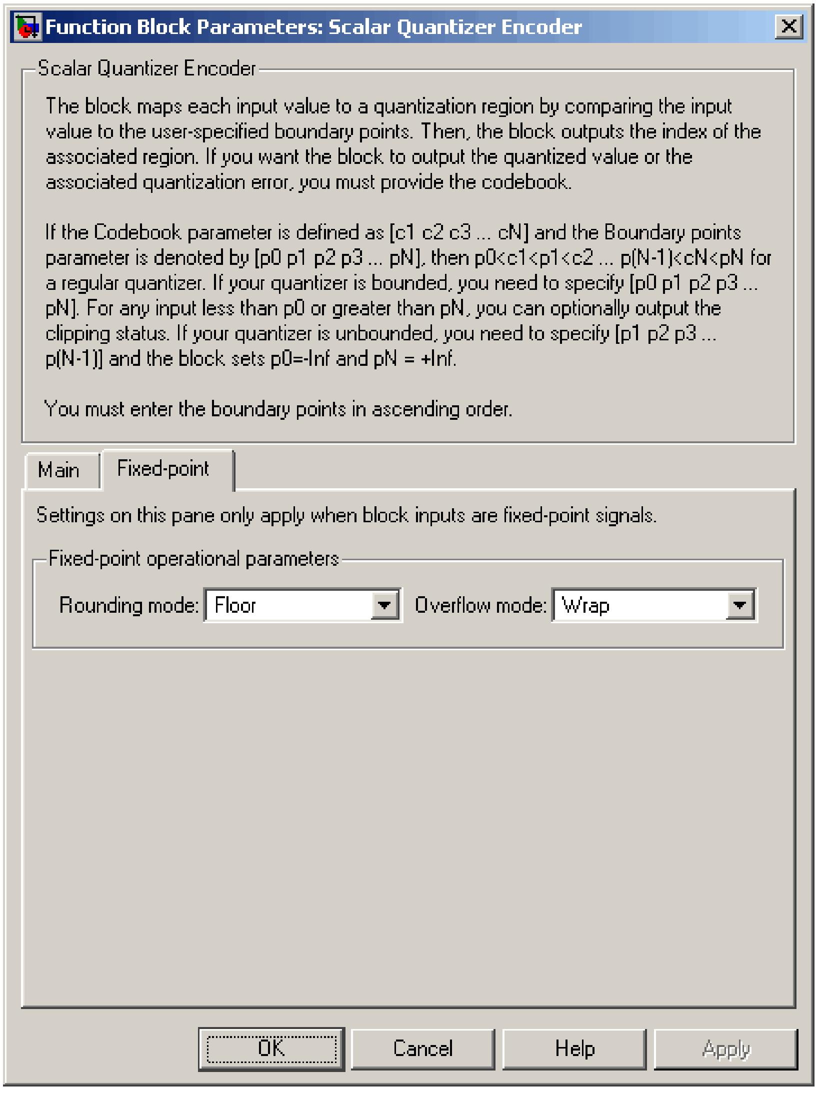
Action for out of range input

Use this parameter to determine the behavior of the block when an input value is outside the range defined by the **Boundary points** parameter. Suppose the boundary points are defined as `[p0 p1 p2 p3 ...  pN]` and the index values are defined as `[i0 i1 i2 ...  i(N-1)]`. When you want any input value less than `p0` to be assigned to index value `i0` and any input values greater than `pN` to be assigned to index value `i(N-1)`, select `Clip`. When you want to be warned when clipping occurs, select `Clip and warn`. When you want the simulation to stop and the block to display an error when the index values are out of range, select `Error`. This parameter is visible when you select `Bounded` from the **Partitioning** list.

Index output data type

Specify the data type of the index output from the block at port I. You can choose `int8`, `uint8`, `int16`, `uint16`, `int32`, or `uint32`.

The **Fixed-point** pane of the Scalar Quantizer Encoder block dialog appears as follows.



Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode to be used when block inputs are fixed point.

References

Gersho, A. and R. Gray. *Vector Quantization and Signal Compression*. Boston: Kluwer Academic Publishers, 1992.

Supported Data Types

Port	Supported Data Types
U	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
B	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
C	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
I	• 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers

Port	Supported Data Types
Q(U)	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
Err	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
S	• Boolean

For more information on what data types are supported for each quantizer mode, see "Data Type Support" on page 2-1054.

See Also

Quantizer	Simulink
Scalar Quantizer Decoder	Signal Processing Blockset
Scalar Quantizer Design	Signal Processing Blockset
Uniform Encoder	Signal Processing Blockset
Uniform Decoder	Signal Processing Blockset

Purpose Select input elements from vector, matrix, or multidimensional signal

Library Signal Management / Indexing

`dspindex`

Description Refer to the Simulink® Selector reference page for more information.

Purpose

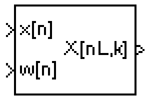
Compute nonparametric estimate of spectrum using short-time, fast Fourier transform (FFT) method

Library

Transforms

`dspxfrm3`

Description



Short-Time FFT

The Short-Time FFT block computes a nonparametric estimate of the spectrum. The block buffers, applies a window, and zero pads the input signal. Then, the block takes the FFT of the signal, transforming it into the frequency domain.

Connect your sample-based or frame-based, single-channel analysis window to the w(n) port. For the **Analysis window length** parameter, enter the length of the analysis window, W. When your analysis window is a sample-based signal, the block buffers it into a frame-based signal with frame length W. When your analysis window is a frame-based signal and its frame length is not W, the block buffers the signal so that its frame length is W.

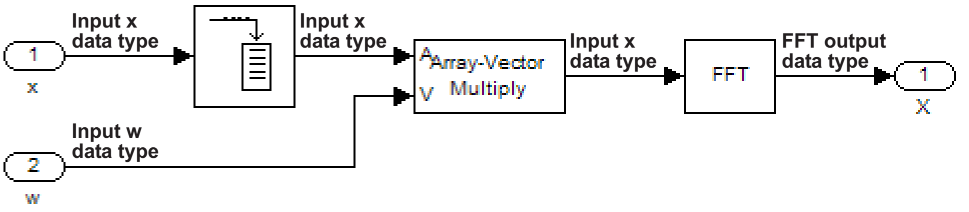
Connect your sample-based or frame-based, single-channel or multichannel input signal to the x(n) port. After the block buffers and windows this signal, it zero-pads the signal before computing the FFT. For the **FFT length** parameter, enter the length to which the block pads the input signal. For the **Overlap between consecutive windows (in samples)** parameter, enter the number of samples to overlap each frame of the input signal.

The complex-valued, sample-based, single-channel or multichannel short-time FFT is output at port X(n,k).

The Short-Time FFT block supports real and complex floating-point and fixed-point signals.

Fixed-Point Data Types

The following diagram shows the data types used within the Short-Time FFT subsystem block for fixed-point signals.



The settings for the fixed-point parameters of the Array-Vector Multiply block in the diagram above are as follows:

- **Rounding Mode** — `Floor`
- **Overflow Mode** — `Wrap`
- **Product output** — `Inherit via internal rule`
- **Accumulator** — `Inherit via internal rule`
- **Output** — `Same as first input`

The settings for the fixed-point parameters of the FFT block in the diagram above are as follows:

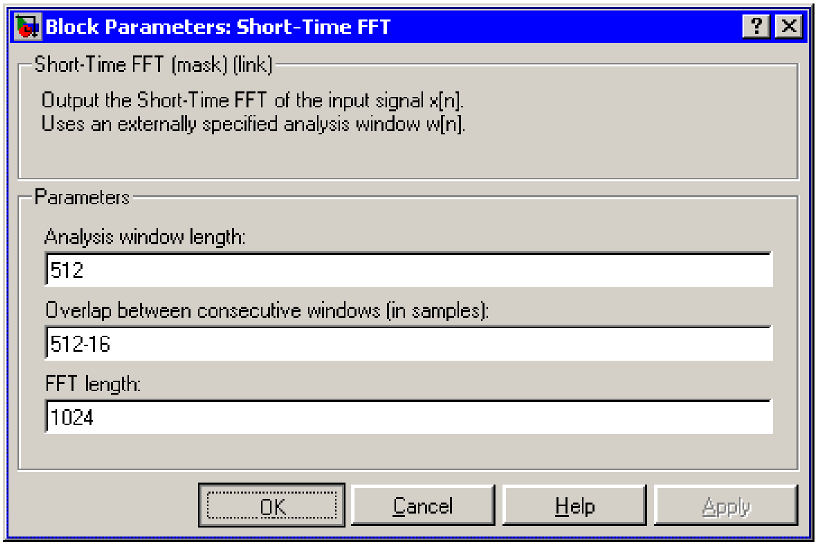
- **Rounding Mode** — `Floor`
- **Overflow Mode** — `Wrap`
- **Sine table** — `Same word length as input`
- **Product output** — `Inherit via internal rule`
- **Accumulator** — `Inherit via internal rule`
- **Output** — `Inherit via internal rule`

See the FFT and Array-Vector Multiply block reference pages for more information.

Examples

The dspstsa_win32 demo illustrates how to use the Short-Time FFT and Inverse Short-Time FFT blocks to remove the background noise from a speech signal.

Dialog Box



Analysis window length
Enter the frame length of the analysis window.

Overlap between consecutive windows (in samples)
Enter the number of samples of overlap for each frame of the input signal.

FFT length
Enter the length to which the block pads the input signal.

References

Quatieri, Thomas E. *Discrete-Time Speech Signal Processing.* Englewood Cliffs, NJ: Prentice-Hall, 2001.

Supported Data Types

Port	Supported Data Types
x(n)	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
w(n)	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
X(n,k)	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers

See Also

Burg Method	Signal Processing Blockset
Inverse Short-Time FFT	Signal Processing Blockset
Magnitude FFT	Signal Processing Blockset
Periodogram	Signal Processing Blockset
Spectrum Scope	Signal Processing Blockset
Window Function	Signal Processing Blockset
Yule-Walker Method	Signal Processing Blockset
`pwelch`	Signal Processing Toolbox

See "Power Spectrum Estimation" for related information.

Purpose

Import signal from MATLAB® workspace

Library

Signal Processing Sources

`dspsrcs4`

Description

The Signal From Workspace block imports a signal from the MATLAB workspace into the Simulink® model. The **Signal** parameter specifies the name of a MATLAB workspace variable containing the signal to import, or any valid MATLAB expression defining a matrix or 3-D array.

When the **Signal** parameter specifies an M-by-N matrix ($M \neq 1$), each of the N columns is treated as a distinct channel. You specify the frame size in the **Samples per frame** parameter, M_o, and the output is an M_o-by-N matrix containing M_o consecutive samples from each signal channel. You specify the output sample period in the **Sample time** parameter, T_s, and the output frame period is M_o*T_s. For $M_o=1$, the output is sample based; otherwise the output is frame based. For convenience, an imported row vector (M=1) is treated as a single channel, so the output dimension is M_o-by-1.

When the **Signal** parameter specifies an M-by-N-by-P array, each of the P pages (an M-by-N matrix) is output in sequence with period T_s. The **Samples per frame** parameter must be set to `1`, and the output is always sample based.

Initial and Final Conditions

Unlike the Simulink From Workspace block, the Signal From Workspace block holds the output value constant between successive output frames (that is, no linear interpolation takes place). Additionally, the initial signal values are always produced immediately at t=0.

When the block has output all of the available signal samples, it can start again at the beginning of the signal, or simply repeat the final value or generate zeros until the end of the simulation. (The block does not extrapolate the imported signal beyond the last sample.) The **Form output after final data value by** parameter controls this behavior:

- When you specify `Setting To Zero`, the block generates zero-valued outputs for the duration of the simulation after generating the last frame of the signal.
- When you specify `Holding Final Value`, the block repeats the final sample for the duration of the simulation after generating the last frame of the signal.
- When you specify `Cyclic Repetition`, the block repeats the signal from the beginning after it reaches the last sample in the signal. If the frame size you specify in the **Samples per frame** parameter does not evenly divide the input length, a buffer block is inserted into the Signal From Workspace subsystem, and the model becomes multirate. If you do not want your model to become multirate, make sure the frame size evenly divides the input signal length.

  Select the **Warn when frame size does not evenly divide input length** parameter to be alerted when the input length is not an integer multiple of the frame size and your model will become multirate. Use the Model Explorer to turn these warnings on or off model-wide:

  a Select **Model Explorer** from the **View** menu in your model.

  b In the **Search** bar of the Model Explorer, search `by Property Name` for the `ignoreOrWarnInputAndFrameLengths` property. Each block with the **Warn when frame size does not evenly divide input length** check box appears in the list in the **Contents** pane.

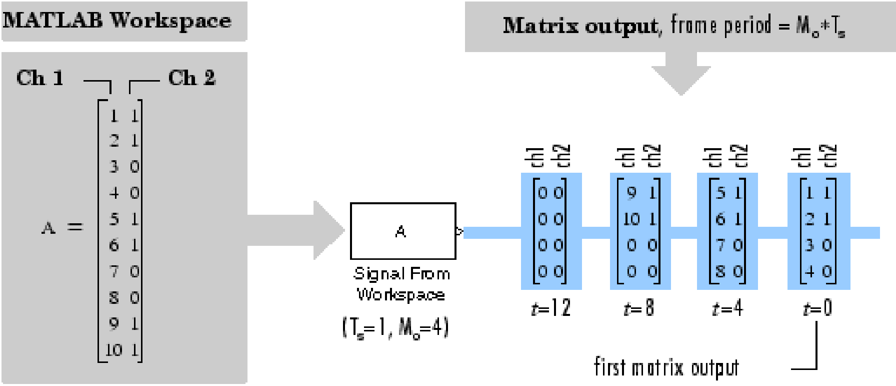
  c Select each of the blocks for which you wish to toggle the warning parameter, and select or deselect the check box in the `ignoreOrWarnInputAndFrameLengths` column.

Examples

Example 1

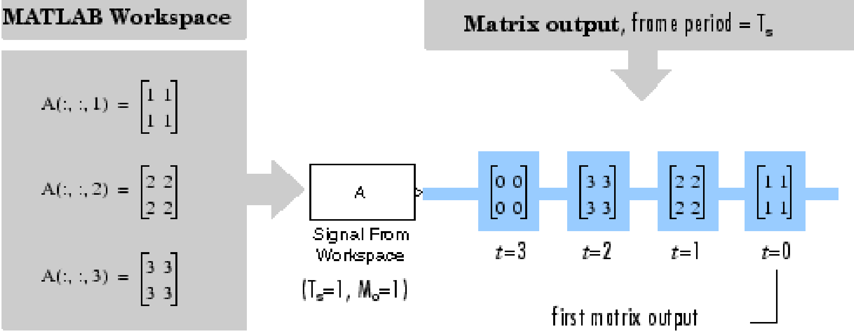
In the first model below, the Signal From Workspace imports a two-channel signal from the workspace matrix `A`. The **Sample time** is set to `1` and the **Samples per frame** is set to `4`, so the output is frame based with a frame size of 4 and a frame period of 4 seconds. The **Form**

output after final data value by parameter specifies `Setting To Zero`, so all outputs after the third frame (at t=8) are zero.



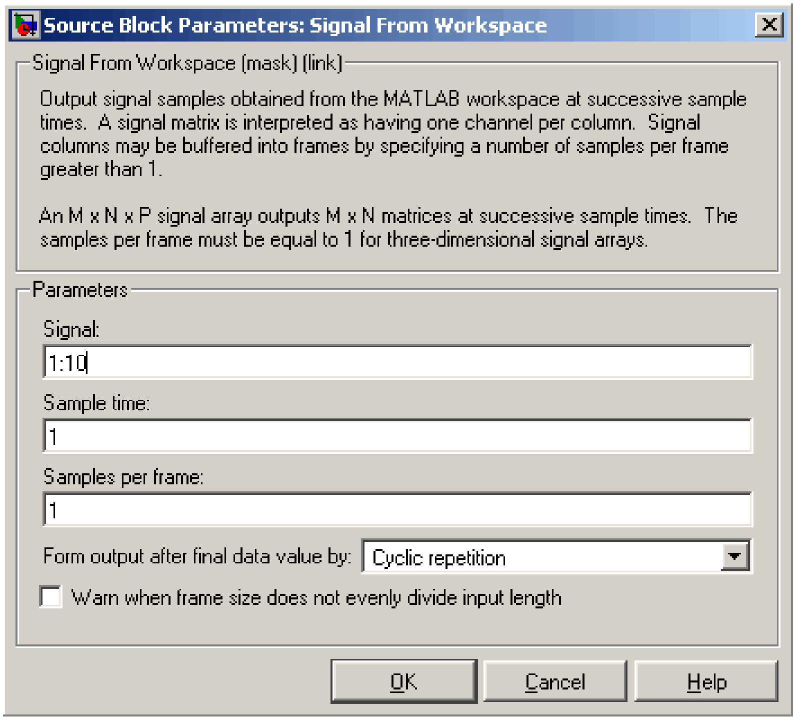
Example 2

In the second model below, the Signal From Workspace block imports a sample-based matrix signal from the 3-D workspace array `A`. Again, the **Form output after final data value by** parameter specifies `Setting To Zero`, so all outputs after the third (at t=2) are zero.



The **Samples per frame** parameter is set to `1` for 3-D input.

Dialog Box



Signal

The name of the MATLAB workspace variable from which to import the signal, or a valid MATLAB expression specifying the signal.

Sample time

The sample period, T_s, of the output. The output frame period is M_o*T_s.

Samples per frame

The number of samples, M_o, to buffer into each output frame. This value must be 1 when you specify a 3-D array in the **Signal** parameter.

Form output after final data value by
: Specifies the output after all of the specified signal samples have been generated. The block can output zeros for the duration of the simulation (`Setting to zero`), repeat the final data sample (`Holding Final Value`) or repeat the entire signal from the beginning (`Cyclic Repetition`).

Warn when frame size does not evenly divide input length
: Select this parameter to be alerted when the input length is not an integer multiple of the frame size and your model will become multirate. For more information, see "Initial and Final Conditions" on page 2-1085.

This parameter is only visible when `Cyclic Repetition` is selected for the **Form output after final data value by** parameter.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

From Audio Device	Signal Processing Blockset
From Wave File	Signal Processing Blockset
Signal From Workspace	Signal Processing Blockset
From Workspace	Simulink
To Workspace	Simulink
Triggered Signal From Workspace	Signal Processing Blockset

See the sections below for related information:

- "Creating Sample-Based Signals"
- "Creating Frame-Based Signals"
- "Importing and Exporting Sample-Based Signals"
- "Importing and Exporting Frame-Based Signals"

Purpose

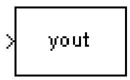
Write simulation data to array in MATLAB® workspace

Library

Signal Processing Sinks

`dspsnks4`

Description



The Signal To Workspace block writes data from your simulation into an array in the MATLAB main workspace. The output array can be 2-D or 3-D, depending on whether the data is 1-D, sample based, or frame based. The Signal To Workspace block and the Simulink® To Workspace block can output the same arrays when their parameters are set appropriately.

For more information on the Signal To Workspace block, see the following sections of this reference page:

- “Parameter Descriptions” on page 2-1091
- “Output Dimension Summary” on page 2-1093
- “Matching the Outputs of Signal To Workspace and To Workspace Blocks” on page 2-1093
- “Examples” on page 2-1094

Parameter Descriptions

The **Variable name** parameter is the name of the array in the MATLAB workspace into which the block logs the simulation data. The array is created in the workspace only after the simulation stops running. When you enter the name of an existing workspace variable, the block overwrites the variable with an array of simulation data after the simulation stops running.

When the block input is sample based or 1-D, the **Limit data points to last** parameter indicates how many *samples of data* to save. When the block input is frame based, this parameter indicates how many *frames of data* to save. When the simulation generates more than the specified maximum number of samples or frames, the simulation saves only the most recently generated data. To capture all data, set **Limit data points to last** to `inf`.

The **Decimation** parameter is the decimation factor. It can be set to any positive integer *d*, and allows you to write data at every *d*th sample. The default decimation, `1`, writes data at every time step.

The **Frames** parameter sets the dimension of the output array to 2-D or 3-D for frame-based inputs. The block ignores this parameter for 1-D and sample-based inputs. The **Frames** parameter has the following two settings:

- `Log frames separately (3-D array)`: Given an M-by-N frame-based input signal, the block outputs an M-by-N-by-K array, where K is the number of frames logged by the end of the simulation. (K is bounded above by the **Limit data points to last** parameter.) Each input frame is an element of the 3-D array. (See "Example 2: Frame-Based Inputs" on page 2-1095.)
- `Concatenate frames (2-D array)`: Given an M-by-N frame-based input signal with frame size *f*, the block outputs a (K**f*)-by-N matrix, where K**f* is the number of samples acquired by the end of the simulation. Each input frame is vertically concatenated to the previous frame to produce the 2-D array output. (See "Example 2: Frame-Based Inputs" on page 2-1095.)

Signal to Workspace always logs sample-based input data as 3-D arrays, regardless of the **Frame** parameter setting. Given an M-by-N sample-based signal, the block outputs an M-by-N-by-L array, where L is the number of samples logged by the end of the simulation (L is bounded above by the **Limit data points to last** parameter). Each sample-based matrix is an element of the 3-D array. (See "Example 1: Sample-Based Inputs" on page 2-1094.)

For 1-D vector inputs, the block outputs a 2-D matrix regardless of the setting of **Frame**. For a length-N 1-D vector input, the block outputs an L-by-N matrix. Each input vector is a row of the output matrix, vertically concatenated to the previous vector.

Output Dimension Summary

The following table summarizes the output array dimensions for various block inputs. In the table, *f* is the frame size of the input, K is the number of *frames* acquired by the end of the simulation, and L is the number of *samples* acquired by the end of the simulation (K and L are bounded above by the **Limit data points to last** parameter).

Input Signal Type	Signal To Workspace Output Dimension
Sample-based M-by-N matrix	M-by-N-by-L array
Length-N 1-D vector	L-by-N matrix
Frame-based M-by-N matrix; **Frame** set to `Log frames separately (3-D array)`	M-by-N-by-K array
Frame-based M-by-N matrix; **Frame** set to `Concatenate frames (2-D array)`	(K**f*)-by-N matrix K*f is the number of samples acquired by the end of the simulation.

Matching the Outputs of Signal To Workspace and To Workspace Blocks

The To Workspace block in the Simulink Sinks Library and the Signal To Workspace block can output the same array when they are given the same inputs. To match the blocks' outputs, set their parameters as follows.

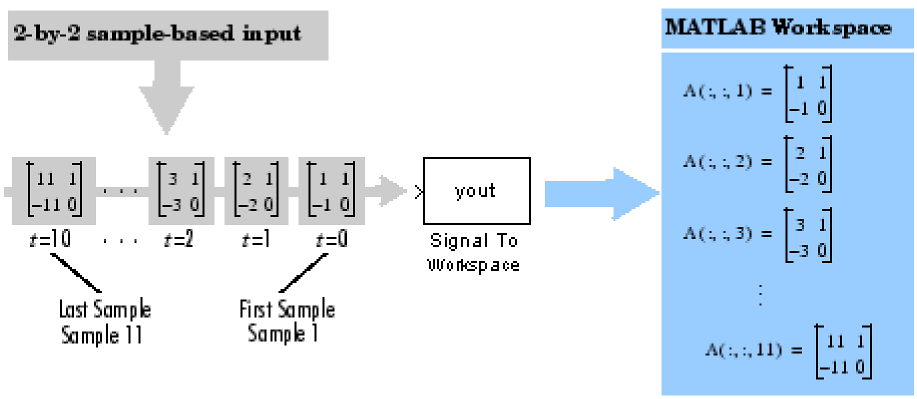
Block Parameters	Signal To Workspace	To Workspace
Limit data points to last	x (any positive integer or `inf`)	x
Decimation	y (any positive integer, not `inf`)	y

Block Parameters	Signal To Workspace	To Workspace
Sample Time	No such parameter	-1
Save format	No such parameter	`Array`
Frames	`Concatenate frames (2-D array)`	No such parameter

Examples

Example 1: Sample-Based Inputs

In the following model, the input to the Signal To Workspace block is a 2-by-2 sample-based matrix signal with a sample time of 1 (generated by a Signal From Workspace block). The Signal To Workspace block logs 11 samples by the end of the simulation, and creates a 2-by-2-by-11 array, A, in the MATLAB workspace.



The block settings are as follows.

Signal To Workspace Block Parameters	
Variable name	`yout`
Limit data points to last	`inf`
Decimation	`1`

Signal To Workspace Block Parameters	
Frames	Ignored since block input is not frame based
Configuration Dialog Box Parameters	
Start time	`0`
Stop time	`10`
Signal From Workspace Parameters (provides Signal To Workspace input)	
Signal	`input1` (defined below)
Sample time	`1`
Samples per frame	`1`
Form output after final data value by	`Setting to zero`

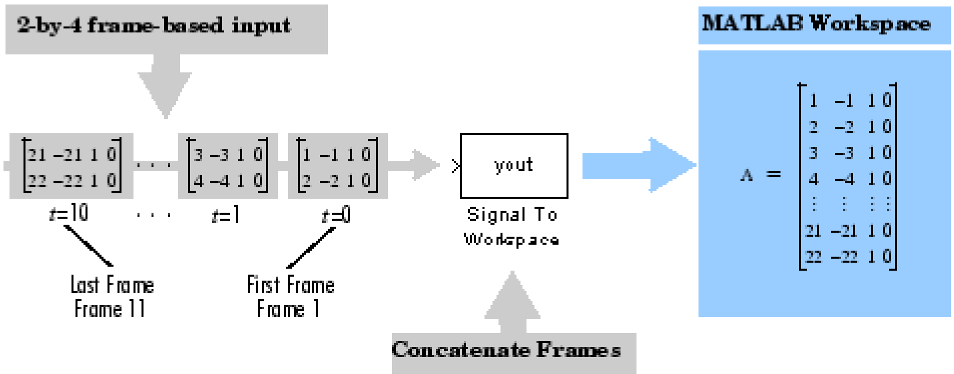
```
input1 = cat(3, [1 1; -1 0], [2 1; -2 0],...,[11 1; -11 0])
```

Example 2: Frame-Based Inputs

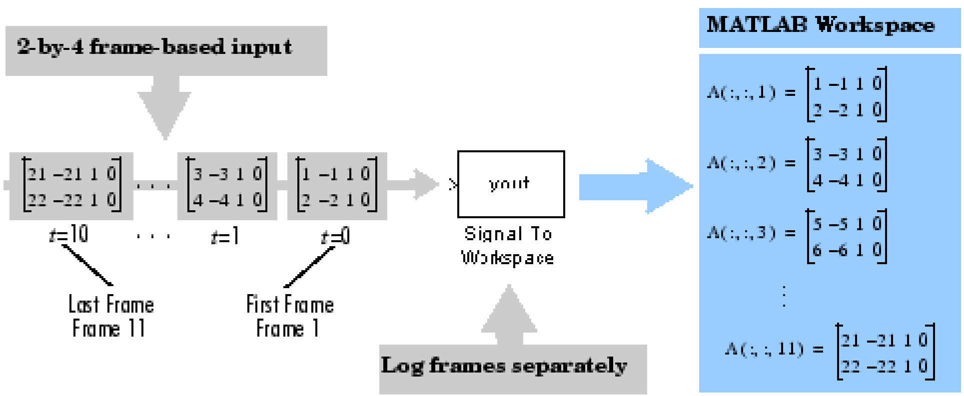
In the following model, the input to the Signal To Workspace block is a 2-by-4 frame-based matrix signal with a frame period of 1 (generated by a Signal From Workspace block). The block logs 11 frames (two samples per frame) by the end of the simulation. The frames are concatenated to create a 22-by-4 matrix, A, in the MATLAB workspace.

The block settings for the following model are similar to the settings used in Example 1, except **Frames** is set to `Concatenate frames (2-D array)` and the Signal From Workspace parameter, **Signal**, is set to `input2`, where

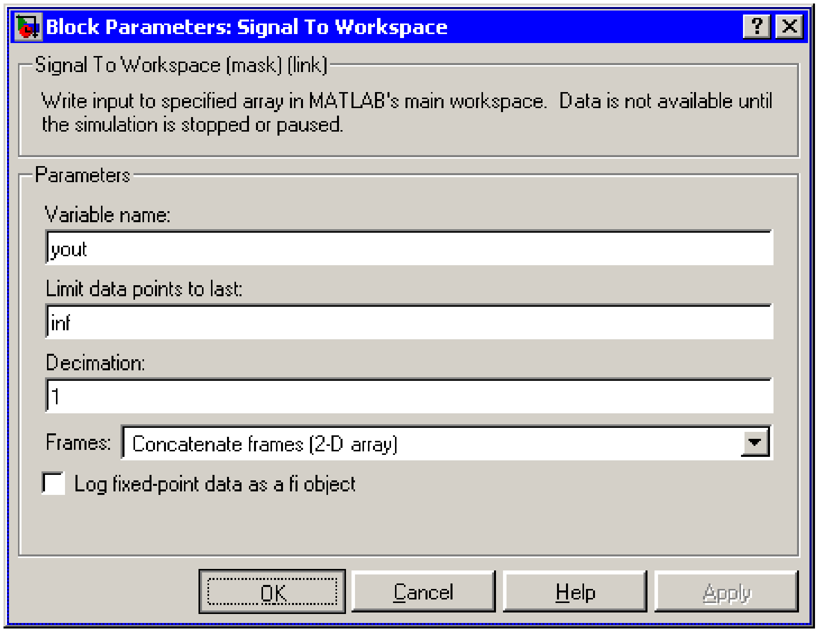
```
input2 = [1 -1 1 0; 2 -2 1 0; 3 -3 1 0;...; 22 -22 1 0]
```



In the 2-D output, there is no indication of where one frame ends and another begins. By setting **Frames** to `Log frames separately (3-D array)` in this model, you can easily see each frame in the MATLAB workspace, as illustrated in the following model. Each of the 11 frames is logged separately to create a 2-by-4-by-11 array, A, in the MATLAB workspace.



Dialog Box



Variable name
: The name of the array that holds the input data.

Limit data points to last
: The maximum number of input samples (for sample-based inputs) or input frames (for frame-based inputs) to be saved.

Decimation
: The decimation factor, d. Data is written at every dth sample.

Frames
: The output dimensionality for frame-based inputs. **Frames** can be set to `Concatenate frames (2-D array)` or `Log frames separately (3-D array)`. This parameter is ignored when inputs are not frame based.

Log fixed-point data as a fi object

Select to log fixed-point data to the MATLAB workspace as a Fixed-Point Toolbox™ `fi` object. Otherwise, fixed-point data is logged to the workspace as `double`.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Triggered To Workspace	Signal Processing Blockset
To Workspace	Simulink

Purpose

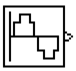
Generate continuous or discrete sine wave

Library

Signal Processing Sources

`dspsrcs4`

Description



The Sine Wave block generates a multichannel real or complex sinusoidal signal, with independent amplitude, frequency, and phase in each output channel. A real sinusoidal signal is generated when the **Output complexity** parameter is set to `Real`, and is defined by an expression of the type

$$y = A\sin(2\pi ft + \phi)$$

where you specify A in the **Amplitude** parameter, f in hertz in the **Frequency** parameter, and ϕ in radians in the **Phase offset** parameter. A complex exponential signal is generated when the **Output complexity** parameter is set to `Complex`, and is defined by an expression of the type

$$y = Ae^{j(2\pi ft+\phi)} = A\left\{\cos(2\pi ft + \phi) + j\sin(2\pi ft + \phi)\right\}$$

Sections of This Reference Page

- “Generating Multichannel Outputs” on page 2-1100
- “Output Sample Time and Samples Per Frame” on page 2-1100
- “Sample Mode” on page 2-1100
- “Discrete Computational Methods” on page 2-1101
- “Examples” on page 2-1104
- “Dialog Box” on page 2-1105
- “Supported Data Types” on page 2-1110
- “See Also” on page 2-1110

Generating Multichannel Outputs

For both real and complex sinusoids, the **Amplitude**, **Frequency**, and **Phase offset** parameter values (A, f, and ϕ) can be scalars or length-N vectors, where N is the desired number of channels in the output. When you specify at least one of these parameters as a length-N vector, scalar values specified for the other parameters are applied to every channel.

For example, to generate the three-channel output containing the real sinusoids below, set **Output complexity** to `Real` and the other parameters as follows:

- **Amplitude** = `[1 2 3]`
- **Frequency** = `[1000 500 250]`
- **Phase offset** = `[0 0 pi/2]`

$$y = \begin{cases} \sin(2000\pi t) & \text{(channel 1)} \\ 2\sin(1000\pi t) & \text{(channel 2)} \\ 3\sin\left(500\pi t + \dfrac{\pi}{2}\right) & \text{(channel 3)} \end{cases}$$

Output Sample Time and Samples Per Frame

In all discrete modes, the block buffers the sampled sinusoids into frames of size M, where you specify M in the **Samples per frame** parameter. The output is a frame-based M-by-N matrix with frame period $M*T_s$, where you specify T_s in the **Sample time** parameter. For M=1, the output is sample based.

Sample Mode

The **Sample mode** parameter specifies the block's sampling property, which can be `Continuous` or `Discrete`:

- `Continuous`

In continuous mode, the sinusoid in the ith channel, y_i, is computed as a continuous function,

$$y_i = A_i \sin(2\pi f_i t + \phi_i) \quad \text{(real)}$$

or

$$y_i = A_i e^{j(2\pi f_i t + \phi_i)} \qquad \text{(complex)}$$

and the block's output is continuous. In this mode, the block's operation is the same as that of a Simulink® Sine Wave block with **Sample time** set to `0`. This mode offers high accuracy, but requires trigonometric function evaluations at each simulation step, which is computationally expensive. Additionally, because this method tracks absolute simulation time, a discontinuity will eventually occur when the time value reaches its maximum limit.

Note also that many Signal Processing Blockset™ blocks do not accept continuous-time inputs.

- `Discrete`

  In discrete mode, the block's discrete-time output can be generated by directly evaluating the trigonometric function, by table lookup, or by a differential method. The three options are explained below.

Discrete Computational Methods

When you select `Discrete` from the **Sample mode** parameter, the secondary **Computation method** parameter provides three options for generating the discrete sinusoid:

- `Trigonometric Fcn`

- `Table Lookup`
- `Differential`

Trigonometric Fcn

The trigonometric function method computes the sinusoid in the *i*th channel, y_i, by sampling the continuous function

$$y_i = A_i \sin(2\pi f_i t + \phi_i) \quad \text{(real)}$$

or

$$y_i = A_i e^{j(2\pi f_i t + \phi_i)} \qquad \text{(complex)}$$

with a period of T_s, where you specify T_s in the **Sample time** parameter. This mode of operation shares the same benefits and liabilities as the `Continuous` sample mode described above.

At each sample time, the block evaluates the sine function at the appropriate time value *within the first cycle* of the sinusoid. By constraining trigonometric evaluations to the first cycle of each sinusoid, the block avoids the imprecision of computing the sine of very large numbers, and eliminates the possibility of discontinuity during extended operations (when an absolute time variable might overflow). This method therefore avoids the memory demands of the table lookup method at the expense of many more floating-point operations.

Table Lookup

The table lookup method precomputes the *unique* samples of every output sinusoid at the start of the simulation, and recalls the samples from memory as needed. Because a table of finite length can only be constructed when all output sequences repeat, the method requires that the period of every sinusoid in the output be evenly divisible by the sample period. That is, $1/(f_i T_s) = k_i$ must be an integer value for every channel i = 1, 2, ..., N. When the **Optimize table for** parameter is set to `Speed`, the table constructed for each channel contains k_i elements. When the **Optimize table for** parameter is set to `Memory`, the table constructed for each channel contains $k_i/4$ elements.

For long output sequences, the table lookup method requires far fewer floating-point operations than any of the other methods, but can demand considerably more memory, especially for high sample rates (long tables). This is the recommended method for models that are intended to emulate or generate code for DSP hardware, and that therefore need to be optimized for execution speed.

Differential

The differential method uses an incremental algorithm. This algorithm computes the output samples based on the output values computed at the previous sample time (and precomputed update terms) by making use of the following identities.

$$\sin(t+T_s) = \sin(t)\cos(T_s) + \cos(t)\sin(T_s)$$
$$\cos(t+T_s) = \cos(t)\cos(T_s) + \sin(t)\sin(T_s)$$

The update equations for the sinusoid in the ith channel, y_i, can therefore be written in matrix form as

$$\begin{bmatrix} \sin\{2\pi f_i(t+T_s)+\phi_i\} \\ \cos\{2\pi f_i(t+T_s)+\phi_i\} \end{bmatrix} = \begin{bmatrix} \cos(2\pi f_i T_s) & \sin(2\pi f_i T_s) \\ -\sin(2\pi f_i T_s) & \cos(2\pi f_i T_s) \end{bmatrix} \begin{bmatrix} \sin(2\pi f_i t+\phi_i) \\ \cos(2\pi f_i t+\phi_i) \end{bmatrix}$$

where you specify T_s in the **Sample time** parameter. Since T_s is constant, the right-hand matrix is a constant and can be computed once at the start of the simulation. The value of $A_i\sin[2\pi f_i(t+T_s)+\phi_i]$ is then computed from the values of $\sin(2\pi f_i t+\phi_i)$ and $\cos(2\pi f_i t+\phi_i)$ by a simple matrix multiplication at each time step.

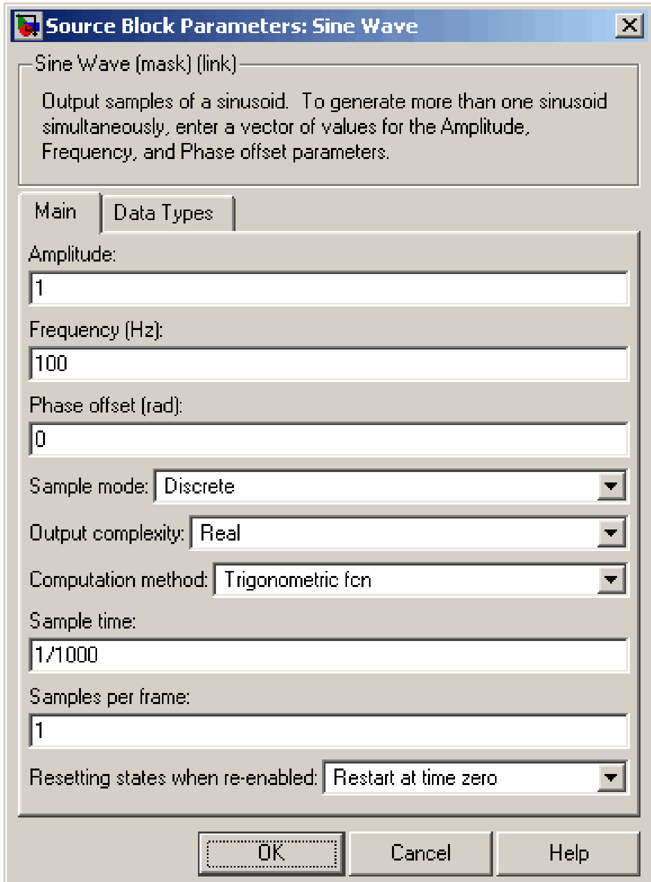
This mode offers reduced computational load, but is subject to drift over time due to cumulative quantization error. Because the method is not contingent on an absolute time value, there is no danger of discontinuity during extended operations (when an absolute time variable might overflow).

Examples

The `dspsinecomp` demo provides a comparison of all the available sine generation methods.

Dialog Box

The **Main** pane of the Sine Wave block dialog appears as follows.



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

Amplitude

A length-*N* vector containing the amplitudes of the sine waves in each of *N* output channels, or a scalar to be applied to all *N* channels. The vector length must be the same as that specified for the **Frequency** and **Phase offset** parameters. Tunable when **Computation method** is to `Trigonometric fcn` or `Differential`.

Frequency

A length-*N* vector containing frequencies, in rad/s, of the sine waves in each of *N* output channels, or a scalar to be applied to all *N* channels. The vector length must be the same as that specified for the **Amplitude** and **Phase offset** parameters. You can specify positive, zero, or negative frequencies. Tunable when **Sample mode** is `Continuous` or **Computation method** is `Trigonometric fcn`.

Phase offset

A length-*N* vector containing the phase offsets, in radians, of the sine waves in each of *N* output channels, or a scalar to be applied to all *N* channels. The vector length must be the same as that specified for the **Amplitude** and **Frequency** parameters. Tunable when **Sample mode** is `Continuous` or **Computation method** is `Trigonometric fcn`.

Sample mode

The block's sampling behavior, `Continuous` or `Discrete`. This parameter is not tunable.

Output complexity

The type of waveform to generate: `Real` specifies a real sine wave, `Complex` specifies a complex exponential. This parameter is not tunable.

Computation method

The method by which discrete-time sinusoids are generated: `Trigonometric fcn`, `Table lookup`, or `Differential`. This parameter is not tunable. This parameter is disabled when you select `Continuous` from the **Sample mode** parameter. For details, see "Discrete Computational Methods" on page 2-1101.

Optimize table for

Optimizes the table of sine values for `Speed` or `Memory` (this parameter is only visible when the **Computation method** parameter is set to `Table lookup`). When optimized for speed, the table contains k elements, and when optimized for memory, the table contains $k/4$ elements, where k is the number of input samples in one full period of the sine wave.

Sample time

The period with which the sine wave is sampled, T_s. The block's output frame period is $M*T_s$, where you specify M in the **Samples per frame** parameter. This parameter is disabled when you select `Continuous` from the **Sample mode** parameter. This parameter is not tunable.

Samples per frame

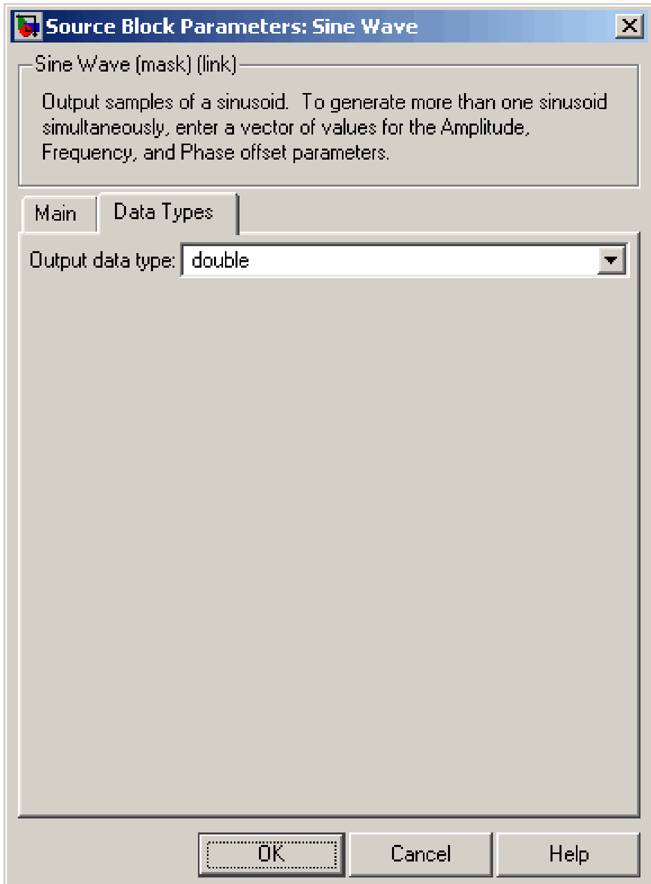
The number of consecutive samples from each sinusoid to buffer into the output frame, M. When the value of this parameter is `1`, the block outputs a sample-based signal.

This parameter is disabled when you select `Continuous` from the **Sample mode** parameter.

Resetting states when re-enabled

This parameter only applies when the Sine Wave block is located inside an enabled subsystem and the **States when enabling** parameter of the Enable block is set to `reset`. This parameter determines the behavior of the Sine Wave block when the subsystem is re-enabled. The block can either reset itself to its starting state (`Restart at time zero`), or resume generating the sinusoid based on the current simulation time (`Catch up to simulation time`). This parameter is disabled when you select `Continuous` from the **Sample mode** parameter.

The **Data types** pane of the Sine Wave block dialog appears as follows.



Output data type

Specify the output data type in out of the following ways:

Choose one of the built-in data types from the list.

Choose `Fixed-point` to specify the output data type and scaling in the **Word length**, **Set fraction length in output to**, and **Fraction length** parameters.

Choose `User-defined` to specify the output data type and scaling in the **User-defined data type**, **Set fraction length in output to**, and **Fraction length** parameters.

Choose `Inherit via back propagation` to set the output data type and scaling to match the next block downstream.

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

User-defined data type
Specify any built-in or fixed-point data type. You can specify fixed-point data types using the following Simulink® Fixed Point™ functions: `sfix`, `ufix`, `sint`, `uint`, `sfrac`, and `ufrac`. This parameter is only visible when you select `User-defined` for the **Output data type** parameter.

Set fraction length in output to
Specify the scaling of the fixed-point output by either of the following two methods:

Choose `Best precision` to have the output scaling automatically set such that the output signal has the best possible precision.

Choose `User-defined` to specify the output scaling in the **Fraction length** parameter.

This parameter is only visible when you select `Fixed-point` for the **Output data type** parameter, or when you select `User-defined` and the specified output data type is a fixed-point data type.

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

the **Output data type** parameter and `User-defined` for the **Set fraction length in output to** parameter.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)
- 8-, 16-, and 32-bit signed integers

See Also

Chirp	Signal Processing Blockset
Complex Exponential	Signal Processing Blockset
Signal From Workspace	Signal Processing Blockset
Signal Generator	Simulink
Sine Wave	Simulink
`sin`	MATLAB

Purpose

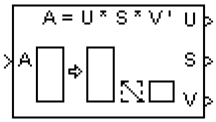
Factor matrix using singular value decomposition

Library

Math Functions / Matrices and Linear Algebra / Matrix Factorizations

`dspfactors`

Description



The Singular Value Decomposition block factors the M-by-N input matrix A such that

$$A = U \cdot diag(S) \cdot V^*$$

where

- U is an M-by-P matrix
- V is an N-by-P matrix
- S is a length-P vector
- P is defined as min(M,N)

When

- $M = N$, U and V are both M-by-M unitary matrices
- $M > N$, V is an N-by-N unitary matrix, and U is an M-by-N matrix whose columns are the first N columns of a unitary matrix
- $N > M$, U is an M-by-M unitary matrix, and V is an N-by-M matrix whose columns are the first M columns of a unitary matrix

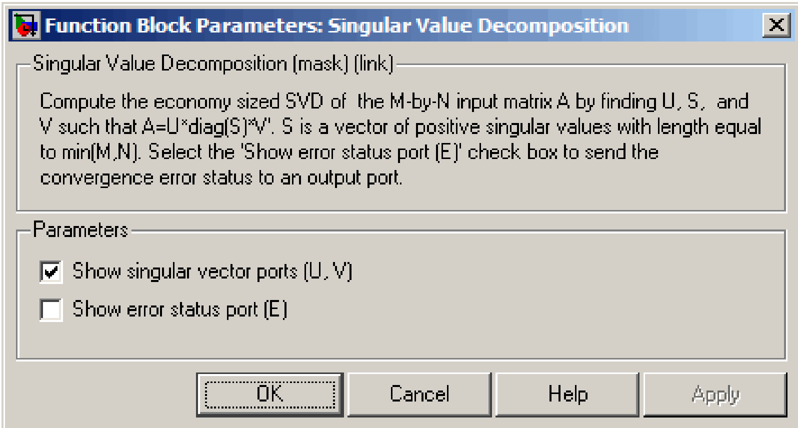
In all cases, S is a 1-D vector of positive singular values having length P.

Length-N row inputs are treated as length-N columns.

Note that the first (maximum) element of output S is equal to the 2-norm of the matrix A.

The output is always sample based.

Dialog Box



Show singular vector ports
Select to enable the `U` and `V` output ports.

Show error status port
Select to enable the E output port, which reports a failure to converge. The possible values you can receive on the port are:

- `0` — The singular value decomposition calculation converges.
- `1` — The singular value decomposition calculation does not converge.

If the singular value decomposition calculation fails to converge, the output at ports U, S, and V are undefined matrices of the correct size.

References

Golub, G. H., and C. F. Van Loan. *Matrix Computations*. 3rd ed. Baltimore, MD: Johns Hopkins University Press, 1996.

Supported Data Types

Port	Supported Data Types
A	• Double-precision floating point • Single-precision floating point
U	• Double-precision floating point • Single-precision floating point
S	• Double-precision floating point • Single-precision floating point
V	• Double-precision floating point • Single-precision floating point
E	• Boolean

See Also

Autocorrelation LPC	Signal Processing Blockset
Cholesky Factorization	Signal Processing Blockset
LDL Factorization	Signal Processing Blockset
LU Inverse	Signal Processing Blockset
Pseudoinverse	Signal Processing Blockset
QR Factorization	Signal Processing Blockset
SVD Solver	Signal Processing Blockset
`svd`	MATLAB

See “Matrix Factorizations” for related information.

Sort

Purpose

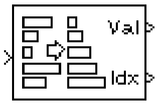
Sort input elements by value

Library

Statistics

`dspstat3`

Description



The Sort block ranks the values of the input elements using either a quick sort or an insertion sort algorithm. The quick sort algorithm uses a recursive sort method and is faster at sorting more than 32 elements. The insertion sort algorithm uses a non-recursive method and is faster at sorting less than 32 elements. You should also always use the insertion sort algorithm when you are generating code from the Sort block if you do not want recursive function calls in your code. To specify the sort method, use the **Sort algorithm** parameter.

The **Mode** parameter specifies the block's mode of operation, and can be set to `Value`, `Index`, or `Value and index`.

The Sort block supports real and complex floating-point and fixed-point inputs. Signed and unsigned fixed-point signals are supported. The block output has the same signedness as the input.

Value Mode

When **Mode** is set to `Value`, the block sorts the elements in each column of the M-by-N input matrix `u` in order of ascending or descending value, as specified by the **Sort order** parameter.

```
val = sort(u)
val = flipud(sort(u))
```

For convenience, length-M 1-D vector inputs and *sample-based* length-M row vector inputs are both treated as M-by-1 column vectors.

The output at each sample time, `val`, is an M-by-N matrix containing the sorted columns of `u`. The output has the same frame status as the input.

Complex inputs are sorted by *magnitude squared*. For complex value `u = a + bi`, the magnitude squared is $a^2 + b^2$.

Index Mode

When **Mode** is set to `Index`, the block sorts the elements in each column of the M-by-N input matrix `u`,

```
[val,idx] = sort(u)
[val,idx] = flipud(sort(u))
```

and outputs the sample-based M-by-N index matrix, `idx`. The `j`th column of `idx` is an index vector that permutes the `j`th column of `u` to the desired sorting order.

```
val(:,j) = u(idx(:,j),j)
```

The index value outputs are always 32-bit unsigned integer values.

As in `Value` mode, length-M 1-D vector inputs and *sample-based* length-M row vector inputs are both treated as M-by-1 column vectors.

Value and Index Mode

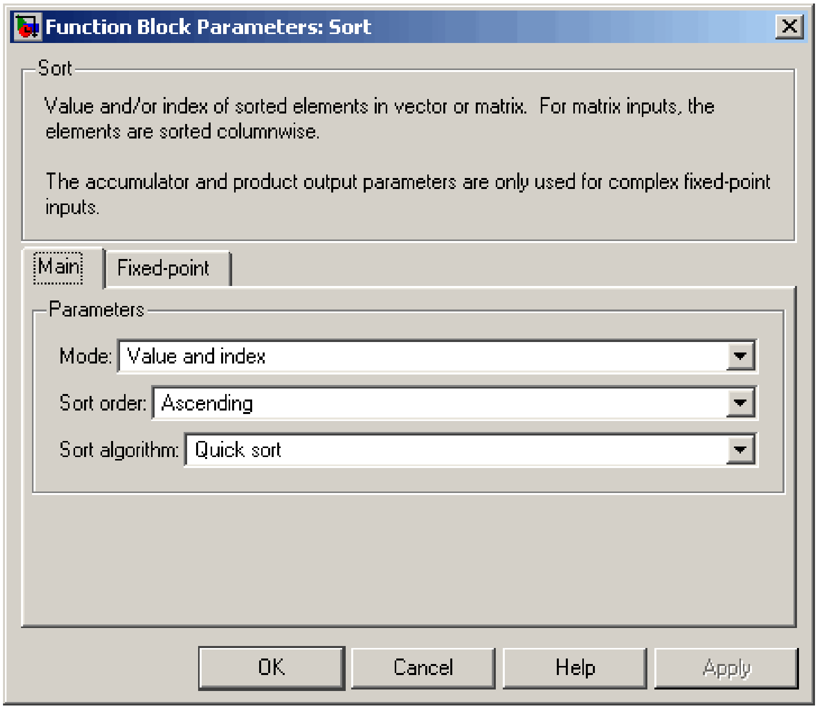
When **Mode** is set to `Value and index`, the block outputs both the sorted matrix, `val`, and the index matrix, `idx`.

Fixed-Point Data Types

The parameters on the **Fixed-point** pane are only used for complex fixed-point inputs. Complex fixed-point inputs are sorted by magnitude squared. The sum of the squares of the real and imaginary parts of such an input are formed before a comparison is made, as described in "Value Mode" on page 2-1114. The results of the squares of the real and imaginary parts are placed into the product output data type. The result of the sum of the squares is placed into the accumulator data type. These parameters are ignored for other types of inputs.

Dialog Box

The **Main** pane of the Sort block dialog appears as follows.



Mode

Specify the block's mode of operation: Output the sorted matrix (`Value`), the index matrix (`Index`), or both (`Value and index`).

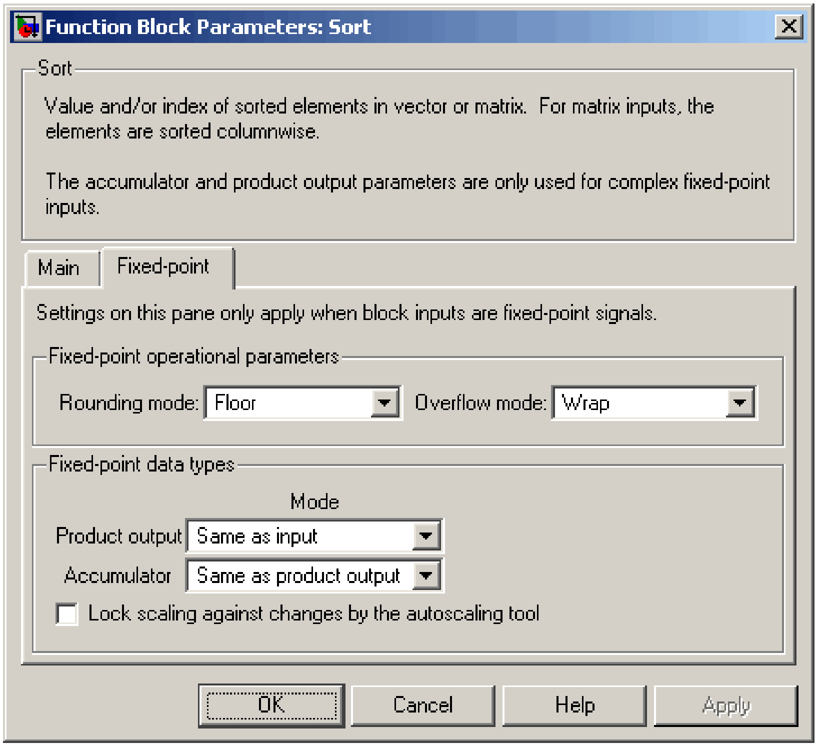
Sort order

Specify the order in which to sort the training points, `Descending` or `Ascending`.

Sort algorithm

Specify whether the elements of the input are sorted using a `Quick sort` or an `Insertion sort` algorithm.

The **Fixed-point** pane of the Sort block dialog appears as follows.



Note The parameters on the **Fixed-point** pane are only used for complex fixed-point inputs. The sum of the squares of the real and imaginary parts of such an input are formed before a comparison is made, as described in "Value Mode" on page 2-1114. The results of the squares of the real and imaginary parts are placed into the product output data type. The result of the sum of the squares is placed into the accumulator data type. These parameters are ignored for other types of inputs.

Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Product output
Use this parameter to specify how you would like to designate the product output word and fraction lengths resulting from a complex-complex multiplication in the block. See "Multiplication Data Types" for more information:

When you select `Same as input`, these characteristics match those of the input to the block.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator
Use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block. See "Multiplication Data Types" for more information:

When you select `Same as product output`, these characteristics match those of the product output

When you select `Same as input`, these characteristics match those of the input to the block.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool
Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • 8-, 16-, 32-, and 128-bit unsigned integers • 8-, 16-, 32-, and 128-bit signed integers
Val	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • 8-, 16-, 32-, and 128-bit unsigned integers • 8-, 16-, 32-, and 128-bit signed integers
Idx	• 32-bit unsigned integers

See Also

Histogram	Signal Processing Blockset
Median	Signal Processing Blockset
`sort`	MATLAB

Spectrum Scope

Purpose

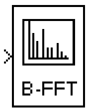
Compute and display periodogram of each input signal

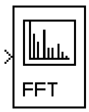
Library

Signal Processing Sinks

`dspsnks4`

Description





The Spectrum Scope block computes and displays the periodogram of the input. The input can be a sample-based or frame-based vector or a frame-based matrix.

Note When the **Buffer input** and **Specify FFT length** parameters are both cleared, the block input length must be a power of two.

Scope Properties Pane

The **Buffer input** check box must be selected for sample-based inputs. Buffering is optional for frame-based inputs. When the block input is buffered, you specify the number of input samples that the block buffers before computing and displaying the magnitude FFT in the **Buffer size** parameter. You also use the **Buffer overlap** parameter to specify the number of samples from the previous buffer to include in the current buffer. The number of new input samples the block acquires before computing and displaying the magnitude FFT is the difference between the buffer size and the buffer overlap.

The display update period is

$$(M_o - L) * T_s$$

where

- M_o = buffer size
- L = buffer overlap
- T_s = input sample period

For negative buffer overlap values, the block discards the appropriate number of input samples after the buffer fills, and updates the scope display at a slower rate than in the zero-overlap case.

The **Window type** and **Window sampling** parameters apply to the specification of the window function; see the Window Function block reference page for more details on these parameters.

The FFT length used by the block, N_{fft}, is determined in the following ways:

- If you clear the **Specify FFT length** check box and select **Buffer input**, the block uses the buffer size as the FFT size.
- If you clear the both the **Specify FFT length** and **Buffer input** check boxes, the block uses the input size as the FFT size.
- If you select the **Specify FFT length** check box, the **FFT length** parameter appears on the dialog box. Enter the number of samples on which you want the block to perform the FFT. This value must be a power of two.

The block zero pads or wraps every channel's buffer to the FFT length before computing the FFT.

The number of spectra to average is set by the **Number of spectral averages** parameter. Setting this parameter to `1` effectively disables averaging; see the Periodogram block reference page for more information.

Display Properties Pane

For information about these parameters, see "Display Properties Pane" on page 2-1352 of the Vector Scope block reference page.

Axis Properties Pane

The **Frequency units** parameter specifies whether the frequency axis values should be in units of Hertz or rad/s. When the **Frequency units** parameter specifies `Hertz`, the spacing between frequency points is $1/(N_{fft}T_s)$. For **Frequency units** of `rad/sec`, the spacing between frequency points is $2\pi/(N_{fft}T_s)$.

The **Frequency range** parameter specifies the range of frequencies over which the magnitudes in the input should be plotted. The available options are `[0..Fs/2], [-Fs/2..Fs/2], and [0..Fs]`, where F_s is the original time-domain signal's sample frequency.

Note that all of the Signal Processing Blockset™ FFT-based blocks, including those in the Power Spectrum Estimation library, compute the FFT at frequencies in the range $[0,F_s)$. The **Frequency range** parameter controls only the displayed range of the signal.

The **Display DC as** parameter allows you to relabel the x-axis of the scope. Specify the new label for the DC frequency (0 Hz).

If you select the **Inherit sample increment from input** check box, the block computes the frequency data from the sample period of the input to the block. This is valid when the following conditions hold:

- The input to the block is the original signal, with no samples added or deleted (by insertion of zeros, for example).
- The sample period of the time-domain signal in the simulation is equal to the period with which the physical signal was originally sampled.

In cases where not all of these conditions hold, specify the appropriate value for the **Sample time of original time series** parameter.

When **Frequency display limits** is set to `Auto`, the block scales the horizontal axis of frequency-domain signals automatically. To do this, the Spectrum Scope block needs to know the actual sample period of the time-domain input. Specify this in the **Sample time of original time series** parameter, T_s.

When **Frequency display limits** is set to `User-defined`, the **Minimum X-limit** and **Maximum X-limit** parameters set the range of the horizontal axis. Setting these parameters is analogous to setting the `xmin` and `xmax` values of the MATLAB® `axis` function.

The **Amplitude scaling** parameter allows you to select `Magnitude-squared` or `dB` scaling along the *y*-axis.

Minimum Y-limit and **Maximum Y-limit** parameters set the range of the vertical axis. Setting these parameters is analogous to setting the `ymin` and `ymax` values of the MATLAB `axis` function.

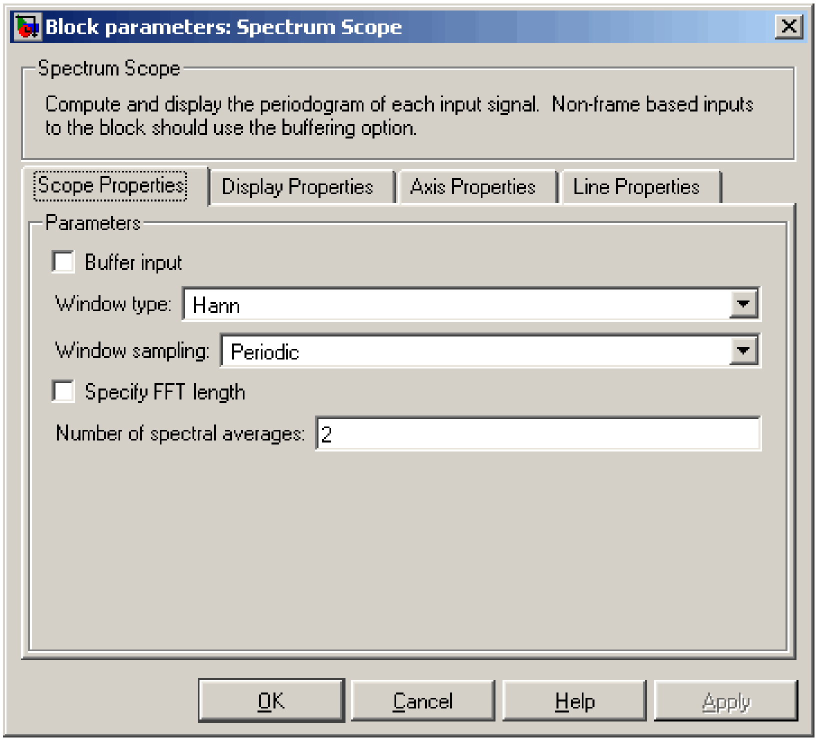
The **Y-axis title** is the text displayed to the left of the y-axis.

Line Properties Pane

For information about these parameters, see "Line Properties Pane" on page 2-1357 in the Vector Scope block reference page.

Dialog Box

Scope Properties Pane



Buffer input

Select this check box to rebuffer the input data. This check box must be selected for sample-based inputs, but is optional for frame-based inputs.

This functionality is not supported for use with external mode. Instead, clear this check box and use a Buffer block prior to the Spectrum Scope in your model.

Buffer size

Specify the number of input samples that the block buffers before computing and displaying the magnitude FFT. When the **Specify FFT length** parameter is not selected, this value must be a power of two.

This parameter is only visible when the **Buffer input** check box is selected.

Buffer overlap

Specify the number of samples from the previous buffer to include in the current buffer. The number of new input samples the block acquires before computing and displaying the magnitude FFT is the difference between the buffer size and the buffer overlap.

This parameter is only visible when the **Buffer input** check box is selected.

Window type

Enter the type of window to apply. See the Window Function block reference page for more details. Tunable.

Stopband attenuation in dB

Enter the level, in dB, of stopband attenuation, R_s, for the Chebyshev window.. Tunable.

This parameter is only visible when `Chebyshev` is selected for the **Window type** parameter.

Beta

Enter the β parameter for the Kaiser window. Increasing **Beta** widens the mainlobe and decreases the amplitude of the window sidelobes in the window's frequency magnitude response. Tunable.

This parameter is only visible if `Kaiser` is selected for the **Window type** parameter.

Window sampling

Choose `Symmetric` or `Periodic`. Tunable.

This parameter is only visible if `Blackman`, `Hamming`, `Hann`, or `Hanning` is selected for the **Window type** parameter.

Specify FFT length

Select this check box to specify the FFT length yourself in the **FFT length** parameter.

FFT length

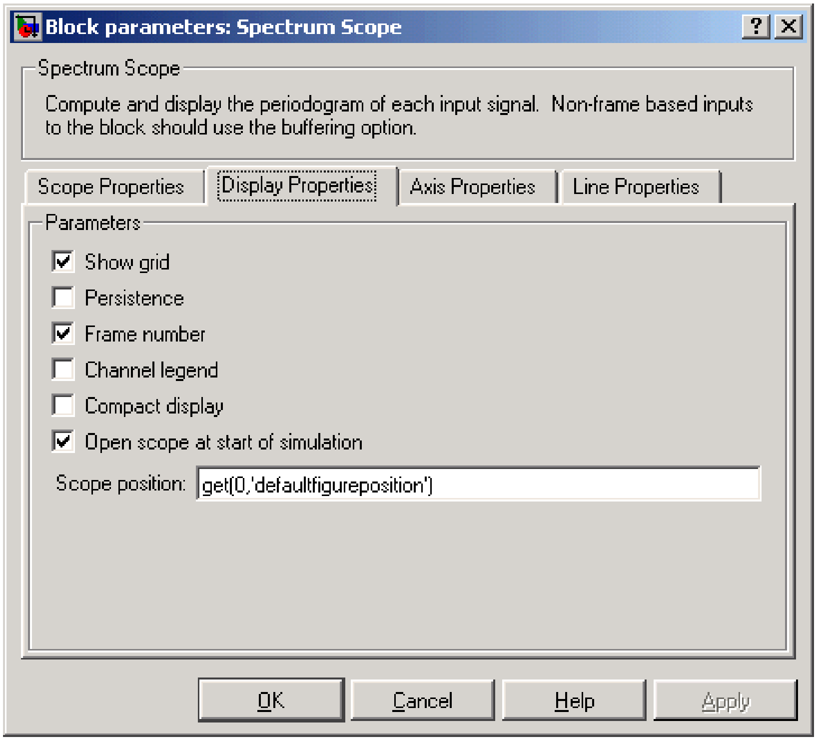
Enter the number of samples on which you want the block to perform the FFT. This value must be a power of two.

This parameter is only visible when then **Specify FFT length** check box is selected.

Number of spectral averages

The number of spectra to average. Setting this parameter to `1` effectively disables averaging. See the Periodogram block reference page for more information.

Display Properties Pane



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

Persistence
Select this check box to maintain successive displays. That is, the scope does not erase the display after each frame (or collection of frames), but overlays successive input frames in the scope display. Tunable.

Frame number
If you select this check box, the number of the current frame in the input sequence appears in the Vector Scope window. Tunable.

Channel legend

Toggles the legend on and off. Tunable.

Compact display

Resizes the scope to fill the window. Tunable.

Open scope at start of simulation

Select this check box to open the scope at the start of the simulation. When this parameter is cleared, the scope not open automatically during the simulation. Tunable.

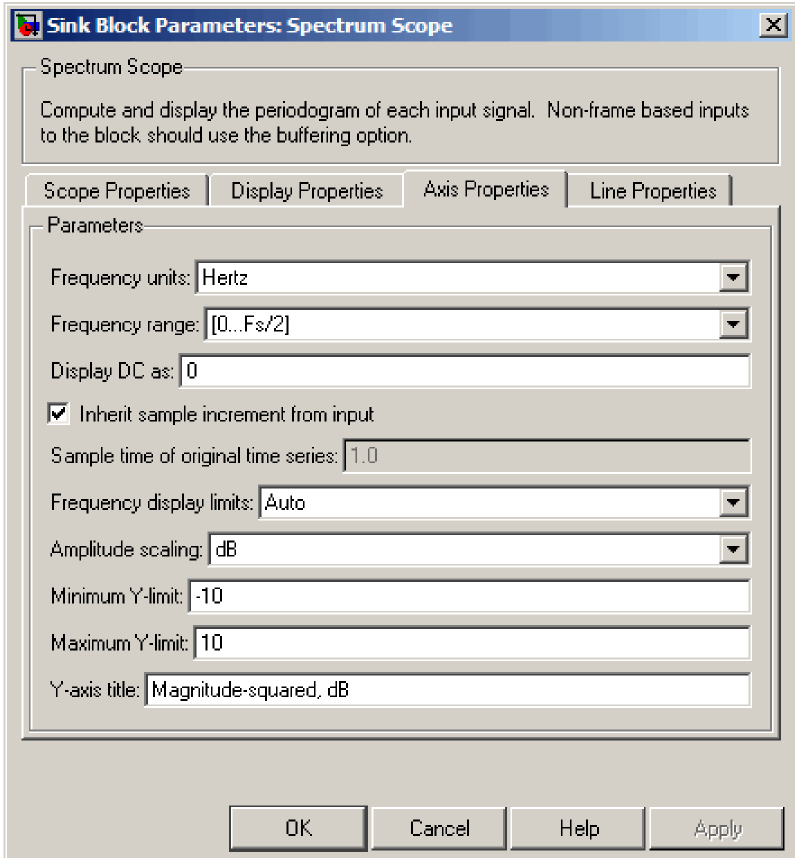
Open scope immediately

If the scope is not open during simulation, select this check box to open it. This parameter is visible only while the simulation is running.

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.

Axis Properties Pane



Frequency units

Choose the frequency units for the horizontal axis, `Hertz` or `rad/sec`. Tunable.

Frequency range

Specify the frequency range over which to plot the data. Tunable.

Display DC as
: This parameter allows you to relabel the x-axis of the scope. Specify the new label for the DC frequency (0 Hz).

Inherit sample increment from input
: If you select this check box, the block computes the time-domain sample period from the frame period and frame size of the frequency-domain input. Use this parameter only when the length of each frame of frequency-domain data is the same as the length of the frame of time-domain data from which it was generated. Tunable.

Sample time of original time series
: Enter the sample period of the original time-domain signal. Tunable.

Frequency display limits
: Select `Auto` to have the limits of the x-axis set for you automatically, or `User-defined` to set the limits yourself in the **Minimum X-limit** and **Maximum X-limit** parameters.

Minimum X-limit
: Specify the minimum value of the x-axis. Setting this parameter is analogous to setting the `xmin` value of the MATLAB `axis` function. This parameter is only visible if the **Frequency display limits** parameter is set to `User-defined`. Tunable.

Maximum X-limit
: Specify the maximum value of the x-axis. Setting this parameter is analogous to setting the `xmax` value of the MATLAB `axis` function. This parameter is only visible if the **Frequency display limits** parameter is set to `User-defined`. Tunable.

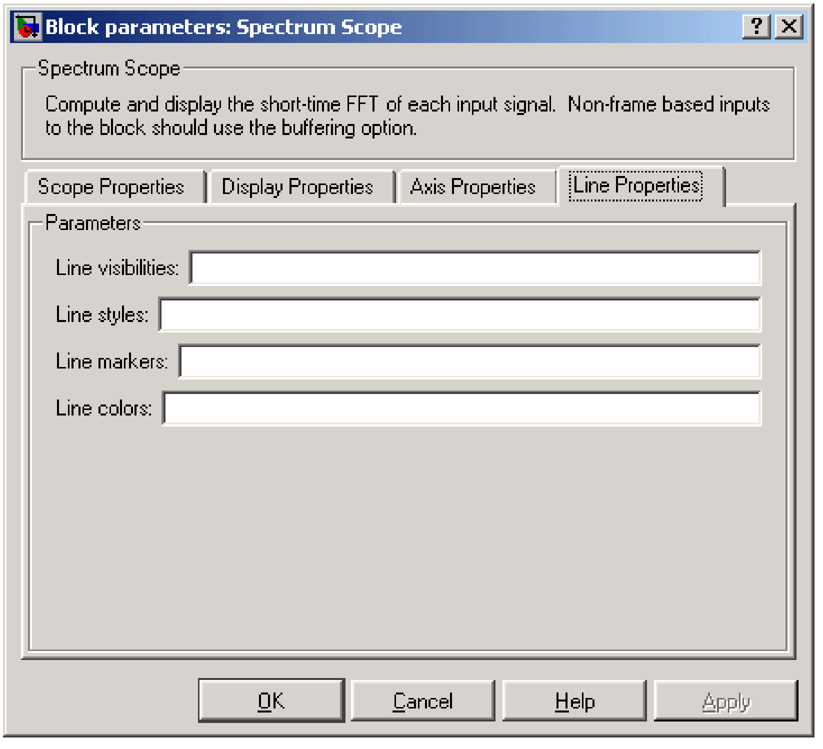
Amplitude scaling
: Choose the scaling for the y-axis, `dB` or `Magnitude-squared`. Tunable.

Minimum Y-limit
: Specify the minimum value of the *y*-axis. Setting this parameter is analogous to setting the `ymin` value of the MATLAB `axis` function. Tunable.

Maximum Y-limit
: Specify the maximum value of the *y*-axis. Setting this parameter is analogous to setting the `ymax` value of the MATLAB `axis` function.Tunable.

Y-axis title
: Specify text to be displayed to the left of the *y*-axis. Tunable.

Line Properties Pane



For more information about these parameters, see “Line Properties Pane” on page 2-1357 in the Vector Scope block reference page.

Line visibilities
Enter `on` or `off` to specify the visibility of the various channels' scope traces. Separate your choices for each channel with by a pipe (`|`) symbol. Tunable.

Line styles
Enter the line styles of the various channels' scope traces using the MATLAB `line` function `LineStyle` formats. Separate your choices for each channel with by a pipe (`|`) symbol. Tunable.

Line markers
Enter the line markers of the various channels' scope traces using the MATLAB `line` function `Marker` formats. Separate your choices for each channel with by a pipe (`|`) symbol. Tunable.

Line colors
Enter the colors of the various channels' scope traces using the MATLAB `ColorSpec` formats. Separate your choices for each channel with by a pipe (`|`) symbol. Tunable.

Supported Data Types

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

See Also

FFT	Signal Processing Blockset
Periodogram	Signal Processing Blockset

Short-Time FFT	Signal Processing Blockset
Vector Scope	Signal Processing Blockset
Window Function	Signal Processing Blockset

Purpose

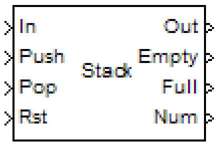
Store inputs into LIFO register

Library

Signal Management / Buffers

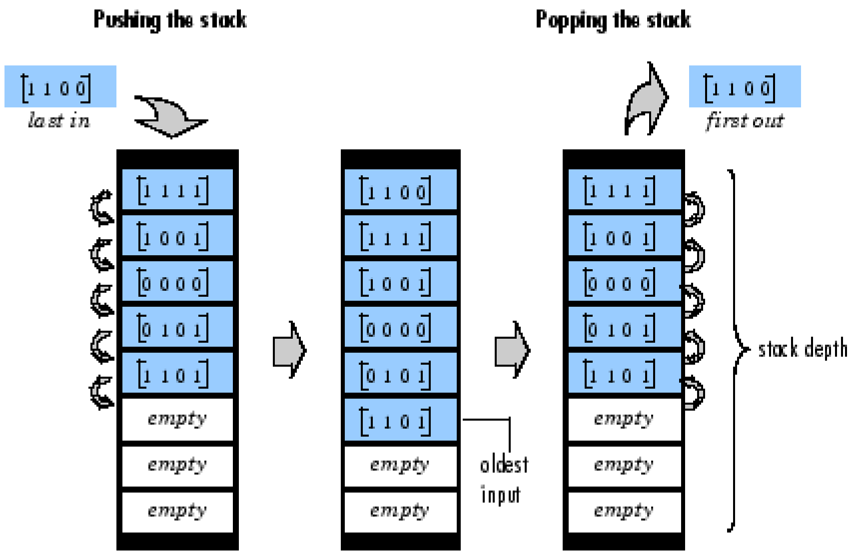
`dspbuff3`

Description



The Stack block stores a sequence of input samples in a last in, first out (LIFO) register. The register capacity is set by the **Stack depth** parameter, and inputs can be scalars, vectors, or matrices.

The block *pushes* the input at the `In` port onto the top of the stack when a trigger event is received at the `Push` port. When a trigger event is received at the `Pop` port, the block *pops* the top element off the stack and holds the `Out` port at that value. The last input to be pushed onto the stack is always the first to be popped off.



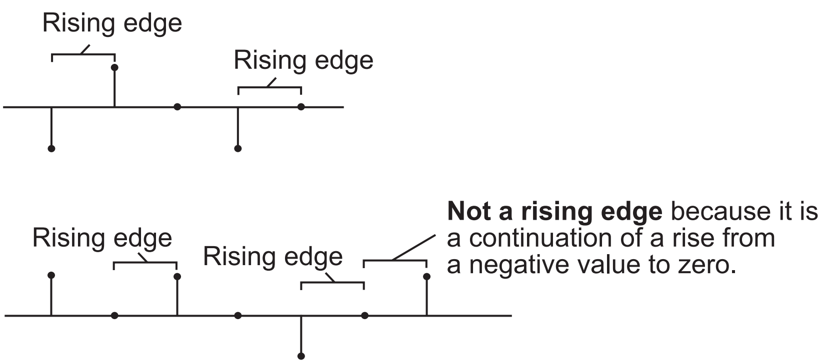
A trigger event at the optional `Rst` port empties the stack contents. When you select **Clear output port on reset**, then a trigger event at the `Rst` port empties the stack *and* sets the value at the `Out` port to zero. This setting also applies when a disabled subsystem containing the Stack block is reenabled; the `Out` port value is only reset to zero in this case when you select **Clear output port on reset**.

When two or more of the control input ports are triggered at the same time step, the operations are executed in the following order:

1. `Rst`
2. `Push`
3. `Pop`

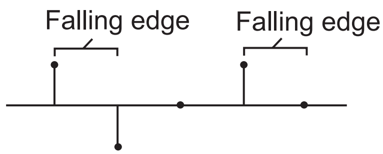
The rate of the trigger signal must be the same as the rate of the data signal input. You specify the triggering event for the `Push`, `Pop`, and `Rst` ports in the **Trigger type** pop-up menu:

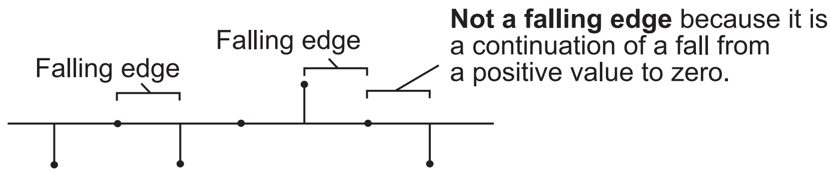
- `Rising edge` — Triggers execution of the block when the trigger input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers execution of the block when the trigger input does one of the following:

- Falls from a positive value to a negative value or zero
- Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)





- `Either edge` — Triggers execution of the block when the trigger input is a `Rising edge` or `Falling edge` (as described above).
- `Non-zero sample` — Triggers execution of the block at each sample time that the trigger input is not zero.

Note When running simulations in the Simulink® `MultiTasking` mode, sample-based trigger signals have a one-sample latency, and frame-based trigger signals have one frame of latency. Thus, there is a one-sample or one-frame delay between the time the block detects a trigger event, and when it applies the trigger. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

The **Push full stack** parameter specifies the block's behavior when a trigger is received at the `Push` port but the register is full. The **Pop empty stack** parameter specifies the block's behavior when a trigger is received at the Pop port but the register is empty. The following options are available for both cases:

- Ignore — Ignore the trigger event, and continue the simulation.
- Warning — Ignore the trigger event, but display a warning message in the MATLAB® command window.
- Error — Display an error dialog box and terminate the simulation.

Note The **Push full stack** and **Pop empty stack** parameters are diagnostic parameters. Like all diagnostic parameters on the Configuration Parameters dialog box, they are set to Ignore in the code generated for this block by Real-Time Workshop code generation software.

The **Push full stack** parameter additionally offers the **Dynamic reallocation** option, which dynamically resizes the register to accept as many additional inputs as memory permits. To find out how many elements are on the stack at a given time, enable the Num output port by selecting the **Show number of stack entries port** parameter.

Note When Dynamic reallocation is selected, the **System target file** parameter on the **Real-Time Workshop** pane of the Configuration Parameters dialog box must be set to grt_malloc.tlc Generic Real-Time Target with dynamic memory allocation.

Examples

Example 1

The table below illustrates the Stack block's operation for a **Stack depth** of 4, **Trigger type** of Either edge, and **Clear output port on reset** enabled. Because the block triggers on both rising and falling edges in this example, each transition from 1 to 0 or 0 to 1 in the Push, Pop, and Rst columns below represents a distinct trigger event. A 1 in the `Empty` column indicates an empty buffer, while a `1` in the `Full` column indicates a full buffer.

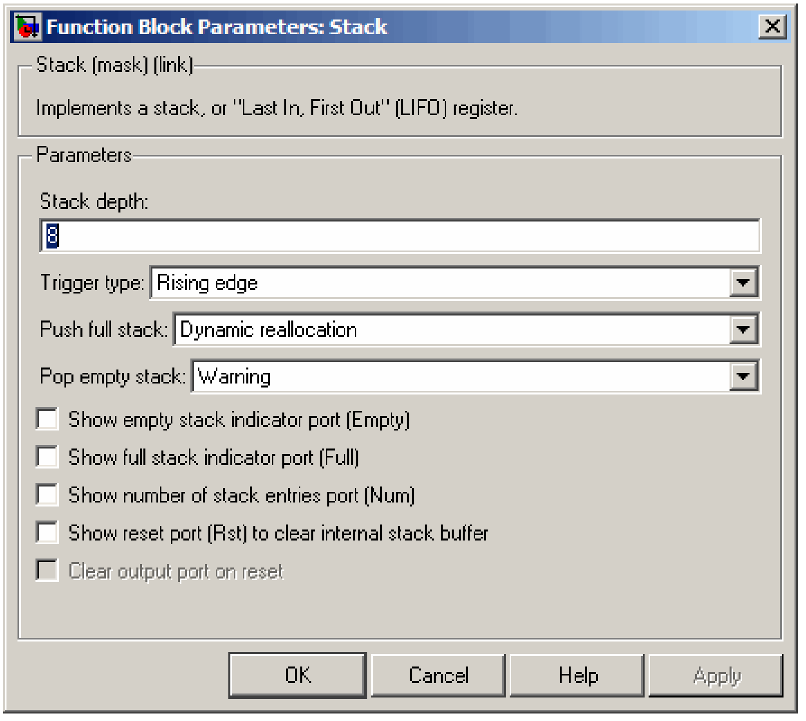
In	Push	Pop	Rst	Stack	Out	Empty	Full	Num
1	0	0	0	top bottom	0	1	0	0
2	1	0	0	top 2 bottom	0	0	0	1
3	0	0	0	top 3 2 bottom	0	0	0	2
4	1	0	0	top 4 3 2 bottom	0	0	0	3
5	0	0	0	top 5 4 3 2 bottom	0	0	1	4
6	0	1	0	top 4 3 2 bottom	5	0	0	3
7	0	0	0	top 3 2 bottom	4	0	0	2
8	0	1	0	top 2 bottom	3	0	0	1
9	0	0	0	top bottom	2	1	0	0
10	1	0	0	top 10 bottom	2	0	0	1
11	0	0	0	top 11 10 bottom	2	0	0	2
12	1	0	1	top 12 bottom	0	0	0	1

Note that at the last step shown, the `Push` and `Rst` ports are triggered simultaneously. The `Rst` trigger takes precedence, and the stack is first cleared and then pushed.

Example 2

The `dspqdemo` demo provides an example of the related Queue block.

Dialog Box



Stack depth
The number of entries that the LIFO register can hold.

Trigger type
The type of event that triggers the block's execution. The rate of the trigger signal must be the same as the rate of the data signal input.

Push full stack

Response to a trigger received at the `Push` port when the register is full. Inputs to this port must have the same built-in data type as inputs to the `Pop` and `Rst` input ports.

When `Dynamic reallocation` is selected, the **System target file** parameter on the **Real-Time Workshop** pane of the Configuration Parameters dialog box must be set to `grt_malloc.tlc  Generic Real-Time Target with dynamic memory allocation`.

Pop empty stack

Response to a trigger received at the `Pop` port when the register is empty. Inputs to this port must have the same built-in data type as inputs to the `Push` and `Rst` input ports.

Show empty stack indicator port

Enable the `Empty` output port, which is high (`1`) when the stack is empty, and low (`0`) otherwise.

Show full stack indicator port

Enable the `Full` output port, which is high (`1`) when the stack is full, and low (`0`) otherwise. The `Full` port remains low when you select **Dynamic reallocation** from the **Push full stack** parameter.

Show number of stack entries port

Enable the `Num` output port, which tracks the number of entries currently on the stack. When inputs to the `In` port are double-precision values, the outputs from the `Num` port are double-precision values. Otherwise, the outputs from the `Num` port are 32-bit unsigned integer values.

Show reset port to clear internal stack buffer

Enable the `Rst` input port, which empties the stack when the trigger specified by the **Trigger type** is received. Inputs to this port must have the same built-in data type as inputs to the `Push` and `Pop` input ports.

Clear output port on reset

Reset the `Out` port to zero (in addition to clearing the stack) when a trigger is received at the `Rst` input port.

Supported Data Types

Port	Supported Data Types
In	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Push	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers Inputs to this port must have the same built-in data type as inputs to the Pop and Rst input ports

<table>
<tr><th>Port</th><th>Supported Data Types</th></tr>
<tr><td>Pop</td><td>Double-precision floating point
Single-precision floating point
Fixed point (signed and unsigned)
Boolean
8-, 16-, and 32-bit signed integers
8-, 16-, and 32-bit unsigned integers
Inputs to this port must have the same built-in data type as inputs to the Push and Rst input ports.</td></tr>
<tr><td>Rst</td><td>Double-precision floating point
Single-precision floating point
Fixed point (signed and unsigned)
Boolean
8-, 16-, and 32-bit signed integers
8-, 16-, and 32-bit unsigned integers
Inputs to this port must have the same built-in data type as inputs to the Push and Pop input ports.</td></tr>
<tr><td>Out</td><td>Double-precision floating point
Single-precision floating point
Fixed point (signed and unsigned)
Boolean
8-, 16-, and 32-bit signed integers
8-, 16-, and 32-bit unsigned integers</td></tr>
</table>

Port	Supported Data Types
Empty	• Double-precision floating point • Boolean The block outputs Boolean values at this port when Boolean support is enabled, as described in "Effects of Enabling and Disabling Boolean Support". To learn how to disable Boolean output support, see "Steps to Disabling Boolean Support"
Full	• Double-precision floating point • Boolean The block outputs Boolean values at this port when Boolean support is enabled, as described in "Effects of Enabling and Disabling Boolean Support". To learn how to disable Boolean output support, see "Steps to Disabling Boolean Support"
Num	• Double-precision floating point The block outputs a double-precision floating-point value at this port when the data type of the In port is double-precision floating-point. • 32-bit unsigned integers The block outputs a 32-bit unsigned integer value at this port when the data type of the In port is anything other than double-precision floating-point.

See Also

Buffer	Signal Processing Blockset
Delay Line	Signal Processing Blockset
Queue	Signal Processing Blockset

Purpose

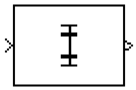
Find standard deviation of input or sequence of inputs

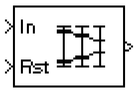
Library

Statistics

`dspstat3`

Description





The Standard Deviation block computes the standard deviation of each row or column of the input, along vectors of a specified dimension of the input, or of the entire input. The Standard Deviation block can also track the standard deviation of a sequence of inputs over a period of time. The **Running standard deviation** parameter selects between basic operation and running operation.

Basic Operation

When you do not select the **Running standard deviation** check box, the block computes the standard deviation of each row or column of the input, along vectors of a specified dimension of the input, or of the entire input at each individual sample time, and outputs the array y. Each element in y contains the standard deviation of the corresponding column, row, vector, or entire input. The output y depends on the setting of the **Find the standard deviation value over** parameter. For example, consider a 3-dimensional input signal of size M-by-N-by-P:

- `Entire input` — The output at each sample time is a scalar that contains the standard deviation of the entire input. In this mode, the output is always sample based.

  ```
  y = std(u(:))     % Equivalent MATLAB code
  ```

- `Each Row` — The output at each sample time consists of an M-by-1-by-P array, where each element contains the standard deviation of each vector over the second dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is an M-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.

  ```
  y = std(u,0,2)     % Equivalent MATLAB code
  ```

- `Each Column` — The output at each sample time consists of a 1-by-N-by-P array, where each element contains the standard deviation of each vector over the first dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is a 1-by-N row vector. In this mode, the frame status of the output is the same as that of the input.

  ```
  y = std(u,0,1)     % Equivalent MATLAB code
  ```

  For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors when the block is in this mode. Sample-based length-M row vector inputs are also treated as M-by-1 column vectors when the **Treat sample-based row input as a column** check box is selected.

- `Specified Dimension` — The output at each sample time depends on **Dimension**. If **Dimension** is set to `1`, the output is the same as when you select `Each column`. If **Dimension** is set to `2`, the output is the same as when you select `Each row`. If **Dimension** is set to `3`, the output at each sample time is an M-by-N matrix containing the standard deviation of each vector over the third dimension of the input. In this mode, the frame status of the output is the same as that of the input.

  ```
  y = std(u,0,Dimension)     % Equivalent MATLAB code
  ```

For purely real or purely imaginary inputs, the standard deviation of the jth column of an M-by-N input matrix is the square root of its variance:

$$y_j = \sigma_j = \sqrt{\frac{\sum_{i=1}^{M} \left| u_{ij} - \mu_j \right|^2}{M-1}} \qquad 1 \le j \le N$$

For complex inputs, the output is the *total standard deviation*, which equals the square root of the *total variance*, or the square root of the sum of the variances of the real and imaginary parts. The standard deviation of each column in an *M*-by-*N* input matrix is given by:

$$\sigma_j = \sqrt{\sigma_{j,\mathrm{Re}}^2 + \sigma_{j,\mathrm{Im}}^2}$$

Note The total standard deviation does *not* equal the sum of the real and imaginary standard deviations.

Running Operation

When you select the **Running standard deviation** check box, the block tracks the standard deviation of successive inputs to the block. For sample-based *M*-by-*N* inputs, the output is a sample-based *M*-by-*N* matrix with each element y_{ij} containing the standard deviation of element u_{ij} over all inputs since the last reset. For frame-based *M*-by-*N* inputs, the output is a frame-based *M*-by-*N* matrix with each element y_{ij} containing the standard deviation of the *j*th column over all inputs since the last reset, up to and including element u_{ij} of the current input.

N-D signals cannot be frame based. When the block is set to `Running` mode, each element of the N-D signal is treated as a separate channel.

There are $\prod d_i$ channels, where d_i is the size of the *i*th dimension.

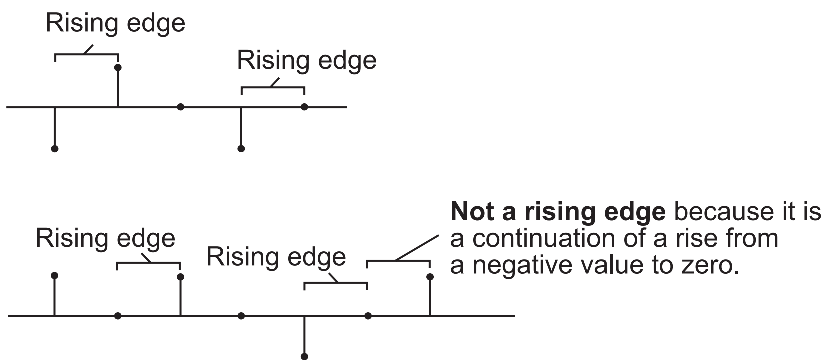
Resetting the Running Standard Deviation

The block resets the running standard deviation whenever a reset event is detected at the optional `Rst` port. The reset sample time must be a positive integer multiple of the input sample time.

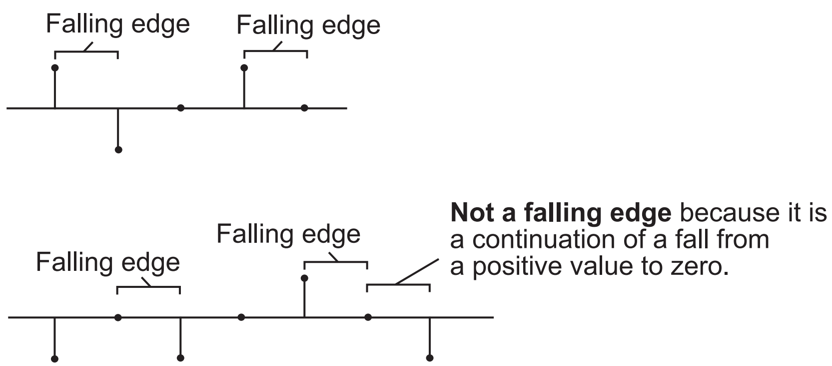
You specify the reset event in the **Reset port** parameter:

- `None` disables the `Rst` port.
- `Rising edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Rises from a negative value to a positive value or zero

- Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)



- `Falling edge` — Triggers a reset operation when the `Rst` input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)



- `Either edge` — Triggers a reset operation when the `Rst` input is a `Rising edge` or `Falling edge` (as described earlier)
- `Non-zero sample` — Triggers a reset operation at each sample time that the `Rst` input is not zero

---

Note When running simulations in the Simulink® MultiTasking mode, reset signals have a one-sample latency. Therefore, when the block detects a reset event, there is a one-sample delay at the reset port rate before the block applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

ROI Processing

To calculate the statistical value within a particular region of interest (ROI) of the input, select the **Enable ROI processing** check box. This option is only available when the **Find the standard deviation value over** parameter is set to `Entire input` and the **Running standard deviation** check box is not selected. ROI processing is only supported for 2-D inputs.

Note Full ROI processing is only available to users who have a Video and Image Processing Blockset™ license. If you only have a Signal Processing Blockset™ license, you can still use ROI processing, but are limited to the **ROI type** `Rectangles`.

Use the **ROI type** parameter to specify whether the ROI is a rectangle, line, label matrix, or binary mask. A binary mask is a binary image that enables you to specify which pixels to highlight, or select. In a label matrix, pixels equal to 0 represent the background, pixels equal to 1 represent the first object, pixels equal to 2 represent the second object, and so on. When the **ROI type** parameter is set to `Label matrix`, the Label and Label Numbers ports appear on the block. Use the Label

Numbers port to specify the objects in the label matrix for which the block calculates statistics. The input to this port must be a vector of scalar values that correspond to the labeled regions in the label matrix. For more information about the format of the input to the ROI port when the ROI is a rectangle or a line, see the Draw Shapes block reference page.

For rectangular ROIs, use the **ROI portion to process** parameter to specify whether to calculate the statistical value for the entire ROI or just the ROI perimeter.

Use the **Output** parameter to specify the block output. The block can output separate statistical values for each ROI or the statistical value for all specified ROIs. This parameter is not available if, for the **ROI type** parameter, you select `Binary mask`.

If, for the **ROI type** parameter, you select `Rectangles` or `Lines`, the **Output flag indicating if ROI is within image bounds** check box appears in the dialog box. If you select this check box, the Flag port appears on the block. The following tables describe the Flag port output based on the block parameters.

Output = Individual statistics for each ROI

Flag Port Output	Description
0	ROI is completely outside the input image.
1	ROI is completely or partially inside the input image.

Output = Single statistic for all ROIs

Flag Port Output	Description
0	All ROIs are completely outside the input image.
1	At least one ROI is completely or partially inside the input image.

If the ROI is partially outside the image, the block only computes the statistical values for the portion of the ROI that is within the image.

If, for the **ROI type** parameter, you select `Label matrix`, the **Output flag indicating if input label numbers are valid** check box appears in the dialog box. If you select this check box, the Flag port appears on the block. The following tables describe the Flag port output based on the block parameters.

Output = Individual statistics for each ROI

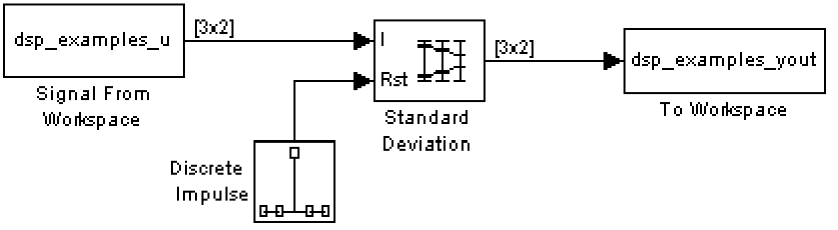
Flag Port Output	Description
0	Label number is not in the label matrix.
1	Label number is in the label matrix.

Output = Single statistic for all ROIs

Flag Port Output	Description
0	None of the label numbers are in the label matrix.
1	At least one of the label numbers is in the label matrix.

Examples

The Standard Deviation block in the next model calculates the running standard deviation of a frame-based 3-by-2 (two-channel) matrix input, `u`. The running standard deviation is reset at t=2 by an impulse to the block's `Rst` port.



The Standard Deviation block has the following settings:

- **Running standard deviation** = ☑
- **Reset port** = `Non-zero sample`

The Signal From Workspace block has the following settings:

- **Signal** = `dsp_examples_u`
- **Sample time** = `1/3`
- **Samples per frame** = `3`

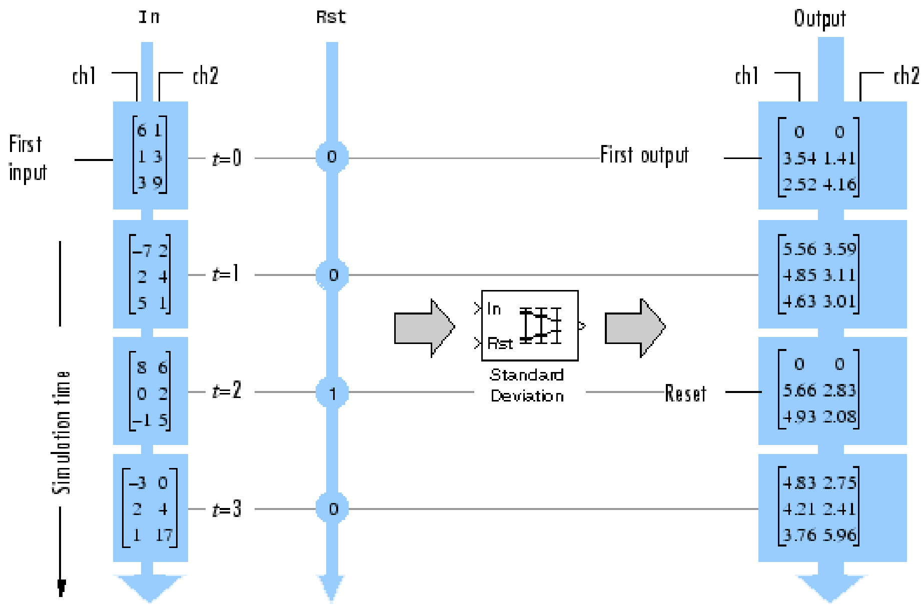
where

```
dsp_examples_u = [6 1 3 -7 2 5 8 0 -1 -3 2 1;1 3 9 2 4 1 6 2 5 0 4 17]'
```

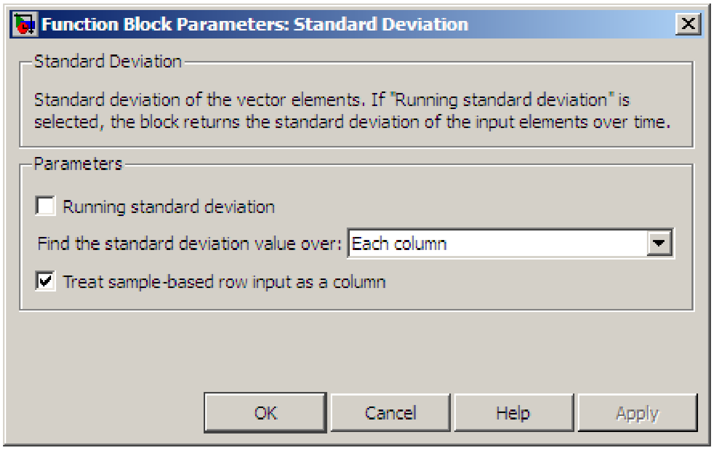
The Discrete Impulse block has the following settings:

- **Delay (samples)** = `2`
- **Sample time** = `1`
- **Samples per frame** = `1`

The block's operation is shown in the next figure.



Dialog Box



Running standard deviation

Enables running operation when selected.

Reset port

Determines the reset event that causes the block to reset the running standard deviation. The reset signal rate must be a positive integer multiple of the rate of the data signal input. This parameter is enabled only when you select **Running standard deviation**. For more information, see "Resetting the Running Standard Deviation" on page 2-1145.

Find the standard deviation value over

Specify whether to find the standard deviation value along rows, columns, entire input, or the dimension specified in the **Dimension** parameter. For more information, see "Basic Operation" on page 2-1143.

Treat sample-based row input as a column

Select to treat sample-based length-M row vector inputs as M-by-1 column vectors. This parameter is only visible when the **Find the standard deviation value over** parameter is set to `Each column`.

Dimension

Specify the dimension (one-based value) of the input signal, over which the standard deviation is computed. The value of this parameter cannot exceed the number of dimensions in the input signal. This parameter is only visible when the **Find the standard deviation value over** parameter is set to `Specified dimension`.

Enable ROI Processing

Select this check box to calculate the statistical value within a particular region of each image. This parameter is only available when the **Find the standard deviation value over** parameter is set to `Entire input`, and the block is not in running mode.

Note Full ROI processing is only available to users who have a Video and Image Processing Blockset license. If you only have a Signal Processing Blockset license, you can still use ROI processing, but are limited to the **ROI type** `Rectangles`.

ROI type
Specify the type of ROI you want to use. Your choices are `Rectangles`, `Lines`, `Label matrix`, or `Binary mask`.

ROI portion to process
Specify whether you want to calculate the statistical value for the entire ROI or just the ROI perimeter. This parameter is only visible if, for the **ROI type** parameter, you specify `Rectangles`.

Output
Specify the block output. The block can output a vector of separate statistical values for each ROI or a scalar value that represents the statistical value for all the specified ROIs. This parameter is not available if, for the **ROI type** parameter, you select `Binary mask`.

Output flag indicating if ROI is within image bounds
If you select this check box, the Flag port appears on the block. For a description of the Flag port output, see the tables in "ROI Processing" on page 2-1147. This parameter is visible if, for the **ROI type** parameter, you select `Rectangles` or `Lines`.

Output flag indicating if label numbers are valid
If you select this check box, the Flag port appears on the block. For a description of the Flag port output, see the tables in "ROI Processing" on page 2-1147. This parameter is visible if, for the **ROI type** parameter, you select `Label matrix`.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
Reset	• Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
ROI	Rectangles and lines: • Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers Binary Mask: • Boolean
Label	• 8-, 16-, and 32-bit unsigned integers
Label Numbers	• 8-, 16-, and 32-bit unsigned integers
Output	• Double-precision floating point • Single-precision floating point
Flag	• Boolean

See Also

Mean	Signal Processing Blockset
RMS	Signal Processing Blockset
Variance	Signal Processing Blockset
`std`	MATLAB

Submatrix

Purpose

Select subset of elements (submatrix) from matrix input

Library

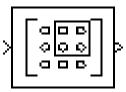
- Math Functions / Matrices and Linear Algebra / Matrix Operations

  `dspmtrx3`

- Signal Management / Indexing

  `dspindex`

Description



The Submatrix block extracts a contiguous submatrix from the M-by-N input matrix `u`. A length-M 1-D vector input is treated as an M-by-1 matrix. The **Row span** parameter provides three options for specifying the range of rows in `u` to be retained in submatrix output `y`:

- `All rows`

  Specifies that `y` contains all M rows of `u`.

- `One row`

  Specifies that `y` contains only one row from `u`. The **Starting row** parameter (described below) is enabled to allow selection of the desired row.

- `Range of rows`

  Specifies that `y` contains one or more rows from `u`. The **Row** and **Ending row** parameters (described below) are enabled to allow selection of the desired range of rows.

The **Column span** parameter contains a corresponding set of three options for specifying the range of columns in `u` to be retained in submatrix `y`: `All columns`, `One column`, or `Range of columns`. The `One column` option enables the **Column** parameter, and `Range of columns` options enable the **Starting column** and **Ending column** parameters.

The output has the same frame status as the input.

Range Specification Options

When you select `One row` or `Range of rows` from the **Row span** parameter, you specify the desired row or range of rows in the **Row** parameter, or the **Starting row** and **Ending row** parameters. Similarly, when you select `One column` or `Range of columns` from the **Column span** parameter, you specify the desired column or range of columns in the **Column** parameter, or the **Starting column** and **Ending column** parameters.

The **Row**, **Column**, **Starting row** or **Starting column** can be specified in six ways:

- `First`

  For rows, this specifies that the first row of `u` should be used as the first row of `y`. When all columns are to be included, this is equivalent to `y(1,:) = u(1,:)`.

  For columns, this specifies that the first column of `u` should be used as the first column of `y`. When all rows are to be included, this is equivalent to `y(:,1) = u(:,1)`.

- `Index`

  For rows, this specifies that the row of `u`, `firstrow`, forward-indexed by the **Row index** parameter or the **Starting row index** parameter, should be used as the first row of `y`. When all columns are to be included, this is equivalent to `y(1,:) = u(firstrow,:)`.

  For columns, this specifies that the column of `u`, forward-indexed by the **Column index** parameter or the **Starting column index** parameter, `firstcol`, should be used as the first column of `y`. When all rows are to be included, this is equivalent to `y(:,1) = u(:,firstcol)`.

- `Offset from last`

  For rows, this specifies that the row of `u` offset from row M by the **Row offset** or **Starting row offset** parameter, `firstrow`, should be used as the first row of y. When all columns are to be included, this is equivalent to `y(1,:)  = u(M-firstrow,:)`.

For columns, this specifies that the column of u offset from column N by the **Column offset** or **Starting column offset** parameter, `firstcol`, should be used as the first column of y. When all rows are to be included, this is equivalent to `y(:,1) = u(:,N-firstcol)`.

- `Last`

  For rows, this specifies that the last row of u should be used as the only row of y. When all columns are to be included, this is equivalent to `y = u(M,:)`.

  For columns, this specifies that the last column of u should be used as the only column of y. When all rows are to be included, this is equivalent to `y = u(:,N)`.

- `Offset from middle`

  For rows, this specifies that the row of u offset from row M/2 by the **Starting row offset** parameter, `firstrow`, should be used as the first row of y. When all columns are to be included, this is equivalent to `y(1,:)  = u(M/2-firstrow,:)`.

  For columns, this specifies that the column of u offset from column N/2 by the **Starting column offset** parameter, `firstcol`, should be used as the first column of y. When all rows are to be included, this is equivalent to `y(:,1) = u(:,N/2-firstcol)`.

- `Middle`

  For rows, this specifies that the middle row of u should be used as the only row of y. When all columns are to be included, this is equivalent to `y = u(M/2,:)`.

  For columns, this specifies that the middle column of u should be used as the only column of y. When all rows are to be included, this is equivalent to `y = u(:,N/2)`.

The **Ending row** or **Ending column** can similarly be specified in five ways:

- `Index`

  For rows, this specifies that the row of `u` forward-indexed by the **Ending row index** parameter, `lastrow`, should be used as the last row of `y`. When all columns are to be included, this is equivalent to `y(end,:)  = u(lastrow,:)`.

  For columns, this specifies that the column of `u` forward-indexed by the **Ending column index** parameter, `lastcol`, should be used as the last column of `y`. When all rows are to be included, this is equivalent to `y(:,end) = u(:,lastcol)`.

- `Offset from last`

  For rows, this specifies that the row of `u` offset from row M by the **Ending row offset** parameter, `lastrow`, should be used as the last row of `y`. When all columns are to be included, this is equivalent to `y(end,:)  = u(M-lastrow,:)`.

  For columns, this specifies that the column of `u` offset from column N by the **Ending column offset** parameter, `lastcol`, should be used as the last column of `y`. When all rows are to be included, this is equivalent to `y(:,end) = u(:,N-lastcol)`.

- `Last`

  For rows, this specifies that the last row of `u` should be used as the last row of `y`. When all columns are to be included, this is equivalent to `y(end,:)  = u(M,:)`.

  For columns, this specifies that the last column of `u` should be used as the last column of `y`. When all rows are to be included, this is equivalent to `y(:,end) = u(:,N)`.

- `Offset from middle`

  For rows, this specifies that the row of `u` offset from row M/2 by the **Ending row offset** parameter, `lastrow`, should be used as the last row of `y`. When all columns are to be included, this is equivalent to `y(end,:)  = u(M/2-lastrow,:)`.

For columns, this specifies that the column of `u` offset from column N/2 by the **Ending column offset** parameter, `lastcol`, should be used as the last column of `y`. When all rows are to be included, this is equivalent to `y(:,end) = u(:,N/2-lastcol)`.

- `Middle`

  For rows, this specifies that the middle row of `u` should be used as the last row of `y`. When all columns are to be included, this is equivalent to `y(end,:)  = u(M/2,:)`.

  For columns, this specifies that the middle column of `u` should be used as the last column of `y`. When all rows are to be included, this is equivalent to `y(:,end) = u(:,N/2)`.

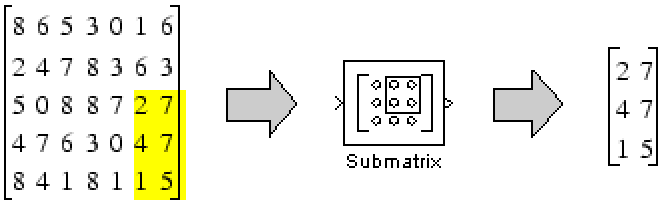
This block supports Simulink® virtual buses.

Examples

To extract the lower-right 3-by-2 submatrix from a 5-by-7 input matrix, enter the following set of parameters:

- **Row span** = `Range of rows`
- **Starting row** = `Index`
- **Starting row index** = `3`
- **Ending row** = `Last`
- **Column span** = `Range of columns`
- **Starting column** = `Offset from last`
- **Starting column offset** = `1`
- **Ending column** = `Last`

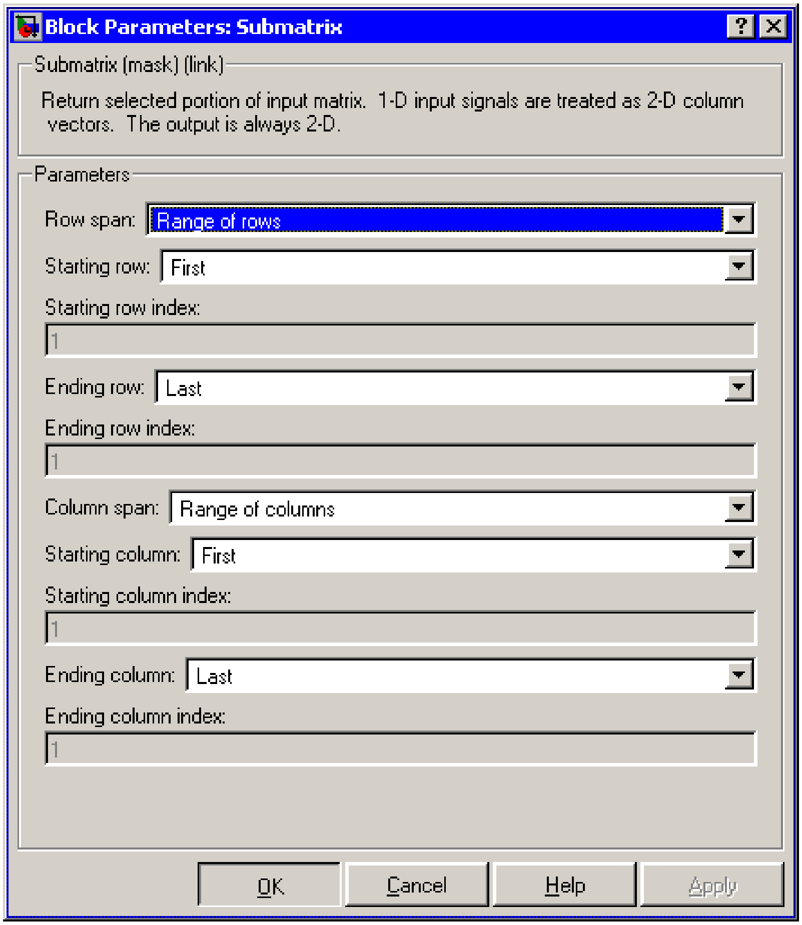
The figure below shows the operation for a 5-by-7 matrix with random integer elements, `randint(5,7,10)`.



There are often several possible parameter combinations that select the *same* submatrix from the input. For example, instead of specifying `Last` for **Ending column**, you could select the same submatrix by specifying

- **Ending column** = `Index`
- **Ending column index** = `7`

Dialog Box



The parameters displayed in the dialog box vary for different menu combinations. Only some of the parameters listed below are visible in the dialog box at any one time.

Row span

The range of input rows to be retained in the output. Options are `All rows`, `One row`, or `Range of rows`.

Row/Starting row
The input row to be used as the first row of the output. **Row** is enabled when you select `One row` from **Row span**, and **Starting row** when you select `Range of rows` from **Row span**.

Row index/Starting row index
The index of the input row to be used as the first row of the output. **Row index** is enabled when you select `Index` from Row, and **Starting row index** when you select `Index` from **Starting row**.

Row offset/Starting row offset
The offset of the input row to be used as the first row of the output. **Row offset** is enabled when you select `Offset from middle` or `Offset from last` from **Row**, and Starting row offset is enabled when you select `Offset from middle` or `Offset from last` from **Starting row**.

Ending row
The input row to be used as the last row of the output. This parameter is enabled when you select `Range of rows` from **Row span** and you select any option but `Last` from **Starting row**.

Ending row index
The index of the input row to be used as the last row of the output. This parameter is enabled when you select `Index` from **Ending row**.

Ending row offset
The offset of the input row to be used as the last row of the output. This parameter is enabled when you select `Offset from middle` or `Offset from last` from **Ending row**.

Column span
The range of input columns to be retained in the output. Options are `All columns`, `One column`, or `Range of columns`.

Column/Starting column
The input column to be used as the first column of the output. **Column** is enabled when you select `One column` from **Column**

span, and **Starting column** is enabled when you select `Range of columns` from **Column span**.

Column index/Starting column index
The index of the input column to be used as the first column of the output. **Column index** is enabled when you select `Index` from Column, and **Starting column index** is enabled when you select `Index` from **Starting column**.

Column offset/Starting column offset
The offset of the input column to be used as the first column of the output. **Column offset** is enabled when you select `Offset from middle` or `Offset from last` from Column. **Starting column offset** is enabled when you select `Offset from middle` or `Offset from last` from **Starting column**.

Ending column
The input column to be used as the last column of the output. This parameter is enabled when you select `Range of columns` from **Column span** and you select any option but `Last` from **Starting column**.

Ending column index
The index of the input column to be used as the last column of the output. This parameter is enabled when you select `Index` from **Ending column**.

Ending column offset
The offset of the input column to be used as the last column of the output. This parameter is enabled when you select `Offset from middle` or `Offset from last` from **Ending column**.

Supported Data Types

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

See Also

Reshape	Simulink
Selector	Simulink
Variable Selector	Signal Processing Blockset
`reshape`	MATLAB

See “Splitting Multichannel Sample-Based Signals into Several Multichannel Signals” for related information.

SVD Solver

Purpose

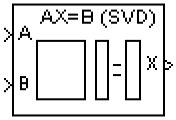
Solve AX=B using singular value decomposition

Library

Math Functions / Matrices and Linear Algebra / Linear System Solvers

`dspsolvers`

Description

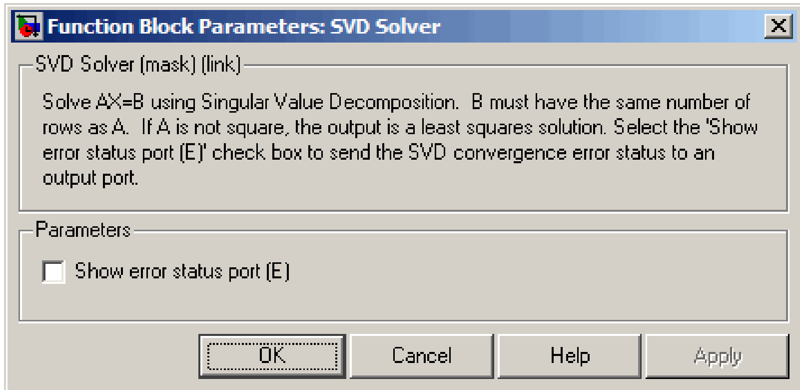


The SVD Solver block solves the linear system AX=B, which can be overdetermined, underdetermined, or exactly determined. The system is solved by applying singular value decomposition (SVD) factorization to the M-by-N matrix A, at the A port. The input to the B port is the right side M-by-L matrix, B. A length-M 1-D vector input at either port is treated as an M-by-1 matrix.

The output at the X port is the N-by-L matrix, X. X is always sample based, and is chosen to minimize the sum of the squares of the elements of B-AX (the residual). When B is a vector, this solution minimizes the vector 2-norm of the residual. When B is a matrix, this solution minimizes the matrix Frobenius norm of the residual. In this case, the columns of X are the solutions to the L corresponding systems AX_k=B_k, where B_k is the kth column of B, and X_k is the kth column of X.

X is known as the minimum-norm-residual solution to AX=B. The minimum-norm-residual solution is unique for overdetermined and exactly determined linear systems, but it is not unique for underdetermined linear systems. Thus when the SVD Solver block is applied to an underdetermined system, the output X is chosen such that the number of nonzero entries in X is minimized.

Dialog Box



Show error status port

Select to enable the E output port, which reports a failure to converge. The possible values you can receive on the port are:

- `0` — The singular value decomposition calculation converges.
- `1` — The singular value decomposition calculation does not converge.

If the singular value decomposition calculation fails to converge, the output at port X is an undefined matrix of the correct size.

Supported Data Types

Port	Supported Data Types
A	• Double-precision floating point • Single-precision floating point
B	• Double-precision floating point • Single-precision floating point
X	• Double-precision floating point • Single-precision floating point
E	• Boolean

See Also

Autocorrelation LPC	Signal Processing Blockset
Cholesky Solver	Signal Processing Blockset
LDL Solver	Signal Processing Blockset
Levinson-Durbin	Signal Processing Blockset
LU Inverse	Signal Processing Blockset
Pseudoinverse	Signal Processing Blockset
QR Solver	Signal Processing Blockset
Singular Value Decomposition	Signal Processing Blockset

See “Linear System Solvers” for related information.

Purpose Display signals generated during simulation

Library Signal Processing Sinks

`dspsnks4`

Description Refer to the Simulink® Scope reference page for more information.

Time-Varying Direct-Form II Transpose Filter

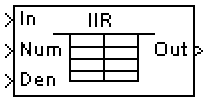
Purpose

Apply variable IIR filter to input

Library

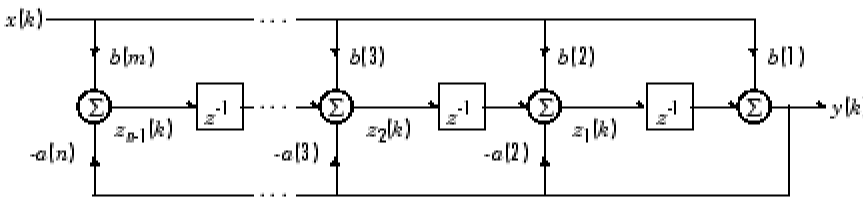
`dspobslib`

Description



Note The Time-Varying Direct-Form II Transpose Filter block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Time-Varying Direct-Form II Transpose Filter block is a version of the Direct-Form II Transpose Filter block whose filter coefficients can be updated during the simulation. The block applies a direct-form II transposed IIR filter to the top input (`In`).



This is a canonical form that has the minimum number of delay elements. The filter order is `max(m,n)-1`.

An M-by-N sample-based matrix input is treated as M*N independent channels, and an M-by-N frame-based matrix input is treated as N independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

The block's two lower inputs (`Num` and `Den`) specify the filter's transfer function,

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2 z^{-1} + \ldots + b_{m+1} z^{-(m-1)}}{a_1 + a_2 z^{-1} + \ldots + a_{n+1} z^{-(n-1)}}$$

By default the filter coefficients are normalized by a_1. To prevent normalization by a_1, deselect the **Support non-normalized filters** check box.

Filter Type

The **Filter type** parameter specifies whether the filter is an all-zero (FIR or MA) filter, all-pole (AR) filter, or pole-zero (IIR or ARMA) filter:

- **Pole-zero**

  The block accepts inputs for both the numerator (`Num`) and denominator (`Den`) vectors.

  Input `Num` is a vector of numerator coefficients,

  ```
  [b(1) b(2) ... b(m)]
  ```

  and input `Den` is a vector of denominator coefficients,

  ```
  [a(1) a(2) ... a(n)]
  ```

- **All-zero**

  The block accepts only the numerator vector (`Num`). The denominator of the all-zero filter is 1.

- **All-pole**

  The block accepts only the denominator vector (`Den`). The numerator of the all-pole filter is 1.

For any of these designs, the coefficient vector inputs can change over time to alter the filter's response characteristics during the simulation.

Initial Conditions

In its default form, the filter initializes the internal filter states to zero, which is equivalent to assuming past inputs and outputs are zero. The block also accepts optional nonzero initial conditions for the filter delays. Note that the number of filter states (delay elements) per input channel is

```
max(m,n)-1
```

The **Initial conditions** parameter may take one of four forms:

- Empty matrix

  The empty matrix, `[]`, causes a zero (0) initial condition to be applied to all delay elements in each filter channel.

- Scalar

  The scalar value is copied to all delay elements in each filter channel. Note that a value of zero is equivalent to setting the **Initial conditions** parameter to the empty matrix, `[]`.

- Vector

  The vector has a length equal to the number of delay elements in each filter channel, `max(m,n)-1`, and specifies a unique initial condition for each delay element in the filter channel. This vector of initial conditions is applied to each filter channel.

- Matrix

  The matrix specifies a unique initial condition for each delay element, and can specify different initial conditions for each filter channel. The matrix must have the same number of rows as the number of delay elements in the filter, `max(m,n)-1`, and must have one column per filter channel.

Filter Update Rate

In frame-based operation, the **Filter update rate** parameter determines how frequently the block updates the filter coefficients (i.e., how often it checks the Num and Den inputs). There are two available options:

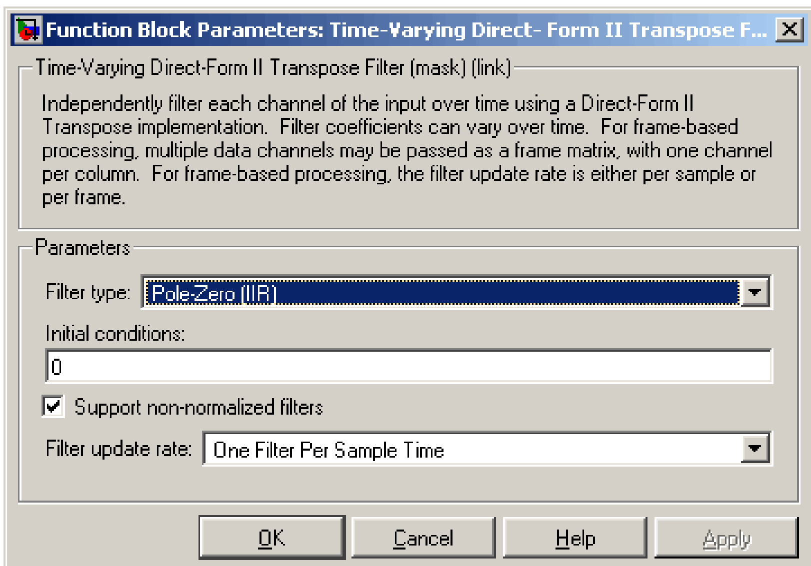
- **One filter per sample time**

  The block updates the filter coefficients (from inputs Num and Den) for each individual scalar sample in the frame-based input. This means that each output sample could potentially be computed by a different filter (assuming that Num and Den inputs are updated frequently enough).

- **One filter per frame time**

  The block updates the filter coefficients (from inputs Num and Den) for each new input frame, rather than at each sample in the frame. This means that each output sample in a given frame is a result of an identical filtering process.

Dialog Box



Filter type

The type of filter to apply: **Pole-Zero (IIR)**, **All-Zero (FIR)**, or **All-Pole (AR)**. The `Num` and `Den` input ports are enabled or disabled as appropriate.

Initial conditions

The filter's initial conditions, a scalar, vector, or matrix.

Support non-normalized filters

Normalizes the filter by a_1 when selected.

Filter update rate

The frequency with which the block updates the filter coefficients; once per sample, or once per frame.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

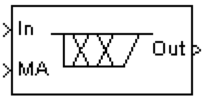
Purpose

Apply variable lattice filter to input

Library

`dspobslib`

Description



Note The Time-Varying Lattice Filter block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Time-Varying Lattice Filter block applies a moving average (MA) or autoregressive (AR) lattice filter to the top input (`In`). The filter reflection coefficients are specified by the vector input to the `MA` or `AR` port, and can vary with time.

An M-by-N sample-based matrix input to the `In` port is treated as M*N independent channels, and an M-by-N frame-based matrix input is treated as N independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

Filter Type

The **Filter type** parameter specifies whether the filter is an all-zero (FIR or MA) filter or all-pole (AR) filter.

- **All-zero**

  The block constructs an nth order MA filter using the n reflection coefficients contained in the vector input to the `MA` port.

  ```
  k = [k(1) k(2) ... k(n)]
  ```

- **All-pole**

  The block constructs an nth order AR filter using the n reflection coefficients contained in the vector input to the `AR` port.

  ```
  k = [k(1) k(2) ... k(n)]
  ```

For both designs, the coefficient vector inputs can change over time to alter the filter's response characteristics during the simulation.

Initial Conditions

In its default form, the filter initializes the internal filter states to zero, which is equivalent to assuming past inputs and outputs are zero. The block also accepts optional nonzero initial conditions for the filter delays. Note that the number of filter states (delay elements) per input channel is

```
length(k)
```

The **Initial conditions** parameter may take one of four forms:

- Empty matrix

  The empty matrix, `[]`, causes a zero (0) initial condition to be applied to all delay elements in each filter channel.

- Scalar

  The scalar value is copied to all delay elements in each filter channel. Note that a value of zero is equivalent to setting the **Initial conditions** parameter to the empty matrix.

- Vector

  The vector has a length equal to the number of delay elements in each filter channel, `length(k)`, and specifies a unique initial condition for each delay element in the filter channel. This vector of initial conditions is applied to each filter channel.

- Matrix

  The matrix specifies a unique initial condition for each delay element, and can specify different initial conditions for each filter channel. The matrix must have the same number of rows as the number of delay elements in the filter, length(k), and must have one column per filter channel.

Filter Update Rate

In frame-based operation, the **Filter update rate** parameter determines how frequently the block updates the filter coefficients (i.e., how often it checks the MA or AR input). There are two available options:

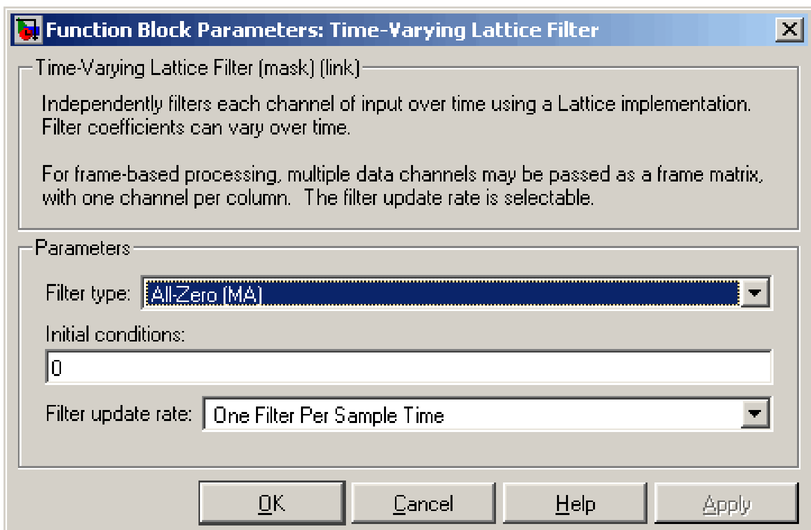
- **One filter per sample time**

  The block updates the filter coefficients (from input MA or AR) for each individual scalar sample in the framed input. This means that each output sample could potentially be computed by a different filter (assuming that the MA or AR input is updated frequently enough).

- **One filter per frame time**

  The block updates the filter coefficients (from input MA or AR) for each new input frame, rather than at each sample in the frame. This means that each output sample in a given frame is a result of an identical filtering process.

Dialog Box



Filter type

The type of filter to apply: MA or AR. The `MA` or `AR` input port is enabled or disabled appropriately.

Initial conditions

The filter's initial conditions.

Filter update rate

The frequency with which the block updates the filter coefficients; once per sample, or once per frame.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

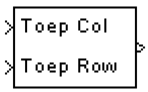
Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

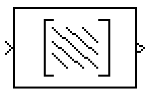
Purpose Generate matrix with Toeplitz symmetry

Library Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description





The Toeplitz block generates a Toeplitz matrix from inputs defining the first column and first row. The top input (`Col`) is a vector containing the values to be placed in the first *column* of the matrix, and the bottom input (`Row`) is a vector containing the values to be placed in the first *row* of the matrix.

```
y = toeplitz(Col,Row)       % Equivalent MATLAB code
```

The other elements of the matrix obey the relationship

```
y(i,j) = y(i-1,j-1)
```

and the output has dimension `[length(Col) length(Row)]`. The y(1,1) element is inherited from the `Col` input. For example, the following inputs

```
Col = [1 2 3 4 5]
Row = [7 7 3 3 2 1 3]
```

produce the Toeplitz matrix

$$\begin{bmatrix} 1 & 7 & 3 & 3 & 2 & 1 & 3 \\ 2 & 1 & 7 & 3 & 3 & 2 & 1 \\ 3 & 2 & 1 & 7 & 3 & 3 & 2 \\ 4 & 3 & 2 & 1 & 7 & 3 & 3 \\ 5 & 4 & 3 & 2 & 1 & 7 & 3 \end{bmatrix}$$

When both of the inputs are sample based, the output is sample based. Otherwise, the output is frame based.

When you select the **Symmetric** check box, the block generates a symmetric (Hermitian) Toeplitz matrix from a single input, u, defining both the first row and first column of the matrix.

```
y = toeplitz(u)      % Equivalent MATLAB code
```

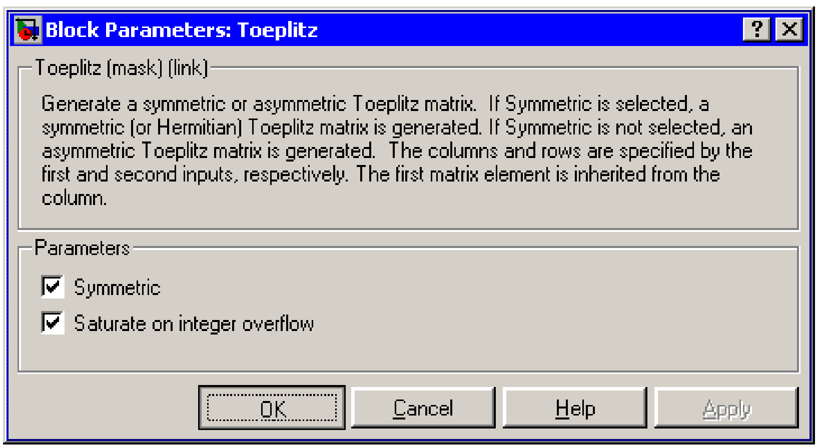
The output has dimension `[length(u) length(u)]`. For example, the Toeplitz matrix generated from the input vector `[1 2 3 4]` is

$$\begin{bmatrix} 1 & 2 & 3 & 4 \\ 2 & 1 & 2 & 3 \\ 3 & 2 & 1 & 2 \\ 4 & 3 & 2 & 1 \end{bmatrix}$$

The output has the same frame status as the input.

The Toeplitz block supports real and complex floating-point and fixed-point inputs.

Dialog Box



Symmetric
When selected, enables the single-input configuration for symmetric Toeplitz matrix output.

Saturate on integer overflow

When you generate a symmetric Toeplitz matrix with this block, if the input vector is complex, the output is a symmetric Hermitian matrix whose elements satisfy the relationship

$$y(i,j) = \mathrm{conj}(y(j,i))$$

For fixed-point signals the conjugate operation could result in an overflow. When you select this parameter, overflows saturate. This parameter is only visible with the **Symmetric** parameter is selected. This parameter is ignored for floating-point signals.

Supported Data Types

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

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

See Also

Constant Diagonal Matrix	Signal Processing Blockset
`toeplitz`	MATLAB

Purpose

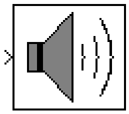
Write audio data to computer's audio device

Library

Signal Processing Sinks

`dspsnks4`

Description



The To Audio Device block sends audio data to your computer's audio device. This block has the following limitations:

- Not supported for use with the Simulink® Model block.
- Not currently supported on Solaris™ platforms.

Use the **Device** parameter to specify the device to which you want to send the audio data. This parameter is automatically populated based on the audio devices installed on your system. If you plug or unplug an audio device from your system, type `clear mex` at the MATLAB® command prompt to update the list.

Select the **Inherit sample rate from input** check box if you want the block to inherit the sample rate of the audio signal from the input to the block. If you clear this check box, the **Sample rate (Hz)** parameter appears on the block. Use this parameter to specify the number of samples per second in the signal.

Use the **Device data type** to specify the data type of the audio data that is sent to the device. You can choose:

- `8-bit integer`
- `16-bit integer`
- `24-bit integer`
- `32-bit float`
- `Determine from input data type`

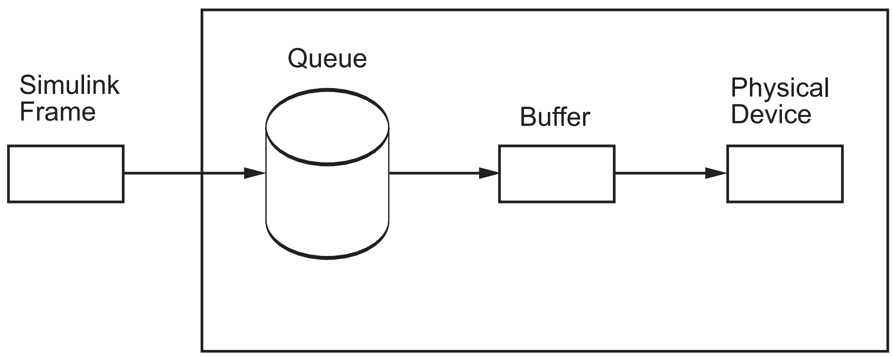
If you choose `Determine from input data type`, the following table summarizes the block's behavior.

Input Data Type	Device Data Type
Double-precision floating point or single-precision floating point	32-bit floating point
32-bit integer	24-bit integer
16-bit integer	16-bit integer
8-bit integer	8-bit integer

If you choose `Determine from input data type` and the device does not support the input data type, the block uses the next lowest-precision data type supported by the device.

Buffering

The To Audio Device block buffers the data from a Simulink signal using the process illustrated by the following figure.



To Audio Device Block

1. At the start of the simulation, the queue is filled with silence. Specify the size of this queue using the **Queue duration (seconds)** parameter. As Simulink runs, the block appends Simulink frames to the bottom of the queue.

2. At each time step, the blocks sends a buffer of samples from the top of the queue to the audio device. Select the **Automatically determine buffer size** check box to allow the block to use a conservative buffer size. See the From Audio Device block reference page for the equation the block uses to calculate this buffer size. If you clear this check box, the **Buffer size (samples)** parameter appears on the block. Use this parameter to specify the size of the buffer in samples.

3. The block writes the buffer of audio data to the device. If the queue did not contain enough data to completely fill the buffer, the block fills the remaining portion of the buffer with zeros. This data has a the data type specified by the **Device data type** parameter.

When the simulation throughput rate is lower than the hardware throughput rate, the queue, which is initially full, becomes empty. If the queue is empty, the block sends zeros (silence) to the audio device. When the simulation throughput rate is higher than the hardware throughput rate, the To Audio Device block waits to write data to the queue.

To minimize the chance of dropouts, the block checks to make sure the queue duration is at least as large as the maximum of the buffer size and the frame size. If it is not, the queue duration is automatically set to this maximum value.

Troubleshooting

When Simulink cannot keep up with an audio device that is operating in real time, the queue becomes empty and gaps occur in the audio data that the block sends to the device. Here are several ways to deal with this situation:

- *Increase the queue duration*.

  The **Queue duration (seconds)** parameter specifies the duration of the signal, in seconds, that can be buffered during the simulation. This is the maximum length of time that the block's data supply can lag the hardware's data demand.

- *Increase the buffer size*.

The size of the buffer processed in each interrupt from the audio device affects the performance of your model. If the buffer is too small, a large portion of hardware resources are used to write data to the device. If the buffer is too big, Simulink must wait for the device to empty the buffer before it can write the data to the queue, which introduces latency.

- *Increase the simulation throughput rate*.

  Two useful methods for improving simulation throughput rates are increasing the signal frame size and compiling the simulation into native code:

  - Increase frame sizes and convert sample-based signals to frame-based signals throughout the model to reduce the amount of block-to-block communication overhead. This can increase throughput rates in many cases. However, larger frame sizes generally result in greater model latency due to initial buffering operations.
  - Generate executable code with Real-Time Workshop® code generation software. Native code runs much faster than Simulink and should provide rates adequate for real-time audio processing.

Other ways to improve throughput rates include simplifying the model and running the simulation on a faster PC processor. For other ideas on improving simulation performance, see "Delay and Latency" and "Improving Simulation Performance and Accuracy" in the Simulink documentation.

Channel-to-Speaker Mapping on Windows® Operating Systems

The To Audio Device and From Audio Device blocks can support multiple channels. On Windows operating systems, the channel-to-speaker mapping is defined as listed below. This mapping only applies when your sound card is properly configured and capable of receiving the audio data you send. If the number of channels on the card does not match the number of channels on the block, or if you specify a data type for the **Device data type** parameter that is not supported by your device, the Windows mixer intervenes to translate from one format to another. If the Windows mixer does intervene, the channel-to-speaker mapping might differ from what is specified here.

- Single channel input — Front center speaker

  On systems with two speakers, the front center channel is split between the right and left speakers.

- Multichannel input — Channels are assigned to speakers as follows:
  - One channel — Front center
  - Two channels — Front left, front right
  - Four channels — Front left, front right, rear left, rear right
  - Six channels — Front left, front right, front center, low frequency, rear left, rear right
  - Eight channels — Front left, front right, front center, low frequency, rear left, rear right, front left center, front right center
  - For all other channel combinations, the channel assignment is dictated by the audio card.

Audio Hardware API

The To Audio Device and From Audio Device blocks use the open-source PortAudio library in order to communicate with the audio hardware on a given computer. The PortAudio library supports a range of API's designed to communicate with the audio hardware on a given platform. The following API choices were made when building the PortAudio library for the Signal Processing Blockset™ product:

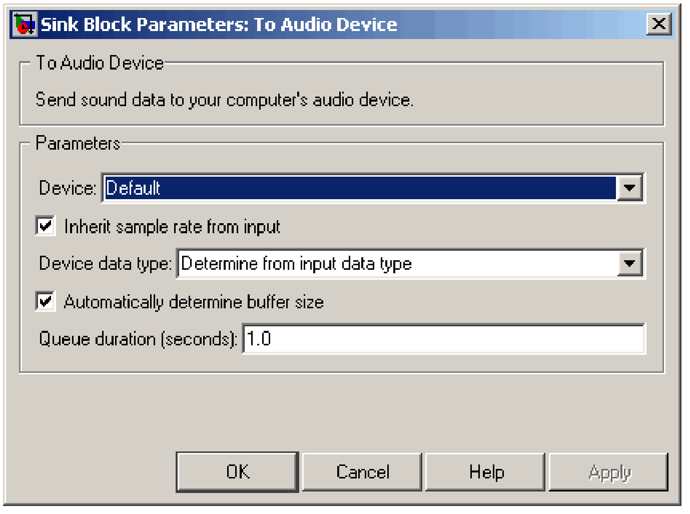
- Windows: DirectSound
- Linux: OSS
- Mac: CoreAudio

If you are interested in using a different audio API, such as ASIO (Windows) or ALSA (Linux®) please search for PortAudio on the Matlab Central website.

Example

See the Positional Audio demo for an example of how to use this block. You can open this demo by typing `dspAudioPos` at the MATLAB command line.

Dialog Box



Device
: Specify which device to send the audio data to.

Inherit sample rate from input
: Select this check box if you want the block to inherit the sample rate of the audio signal from the input to the block.

Sample rate (Hz)
: Specify the number of samples per second in the signal. This parameter is visible when the **Inherit sample rate from input** check box is cleared.

Device data type
Specify the data type of the audio data sent to the device.

Automatically determine buffer size
Select this check box to allow the block to calculate a conservative buffer size.

Buffer size (samples)
Specify the size of the buffer. This parameter is visible when the **Automatically determine buffer size** check box is cleared.

Queue duration (seconds)
Specify the size of the queue in seconds.

Supported Data Types

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

See Also

From Audio Device	Signal Processing Blockset
To Wave File	Signal Processing Blockset
`audioplayer`	MATLAB
`sound`	MATLAB

Purpose

Write video frames and/or audio samples to multimedia file

Library

Signal Processing Sinks

`dspsnks4`

Description

The To Multimedia File block writes video frames and/or audio samples to a multimedia (`.avi`) file. Video processing requires the Video and Image Processing Blockset™ product.

You can also compress the video frames or audio samples by selecting a compression algorithm. You can connect as many of the input ports as you want. Therefore, you can control what type of video and/or audio is sent to the multimedia file.

Note This block supports code generation for platforms that have file I/O available. This excludes RTWin (Real-Time Windows Target™ software, which does not support file I/O). This block performs best on platforms with Version 9.0 or later of DirectX® software and Version 9.0 or later of Windows Media® software. On UNIX® and Linux® platforms, this block supports only uncompressed RGB24 AVI files whose size is less than 2 GB.

Port	Input	Supported Data Types	Supports Complex Values?
Image	M-by-N-by-3 matrix RGB signal. To record M-by-N intensity video, use the Matrix Concatenate block to create an RGB signal.	• Double-precision floating point • Single-precision floating point • Boolean • 8-, 16- 32-bit signed integers • 8-, 16- 32-bit unsigned integers	No
R, G, B	Matrix that represents one plane of the RGB video stream. Inputs to the R, G, or B port must have the same dimensions and data type.	Same as Image port	No
Audio	Vector of audio data	• Double-precision floating point • Single-precision floating point • 16-bit signed integers • 8-bit unsigned integers	No

For the block to display video data properly, double- and single-precision floating-point pixel values must be between 0 and 1. For any other data type, the pixel values must be between the minimum and maximum values supported by their data type.

Use the **Output file name** parameter to specify the name of the multimedia file to which to write. This file is saved in your current directory. To specify a different file, click the **Browse** button, and then navigate to the new file.

Use the **Write** parameter to specify whether the block writes video frames and/or audio samples to the multimedia file. The choices are `Video and audio`, `Video only`, or `Audio only`.

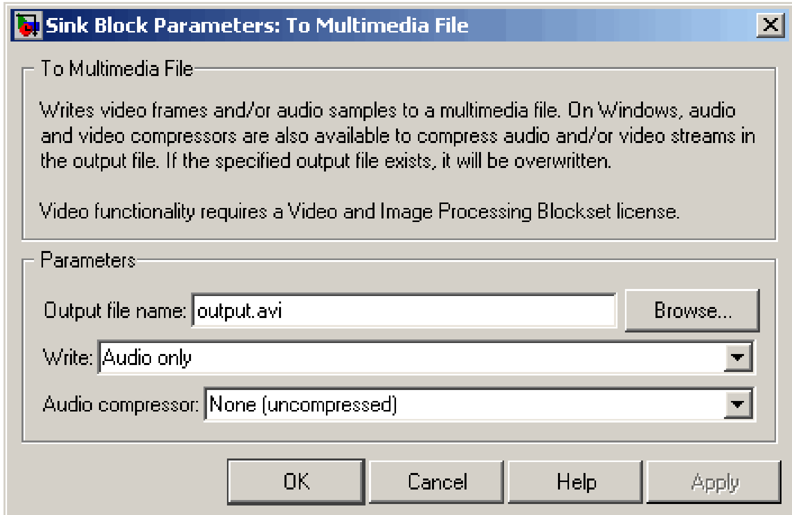
Use the **Audio compressor** parameter to specify the type of compression algorithm to use to compress the audio data. This compression reduces the size of the multimedia file. Choose `None (uncompressed)` to save uncompressed audio data to the multimedia file. The other items available in this parameter list are the audio compression algorithms installed on your system. For information about a specific audio compressor, see its documentation.

Use the **Video compressor** parameter to specify the type of compression algorithm to use to compress the video data. This compression reduces the size of the multimedia file. Choose `None (uncompressed)` to save uncompressed video data to the multimedia file. The other items available in this parameter list are the video compression algorithms installed on your system. For information about a specific video compressor, see its documentation.

Use the **Image signal** parameter to specify how the block accepts a color video signal. If you select `One multidimensional signal`, the block accepts an M-by-N-by-3 color video signal. To record M-by-N intensity video, use the Matrix Concatenate block to create an RGB signal. If you select `Separate color signals`, additional ports appear on the block. Each port accepts one M-by-N plane of an RGB video stream.

Note All the To Multimedia File block input signals must have the same frame period. You might need to adjust the frame size of the audio signal so that the frame period of the video signal is the same as the frame period of the audio signal. To calculate the frame size, divide the frequency of the audio signal (samples per second) by the frame rate of the video signal (frames per second).

Dialog Box



Output file name

Specify the name of the multimedia file to which to write. This file is saved in your current directory. To specify a different file, click the **Browse** button, and then navigate to the new file.

Write

Specify whether the block writes video frames and/or audio samples to the multimedia file. The choices are `Video and audio`, `Video only`, or `Audio only`.

Audio compressor

Select the type of compression algorithm to use to compress the audio data.

Video compressor

Select the type of compression algorithm to use to compress the video data.

Image signal

Specify how the block accepts a color video signal. If you select `One multidimensional signal`, the block accepts an M-by-N-by-P color video signal, where P is the number of color planes, at one port. If you select `Separate color signals`, additional ports appear on the block. Each port accepts one M-by-N plane of an RGB video stream.

See Also

From Multimedia File	Signal Processing Blockset
To Wave File	Signal Processing Blockset
Frame Rate Display	Video and Image Processing Blockset
To Video Display	Video and Image Processing Blockset
Video To Workspace	Video and Image Processing Blockset
Video Viewer	Video and Image Processing Blockset

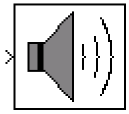
Purpose

Send audio data to standard Windows® audio device in real time

Library

`dspwin32`

Description



Note The To Wave Device block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the To Audio Device block.

The To Wave Device block sends audio data to a standard Windows audio device in real time. It is compatible with most popular Windows hardware, including Sound Blaster cards. The data is sent to the hardware in uncompressed pulse code modulation (PCM) format, and should typically be sampled at one of the standard Windows audio device rates: 8000, 11025, 22050, or 44100 Hz. Some hardware might support other rates in addition to these.

Note Models that contain both the To Wave Device block and the From Wave Device block require a duplex-capable sound card.

The **Use default audio device** check box allows the To Wave Device block to detect and use the system's default audio hardware. You should select this option for systems that have a single sound device installed, or when the default sound device on a multiple-device system is your desired target. When the default sound device is *not* your desired output device, clear **Use default audio device**, and set the desired hardware in the **Audio device** parameter. This parameter lists the names of the installed audio devices.

The block input can contain audio data from a mono or stereo signal. A mono signal is represented as either a sample-based scalar or a frame-based length-M vector, where M is frame size. A stereo signal is represented as a sample-based length-2 vector or a frame-based M-by-2 matrix.

When the input data type is `uint8`, the block conveys the signal samples to the audio device using 8 bits. When the input data type is `double`, `single`, `int16`, or fixed point with a word length of 16 and a fraction length of 15, the block conveys the signal samples to the audio device using 16 bits by default. For inputs of data type `double` and `single`, you can also set the block to convey the signal samples using 24 bits by selecting the **Enable 24-bit output for double- and single-precision input signals** check box. The 24-bit sample width requires more memory but in general yields better fidelity.

The amplitude of the input must be in a valid range that depends on the input data type, as shown in the following table. Amplitudes outside the valid range are clipped to the nearest allowable value.

Input Data Type	Valid Input Amplitude Range
`double`	$-1 \le amplitude < 1$
`single`	$-1 \le amplitude < 1$
`int16`	$-32768 \le amplitude \le 32767$
`uint8`	$0 \le amplitude \le 255$
Fixed point with a word length of 16 and a fraction length of 15	$-1 \le amplitude \le 1 - 2^{-15}$

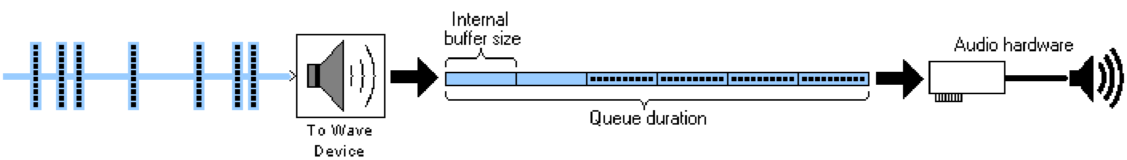
Buffering

Because audio devices generate real-time audio output, the Simulink® environment must maintain a continuous flow of data to a device throughout simulation. Delays in passing data to the audio hardware can result in hardware errors or distortion of the output. This means that the To Wave Device block must in principle supply data to the audio hardware as quickly as the hardware reads the data. However, the To Wave Device block often *cannot* match the throughput rate of the audio hardware, especially when the simulation is running within Simulink rather than as generated code. Simulink execution speed can vary during the simulation as the host operating system services other processes. The block must therefore rely on a buffering strategy to ensure that signal data is available to the hardware on demand.

Note This block requires real-time execution of the parent model for best performance.

The following block parameters control the memory management for this block:

- **Queue duration**
- **Automatically determine internal buffer size** or **User-defined internal buffer size**
- **Initial output delay**



The **Queue duration** parameter defines the overall size of the block's buffer. The block reads in chunks of data in the size of the input dimensions and stores them in the buffer. The internal buffer size defines the dimensions of the block output to the hardware. You can define the internal buffer size yourself in the **User-defined internal buffer size parameter**. If you select **Automatically determine internal buffer size** instead, the internal buffer size is calculated for you according to the following rules:

- If the input to the block has a frame size of 32 samples or larger, the internal buffer size be the same as the input frame size.
- If the input to the block has a frame size smaller than 32 samples, the internal buffer size is based on the input sample rate according to the following table, where

$$F_s = sampling\ frequency = \frac{1}{sample\ time}$$

F_s (Hz)	Internal Buffer Size (samples)
$F_s < 8000$	$\min(64, 2 * F_s)$
$8000 \le F_s < 22{,}050$	128
$22{,}050 \le F_s < 44{,}100$	256
$44{,}100 \le F_s < 96{,}000$	512
$F_s \ge 96{,}000$	1024

To minimize the chance of dropouts, the block checks to make sure that the queue duration is at least as big as twice the internal buffer size. If it is not, the queue duration is automatically set to twice the internal buffer size.

The **Initial output delay** parameter enables you to preload the buffer before the block starts to output data to the audio device, which can be helpful for models that do not run in real time. However, for real-time applications, it is best to set the initial output delay to zero (one frame of delay), or as close to zero as possible.

Troubleshooting

If you are getting undesirable audio output using the To Wave Device block, first determine whether your model can run in real time. Replace the To Wave Device block with a To Wave File block, run the model, and compare the model's simulation stop time to the elapsed time on your watch. If the model simulation stop time is less than the elapsed time on your watch, your model can probably run in real time. Then,

- If your model can run in real time,

  a Select **Automatically determine internal buffer size**. This alone might solve the problem. If not,

  b Try increasing the **Queue duration** parameter to a relatively large value, such as 0.5 s.

If one or both of these options restores desirable audio output, you can try reducing the internal buffer size and/or queue duration until the quality of the audio output again degrades.

- If your model is not running in real time, try to make it run in real time by

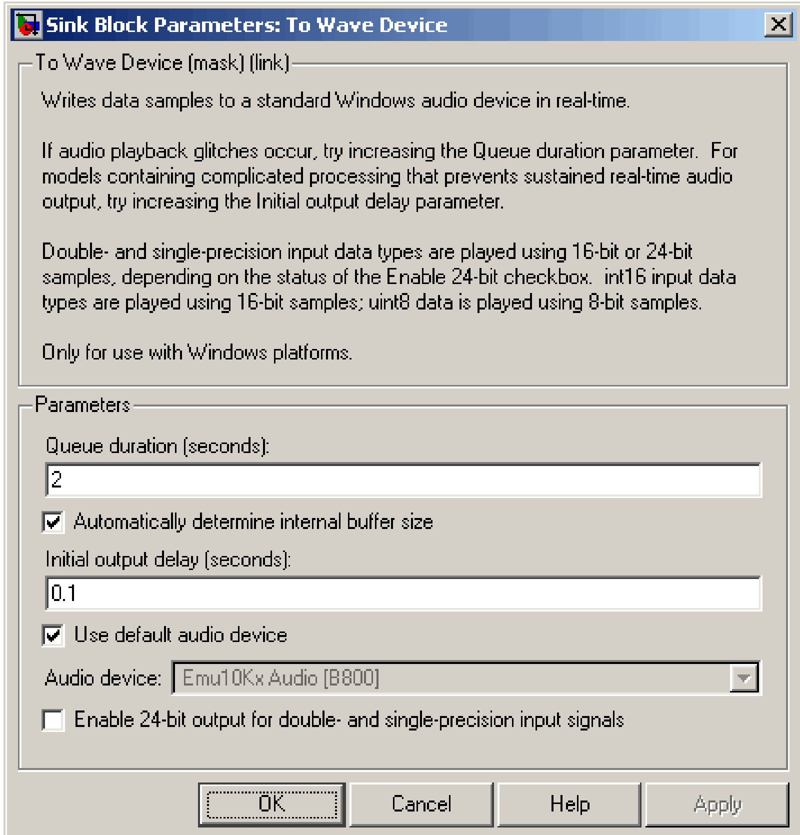
  a Optimizing the model (using a more efficient implementation), or

  b Using the Simulink Accelerator mode, or

  c Generating stand-alone code

  If none of these are possible, but the model only runs for a short period of time, set the **Queue duration** parameter to a size equal to a significant fraction of the model stop time and use a similarly large initial delay. This is not an optimal solution, but might work in some cases.

Dialog Box



Queue duration (seconds)

Specify the overall buffer size. To minimize the chance of dropouts, the block checks to make sure that the queue duration is as least as large as twice the internal buffer size. If it is not, the queue duration is automatically set to twice the internal buffer size.

Automatically determine internal buffer size

Select to have the block automatically select the internal buffer size for you. For details, see "Buffering" on page 2-1196.

User-defined internal buffer size (samples)

Define the internal buffer size, or the size of the chunks of data sent by the block to the audio hardware device.

This parameter is only visible when **Automatically determine internal buffer size** is not selected.

Initial output delay (seconds)

Specify the amount of time by which to delay the initial output to the audio device. During this time data accumulates in the block's buffer. Any value less than or equal to the queue duration specifies the smallest possible initial delay, which is a single frame.

Use default audio device

Select to direct audio output to the system's default audio device.

Audio device

This parameter lists the names of the installed audio devices. Specify the name of the audio device to receive the audio output. Select **Use default audio device** when the system has only a single audio card installed.

This parameter is only enabled when the **Use default audio device** check box is not selected.

Enable 24-bit output for double and single precision input signals

Select to output 24-bit data when inputs are double- or single-precision. Otherwise, the block outputs 16-bit data for double- and single-precision inputs.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point • Signed fixed point with a word length of 16 and a fraction length of 15 • 16-bit signed integers • 8-bit unsigned integers

See Also

From Wave Device	Signal Processing Blockset
To Wave File	Signal Processing Blockset
`audioplayer`	MATLAB
`sound`	MATLAB

To Wave File

Purpose

Write audio data to file in Microsoft® Wave (`.wav`) format

Library

Signal Processing Sinks

`dspsnks4`

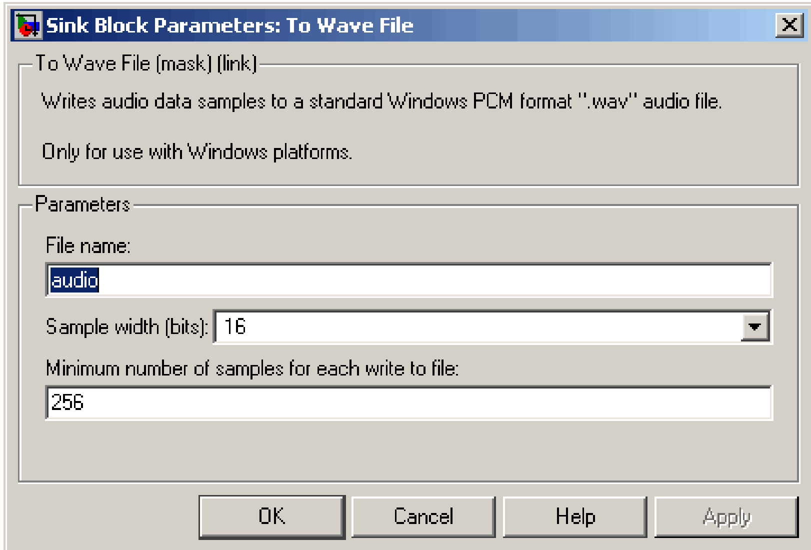
Description

The To Wave File block streams audio data to a Microsoft Wave (`.wav`) file in the uncompressed pulse code modulation (PCM) format. For compatibility reasons, the sample rate of the discrete-time input signal should typically be one of the standard Windows® audio device rates (8000, 11025, 22050, or 44100 Hz), although the block supports arbitrary rates.

The input to the block, u, can contain audio data with one or more channels. A signal with C channels is represented as a sample-based length-C vector or a frame-based M-by-C matrix. The amplitude of the input should be in the range ±1. Values outside this range are clipped to the nearest allowable value.

```
wavwrite(u,Fs,bits,'filename')    % Equivalent MATLAB code
```

Dialog Box



File name

Specify the path and name of the file to write. Paths can be relative or absolute. You do not need to specify the `.wav` extension.

Sample width (bits)

Specify the number of bits used to represent the signal samples in the file. The higher sample width settings require more memory but yield better fidelity for double- and single-precision inputs:

- `8` — Allocates 8 bits to each sample, allowing a resolution of 256 levels
- `16` — Allocates 16 bits to each sample, allowing a resolution of 65536 levels
- `24` — Allocates 24 bits to each sample, allowing a resolution of 16777216 levels
- `32` — Allocates 32 bits to each sample, allowing a resolution of 232 levels ranging from -1 to 1

The 8-, 16-, and 24-bit modes output integer data, while the 32-bit mode outputs single-precision floating-point data.

Minimum number of samples for each write to file

Specify the number of consecutive samples, L, to write with each file access. To reduce the required number of file accesses, the block writes L consecutive samples to the file during each access for $L \geq M$. For $L < M$, the block instead writes M consecutive samples during each access. Larger values of L result in fewer file accesses, which reduces run-time overhead.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point • Signed fixed point with a word length of 16 and a fraction length of 15 • 16-bit signed integers • 8-bit unsigned integers

See Also

From Wave File	Signal Processing Blockset
To Audio Device	Signal Processing Blockset
To Workspace	Simulink
`wavwrite`	MATLAB

Purpose

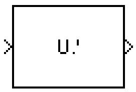
Compute matrix transpose

Library

Math Functions / Matrices and Linear Algebra / Matrix Operations

`dspmtrx3`

Description



The Transpose block transposes the M-by-N input matrix to size N-by-M. When you select the **Hermitian** check box, the block performs the Hermitian (complex conjugate) transpose.

```
y = u'    % Equivalent MATLAB code
```

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \end{bmatrix} \xrightarrow{\texttt{u'}} \begin{bmatrix} u_{11}^* & u_{21}^* \\ u_{12}^* & u_{22}^* \\ u_{13}^* & u_{23}^* \end{bmatrix}$$

When you do not select the **Hermitian** check box, the block performs the nonconjugate transpose.

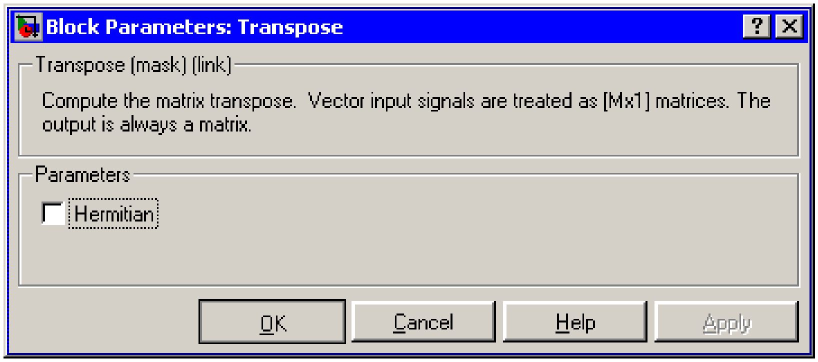
```
y = u.'    % Equivalent MATLAB code
```

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \end{bmatrix} \xrightarrow{\texttt{u.'}} \begin{bmatrix} u_{11} & u_{21} \\ u_{12} & u_{22} \\ u_{13} & u_{23} \end{bmatrix}$$

A length-M 1-D vector input is treated as an M-by-1 matrix. The output is always sample based.

The Transpose block supports real and complex floating-point and fixed-point data types. When **Hermitian** is selected, the block input must be a signed data type.

Dialog Box



Hermitian
When selected, specifies the complex conjugate transpose.

Saturate on integer overflow
This parameter is only visible when the **Hermitian** parameter is selected because overflows can occur when computing the complex conjugate of complex fixed-point signals. When you select this parameter, such overflows saturate. This parameter is ignored for floating-point signals and for real-valued fixed-point signals.

Supported Data Types

When **Hermitian** is selected, the block input must be a signed data type.

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

See Also

Math Function	Simulink
Permute Matrix	Signal Processing Blockset
Reshape	Simulink
Submatrix	Signal Processing Blockset

Triggered Delay Line

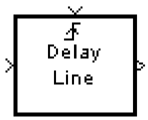
Purpose

Buffer sequence of inputs into frame-based output

Library

`dspobslib`

Description



Note The Triggered Delay Line block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Delay Line block.

The Triggered Delay Line block acquires a collection of M_o input samples into a frame, where you specify M_o in the **Delay line size** parameter. The block buffers a single sample from input 1 whenever it is triggered by the control signal at input 2 (). When the next triggering event occurs, the newly acquired input sample is appended to the output frame so that the new output overlaps the previous output by M_o-1 samples. Between triggering events the block ignores input 1 and holds the output at its last value.

You specify the triggering event at input 2 in the **Trigger type** pop-up menu:

- `Rising edge` triggers execution of the block when the trigger input rises from a negative value to zero or a positive value, or from zero to a positive value.
- `Falling edge` triggers execution of the block when the trigger input falls from a positive value to zero or a negative value, or from zero to a negative value.
- `Either edge` triggers execution of the block when either a rising or falling edge (as described above) occurs.

The Triggered Delay Line block has zero latency, so the new input appears at the output in the same simulation time step. The output frame period is the same as the input sample period, $T_{fo}=T_{si}$.

Sample-Based Operation

In sample-based operation, the Triggered Delay Line block buffers a sequence of sample-based length-N vector inputs (1-D, row, or column) into a sequence of overlapping sample-based M_o-by-N matrix outputs, where you specify M_o in the **Delay line size** parameter (M_o>1). That is, each input vector becomes a *row* in the sample-based output matrix. When M_o=1, the input is simply passed through to the output, and retains the same dimension. Sample-based full-dimension matrix inputs are not accepted.

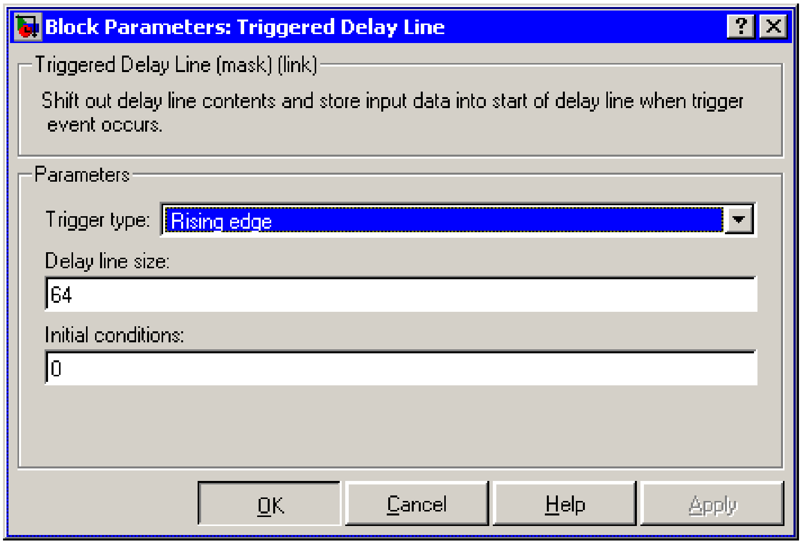
Frame-Based Operation

In frame-based operation, the Triggered Delay Line block rebuffers a sequence of frame-based M_i-by-N matrix inputs into an sequence of overlapping frame-based M_o-by-N matrix outputs, where M_o is the output frame size specified by the **Delay line size** parameter (that is, the number of consecutive samples from the input frame to rebuffer into the output frame). M_o can be greater or less than the input frame size, M_i. Each of the N input channels is rebuffered independently.

Initial Conditions

The Triggered Delay Line block's buffer is initialized to the value specified by the **Initial condition** parameter. The block always outputs this buffer at the first simulation step (*t*=0). When the block's output is a vector, the **Initial condition** can be a vector of the same size or a scalar value to be repeated across all elements of the initial output. When the block's output is a matrix, the **Initial condition** can be a matrix of the same size or a scalar to be repeated across all elements of the initial output.

Dialog Box



Trigger type

The type of event that triggers the block's execution.

Delay line size

The length of the output frame (number of rows in output matrix), M_o.

Initial condition

The value of the block's initial output, a scalar, vector, or matrix.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Trigger	• Any data type supported by the Trigger block
Output	• Double-precision floating point • Single-precision floating point • Fixed point (signed and unsigned) • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers

See Also

Buffer	Signal Processing Blockset
Delay Line	Signal Processing Blockset
Unbuffer	Signal Processing Blockset

Triggered Signal From Workspace

Purpose

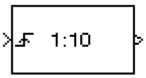
Import signal samples from MATLAB® workspace when triggered

Library

Signal Operations

`dspsigops`

Description



The Triggered Signal From Workspace block imports signal samples from the MATLAB workspace into the Simulink® model when triggered by the control signal at the input port (). The **Signal** parameter specifies the name of a MATLAB workspace variable containing the signal to import, or any valid MATLAB expression defining a matrix or 3-D array.

When the **Signal** parameter specifies an M-by-N matrix ($M \neq 1$), each of the N columns is treated as a distinct channel. You specify the frame size in the **Samples per frame** parameter, M_o, and the output when triggered is an M_o-by-N matrix containing M_o consecutive samples from each signal channel. For $M_o=1$, the output is sample based; otherwise the output is frame based. For convenience, an imported row vector ($M=1$) is treated as a single channel, so the output dimension is M_o-by-1.

When the **Signal** parameter specifies an M-by-N-by-P array, the block generates a single page of the array (an M-by-N matrix) at each trigger time. The **Samples per frame** parameter must be set to `1`, and the output is always sample based.

Trigger Event

You specify the triggering event at the input port in the **Trigger type** pop-up menu:

- `Rising edge` triggers execution of the block when the trigger input rises from a negative value to zero or a positive value, or from zero to a positive value.
- `Falling edge` triggers execution of the block when the trigger input falls from a positive value to zero or a negative value, or from zero to a negative value.

- `Either edge` triggers execution of the block when either a rising or falling edge (as described above) occurs.

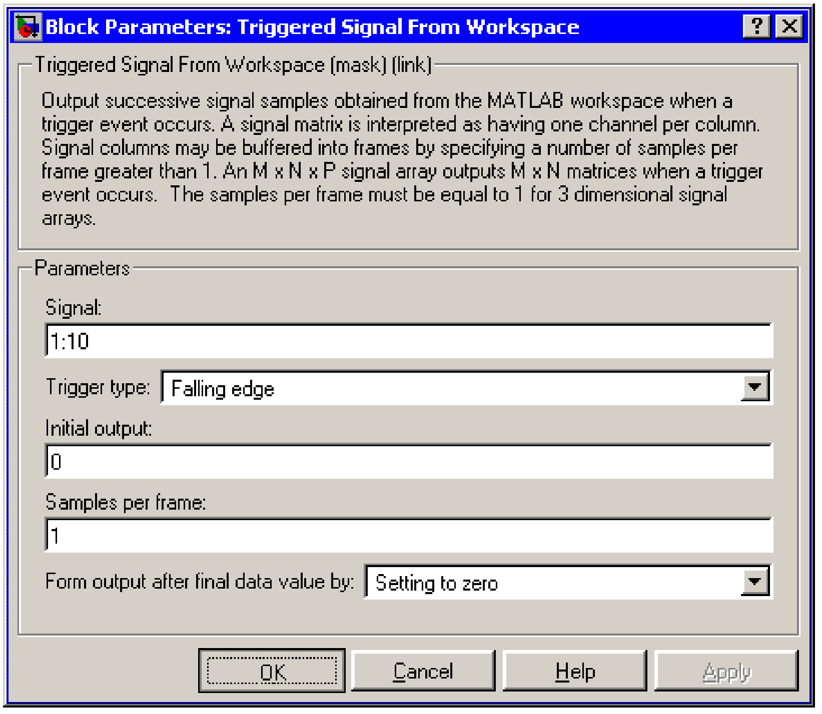
Initial and Final Conditions

The **Initial output** parameter specifies the output of the block from the start of the simulation until the first trigger event arrives. Between trigger events, the block holds the output value constant at its most recent value (that is, no linear interpolation takes place). For single-channel signals, the **Initial output** parameter value can be a vector of length M_o or a scalar to repeat across the M_o elements of the initial output frames. For matrix outputs (M_o-by-N or M-by-N), the **Initial output** parameter value can be a matrix of the same size or a scalar to be repeated across all elements of the initial output.

When the block has output all of the available signal samples, it can start again at the beginning of the signal, or simply repeat the final value or generate zeros until the end of the simulation. (The block does not extrapolate the imported signal beyond the last sample.) The **Form output after final data value by** parameter controls this behavior:

- When you specify `Setting To Zero`, the block generates zero-valued outputs for the duration of the simulation after generating the last frame of the signal.
- When you specify `Holding Final Value`, the block repeats the final sample for the duration of the simulation after generating the last frame of the signal.
- When you specify `Cyclic Repetition`, the block repeats the signal from the beginning after generating the last frame. When there are not enough samples at the end of the signal to fill the final frame, the block zero-pads the final frame as necessary to ensure that the output for each cycle is identical (for example, the *i*th frame of one cycle contains the same samples as the *i*th frame of any other cycle).

Dialog Box



Signal
: The name of the MATLAB workspace variable from which to import the signal, or a valid MATLAB expression specifying the signal.

Trigger type
: The type of event that triggers the block's execution.

Initial output
: The value to output until the first trigger event is received.

Samples per frame
: The number of samples, M_o, to buffer into each output frame. This value must be 1 when you specify a 3-D array in the **Signal** parameter.

Form output after final data value by

Specifies the output after all of the specified signal samples have been generated. The block can output zeros for the duration of the simulation (`Setting to zero`), repeat the final data sample (`Holding Final Value`) or repeat the entire signal from the beginning (`Cyclic Repetition`).

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed and unsigned)
- 8-, 16-, and 32-bit signed integers
- 8-, 16-, and 32-bit unsigned integers

See Also

Signal From Workspace	Signal Processing Blockset
Signal To Workspace	Signal Processing Blockset
Triggered To Workspace	Signal Processing Blockset

Purpose

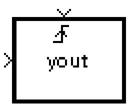
Write input sample to MATLAB® workspace when triggered

Library

Signal Processing Sinks

`dspsnks4`

Description



The Triggered To Workspace block creates a matrix or array variable in the MATLAB workspace, where it stores the acquired inputs at the end of a simulation. The block overwrites an existing variable with the same name.

For an M-by-N frame-based input, the block creates an N-column workspace matrix in which each group of M rows represents a single input frame from each of N channels (the most recent frame occupying the last M rows). The maximum size of this workspace variable is limited to P-by-N, where P is the **Maximum number of rows** parameter. (When the simulation progresses long enough for the block to acquire more than P samples, it stores only the most recent P samples.) The **Decimation factor**, D, allows you to store only every Dth input frame.

For an M-by-N sample-based input, the block creates a three-dimensional array in which each M-by-N page represents a single sample from each of M*N channels (the most recent input matrix occupying the last page). The maximum size of this variable is limited to M-by-N-by-P, where P is the **Maximum number of rows** parameter. (When the simulation progresses long enough for the block to acquire more than P inputs, it stores only the last P inputs.) The **Decimation factor**, D, allows you to store only every Dth input matrix.

The block acquires and buffers a single frame from input 1 whenever it is triggered by the control signal at input 2 (). At all other times, the block ignores input 1. You specify the triggering event at input 2 in the **Trigger type** pop-up menu:

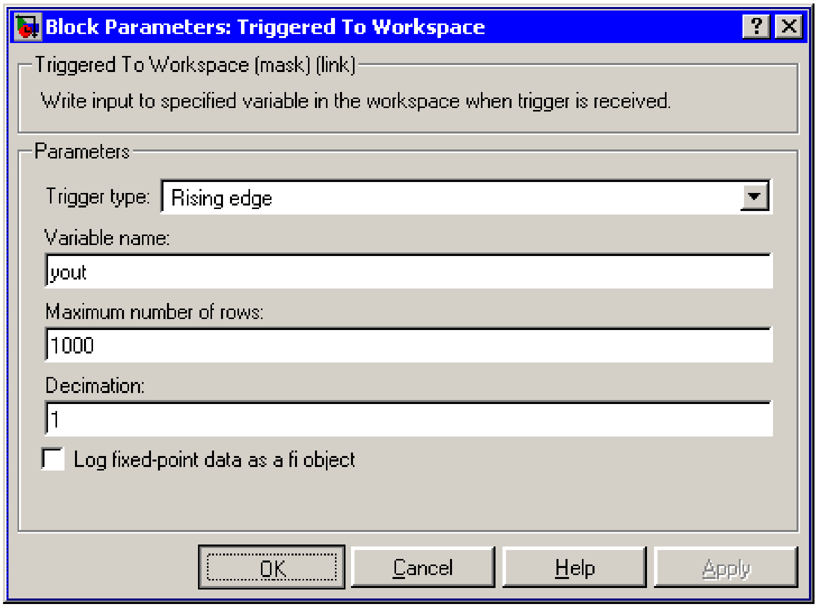
- `Rising edge` triggers execution of the block when the trigger input rises from a negative value to zero or a positive value, or from zero to a positive value.

- `Falling edge` triggers execution of the block when the trigger input falls from a positive value to zero or a negative value, or from zero to a negative value.
- `Either edge` triggers execution of the block when either a rising or falling edge (as described above) occurs.

To save a record of the sample time corresponding to each sample value, open the Configuration Parameters dialog box. In the **Select** pane, click **Data Import/Export**. In the **Save to workspace** section, select the **Time** check box.

The nontriggered version of this block is the Simulink® To Workspace block.

Dialog Box



Trigger type
The type of event that triggers the block's execution.

Variable name
The name of the workspace matrix in which to store the data.

Maximum number of rows
The maximum number of rows (one row per time step) to be saved, P.

Decimation
The decimation factor, D.

Log fixed-point data as a fi object
Select to log fixed-point data to the MATLAB workspace as a Fixed-Point Toolbox™ `fi` object. Otherwise, fixed-point data is logged to the workspace as `double`.

Supported Data Types

Port	Supported Data Types
Input	• Any data type supported by the To Workspace block
Trigger	• Any data type supported by the Trigger block

See Also

Signal From Workspace	Signal Processing Blockset
Signal To Workspace	Signal Processing Blockset
Triggered Signal From Workspace	Signal Processing Blockset

Purpose

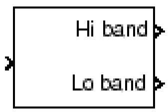
Decompose signal into high-frequency subband and low-frequency subband

Library

Filtering / Multirate Filters

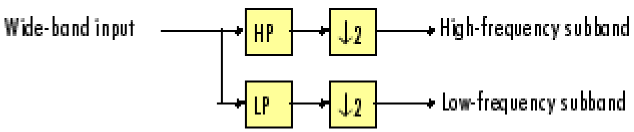
`dspmlti4`

Description



The Two-Channel Analysis Subband Filter block decomposes the input into a high-frequency subband and a low-frequency subband, each with half the bandwidth and half the sample rate of the input.

The block filters the input with a pair of highpass and lowpass FIR filters, and then downsamples the results by 2, as illustrated in the following figure.



Note that the block implements the FIR filtering and downsampling steps together using a polyphase filter structure, which is more efficient than the straightforward filter-then-decimate algorithm illustrated above. Each subband is the first phase of the respective polyphase filter.

You must provide the vector of filter coefficients for the two filters. Each filter should be a half-band filter that passes the frequency band that the other filter stops. For frame-based inputs, you also need to specify whether the change in the sample rate of the output gets reflected by a change in the frame size, or the frame rate.

Note By connecting many copies of this block, you can implement a multilevel dyadic analysis filter bank. In some cases, it is more efficient to use the Dyadic Analysis Filter Bank block instead. For more information, see "Creating Multilevel Dyadic Analysis Filter Banks" on page 2-1225.

Sections of This Reference Page

- "Specifying the FIR Filters" on page 2-1222
- "Sample-Based Operation" on page 2-1223
- "Frame-Based Operation" on page 2-1223
- "Latency" on page 2-1224
- "Creating Multilevel Dyadic Analysis Filter Banks" on page 2-1225
- "Fixed-Point Data Types" on page 2-1227
- "Examples" on page 2-1227
- "Dialog Box" on page 2-1228
- "References" on page 2-1234
- "Supported Data Types" on page 2-1235
- "See Also" on page 2-1235

Specifying the FIR Filters

You must provide the vector of numerator coefficients for the lowpass and highpass filters in the **Lowpass FIR filter coefficients** and **Highpass FIR filter coefficients** parameters.

For example, to specify a filter with the following transfer function, enter the vector `[b(1) b(2) ...  b(m)]`.

$$H(z) = B(z) = b_1 + b_2 z^{-1} + \ldots + b_m z^{-(m-1)}$$

Each filter should be a half-band filter that passes the frequency band that the other filter stops. When you plan to use the Two-Channel Synthesis Subband Filter block to reconstruct the input to this block, you need to design perfect reconstruction filters to use in the synthesis subband filter.

The best way to design perfect reconstruction filters is to use the Wavelet Toolbox™ `wfilters` function in to design both the filters both in this block and in the Two-Channel Synthesis Subband Filter block. You can also use Filter Design Toolbox™ functions and Signal Processing Toolbox™ functions. To learn how to design your own perfect reconstruction filters, see "References" on page 2-1234.

The block initializes all filter states to zero.

Sample-Based Operation

- "Valid Sample-Based Inputs" on page 2-1223
- "Sample-Based Outputs" on page 2-1223

Valid Sample-Based Inputs

The block accepts all M-by-N sample-based matrix inputs. The block treats such inputs as $M \cdot N$ independent channels, and decomposes each channel over time.

Sample-Based Outputs

Given a sample-based M-by-N input, the block outputs two M-by-N sample-based matrices whose sample rates are half the input sample rate. Each output matrix element is the high- or low-frequency subband output of the corresponding input matrix element. Depending on the Simulink® configuration parameters, some sample-based outputs can have one sample of latency, as described in "Latency" on page 2-1224.

Frame-Based Operation

- "Valid Frame-Based Inputs" on page 2-1223
- "Frame-Based Outputs" on page 2-1224

Valid Frame-Based Inputs

The block accepts M-by-N frame-based matrix inputs where M is a multiple of two. The block treats such inputs as N independent channels, and decomposes each channel over time.

Frame-Based Outputs

Given a valid frame-based input, the block outputs two frame-based matrices. Each output column is the high- or low-frequency subband of the corresponding input column.

The sample rate of the outputs are half that of the input. The **Framing** parameter sets whether the block halves the sample rate by halving the output frame size, or halving the output frame rate:

- `Maintain input frame size` — The input and output frame *sizes* are the same, but the frame *rate* of the outputs are half that of the input. So, the overall sample rate of the output is half that of the input. This setting causes the block to have one frame of latency, as described in “Latency” on page 2-1224.
- `Maintain input frame rate` — The input and output frame *rates* are the same, but the frame *size* of the outputs are half that of the input (the input frame size must be a multiple of two). So, the overall sample rate of the output is half that of the input.

Latency

In some cases, the block has nonzero tasking latency, which means that there is a constant delay between the time that the block receives an input, and produces the corresponding output, as summarized below and in the following table:

- For sample-based inputs, there are cases where the block exhibits *one-sample latency*. In such cases, when the block receives the *n*th input sample, it produces the outputs corresponding to the *n-1*th input sample. When the block receives the first input sample, the block outputs an initial value of zero in each output channel.
- For frame-based inputs, there are cases where the block exhibits *one-frame latency*. In such cases, when the block receives the *n*th input frame, it produces the outputs corresponding to the *n-1*th input frame. When the block receives the first input frame, the block outputs a frame of zeros.

Note For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Amount of Block Latency for All Possible Block Settings

Input	Latency	No Latency
Sample based	One sample of latency when the **Tasking mode for periodic sample times** parameter is set to `MultiTasking` or `Auto` in the **Solver** pane of the Configuration Parameters dialog box. The first output sample of each channel is always `0`.	The **Tasking mode for periodic sample times** parameter is set to `SingleTasking` in the **Solver** pane of the Configuration Parameters dialog box.
Frame based	One frame of latency when the **Framing** parameter is set to `Maintain input frame size`. The first output frame is always all zeros.	The **Framing** parameter is set to `Maintain input frame rate`.

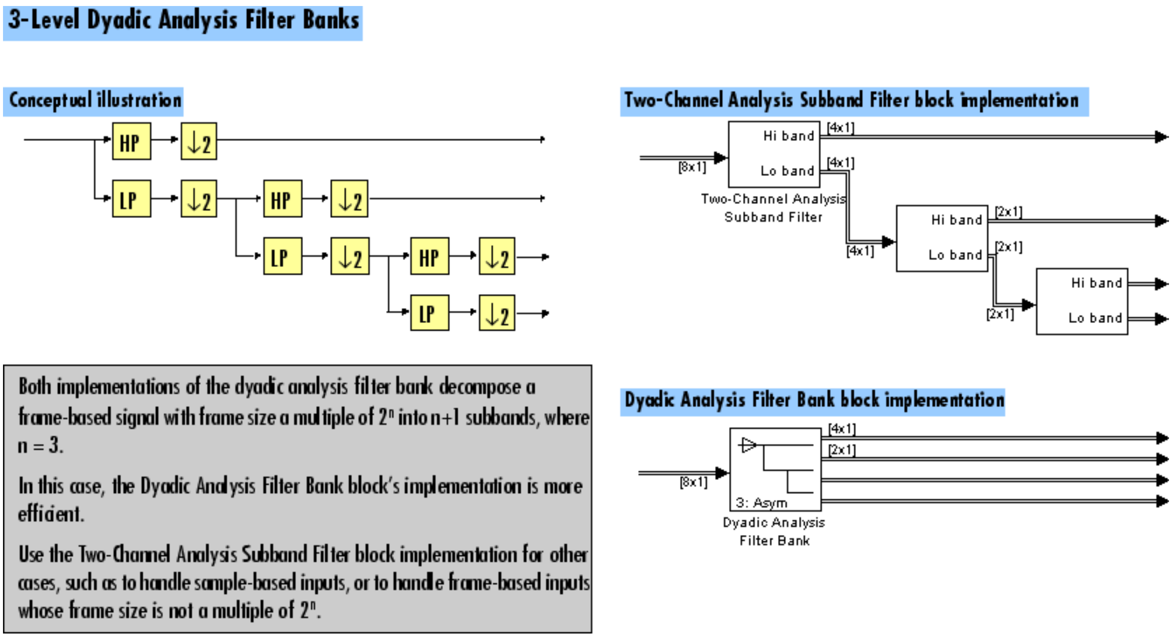
Creating Multilevel Dyadic Analysis Filter Banks

The Two-Channel Analysis Subband Filter block is the basic unit of a dyadic analysis filter bank. You can connect several of these blocks to implement an n-level filter bank, as illustrated in the following figure. For a review of dyadic analysis filter banks, see the Dyadic Analysis Filter Bank block reference page.

When you create a filter bank by connecting multiple copies of this block, the output values of the filter bank differ depending on whether there is latency. See the previous table,

For instance, for frame-based inputs, the filter bank output values differ depending on whether you set the **Framing** parameter to `Maintain input frame rate` (no latency), or `Maintain input frame size` (one frame of latency for every block). Though the output values differ, both sets of values are valid; the difference arises from changes in latency.

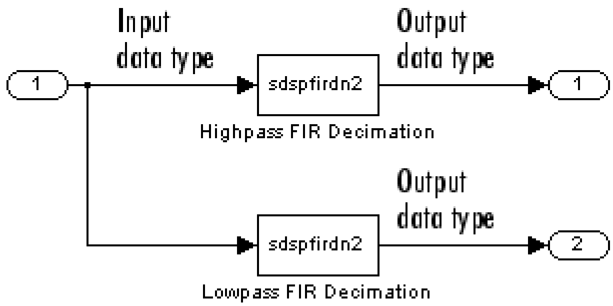
In some cases, rather than connecting several Two-Channel Analysis Subband Filter blocks, it is faster and requires less memory to use the Dyadic Analysis Filter Bank block. In particular, use the Dyadic Analysis Filter Bank block when you want to decompose a frame-based signal with frame size a multiple of 2^n into n+1 or 2^n subbands. In all other cases, use Two-Channel Analysis Subband Filter blocks to implement your filter banks.



The Dyadic Analysis Filter Bank block allows you to specify the filter bank filters by providing vectors of filter coefficients, just as this block does. The Dyadic Analysis Filter Bank block provides an additional option of using wavelet-based filters that the block designs by using a wavelet you specify.

Fixed-Point Data Types

The Two-Channel Analysis Subband Filter block is comprised of two FIR Decimation blocks as shown in the following diagram.



For fixed-point signals, you can set the coefficient, product output, accumulator, and output data types of the FIR Decimation blocks as discussed in "Dialog Box" on page 2-1228. For a diagram showing the usage of these data types, see the FIR Decimation block reference page.

Examples

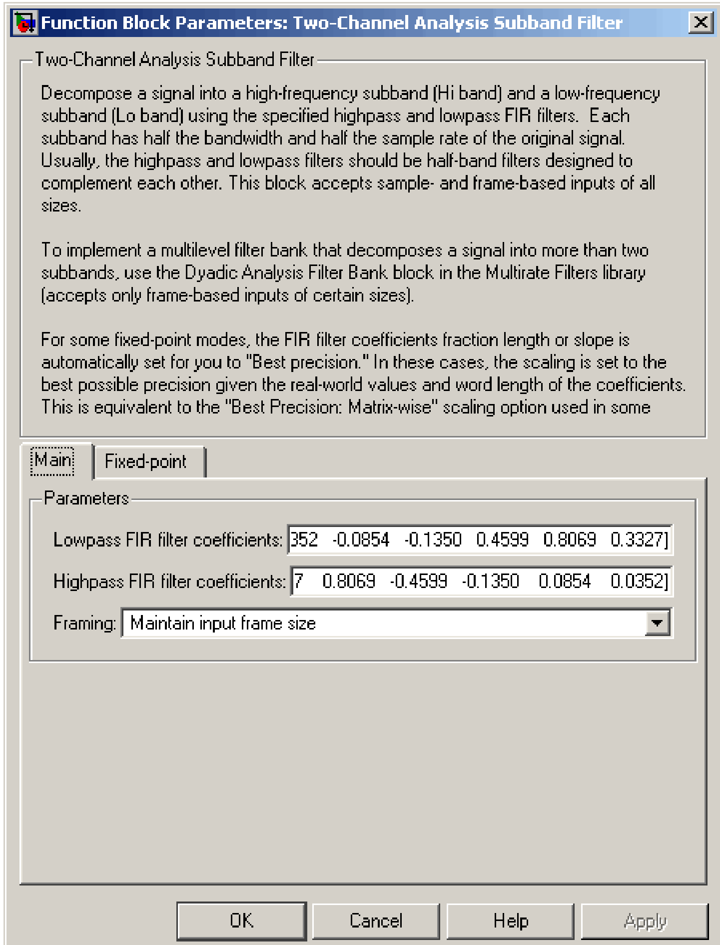
See the following Signal Processing Blockset™ demos, which use the Two-Channel Analysis Subband Filter block:

- Multilevel PR filter bank
- Denoising
- Wavelet transmultiplexer (WTM)

Note By default, the demos open the versions using the Two-Channel Analysis Subband Filter block. You can also see the version of the demos that use the Dyadic Analysis Filter Bank block by clicking the **Frame-Based Demo** button in the demos.

Dialog Box

The **Main** pane of the Two-Channel Analysis Subband Filter block dialog appears as follows.



Lowpass FIR filter coefficients

Specify a vector of lowpass FIR filter coefficients, in descending powers of z. The lowpass filter should be a half-band filter that passes the frequency band stopped by the filter specified in the **Highpass FIR filter coefficients** parameter. The default values of this parameter specify a filter based on a 3rd-order Daubechies wavelet. When you use the Two-Channel Synthesis Subband Filter block to reconstruct the input to this block, you need to design perfect reconstruction filters to use in the synthesis subband filter. For more information, see "Specifying the FIR Filters" on page 2-1222.

Highpass FIR filter coefficients

Specify a vector of highpass FIR filter coefficients, in descending powers of z. The highpass filter should be a half-band filter that passes the frequency band stopped by the filter specified in the **Lowpass FIR filter coefficients** parameter. The default values of this parameter specify a filter based on a 3rd-order Daubechies wavelet. When you use the Two-Channel Synthesis Subband Filter block to reconstruct the input to this block, you need to design perfect reconstruction filters to use in the synthesis subband filter. For more information, see "Specifying the FIR Filters" on page 2-1222.

Framing

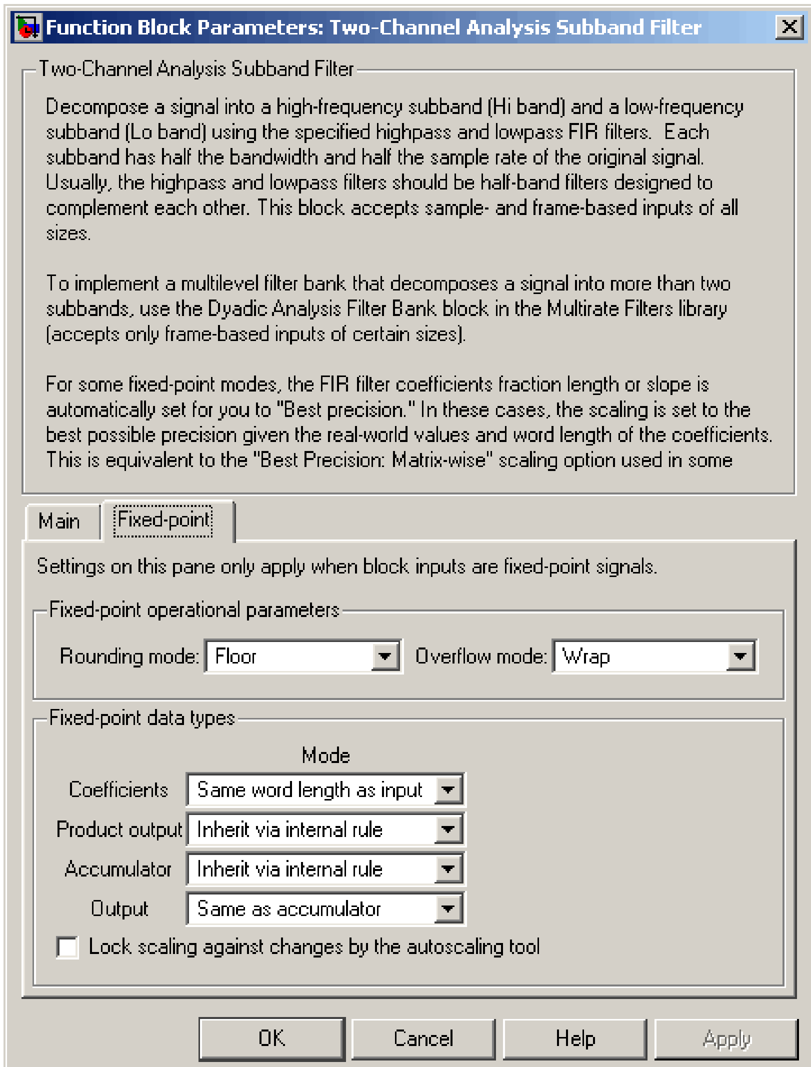
Specify the method by which to implement the decimation for frame-based inputs:

Select `Maintain input frame size` to halve the output frame rate

Select `Maintain input frame rate` to halve the output frame size

For more information, see "Frame-Based Operation" on page 2-1223. Some settings of this parameter causes the block to have nonzero latency, as described in "Latency" on page 2-1224.

The **Fixed-point** pane of the Two-Channel Analysis Subband Filter block dialog appears as follows.



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

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

Coefficients
Choose how you specify the word length and the fraction length of the FIR filter coefficients:

When you select `Same word length as input`, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.

When you select `Specify word length`, you can enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the coefficients, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the coefficients. This block requires power-of-two slope and a bias of zero.

The filter coefficients do not obey the **Rounding mode** and the **Overflow mode** parameters; they are always saturated and rounded to `Nearest`.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-475 of the FIR Decimation reference page and "Multiplication Data Types" for illustrations depicting the use of the product output data type in the FIR Decimation blocks of this block:

When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".

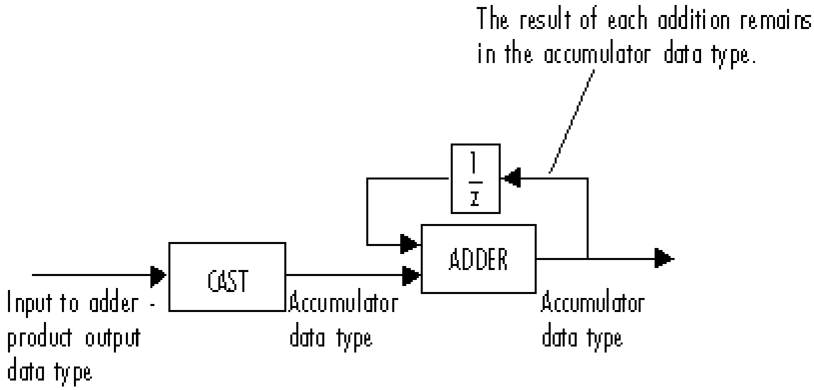
Note The actual product output word length may be equal to or greater than the calculated ideal product output word length, depending on the settings on the **Hardware Implementation** pane of the Configuration Parameters dialog box.

When you select `Same as input`, these characteristics match those of the input to the block.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator



As depicted above, inputs to the accumulator are cast to the accumulator data type. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you would like to designate this accumulator word and fraction lengths.

You also use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the FIR Decimation blocks in this block. See "Multiplication Data Types" for more information:

When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".

When you select `Same as product output`, these characteristics match those of the product output

When you select `Same as input`, these characteristics match those of the input to the block.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length of the FIR Decimation blocks, as well as of the final overall filter output:

When you select `Same as accumulator`, these characteristics match those of the accumulator.

A special case occurs when `Inherit via internal rule` is specified for **Accumulator**, and block inputs and coefficients are complex. In that case, the output word length is one less than the accumulator word length.

When you select `Same as product output`, these characteristics match those of the product output

When you select `Same as input`, these characteristics match those of the input to the block.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

References

Fliege, N. J. *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets*. West Sussex, England: John Wiley & Sons, 1994.

Strang, G. and T. Nguyen. *Wavelets and Filter Banks*. Wellesley, MA: Wellesley-Cambridge Press, 1996.

Vaidyanathan, P. P. *Multirate Systems and Filter Banks*. Englewood Cliffs, NJ: Prentice Hall, 1993.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)
- 8-, 16-, and 32-bit signed integers

See Also

Dyadic Analysis Filter Bank	Signal Processing Blockset
FIR Decimation	Signal Processing Blockset
Two-Channel Synthesis Subband Filter	Signal Processing Blockset
`fir1`	Signal Processing Toolbox
`fir2`	Signal Processing Toolbox
`firls`	Signal Processing Toolbox
`wfilters`	Wavelet Toolbox

For related information, see “Multirate Filters”.

Two-Channel Synthesis Subband Filter

Purpose

Reconstruct signal from high-frequency subband and low-frequency subband

Library

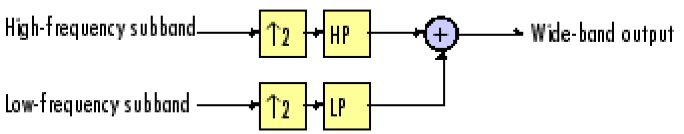
Filtering / Multirate Filters

`dspmlti4`

Description

The Two-Channel Synthesis Subband Filter block reconstructs a signal from its high-frequency subband and low-frequency subband, each with half the bandwidth and half the sample rate of the original signal. Use this block to reconstruct signals decomposed by the Two-Channel Analysis Subband Filter block.

The block upsamples the high- and low-frequency subbands by 2, and then filters the results with a pair of highpass and lowpass FIR filters, as illustrated in the following figure.



Note that the block implements the FIR filtering and downsampling steps together using a polyphase filter structure, which is more efficient than the straightforward interpolate-then-filter algorithm illustrated above.

You must provide the vector of filter coefficients for the two filters. Each filter should be a half-band filter that passes the frequency band that the other filter stops. To use this block to reconstruct the output of a Two-Channel Analysis Subband Filter block, the filters in this block *must* be designed to perfectly reconstruct the outputs of the analysis filters.

Note By connecting many copies of this block, you can implement a multilevel dyadic synthesis filter bank. In some cases, it is more efficient to use the Dyadic Synthesis Filter Bank block instead. For more information, see "Creating Multilevel Dyadic Synthesis Filter Banks" on page 2-1241.

Sections of This Reference Page

- "Specifying the FIR Filters" on page 2-1237
- "Sample-Based Operation" on page 2-1238
- "Frame-Based Operation" on page 2-1239
- "Latency" on page 2-1239
- "Creating Multilevel Dyadic Synthesis Filter Banks" on page 2-1241
- "Fixed-Point Data Types" on page 2-1242
- "Examples" on page 2-1243
- "Dialog Box" on page 2-1244
- "References" on page 2-1250
- "Supported Data Types" on page 2-1251
- "See Also" on page 2-1251

Specifying the FIR Filters

You must provide the vector of numerator coefficients for the lowpass and highpass filters in the **Lowpass FIR filter coefficients** and **Highpass FIR filter coefficients** parameters.

For example, to specify a filter with the following transfer function, enter the vector `[b(1) b(2) ...  b(m)]`.

$$H(z) = B(z) = b_1 + b_2 z^{-1} + \ldots + b_m z^{-(m-1)}$$

Each filter should be a half-band filter that passes the frequency band that the other filter stops. To use this block to reconstruct the output of a Two-Channel Analysis Subband Filter block, the filters in this block must be designed to perfectly reconstruct the outputs of the analysis filters.

The best way to design perfect reconstruction filters is to use the Wavelet Toolbox™ `wfilters` function for the filters in both this block *and* in the corresponding Two-Channel Analysis Subband Filter block. You can also use Filter Design Toolbox™ functions and Signal Processing Toolbox™ functions. To learn how to design your own perfect reconstruction filters, see "References" on page 2-1250.

The block initializes all filter states to zero.

Sample-Based Operation

- "Valid Sample-Based Inputs" on page 2-1238
- "Sample-Based Outputs" on page 2-1238

Valid Sample-Based Inputs

The block accepts any two M-by-N sample-based matrices with the same sample rates. The block treats each M-by-N matrix as MxN independent subbands, where MxN is the product of the matrix dimensions. Each matrix element is the high- or low-frequency subband of the corresponding channel in the output matrix. The input to the topmost input port should contain the high-frequency subbands.

Sample-Based Outputs

Given valid sample-based inputs, the block outputs one sample-based matrix with the same dimensions as the inputs. The output sample rate is twice that of the input. Each element of the output is a single channel, reconstructed from the corresponding elements in each input matrix. Depending on the Simulink® configuration parameters, some sample-based outputs can have one sample of latency, as described in "Latency" on page 2-1239.

Frame-Based Operation

- "Valid Frame-Based Inputs" on page 2-1239
- "Frame-Based Outputs" on page 2-1239

Valid Frame-Based Inputs

The block accepts any two M-by-N frame-based matrices with the same frame rates. The block treats each input column as the high- or low-frequency subbands of the corresponding output channel. The input to the topmost input port should contain the high-frequency subbands.

Frame-Based Outputs

Given valid frame-based inputs, the block outputs a frame-based matrix. Each output column is a single channel, reconstructed from the corresponding columns in each input matrix.

The sample rate of the output is twice that of the input. The **Framing** parameter sets whether the block doubles the sample rate by doubling the output frame size, or doubling the output frame rate:

- `Maintain input frame size` — The input and output frame *sizes* are the same, but the frame *rate* of the output is twice that of the input. So, the overall sample rate of the output is twice that of the input. This setting causes the block to have one frame of latency, as described in "Latency" on page 2-1224.
- `Maintain input frame rate` — The input and output frame *rates* are the same, but the frame *size* of the output is twice that of the input. So, the overall sample rate of the output is twice that of the input.

Latency

In some cases, the block has nonzero tasking latency, which means that there is a constant delay between the time that the block receives an input, and produces the corresponding output, as summarized below and in the following table:

- For sample-based inputs, there are cases where the block exhibits *one-sample latency*. In such cases, when the block receives the *n*th input sample, it produces the outputs corresponding to the *n-1*th input sample. When the block receives the first input sample, the block outputs an initial value of zero in each output channel.
- For frame-based inputs, there are cases where the block exhibits *one-frame latency*. In such cases, when the block receives the *n*th input frame, it produces the outputs corresponding to the *n-1*th input frame. When the block receives the first input frame, the block outputs a frame of zeros.

Note For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Amount of Block Latency for All Possible Block Settings

Input	Latency	No Latency
Sample based	One sample of latency when the **Tasking mode for periodic sample times** parameter is set to `MultiTasking` or `Auto` in the **Solver** pane of the Configuration Parameters dialog box. The first output sample of each channel is always `0`.	The **Tasking mode for periodic sample times** parameter is set to `SingleTasking` in the **Solver** pane of the Configuration Parameters dialog box.
Frame based	One frame of latency when the **Framing** parameter is set to `Maintain input frame size`. The first output frame is always all zeros.	The **Framing** parameter is set to `Maintain input frame rate`.

Creating Multilevel Dyadic Synthesis Filter Banks

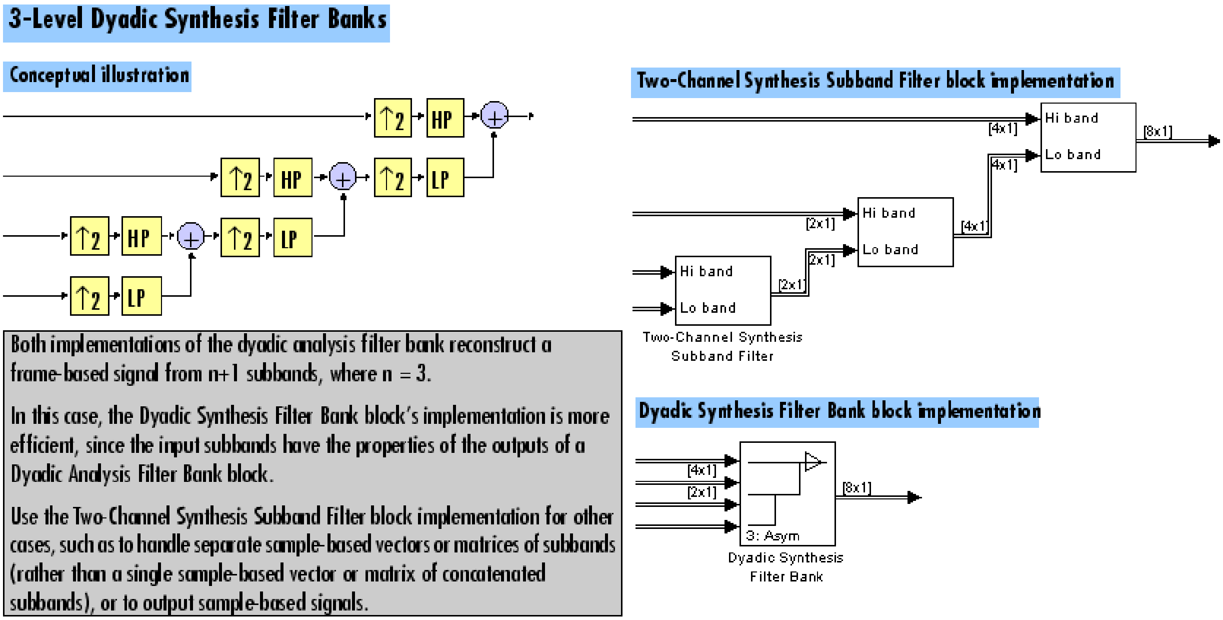
The Two-Channel Synthesis Subband Filter block is the basic unit of a dyadic synthesis filter bank. You can connect several of these blocks to implement an n-level filter bank, as illustrated in the following figure. For a review of dyadic synthesis filter banks, see the Dyadic Synthesis Filter Bank block reference page.

When you create a filter bank by connecting multiple copies of this block, the output values of the filter bank differ depending on whether there is latency. See the previous table, Amount of Block Latency for All Possible Block Settings on page 2-1240.

For instance, for frame-based inputs, the filter bank output values differ depending on whether you set the **Framing** parameter to `Maintain input frame rate` (no latency), or `Maintain input frame size` (one frame of latency for every block). Though the output values differ, both sets of values are valid; the difference arises from changes in latency.

In some cases, rather than connecting several Two-Channel Synthesis Subband Filter blocks, it is faster and requires less memory to use the Dyadic Synthesis Filter Bank block. In particular, use the Dyadic Synthesis Filter Bank block to reconstruct a frame-based signal (with frame size a multiple of 2^n) from 2^n or $n+1$ subbands whose properties match those of the Dyadic Analysis Filter Bank block's outputs. These properties are described in the Dyadic Analysis Filter Bank reference page.

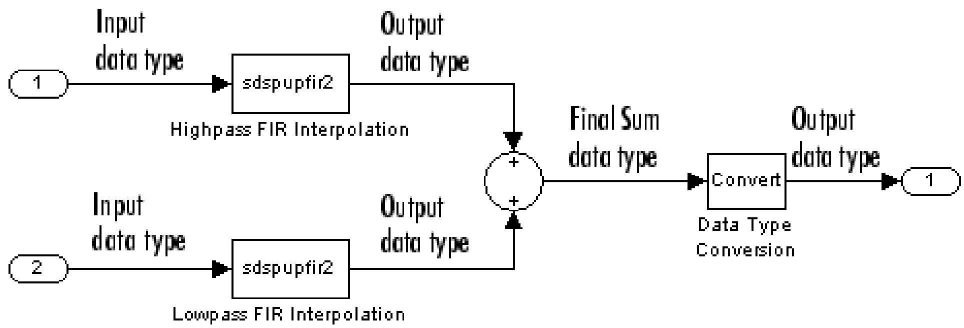
Two-Channel Synthesis Subband Filter



The Dyadic Synthesis Filter Bank block allows you to specify the filter bank filters by providing vectors of filter coefficients, just as this block does. The Dyadic Synthesis Filter Bank block provides an additional option of using wavelet-based filters that the block designs by using a wavelet you specify.

Fixed-Point Data Types

The Two-Channel Synthesis Subband Filter block is comprised of two FIR Interpolation blocks as shown in the following diagram.



For fixed-point signals, you can set the coefficient, product output, accumulator, and output data types used in the FIR Interpolation blocks as discussed in "Dialog Box" on page 2-1244 below. For a diagram showing the usage of these data types within the FIR blocks, see the FIR Interpolation block reference page.

In addition, the inputs to the Sum block in the diagram above are accumulated using the accumulator data type. The output of the Sum block is then cast from the accumulator data type to the output data type. Therefore the output of the Two-Channel Synthesis Subband Filter block is in the output data type. You also set these data types in the block dialog as discussed in "Dialog Box" on page 2-1244 below.

Examples

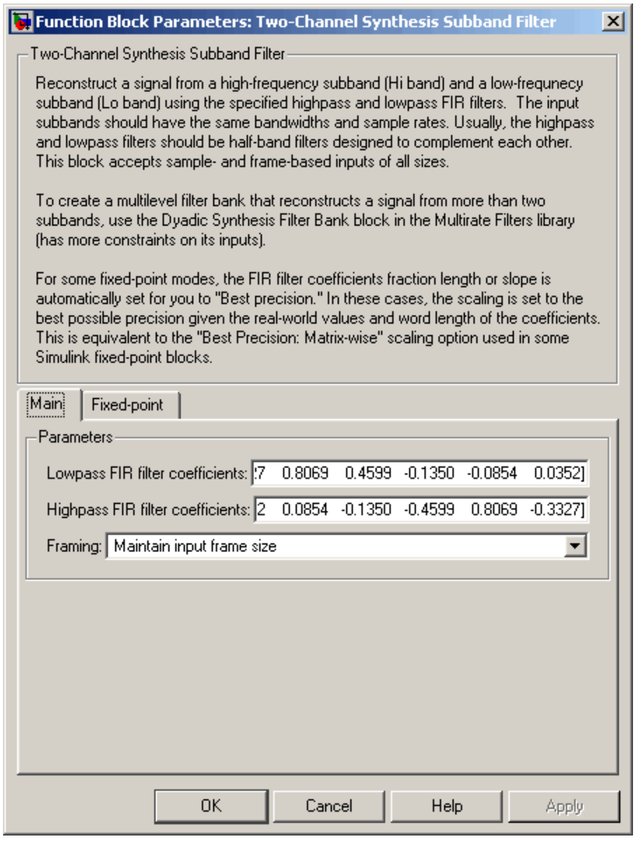
See the following Signal Processing Blockset™ demos, which use the Two-Channel Synthesis Subband Filter block:

- Multilevel PR filter bank
- Denoising
- Wavelet transmultiplexer (WTM)

Note By default, the demos open the versions using the Two-Channel Synthesis Subband Filter block. You can also see the version of the demos that use the Dyadic Synthesis Filter Bank block by clicking the **Frame-Based Demo** button in the demos.

Dialog Box

The **Main** pane of the Two-Channel Synthesis Subband Filter block dialog appears as follows.



Lowpass FIR filter coefficients

A vector of lowpass FIR filter coefficients, in descending powers of z. The lowpass filter should be a half-band filter that passes the frequency band stopped by the filter specified in the **Highpass FIR filter coefficients** parameter. To use this block to reconstruct the output of a Two-Channel Analysis Subband Filter block, you must design the filters in this block to perfectly reconstruct the outputs of the analysis filters. For more information, see "Specifying the FIR Filters" on page 2-1237.

Highpass FIR filter coefficients

A vector of highpass FIR filter coefficients, in descending powers of z. The highpass filter should be a half-band filter that passes the frequency band stopped by the filter specified in the **Lowpass FIR filter coefficients** parameter. To use this block to reconstruct the output of a Two-Channel Analysis Subband Filter block, you must design the filters in this block to perfectly reconstruct the outputs of the analysis filters. For more information, see "Specifying the FIR Filters" on page 2-1237.

Framing

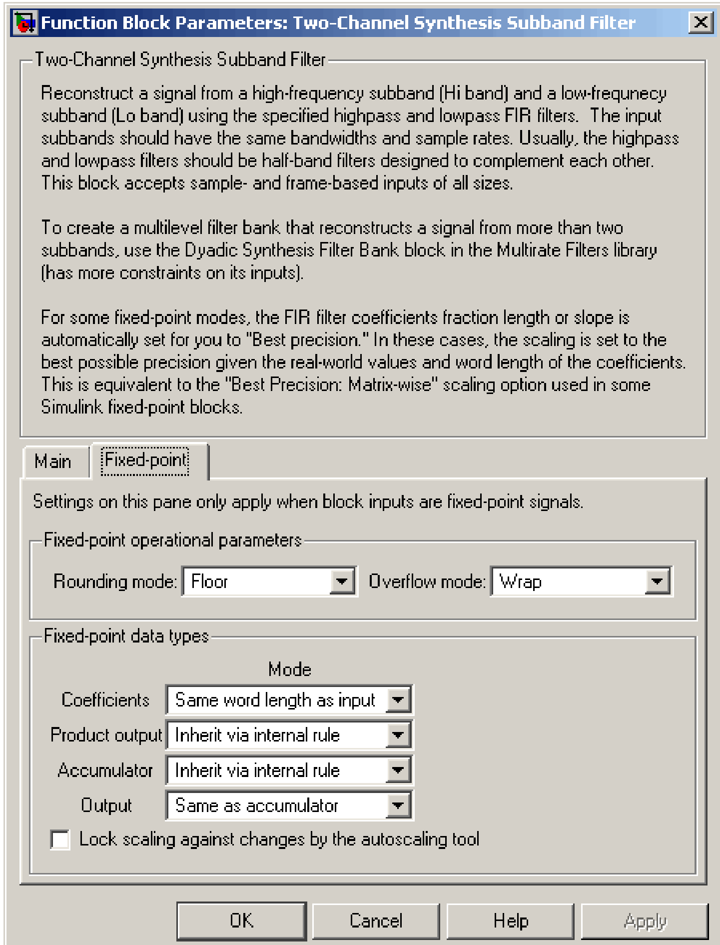
Select the method by which to implement the interpolation for frame-based inputs:

Select `Maintain input frame size` to double the output frame rate

Select `Maintain input frame rate` to double the output frame size

For more information, see "Frame-Based Operation" on page 2-1223. Some settings of this parameter causes the block to have nonzero latency, as described in "Latency" on page 2-1224.

The **Fixed-point** pane of the Two-Channel Synthesis Subband Filter block dialog appears as follows.



Round mode

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

Overflow mode

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

Coefficients

Choose how you specify the word length and the fraction length of the FIR filter coefficients:

When you select `Same word length as input`, the word length of the filter coefficients match that of the input to the block. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.

When you select `Specify word length`, you can enter the word length of the coefficients, in bits. In this mode, the fraction length of the coefficients is automatically set to the binary-point only scaling that provides you with the best precision possible given the value and word length of the coefficients.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the coefficients, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the coefficients. This block requires power-of-two slope and a bias of zero.

The filter coefficients do not obey the **Rounding mode** and the **Overflow mode** parameters; they are always saturated and rounded to `Nearest`.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-494 of the FIR Interpolation reference page and "Multiplication Data Types" for illustrations depicting the use of the product output data type in the FIR Interpolation blocks of this block:

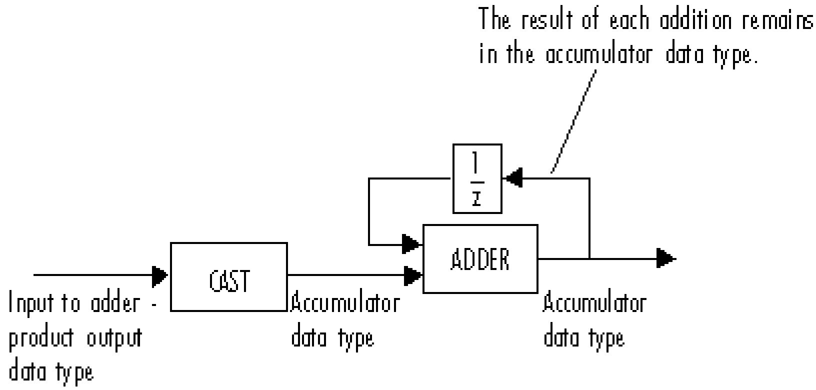
When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".

When you select `Same as input`, these characteristics match those of the input to the block.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Accumulator



As depicted above, inputs to the accumulator are cast to the accumulator data type. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you would like to designate this accumulator word and fraction lengths.

You also use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the FIR Interpolation blocks in this block. See "Multiplication Data Types" for more information:

When you select `Inherit via internal rule`, the accumulator word length and fraction length are calculated automatically. For information about how the accumulator word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".

When you select `Same as product output`, these characteristics match those of the product output

When you select `Same as input`, these characteristics match those of the input to the block.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length of the FIR Interpolation blocks, as well as of the final overall filter output:

When you select `Same as accumulator`, these characteristics match those of the accumulator.

A special case occurs when `Inherit via internal rule` is specified for **Accumulator**, and block inputs and coefficients are complex. In that case, the output word length is one less than the accumulator word length.

When you select `Same as product output`, these characteristics match those of the product output

When you select `Same as input`, these characteristics match those of the input to the block.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

References

Fliege, N. J. *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets*. West Sussex, England: John Wiley & Sons, 1994.

Strang, G. and T. Nguyen. *Wavelets and Filter Banks*. Wellesley, MA: Wellesley-Cambridge Press, 1996.

Vaidyanathan, P. P. *Multirate Systems and Filter Banks*. Englewood Cliffs, NJ: Prentice Hall, 1993.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)
- 8-, 16-, and 32-bit signed integers

See Also

Dyadic Synthesis Filter Bank	Signal Processing Blockset
FIR Interpolation	Signal Processing Blockset
Two-Channel Analysis Subband Filter	Signal Processing Blockset
`fir1`	Signal Processing Toolbox
`fir2`	Signal Processing Toolbox
`firls`	Signal Processing Toolbox
`wfilters`	Wavelet Toolbox

For related information, see "Multirate Filters".

Unbuffer

Purpose

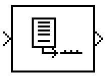
Unbuffer input frame into sequence of scalar outputs

Library

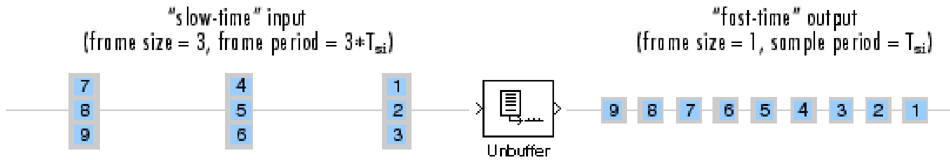
Signal Management / Buffers

`dspbuff3`

Description

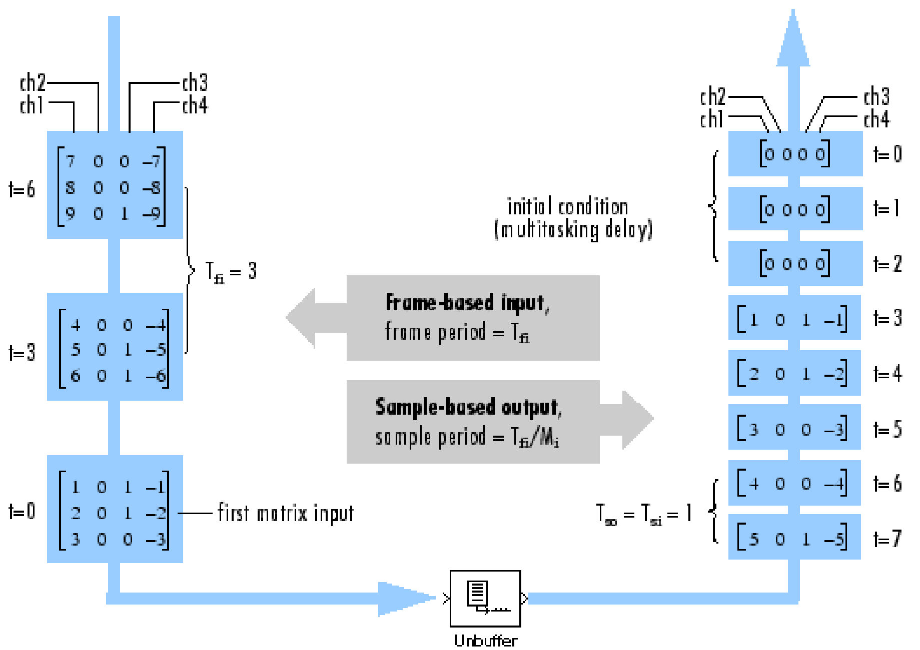


The Unbuffer block unbuffers an M_i-by-N frame-based input into a 1-by-N sample-based output. That is, inputs are unbuffered *row-wise* so that each matrix row becomes an independent time-sample in the output. The rate at which the block receives inputs is generally less than the rate at which the block produces outputs.



The block adjusts the output rate so that the *sample period* is the same at both the input and output, T_{so}=T_{si}. Therefore, the output sample period for an input of frame size M_i and frame period T_{fi} is T_{fi}/M_i, which represents a *rate* M_i times higher than the input frame rate. In the example above, the block receives inputs only once every three sample periods, but produces an output once every sample period. To rebuffer frame-based inputs to a larger or smaller frame size, use the Buffer block.

In the model below, the block unbuffers a four-channel frame-based input with frame size 3. The **Initial conditions** parameter is set to zero and the tasking mode is set to multitasking, so the first three outputs are zero vectors.



Zero Latency

The Unbuffer block has *zero tasking latency* in the Simulink® single-tasking mode. Zero tasking latency means that the first input sample (received at *t*=0) appears as the first output sample.

Nonzero Latency

For *multitasking* operation, the Unbuffer block's buffer is initialized with the value specified by the **Initial condition** parameter, and the block begins unbuffering this frame at the start of the simulation. Inputs to the block are therefore delayed by one buffer length, or M_i samples.

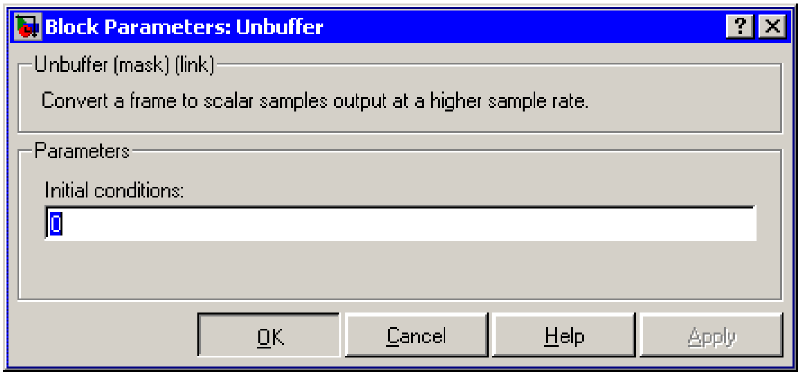
The **Initial condition** parameter can be one of the following:

- A scalar to be repeated for the first M_i output samples of every channel

- A length-M_i vector containing the values of the first M_i output samples for every channel
- An M_i-by-N matrix containing the values of the first M_i output samples in each of N channels

Note For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Dialog Box



Initial conditions

The value of the block's initial output for cases of nonzero latency; a scalar, vector, or matrix.

Supported Data Types

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

See Also

Buffer	Signal Processing Blockset

See "Unbuffering Frame-Based Signals into Sample-Based Signals" for related information.

Uniform Decoder

Purpose

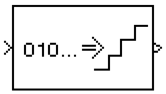
Decode integer input into floating-point output

Library

Quantizers

`dspquant2`

Description



The Uniform Decoder block performs the inverse operation of the Uniform Encoder block, and reconstructs quantized floating-point values from encoded integer input. The block adheres to the definition for uniform decoding specified in ITU-T Recommendation G.701.

Inputs can be real or complex values of the following six integer data types: `uint8`, `uint16`, `uint32`, `int8`, `int16`, or `int32`.

The block first casts the integer input values to floating-point values, and then uniquely maps (decodes) them to one of 2^B uniformly spaced floating-point values in the range $[-V, (1-2^{1-B})V]$, where you specify B in the **Bits** parameter (as an integer between 2 and 32) and V is a floating-point value specified by the **Peak** parameter. The smallest input value representable by B bits (0 for an unsigned input data type; -2^{B-1} for a signed input data type) is mapped to the value -V. The largest input value representable by B bits (2^B-1 for an unsigned input data type; $2^{B-1}-1$ for a signed input data type) is mapped to the value $(1-2^{1-B})V$. Intermediate input values are linearly mapped to the intermediate values in the range $[-V, (1-2^{1-B})V]$.

To correctly decode values encoded by the Uniform Encoder block, the **Bits** and **Peak** parameters of the Uniform Decoder block should be set to the same values as the **Bits** and **Peak** parameters of the Uniform Encoder block. The **Overflow mode** parameter specifies the Uniform Decoder block's behavior when the integer input is outside the range representable by B bits. When you select **Saturate**, *unsigned* input values greater than 2^B-1 saturate at 2^B-1; *signed* input values greater than $2^{B-1}-1$ or less than -2^{B-1} saturate at those limits. The real and imaginary components of complex inputs saturate independently.

When you select **Wrap**, *unsigned* input values, `u`, greater than 2^B-1 are wrapped back into the range $[0, 2^B-1]$ using mod-2^B arithmetic.

```
u = mod(u,2^B)
```

Signed input values, u, greater than 2^{B-1}-1 or less than -2^{B-1} are wrapped back into that range using mod-2^B arithmetic.

```
u = (mod(u+2^B/2,2^B)-(2^B/2))
```

The real and imaginary components of complex inputs wrap independently.

The **Output type** parameter specifies whether the decoded floating-point output is single or double precision. Either level of output precision can be used with any of the six integer input data types.

Examples

Consider a Uniform Decoder block with the following parameter settings:

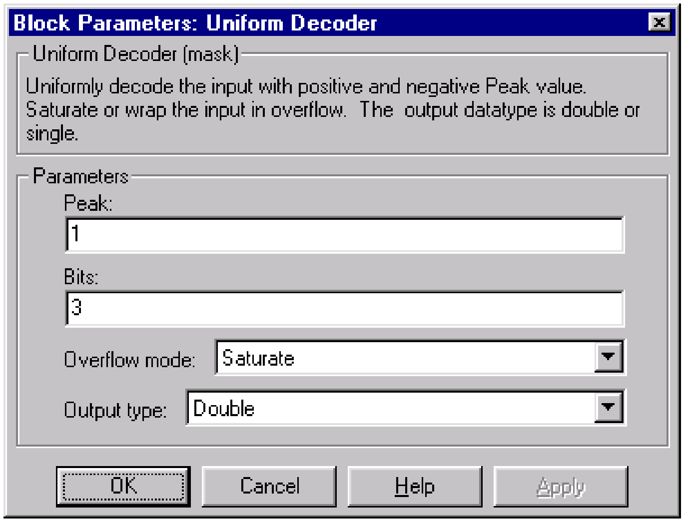
- **Peak** = 2
- **Bits** = 3

The input to the block is the `uint8` output of a Uniform Encoder block with comparable settings: **Peak** = 2, **Bits** = 3, and **Output type** = `Unsigned`. (Comparable settings ensure that inputs to the Uniform Decoder block do not saturate or wrap. See the example on the Uniform Encoder block reference page for more about these settings.)

The real and complex components of each input are independently mapped to one of 2^3 distinct levels in the range `[-2.0,1.5]`.

```
0  is mapped to  -2.0
1  is mapped to  -1.5
2  is mapped to  -1.0
3  is mapped to  -0.5
4  is mapped to   0.0
5  is mapped to   0.5
6  is mapped to   1.0
7  is mapped to   1.5
```

Dialog Box



Peak

The largest amplitude represented in the encoded input. To correctly decode values encoded with the Uniform Encoder block, set the **Peak** parameters in both blocks to the same value.

Bits

The number of input bits, B, used to encode the data. (This can be less than the total number of bits supplied by the input data type.) To correctly decode values encoded with the Uniform Encoder block, set the **Bits** parameters in both blocks to the same value.

Overflow mode

The block's behavior when the integer input is outside the range representable by B bits. Out-of-range inputs can either saturate at the extreme value, or wrap back into range.

Output type

The precision of the floating-point output, `single` or `double`.

References

General Aspects of Digital Transmission Systems: Vocabulary of Digital Transmission and Multiplexing, and Pulse Code Modulation

(PCM) Terms, International Telecommunication Union, ITU-T Recommendation G.701, March, 1993

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

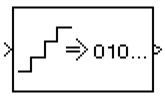
Data Type Conversion	Simulink
Quantizer	Simulink
Scalar Quantizer Decoder	Signal Processing Blockset
Uniform Encoder	Signal Processing Blockset
`udecode`	Signal Processing Toolbox
`uencode`	Signal Processing Toolbox

Purpose Quantize and encode floating-point input into integer output

Library Quantizers

`dspquant2`

Description The Uniform Encoder block performs the following two operations on each floating-point sample in the input vector or matrix:



1. Quantizes the value using the same precision
2. Encodes the quantized floating-point value to an integer value

In the first step, the block quantizes an input value to one of 2^B uniformly spaced levels in the range $[-V, (1-2^{1-B})V]$, where you specify B in the **Bits** parameter and you specify V in the **Peak** parameter. The quantization process rounds both positive and negative inputs *downward* to the nearest quantization level, with the exception of those that fall exactly on a quantization boundary. The real and imaginary components of complex inputs are quantized independently.

The number of bits, B, can be any integer value between 2 and 32, inclusive. Inputs greater than $(1-2^{1-B})V$ or less than -V saturate at those respective values. The real and imaginary components of complex inputs saturate independently.

In the second step, the quantized floating-point value is uniquely mapped (encoded) to one of 2^B integer values. When the **Output type** is set to `Unsigned integer`, the smallest quantized floating-point value, -V, is mapped to the integer 0, and the largest quantized floating-point value, $(1-2^{1-B})V$, is mapped to the integer 2^B-1. Intermediate quantized floating-point values are linearly (uniformly) mapped to the intermediate integers in the range $[0, 2^B-1]$. For efficiency, the block automatically selects an *unsigned* output data type (`uint8`, `uint16`, or `uint32`) with the minimum number of bits equal to or greater than B.

When the **Output type** is set to `Signed integer`, the smallest quantized floating-point value, -V, is mapped to the integer -2^{B-1}, and the largest quantized floating-point value, $(1-2^{1-B})V$, is mapped to the integer 2^{B-1}-1. Intermediate quantized floating-point values are linearly mapped to the intermediate integers in the range [-2^{B-1}, 2^{B-1}-1]. The block automatically selects a *signed* output data type (`int8`, `int16`, or `int32`) with the minimum number of bits equal to or greater than B.

Inputs can be real or complex, double or single precision. The output data types that the block uses are shown in the table below. Note that most of the Signal Processing Blockset™ blocks accept only double-precision inputs. Use the Simulink® Data Type Conversion block to convert integer data types to double precision. See "Working with Data Types" in the Simulink documentation for a complete discussion of data types, as well as a list of Simulink blocks capable of reduced-precision operations.

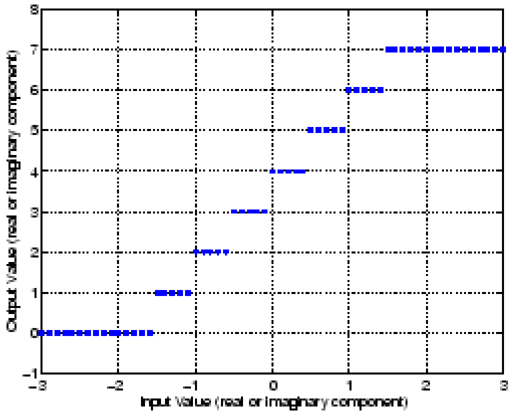
Bits	Unsigned Integer	Signed Integer
2 to 8	`uint8`	`int8`
9 to 16	`uint16`	`int16`
17 to 32	`uint32`	`int32`

The Uniform Encoder block operations adhere to the definition for uniform encoding specified in ITU-T Recommendation G.701.

Examples

The following figure illustrates uniform encoding with the following parameter settings:

- **Peak** = 2
- **Bits** = 3
- **Output type** = `Unsigned`



The real and complex components of each input (horizontal axis) are independently quantized to one of 2^3 distinct levels in the range `[-2,1.5]` and then mapped to one of 2^3 integer values in the range `[0,7]`.

```
-2.0  is mapped to  0
-1.5  is mapped to  1
-1.0  is mapped to  2
-0.5  is mapped to  3
 0.0  is mapped to  4
 0.5  is mapped to  5
 1.0  is mapped to  6
 1.5  is mapped to  7
```

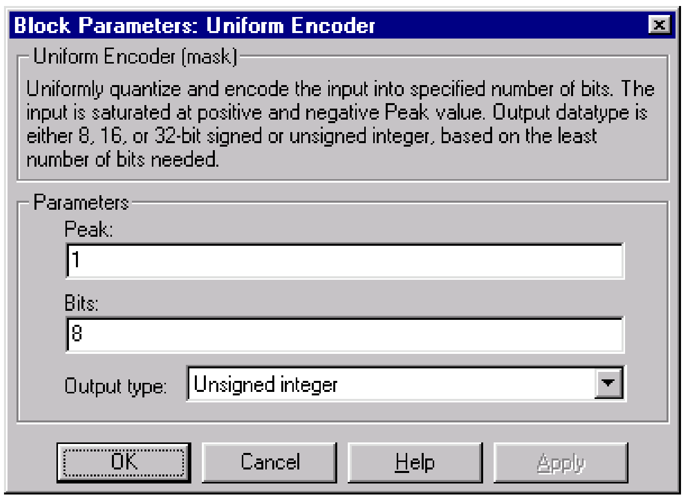
The table below shows the results for a few particular inputs.

Input	Quantized Input	Output	Notes
1.6	1.5+0.0i	7+4i	
-0.4	-0.5+0.0i	3+4i	
-3.2	-2.0+0.0i	4i	Saturation (real)

Input	Quantized Input	Output	Notes
`0.4-1.2i`	`0.0-1.5i`	`4+i`	
`0.4-6.0i`	`0.0-2.0i`	`4`	Saturation (imaginary)
`-4.2+3.5i`	`-2.0+2.0i`	`7i`	Saturation (real and imaginary)

The output data type is automatically set to `uint8`, the most efficient format for this input range.

Dialog Box



Peak

The largest input amplitude to be encoded, V. Real or imaginary input values greater than $(1\text{-}2^{1\text{-}B})$V or less than -V saturate (independently for complex inputs) at those limits.

Bits

The number of levels at which to quantize the floating-point input. (Also the number of bits needed to represent the integer output.)

Output type

The data type of the block's output, `Unsigned integer` or `Signed integer`. Unsigned outputs are `uint8`, `uint16`, or `uint32`, while signed outputs are `int8`, `int16`, or `int32`.

References

General Aspects of Digital Transmission Systems: Vocabulary of Digital Transmission and Multiplexing, and Pulse Code Modulation (PCM) Terms, International Telecommunication Union, ITU-T Recommendation G.701, March, 1993

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Data Type Conversion	Simulink
Quantizer	Simulink
Scalar Quantizer Decoder	Signal Processing Blockset
Uniform Decoder	Signal Processing Blockset
`udecode`	Signal Processing Toolbox
`uencode`	Signal Processing Toolbox

Unwrap

Purpose

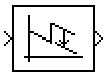
Unwrap signal phase

Library

Signal Operations

`dspsigops`

Description



The Unwrap block unwraps each input channel by adding or subtracting appropriate multiples of 2π to each channel element. The input can be any matrix or 1-D vector, and must have radian phase entries. The block recognizes phase discontinuities larger than the **Tolerance** parameter setting.

The block preserves the input size, dimension, and frame status, and the output port rate equals the input port rate. For a detailed discussion of the Unwrap block, see other sections of this reference page.

Sections of This Reference Page

- “Acceptable Inputs and Corresponding Output Characteristics”
- “The Two Unwrap Modes”
- “Unwrap Method”
- “Definition of Phase Unwrap”

Acceptable Inputs and Corresponding Output Characteristics

The Unwrap block preserves the input size, dimension, and frame status, and the output port rate equals the input port rate.

Characteristics of Valid Input	Characteristics of Corresponding Output
Input elements must be phase values in radians. Sample- or frame-based M-by-N 2-D matrix or a 1-D vector	Output elements are phase values in radians. Same frame status as input Same size and dimension as input Output port rate = input port rate

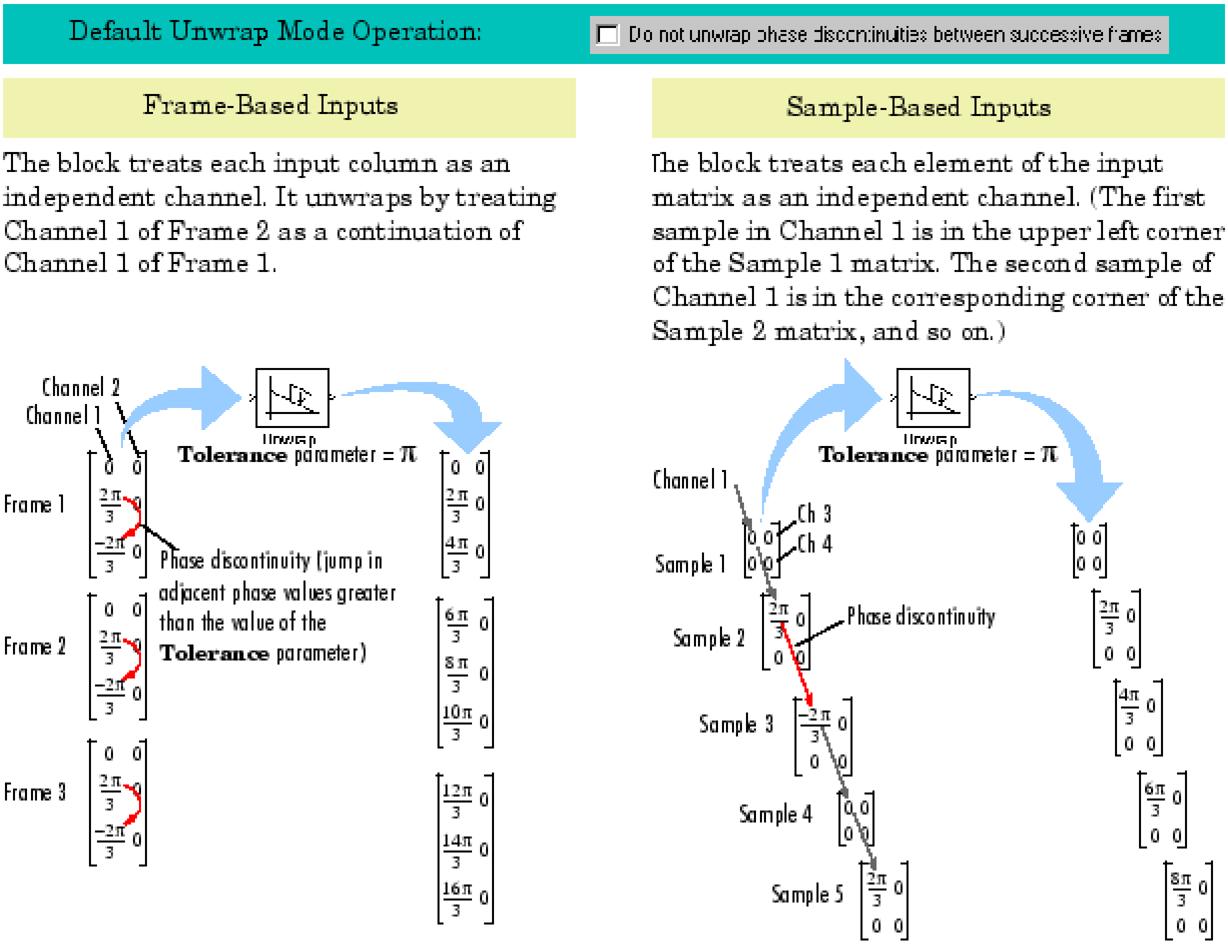
The Two Unwrap Modes

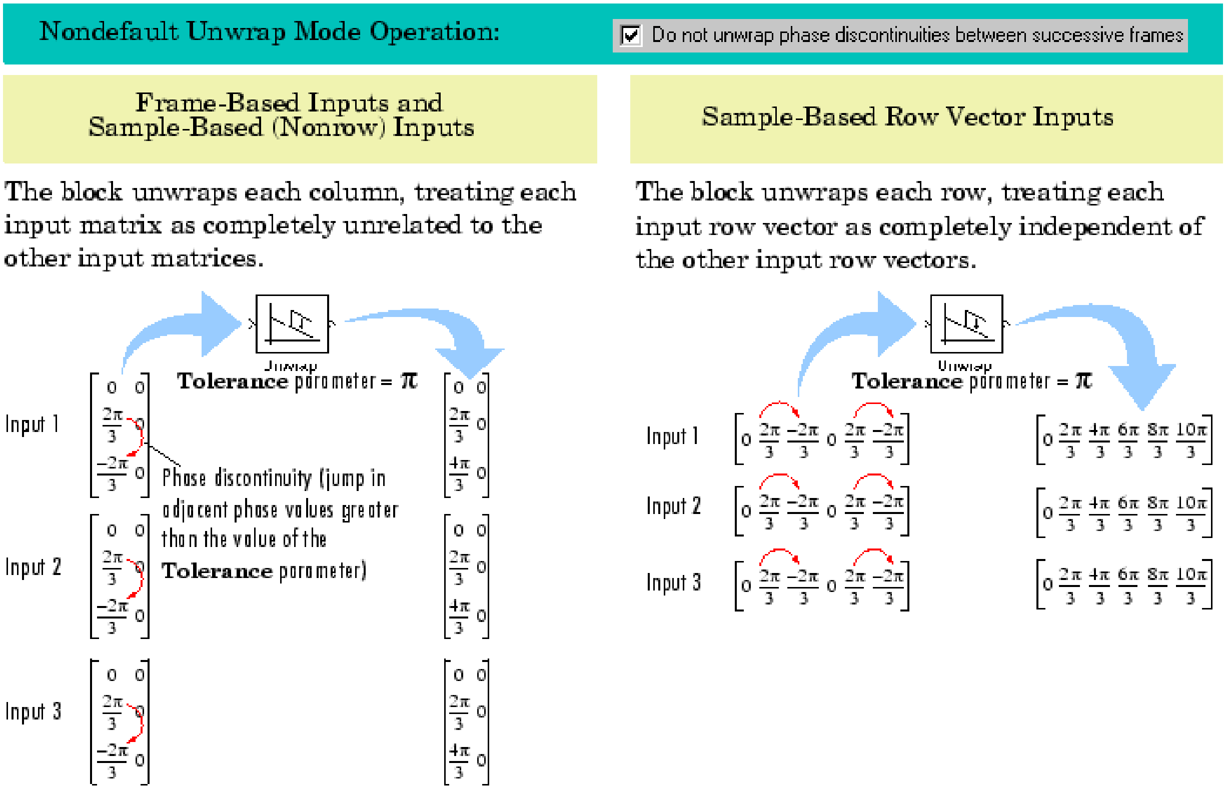
You must specify the unwrap mode by setting the parameter, **Do not unwrap phase discontinuities between successive frames**. The unwrap modes are summarized in the next table.

Two Unwrap Modes	
In both unwrap modes, the block adds 2π*k* to each input channel's elements, where it updates *k* at each phase discontinuity. (For more on the updating of *k*, see "Unwrap Method" on page 2-1269.) The number of times that *k* is reset to 0 depends on the unwrap mode.	
Default Unwrap Mode: Initialize k to 0 for Only the First Input Frame	**Nondefault Unwrap Mode: Set k to 0 for Each Successive Input Matrix or Input Vector**
☐ Do not unwrap phase discontinuities between successive frames In this mode, *k* is initialized to 0 for only the first input matrix or input vector. As *k* gets updated, the value of *k* is retained between successive input matrices or input vectors. That is, the block unwraps each input's channel by considering phase discontinuities in all previous frames and the current frame.	☑ Do not unwrap phase discontinuities between successive frames In this mode, *k* is reset to 0 for each successive input matrix or input vector. As *k* gets updated, the value of *k* is only retained within the current input matrix or vector. That is, the block unwraps each input's channel by considering phase discontinuities in the current input matrix or input vector only, ignoring discontinuities in previous inputs.

Two Unwrap Modes	
In both unwrap modes, the block adds $2\pi k$ to each input channel's elements, where it updates *k* at each phase discontinuity. (For more on the updating of *k*, see "Unwrap Method" on page 2-1269.) The number of times that *k* is reset to 0 depends on the unwrap mode.	
Default Unwrap Mode: Initialize k to 0 for Only the First Input Frame	**Nondefault Unwrap Mode: Set k to 0 for Each Successive Input Matrix or Input Vector**
In this mode, the block unwraps the columns or each individual element of the input: • Frame-based inputs — unwrap columns • Sample-based inputs — unwrap each element of the input. • 1-D vector inputs — treat as frame-based column	In this mode, the block unwraps the columns or rows of the input: • Frame-based inputs — unwrap columns • Sample-based nonrow inputs — unwrap columns • Sample-based row vector inputs — unwrap the row. • 1-D vector inputs — treat as frame-based column
See the following diagrams.	See the following diagrams.

The following diagrams illustrate how the two unwrap modes operate on various inputs.





Unwrap Method

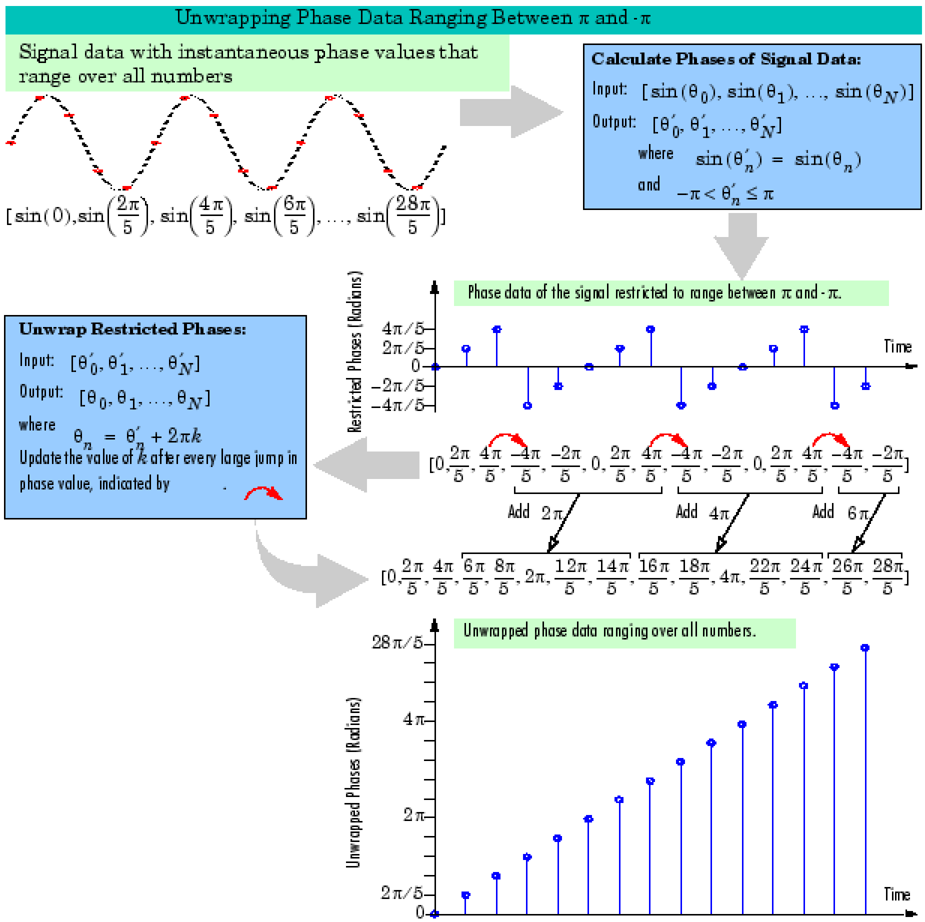
The Unwrap block unwraps each channel of its input matrix or input vector by adding $2\pi k$ to each successive channel element, and updating k at each *phase jump*. See the following steps to the unwrap method for details.

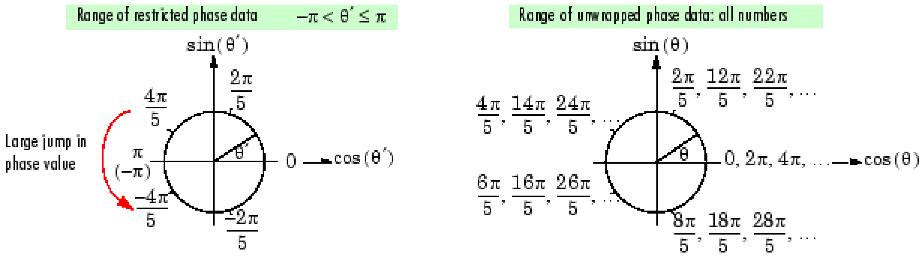
Relevant Unwrap Terms:
• u_i — ith element of the input channel on which the algorithm operates • α —**Tolerance** parameter value • Phase jump or phase discontinuity — difference between phase values of two adjacent channel entries that exceeds α. The diagram in the next section indicates phase jumps with red arrows.
Steps to the Unwrap Method:
1 Set k to 0 (See "The Two Unwrap Modes" on page 2-1266 for more on how often this step occurs.) **2** Check for a phase jump between adjacent channel elements u_i and u_{i+1}: • When there is no phase jump between u_i and u_{i+1} $\left(\lvert u_{i+1} - u_i \rvert \le \lvert \alpha \rvert\right)$, add 2π$k$ to u_i, and then repeat step 2 to continue checking for phase jumps. • When there is a phase jump between u_i and u_{i+1} $\left(\lvert u_{i+1} - u_i \rvert > \lvert \alpha \rvert\right)$, add 2π$k$ to u_i, and then go to step 3 to update k. **3** Update k as follows when there is a phase jump between u_i and u_{i+1}. Then go back to step 2 to add the updated 2πk value to u_{i+1} and succeeding channel elements until the next phase jump: • When $u_{i+1} < u_i$ (phase jump is negative), increment k. • When $u_{i+1} > u_i$ (phase jump is positive), decrement k.

Definition of Phase Unwrap

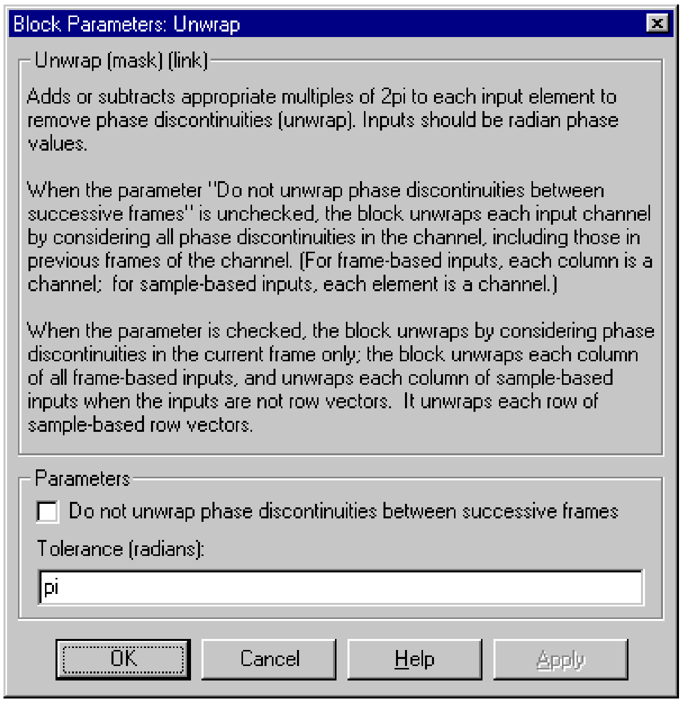
Algorithms that compute the phase of a signal often only output phases between –π and π. For instance, such algorithms compute the phase of sin(2π + 3) to be 3, since sin(3) = sin(2π + 3), and since the actual phase, 2π + 3, is not between –π and π. Such algorithms compute the phases of sin(–4π + 3) and sin(16π + 3) to be 3 as well.

Phase unwrap or unwrap is a process often used to reconstruct a signal's original phase. Unwrap algorithms add appropriate multiples of 2π to each phase input to restore original phase values, as illustrated in the following diagram. For more on phase unwrap, see the previous section, "Unwrap Method" on page 2-1269.





Dialog Box



Do not unwrap phase discontinuities between successive frames

When this parameter is cleared, the block unwraps each input's channels (the input channels are the columns of frame-based inputs and each element of sample-based inputs). When you select this parameter, the block unwraps each row of sample-based row vector inputs, and unwraps the columns of all other inputs, where each input matrix or input vector is treated as completely unrelated to the other input matrices or input vectors. 1-D vector inputs are always treated as frame-based column vectors. See "The Two Unwrap Modes" on page 2-1266.

Tolerance

The jump size that the block recognizes as a true phase discontinuity. The default is set to π (rather than a smaller value) to avoid altering legitimate signal features. To increase the block's sensitivity, set **Tolerance** to a value slightly less than π.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

`unwrap`	MATLAB

Upsample

Purpose

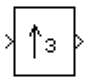
Resample input at higher rate by inserting zeros

Library

Signal Operations

`dspsigops`

Description



The Upsample block resamples each channel of the M_i-by-N input at a rate L times higher than the input sample rate by inserting L-1 zeros between consecutive samples. You specify the integer L in the **Upsample factor** parameter. The **Sample offset** parameter delays the output samples by an integer number of sample periods D, where

$0 \leq D < (L-1)$, so that any of the L possible output phases can be selected.

This block supports triggered subsystems if, for **Frame-based mode**, you select `Maintain input frame rate`.

Sample-Based Operation

When the input is sample based, the block treats each of the M*N matrix elements as an independent channel, and upsamples each channel over time. The **Frame-based mode** parameter must be set to `Maintain input frame size`. The output sample rate is L times higher than the input sample rate ($T_{so} = T_{si}/L$), and the input and output sizes are identical.

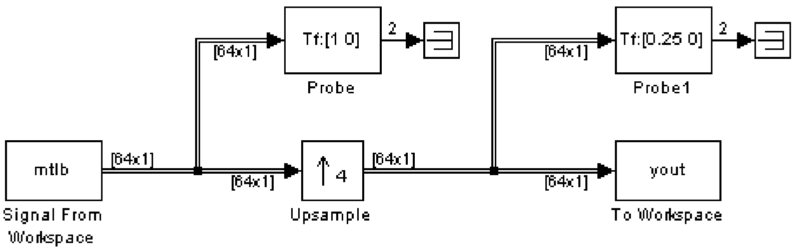
Frame-Based Operation

When the input is frame based, the block treats each of the N input columns as a frame containing M_i sequential time samples from an independent channel. The block upsamples each channel independently by inserting L-1 rows of zeros between each row in the input matrix. The **Frame-based mode** parameter determines how the block adjusts the rate at the output to accommodate the added rows. There are two available options:

- `Maintain input frame size`

The block generates the output at the faster (upsampled) rate by using a proportionally shorter frame *period* at the output port than at the input port. For upsampling by a factor of L, the output frame period is L times shorter than the input frame period ($T_{fo} = T_{fi}/L$), but the input and output frame sizes are equal.

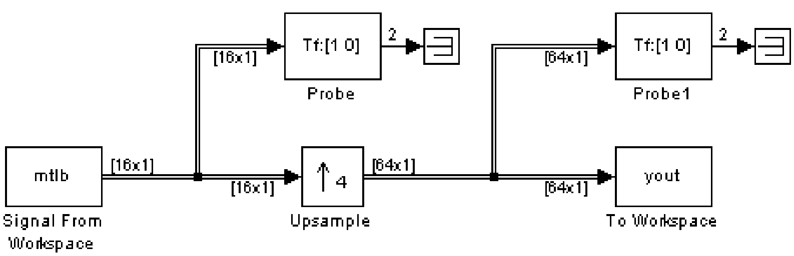
The model below shows a single-channel input with a frame period of 1 second being upsampled by a factor of 4 to a frame period of 0.25 second. The input and output frame sizes are identical.



- `Maintain input frame rate`

  The block generates the output at the faster (upsampled) rate by using a proportionally larger frame *size* than the input. For upsampling by a factor of L, the output frame size is L times larger than the input frame size ($M_o = M_i$*L), but the input and output frame rates are equal.

  The model below shows a single-channel input of frame size 16 being upsampled by a factor of 4 to a frame size of 64. The input and output frame rates are identical.



Zero Latency

The Upsample block has *zero tasking latency* for all single-rate operations. The block is single-rate for the particular combinations of sampling mode and parameter settings shown in the table below.

Sampling Mode	Parameter Settings
Sample based	**Upsample factor** parameter, L, is `1`.
Frame based	**Upsample factor** parameter, L, is `1`, *or* **Frame-based mode** parameter is `Maintain input frame rate`.

The block also has zero latency for all multirate operations in the Simulink® single-tasking mode.

Zero tasking latency means that the block propagates the first input (received at *t*=0) immediately following the D consecutive zeros specified by the **Sample offset** parameter. This output (D+1) is followed in turn by the L-1 inserted zeros and the next input sample. The **Initial condition** parameter value is not used.

Nonzero Latency

The Upsample block has tasking latency only for multirate operation in the Simulink multitasking mode:

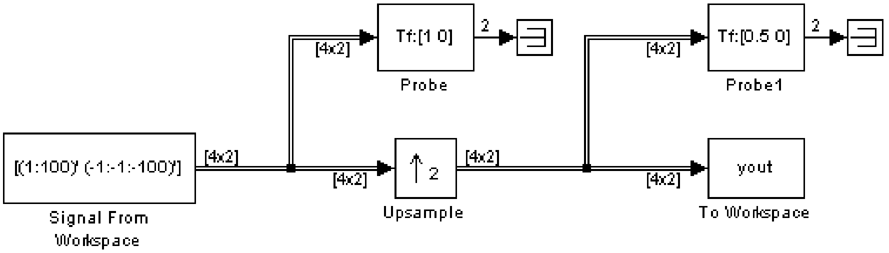
- In sample-based mode, the initial condition for each channel appears as output sample D+1, and is followed by L-1 inserted zeros. The channel's first input appears as output sample D+L+1. The **Initial condition** value can be an M_i-by-N matrix containing one value for each channel, or a scalar to be applied to all signal channels.
- In frame-based mode, the first row of the initial condition matrix appears as output sample D+1, and is followed by L-1 inserted rows of zeros, the second row of the initial condition matrix, and so on. The first row of the first input matrix appears in the output as sample M_iL+D+1. The **Initial condition** value can be an M_i-by-N matrix, or

a scalar to be repeated across all elements of the M_i-by-N matrix. See the example below for an illustration of this case.

Note For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and "Models with Multiple Sample Rates" in the *Real-Time Workshop® User's Guide*.

Examples

Construct the frame-based model shown below.



Adjust the block parameters as follows:

- Configure the Signal From Workspace block to generate a two-channel signal with frame size of 4 and sample period of 0.25. This represents an output frame period of 1 (0.25*4). The first channel should contain the positive ramp signal 1, 2, ..., 100, and the second channel should contain the negative ramp signal -1, -2, ..., -100.
  - **Signal** = `[(1:100)' (-1:-1:-100)']`
  - **Sample time** = `0.25`
  - **Samples per frame** = `4`
- Configure the Upsample block to upsample the two-channel input by increasing the output frame rate by a factor of 2 relative to the input frame rate. Set a sample offset of 1, and an initial condition matrix of

$$\begin{bmatrix} 11 & -11 \\ 12 & -12 \\ 13 & -13 \\ 14 & -14 \end{bmatrix}$$

- **Upsample factor** = `2`
- **Sample offset** = `1`
- **Initial condition** = `[11 -11;12 -12;13 -13;14 -14]`
- **Frame-based mode** = `Maintain input frame size`

- Configure the Probe blocks by clearing the **Probe width** and **Probe complex signal** check boxes (if desired).

This model is multirate because there are at least two distinct frame rates, as shown by the two Probe blocks. To run this model in the Simulink multitasking mode, open the Configuration Parameters dialog box. In the **Select** pane, click **Solver**. From the **Type** list, select `Fixed-step`, and from the **Solver** list, select `discrete (no continuous states)`. From the **Tasking mode for periodic sample times** list, select `MultiTasking`. Also set the **Stop time** to `30`.

Run the model and look at the output, `yout`. The first few samples of each channel are shown below.

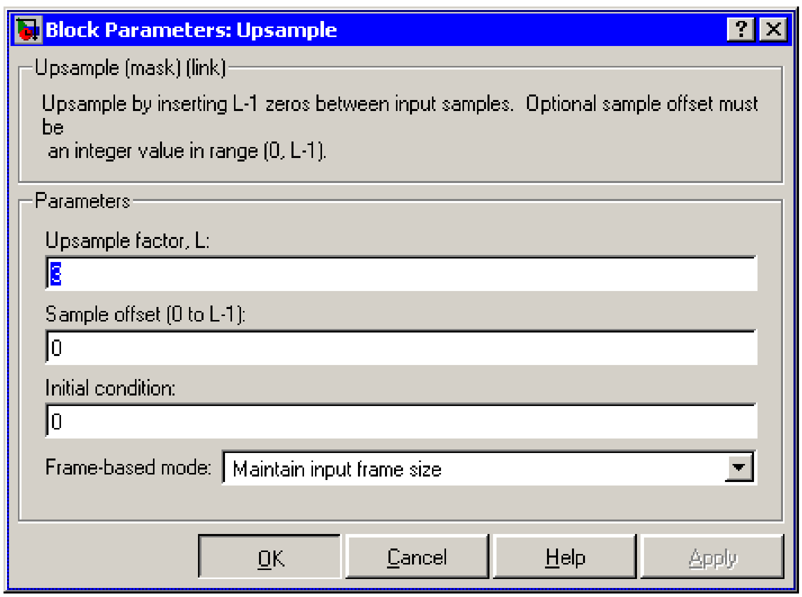
```
yout =

      0       0
     11     -11
      0       0
     12     -12
      0       0
     13     -13
      0       0
     14     -14
      0       0
      1      -1
```

0	0
2	-2
0	0
3	-3
0	0
4	-4
0	0
5	-5
0	0

Since we ran this frame-based multirate model in multitasking mode, the first row of the initial condition matrix appears as output sample 2 (that is, sample D+1, where D is the **Sample offset** value). It is followed by the other three initial condition rows, each separated by L-1 inserted rows of zeros, where L is the **Upsample factor** value of 2. The first row of the first input matrix appears in the output as sample 10 (that is, sample M_iL+D+1, where M_i is the input frame size).

Dialog Box



Upsample factor
The integer factor, L, by which to increase the input sample rate.

Sample offset
The sample offset, D, which must be an integer in the range [0,L-1].

Initial condition
The value with which the block is initialized for cases of nonzero latency, a scalar or matrix. This value (first row in frame-based mode) appears in the output as sample D+1.

Frame-based mode
For frame-based operation, the method by which to implement the upsampling: `Maintain input frame size` (that is, increase the frame rate), or `Maintain input frame rate` (that is, increase the frame size). The **Framing** parameter must be set to `Maintain input frame size` for sample-base inputs.

Supported Data Types

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

See Also

Downsample	Signal Processing Blockset
FIR Interpolation	Signal Processing Blockset
FIR Rate Conversion	Signal Processing Blockset
Repeat	Signal Processing Blockset

Variable Fractional Delay

Purpose

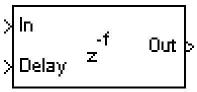
Delay input by time-varying fractional number of sample periods

Library

Signal Operations

`dspsigops`

Description



The Variable Fractional Delay block delays each element of the N-D input array, u, by a variable (possibly noninteger) number of sample intervals.

The block computes the value for each channel of the output based on the stored samples in memory most closely indexed by the `Delay` input, v, and the interpolation method specified by the **Mode** parameter. In `Linear Interpolation` mode, the block stores the D+1 most recent samples received at the In port for each channel, where D is the **Maximum delay**. In `FIR Interpolation` mode, the block stores the D+P+1 most recent samples received at the In port for each channel, where P is the **Interpolation filter half-length**.

See the Variable Integer Delay block for further discussion of how input samples are stored in the block's memory. The Variable Fractional Delay block differs only in the way that these stored sample are accessed; a fractional delay requires the computation of a value by interpolation from the nearby samples in memory.

Sample-Based Operation

When the input is sample based, the block treats each element of the N-D input array, u, as an independent channel. The input to the Delay port, v, must either be an N-D array of the same size and dimension as the input u, or be a scaler value, such that $0 \leq v \leq$ D.

For example, consider an M-by-N input matrix. Each of the M_i*N matrix elements are treated as independent channels. The input to the Delay port can be an M_i-by-N matrix of floating-point values in the range $0 \leq v \leq D$ that specifies the number of sample intervals to delay each channel of the input, or it can be a scalar floating-point value, $0 \leq v \leq$ D, by which to equally delay all channels.

A 1-D vector input is treated as an M_i-by-1 matrix, and the output is 1-D.

The **Initial conditions** parameter specifies the values in the block's memory at the start of the simulation in the same manner as the Variable Integer Delay block. See the Variable Integer Delay block reference page for more information.

Frame-Based Operation

When the input is frame based, the block treats each of the N input columns as a frame containing M_i sequential time samples from an independent channel.

The input to the Delay port, v, contains floating-point values in the range $0 \leq v \leq D$ that specify the number of sample intervals to delay the current input. The input to the Delay port can be a scalar value to uniformly delay every sample in every channel. It can also be a column-based length-M vector, containing one delay for each sample in the input frame(s). The set of delays contained in the vector is applied identically to every channel of a multichannel input. The Delay port entry can also be a row-based length-N vector, containing one delay for each channel. Finally, the Delay port entry can also be an M-by-N matrix, containing a different delay for each corresponding element of the input.

For example, if v is the M_i-by-1 matrix `[v(1) v(2) ... v(Mi)]'`, the earliest sample in the current frame is delayed by `v(1)` fractional sample intervals, the following sample in the frame is delayed by `v(2)` fractional sample intervals, and so on. The set of fractional delays contained in v is applied identically to every channel of a multichannel input.

The **Initial conditions** parameter specifies the values in the block's memory at the start of the simulation in the same manner as the Variable Integer Delay block. See the Variable Integer Delay block reference page for more information.

Interpolation Modes

The delay value specified at the Delay port is used as an index into the block's memory, `U`, which stores the D+1 most recent samples received at the In port for each channel. For example, an integer delay of `5` on a scalar input sequence retrieves and outputs the fifth most recent input sample from the block's memory, `U(6)`. Fractional delays are computed by interpolating between stored samples; the two available interpolation modes are as follows.

Linear Interpolation Mode

For noninteger delays, at each sample time, the `Linear Interpolation` mode uses the two samples in memory nearest to the specified delay to compute a value for the sample at that time. If `v` is the specified fractional delay for a scalar input, the output sample, `y`, is computed as follows.

```
vi = floor(v)     % vi = integer delay
vf = v-vi     % vf = fractional delay
y = (1-vf)*U(vi+1) + vf*U(vi)
```

Delay values less than 0 are clipped to 0, and delay values greater than D are clipped to D, where D is the **Maximum delay**. A delay value of 0 causes the block to pass through the current input sample, `U(1)`, in the same simulation step that it is received.

FIR Interpolation Mode

In `FIR Interpolation` mode, the block computes a value for the sample at the desired delay by applying an FIR filter of order $2P$ to the stored samples on either side of the desired delay, where P is the **Interpolation filter half-length**. For periodic signals, a larger value of P (that is, a higher order filter) yields a better estimate of the sample at the specified delay. A value between 4 and 6 for this parameter (that is, a 7th to 11th order filter) is usually adequate.

A vector of $2P$ filter tap weights is precomputed at the start of the simulation for each of Q-1 discrete points between input samples, where you specify Q in the **Interpolation points per input sample** parameter. For a delay corresponding to one of the Q interpolation points, the unique filter computed for that interpolation point is applied to obtain a value for the sample at the specified delay. For delay times that fall between interpolation points, the value computed at the nearest interpolation point is used. Since Q controls the number of locations where a unique interpolation filter is designed, a larger value results in a better estimate of the sample at a given delay.

Increasing the **Interpolation filter half length** (*P*) increases the number of computations performed per input sample, as well as the amount of memory needed to store the filter coefficients. Increasing the **Interpolation points per input sample** (*Q*) increases the simulation's memory requirements, but does not affect the computational load per sample.

The **Normalized input bandwidth** parameter allows you to take advantage of the bandlimited frequency content of the input. For example, if you know that the input signal does not have frequency content above $F_s/4$, you can specify a value of `0.5` for the **Normalized input bandwidth** to constrain the frequency content of the output to that range.

Note Each of the *Q* interpolation filters can be considered to correspond to one output phase of an "upsample-by-*Q*" FIR filter. In this view, the **Normalized input bandwidth** value improves the stopband in critical regions, and relaxes the stopband requirements in frequency regions where there is no signal energy.

For delay values less than P-1, the output is computed using linear interpolation. Delay values greater than *D* are clipped to *D*, where *D* is the **Maximum delay**.

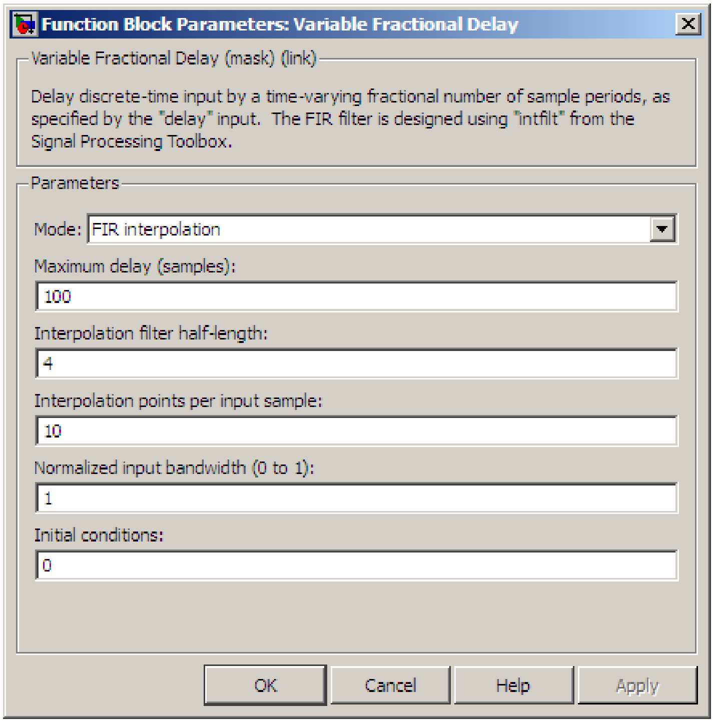
The block uses the Signal Processing Toolbox™ `intfilt` function to compute the FIR filters.

Note When the Variable Fractional Delay block is used in a feedback loop, at least one block with nonzero delay (for example, a Delay block with **Delay** > `0`) should be included in the loop as well. This prevents the occurrence of an algebraic loop when the delay of the Variable Fractional Delay block is driven to zero.

Examples

The `dspaudioeffects` demo illustrates three audio effects applied to a short segment of music. When you set the **Audio effect** of the Effect block to `Flanging`, the model uses the Variable Fractional Delay block to mix the original signal with a delayed version of itself. To see the Flanging subsystem, right-click the Effect block and select **Look Under Mask**. Next, double-click the Flanging block in the Effect block subsystem that just opened. This opens the Flanging subsystem and allows you to see the parameters of the Variable Fractional Delay block.

Dialog Box



Mode

The method by which to interpolate between adjacent stored samples to obtain a value for the sample indexed by the input at the Delay port.

Maximum delay

The maximum delay that the block can produce, D. Delay input values exceeding this maximum are clipped at the maximum.

Interpolation filter half-length

Half the number of input samples to use in the FIR interpolation filter.

Interpolation points per input sample
The number of points per input sample, Q, at which a unique FIR interpolation filter is computed.

Normalized input bandwidth
The bandwidth to which the interpolated output samples should be constrained. A value of 1 specifies half the sample frequency.

Initial conditions
The values with which the block's memory is initialized. See the Variable Integer Delay block for more information.

Supported Data Types

- Double-precision floating point
- Single-precision floating point

See Also

Delay	Signal Processing Blockset
Unit Delay	Simulink
Variable Integer Delay	Signal Processing Blockset

Variable Integer Delay

Purpose

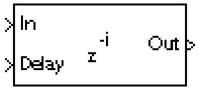
Delay input by time-varying integer number of sample periods

Library

Signal Operations

`dspsigops`

Description



The Variable Integer Delay block delays the discrete-time input at the In port by the integer number of sample intervals specified by the input to the Delay port. The sample rate of the input signal at the Delay port must be the same as the sample rate of the input signal at the In port. When these sample rates are not the same, you need to insert a Zero-Order Hold or Rate Transition block in order to make the sample rates identical. The delay for an N-D sample-based input can be a scalar value to uniformly delay every sample in every channel, or a matrix containing one delay value for each channel of the input. The delay for a frame-based input sequence can be a scalar value to uniformly delay every sample in every channel, a vector containing one delay value for each sample in the input frame, or a vector containing one delay value for each channel in the input frame.

The delay values should be in the range of 0 to D, where D is the **Maximum delay**. Delay values greater than D or less than 0 are clipped to those respective values and noninteger delays are rounded to the nearest integer value.

The Variable Integer Delay block differs from the Delay block in the following ways.

Variable Integer Delay Block	Delay Block
The delay is provided as an input to the Delay port.	You specify the delay as a parameter setting in the dialog box.

Variable Integer Delay Block	Delay Block
Delay can vary with time; for example, for a frame-based input, the *n*th element's delay in the first input frame can differ from the *n*th element's delay in the second input frame.	Delay cannot vary with time; for example, for a frame-based input, the *n*th element's delay is the same for every input frame.
When the Variable Integer Delay block is used in a feedback loop, at least one block with nonzero delay (for example, a Delay block with **Delay** > `0`) should be included in the loop as well. This prevents the occurrence of an algebraic loop when the delay of the Variable Integer Delay block is driven to zero.	You can use the Delay block to break an algebraic loop.

Sample-Based Operation

The Variable Integer Delay block supports N-D input arrays. When the input is an *M*-by-*N*-by-*P* sample-based array, the block treats each of the *M**N*P* elements as independent channels, and applies the delay at the Delay port to each channel.

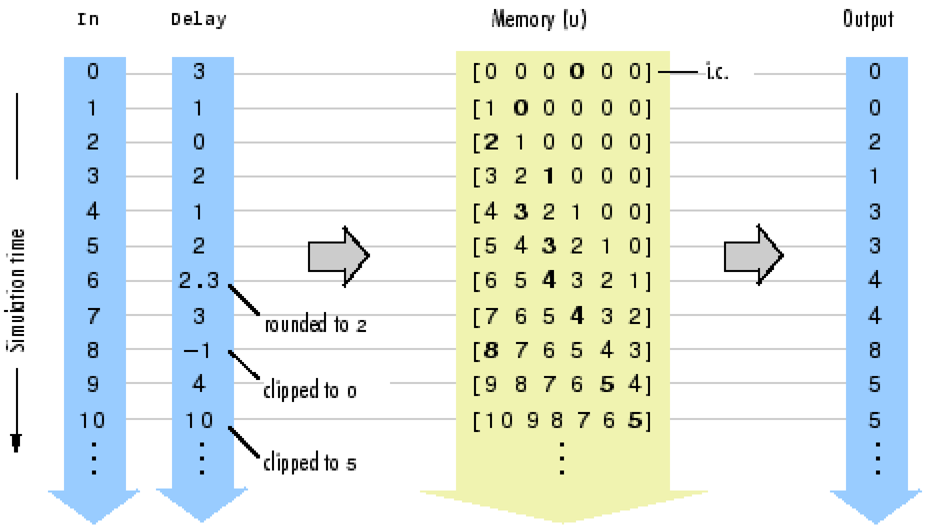
The Variable Integer Delay block stores the D+1 most recent samples received at the In port for each channel. At each sample time the block outputs the stored sample(s) indexed by the input to the Delay port.

For example, when the input to the In port, `u`, is a scalar signal, the block stores a vector, `U`, of the D+1 most recent signal samples. When the current input sample is `U(1)`, the previous input sample is `U(2)`, and so on, then the block's output is

```
y = U(v+1);     % Equivalent MATLAB code
```

where v is the input to the Delay port. A delay value of 0 (v=0) causes the block to pass through the sample at the In port in the same simulation step that it is received. The block's memory is initialized to the **Initial conditions** value at the start of the simulation (see below).

The next figure shows the block output for a scalar ramp sequence at the In port, a **Maximum delay** of 5, an **Initial conditions** of 0, and a variety of different delays at the Delay port.



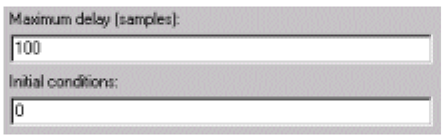
The current input at each time step is immediately stored in memory as `U(1)`. This allows the current input to be available at the output for a delay of 0 (v=0).

The **Initial conditions** parameter specifies the values in the block's memory at the start of the simulation. Unlike the Delay block, the Variable Integer Delay block does not have a fixed initial delay period during which the initial conditions appear at the output. Instead, the initial conditions are propagated to the output only when they are indexed in memory by the value at the Delay port. Both fixed and time-varying initial conditions can be specified in a variety of ways to suit the dimensions of the input sequence.

Fixed Initial Conditions

The settings in this section specify fixed initial conditions. For a fixed initial condition, the block initializes each of D samples in memory to the value entered in the **Initial conditions** parameter. A fixed initial condition in sample-based mode can be specified in one of the following ways:

- Scalar value with which to initialize every sample of every channel in memory. For a general M-by-N input and the parameter settings in this figure,

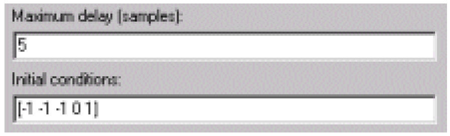
  the block initializes 100 M-by-N matrices in memory with zeros.

- Array of size M-by-N-by-D. In this case, you can specify different fixed initial conditions for each channel. See the Array bullet in "Time-Varying Initial Conditions" on page 2-1292 below for details.

Time-Varying Initial Conditions

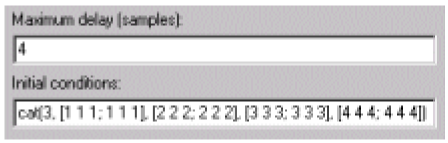
The following settings specify time-varying initial conditions. For a time-varying initial condition, the block initializes each of D samples in memory to one of the values entered in the **Initial conditions** parameter. This allows you to specify a unique output value for each sample in memory. A time-varying initial condition in sample-based mode can be specified in one of the following ways:

- Vector containing D elements with which to initialize memory samples `U(2:D+1)`, where D is the **Maximum delay**. For a scalar input and the parameters in the next figure, the block initializes `U(2:6)` with values `[-1, -1, -1, 0, 1]`.



- Array of dimension M-by-N-by-D with which to initialize memory samples `U(2:D+1)`, where D is the **Maximum delay** and M and N are the number of rows and columns, respectively, in the input matrix. For a 2-by-3 input and the following parameters, the block initializes memory locations `U(2:5)` with values

$$\mathbf{U}(2) = \begin{bmatrix} 1 & 1 & 1 \\ 1 & 1 & 1 \end{bmatrix},\ \mathbf{U}(3) = \begin{bmatrix} 2 & 2 & 2 \\ 2 & 2 & 2 \end{bmatrix},\ \mathbf{U}(4) = \begin{bmatrix} 3 & 3 & 3 \\ 3 & 3 & 3 \end{bmatrix},\ \mathbf{U}(5) = \begin{bmatrix} 4 & 4 & 4 \\ 4 & 4 & 4 \end{bmatrix}$$



An M-by-N-by-P-by-D matrix can be entered for the **Initial Conditions** parameter when the input is an M-by-N-by-P array. The (M,N,P,T)th sample of the **Initial Conditions** matrix provides the initial condition value for the (M,N,P)th channel of the input matrix at delay = $D–t+1$ samples.

Frame-Based Operation

When the input is an M-by-N frame-based matrix, the block treats each of the N input columns as a frame containing M sequential time samples from an independent channel.

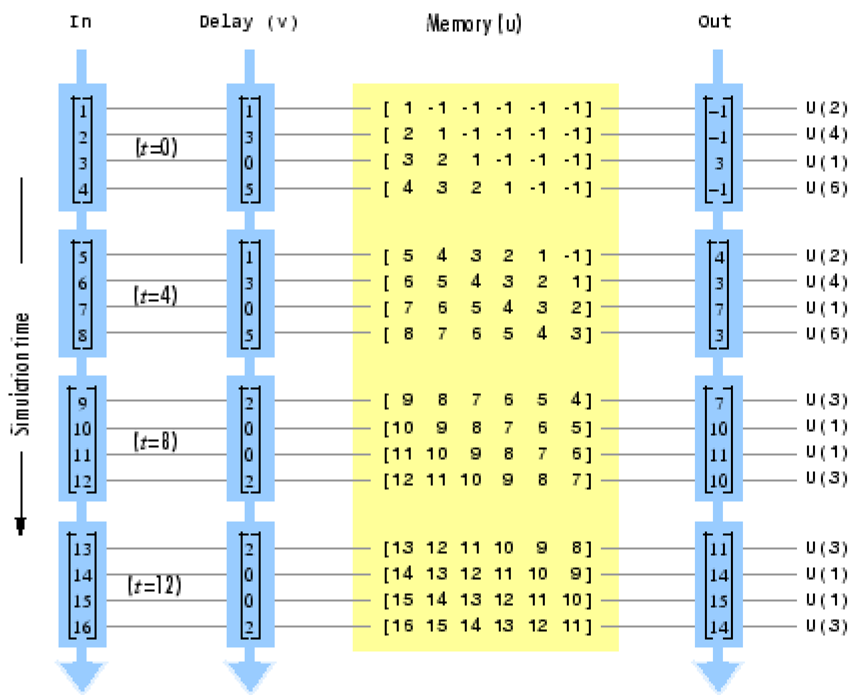
In frame-based mode, the input at the Delay port can be a scalar value to uniformly delay every sample in every channel. It can also be a column-based length-M vector, containing one delay for each sample in the input frame(s). The set of delays contained in the vector is applied identically to every channel of a multichannel input. The Delay port entry can also be a row-based length-N vector, containing one delay for each channel. Finally, the Delay port entry can also be an M-by-N

matrix, containing a different delay for each corresponding element of the input.

Vector `v` does not specify when the samples in the current input frame will appear in the output. Rather, `v` indicates which previous input samples (stored in memory) should be included in the current output frame. The first sample in the current output frame is the input sample `v(1)` intervals earlier in the sequence, the second sample in the current output frame is the input sample `v(2)` intervals earlier in the sequence, and so on.

The illustration below shows how this works for an input with a sample period of 1 and frame size of 4. The **Maximum delay** (`Dmax`) is `5`, and the **Initial conditions** parameter is set to `-1`. The delay input changes from `[1 3 0 5]` to `[2 0 0 2]` after the second input frame. The samples in each output frame are the values in memory indexed by the elements of `v`:

```
y(1) = U(v(1)+1)
y(2) = U(v(2)+1)
y(3) = U(v(3)+1)
y(4) = U(v(4)+1)
```

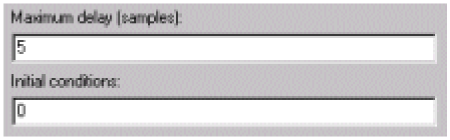


The **Initial conditions** parameter specifies the values in the block's memory at the start of the simulation. Both fixed and time-varying initial conditions can be specified.

Fixed Initial Conditions

The settings shown in this section specify fixed initial conditions. For a fixed initial condition, the block initializes each of *D* samples in memory to the value entered in the **Initial conditions** parameter. A fixed initial condition in frame-based mode can be one of the following:

- Scalar value with which to initialize every sample of every channel in memory. For a general *M*-by-*N* input with the parameter settings below, the block initializes five samples in memory with zeros.



- Array of size 1-by-N-by-D. In this case, you can specify different fixed initial conditions for each channel. See the Array bullet in "Time-Varying Initial Conditions" on page 2-1296 below for details.

Time-Varying Initial Conditions

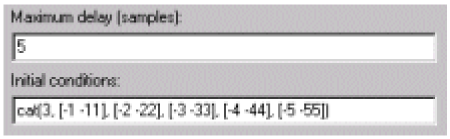
The following setting specifies a time-varying initial condition. For a time-varying initial condition, the block initializes each of D samples in memory to one of the values entered in the **Initial conditions** parameter. This allows you to specify a unique output value for each sample in memory. A time-varying initial condition in frame-based mode can be specified in the following ways:

- Vector of dimensions 1-by-D. In this case, all channels have the same set of time-varying initial conditions specified by the entries of the vector. For the ramp input `[100; 100]'` with a frame size of `4`, delay of `5`, and the following parameter settings, the block outputs the following sequence of frames at the start of the simulation:

$$\begin{bmatrix} -1 & -1 \\ -2 & -2 \\ -3 & -3 \\ -4 & -4 \end{bmatrix}, \begin{bmatrix} -5 & -5 \\ 1 & 1 \\ 2 & 2 \\ 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 \\ 5 & 5 \\ 6 & 6 \\ 7 & 7 \end{bmatrix}, \ldots$$

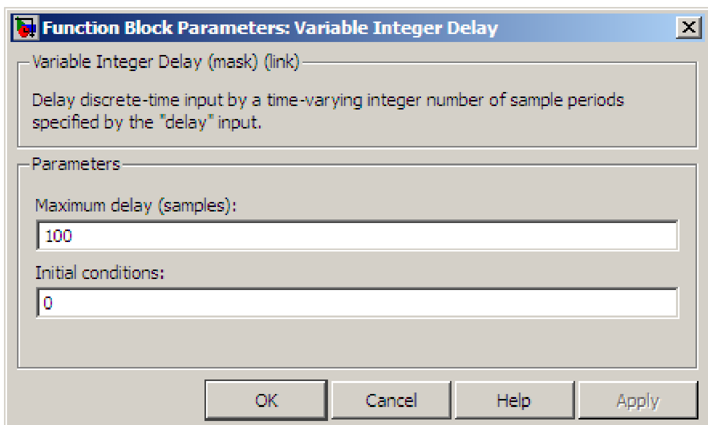
- Array of size 1-by-N-by-D. In this case, you can specify different time-varying initial conditions for each channel. For the ramp input `[100; 100]'` with a frame size of `4`, delay of `5`, and the following parameter settings, the block outputs the following sequence of frames at the start of the simulation:

$$\begin{bmatrix} -1 & -11 \\ -2 & -22 \\ -3 & -33 \\ -4 & -44 \end{bmatrix}, \begin{bmatrix} -5 & -55 \\ 1 & 1 \\ 2 & 2 \\ 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 \\ 5 & 5 \\ 6 & 6 \\ 7 & 7 \end{bmatrix}, \ldots$$



By specifying a 1-by-*N*-by-*D* initial condition array such that each 1-by-*N* vector entry is identical, you can implement different fixed initial conditions for each channel.

Dialog Box



Maximum delay

The maximum delay that the block can produce for any sample. Delay input values exceeding this maximum are clipped at the maximum.

Initial conditions
The values with which the block's memory is initialized.

Supported Data Types

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

See Also

Delay	Signal Processing Blockset
Variable Fractional Delay	Signal Processing Blockset

Purpose

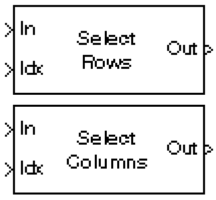
Select subset of rows or columns from input

Library

Signal Management / Indexing

`dspindex`

Description



The Variable Selector block extracts a subset of rows or columns from the M-by-N input matrix `u` at each input port. You specify the number of input and output ports in the **Number of input signals** parameter.

When the **Select** parameter is set to `Rows`, the Variable Selector block extracts rows from each input matrix, while if the **Select** parameter is set to `Columns`, the block extracts columns.

When the **Selector mode** parameter is set to `Variable`, the length-L vector input to the Idx port selects L rows or columns of each input to pass through to the output. The elements of the indexing vector can be updated at each sample time, but the vector length must remain the same throughout the simulation.

When the **Selector mode** parameter is set to `Fixed`, the Idx port is disabled, and the length-L vector specified in the **Elements** parameter selects L rows or columns of each input to pass through to the output. The **Elements** parameter is tunable, so you can change the values of the indexing vector elements at any time during the simulation; however, the vector length must remain the same.

For both variable and fixed indexing modes, the row selection operation is equivalent to

```
y = u(idx,:)      % Equivalent MATLAB code
```

and the column selection operation is equivalent to

```
y = u(:,idx)      % Equivalent MATLAB code
```

where `idx` is the length-L indexing vector. The row selection output size is L-by-N and the column selection output size is M-by-L. Input rows or columns can appear any number of times in the output, or not at all.

When the input is a 1-*D* vector, the **Select** parameter is ignored; the output is a 1-*D* vector of length *L* containing those elements specified by the length-*L* indexing vector.

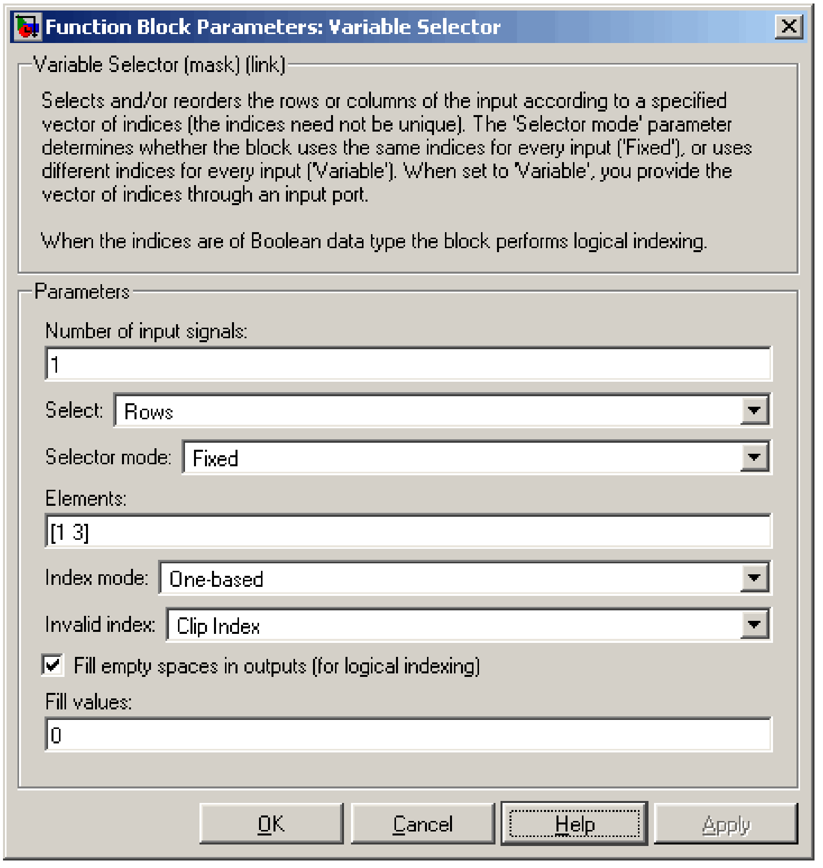
When an element of the indexing vector references a nonexistent row or column of the input, the block reacts with the behavior specified by the **Invalid index** parameter. The following options are available:

- `Clip index` — Clip the index to the nearest valid value, and do not issue an alert. Example: For a 64-by-*N* input, an index of 72 is clipped to 64; an index of -2 is clipped to 1.
- `Clip and warn` — Display a warning message in the MATLAB® Command Window, and clip as above.
- `Generate error` — Display an error dialog box and terminate the simulation.

When the indexing vector elements are of Boolean data type, the block performs logical indexing. Select **Fill empty spaces in outputs (for logical indexing)** to access the **Fill values** parameter. These values are appended to the output to make it as long as the input elements.

Note The Variable Selector block always copies the selected input rows to a contiguous block of memory (unlike the Simulink® Selector block).

Dialog Box



Number of input signals

Specify the number of input signals. An input port is created on the block for each input signal.

Select

The dimension of the input to select, `Rows` or `Columns`.

Selector mode

The type of indexing operation to perform, `Variable` or `Fixed`. Variable indexing uses the input at the Idx port to select rows or columns from the input at the `In` port. Fixed indexing uses the **Elements** parameter value to select rows from the input at the In port, and disables the Idx port.

Elements

A vector containing the indices of the input rows or columns that will appear in the output matrix. This parameter is only visible when you select `Fixed` for the **Selector mode** parameter.

Index mode

When set to `One-based`, an index value of `1` refers to the first row or column of the input. When set to `Zero-based`, an index value of `0` refers to the first row or column of the input.

Invalid index

Response to an invalid index value. Tunable.

Fill empty spaces in outputs (for logical indexing)

When the indexing vector elements are of Boolean data type, the block performs logical indexing. This can cause empty spaces in the output. Select this parameter to designate values to be appended to the output in the **Fill values** parameter.

Fill values

Specify the fill values when the block performs logical indexing. This parameter is only visible when the **Fill empty spaces in outputs (for logical indexing)** parameter is selected.

Supported Data Types

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

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

See Also

Multiport Selector	Signal Processing Blockset
Permute Matrix	Signal Processing Blockset
Selector	Simulink
Submatrix	Signal Processing Blockset

Variance

Purpose

Compute variance of input or sequence of inputs

Library

Statistics

`dspstat3`

Description

The Variance block computes the unbiased variance of each row or column of the input, along vectors of a specified dimension of the input, or of the entire input. The Variance block can also track the variance of a sequence of inputs over a period of time. The **Running variance** parameter selects between basic operation and running operation.

Basic Operation

When you do not select the **Running variance** check box, the block computes the variance of each row or column of the input, along vectors of a specified dimension of the input, or of the entire input at each individual sample time, and outputs the array y. Each element in y is the variance of the corresponding column, row, vector, or entire input. The output y depends on the setting of the **Find the variance value over** parameter. For example, consider a 3-dimensional input signal of size M-by-N-by-P:

- `Entire input` — The output at each sample time is a scalar that contains the variance of the entire input. In this mode, the output is always sample based.

  ```
  y = var(u(:))     % Equivalent MATLAB code
  ```

- `Each row` — The output at each sample time consists of an M-by-1-by-P array, where each element contains the variance of each vector over the second dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is an M-by-1 column vector. In this mode, the frame status of the output is the same as that of the input.

  ```
  y = var(u,0,2)     % Equivalent MATLAB code
  ```

- `Each column` — The output at each sample time consists of a 1-by-N-by-P array, where each element contains the variance of each vector over the first dimension of the input. For an input that is an M-by-N matrix, the output at each sample time is a 1-by-N row vector. In this mode, the frame status of the output is the same as that of the input.

  ```
  y = var(u,0,1)     % Equivalent MATLAB code
  ```

  For convenience, length-M 1-D vector inputs are treated as M-by-1 column vectors when the block is in this mode. Sample-based length-M row vector inputs are also treated as M-by-1 column vectors when the **Treat sample-based row input as a column** check box is selected.

- `Specified dimension` — The output at each sample time depends on **Dimension**. If **Dimension** is set to 1, the output is the same as that when you select `Each column`. If **Dimension** is set to `2`, the output is the same as when you select `Each row`. If **Dimension** is set to `3`, the output at each sample time is an M-by-N matrix containing the variance of each vector over the third dimension of the input. In this mode, the frame status of the output is the same as that of the input.

  ```
  y = var(u,0,Dimension)     % Equivalent MATLAB code
  ```

For purely real or purely imaginary inputs, the variance of an M-by-N matrix is the square of the standard deviation:

$$y = \sigma^2 = \frac{\sum_{i=1}^{M}\sum_{j=1}^{N}\left|u_{ij}\right|^2 - \frac{\left|\sum_{i=1}^{M}\sum_{j=1}^{N}u_{ij}\right|^2}{M*N}}{M*N-1}$$

For complex inputs, the variance is given by the following equation:

$$\sigma^2 = \sigma_{\mathrm{Re}}{}^2 + \sigma_{\mathrm{Im}}{}^2$$

Running Operation

When you select the **Running variance** check box, the block tracks the variance of successive inputs to the block. For sample-based M-by-N inputs, the output is a sample-based M-by-N matrix with each element y_{ij} containing the variance of element u_{ij} over all inputs since the last reset. For frame-based M-by-N inputs, the output is a frame-based M-by-N matrix with each element y_{ij} containing the variance of the jth column over all inputs since the last reset, up to and including element u_{ij} of the current input.

N-D signals cannot be frame based. When the block is set to `Running` mode, each element of the N-D signal is treated as a separate channel.

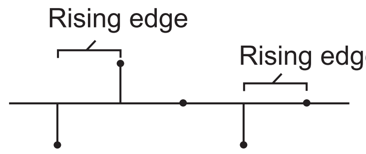
There are $\prod d_i$ channels, where d_i is the size of the ith dimension.

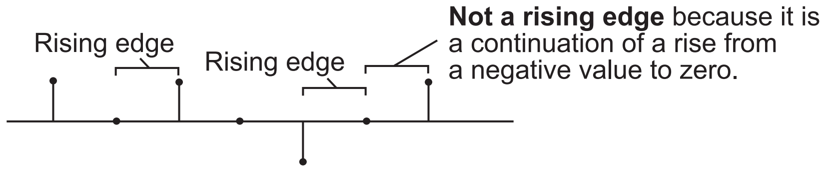
Resetting the Running Variance

The block resets the running variance whenever a reset event is detected at the optional Rst port. The reset sample time must be a positive integer multiple of the input sample time.

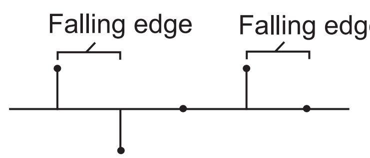
You specify the reset event in the **Reset port** parameter:

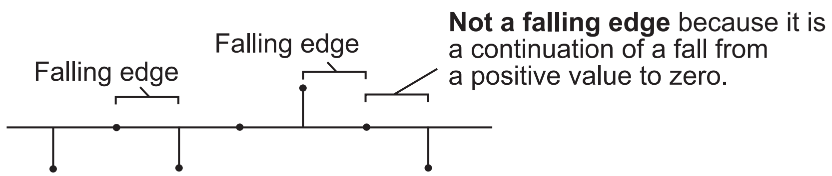
- `None` disables the Rst port.
- `Rising edge` — Triggers a reset operation when the Rst input does one of the following:
  - Rises from a negative value to a positive value or zero
  - Rises from zero to a positive value, where the rise is not a continuation of a rise from a negative value to zero (see the following figure)





- `Falling edge` — Triggers a reset operation when the Rst input does one of the following:
  - Falls from a positive value to a negative value or zero
  - Falls from zero to a negative value, where the fall is not a continuation of a fall from a positive value to zero (see the following figure)





- `Either edge` — Triggers a reset operation when the Rst input is a `Rising edge` or `Falling edge` (as described earlier)
- `Non-zero sample` — Triggers a reset operation at each sample time that the Rst input is not zero

---

Note When running simulations in the Simulink® MultiTasking mode, reset signals have a one-sample latency. Therefore, when the block detects a reset event, there is a one-sample delay at the reset port rate before the block applies the reset. For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and the topic on models with multiple sample rates in the Real-Time Workshop® documentation.

ROI Processing

To calculate the statistical value within a particular region of interest (ROI) of the input, select the **Enable ROI processing** check box. This option is only available when the **Find the variance value over** parameter is set to `Entire input` and the **Running variance** check box is not selected. ROI processing is only supported for 2-D inputs.

Note Full ROI processing is only available to users who have a Video and Image Processing Blockset™ license. If you only have a Signal Processing Blockset™ license, you can still use ROI processing, but are limited to the **ROI type** `Rectangles`.

Use the **ROI type** parameter to specify whether the ROI is a binary mask, label matrix, rectangle, or line. ROI processing is only supported for 2-D inputs.

- A binary mask is a binary image that enables you to specify which pixels to highlight, or select.
- In a label matrix, pixels equal to 0 represent the background, pixels equal to 1 represent the first object, pixels equal to 2 represent the second object, and so on. When the **ROI type** parameter is set to `Label matrix`, the Label and Label Numbers ports appear on the block. Use the Label Numbers port to specify the objects in the label matrix for which the block calculates statistics. The input to this

port must be a vector of scalar values that correspond to the labeled regions in the label matrix.

- For more information about the format of the input to the ROI port when the ROI is a rectangle or a line, see the Draw Shapes reference page.

Note For rectangular ROIs, use the **ROI portion to process** parameter to specify whether to calculate the statistical value for the entire ROI or just the ROI perimeter.

Use the **Output** parameter to specify the block output. The block can output separate statistical values for each ROI or the statistical value for all specified ROIs. This parameter is not available if, for the **ROI type** parameter, you select `Binary mask`.

If, for the **ROI type** parameter, you select `Rectangles` or `Lines`, the **Output flag indicating if ROI is within image bounds** check box appears in the dialog box. If you select this check box, the Flag port appears on the block. The following tables describe the Flag port output based on the block parameters.

Output = Individual Statistics for Each ROI

Flag Port Output	Description
0	ROI is completely outside the input image.
1	ROI is completely or partially inside the input image.

Output = Single Statistic for All ROIs

Flag Port Output	Description
0	All ROIs are completely outside the input image.
1	At least one ROI is completely or partially inside the input image.

If the ROI is partially outside the image, the block only computes the statistical values for the portion of the ROI that is within the image.

If, for the **ROI type** parameter, you select `Label matrix`, the **Output flag indicating if input label numbers are valid** check box appears in the dialog box. If you select this check box, the Flag port appears on the block. The following tables describe the Flag port output based on the block parameters.

Output = Individual Statistics for Each ROI

Flag Port Output	Description
0	Label number is not in the label matrix.
1	Label number is in the label matrix.

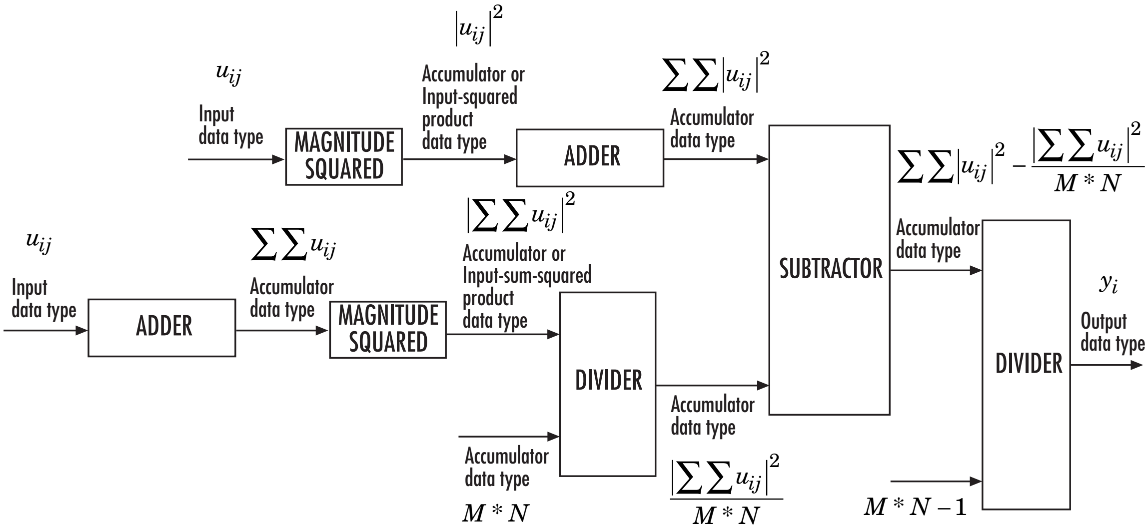
Output = Single Statistic for All ROIs

Flag Port Output	Description
0	None of the label numbers are in the label matrix.
1	At least one of the label numbers is in the label matrix.

Fixed-Point Data Types

The parameters on the Fixed-Point pane of the block dialog are only used for fixed-point inputs. For purely real or purely imaginary inputs, the variance of the input is the square of its standard deviation. For complex inputs, the output is the sum of the variance of the real and imaginary parts of the input.

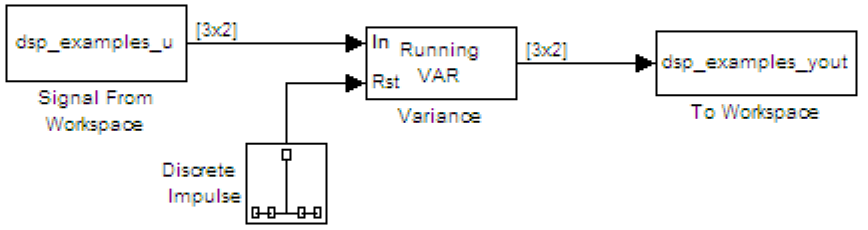
The following diagram shows the data types used within the Variance block for fixed-point signals.



The results of the magnitude-squared calculations in the figure are in the product output data type. You can set the accumulator, product output, and output data types in the block dialog as discussed in "Dialog Box" on page 2-1314.

Examples

The Variance block in the next model calculates the running variance of a frame-based 3-by-2 (two-channel) matrix input, u. The running variance is reset at t=2 by an impulse to the block's Rst port.



The Variance block has the following settings:

- **Running variance** = ☑
- **Reset port** = `Non-zero sample`

The Signal From Workspace block has the following settings:

- **Signal** = `dsp_examples_u`
- **Sample time** = `1/3`
- **Samples per frame** = `3`

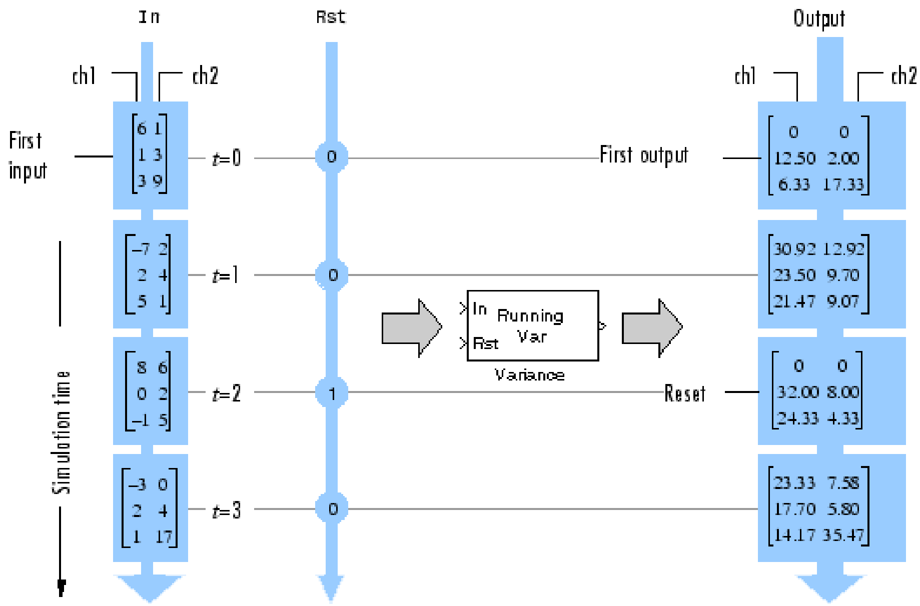
where

```
dsp_examples_u = [6 1 3 -7 2 5 8 0 -1 -3 2 1;1 3 9 2 4 1 6 2 5 0 4 17]'
```

The Discrete Impulse block has the following settings:

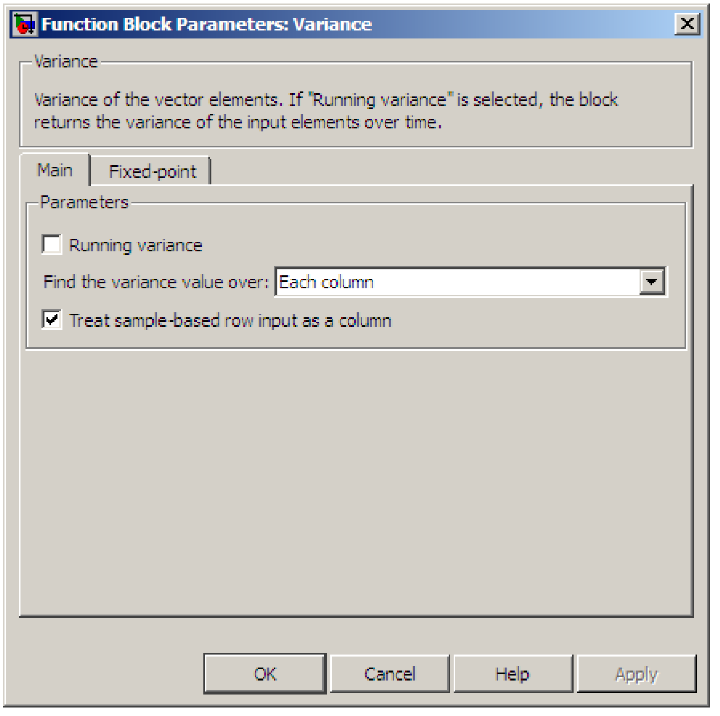
- **Delay (samples)** = `2`
- **Sample time** = `1`
- **Samples per frame** = `1`

The next figure shows the block's operation.


In
Rst
Output
ch1
ch2
ch1
ch2
First input
First output
Reset
Simulation time
t=0
t=1
t=2
t=3
In
Rst
Running Var
Variance

Dialog Box

The **Main** pane of the Variance block dialog appears as follows.



Running variance

Enables running operation when selected.

Reset port

Determines the reset event that causes the block to reset the running variance. The rate of the reset signal must be a positive integer multiple of the rate of the data signal input. This parameter is enabled only when you select the **Running variance** parameter. For more information, see "Resetting the Running Variance" on page 2-1306

Find the variance value over
Specify whether to find the variance along rows, columns, entire input, or the dimension specified in the **Dimension** parameter. For more information, see "Basic Operation" on page 2-1304.

Treat sample-based row input as a column
Select to treat sample-based length-*M* row vector inputs as *M*-by-1 column vectors. This parameter is only visible when the **Find the variance value over** parameter is set to `Each column`.

Dimension
Specify the dimension (one-based value) of the input signal, over which the variance is computed. The value of this parameter cannot exceed the number of dimensions in the input signal. This parameter is only visible when the **Find the variance value over** parameter is set to `Specified dimension`.

Enable ROI Processing
Select this check box to calculate the statistical value within a particular region of each image. This parameter is only available when the **Find the variance value over** parameter is set to `Entire input`, and the block is not in running mode.

Note Full ROI processing is only available to users who have a Video and Image Processing Blockset license. If you only have a Signal Processing Blockset license, you can still use ROI processing, but are limited to the **ROI type** `Rectangles`.

ROI type
Specify the type of ROI you want to use. Your choices are `Rectangles`, `Lines`, `Label matrix`, or `Binary mask`.

ROI portion to process
Specify whether you want to calculate the statistical value for the entire ROI or just the ROI perimeter. This parameter is only visible if, for the **ROI type** parameter, you specify `Rectangles`.

Output

Specify the block output. The block can output a vector of separate statistical values for each ROI or a scalar value that represents the statistical value for all the specified ROIs. This parameter is not available if, for the **ROI type** parameter, you select `Binary mask`.

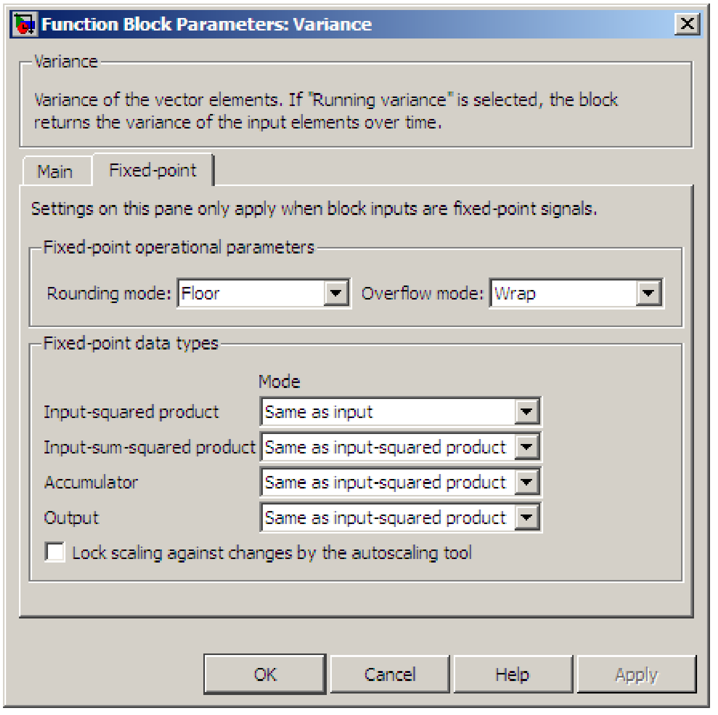
Output flag indicating if ROI is within image bounds

If you select this check box, the Flag port appears on the block. For a description of the Flag port output, see the tables in "ROI Processing" on page 2-1308. This parameter is visible if, for the **ROI type** parameter, you select `Rectangles` or `Lines`.

Output flag indicating if label numbers are valid

If you select this check box, the Flag port appears on the block. For a description of the Flag port output, see the tables in "ROI Processing" on page 2-1308. This parameter is visible if, for the **ROI type** parameter, you select `Label matrix`.

The **Fixed-point** pane of the Variance block dialog appears as follows.



Rounding mode
Select the rounding mode for fixed-point operations.

Overflow mode
Select the overflow mode for fixed-point operations.

Note See "Fixed-Point Data Types" on page 2-1311 for more information on how the product output, accumulator, and output data types are used in this block.

Input-squared product

Use this parameter to specify how to designate the input-squared product word and fraction lengths:

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the input-squared product, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the input-squared product. This block requires power-of-two slope and a bias of zero.

Input-sum-squared product

Use this parameter to specify how to designate the input-sum-squared product word and fraction lengths:

- When you select `Same as input-squared product`, these characteristics match those of the input-squared product.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the input-sum-squared product, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the input-sum-squared product. This block requires power-of-two slope and a bias of zero.

Accumulator

Use this parameter to specify the accumulator word and fraction lengths resulting from a complex-complex multiplication in the block:

- When you select `Same as input-squared product`, these characteristics match those of the input-squared product.
- When you select `Same as input`, these characteristics match those of the input to the block.

- When you select `Binary point scaling`, you can enter the word length and the fraction length of the accumulator, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the accumulator. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the output word length and fraction length:

- When you select `Same as accumulator`, these characteristics match those of the accumulator.
- When you select `Same as input-squared product`, these characteristics match those of the input-squared product.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Reset	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
ROI	Rectangles and lines: • Double-precision floating point • Single-precision floating point • Boolean • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers Binary Mask: • Boolean
Label	• 8-, 16-, and 32-bit unsigned integers
Label Numbers	• 8-, 16-, and 32-bit unsigned integers

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

See Also

Mean	Signal Processing Blockset
RMS	Signal Processing Blockset
Standard Deviation	Signal Processing Blockset
`var`	MATLAB

Vector Quantizer Decoder

Purpose

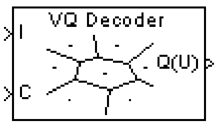
Find vector quantizer codeword that corresponds to given, zero-based index value

Library

Quantizers

`dspquant2`

Description



Vector Quantizer Decoder

The Vector Quantizer Decoder block associates each input index value with a codeword, a column vector of quantized output values defined in the **Codebook values** parameter. When you input multiple index values into this block, the block outputs a matrix of quantized output vectors. This matrix is created by horizontally concatenating the codeword vectors that correspond to each index value.

You can select how you want to enter the codebook values using the **Source of codebook** parameter. When you select `Specify via dialog`, you can type the codebook values into the block parameters dialog box. Select `Input port` and port C appears on the block. The block uses the input to port C as the **Codebook values** parameter.

The **Codebook values** parameter is a k-by-N matrix of values, where $k \geq 1$ and $N \geq 1$. Each column of this matrix is a codeword vector, and each codeword vector corresponds to an index value. The index values are zero based; therefore, the first codeword vector corresponds to an index value of 0, the second codeword vector corresponds to an index value of 1, and so on.

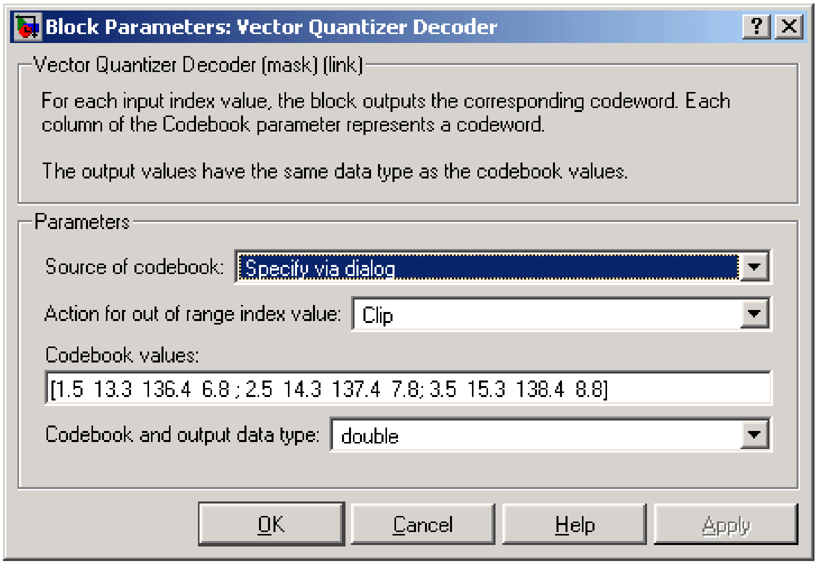
The input to this block is a vector of index values, where $0 \leq index < N$ and N is the number of columns of the codebook matrix. Use the **Action for out of range index value** parameter to determine how the block behaves when an input index value is out of this range. When you want any index values less than 0 to be set to 0 and any index values greater than or equal to N to be set to N-1, select `Clip`. When you want to be warned when any index values less than 0 are set to 0 and any index values greater than or equal to N are set to N-1, select `Clip and warn`. When you want the simulation to stop and display an error when the index values are out of range, select `Error`.

Data Type Support

The input to the block can be the index values and the codebook values. The data type of the index input to the block at port I can be `uint8`, `uint16`, `uint32`, `int8`, `int16`, or `int32`. The data type of the codebook values can be `double`, `single`, or Fixed-point.

The output of the block is the quantized output values. These quantized output values always have the same data type as the codebook values. When the codebook values are specified via an input port, the block assigns the same data type to the Q(U) output port. When the codebook values are specified via the dialog, use the **Codebook and output data type** parameter to specify the data type of the Q(U) output port. The data type of the codebook and quantized output can be `Same as input`, `double`, `single`, `Fixed-point`, `User-defined`, or `Inherit via back propagation`.

Dialog Box



Source of codebook

Choose `Specify via dialog` to type the codebook values into the block parameters dialog box. Select `Input port` to specify the codebook values using the block's input port, C.

Action for out of range index value

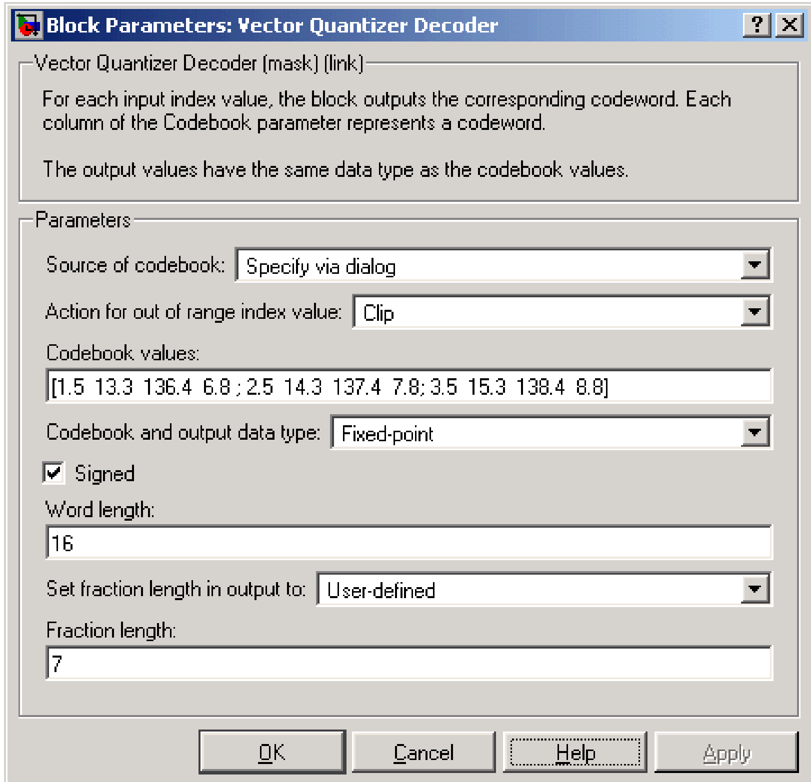
Choose the behavior of the block when an input index value is out of range, where $0 \le index < N$ and N is the length of the codebook vector. Select `Clip` when you want any index values less than 0 to be set to 0 and any index values greater than or equal to N to be set to N-1. Select `Clip and warn` when you want to be warned when any index values less than 0 are set to 0 and any index values greater than or equal to N are set to N-1. Select `Error` when you want the simulation to stop and display an error when the index values are out of range.

Codebook values

Enter a k-by-N matrix of quantized output values, where $1 \le k$ and $1 \le N$. Each column of your matrix corresponds to an index value. This parameter is visible if, from the **Source of codebook** list, you select `Specify via dialog`.

Codebook and output data type

Use this parameter to specify the data type of the codebook and quantized output values. The data type can be `Same as input`, `double`, `single`, `Fixed-point`, `User-defined`, or`Inherit via back propagation`. This parameter becomes visible when you select `Specify via dialog` for the **Source of codebook** parameter.



Signed

Select to output a signed fixed-point signal. Otherwise, the signal is unsigned. This parameter is only visible if, from the **Codebook and output data type** list, you select `Fixed-point`.

Word length

Specify the word length, in bits, of the fixed-point output data type. This parameter is only visible if, from the **Codebook and output data type** list, you select `Fixed-point`.

Set fraction length in output to

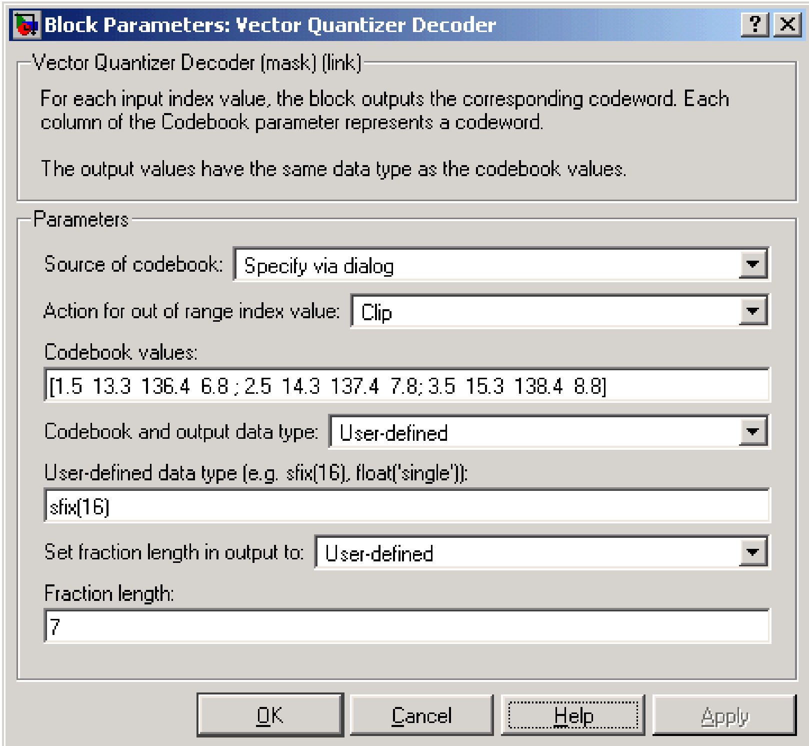
Specify the scaling of the fixed-point output by either of the following two methods:

- Choose `Best precision` to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose `User-defined` to specify the output scaling in the **Fraction length** parameter.

This parameter is only visible if, from the **Codebook and output data type** list, you select `Fixed-point` or when you select `User-defined` and the specified output data type is a fixed-point data type.

Fraction length

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



User-defined data type

Specify any built-in or fixed-point data type. You can specify fixed-point data types using the following Simulink Fixed Point functions: `sfix`, `ufix`, `sint`, `uint`, `sfrac`, and `ufrac`. This parameter is only visible when you select `User-defined` for the **Codebook and output data type** parameter.

References

Gersho, A. and R. Gray. *Vector Quantization and Signal Compression*. Boston: Kluwer Academic Publishers, 1992.

Supported Data Types

Port	Supported Data Types
I	• 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
C	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers
Q(U)	• Double-precision floating point • Single-precision floating point • Fixed point • 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers

See Also

Quantizer	Simulink
Scalar Quantizer Decoder	Signal Processing Blockset
Scalar Quantizer Design	Signal Processing Blockset
Uniform Encoder	Signal Processing Blockset
Uniform Decoder	Signal Processing Blockset
Vector Quantizer Encoder	Signal Processing Blockset

Purpose

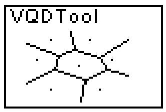
Design vector quantizer using Vector Quantizer Design Tool (VQDTool)

Library

Quantizers

`dspquant2`

Description



Vector Quantizer Design

Double-click on the Vector Quantizer Design block to start VQDTool, a GUI that allows you to design and implement a vector quantizer. You can also start VQDTool by typing `vqdtool` at the MATLAB® command prompt. Based on your specifications, VQDTool iteratively calculates the codebook values that minimize the mean squared error between the training set and the codebook until the stopping criteria for the design process is satisfied. The block uses the resulting codebook values to implement your vector quantizer.

For the **Training Set** parameter, enter a k-by-M matrix of values you want to use to train the quantizer codebook. The variable k, where $k \leq 1$, is the length of each training vector. It also represents the dimension of your quantizer. The variable M, where $M \leq 2$, is the number of training vectors. This data can be created using a MATLAB function, such as the default value `randn(10,1000)`, or it can be any variable defined in the MATLAB workspace.

You have two choices for the **Source of initial codebook** parameter. Select `Auto-generate` to have the block choose the values of the initial codebook. In this case, the block picks N random training vectors as the initial codebook, where N is the **Number of levels** parameter and $N \geq 2$. When you select `User defined`, enter the initial codebook values in the **Initial codebook** field. The initial codebook matrix must have the same number of rows as the training set. Each column of the codebook is a codeword, and your codebook must have at least two codewords.

For the given training set and initial codebook, the block performs an iterative process, using the Generalized Lloyd Algorithm (GLA), to design a final codebook. For each iteration of the GLA, the block first associates each training vector with its nearest codeword by calculating the distortion. You can specify one of the two possible methods for calculating distortion using the **Distortion measure** parameter.

When you select `Squared error` for the **Distortion measure** parameter, the block finds the nearest codeword by calculating the squared error (unweighted). Consider the codebook

$CB = [CW_1 \quad CW_2 \quad \ldots \quad CW_N]$. This codebook has N codewords; each codeword has k elements. The i-th codeword is defined as

$CW_i = [a_{1i} \quad a_{2i} \quad \ldots \quad a_{ki}]$. The training set has M columns and is

defined as $U = [U_1 \quad U_2 \quad \ldots \quad U_M]$, where the p-th training vector

is $U_p = \left[u_{1p} \quad u_{2p} \quad \ldots \quad u_{kp}\right]'$. The squared error (unweighted) is calculated using the equation

$$D = \sum_{j=1}^{k} \left(a_{ji} - u_{jp}\right)^2$$

When you select `Weighted squared error` for the **Distortion measure** parameter, enter a vector or matrix for the **Weighting factor** parameter. When the weighting factor is a vector, its length must be equal to the number of rows in the training set. This weighting factor is used for each training vector. When the weighting factor is a matrix, it must be the same size as the training set matrix. The block finds the nearest codeword by calculating the weighted squared error. If

the weighting factor for the p-th column of the training vector, U_p,

is defined as $Wp = \left[w_{1p} \quad w_{2p} \quad \ldots \quad w_{kp}\right]'$, then the weighted squared error is defined by the equation

$$D = \sum_{j=1}^{k} w_{jp} \left(a_{ji} - u_{jp}\right)^2$$

Once the block has associated all the training vectors with their nearest codeword vectors, the block calculates the mean squared error for the codebook and checks to see if the stopping criteria for the process has been satisfied.

The two possible options for the **Stopping criteria** parameter are `Relative threshold` and `Maximum iteration`. When you want the design process to stop when the fractional drop in the squared error is below a certain value, select `Relative threshold`. Then, type the maximum acceptable fractional drop in the **Relative threshold** field. The fraction drop in the squared error is defined as

$$\frac{\text{error at previous iteration} - \text{error at current iteration}}{\text{error at previous iteration}}$$

When you want the design process to stop after a certain number of iterations, choose `Maximum iteration`. Then, enter the maximum number of iterations you want the block to perform in the **Maximum iteration** field. For **Stopping criteria**, you can also choose `Whichever comes first` and enter **Relative threshold** and **Maximum iteration** values. The block stops iterating as soon as one of these conditions is satisfied.

When a training vector has the same distortion for two different codeword vectors, the algorithm uses the **Tie-breaking rule** parameter to determine which codeword vector the training vector is associated with. When you want the training vector to be associated with the lower indexed codeword, select `Lower indexed codeword`. To associate the training vector with the higher indexed codeword, select `Higher indexed codeword`.

With each iteration, the block updates the codeword values in order to minimize the distortion. The **Codebook update method** parameter defines the way the block calculates these new codebook values.

Note If, for the **Distortion measure** parameter, you choose `Squared error`, the **Codebook update method** parameter is set to `Mean`.

If, for the **Distortion measure** parameter, you choose `Weighted squared error` and you choose `Mean` for the **Codebook update method** parameter, the new codeword vector is found as follows. Suppose there are three training vectors associated with one codeword vector. The training vectors are

$$TS_1 = \begin{bmatrix} 1 \\ 2 \end{bmatrix},\ TS_3 = \begin{bmatrix} 10 \\ 12 \end{bmatrix},\ \text{and}\ TS_7 = \begin{bmatrix} 11 \\ 12 \end{bmatrix}.$$

The new codeword vector is calculated as

$$CW_{new} = \begin{bmatrix} \dfrac{1+10+11}{3} \\ \dfrac{2+12+12}{3} \end{bmatrix}$$

where the denominator is the number of training vectors associated with this codeword. If, for the **Codebook update method** parameter, you choose `Centroid` and you specify the weighting factors $W_1 = \begin{bmatrix} 0.1 \\ 0.2 \end{bmatrix}$, $W_3 = \begin{bmatrix} 1 \\ 0.6 \end{bmatrix}$, and $W_7 = \begin{bmatrix} 0.3 \\ 0.4 \end{bmatrix}$, the new codeword vector is calculated as

$$CW_{new} = \begin{bmatrix} \dfrac{(0.1)(1)+(1)(10)+(0.3)(11)}{0.1+1+0.3} \\ \dfrac{(0.2)(2)+(0.6)(12)+(0.4)(12)}{0.2+0.6+0.4} \end{bmatrix}$$

Click **Design and Plot** to design the quantizer with the parameter values specified on the left side of the GUI. The performance curve and the entropy of the quantizer are updated and displayed in the figures on the right side of the GUI.

Note You must click **Design and Plot** to apply any changes you make to the parameter values in the VQDTool dialog box.

The following is an example of how the block calculates the entropy of the quantizer at each iteration. Suppose you have a codebook with four codewords and a training set with 200 training vectors. Also suppose that, at the i-th iteration, 40 training vectors are associated with the first codeword, 60 training vectors are associated with the second codeword, 20 training vectors are associated with the third codeword, and 80 training vectors are associated with the fourth codeword. The probability that a training vector is associated with the first codeword is $\frac{40}{200}$. The probabilities that training vectors are associated with the second, third, and fourth codewords are $\frac{60}{200}$, $\frac{20}{200}$, and $\frac{80}{200}$, respectively. The GUI uses these probabilities to calculate the entropy according to the equation

$$H = \sum_{i=1}^{N} -p_i \log_2 p_i$$

where N is the number of codewords. Based on these probabilities, the GUI calculates the entropy of the quantizer at the i-th iteration as

$$H = -\left(\frac{40}{200}\log_2\frac{40}{200} + \frac{60}{200}\log_2\frac{60}{200} + \frac{20}{200}\log_2\frac{20}{200} + \frac{80}{200}\log_2\frac{80}{200}\right)$$
$$H = 1.8464$$

VQDTool can export parameter values that correspond to the figures displayed in the GUI. Click the **Export Outputs** button, or press **Ctrl+E**, to export the **Final Codebook**, **Mean Square Error**, and **Entropy** values to the workspace, a text file, or a MAT-file.

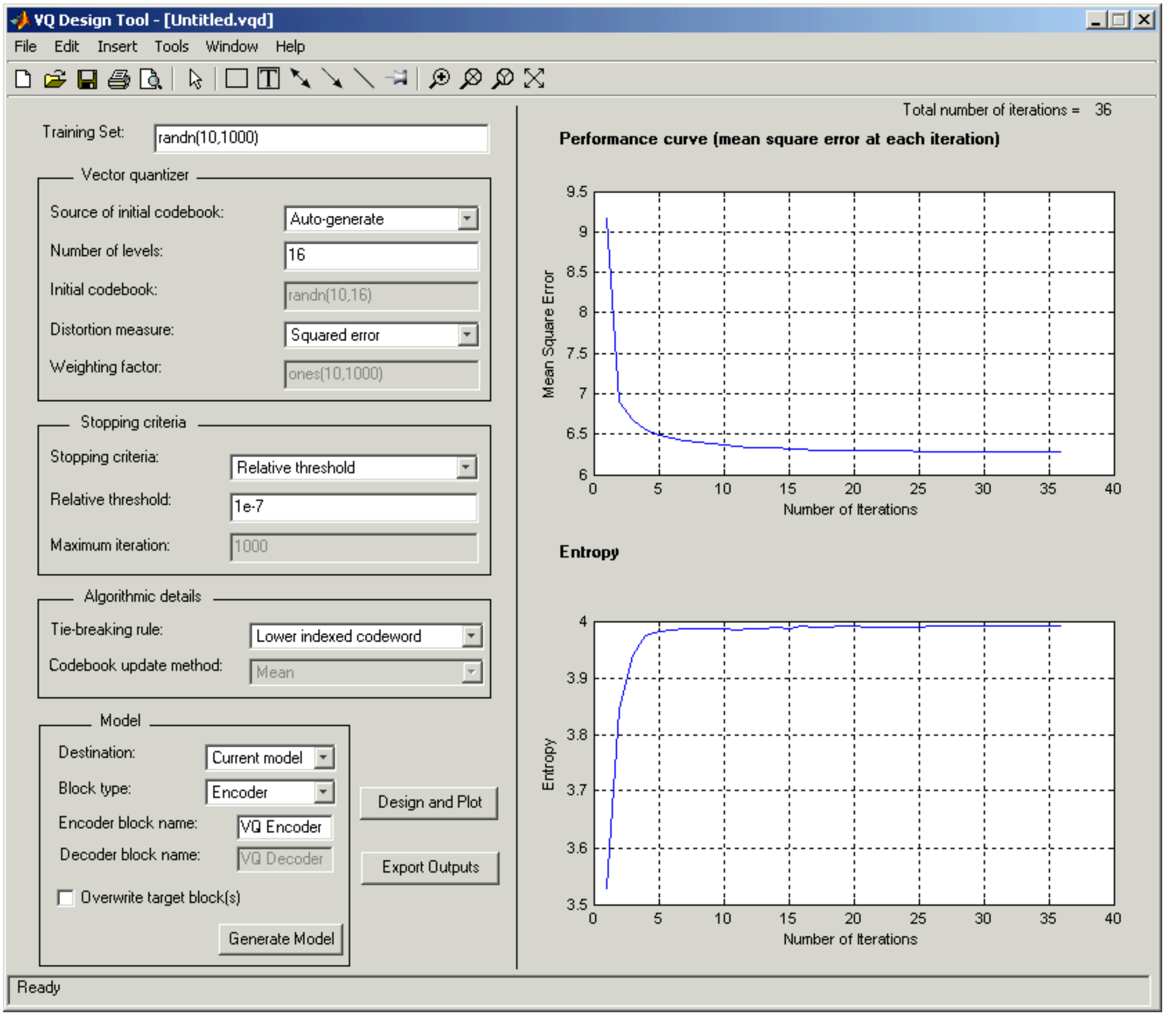
In the **Model** section of the GUI, specify the destination of the block that will contain the parameters of your quantizer. For **Destination**, select `Current model` to create a block with your parameters in the model you most recently selected. Type `gcs` in the MATLAB Command Window to display the name of your current model. Select `New model` to create a block in a new model file.

From the **Block type** list, select `Encoder` to design a Vector Quantizer Encoder block. Select `Decoder` to design a Vector Quantizer Decoder block. Select `Both` to design a Vector Quantizer Encoder block and a Vector Quantizer Decoder block.

In the **Encoder block name** field, enter a name for the Vector Quantizer Encoder block. In the **Decoder block name** field, enter a name for the Vector Quantizer Decoder block. When you have a Vector Quantizer Encoder and/or Decoder block in your destination model with the same name, select the **Overwrite target block** check box to replace the block's parameters with the current parameters. When you do not select this check box, a new Vector Quantizer Encoder and/or Decoder block is created in your destination model.

Click **Generate Model**. VQDTool uses the parameters that correspond to the current plots to set the parameters of the Vector Quantizer Encoder and/or Decoder blocks.

Dialog Box


VQ Design Tool - [Untitled.vqd]
File Edit Insert Tools Window Help
Training Set: randn(10,1000)
Vector quantizer
Source of initial codebook: Auto-generate
Number of levels: 16
Initial codebook: randn(10,16)
Distortion measure: Squared error
Weighting factor: ones(10,1000)
Stopping criteria
Stopping criteria: Relative threshold
Relative threshold: 1e-7
Maximum iteration: 1000
Algorithmic details
Tie-breaking rule: Lower indexed codeword
Codebook update method: Mean
Model
Destination: Current model
Block type: Encoder
Encoder block name: VQ Encoder
Decoder block name: VQ Decoder
Overwrite target block(s)
Generate Model
Design and Plot
Export Outputs
Total number of iterations = 36
Performance curve (mean square error at each iteration)
Mean Square Error
Number of Iterations
Entropy
Ready

Training Set
Enter the samples of the signal you would like to quantize. This data set can be a MATLAB function or a variable defined in the MATLAB workspace. The typical length of this data vector is `1e5`.

Source of initial codebook
Select `Auto-generate` to have the block choose the initial codebook values. Choose `User defined` to enter your own initial codebook values.

Number of levels
Enter the number of codeword vectors, N, in your codebook matrix, where $N \geq 2$.

Initial codebook
Enter your initial codebook values. From the **Source of initial codebook** list, select `User defined` in order to activate this parameter. The codebook must have the same number of rows as the training set. You must provide at least two codeword vectors.

Distortion measure
When you select `Squared error`, the block finds the nearest codeword by calculating the squared error (unweighted). When you select `Weighted squared error`, the block finds the nearest codeword by calculating the weighted squared error.

Weighting factor
Enter a vector or matrix. The block uses these values to compute the weighted squared error. When the weighting factor is a vector, its length must be equal to the number of rows in the training set. This weighting factor is used for each training vector. When the weighting factor is a matrix, it must be the same size as the training set matrix. The individual weighting factors cannot be negative. The weighting factor vector or matrix cannot contain all zeros.

Stopping criteria
Choose `Relative threshold` to enter the maximum acceptable fractional drop in the squared quantization error. Choose `Maximum iteration` to specify the number of iterations at which to stop.

Choose `Whichever comes first` and the block stops the iteration process as soon as the relative threshold or maximum iteration value is attained.

Relative threshold
This parameter is available when you choose `Relative threshold` or `Whichever comes first` for the **Stopping criteria** parameter. Enter the value that is the maximum acceptable fractional drop in the squared quantization error.

Maximum iteration
This parameter is available when you choose `Maximum iteration` or `Whichever comes first` for the **Stopping criteria** parameter. Enter the maximum number of iterations you want the block to perform.

Tie-breaking rules
When a training vector has the same distortion for two different codeword vectors, select `Lower indexed codeword` to associate the training vector with the lower indexed codeword. Select `Higher indexed codeword` to associate the training vector with the lower indexed codeword.

Codebook update method
When you choose `Mean`, the new codeword vector is calculated by taking the average of all the training vector values that were associated with the original codeword vector. When you choose `Centroid`, the block calculates the new codeword vector by taking the weighted average of all the training vector values that were associated with the original codeword vector Note that if, for the **Distortion measure** parameter, you choose `Squared error`, the **Codebook update method** parameter is set to `Mean`.

Destination
Choose `Current model` to create a Vector Quantizer block in the model you most recently selected. Type `gcs` in the MATLAB Command Window to display the name of your current model. Choose `New model` to create a block in a new model file.

Block type

Select `Encoder` to design a Vector Quantizer Encoder block. Select `Decoder` to design a Vector Quantizer Decoder block. Select `Both` to design a Vector Quantizer Encoder block and a Vector Quantizer Decoder block.

Encoder block name

Enter a name for the Vector Quantizer Encoder block.

Decoder block name

Enter a name for the Vector Quantizer Decoder block.

Overwrite target block

When you do not select this check box and a Vector Quantizer Encoder and/or Decoder block with the same block name exists in the destination model, a new Vector Quantizer Encoder and/or Decoder block is created in the destination model. When you select this check box and a Vector Quantizer Encoder and/or Decoder block with the same block name exists in the destination model, the parameters of these blocks are overwritten by new parameters.

Generate Model

Click this button and VQDTool uses the parameters that correspond to the current plots to set the parameters of the Vector Quantizer Encoder and/or Decoder blocks.

Design and Plot

Click this button to design a quantizer using the parameters on the left side of the GUI and to update the performance curve and entropy plots on the right side of the GUI.

You must click **Design and Plot** to apply any changes you make to the parameter values in the VQDTool GUI.

Export Outputs

Click this button, or press **Ctrl+E**, to export the **Final Codebook**, **Mean Squared Error**, and **Entropy** values to the workspace, a text file, or a MAT-file.

References

Gersho, A. and R. Gray. *Vector Quantization and Signal Compression*. Boston: Kluwer Academic Publishers, 1992.

Supported Data Types

- Double-precision floating point

See Also

Quantizer	Simulink
Scalar Quantizer Decoder	Signal Processing Blockset
Scalar Quantizer Design	Signal Processing Blockset
Uniform Encoder	Signal Processing Blockset
Uniform Decoder	Signal Processing Blockset
Vector Quantizer Decoder	Signal Processing Blockset
Vector Quantizer Encoder	Signal Processing Blockset

Vector Quantizer Encoder

Purpose

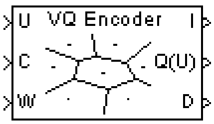
For given input, find index of nearest codeword based on Euclidean or weighted Euclidean distance measure

Library

Quantizers

`dspquant2`

Description



Vector Quantizer Encoder

The Vector Quantizer Encoder block compares each input column vector to the codeword vectors in the codebook matrix. Each column of this codebook matrix is a codeword. The block finds the codeword vector nearest to the input column vector and returns its zero-based index. This block supports real floating-point and fixed-point signals on all input ports.

The block finds the nearest codeword by calculating the distortion. The block uses two methods for calculating distortion: Euclidean squared error (unweighted) and weighted Euclidean squared error. Consider the codebook, $CB = [CW_1 \quad CW_2 \quad \ldots \quad CW_N]$. This codebook has N codewords; each codeword has k elements. The i-th codeword is defined as a column vector, $CW_i = [a_{1i} \quad a_{2i} \quad \ldots \quad a_{ki}]$. The multichannel input has M columns and is defined as $U = [U_1 \quad U_2 \quad \ldots \quad U_M]$, where the p-th input column vector is $U_p = \left[u_{1p} \quad u_{2p} \quad \ldots \quad u_{kp}\right]'$. The squared error (unweighted) is calculated using the equation

$$D = \sum_{j=1}^{k} \left(a_{ji} - u_{jp}\right)^2$$

The weighted squared error is calculated using the equation

$$D = \sum_{j=1}^{k} w_j \left(a_{ji} - u_{jp}\right)^2$$

where the weighting factor is defined as $W = [w_1 \quad w_2 \quad \ldots \quad w_k]$. The index of the codeword that is associated with the minimum distortion is assigned to the input column vector.

You can select how you want to enter the codebook values using the **Source of codebook** parameter. When you select `Specify via dialog`, you can type the codebook values into the block parameters dialog box. Select `Input port` and port C appears on the block. The block uses the input to port C as the **Codebook** parameter.

The **Codebook** parameter is an k-by-N matrix of values, where $k \geq 1$ and $N \geq 1$. Each input column vector is compared to this codebook. Each column of the codebook matrix is a codeword, and each codeword has an index value. The first codeword vector corresponds to an index value of 0, the second codeword vector corresponds to an index value of 1, and so on. The codeword vectors must have the same number of rows as the input, U.

For the **Distortion measure** parameter, select `Squared error` when you want the block to calculate the distortion by evaluating the Euclidean distance between the input column vector and each codeword in the codebook. Select `Weighted squared error` when you want to use a weighting factor to emphasize or deemphasize certain input values.

For the **Source of weighting factor** parameter, select `Specify via dialog` to enter a weighting factor vector in the dialog box. Choose `Input port` to specify the weighting factor using port W.

Use the **Weighting factor** parameter to emphasize or deemphasize certain input values when calculating the distortion measure. For example, consider the p-th input column vector, U_p, as previously defined. When you want to neglect the effect of the first element of this vector, enter `[0 1 1 ... 1]` as the **Weighting factor** parameter. This weighting factor is used to calculate the weighted squared error using the equation

$$D = \sum_{j=1}^{k} w_j \left(a_{ji} - u_{jp}\right)^2$$

Because of the weighting factor used in this example, the weighted squared error is not affected by the first element of the input matrix. Therefore, the first element of the input column vector no longer impacts the choice of index value output by the Vector Quantizer Encoder block.

Use the **Index output data type** parameter to specify the data type of the index values output at port I. The data type of the index values can be `int8`, `uint8`, `int16`, `uint16`, `int32`, or `uint32`.

When an input vector is equidistant from two codewords, the block uses the **Tie-breaking rule** parameter to determine which index value the block chooses. When you want the input vector to be represented by the lower index valued codeword, select `Choose the lower index`. To represent the input column vector by the higher index valued codeword, select `Choose the higher index`.

Select the **Output codeword** check box to output at port Q(U) the codeword vectors that correspond to each index value. When the input is a matrix, the corresponding codeword vectors are horizontally concatenated into a matrix.

Select the **Output quantization error** check box to output at port D the quantization error that results when the block represents the input column vector by its nearest codeword. When the input is a matrix, the quantization error values are horizontally concatenated.

The Vector Quantizer Encoder block accepts real floating-point and fixed-point inputs. For more information on the data types accepted by each port, see “Data Type Support” on page 2-1342 or “Supported Data Types” on page 2-1349.

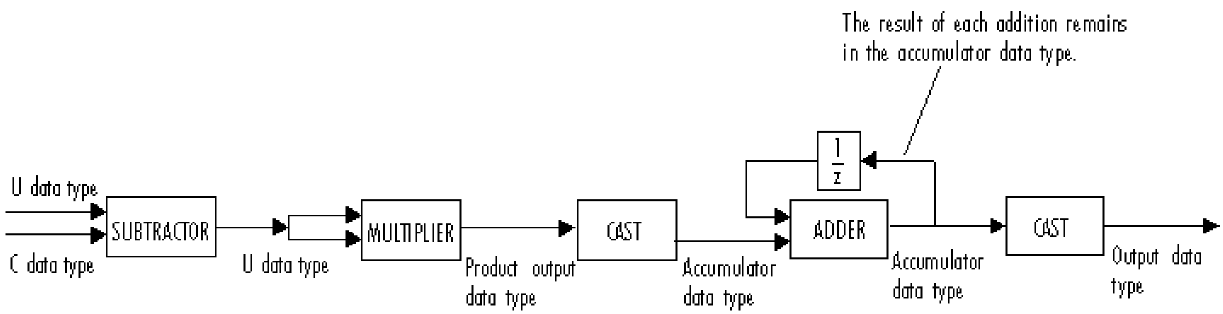
Data Type Support

The input data values, codebook values, and weighting factor values are input to the block at ports U, C, and W, respectively. The data type of the input data values, codebook values, and weighting factor values can be `double`, `single`, or Fixed-point. The input data, codebook values, and weighting factor must be the same data type.

The outputs of the block are the index values, output codewords, and quantization error. Use the **Index output data type** parameter to specify the data type of the index output from the block at port I. The data type of the index can be `int8`, `uint8`, `int16`, `uint16`, `int32`, or `uint32`. The data type of the output codewords and the quantization error can be `double`, `single`, or Fixed-point. The block assigns the data type of the output codewords and the quantization error based on the data type of the input data.

Fixed-Point Data Types

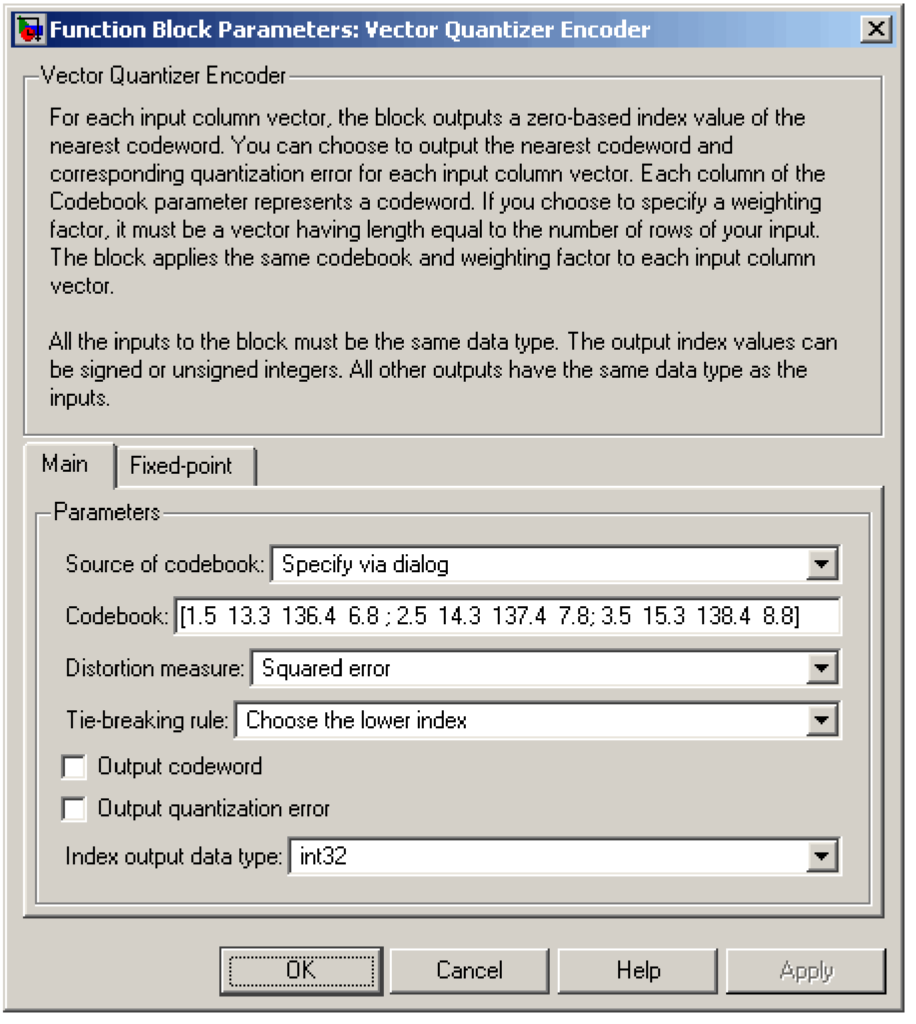
The following diagram shows the data types used within the Vector Quantizer Encoder block for fixed-point signals.



You can set the product output, accumulator, and index output data types in the block dialog as discussed below.

Dialog Box

The **Main** pane of the Vector Quantizer Encoder block dialog appears as follows.



Source of codebook

Choose `Specify via dialog` to type the codebook values into the block parameters dialog box. Select `Input port` to specify the codebook values using the block's input port, C.

Codebook

Enter a k-by-N matrix of values, where $1 \leq k$ and $1 \leq N$, to which your input column vector or matrix is compared. This parameter is visible if, from the **Source of codebook** list, you select `Specify via dialog`.

Distortion measure

Select `Squared error` when you want the block to calculate the distortion by evaluating the Euclidean distance between the input column vector and each codeword in the codebook. Select `Weighted squared error` when you want the block to calculate the distortion by evaluating a weighted Euclidean distance using a weighting factor to emphasize or deemphasize certain input values.

Source of weighting factor

Select `Specify via dialog` to enter a value for the weighting factor in the dialog box. Choose `Input port` and specify the weighting factor using port W on the block. This parameter is visible if, for the **Distortion measure** parameter, you select `Weighted squared error`.

Weighting factor

Enter a vector of values. This vector must have length equal to the number of rows of the input, U. This parameter is visible if, for the **Source of weighting factor** parameter, you select `Specify via dialog`.

Tie-breaking rule

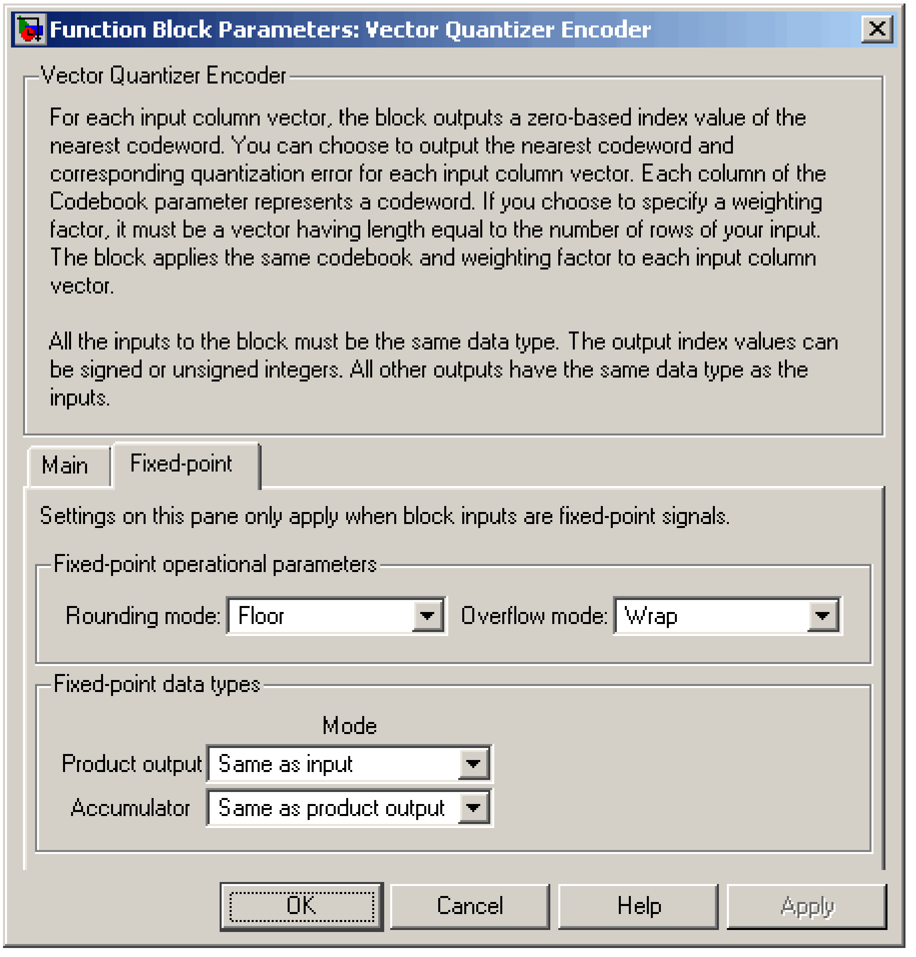
Set this parameter to determine the behavior of the block when an input column vector is equidistant from two codewords. When you want the input column vector to be represented by the lower index valued codeword, select `Choose the lower index`. To represent the input column vector by the higher index valued codeword, select `Choose the higher index`.

Output codeword
Select this check box to output the codeword vectors nearest to the input column vectors.

Output quantization error
Select this check box to output the quantization error value that results when the block represents the input column vector by the nearest codeword.

Index output data type
Select `int8`, `uint8`, `int16`, `uint16`, `int32`, or `uint32` as the data type of the index output at port I.

The **Fixed-point** pane of the Vector Quantizer Encoder block dialog appears as follows.



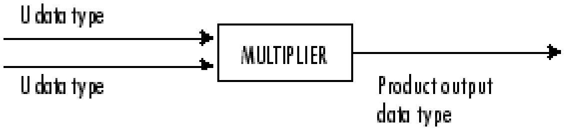
Rounding mode

Select the rounding mode for fixed-point operations.

Overflow mode

Select the overflow mode to be used when block inputs are fixed point.

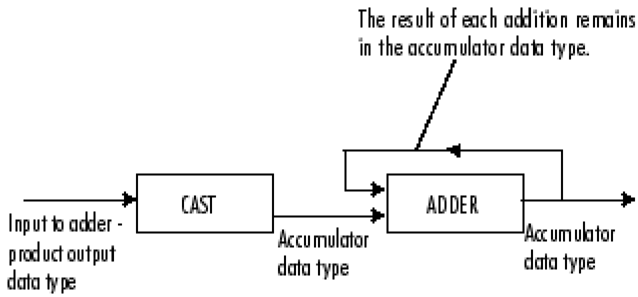
Product output



As depicted above, the output of the multiplier is placed into the product output data type and scaling. Use this parameter to specify how you would like to designate this product output word and fraction lengths.

- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and zero bias.

Accumulator



As depicted above, inputs to the accumulator are cast to the accumulator data type. The output of the adder remains in the accumulator data type as each element of the input is added to it. Use this parameter to specify how you would like to designate the accumulator word and fraction lengths.

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

References

Gersho, A. and R. Gray. *Vector Quantization and Signal Compression*. Boston: Kluwer Academic Publishers, 1992.

Supported Data Types

Port	Supported Data Types
U	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
C	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
W	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers

Port	Supported Data Types
I	• 8-, 16-, and 32-bit signed integers • 8-, 16-, and 32-bit unsigned integers
Q(U)	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers
D	• Double-precision floating point • Single-precision floating point • Fixed point (signed only) • 8-, 16-, and 32-bit signed integers

See Also

Quantizer	Simulink
Scalar Quantizer Decoder	Signal Processing Blockset
Scalar Quantizer Design	Signal Processing Blockset
Uniform Encoder	Signal Processing Blockset
Uniform Decoder	Signal Processing Blockset
Vector Quantizer Decoder	Signal Processing Blockset

Purpose

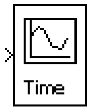
Display vector or matrix of time-domain, frequency-domain, or user-defined data

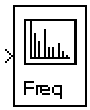
Library

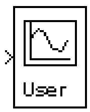
Signal Processing Sinks

`dspsnks4`

Description







The Vector Scope block is a comprehensive display tool similar to a digital oscilloscope. The block can display time-domain, frequency-domain, or user-defined signals. You can use the Vector Scope block to plot consecutive time samples from a frame-based vector, or to plot vectors containing data such as filter coefficients or spectral magnitudes. To compute and plot the periodogram of a signal with a single block, use the Spectrum Scope block.

The input to the Vector Scope block can be any real-valued M-by-N matrix, column or row vector, or 1-D (unoriented) vector, where 1-D vectors are treated as column vectors. Regardless of the input frame status, the block treats each column of an M-by-N input as an independent channel of data with M consecutive samples.

The block plots each sample of each input channel sequentially across the horizontal axis of the plot.

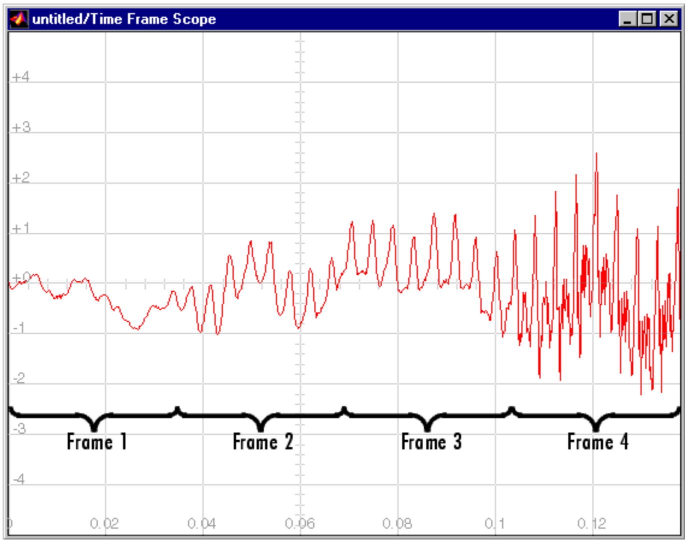
Scope Properties Pane

The **Scope Properties** pane enables you to plot time-domain, frequency-domain, or user-defined data, and adjust the horizontal display span of the plot. The scope displays frames of data, and updates the display for each new input frame.

The **Input domain** parameter specifies the domain of the input data. If you select `Time`, for M-by-N inputs containing time-domain data, the block treats each of the N input frames (columns) as a succession of M consecutive samples taken from a time series. That is, each data point in the input frame is assumed to correspond to a unique time value. Also, the **Time display span (number of frames)** parameter appears on the pane. Enter a scalar value greater than or equal to one that corresponds to the number of frames to be displayed across the width of the scope window.

If you select `Frequency` for the **Input domain** parameter, for *M*-by-*N* inputs containing frequency-domain data, the block treats each of the *N* input frames (columns) as a vector of spectral magnitude data corresponding to *M* consecutive ascending frequency indices. That is, when the input is a single column vector, u, each value in the input frame, `u(i)`, is assumed to correspond to a unique frequency value, `f(i)`, where `f(i+1)>f(i)`.

If you select `User-defined` for the **Input domain** parameter, the block does not assume that the input frame data is time-domain or frequency-domain data. You can plot the data in the appropriate manner. Also, the **Horizontal display span (number of frames)** parameter appears on the pane. Enter a scalar value greater than or equal to one that corresponds to the number of frames to be displayed across the width of the scope window.



Display Properties Pane

The **Display Properties** pane enables you to control how the block displays your data.

The **Show grid** parameter toggles the background grid on and off.

If you select the **Persistence** check box, the window maintains successive displays. That is, the scope does not erase the display after each frame (or collection of frames), but overlays successive input frames in the scope display.

If you select the **Frame number** check box, the block displays the number of the current frame in the input sequence on the scope window, and the block increments the count as each new input is received. Counting starts at 1 with the first input frame, and continues until the simulation stops.

If you select the **Channel legend** check box, a legend indicating the line color, style, and marker of each channel's data is added. When the input signal is labeled, that label is displayed in the channel legend. When the input signal is not labeled, but comes from a Concatenate block with labeled inputs, those labels are displayed in the channel legend. Otherwise, each channel in the legend is labeled with the channel number (`CH 1`, `CH 2`, etc.). Click-and-drag the legend to reposition it in the scope window; double-click on the line label to edit the text. If you rerun the simulation, the labels revert to the defaults.

If you select the **Compact display** check box, the scope completely fills the figure window. The scope does not display menus and axis titles, and it shows the numerical axis labels within the axes. If you clear the **Compact display** check box, the scope displays the axis labels and titles in a gray border surrounding the scope axes, and the window's menus and toolbar are visible.

If you select the **Open scope at start of simulation** check box, the scope opens at the start of the simulation. If you clear this parameter, the scope does not open automatically during the simulation. To view the scope, double-click the Vector Scope block, which brings up the scope as well as the block parameter dialog box. Use this feature when you have several scope blocks in a model and you do not want to view all the associated scopes during the simulation.

If the scope is not open during the simulation and you select the **Open scope immediately** check box, the block opens the scope and clears the check box.

The **Scope position** parameter specifies a four-element vector of the form

```
[left bottom width height]
```

specifying the position of the scope window on the screen, where (0,0) is the lower-left corner of the display. See the MATLAB® `figure` function for more information.

Axis Properties Pane

The parameters that are available on the **Axis Properties** pane depend on the setting of the **Input domain** parameter on the **Scope Properties** pane.

Time Domain Inputs

When **Time display limits** is set to `Auto`, the block scales the horizontal axis of time-domain signals automatically. The range of the time axis is $[0, S * T_{fi}]$, where T_{fi} is the input frame period, and S is the **Time display span (number of frames)** parameter on the **Scope Properties** pane. The spacing between time points is $T_{fi}/(M-1)$, where M is the number of samples in each consecutive input frame.

When **Time display limits** is set to `User-defined`, the **Minimum X-limit** and **Maximum X-limit** parameters set the range of the horizontal axis. Setting these parameters is analogous to setting the `xmin` and `xmax` values of the MATLAB `axis` function.

Minimum Y-limit and **Maximum Y-limit** parameters set the range of the vertical axis. Setting these parameters is analogous to setting the `ymin` and `ymax` values of the MATLAB axis function.

The **Y-axis title** is the text displayed to the left of the y-axis.

Frequency Domain Inputs

The **Frequency units** parameter specifies whether the frequency axis values should be in units of Hertz or rad/sec. When the **Frequency units** parameter is set to `Hertz`, the spacing between frequency points is $1/(M * T_s)$, where T_s is the sample time of the original time-domain signal. When the **Frequency units** parameter is set to `rad/sec`, the spacing between frequency points is $2\pi/(M * T_s)$.

The **Frequency range** parameter specifies the range of frequencies over which the magnitudes in the input should be plotted. The available options are `[0..Fs/2]`, `[-Fs/2..Fs/2]`, and `[0..Fs]`, where `Fs` is the original time-domain signal's sample frequency. The Vector Scope block assumes that the input data spans the range $[0, F_s)$, which is the same as the output from an FFT. To plot over the range `[0..Fs/2]` the scope truncates the input vector, leaving only the first half of the data, then plots these remaining samples over half the frequency range. To plot over the range `[-Fs/2..Fs/2]`, the scope reorders the input vector elements such that the last half of the data becomes the first half, and vice versa; then it relabels the *x*-axis accordingly.

If you select the **Inherit sample time from input** check box for frequency domain inputs, the block scales the frequency axis by reconstructing the frequency data from the frame-period of the frequency-domain input. This is valid when the following conditions hold:

- Each frame of frequency-domain data shares the same length as the frame of time-domain data from which it was generated; for example, when the FFT is computed on the same number of points as are contained in the time-domain input.
- The sample period of the time-domain signal in the simulation is equal to the period with which the physical signal was originally sampled.

- Consecutive frames containing the time-domain signal do not overlap each other; that is, a particular signal sample does not appear in more than one sequential frame.

In cases where not all of these conditions hold, specify the appropriate value for the **Sample time of original time series** parameter.

When **Frequency display limits** is set to `Auto`, the block scales the horizontal axis of frequency-domain signals automatically. To do this, the Vector Scope block needs to know the sample period of the original time-domain sequence represented by the frequency-domain data. Specify this period by entering a value for the **Sample time of original time series** parameter.

When **Frequency display limits** is set to `User-defined`, the **Minimum X-limit** and **Maximum X-limit** parameters set the range of the horizontal axis. Setting these parameters is analogous to setting the `xmin` and `xmax` values of the MATLAB `axis` function.

The **Amplitude scaling** parameter allows you to select `Magnitude` or `dB` scaling along the *y*-axis.

Minimum Y-limit and **Maximum Y-limit** parameters set the range of the vertical axis. Setting these parameters is analogous to setting the `ymin` and `ymax` values of the MATLAB axis function.

The **Y-axis title** is the text displayed to the left of the *y*-axis.

User-Defined Inputs

If you select the **Inherit sample increment from input** check box for user-defined input domains, the block scales the horizontal axis by computing the horizontal interval between samples in the input frame from the frame period of the input. For example, when the input frame period is 1, and there are 64 samples per input frame, the interval between samples is computed to be 1/64. Computing the interval this way is usually only valid when the following conditions hold:

- The input is a nonoverlapping time series; the *x*-axis on the scope represents time.

- The input's sample period (1/64 in the above example) is equal to the period with which the physical signal was originally sampled.

In cases where not all of these conditions hold, use the **X display offset** and **Increment per sample in input** parameters.

The **X-axis title** is the text displayed below the x-axis.

When **X display limits** is set to `Auto`, the block scales the horizontal axis of user-defined domain signals automatically. To do this, the Vector Scope block needs to know the spacing of the input data. Specify this spacing using the **Increment per sample in input** parameter, I_s. This parameter represents the numerical interval between adjacent x-axis points corresponding to the input data. The range of the horizontal axis is $[0, M * I_s * S]$, where M is the number of samples in each consecutive input frame, and S is the **Horizontal display span (number of frames)** parameter that you specify in the **Scope Properties** pane.

When **X display limits** is set to `User-defined`, the **Minimum X-limit** and **Maximum X-limit** parameters set the range of the horizontal axis. Setting these parameters is analogous to setting the `xmin` and `xmax` values of the MATLAB `axis` function.

Minimum Y-limit and **Maximum Y-limit** parameters set the range of the vertical axis. Setting these parameters is analogous to setting the `ymin` and `ymax` values of the MATLAB `axis` function.

The **Y-axis title** is the text displayed to the left of the y-axis.

Line Properties Pane

Use the parameters on the **Line Properties** pane to help you distinguish between two or more independent channels of data on the scope.

The **Line visibilities** parameter specifies which channel's data is displayed on the scope, and which is hidden. The syntax specifies the visibilities in list form, where the term `on` or `off` as a list entry specifies the visibility of the corresponding channel's data. The list entries are separated by the pipe symbol, `|`.

For example, a five-channel signal would ordinarily generate five distinct plots on the scope. To disable plotting of the third and fifth lines, enter the following visibility specification in the **Line visibilities** parameter.

```
 on | on | off | on | off
ch 1  ch 2  ch 3  ch 4  ch 5
```

Note that the first (leftmost) list item corresponds to the first signal channel (leftmost column of the input matrix).

The **Line styles** parameter specifies the line style with which each channel's data is displayed on the scope. The syntax specifies the channel line styles in list form, with each list entry specifying a style for the corresponding channel's data. The list entries are separated by the pipe symbol, `|`.

For example, a five-channel signal would ordinarily generate all five plots with a solid line style. To plot each line with a different style, enter

```
  - | -- | : | -. | -
ch 1  ch 2  ch 3 ch 4  ch 5
```

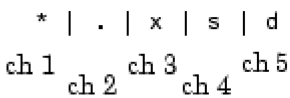
These settings plot the signal channels with the following styles.

Line Style	Command to Type in Line Style Parameter	Appearance
Solid	`-`	
Dashed	`--`	
Dotted	`:`	
Dash-dot	`-.`	
No line	`none`	No line appears

Note that the first (leftmost) list item, `'-'`, corresponds to the first signal channel (leftmost column of the input matrix). See the LineStyle property of the MATLAB `line` function for more information about the style syntax.

The **Line markers** parameter specifies the marker style with which each channel's samples are represented on the scope. The syntax specifies the channels' marker styles in list form, with each list entry specifying a marker for the corresponding channel's data. The list entries are separated by the pipe symbol, `|`.

For example, a five-channel signal would ordinarily generate all five plots with no marker symbol (that is, the individual sample points are not marked on the scope). To instead plot each line with a different marker style, you could enter



These settings plot the signal channels with the following styles.

Marker Style	Command to Type in Marker Style Parameter	Appearance
Asterisk	*	
Point	.	
Cross	x	

Marker Style	Command to Type in Marker Style Parameter	Appearance
Square	s	
Diamond	d	

Note that the leftmost list item, '*', corresponds to the first signal channel or leftmost column of the input matrix. See the Marker property of the MATLAB line function for more information about the available markers.

To produce a stem plot for the data in a particular channel, type the word stem instead of one of the basic marker shapes.

The **Line colors** parameter specifies the color in which each channel's data is displayed on the scope. The syntax specifies the channel colors in list form, with each list entry specifying a color (in one of the MATLAB ColorSpec formats) for the corresponding channel's data. The list entries are separated by the pipe symbol, |.

For example, a five-channel signal would ordinarily generate all five plots in the color black. To instead plot the lines with the color order below, enter

```
[0 0 0] | [0 0 1] | [1 0 0 ] | [0 1 0] | [.7529 0 .7529]
 ch 1      ch 2      ch 3       ch 4        ch 5
```

or

```
 'k' | 'b' | 'r' | 'g' | [.7529 0 .7529]
ch 1  ch 2  ch 3  ch 4       ch 5
```

These settings plot the signal channels in the following colors (8-bit RGB equivalents 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)	

Note that the leftmost list item, `'k'`, corresponds to the first signal channel or leftmost column of the input matrix. See the MATLAB function `ColorSpec` for more information about the color syntax.

Vector Scope Window

The title in the window title bar is the same as the block title. In addition to the standard MATLAB figure window menus such as **File**, **Window**, and **Help**, the Vector Scope window contains **Axes** and **Channels** menus.

The parameters that you set using the **Axes** menu apply to all channels. Many of the parameters in this menu are also accessible through the block parameters dialog box. For descriptions of these parameters, see "Display Properties Pane" on page 2-1352. Below are descriptions of other parameters in the **Axes** menu:

- **Refresh** erases all data on the scope display, except for the most recent trace. This command is useful in conjunction with the **Persistence** setting.
- **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 edit these values.

- **Save position** automatically updates the **Scope position** parameter in the **Axis properties** field 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, drag the window to the desired location, resize it, and select **Save position**. Note that the parameter dialog box must be closed when you select **Save position** in order for the **Scope position** parameter to be updated.

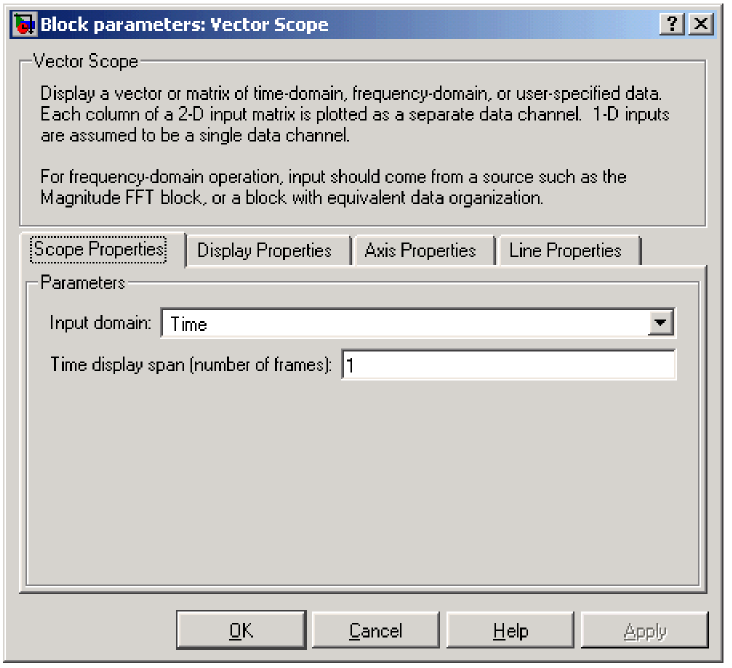
The properties listed in the **Channels** menu apply to a particular channel. All of the parameters in this menu are also accessible through the block parameters dialog box. For descriptions of these parameters, see "Line Properties Pane" on page 2-1357.

Many of these options can also be accessed by right-clicking with the mouse anywhere on the scope display. The menu that is displayed contains a combination of the options available in both the **Axes** and **Channels** menus.

Note When you select **Compact display** from the **Axes** menu, the **Axes** and **Channels** menus are no longer visible. Right-click in the Vector Scope window and click **Compact display** in order to make the menus reappear.

Dialog Box

Scope Properties Pane

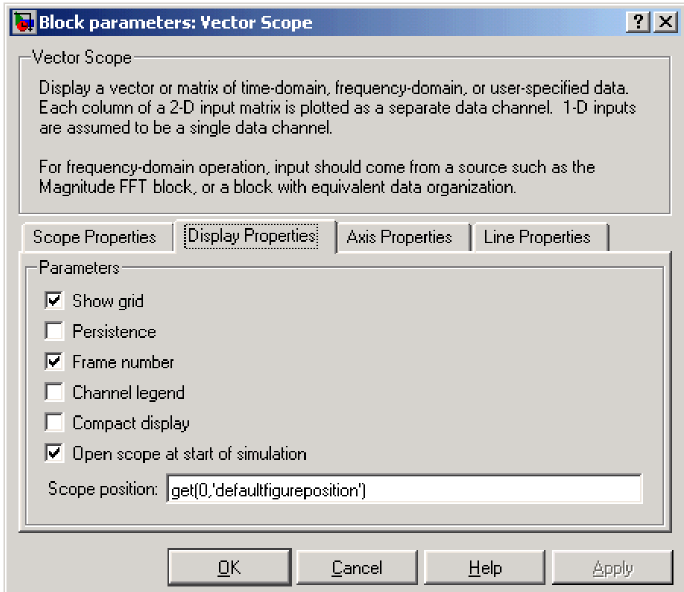


Input domain
: Select the domain of the input. Your choices are `Time`, `Frequency`, or `User-defined`. Tunable.

Time display span (number of frames)
: The number of consecutive frames to display (horizontally) on the scope at any one time. This parameter is visible when the **Input domain** parameter is set to `Time`.

Horizontal display span (number of frames)
: The number of consecutive frames to display (horizontally) on the scope at any one time. This parameter is visible when the **Input domain** parameter is set to `User-defined`.

Display Properties Pane



Show grid
: Toggle the scope grid on and off. Tunable.

Persistence
: Select this check box to maintain successive displays. That is, the scope does not erase the display after each frame (or collection of frames), but overlays successive input frames in the scope display. Tunable.

Frame number
: If you select this check box, the number of the current frame in the input sequence appears in the Vector Scope window. Tunable.

Channel legend
: Toggles the legend on and off. Tunable.

Compact display
Resizes the scope to fill the window. Tunable.

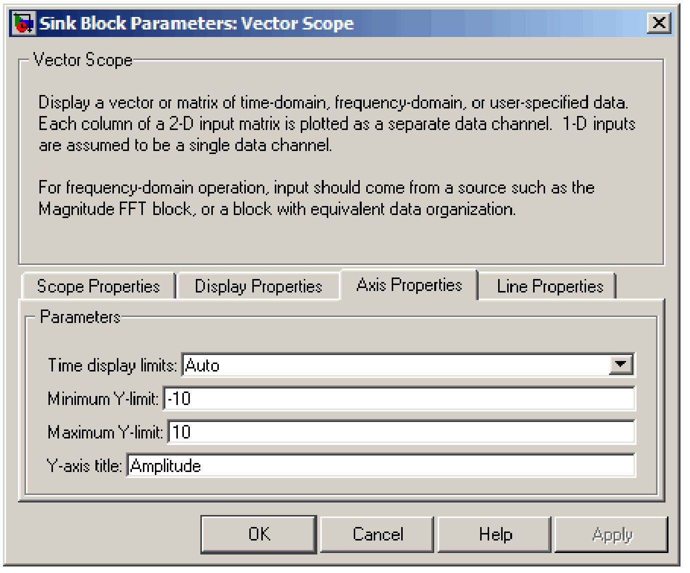
Open scope at start of simulation
Select this check box to open the scope at the start of the simulation. When this parameter is cleared, the scope does not open automatically during the simulation. Tunable.

Open scope immediately
If the scope is not open during simulation, select this check box to open it. This parameter is visible only while the simulation is running.

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.

Axis Properties Pane

The parameters that are available on the **Axis Properties** pane depend on the setting of the **Input domain** parameter on the **Scope Properties** pane. When `Time` is selected for the **Input domain** parameter, the following parameters are available on the **Axis Properties** pane:



Time display limits

Select `Auto` to have the limits of the *x*-axis set for you automatically, or `User-defined` to set the limits yourself in the **Minimum X-limit** and **Maximum X-limit** parameters.

Minimum X-limit

Specify the minimum value of the *x*-axis. Setting this parameter is analogous to setting the `xmin` value of the MATLAB `axis` function. This parameter is only visible if the **Time display limits** parameter is set to `User-defined`. Tunable.

Maximum X-limit

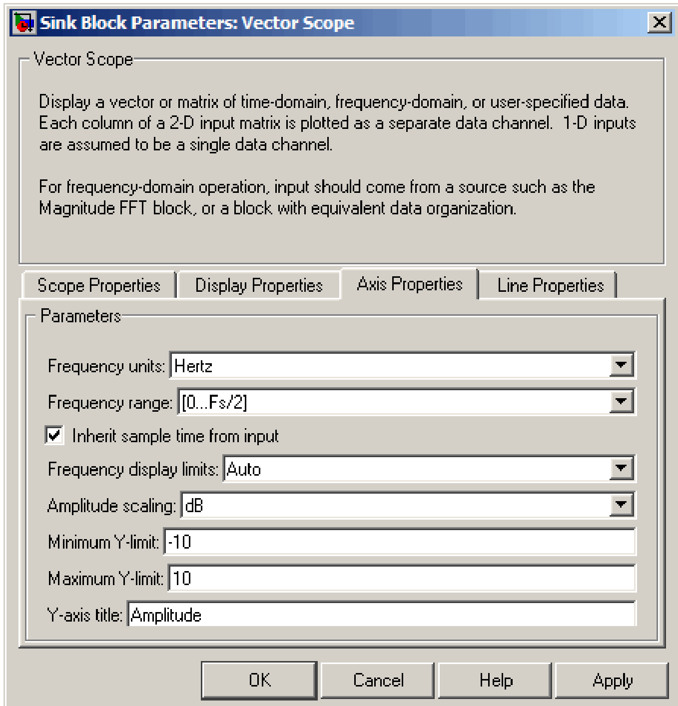
Specify the maximum value of the *x*-axis. Setting this parameter is analogous to setting the `xmax` value of the MATLAB `axis` function. This parameter is only visible if the **Time display limits** parameter is set to `User-defined`. Tunable.

Minimum Y-limit
: Specify the minimum value of the *y*-axis. Setting this parameter is analogous to setting the `ymin` value of the MATLAB `axis` function. Tunable.

Maximum Y-limit
: Specify the maximum value of the *y*-axis. Setting this parameter is analogous to setting the `ymax` value of the MATLAB `axis` function.Tunable.

Y-axis title
: Specify text to be displayed to the left of the *y*-axis. Tunable.

When `Frequency` is selected for the **Input domain** parameter, the following parameters are available on the **Axis Properties** pane:



Frequency units

Choose the frequency units for the *x*-axis, `Hertz` or `rad/sec`. Tunable.

Frequency range

Specify the frequency range over which to plot the data. Tunable.

Inherit sample time from input

If you select this check box, the block computes the time-domain sample period from the frame period and frame size of the frequency-domain input. Use this parameter only when the length of the each frame of frequency-domain data is the same as the length of the frame of time-domain data from which it was generated. Tunable.

Sample time of original time series
: Enter the sample period, T_s, of the original time-domain signal. This parameter is only visible when the **Inherit sample time from input** check box is not selected. Tunable.

Frequency display limits
: Select `Auto` to have the limits of the *x*-axis set for you automatically, or `User-defined` to set the limits yourself in the **Minimum X-limit** and **Maximum X-limit** parameters.

Minimum X-limit
: Specify the minimum value of the *x*-axis. Setting this parameter is analogous to setting the `xmin` value of the MATLAB `axis` function. This parameter is only visible if the **Frequency display limits** parameter is set to `User-defined`. Tunable.

Maximum X-limit
: Specify the maximum value of the *x*-axis. Setting this parameter is analogous to setting the `xmax` value of the MATLAB `axis` function. This parameter is only visible if the **Frequency display limits** parameter is set to `User-defined`. Tunable.

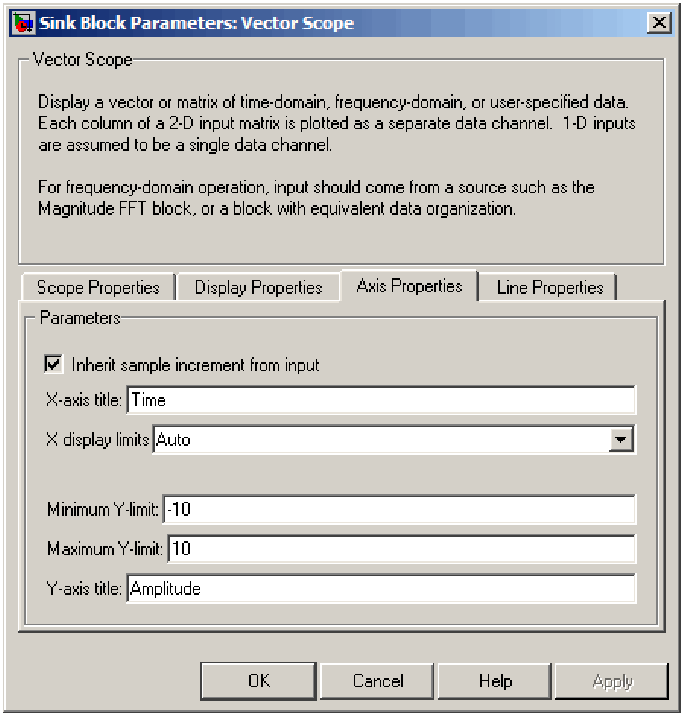
Amplitude scaling
: Choose the scaling for the *y*-axis, `dB` or `Magnitude`. Tunable.

Minimum Y-limit
: Specify the minimum value of the *y*-axis. Setting this parameter is analogous to setting the `ymin` value of the MATLAB `axis` function. Tunable.

Maximum Y-limit
: Specify the maximum value of the *y*-axis. Setting this parameter is analogous to setting the `ymax` value of the MATLAB `axis` function.Tunable.

Y-axis title
: Specify text to be displayed to the left of the *y*-axis. Tunable.

When `User-defined` is selected for the **Input domain** parameter, the following parameters are available on the **Axis Properties** pane:



Inherit sample increment from input

When you select this check box, the block scales the horizontal axis by computing the horizontal interval between samples in the input frame from the frame period of the input. Use this parameter only when the input's sample period is equal to the period with which the physical signal was originally sampled. Tunable.

X display offset

Specify an offset for the *x*-axis display. This parameter is only visible when the **Inherit sample increment from input** check box is not selected. Tunable.

Increment per sample in input

Enter the numerical interval between adjacent *x*-axis points corresponding to the user-defined input data. This parameter is only visible when the **Inherit sample increment from input** check box is not selected. Tunable.

X-axis title

Enter the text to be displayed below the *x*-axis. Tunable.

X display limits

Select `Auto` to have the limits of the *x*-axis set for you automatically, or `User-defined` to set the limits yourself in the **Minimum X-limit** and **Maximum X-limit** parameters.

Minimum X-limit

Specify the minimum value of the *x*-axis. Setting this parameter is analogous to setting the `xmin` value of the MATLAB `axis` function. This parameter is only visible if the **X display limits** parameter is set to `User-defined`. Tunable.

Maximum X-limit

Specify the maximum value of the *x*-axis. Setting this parameter is analogous to setting the `xmax` value of the MATLAB `axis` function. This parameter is only visible if the **X display limits** parameter is set to `User-defined`. Tunable.

Minimum Y-limit

Specify the minimum value of the *y*-axis. Setting this parameter is analogous to setting the `ymin` value of the MATLAB `axis` function. Tunable.

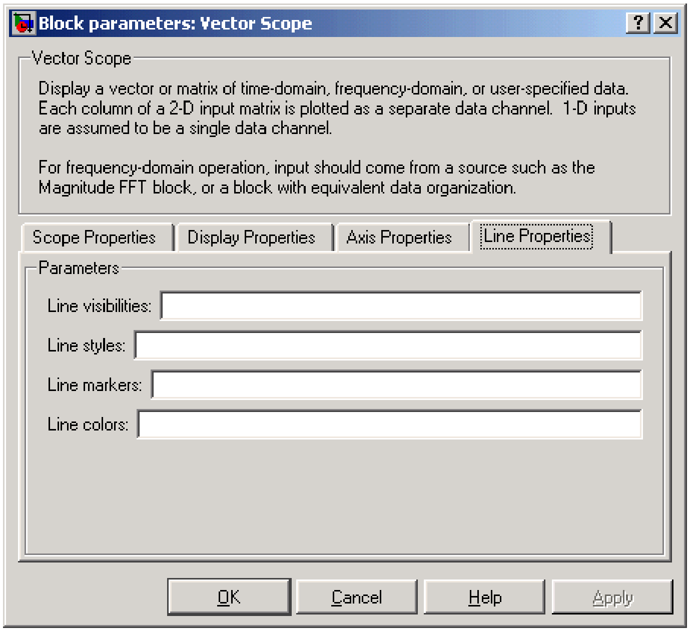
Maximum Y-limit

Specify the maximum value of the *y*-axis. Setting this parameter is analogous to setting the `ymax` value of the MATLAB `axis` function.Tunable.

Y-axis title

Specify text to be displayed to the left of the *y*-axis. Tunable.

Line Properties Pane



Line visibilities
: Enter `on` or `off` to specify the visibility of the various channels' scope traces. Separate your choices for each channel with by a pipe (|) symbol. Tunable.

Line styles
: Enter the line styles of the various channels' scope traces. Separate your choices for each channel with by a pipe (|) symbol. Tunable.

Line markers
: Enter the line markers of the various channels' scope traces. Separate your choices for each channel with by a pipe (|) symbol. Tunable.

Line colors

Enter the colors of the various channels' scope traces using the `ColorSpec` formats. Separate your choices for each channel with by a pipe (`|`) symbol. Tunable.

Supported Data Types

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

See Also

Matrix Viewer	Signal Processing Blockset
Spectrum Scope	Signal Processing Blockset

Waterfall

Purpose

View vectors of data over time

Library

Signal Processing Sinks

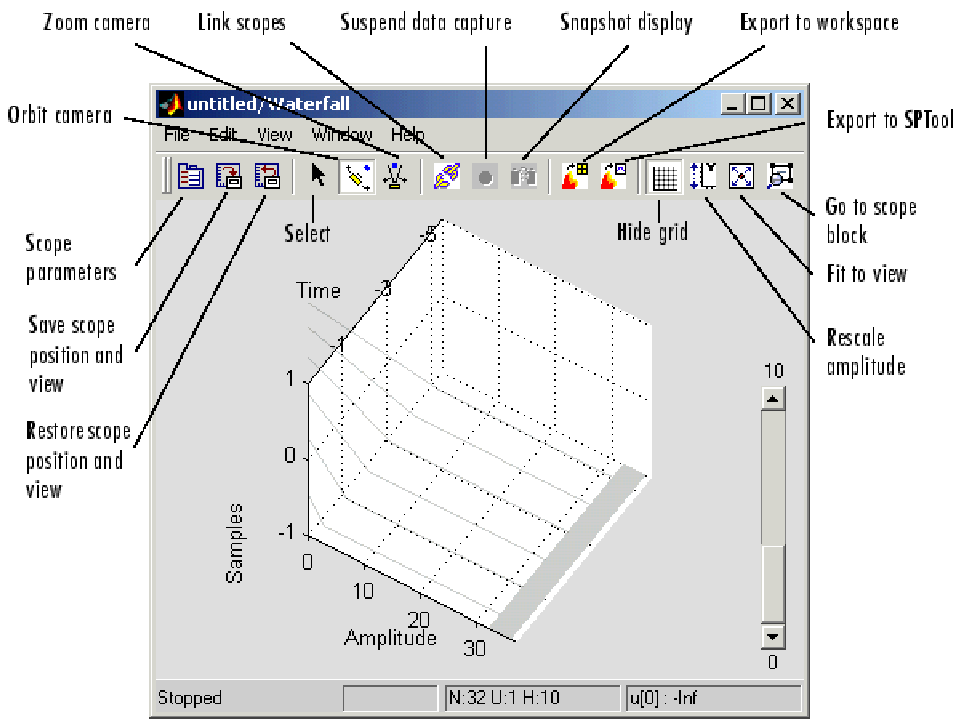
`dspsnks4`

Description

Waterfall

The Waterfall block displays multiple vectors of data at one time. These vectors represent the input data at consecutive sample times. The input to the block can be real or complex-valued data vectors of any data type including fixed-point data types. However, the input is converted to double-precision before the block processes the data. The Waterfall block displays only real-valued, double-precision vectors of data.

The data is displayed in a three-dimensional axis in the Waterfall window. By default, the x-axis represents amplitude, the y-axis represents samples, and the z-axis represents time. You can adjust the number of sample vectors that the block displays, move and resize the Waterfall window, and modify block parameter values during the simulation. The Waterfall window has toolbar buttons that enable you to zoom in on displayed data, suspend data capture, freeze the scope's display, save the scope position, and export data to the workspace. The toolbar buttons are labeled in the following figure, which shows the Waterfall window as it appears when you double-click a Waterfall block.



Sections of This Reference Page

- "Waterfall Parameters" on page 2-1376
- "Display Parameters" on page 2-1377
- "Axes Parameters" on page 2-1378
- "Data History Parameters" on page 2-1379
- "Triggering Parameters" on page 2-1380
- "Scope Trigger Function" on page 2-1383
- "Transform Parameters" on page 2-1386
- "Scope Transform Function" on page 2-1388

- "Examples" on page 2-1388

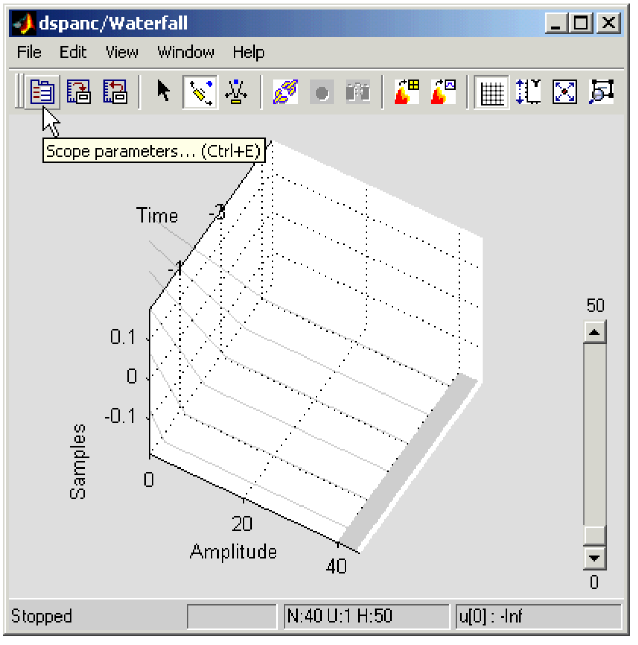
Waterfall Parameters

You can control the display and behavior of the Waterfall window using the Parameters dialog box.

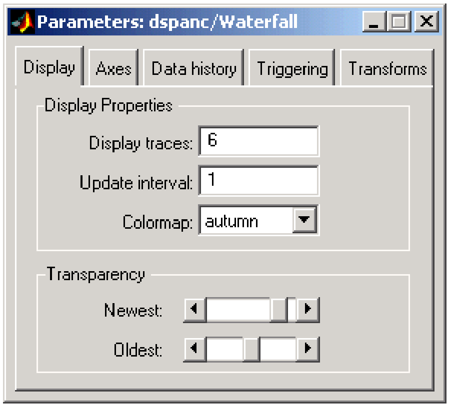
Note You can alter the Waterfall parameters while the simulation is running. However, when you make changes to values in text boxes, you must click **Enter** or click outside the text box before the block accepts your changes.

---

1 To open the Parameters dialog box, click the **Scope parameters** button.



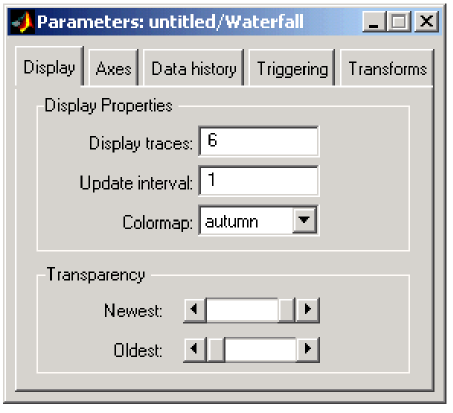
The Parameters dialog box appears.



2 Click on the different panes to enter parameter settings.

Display Parameters

The following parameters control the Waterfall window's display.



Display traces

Enter the number of vectors of data to be displayed in the Waterfall window.

Update interval

Enter the number of vectors the block should store before it displays them to the window.

Colormap

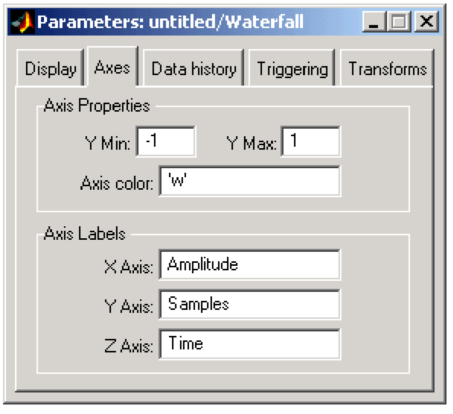
Choose a colormap for the displayed data.

Transparency

Specify the transparency of the newest and oldest data vectors. Placing the slider in the left-most position tells the block to make the data vector transparent. Placing the slider in the right-most position tells the block to make the data vector opaque. The intermediate data vectors transition between the two chosen transparency values.

Axes Parameters

The following parameters control the axes in the Waterfall window.



Y Min

Enter the minimum value of the *y*-axis.

Y Max

Enter the maximum value of the *y*-axis.

Axis color

Enter a background color for the axes. Specify the color using a character string. For example, to specify black, enter `'k'`.

X Axis

Enter the *x*-axis label.

Y Axis

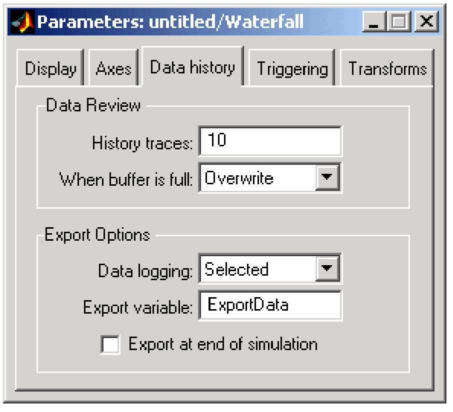
Enter the *y*-axis label.

Z Axis

Enter the *z*-axis label.

Data History Parameters

The following parameters control how many input data vectors the Waterfall block stores. They also control how the data is exported to the MATLAB® workspace or SPTool.



History traces

Enter the number of vectors (traces) that you want the block to store.

When the buffer is full

Use this parameter to control the behavior of the block when the buffer is filled:

- `Overwrite` — The old data is replaced with the new data.
- `Suspend` — The block stops storing data in the buffer; however, the simulation continues to run.
- `Extend` — The block extends the buffer so that it can continue to store all the input data.

Data logging
Use this parameter to control which data is exported from the block:

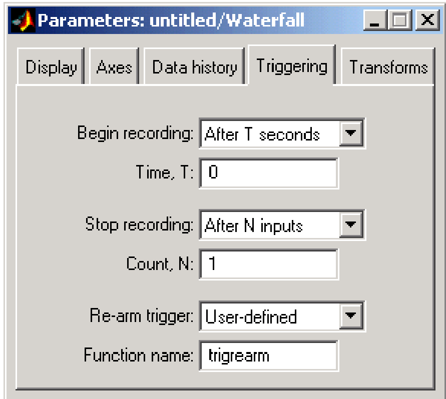
- `Selected` — The selected data vector is exported.
- `All visible` — All of the data vectors displayed in the Waterfall window are exported.
- `All history` — All of the data vectors stored in the block's history buffer are exported.

Export variable
Enter the name of the variable that represents your data in the MATLAB workspace or SPTool. The default variable name is `ExportData`.

Export at end of simulation
Select this check box to automatically export the data to the MATLAB workspace when the simulation stops.

Triggering Parameters

The following parameters control when the Waterfall block starts and stops capturing data.



Begin recording

This parameter controls when the Waterfall block starts capturing data:

- `Immediately` — The Waterfall window captures the input data as soon as the simulation starts.
- `After T seconds` — The **Time, T** parameter appears in the dialog box. Enter the number of seconds the block should wait before it begins capturing data.
- `After N inputs` — The **Count, N** parameter appears in the dialog box. Enter the number of inputs the block should receive before it begins capturing data.
- `User-defined` — The **Function name** parameter appears in the dialog box. Enter the name of a MATLAB function that defines when the block should begin capturing data. For more information about how you define this function, see “Scope Trigger Function” on page 2-1383.

Stop recording

This parameter controls when the Waterfall block stops capturing data:

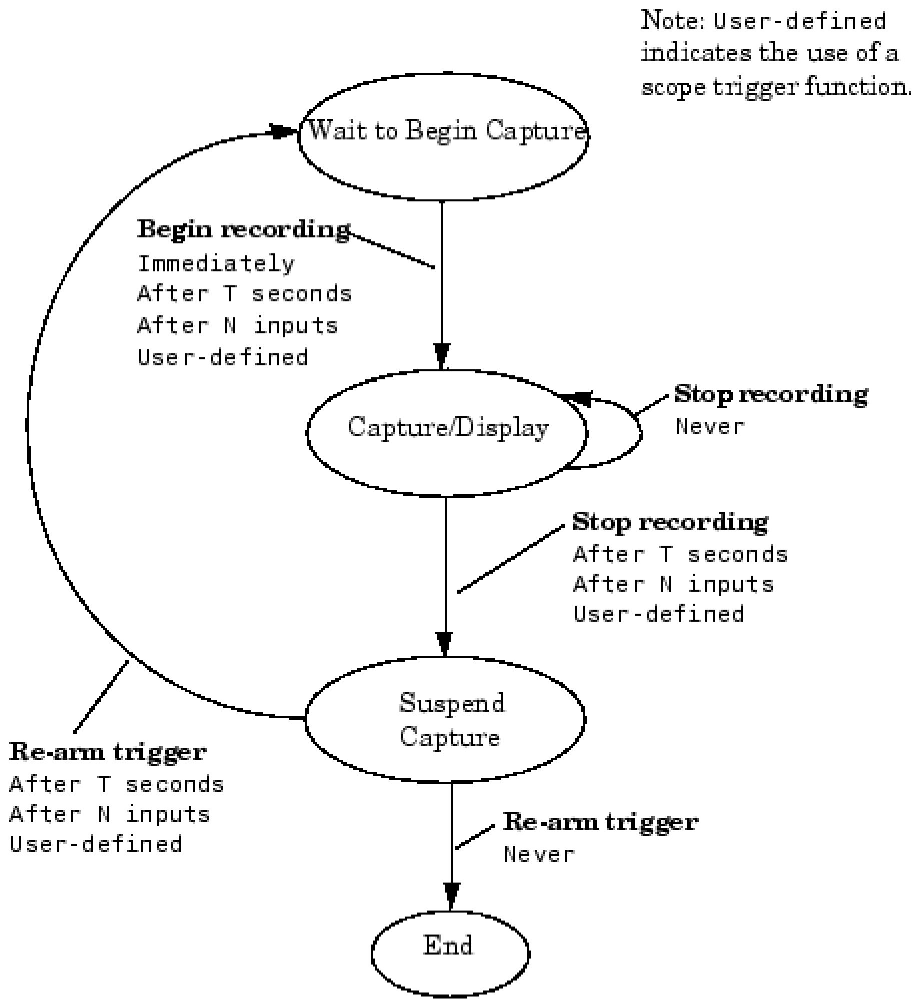
- `Never` — The block captures the input data as long as the simulation is running.
- `After T seconds` — The **Time, T** parameter appears in the dialog box. Enter the number of seconds the block should wait before it stops capturing data.
- `After N inputs` —The **Count, N** parameter appears in the dialog box. Enter the number of inputs the block should receive before it stops capturing data.
- `User-defined` — The **Function name** parameter appears in the dialog box. Enter the name of a MATLAB function that defines when the block should stop capturing data. For more information about how you define this function, see "Scope Trigger Function" on page 2-1383.

Re-arm trigger

This parameter controls when the Waterfall block begins waiting to capture data. It is available only when you select `After T seconds`, `After N inputs`, or `User-defined` for the **Stop recording** parameter:

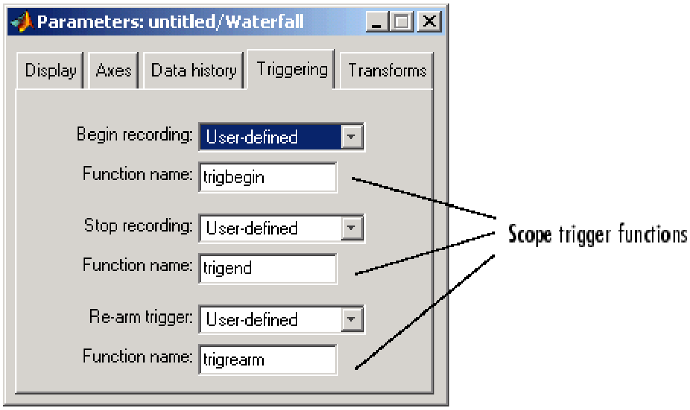
- `Never` — The Waterfall Scope block starts and stops capturing data as defined by the **Begin recording** and **Stop recording** parameters.
- `After T seconds` — The **Time, T** parameter appears in the dialog box. Enter the number of seconds the block should wait before it begins waiting to capture data.
- `After N inputs` — The **Count, N** parameter appears in the dialog box. Enter the number of inputs the block should receive before it begins waiting to capture data.
- `User-defined` — The **Function name** parameter appears in the dialog box. Enter the name of a MATLAB function that defines when the block should begin waiting to capture data. For more information about how you define this function, see "Scope Trigger Function" on page 2-1383.

The triggering process is illustrated in the state diagram below.



Scope Trigger Function

You can create custom scope trigger functions to control when the scope starts, stops, or begins waiting to capture data.



These functions must be valid MATLAB functions and be located either in the current directory or on the MATLAB path.

Each scope trigger function must have the following form

```
y = functionname(blk,t,u),
```

where `functionname` refers to the name you give your scope trigger function. The variable `blk` is the Simulink® block handle. When the scope trigger function is called by the block, Simulink automatically populates this variable with the handle of the Waterfall block. The variable `t` is the current simulation time, represented by a real, double-precision, scalar value. The variable `u` is the vector input to the block. The output of the scope trigger function, `y`, is interpreted as a logical signal. It is either true or false:

- Begin recording scope trigger function
  - When the output of this scope trigger function is true, the Waterfall block starts capturing data.
  - When the output is false, the block remains in its current state.
- Stop recording scope trigger function
  - When the output of this scope trigger function is true, the block stops capturing data.

- When the output is false, the block remains in its current state.

- Re-arm trigger scope trigger function
  - When the output of this scope trigger function is true, the block waits for a begin recording event.
  - When the output is false, the block remains in its current state.

Note The Waterfall block passes its input data directly to the scope trigger functions. These functions do not use the transformed data defined by the Transform parameters.

The following is an example of a scope trigger function. This function, called `trigPower` detects when the energy in u exceeds a certain threshold.

```
function y = trigPower(blk, t, u)

y = (u'*u > 2300);
```

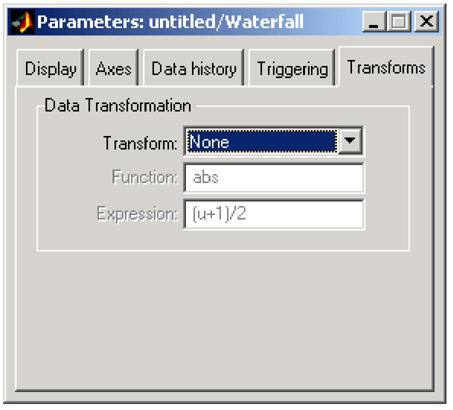
The following is another example of a scope trigger function. This function, called `count3`, triggers the scope once three vectors with positive means are input to the block. Then, the function resets itself and begins searching for the next three input vectors with positive means. This scope trigger function is valid only when one Waterfall block is present in your model.

```
function y = count3(blk, t, u)

persistent state;
if isempty(state); state = 0; end
if mean(u)>0; state = state+1; end
y = (state>=3);
if y; state = 0; end
```

Transform Parameters

The following parameters transform the input data to the Waterfall block. The result of the transform is displayed in the Waterfall window.



Note The block assumes that the input to the block corresponds to the **Transform** parameter you select. For example, when you choose `Complex-> Angle`, the block assumes that the input is complex. The block does not produce an error when the input is not complex. Therefore, you must verify the format of your input data to guarantee that a meaningful result is displayed in the Waterfall window.

Transform

Choose a transform that you would like to apply to the input of the Waterfall block:

- `None` — The input is displayed as it is received by the block.
- `Amplitude-> dB` — The block converts the input amplitude into decibels.
- `Complex-> Mag Lin` — The block converts the complex input into linear magnitude.

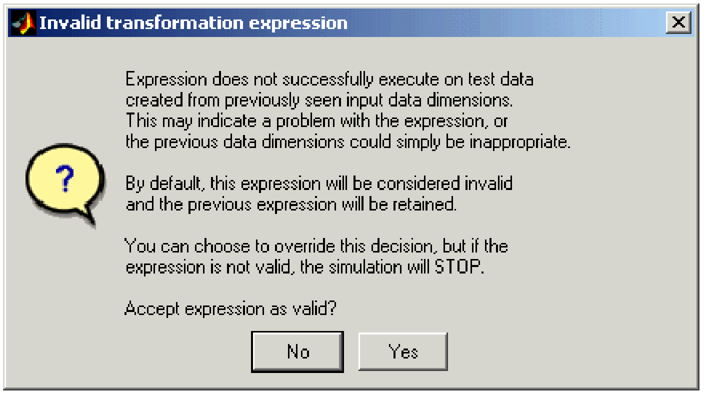
- `Complex-> Mag dB` — The block converts the complex input into magnitude in decibels.
- `Complex-> Angle` — The block converts the complex input into phase.
- `FFT-> Mag Lin Fs/2` — The block takes the linear magnitude of the FFT input and plots it from 0 to the Nyquist frequency.
- `FFT-> Mag dB Fs/2` — The block takes the magnitude of the FFT input, converts it to decibels, and plots it from 0 to the Nyquist frequency.
- `FFT-> Angle Fs/2` — The block converts the FFT input into phase and plots it from 0 to the Nyquist frequency.
- `Power-> dB` — The block converts the input power into decibels.

Function

This parameter is only available when you select `User-defined fcn` for the **Transform** parameter. Enter a function that you would like to apply to the input of the Waterfall block. For more information about how you define this function, see "Scope Transform Function" on page 2-1388.

Expression

This parameter is only available when you select `User-defined expr` for the **Transform** parameter. Enter an expression that you would like to apply to the input of the Waterfall block. The result of this expression must be real-valued. When you write the expression, be sure to include only one unknown variable. The block assumes this unknown variable represents the input to the block. When the block believes your expression is invalid, the following window appears.



When you click **No**, your expression is not applied to the input. When you click **Yes** and your expression is invalid, your simulation stops and Simulink displays an error.

Scope Transform Function

You can create a scope transform function to control how the Waterfall block transforms your input data. This function must have a valid MATLAB function name and be located either in the current directory or on the MATLAB path.

Your scope transform function must have the following form

```
y = functionname(u),
```

where `functionname` refers to the name you give your function. The variable `u` is the real or complex vector input to the block. The output of the scope transform function, `y`, must be a double-precision, real-valued vector. When it is not, the simulation stops and Simulink displays an error. Note that the output vector does not need to be the same size as the input vector.

Examples

The following examples illustrate some capabilities of the Waterfall block.

- "Exporting Data" on page 2-1389

- "Capturing Data" on page 2-1390
- "Linking Scopes" on page 2-1390
- "Selecting Data" on page 2-1392
- "Zooming" on page 2-1394
- "Rotating the Display" on page 2-1394
- "Scaling the Axes" on page 2-1394
- "Saving Scope Settings" on page 2-1395

Exporting Data

You can use the Waterfall block to export data to the MATLAB workspace or to SPTool:

1. Open and run the `dspanc` demo.

2. While the simulation is running, click the **Export to Workspace** button.

3. Type `whos` at the MATLAB command line.

   The variable `ExportData` appears in your MATLAB workspace. `ExportData` is a 40-by-6 matrix. This matrix represents the six data vectors that were present in the Waterfall window at the time you clicked the **Export to Workspace** button. Each column of this matrix contains 40 filter coefficients. The columns of data were captured at six consecutive instants in time.

   You can control what data is exported using the **Data logging** parameter in **Data history** pane of the Parameters dialog box. For more information, see "Data History Parameters" on page 2-1379.

4. While the simulation is running, click the **Export to SPTool** button.

   The SPTool GUI opens and the variable `ExportData` is displayed in the **Signals** list.

For more information about SPTool, see the Signal Processing Toolbox™ documentation.

Capturing Data

You can use the Waterfall block to interact with your data while it is being captured:

1. Open and run the `dspanc` demo.
2. While the simulation is running, click the **Suspend data capture** button.

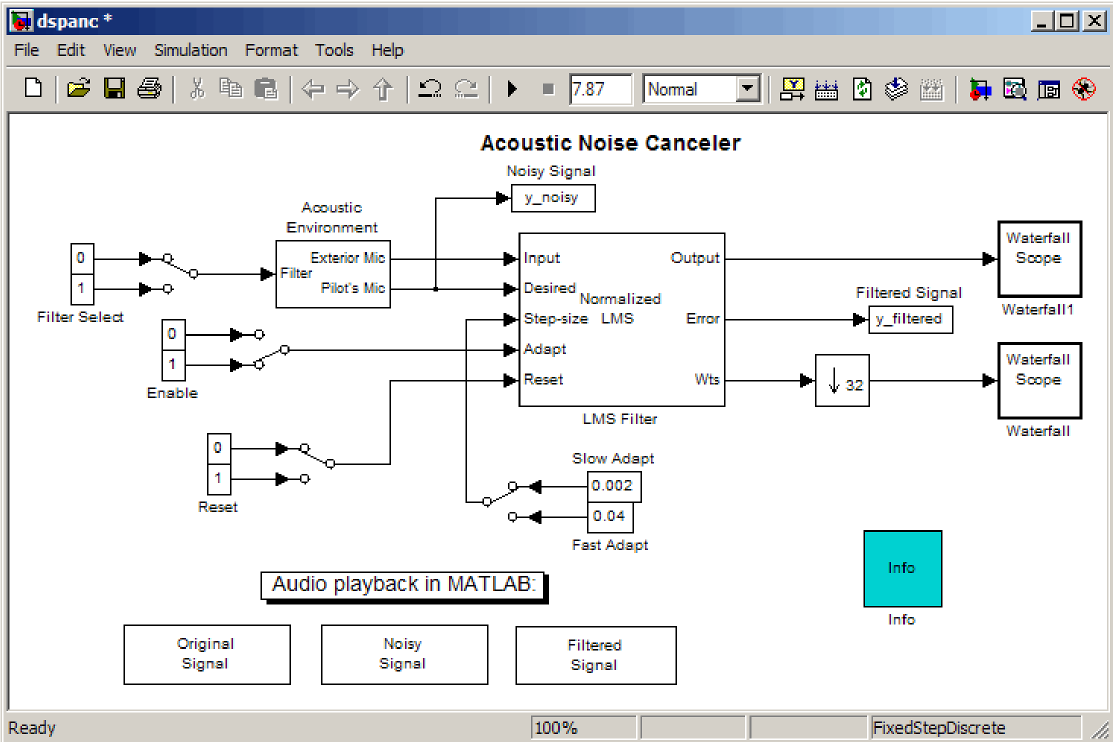
   The Waterfall block no longer captures or displays the data coming from the Downsample block.
3. To continue capturing data, click the **Resume data capture** button.
4. To freeze the data display while continuing to capture data, click the **Snapshot display** button.
5. To view the Waterfall block that the data is coming from, click the **Go to scope block** button.

   In the Simulink model window, the Waterfall block that corresponds to the active Waterfall window flashes. This feature is helpful when you have more than one Waterfall block in a model and you want to clarify which data is being displayed.

Linking Scopes

You can link several Waterfall blocks together in order to capture the effect of a model event in all of the Waterfall windows in the model:

1. Open the `dspanc` demo.
2. Drag a second Waterfall block into the demo model.
3. Connect this block to the Output port of the LMS Filter block as shown in the figure below.



4 Run the model and view the model behavior in both Waterfall windows.

5 In the dspanc/Waterfall window, click the **Link scopes** button.

6 In the same window, click the **Suspend data capture** button.

The data capture is suspended in both scope windows.

7 Click the **Resume data capture** button.

The data capture resumes in both scope windows.

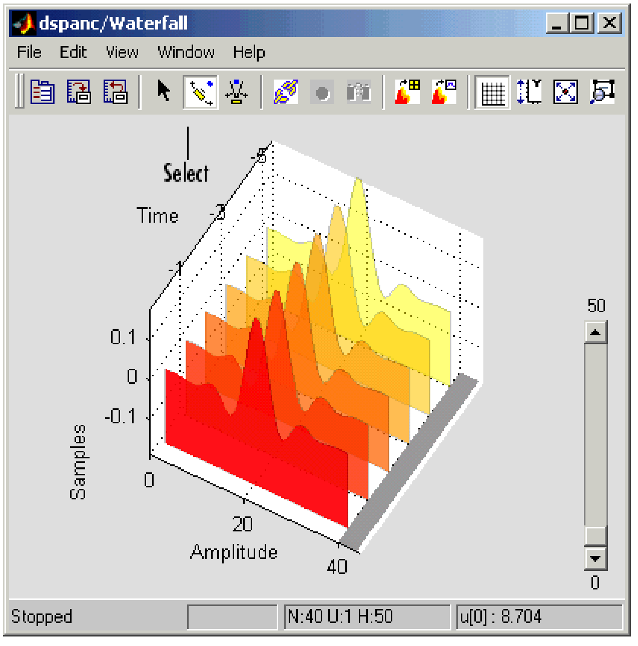
8 In the dspanc/Waterfall window, click the **Snapshot display** button.

In both scope windows, the data display freezes while the block continues to capture data.

9 To continue displaying the captured data, click the **Resume display** button.

Selecting Data

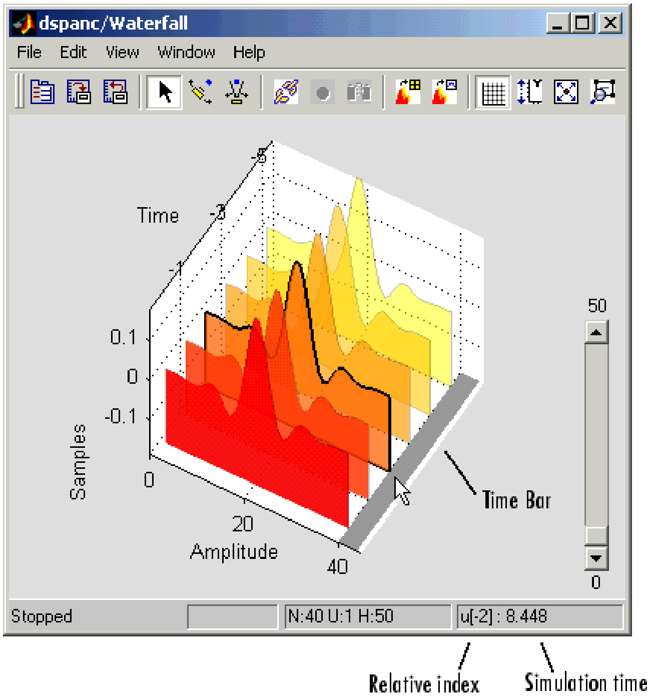
The following figure shows the Waterfall window displaying the output of the `dspanc` demo:



1 To select a particular set of data, click the **Select** button.

2 Click on the Time Bar at the bottom right of the axes to select a vector of data.

The Waterfall block highlights the selected trace.



While the simulation is running, in the bottom right corner, the Waterfall window displays the relative index of the selected trace. For example, in the previous figure, the selected vector is two sample times away from the most current data vector. When the simulation is stopped, the Waterfall window displays both the relative index and the simulation time associated with the selected trace.

3 To deselect the data vector, click it again.

4 Click-and-drag along the Time Bar.

Your selection follows the movement of the pointer.

You can use this feature to choose a particular vector to export to the MATLAB workspace or SPTool. For more information, see “Data History Parameters” on page 2-1379.

Zooming

You can use the Waterfall window to zoom in on data:

1. Click the **Zoom camera** button.
2. In the Waterfall window, click and hold down the left mouse button.
3. Move the mouse up and down and side-to-side to move closer and farther away from the axes.
4. To resize the axes to fit the Waterfall window, click the **Fit to view** button.

Rotating the Display

You can rotate the data displayed in the Waterfall window:

1. Click on the **Orbit camera** button.
2. In the Waterfall window, click and hold down the left mouse button.
3. Move the mouse in a circular motion to rotate the axes.
4. To return to the position of the original axes, click the **Restore scope position and view** button.

Scaling the Axes

You can use the Waterfall window to rescale the *y*-axis values:

1. Open and run the `dspanc` demo.
2. Click the **Rescale amplitude** button.

   The *y*-axis changes so that its minimum value is zero. The maximum value is scaled to fit the data displayed.

Alternatively, you can scale the *y*-axis using the **Y Min** and **Y Max** parameters in the **Axes** pane of the Parameters dialog box. This is helpful when you want to undo the effects of rescaling the amplitude. For more information, see "Axes Parameters" on page 2-1378.

Saving Scope Settings

The Waterfall block can save the screen position and viewpoint of the Waterfall window:

1. Click the **Save scope position and view** button.
2. Close the Waterfall window.
3. Reopen the Waterfall window.

   It reopens at the same place on your screen. The viewpoint of the axes also remains the same.

Supported Data Types

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

The Waterfall block accepts any of these data types as input. However, the input is converted to double-precision before the block processes the data. The Waterfall block displays only real-valued, double-precision vectors of data.

See Also

Scope	Simulink
Time Scope	Signal Processing Blockset

Vector Scope	Signal Processing Blockset
Spectrum Scope	Signal Processing Blockset
Matrix Viewer	Signal Processing Blockset
Signal To Workspace	Signal Processing Blockset
Triggered To Workspace	Signal Processing Blockset

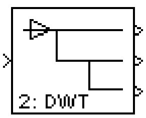
Purpose

Decompose signal into components of logarithmically decreasing frequency intervals and sample rates (requires the Wavelet Toolbox™ product)

Library

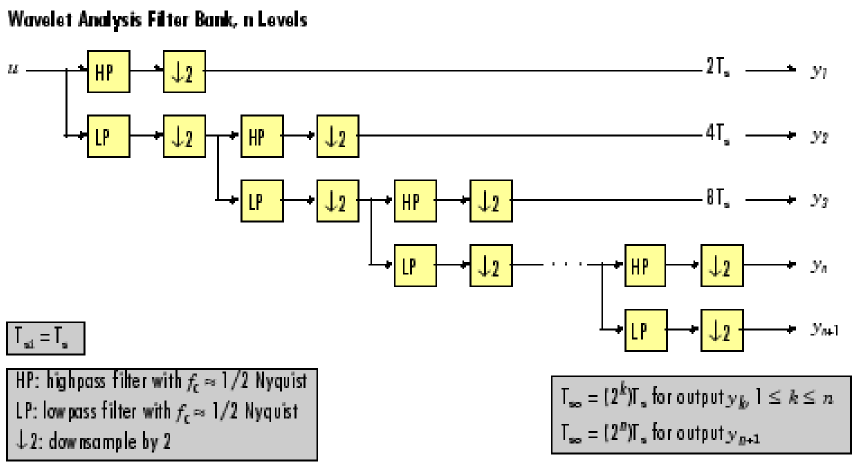
`dspobslib`

Description



Note The Wavelet Analysis block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the DWT block.

The Wavelet Analysis block uses the Wavelet Toolbox `wfilters` function to construct a dyadic analysis filter bank that decomposes a broadband signal into a collection of successively more bandlimited components. An n-level filter bank structure is shown below, where n is specified by the **Number of levels** parameter.



At each level, the low-frequency output of the previous level is decomposed into adjacent high- and low-frequency subbands by a highpass (HP) and lowpass (LP) filter pair. Each of the two output subbands is half the bandwidth of the input to that level. The bandlimited output of each filter is maximally decimated by a factor of 2 to preserve the bit rate of the original signal.

Filter Coefficients

The filter coefficients for the highpass and lowpass filters are computed by the Wavelet Toolbox function `wfilters`, based on the wavelet specified in the **Wavelet name** parameter. The table below lists the available options.

Wavelet Name	Sample Wavelet Function Syntax
Haar	`wfilters('haar')`
Daubechies	`wfilters('db4')`
Symlets	`wfilters('sym3')`
Coiflets	`wfilters('coif1')`
Biorthogonal	`wfilters('bior3.1')`
Reverse Biorthogonal	`wfilters('rbio3.1')`
Discrete Meyer	`wfilters('dmey')`

The **Daubechies**, **Symlets**, and **Coiflets** options enable a secondary **Wavelet order** parameter that allows you to specify the wavelet order. For example, if you specify a **Daubechies** wavelet with **Wavelet order** equal to `6`, the Wavelet Analysis block calls the `wfilters` function with input argument `'db6'`.

The **Biorthogonal** and **Reverse Biorthogonal** options enable a secondary **Filter order [synthesis / analysis]** parameter that allows you to independently specify the wavelet order for the analysis and synthesis filter stages. For example, if you specify a **Biorthogonal** wavelet with **Filter order [synthesis / analysis]** equal to `[2 / 6]`, the Wavelet Analysis block calls the `wfilters` function with input argument `'bior2.6'`.

See the Wavelet Toolbox documentation for more information about the `wfilters` function. If you want to explicitly specify the FIR coefficients for the analysis filter bank, use the Dyadic Analysis Filter Bank block.

Tree Structure

The wavelet tree structure has n+1 outputs, where n is the number of levels. The sample rate and bandwidth of the top output are half the input sample rate and bandwidth. The sample rate and bandwidth of each additional output (except the last) are half that of the output from the previous level. In general, for an input with sample period $T_{si} = T_s$, and bandwidth BW, output y_k has sample period $T_{so,k}$ and bandwidth BW_k.

$$T_{so,k} = \begin{cases} \left(2^k\right)T_s & (1 \le k \le n) \\ \left(2^n\right)T_s & (k = n+1) \end{cases}$$

$$BW_k = \begin{cases} \dfrac{BW}{2^k} & (1 \le k \le n) \\ \dfrac{BW}{2^n} & (k = n+1) \end{cases}$$

Note that in frame-based mode, the change in the sample period of output y_k is reflected by its frame size $M_{o,k}$, rather than by its frame rate.

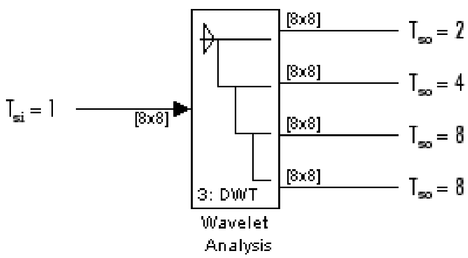
$$M_{o,k} = \begin{cases} \dfrac{M_i}{2^k} & (1 \le k \le n) \\ \dfrac{M_i}{2^n} & (k = n+1) \end{cases}$$

The bottom two outputs (y_n and y_{n+1}) share the same sample period, bandwidth, and frame size because they originate at the same tree level.

Sample-Based Operation

An M-by-N sample-based matrix input is treated as $M*N$ independent channels, and the block filters each channel independently over time. The output at each port is the same size as the input, one output channel for each input channel. As described earlier, each output port has a different sample period.

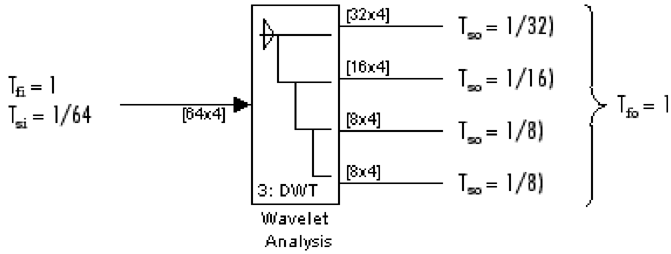
The figure below shows the input and output sample periods for a 64-channel sample-based input to a three-level filter bank. The input has a period of 1, so the fastest output has a period of 2.



Frame-Based Operation

An M_i-by-N frame-based matrix input is treated as N independent channels, and the block filters each channel independently over time. The input frame size M_i must be a multiple of 2^n, and n is the number of filter bank levels. For example, a frame size of 8 would be appropriate for a three-level tree (2^3=8). The number of columns in each output is the same as the number of columns in the input.

Each output port has the same frame period as the input. The reduction in the output sample rates results from the smaller output frame sizes, as shown in the example below for a four-channel input to a three-level filter bank.



Zero Latency

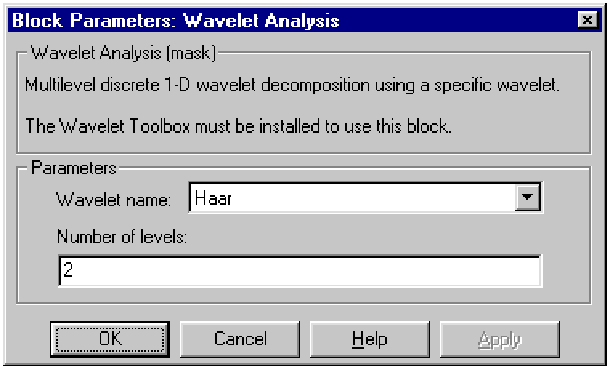
The Wavelet Analysis block has no tasking latency for frame-based operation, which is always single-rate. The block therefore analyzes the first input sample (received at t=0) to produce the first output sample at each port.

Nonzero Latency

For sample-based operation, the Wavelet Analysis block is multirate and has 2^{n-1} samples of latency in both Simulink® tasking modes. As a result, the block repeats a zero initial condition in each channel for the first 2^{n-1} output samples, before propagating the first analyzed input sample (computed from the input received at t=0).

Note For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and the topic on models with multiple sample rates in the Real-Time Workshop® documentation.

Dialog Box



The parameters displayed in the dialog box vary for different wavelet types. Only some of the parameters listed below are visible in the dialog box at any one time.

Wavelet name
: The wavelet used in the analysis.

Wavelet order
: The order for the **Daubechies**, **Symlets**, and **Coiflets** wavelets. This parameter is available only when one of these wavelets is selected in the **Wavelet name** menu.

Filter order [synthesis / analysis]
: The filter orders for the synthesis and analysis stages of the **Biorthogonal** and **Reverse Biorthogonal** wavelets. For example, `[2 / 6]` selects a second-order synthesis stage and a sixth-order analysis stage. The **Filter order** parameter is available only when one of the above wavelets is selected in the **Wavelet name** menu.

Number of levels
: The number of filter bank levels. An n-level structure has $n+1$ outputs.

References

Fliege, N. J. *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets*. West Sussex, England: John Wiley & Sons, 1994.

Strang, G. and T. Nguyen. *Wavelets and Filter Banks*. Wellesley, MA: Wellesley-Cambridge Press, 1996.

Vaidyanathan, P. P. *Multirate Systems and Filter Banks*. Englewood Cliffs, NJ: Prentice Hall, 1993.

Supported Data Types

- Double-precision floating point

See Also

Dyadic Analysis Filter Bank	Signal Processing Blockset
Wavelet Synthesis	Signal Processing Blockset
`wfilters`	Wavelet Toolbox

Wavelet Synthesis

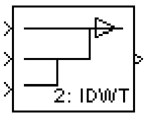
Purpose

Reconstruct signal from its multirate bandlimited components (requires the Wavelet Toolbox™ product)

Library

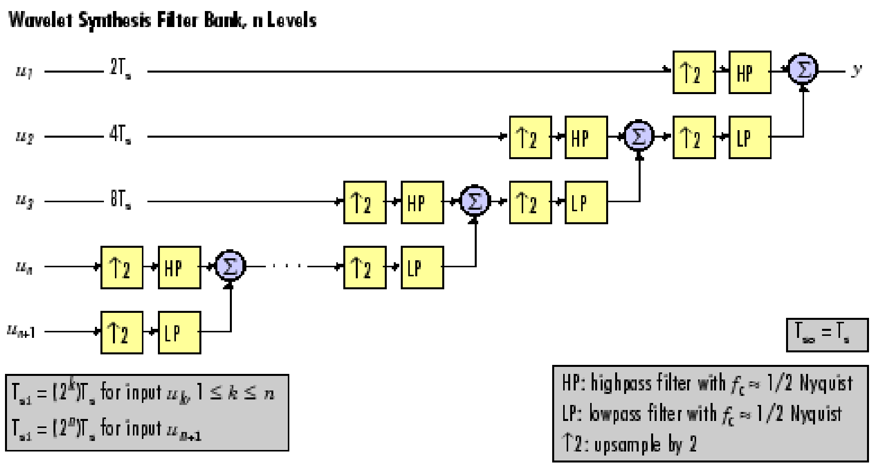
`dspobslib`

Description



Note The Wavelet Synthesis block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the IDWT block.

The Wavelet Synthesis block uses the Wavelet Toolbox `wfilters` function to reconstruct a signal that was decomposed by the Wavelet Analysis block. The reconstruction or synthesis process is the inverse of the analysis process, and restores the original signal by upsampling, filtering, and summing the bandlimited inputs in stages corresponding to the analysis process. An n-level synthesis filter bank structure is shown below, where n is specified by the **Number of levels** parameter.



At each level, the two bandlimited inputs (one low-frequency, one high-frequency, both with the same sample rate) are upsampled by a factor of 2 to match the sample rate of the input to the next stage.

They are then filtered by a highpass (HP) and lowpass (LP) filter pair with coefficients calculated to cancel (in the subsequent summation) the aliasing introduced in the corresponding analysis filter stage. The output from each (upsample-filter-sum) level has twice the bandwidth and twice the sample rate of the input to that level.

For perfect reconstruction, the Wavelet Synthesis and Wavelet Analysis blocks must have the same parameter settings.

Filter Coefficients

The filter coefficients for the highpass and lowpass filters are computed by the Wavelet Toolbox function `wfilters`, based on the wavelet specified in the **Wavelet name** parameter. The table below lists the available options.

Wavelet Name	Sample Wavelet Function Syntax
Haar	`wfilters('haar')`
Daubechies	`wfilters('db4')`
Symlets	`wfilters('sym3')`
Coiflets	`wfilters('coif1')`
Biorthogonal	`wfilters('bior3.1')`
Reverse Biorthogonal	`wfilters('rbio3.1')`
Discrete Meyer	`wfilters('dmey')`

The **Daubechies**, **Symlets**, and **Coiflets** options enable a secondary **Wavelet order** parameter that allows you to specify the wavelet order. For example, if you specify a **Daubechies** wavelet with **Wavelet order** equal to `6`, the Wavelet Synthesis block calls the `wfilters` function with input argument `'db6'`.

The **Biorthogonal** and **Reverse Biorthogonal** options enable a secondary **Filter order [synthesis / analysis]** parameter that allows you to independently specify the wavelet order for the analysis and synthesis filter stages. For example, if you specify a **Biorthogonal** wavelet with **Filter order [synthesis / analysis]** equal to `[2 / 6]`, the Wavelet Synthesis block calls the `wfilters` function with input argument `'bior2.6'`.

See the Wavelet Toolbox documentation for more information about the `wfilters` function. If you want to explicitly specify the FIR coefficients for the synthesis filter bank, use the Dyadic Synthesis Filter Bank block.

Tree Structure

The wavelet tree structure has n+1 inputs, where n is the number of levels. The sample rate and bandwidth of the output are twice the sample rate and bandwidth of the top input. The sample rate and bandwidth of each additional input (except the last) are half that of the input to the previous level.

$$T_{si,k+1} = 2T_{si,k} \quad 1 \le k < n$$

$$BW_{k+1} = \frac{BW_k}{2} \quad 1 \le k < n$$

The bottom two inputs (u_n and u_{n+1}) should have the same sample rate and bandwidth since they are processed by the same level.

$$T_{si,n+1} = T_{si,n}$$

$$BW_{n+1} = BW_n$$

Note that in frame-based mode, the sample period of input u_k is reflected by its frame size$M_{i,k}$, rather than by its frame rate.

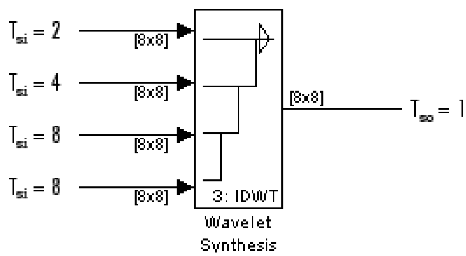
$$M_{i,k+1} = \frac{M_{i,k}}{2} \qquad 1 \le k < n$$

$$M_{i,n+1} = M_{i,n}$$

Sample-Based Operation

An M-by-N sample-based matrix input is treated as $M*N$ independent channels, and the block filters each channel independently over time. The output is the same size as the input at each port, one output channel for each input channel. As described earlier, each input port has a different sample period.

The figure below shows the input and output sample periods for the four 64-channel sample-based inputs to a three-level filter bank. The fastest input has a period of 2, so the output period is 1.

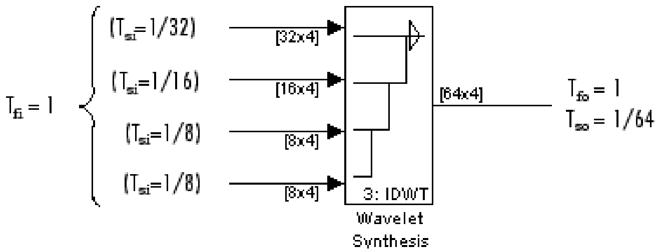


Frame-Based Operation

An M_i-by-N frame-based matrix input is treated as N independent channels, and the block filters each channel independently over time. The number of columns in the output is the same as the number of columns in the input.

All inputs must have the same frame period, which is also the output frame period. The different input sample rates should be represented by the input frame sizes: If the input to the top port has frame size M_i, the input to the second-from-top port should have frame size $M_i/2$, the input to the third-from-top port should have frame size $M_i/4$, and so on. The input to the bottom port should have the same frame size as the second-from-bottom port. The increase in the sample rate of the output is also represented by its frame size, which is twice the largest input frame size.

The relationship between sample periods, frame periods, and frame sizes is shown below for a four-channel frame-based input to a 3-level filter bank.



Zero Latency

The Wavelet Synthesis block has no tasking latency for frame-based operation, which is always single-rate. The block therefore uses the first input samples (received at t=0) to synthesize the first output sample.

Nonzero Latency

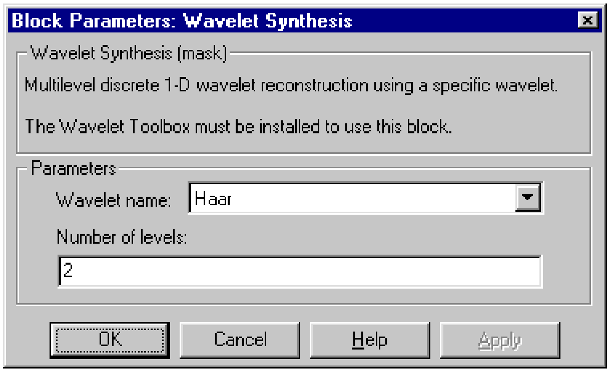
For sample-based operation, the Wavelet Synthesis block is multirate and has the following tasking latencies:

- 2^n-2 samples in Simulink®'s single-tasking mode
- 2^n samples in Simulink's multitasking mode

In the above cases, the block repeats a zero initial condition in each channel for the first D output samples, where D is the latency shown above. For example, in single-tasking mode the block generates 2^n-2 zero-valued output samples in each channel before propagating the first synthesized output sample (computed from the inputs received at t=0).

Note For more information on latency and the Simulink tasking modes, see "Excess Algorithmic Delay (Tasking Latency)" and the topic on models with multiple sample rates in the Real-Time Workshop® documentation.

Dialog Box



The parameters displayed in the dialog box vary for different wavelet types. Only some of the parameters listed below are visible in the dialog box at any one time.

Wavelet name
The wavelet used in the synthesis.

Wavelet order
The order for the **Daubechies**, **Symlets**, and **Coiflets** wavelets. This parameter is available only when one of these wavelets is selected in the **Wavelet name** menu.

Filter order [synthesis / analysis]
The filter orders for the synthesis and analysis stages of the **Biorthogonal** and **Reverse Biorthogonal** wavelets. For example, `[2 / 6]` selects a second-order synthesis stage and a sixth-order analysis stage. The **Filter order** parameter is available only when one of the above wavelets is selected in the **Wavelet name** menu.

Number of levels
The number of filter bank levels. An n-level structure has n+1 outputs.

References

Fliege, N. J. *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets*. West Sussex, England: John Wiley & Sons, 1994.

Strang, G. and T. Nguyen. *Wavelets and Filter Banks*. Wellesley, MA: Wellesley-Cambridge Press, 1996.

Vaidyanathan, P. P. *Multirate Systems and Filter Banks*. Englewood Cliffs, NJ: Prentice Hall, 1993.

Supported Data Types

- Double-precision floating point

See Also

Dyadic Synthesis Filter Bank	Signal Processing Blockset
Wavelet Analysis	Signal Processing Blockset
`wfilters`	Wavelet Toolbox

Purpose

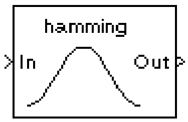
Compute and/or apply window to input signal

Library

Signal Operations

`dspsigops`

Description



The Window Function block computes a window, and/or applies a window to an input signal. This block supports real and complex floating-point and fixed-point inputs.

Operation Modes

The Window Function block has three modes of operation that you can select via the **Operation** parameter. In each mode, the block first creates a window vector w by sampling the window specified in the **Window type** parameter at M discrete points. The operation modes are

- `Apply window to input`

  In this mode, the block computes an M-by-1 window vector w and multiplies it element-wise with each of the N channels in the M-by-N input matrix u. This is equivalent to the following MATLAB® code.

  ```
  y = repmat(w,1,N) .* u        % Equivalent MATLAB code
  ```

  In this mode, a length-M 1-D vector input is treated as an M-by-1 matrix. The output y always has the same dimension as the input. When the input is frame based, the output is frame based; otherwise, the output is sample based.

- `Generate window`

  In this mode the block generates a sample-based 1-D window vector w with length M specified by the **Window length** parameter. The In port is disabled for this mode.

- `Generate and apply window`

  In this mode, the block computes an M-by-1 window vector w and multiplies it element-wise with each of the N channels in the M-by-N input matrix u. This is equivalent to the following MATLAB code.

```
y = repmat(w,1,N) .* u        % Equivalent MATLAB code
```

In this mode, a length-M 1-D vector input is treated as an M-by-1 matrix. The block produces two outputs:

- At the Out port, the block produces the result of the multiplication y, which has the same dimension as the input. When the input is frame based, the output y is frame based; otherwise, the output y is sample based.
- At the Win port, the block produces the M-by-1 window vector w. Output w is always sample based.

Window Type

The available window types are shown in the table below. For complete information about the window functions, consult the "Signal Processing Toolbox" documentation.

Window Type	Description
Bartlett	Computes a Bartlett window. `w = bartlett(M)`
Blackman	Computes a Blackman window. `w = blackman(M)`
Boxcar	Computes a rectangular window. `w = rectwin(M)`
Chebyshev	Computes a Chebyshev window with stopband ripple `R`. `w = chebwin(M,R)`

Window Type	Description
Hamming	Computes a Hamming window. `w = hamming(M)`
Hann	Computes a Hann window (also known as a Hanning window). `w = hann(M)`
Hanning	Obsolete. This window option is included only for compatibility with older models. Use the `Hann` option instead of `Hanning` whenever possible.
Kaiser	Computes a Kaiser window with Kaiser parameter `beta`. `w = kaiser(M,beta)`
Taylor	Computes a Taylor window. `w = taylorwin(M)`
Triang	Computes a triangular window. `w = triang(M)`

User Defined	Computes the user-defined window function specified by the entry in the **Window function name** parameter, `usrwin`.
	
	`w = usrwin(M) % Window takes no extra parameters`
	`w = usrwin(M,`x_1`,...,`x_n`) % Window takes extra parameters {`x_1` ... `x_n`}`

Window Sampling

For the generalized-cosine windows (`Blackman`, `Hamming`, `Hann`, and `Hanning`), the **Sampling** parameter determines whether the window samples are computed in a periodic or a symmetric manner. For example, when **Sampling** is set to `Symmetric`, a Hamming window of length M is computed as

```
w = hamming(M)     % Symmetric (aperiodic) window
```

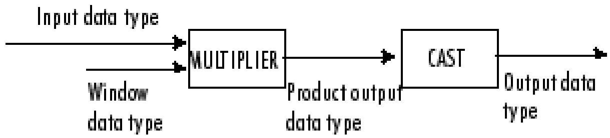
When **Sampling** is set to `Periodic`, the same window is computed as

```
w = hamming(M+1)     % Periodic (asymmetric) window
w = w(1:M)
```

Fixed-Point Data Types

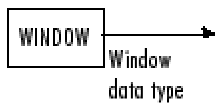
The following diagram shows the data types used within the Window block for fixed-point signals for each of the three operating modes.

Apply window to input



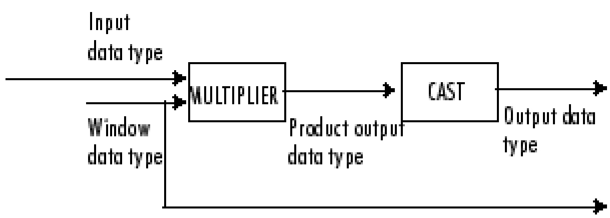
The input data type comes from the driving block. You can set the window, product output, and output data types in the block dialog. In this mode, the window vector is not output from the block.

Generate window



In this mode, the block acts as a source. The window vector is output in the window data type you specify in the block dialog.

Generate and apply window

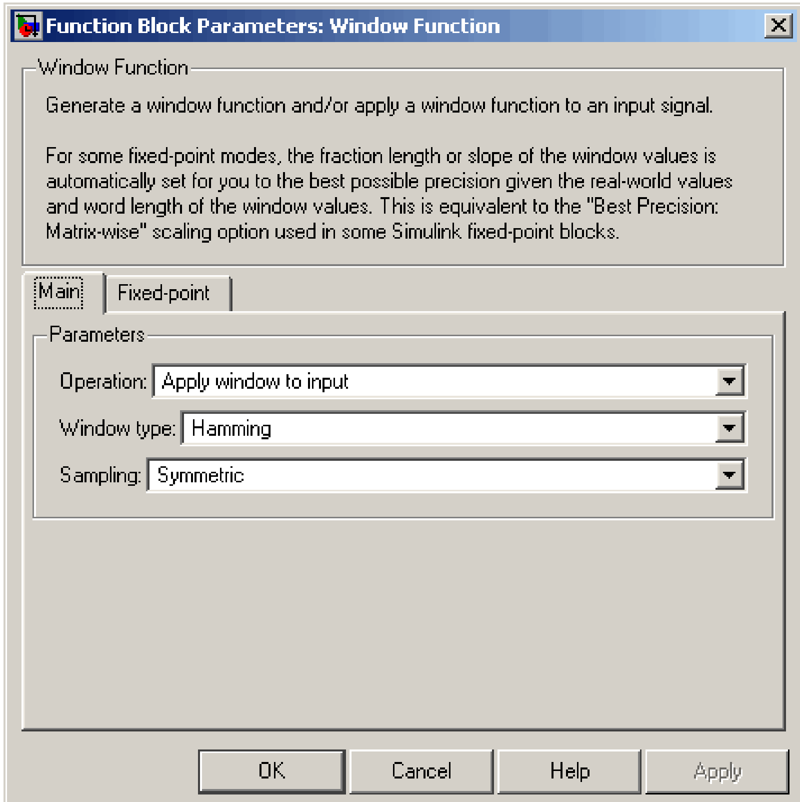


The input data type comes from the driving block. You can set the window, product output, and output data types in the block dialog. In this mode, the window vector is output from the block.

You can set the window, product output, and output data types in the block dialog as discussed below.

Dialog Box

The **Main** pane of the Window Function block dialog appears as follows.



Operation

Specify the block's operation as discussed in "Operation Modes" on page 2-1411. The port configuration of the block is updated to match the setting of this parameter.

Window type

Specify the type of window to apply as listed in "Window Type" on page 2-1412. Tunable in simulation only.

Sample Mode

Specify the sample mode for the block, `Continuous` or `Discrete`, when it is in `Generate Window` mode. In the `Apply window to output` and `Generate and apply window` modes, the block inherits the sample time from its driving block. Therefore, this parameter is only visible when you select `Generate window` for the **Operation** parameter.

Sample time

Specify the sample time for the block when it is in `Generate window` and `Discrete` modes. In `Apply window to output` and `Generate and apply window` modes, the block inherits the sample time from its driving block. This parameter is only visible when you select `Discrete` for the **Sample Mode** parameter.

Window length

Specify the length of the window to apply. This parameter is only visible when you select `Generate window` for the **Operation** parameter. Otherwise, the window vector length is computed to match the input frame size, M.

Sampling

Specify the window sampling for generalized-cosine windows. This parameter is only visible when you select `Blackman`, `Hamming`, `Hann`, or `Hanning` for the **Window type** parameter. Tunable in simulation only.

Stopband attenuation in dB

Specify the level of stopband attenuation, R_s, in decibels. This parameter is only visible when you select `Chebyshev` for the **Window type** parameter. Tunable in simulation only.

Beta

Specify the `Kaiser` window β parameter. Increasing β widens the mainlobe and decreases the amplitude of the window sidelobes in the window's frequency magnitude response. This parameter is only visible when you select `Kaiser` for the **Window type** parameter. Tunable in simulation only.

Window function name
Specify the name of the user-defined window function to be calculated by the block. This parameter is only visible when you select `User defined` for the **Window type** parameter.

Specify additional arguments to the hamming function
Select to enable the **Cell array of additional arguments** parameter, when the user-defined window requires parameters other than the window length. This parameter is only visible when you select `User defined` for the **Window type** parameter.

Cell array of additional arguments
Specify the extra parameters required by the user-defined window function, besides the window length. This parameter is only available when you select the **Specify additional arguments to the hamming function** parameter. The entry must be a cell array.

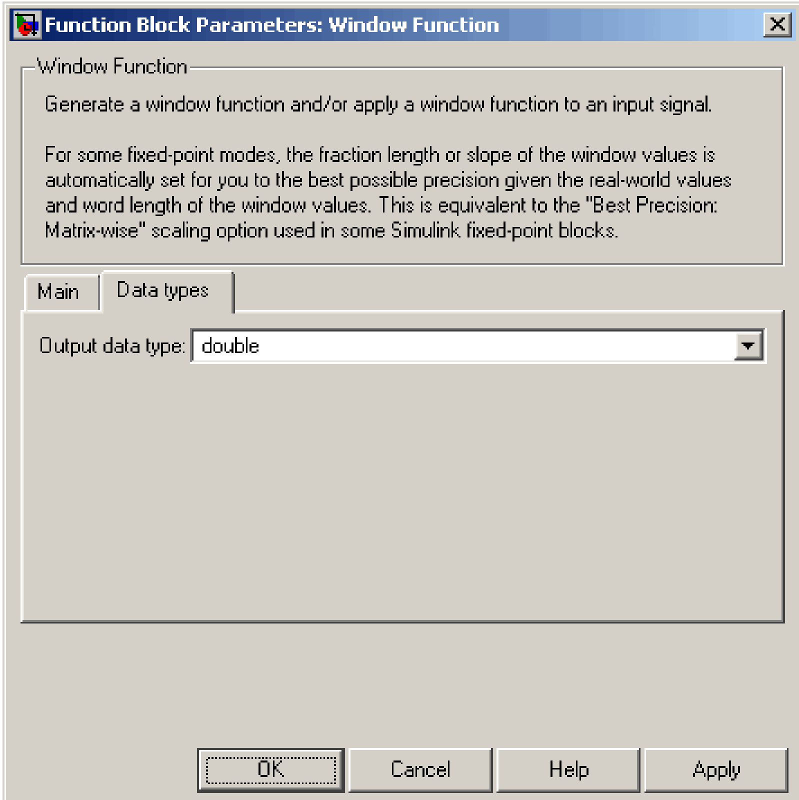
The **Data types** pane of the Window block dialog is discussed in the following sections:

"Parameters for Generate Window Only Mode" on page 2-1418

"Parameters for Apply Window Modes" on page 2-1421

Parameters for Generate Window Only Mode

The **Data types** pane of the Window Function block dialog appears as follows when the **Operation** parameter is set to `Generate window`.



Output data type

Specify the output data type in one of the following ways:

- Choose `double` or `single` from the list.
- Choose `Fixed-point` to specify the output data type and scaling in the **Signed**, **Word length**, **Set fraction length in output to,** and **Fraction length** parameters.
- Choose `User-defined` to specify the output data type and scaling in the **User-defined data type**, **Set fraction length in output to,** and **Fraction length** parameters.

- Choose `Inherit via back propagation` to set the output data type and scaling to match the following block.

Signed
Select to output a signed fixed-point signal. Otherwise, the signal is unsigned.

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

User-defined data type
Specify any built-in or fixed-point data type. You can specify fixed-point data types using the Simulink® Fixed Point™ functions `sfix`, `ufix`, `sint`, `uint`, `sfrac`, and `ufrac`. This parameter is only visible when you select `User-defined` for the **Output data type** parameter.

Set fraction length in output to
Specify the scaling of the fixed-point output by either of the following two methods:

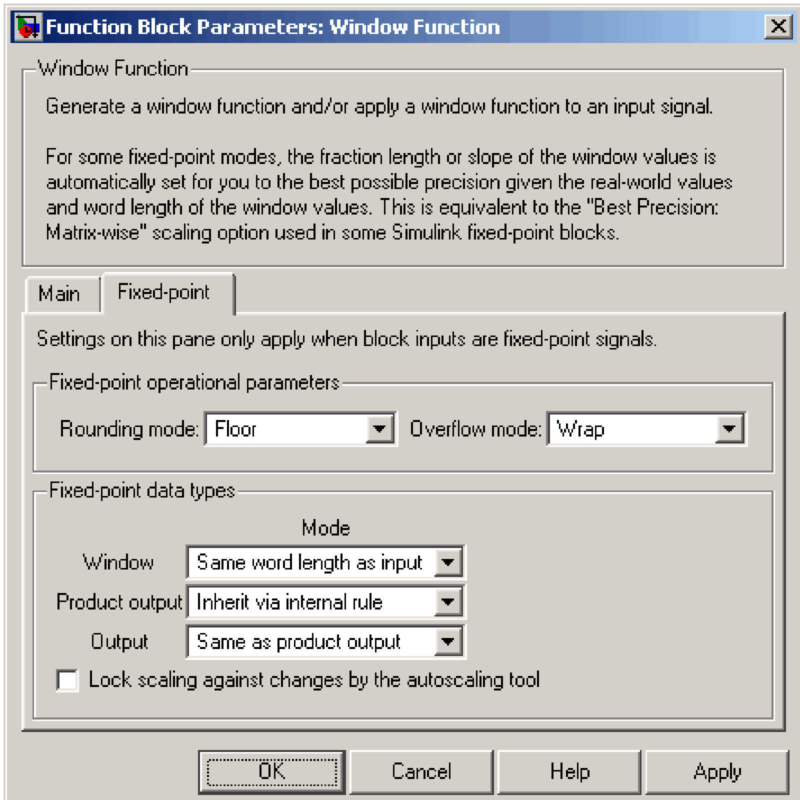
- Choose `Best precision` to have the output scaling automatically set such that the output signal has the best possible precision.
- Choose `User-defined` to specify the output scaling in the **Fraction length** parameter.

This parameter is only visible when you select `Fixed-point` or `User-defined` for the **Output data type** parameter, and when the specified output data type is a fixed-point data type.

Fraction length
Specify the fraction length, in bits, of the fixed-point output data type. This parameter is only visible when you select `Fixed-point` or `User-defined` for the **Output data type** parameter and `User-defined` for the **Set fraction length in output to** parameter.

Parameters for Apply Window Modes

The **Fixed-point** pane of the Window Function block dialog appears as follows when the **Operation** parameter is set to either `Apply window to input` or `Generate and apply window`.



Rounding mode

Select the rounding mode for fixed-point operations.

The window vector w does not obey this parameter; it always rounds to `Nearest`.

Overflow mode

Select the overflow mode for fixed-point operations.

The window vector w does not obey this parameter; it is always saturated.

Window

Choose how you specify the word length and fraction length of the window vector w.

When you select `Same word length as input`, the word length of the window vector elements is the same as the word length of the input. The fraction length is automatically set to the best precision possible.

When you select `Specify word length`, you can enter the word length of the window vector elements in bits. The fraction length is automatically set to the best precision possible.

When you select `Binary point scaling`, you can enter the word length and the fraction length of the window vector elements in bits.

When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the window vector elements. This block requires power-of-two slope and a bias of zero.

The window vector does not obey the **Rounding mode** and **Overflow mode** parameters; it is always saturated and rounded to `Nearest`.

Product output

Use this parameter to specify how you would like to designate the product output word and fraction lengths. See "Fixed-Point Data Types" on page 2-1311 for illustrations depicting the use of the product output data type in this block:

- When you select `Inherit via internal rule`, the product output word length and fraction length are calculated automatically. For information about how the product output word and fraction lengths are calculated when an internal rule is used, see "Inherit via Internal Rule".
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the product output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the product output. This block requires power-of-two slope and a bias of zero.

Output

Choose how you specify the word length and fraction length of the output of the block:

- When you select `Same as product output`, these characteristics match those of the product output.
- When you select `Same as input`, these characteristics match those of the input to the block.
- When you select `Binary point scaling`, you can enter the word length and the fraction length of the output, in bits.
- When you select `Slope and bias scaling`, you can enter the word length, in bits, and the slope of the output. This block requires power-of-two slope and a bias of zero.

Lock scaling against changes by the autoscaling tool

Select this parameter to prevent any fixed-point scaling you specify in this block mask from being overridden by the autoscaling feature of the Fixed-Point Tool. See the `fxptdlg` reference page for more information.

Supported Data Types

- Double-precision floating point
- Single-precision floating point
- Fixed point (signed only)
- 8-, 16-, and 32-bit signed integers

See Also

FFT	Signal Processing Blockset
`bartlett`	Signal Processing Toolbox
`blackman`	Signal Processing Toolbox
`rectwin`	Signal Processing Toolbox
`chebwin`	Signal Processing Toolbox
`hamming`	Signal Processing Toolbox
`hann`	Signal Processing Toolbox
`kaiser`	Signal Processing Toolbox
`taylorwin`	Signal Processing Toolbox
`triang`	Signal Processing Toolbox

Yule-Walker AR Estimator

Purpose

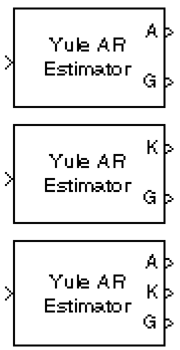
Compute estimate of autoregressive (AR) model parameters using Yule-Walker method

Library

Estimation / Parametric Estimation

`dspparest3`

Description



The Yule-Walker AR Estimator block uses the Yule-Walker AR method, also called the autocorrelation method, to fit an autoregressive (AR) model to the windowed input data by minimizing the forward prediction error in the least squares sense. This formulation leads to the Yule-Walker equations, which are solved by the Levinson-Durbin recursion. Block outputs are always nonsingular.

The Yule-Walker AR Estimator block can output the AR model coefficients as polynomial coefficients, reflection coefficients, or both. Each channel of the input is a sample-based vector (row, column, or 1-D) or frame-based vector (column only) representing a frame of consecutive time samples from a signal that is assumed to be the output of an AR system driven by white noise. The block computes the normalized estimate of the AR system parameters, $A(z)$, independently for each successive input frame.

$$H(z) = \frac{\sqrt{G}}{A(z)} = \frac{\sqrt{G}}{1 + a(2)z^{-1} + \ldots + a(p+1)z^{-p}}$$

When you select **Inherit estimation order from input dimensions**, the order p of the all-pole model is one less than the length of each input channel. Otherwise, the order is the value specified by the **Estimation order** parameter. To guarantee a valid output, you must set the **Estimation order** parameter to be a scalar less than or equal to half the input channel length. The Yule-Walker AR Estimator and Burg AR Estimator blocks return similar results for large frame sizes.

When **Output(s)** is set to `A`, port A is enabled. For each channel, port A outputs a column of length p+1 that contains the normalized estimate of the AR model coefficients in descending powers of z

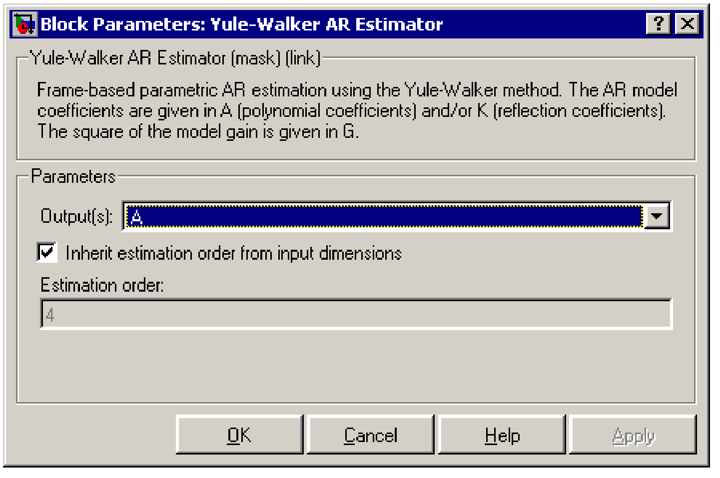
```
[1 a(2) ... a(p+1)]
```

When **Output(s)** is set to `K`, port K is enabled. For each channel, port K outputs a length-*p* column whose elements are the AR model reflection coefficients. When **Output(s)** is set to `A and K`, both port A and K are enabled, and each port outputs the respective AR model coefficients for each channel.

The square of the model gain, *G*, is provided at port G. *G* is a scalar for each channel.

See the Burg AR Estimator block reference page for a comparison of the Burg AR Estimator, Covariance AR Estimator, Modified Covariance AR Estimator, and Yule-Walker AR Estimator blocks.

Dialog Box



Output(s)

The type of AR model coefficients output by the block. The block can output polynomial coefficients (`A`), reflection coefficients (`K`), or both (`A and K`).

Inherit estimation order from input dimensions

When selected, sets the estimation order *p* to one less than the length of each input channel.

Estimation order

The order of the AR model, *p*. This parameter is enabled when you do not select **Inherit estimation order from input dimensions**.

References

Kay, S. M. *Modern Spectral Estimation: Theory and Application.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

Marple, S. L., Jr., *Digital Spectral Analysis with Applications.* Englewood Cliffs, NJ: Prentice-Hall, 1987.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
A	• Double-precision floating point • Single-precision floating point
K	• Double-precision floating point • Single-precision floating point
G	• Double-precision floating point • Single-precision floating point

See Also

Burg AR Estimator	Signal Processing Blockset
Covariance AR Estimator	Signal Processing Blockset
Modified Covariance AR Estimator	Signal Processing Blockset

Yule-Walker Method	Signal Processing Blockset
`aryule`	Signal Processing Toolbox

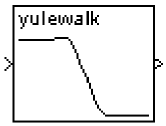
Purpose

Design and apply IIR filter

Library

`dspobslib`

Description



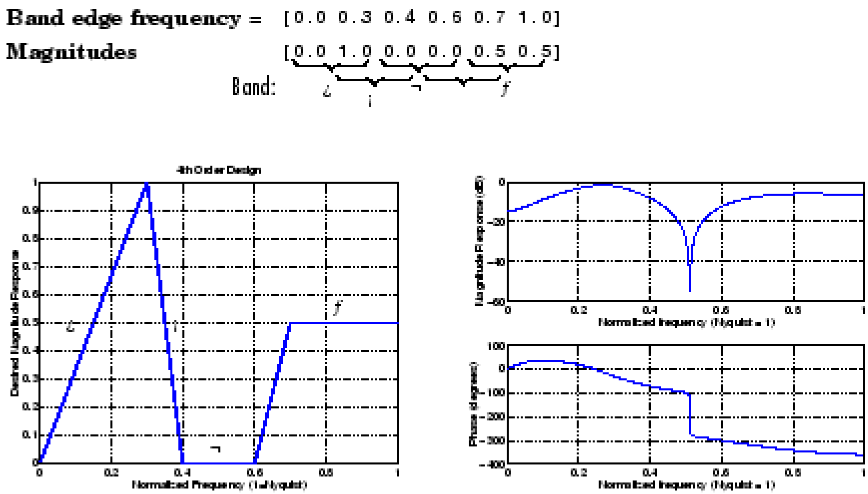
Note The Yule-Walker IIR Filter Design block is still supported but is likely to be obsoleted in a future release. We strongly recommend replacing this block with the Digital Filter block.

The Yule-Walker IIR Filter Design block designs a recursive (ARMA) digital filter with arbitrary multiband magnitude response, and applies it to a discrete-time input using the Direct-Form II Transpose Filter block. The filter design, which uses the Signal Processing Toolbox™ `yulewalk` function, performs a least-squares fit to the specified frequency response.

An *M*-by-*N* sample-based matrix input is treated as *M*N* independent channels, and an *M*-by-*N* frame-based matrix input is treated as *N* independent channels. In both cases, the block filters each channel independently over time, and the output has the same size and frame status as the input.

The **Band-edge frequency vector** parameter is a vector of frequency points in the range `0` to `1`, where `1` corresponds to half the sample frequency. The first element of this vector must be `0` and the last element `1`, and intermediate points must appear in ascending order. The **Magnitudes at these frequencies** parameter is a vector containing the desired magnitude response at the points specified in the **Band-edge frequency vector**.

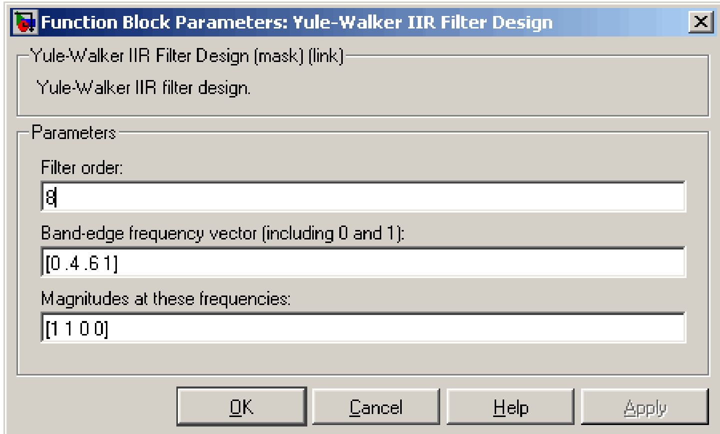
Note that, unlike the Remez FIR Filter Design block, each frequency-magnitude pair specifies the junction of two adjacent frequency bands, so there are no “don’t care” regions.



When specifying the **Band-edge frequency vector** and **Magnitudes at these frequencies** vectors, avoid excessively sharp transitions from passband to stopband. You may need to experiment with the slope of the transition region to get the best filter design.

For more details on the Yule-Walker filter design algorithm, see the description of the `yulewalk` function in the Signal Processing Toolbox documentation.

Dialog Box



Filter order

The order of the filter.

Band-edge frequency vector

A vector of frequency points. The value 1 corresponds to half the sample frequency. The first element of this vector must be 0 and the last element 1. Tunable.

Magnitudes at these frequencies

A vector of frequency response magnitudes corresponding to the points in the **Band-edge frequency vector**. This vector must be the same length as the **Band-edge frequency vector**. Tunable.

References

Oppenheim, A. V. and R. W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

Proakis, J. and D. Manolakis. *Digital Signal Processing.* 3rd ed. Englewood Cliffs, NJ: Prentice-Hall, 1996.

Yule-Walker Method

Purpose

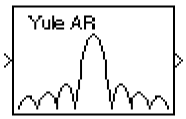
Compute parametric estimate of tspectrum using Yule-Walker autoregressive (AR) method

Library

Estimation / Power Spectrum Estimation

`dspspect3`

Description



The Yule-Walker Method block estimates the power spectral density (PSD) of the input using the Yule-Walker AR method. This method, also called the autocorrelation method, fits an autoregressive (AR) model to the windowed input data by minimizing the forward prediction error in the least squares sense. This formulation leads to the Yule-Walker equations, which are solved by Levinson-Durbin recursion. Block outputs are always nonsingular.

The input is a sample-based vector (row, column, or 1-D) or frame-based vector (column only) representing a frame of consecutive time samples from a single-channel signal. The block's output (a column vector) is the estimate of the signal's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where F_s is the signal's sample frequency.

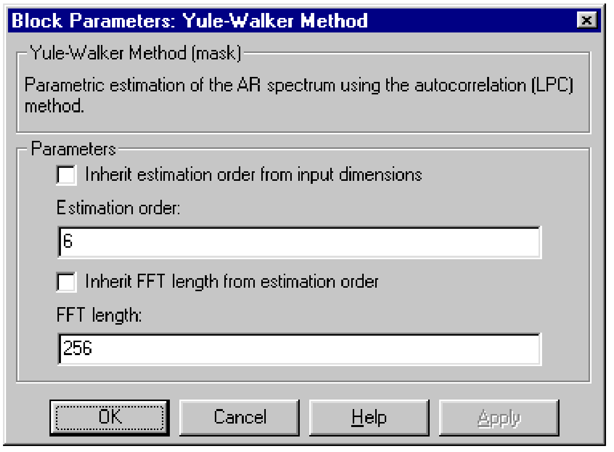
When you select **Inherit estimation order from input dimensions**, the order of the all-pole model is one less that the input frame size. Otherwise, the order is the value specified by the **Estimation order** parameter. To guarantee a valid output, you must set the **Estimation order** parameter to be less than or equal to half the input vector length. The spectrum is computed from the FFT of the estimated AR model parameters.

When you select **Inherit FFT length from estimation order**, N_{fft} is specified by (estimation order + 1), which must be a power of 2. When you do not select **Inherit FFT length from estimation order**, N_{fft} is specified as a power of 2 by the **FFT length** parameter, and the block zero pads or wraps the input to N_{fft} before computing the FFT. The output is always sample based.

See the Burg Method block reference for a comparison of the Burg Method, Covariance Method, Modified Covariance Method, and

Yule-Walker AR Estimator blocks. The Yule-Walker AR Estimator and Burg Method blocks return similar results for large buffer lengths.

Dialog Box



Inherit estimation order from input dimensions
: When selected, sets the estimation order to one less than the length of the input vector.

Estimation order
: The order of the AR model. This parameter is enabled when you do not select **Inherit estimation order from input dimensions**.

Inherit FFT length from estimation order
: When selected, uses the estimation order to determine the number of data points, $\mathrm{N}_{\mathrm{fft}}$, on which to perform the FFT. Sets $\mathrm{N}_{\mathrm{fft}}$ equal to (estimation order + 1). Note that $\mathrm{N}_{\mathrm{fft}}$ must be a power of 2, so (estimation order + 1) must be a power of 2.

FFT length
: Enter the number of data points on which to perform the FFT, N_{fft}. When N_{fft} is larger than the input frame size, each frame is zero-padded as needed. When N_{fft} is smaller than the input frame size, each frame is wrapped as needed. This parameter is

enabled when you clear the **Inherit FFT length from input dimensions** check box.

References

Kay, S. M. *Modern Spectral Estimation: Theory and Application.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

Marple, S. L., Jr., *Digital Spectral Analysis with Applications.* Englewood Cliffs, NJ: Prentice-Hall, 1987.

Orfanidis, S. J. *Introduction to Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1995.

Supported Data Types

Port	Supported Data Types
Input	• Double-precision floating point • Single-precision floating point
Output	• Double-precision floating point • Single-precision floating point

The output data type is the same as the input data type.

See Also

Burg Method	Signal Processing Blockset
Covariance Method	Signal Processing Blockset
Levinson-Durbin	Signal Processing Blockset
Autocorrelation LPC	Signal Processing Blockset
Short-Time FFT	Signal Processing Blockset
Yule-Walker AR Estimator	Signal Processing Blockset
`pyulear`	Signal Processing Toolbox

See "Power Spectrum Estimation" for related information.

Purpose

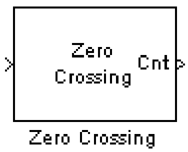
Count number of times signal crosses zero in single time step

Library

Signal Operations

`dspsigops`

Description



The Zero Crossing block concludes that a signal in a given channel has passed through zero if it meets any of the following criteria, where x_i is the current signal value, x_{i-1} is the previous signal value, and so on:

- $x_i < 0$ and $x_{i-1} > 0$
- $x_i > 0$ and $x_{i-1} < 0$
- For some positive integer L, $x_i < 0$, $x_{i-l} = 0$, and $x_{i-L-1} > 0$, where $0 \le l \le L$.
- For some positive integer L, $x_i > 0$, $x_{i-l} = 0$, and $x_{i-L-1} < 0$, where $0 \le l \le L$.

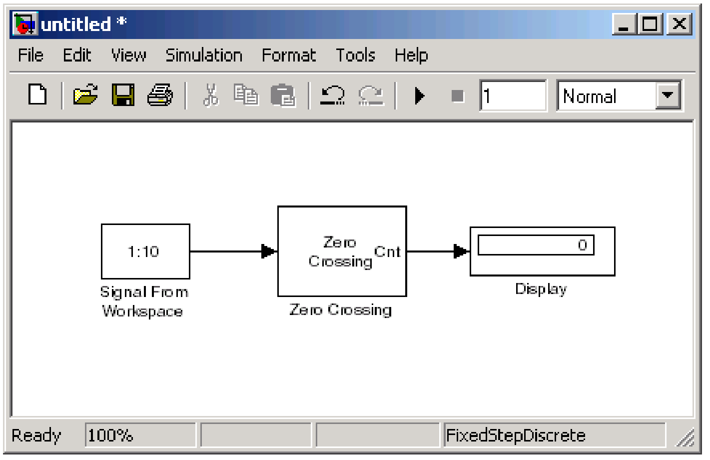
For the first input value, x_{i-1} and x_{i-2} are zero. The block outputs the number of times the signal crosses zero in a single time step at the Cnt port.

The input to this block must be a real-valued fixed-point or floating-point signal. If the input is a sample-based vector or matrix, then each entry is treated as a time-varying channel. If the input is a frame-based row vector of length N, it is treated as N independent channels. If the input is a frame-based column vector, it is treated as a single channel.

Examples

The following example illustrates the behavior of the Zero Crossing block.

1 Create the following Simulink® model.



2 Use the Signal From Workspace block to create a frame-based signal. Set the parameters as follows:

- **Signal** = `[-3:3]'`
- **Sample time** = `1/7`
- **Samples per frame** = `7`
- **Form output after final data value by** = `Cyclic repetition`

The block outputs a single frame of the frame-based signal at the first time step, and identical frames at each additional time step.

3 Use the Zero Crossing block to detect the number of zero crossing in each time step. Use the default parameters.

4 Use the Display block to view the number of zero crossings.

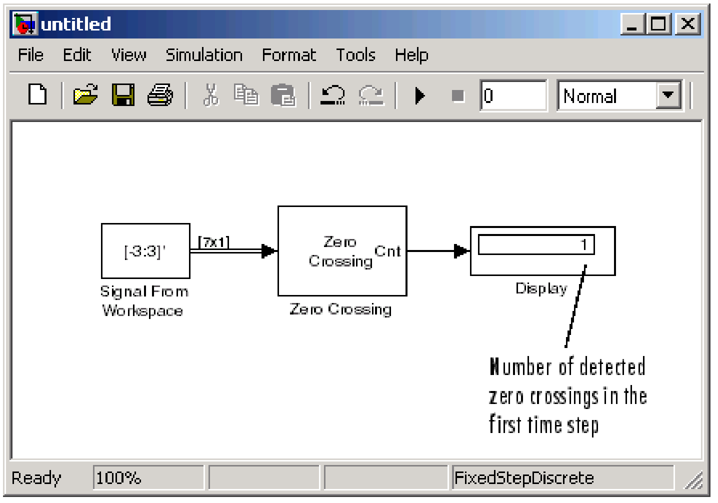
5 To run the model for one time step, set the configuration parameters. Open the Configuration Parameters dialog box by selecting **Configuration Parameters** from the **Simulation** menu. In the **Solver** pane, set the parameters as follows:

- **Stop time** = `0`

- **Type** = `Fixed-step`
- **Solver** = `discrete (no continuous states)`

6 Run the model.

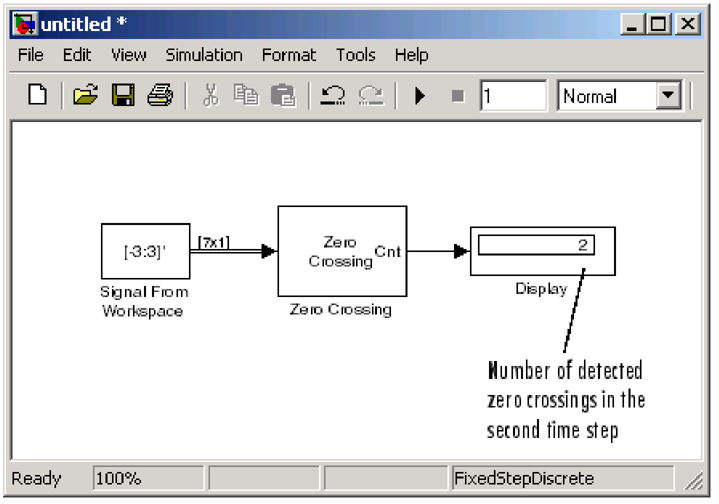
Because the signal passes through zero once during the first time step, the Zero Crossing block finds one zero crossing as shown in the figure below.



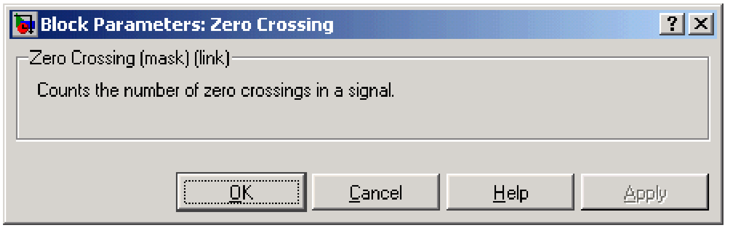
7 To run the model for two time steps, change the simulation **Stop time** to `1`.

8 Run the model.

The Zero Crossing block remembers that the last value of the last frame was 3. Therefore, the signal passes through zero twice during the second time step. It passes through zero while going from 3 to -3, and it passes through zero again while going from -3 to 3. The Zero Crossing block finds two zero crossings in the second time step as shown in the figure below.



Dialog Box



Supported Data Types

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

See Also

Hit Crossing Simulink

3

Function Reference

dsp_links

Purpose Library link information for Signal Processing Blockset™ blocks

Syntax

```
dsp_links
dsp_links()
```

Description The `dsp_links` function displays and returns library link information for blocks linked to the Signal Processing Blockset libraries.

Signal Processing Blockset blocks can be obsolete, deprecated, or current. Obsolete blocks are blocks that are no longer supported. They might or might not work properly. Deprecated blocks are still supported but are likely to become obsolete in a future release. Current blocks are supported and represent the latest block functionality.

`dsp_links()` returns a structure with three elements for the current model. Each element represents one of the three block categories and contains a cell array of strings. Each string is the name of a library block in the current model.

`dsp_links(sys)` returns a structure with three elements for the named system.

See Also

`liblinks`	Signal Processing Blockset

Purpose Open main Signal Processing Blockset™ library

Syntax `dsplib`

Description `dsplib` opens the current version of the main Signal Processing Blockset library.

dspstartup

Purpose

Configure Simulink® environment for signal processing systems

Syntax

```
dspstartup
```

Description

`dspstartup` configures a number of Simulink environment parameters with settings appropriate for a typical signal processing project. When the Simulink environment has been successfully configured, the function displays the following message in the command window.

```
Changed default Simulink settings for signal processing
systems (dspstartup.m).
```

To automatically configure the Simulink environment at startup, add a call to `dspstartup.m` from your `startup.m` file. If you do not have a `startup.m` file on your path, you can create one from the `startupsav.m` template in the `toolbox/local` directory.

To edit `startupsav.m`, simply replace the `load matlab.mat` command with a call to `dspstartup.m`, and save the file as `startup.m`. The result should look like this.

```
%STARTUP Startup file
%   This file is executed when MATLAB starts up,
%   if it exists anywhere on the path.
dspstartup;
```

For more information, see the description for the `startup` command in the MATLAB® documentation and Configuring Simulink for Signal Processing Models in the *Signal Processing Blockset™ Getting Started Guide*.

The `dspstartup.m` script executes the following commands. See "Model and Block Parameters" in the Simulink documentation for complete information about a particular setting.

```
set_param(0, ...
         'SingleTaskRateTransMsg','error', ...
         'multiTaskRateTransMsg', 'error', ...
```

```
                 'Solver',                'fixedstepdiscrete', ...
                 'SolverMode',            'SingleTasking', ...
                 'StartTime',             '0.0', ...
                 'StopTime',              'inf', ...
                 'FixedStep',             'auto', ...
                 'SaveTime',              'off', ...
                 'SaveOutput',            'off', ...
                 'AlgebraicLoopMsg',      'error', ...
                 'SignalLogging',         'off');
       set_param(getActiveConfigSet(0), 'RollThreshold', 2);
```

See Also

`startup` MATLAB

liblinks

Purpose Library link information for Signal Processing Blockset™ blocks

Description Please see the command line help for `liblinks`. Type

```
help liblinks
```

in the MATLAB® Command Window.

See Also

`dsp_links`	Signal Processing Blockset

Purpose

Compute number of samples of delay introduced by buffering and unbuffering operations

Syntax

```
d = rebuffer_delay(f,n,m)
d = rebuffer_delay(f,n,m,'singletasking')
```

Description

`d = rebuffer_delay(f,n,m)` returns the delay (in samples) introduced by the buffering and unbuffering blocks in multitasking operations, where `f` is the input frame size, `n` is the **Output buffer size** parameter setting, and `m` is the **Buffer overlap** parameter setting.

The blocks whose delay can be computed by `rebuffer_delay` are

- Buffer
- Unbuffer

`d = rebuffer_delay(f,n,m,'singletasking')` returns the delay (in samples) introduced by these blocks in single-tasking operations.

The table below shows the appropriate `rebuffer_delay` parameter values to use in computing delay for the two blocks.

Block	Parameter Values
Buffer	`f` = input frame size (`f=1` for sample-based mode) `n` = **Output buffer size** `m` = **Buffer overlap**

Unbuffer	`f` = input frame size
	`n = 1`
	`m = 0`

See Also

Buffer	Signal Processing Blockset
Unbuffer	Signal Processing Blockset

Glossary

This glossary defines terms related to fixed-point data types and numbers. These terms may appear in some or all of the documents that describe products from The MathWorks™ that have fixed-point support.

arithmetic shift

Shift of the bits of a binary word for which the sign bit is recycled for each bit shift to the right. A zero is incorporated into the least significant bit of the word for each bit shift to the left. In the absence of overflows, each arithmetic shift to the right is equivalent to a division by 2, and each arithmetic shift to the left is equivalent to a multiplication by 2.

See also binary point, binary word, bit, logical shift, most significant bit

bias

Part of the numerical representation used to interpret a fixed-point number. Along with the slope, the bias forms the scaling of the number. Fixed-point numbers can be represented as

$$\textit{real-world value} = (\textit{slope} \times \textit{stored integer}) + \textit{bias}$$

where the slope can be expressed as

$$\textit{slope} = \textit{fractional slope} \times 2^{\textit{exponent}}$$

See also fixed-point representation, fractional slope, integer, scaling, slope, [Slope Bias]

binary number

Value represented in a system of numbers that has two as its base and that uses 1’s and 0’s (bits) for its notation.

See also bit

binary point

Symbol in the shape of a period that separates the integer and fractional parts of a binary number. Bits to the left of the binary point are integer bits and/or sign bits, and bits to the right of the binary point are fractional bits.

See also binary number, bit, fraction, integer, radix point

binary point-only scaling

Scaling of a binary number that results from shifting the binary point of the number right or left, and which therefore can only occur by powers of two.

See also binary number, binary point, scaling

binary word

Fixed-length sequence of bits (1's and 0's). In digital hardware, numbers are stored in binary words. The way in which hardware components or software functions interpret this sequence of 1's and 0's is described by a data type.

See also bit, data type, word

bit

Smallest unit of information in computer software or hardware. A bit can have the value 0 or 1.

ceiling (round toward)

Rounding mode that rounds to the closest representable number in the direction of positive infinity. This is equivalent to the Fixed-Point Toolbox™ `ceil` mode.

See also convergent rounding, floor (round toward), nearest (round toward), rounding, truncation, zero (round toward)

contiguous binary point

Binary point that occurs within the word length of a data type. For example, if a data type has four bits, its contiguous binary point must be understood to occur at one of the following five positions:

.0000

0.000

00.00

000.0

0000.

See also data type, noncontiguous binary point, word length

convergent rounding

Rounding mode that rounds to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to `0`.

See also ceiling (round toward), floor (round toward), nearest (round toward), rounding, truncation, zero (round toward)

data type

Set of characteristics that define a group of values. A fixed-point data type is defined by its word length, its fraction length, and whether it is signed or unsigned. A floating-point data type is defined by its word length and whether it is signed or unsigned.

See also fixed-point representation, floating-point representation, fraction length, word length

data type override

Parameter in the Fixed-Point Tool that allows you to set the output data type and scaling of fixed-point blocks on a system or subsystem level.

See also data type, scaling

exponent

Part of the numerical representation used to express a floating-point or fixed-point number.

1. Floating-point numbers are typically represented as

$$real\text{-}world\ value = mantissa \times 2^{exponent}$$

2. Fixed-point numbers can be represented as

$$real\text{-}world\ value = (slope \times stored\ integer) + bias$$

where the slope can be expressed as

$$slope = fractional\ slope \times 2^{exponent}$$

The exponent of a fixed-point number is equal to the negative of the fraction length:

$$exponent = -1 \times fraction\ length$$

See also bias, fixed-point representation, floating-point representation, fraction length, fractional slope, integer, mantissa, slope

fixed-point representation

Method for representing numerical values and data types that have a set range and precision.

1. Fixed-point numbers can be represented as

$$real\text{-}world\ value = (slope \times stored\ integer) + bias$$

where the slope can be expressed as

$$slope = fractional\ slope \times 2^{exponent}$$

The slope and the bias together represent the scaling of the fixed-point number.

2. Fixed-point data types can be defined by their word length, their fraction length, and whether they are signed or unsigned.

See also bias, data type, exponent, fraction length, fractional slope, integer, precision, range, scaling, slope, word length

floating-point representation

Method for representing numerical values and data types that can have changing range and precision.

1. Floating-point numbers can be represented as

$$real\text{-}world\ value = mantissa \times 2^{exponent}$$

2. Floating-point data types are defined by their word length.

See also data type, exponent, mantissa, precision, range, word length

floor (round toward)

Rounding mode that rounds to the closest representable number in the direction of negative infinity.

See also ceiling (round toward), convergent rounding, nearest (round toward), rounding, truncation, zero (round toward)

fraction

Part of a fixed-point number represented by the bits to the right of the binary point. The fraction represents numbers that are less than one.

See also binary point, bit, fixed-point representation

fraction length

Number of bits to the right of the binary point in a fixed-point representation of a number.

See also binary point, bit, fixed-point representation, fraction

fractional slope

Part of the numerical representation used to express a fixed-point number. Fixed-point numbers can be represented as

$$\textit{real-world value} = (\textit{slope} \times \textit{stored integer}) + \textit{bias}$$

where the slope can be expressed as

$$\textit{slope} = \textit{fractional slope} \times 2^{\textit{exponent}}$$

The term *slope adjustment* is sometimes used as a synonym for fractional slope.

See also bias, exponent, fixed-point representation, integer, slope

guard bits

Extra bits in either a hardware register or software simulation that are added to the high end of a binary word to ensure that no information is lost in case of overflow.

See also binary word, bit, overflow

integer

1. Part of a fixed-point number represented by the bits to the left of the binary point. The integer represents numbers that are greater than or equal to one.

2. Also called the "stored integer." The raw binary number, in which the binary point is assumed to be at the far right of the word. The integer is part of the numerical representation used to express a fixed-point number. Fixed-point numbers can be represented as

$$\textit{real-world value} = 2^{-\textit{fraction length}} \times \textit{stored integer}$$

or

$$\textit{real-world value} = (\textit{slope} \times \textit{stored integer}) + \textit{bias}$$

where the slope can be expressed as

$$slope = fractional\ slope \times 2^{exponent}$$

See also bias, fixed-point representation, fractional slope, integer, real-world value, slope

integer length

Number of bits to the left of the binary point in a fixed-point representation of a number.

See also binary point, bit, fixed-point representation, fraction length, integer

least significant bit (LSB)

Bit in a binary word that can represent the smallest value. The LSB is the rightmost bit in a big-endian-ordered binary word. The weight of the LSB is related to the fraction length according to

$$weight\ of\ LSB = 2^{-fraction\ length}$$

See also big-endian, binary word, bit, most significant bit

logical shift

Shift of the bits of a binary word, for which a zero is incorporated into the most significant bit for each bit shift to the right and into the least significant bit for each bit shift to the left.

See also arithmetic shift, binary point, binary word, bit, most significant bit

mantissa

Part of the numerical representation used to express a floating-point number. Floating-point numbers are typically represented as

$$real\text{-}world\ value = mantissa \times 2^{exponent}$$

See also exponent, floating-point representation

most significant bit (MSB)

Bit in a binary word that can represent the largest value. The MSB is the leftmost bit in a big-endian-ordered binary word.

See also binary word, bit, least significant bit

nearest (round toward)

Rounding mode that rounds to the closest representable number, with the exact midpoint rounded to the closest representable number in the direction of positive infinity. This is equivalent to the Fixed-Point Toolbox™ `nearest` mode.

See also ceiling (round toward), convergent rounding, floor (round toward), rounding, truncation, zero (round toward)

noncontiguous binary point

Binary point that is understood to fall outside the word length of a data type. For example, the binary point for the following 4-bit word is understood to occur two bits to the right of the word length,

0000__.

thereby giving the bits of the word the following potential values:

$2^5 2^4 2^3 2^2$__.

See also binary point, data type, word length

one's complement representation

Representation of signed fixed-point numbers. Negating a binary number in one's complement requires a bitwise complement. That is, all 0's are flipped to 1's and all 1's are flipped to 0's. In one's complement notation there are two ways to represent zero. A binary word of all 0's represents "positive" zero, while a binary word of all 1's represents "negative" zero.

See also binary number, binary word, sign/magnitude representation, signed fixed-point, two's complement representation

overflow

Situation that occurs when the magnitude of a calculation result is too large for the range of the data type being used. In many cases you can choose to either saturate or wrap overflows.

See also saturation, wrapping

padding

Extending the least significant bit of a binary word with one or more zeros.

See also least significant bit

precision

1. Measure of the smallest numerical interval that a fixed-point data type and scaling can represent, determined by the value of the number's least significant bit. The precision is given by the slope, or the number of fractional bits. The term *resolution* is sometimes used as a synonym for this definition.

2. Measure of the difference between a real-world numerical value and the value of its quantized representation. This is sometimes called quantization error or quantization noise.

See also data type, fraction, least significant bit, quantization, quantization error, range, slope

Q format

Representation used by Texas Instruments to encode signed two's complement fixed-point data types. This fixed-point notation takes the form

$$Qm.n$$

where

- Q indicates that the number is in Q format.
- m is the number of bits used to designate the two's complement integer part of the number.
- n is the number of bits used to designate the two's complement fractional part of the number, or the number of bits to the right of the binary point.

In Q format notation, the most significant bit is assumed to be the sign bit.

See also binary point, bit, data type, fixed-point representation, fraction, integer, two's complement

quantization

Representation of a value by a data type that has too few bits to represent it exactly.

See also bit, data type, quantization error

quantization error

Error introduced when a value is represented by a data type that has too few bits to represent it exactly, or when a value is converted from one data type to a shorter data type. Quantization error is also called quantization noise.

See also bit, data type, quantization

radix point

Symbol in the shape of a period that separates the integer and fractional parts of a number in any base system. Bits to the left of the radix point are integer and/or sign bits, and bits to the right of the radix point are fraction bits.

See also binary point, bit, fraction, integer, sign bit

range

Span of numbers that a certain data type can represent.

See also data type, precision

real-world value

Stored integer value with fixed-point scaling applied. Fixed-point numbers can be represented as

$$real\text{-}world\ value = 2^{-fraction\ length} \times stored\ integer$$

or

$$real\text{-}world\ value = (slope \times stored\ integer) + bias$$

where the slope can be expressed as

$$slope = fractional\ slope \times 2^{exponent}$$

See also integer

resolution

See **precision**

rounding

Limiting the number of bits required to express a number. One or more least significant bits are dropped, resulting in a loss of precision. Rounding is necessary when a value cannot be expressed exactly by the number of bits designated to represent it.

See also bit, ceiling (round toward), convergent rounding, floor (round toward), least significant bit, nearest (round toward), precision, truncation, zero (round toward)

saturation

Method of handling numeric overflow that represents positive overflows as the largest positive number in the range of the data type being used, and negative overflows as the largest negative number in the range.

See also overflow, wrapping

scaled double

A double data type that retains fixed-point scaling information. For example, in Simulink® and Fixed-Point Toolbox™ software you can use data type override to convert your fixed-point data types to scaled doubles. You can then simulate to determine the ideal floating-point behavior of your system. After you gather that information you can turn data type override off to return to fixed-point data types, and your quantities still have their original scaling information because it was held in the scaled double data types.

scaling

1. Format used for a fixed-point number of a given word length and signedness. The slope and bias together form the scaling of a fixed-point number.

2. Changing the slope and/or bias of a fixed-point number without changing the stored integer.

See also bias, fixed-point representation, integer, slope

shift

Movement of the bits of a binary word either toward the most significant bit ("to the left") or toward the least significant bit ("to the right"). Shifts to the right can be either logical, where the spaces emptied at the front of the word with each shift are filled in with zeros, or arithmetic, where the word is sign extended as it is shifted to the right.

See also arithmetic shift, logical shift, sign extension

sign bit

Bit (or bits) in a signed binary number that indicates whether the number is positive or negative.

See also binary number, bit

sign extension

Addition of bits that have the value of the most significant bit to the high end of a two's complement number. Sign extension does not change the value of the binary number.

See also binary number, guard bits, most significant bit, two's complement representation, word

sign/magnitude representation

Representation of signed fixed-point or floating-point numbers. In sign/magnitude representation, one bit of a binary word is always the dedicated sign bit, while the remaining bits of the word encode the magnitude of the number. Negation using sign/magnitude representation consists of flipping the sign bit from 0 (positive) to 1 (negative), or from 1 to 0.

See also binary word, bit, fixed-point representation, floating-point representation, one's complement representation, sign bit, signed fixed-point, two's complement representation

signed fixed-point

Fixed-point number or data type that can represent both positive and negative numbers.

See also data type, fixed-point representation, unsigned fixed-point

slope

Part of the numerical representation used to express a fixed-point number. Along with the bias, the slope forms the scaling of a fixed-point number. Fixed-point numbers can be represented as

$$\textit{real-world value} = (\textit{slope} \times \textit{stored integer}) + \textit{bias}$$

where the slope can be expressed as

$$\textit{slope} = \textit{fractional slope} \times 2^{\textit{exponent}}$$

See also bias, fixed-point representation, fractional slope, integer, scaling, [Slope Bias]

slope adjustment

See **fractional slope**

[Slope Bias]

Representation used to define the scaling of a fixed-point number.

See also bias, scaling, slope

stored integer

See **integer**

trivial scaling

Scaling that results in the real-world value of a number being simply equal to its stored integer value:

$$\textit{real-world value} = \textit{stored integer}$$

In [Slope Bias] representation, fixed-point numbers can be represented as

$$\textit{real-world value} = (\textit{slope} \times \textit{stored integer}) + \textit{bias}$$

In the trivial case, slope = 1 and bias = 0.

In terms of binary point-only scaling, the binary point is to the right of the least significant bit for trivial scaling, meaning that the fraction length is zero:

$$\textit{real-world value} = \textit{stored integer} \times 2^{-\textit{fraction length}} = \textit{stored integer} \times 2^{0}$$

Scaling is always trivial for pure integers, such as `int8`, and also for the true floating-point types `single` and `double`.

See also bias, binary point, binary point-only scaling, fixed-point representation, fraction length, integer, least significant bit, scaling, slope, [Slope Bias]

truncation

Rounding mode that drops one or more least significant bits from a number.

See also ceiling (round toward), convergent rounding, floor (round toward), nearest (round toward), rounding, zero (round toward)

two's complement representation

Common representation of signed fixed-point numbers. Negation using signed two's complement representation consists of a translation into one's complement followed by the binary addition of a one.

See also binary word, one's complement representation, sign/magnitude representation, signed fixed-point

unsigned fixed-point

Fixed-point number or data type that can only represent numbers greater than or equal to zero.

See also data type, fixed-point representation, signed fixed-point

word

Fixed-length sequence of binary digits (1's and 0's). In digital hardware, numbers are stored in words. The way hardware components or software functions interpret this sequence of 1's and 0's is described by a data type.

See also binary word, data type

word length

Number of bits in a binary word or data type.

See also binary word, bit, data type

wrapping

Method of handling overflow. Wrapping uses modulo arithmetic to cast a number that falls outside of the representable range the data type being used back into the representable range.

See also data type, overflow, range, saturation

zero (round toward)

Rounding mode that rounds to the closest representable number in the direction of zero. This is equivalent to the Fixed-Point Toolbox™ `fix` mode.

See also ceiling (round toward), convergent rounding, floor (round toward), nearest (round toward), rounding, truncation

Index

A

acquiring data
  blocks for 1-16
adaptive filter designs
  FIR 2-712
  Kalman 2-657
  LMS 2-712
  RLS 2-1026
addition
  cumulative 2-237
Analog Filter Design block 2-2
analytic signal 2-7
Analytic Signal block 2-7
Arbitrary Magnitude Filter block 2-9
Array-Vector Add block 2-11
Array-Vector Divide block 2-19
Array-Vector Multiply block 2-26
Array-Vector Subtract block 2-35
audio
  From Audio Device block 2-546
  From Wave Device block 2-562
  From Wave File block 2-568
  To Audio Device block 2-1183
  To Wave Device block 2-1195
  To Wave File block 2-1204
autocorrelation
  and Levinson-Durbin recursion 2-697
  of a real vector 2-43
  sequence 2-1432
Autocorrelation block 2-43
Autocorrelation LPC block 2-53
autocorrelation method 2-1425
autoregressive models, using
  Burg AR Estimator block 2-91
  Burg Method block 2-96
  Covariance AR Estimator block 2-215
  Covariance Method block 2-218
  Modified Covariance AR Estimator
    block 2-862
  Modified Covariance Method block 2-865
  Yule-Walker AR Estimator block 2-1425
  Yule-Walker Method block 2-1432

B

Backward Substitution block 2-57
Bandpass Filter block 2-65
bandpass filter designs
  using Analog Filter Design block 2-2
Bandstop Filter block 2-67
bandstop filter designs
  using Analog Filter Design block 2-2
Bartlett windows 2-1412
binary clock signals 2-868
bins
  histogram 2-584
bit-reversed order 2-449
Blackman windows 2-1412
Block LMS Filter block 2-73
Buffer block 2-81
buffering
  blocks for 1-12
  Buffer block 2-81
  Delay Line block 2-279
  first input, first output (FIFO) register 2-983
  last input, first output (LIFO)
    register 2-1133
  Queue block 2-983
  Stack block 2-1133
Burg AR Estimator block 2-91
Burg Method block 2-96
  power spectrum estimation 2-96
Butterworth filter designs
  using Analog Filter Design block 2-2

C

changing
  size of frames 2-81
Chebyshev type I filter designs

using Analog Filter Design block 2-2
Chebyshev type II filter designs
using Analog Filter Design block 2-2
Chebyshev windows 2-1412
Check Signal Attributes block 2-101
Chirp block 2-108
Cholesky Factorization block 2-133
Cholesky Inverse block 2-137
Cholesky Solver block 2-140
CIC Compensator block 2-143
CIC Decimation block 2-145
CIC Filter block 2-154
CIC Interpolation block 2-156
clocks
binary 2-868
multiphase 2-868
complex analytic signal 2-7
Complex Cepstrum block 2-166
Complex Exponential block 2-168
complex exponentials 2-168
computing
frequency distributions 2-584
histograms 2-584
Constant Diagonal Matrix block 2-169
Constant Ramp block 2-173
constants
matrix 2-169
ramp 2-173
control signals, for
Triggered Shift Register block 2-1210
Triggered Signal From Workspace block 2-1214
Triggered Signal To Workspace block 2-1218
controller canonical forms 2-3
Convert 1-D to 2-D block 2-180
Convert 2-D to 1-D block 2-183
convolution
of two real vectors 2-185
Convolution block 2-185
correlation
of two real vectors 2-194
Correlation block 2-194
correlation matrices 2-658
Counter block 2-203
Covariance AR Estimator block 2-215
Covariance Method block 2-218
spectral analysis 2-218
Create Diagonal Matrix block 2-221
Cumulative Product block 2-223
Cumulative Sum block 2-237

D

Data Type Conversion block 2-249
dB
converting to 2-250
dB Conversion block 2-250
dB Gain block 2-253
dBm
converting to 2-250
DC component of an analytic signal 2-7
DCT block 2-256
DCTs
computing 2-256
decimation
process of 2-472
using FIR Decimation block 2-472
using FIR Rate Conversion block 2-510
delay
fractional 2-1282
`rebuffer_delay` function 3-7
Delay block 2-264
Delay Line block 2-279
Detrend block 2-285
diagonal matrix constants 2-169
difference
between elements in a vector 2-287
Difference block 2-287
Differentiator Filter block 2-294
Digital Filter block 2-296

Digital Filter Design block 2-357
discrete cosine transforms 2-256
Discrete Impulse block 2-381
discrete wavelet transform 2-402
Display block 2-387
display span
    Vector Scope Block 2-1351
displaying
    blocks for 1-15
    frame-based data 2-1351
    matrices as images 2-801
Downsample block 2-388
downsampling
    Downsample block 2-388
    FIR Decimation block 2-472
    FIR Rate Conversion block 2-510
DSP Constant block 2-397
`dsplib` function 3-3
`dspstartup` M-file 3-4
DWT block 2-402
Dyadic Analysis Filter Bank block 2-404
Dyadic Synthesis Filter Bank block 2-416

E

Edge Detector block 2-427
elements of a vector
    selecting 2-1299
elliptic filter designs
    using Analog Filter Design block 2-2
estimation
    blocks for 1-2
    Burg AR Estimator block 2-91
    Burg Method block 2-96
    Covariance AR Estimator block 2-215
    Covariance Method block 2-218
    Modified Covariance AR Estimator block 2-862
    Modified Covariance Method block 2-865
    nonparametric with Magnitude FFT block 2-772
    nonparametric with Short-Time FFT block 2-1080
    Yule-Walker AR Estimator block 2-1425
    Yule-Walker Method block 2-1432
Event-Count Comparator block 2-430
events, triggering for
    N-Sample Enable block 2-876
    N-Sample Switch block 2-881
    Sample and Hold block 2-1041
    Stack block 2-1134
    Triggered Shift Register block 2-1210
    Triggered Signal From Workspace block 2-1214
    Triggered Signal To Workspace block 2-1218
exponentials
    complex 2-168
exporting
    blocks for 1-15
    using Triggered Signal To Workspace block 2-1218
Extract Diagonal block 2-433
Extract Triangular Matrix block 2-435

F

Fast Block LMS Filter block 2-439
fast Fourier transform (FFT) 2-446
FDATool
    in Signal Processing Blockset 2-357
FFT block 2-446
FFTs
    computing 2-446
    overlap-add filtering 2-917
    overlap-save filtering 2-921
filter designs
    Digital Filter block 2-296
    Digital Filter Design block 2-357
    Levinson-Durbin block 2-698

using Analog Filter Design block 2-2
Filter Realization Wizard 2-461
filter realizations
using Filter Realization Wizard 2-461
filters
blocks for 1-4
overlap-add method 2-917
overlap-save method 2-921
FIR
interpolation 2-641
FIR Decimation block 2-472
FIR filter designs
using Levinson-Durbin block 2-698
with prescribed autocorrelation sequence 2-698
FIR Interpolation block 2-490
FIR Rate Conversion block 2-510
first-input, first-output (FIFO) registers 2-983
Flip block 2-527
forms
controller canonical 2-3
state-space 2-3
Forward Substitution block 2-530
Fractional Delay Filter block 2-538
Frame Conversion 2-540
Frame Status Conversion block 2-543
frames
changing size of 2-81
unbuffering to scalars 2-1252
frequency distributions 2-584
computing 2-584
From Audio Device block 2-546
From Multimedia Fiel block 2-553
From Wave Device block 2-562
From Wave File block 2-568
functions, utility
`dsplib` 3-3
`dspstartup` 3-4
`rebuffer_delay` 3-7

G

G.711 Codec block 2-573
gain
applying in dB 2-253

H

Halfband Filter block 2-578
Hamming windows 2-1413
Hann windows 2-1413
Highpass Filter block 2-580
highpass filter designs
using Analog Filter Design block 2-2
Hilbert Filter block 2-582
Hilbert transformer filter designs 2-7
Histogram block 2-584
histograms
computing 2-584

I

IDCT block 2-598
IDCTs 2-598
computing 2-598
identity matrices 2-606
Identity Matrix block 2-606
IDWT block 2-611
IFFT block 2-613
IFFTs
computing 2-613
images
displaying matrices as 2-801
importing
blocks for 1-16
scalars 2-568
signals from the workspace 2-1085
Triggered Signal From Workspace block 2-1214
vectors 2-568
indexing

blocks for 1-13
Inherit Complexity block 2-628
Integer Delay block 2-631
interpolating
FIR Interpolation block 2-490
FIR Rate Conversion block 2-510
procedure 2-490
Interpolation block 2-641
inverse discrete cosine transforms 2-598
inverse fast Fourier transform (FFT) 2-613
Inverse Short-Time FFT block 2-651
Inverse Sinc Filter block 2-655

K

Kaiser windows 2-1413
Kalman Adaptive Filter block 2-657
Kalman Filter block 2-662

L

last-input, first-output (LIFO) registers 2-1133
LDL Factorization block 2-672
LDL Inverse block 2-681
LDL Solver block 2-684
least mean-square algorithm 2-712
Least Squares Polynomial Fit block 2-693
Levinson-Durbin block 2-696
libraries
displaying link information 3-2
linear algebra
blocks for 1-7
linear prediction
using Autocorrelation LPC block 2-53
LMS Adaptive Filter block 2-707
LMS algorithm
Block LMS Filter block 2-73
Fast Block LMS Filter block 2-439
LMS Filter block 2-712
LMS Filter block 2-712
Lowpass Filter block 2-729
lowpass filter designs
using Analog Filter Design block 2-2
LPC to LSP/LSF Conversion block 2-731
LPC to/from Cepstral Coefficients block 2-748
LPC to/from RC block 2-753
LPC/RC to Autocorrelation block 2-758
LSF/LSP to LPC Conversion block 2-746
LU Factorization block 2-761
LU Inverse block 2-768
LU Solver block 2-770

M

M-files
`dspstartup` 3-4
Magnitude FFT block 2-772
magnitude response of filters 2-2
magnitudes
converting to dB 2-250
matrices
constant diagonal 2-169
create diagonal 2-221
displaying as images 2-801
extracting diagonal of 2-433
extracting triangle from 2-435
identity 2-169
Identity Matrix block 2-606
multiplying 2-784
multiplying within 2-785
normalizing 2-777
overwriting elements of 2-925
permuting 2-965
selecting elements from 2-1156
summing 2-794
Toeplitz 2-1179
transposing 2-1207
Matrix 1-Norm block 2-777
Matrix Concatenate block 2-781
Matrix Exponential block 2-782

Matrix Multiply block 2-784
matrix operations
  blocks for 1-7
Matrix Product block 2-785
Matrix Square block 2-792
Matrix Sum block 2-794
Matrix Viewer block 2-801
Maximum block 2-808
Mean block 2-823
Median block 2-839
Minimum block 2-847
minimum mean-square estimate (MMSE) 2-657
Modified Covariance AR Estimator block 2-862
Modified Covariance Method block 2-865
modifying signal attributes
  blocks for 1-13
Multiphase Clock block 2-868
multiplication
  cumulative 2-223
multiplying
  by dB gain 2-253
Multiport Selector block 2-872

N

N-Sample Enable block 2-876
N-Sample Switch block 2-881
n-step forward linear predictors 2-53
NCO block 2-886
Normalization block 2-902
Numerically Controlled Oscillator block 2-886
Nyquist Filter block 2-909

O

Octave Filter block 2-911
Offset block 2-913
ones
  outputting 2-876
Overlap-Add FFT Filter block 2-917
overlap-add method 2-917
Overlap-Save FFT Filter block 2-921
overlap-save method 2-921
Overwrite Values block 2-925

P

Pad block 2-943
Parametric Equalizer block 2-950
Peak Finder block 2-952
Peak-Notch Filter block 2-959
Periodogram block 2-961
periodograms 2-772
Permute Matrix block 2-965
phase angles
  unwrapping 2-1265
phase unwrap 2-1265
Polynomial Evaluation block 2-970
Polynomial Stability Test block 2-972
polyphase filter structures
  FIR Decimation 2-472
  FIR Interpolation block 2-490
power spectrum estimation, using
  Burg method 2-96
  short-time, fast Fourier transform (ST-FFT) 2-1080
  Yule-Walker AR method 2-1432
prediction
  linear 2-53
predictor algorithm 2-657
Pseudoinverse block 2-974

Q

QR Factorization block 2-977
QR Solver block 2-980
quantization
  blocks for 1-11
Quantizer block 2-982
Queue block 2-983

R

ramp signal 2-173
random signals 2-994
Random Source block 2-994
random-walk Kalman filter 2-658
Real Cepstrum block 2-1004
`rebuffer_delay` function 3-7
rebuffering
  `rebuffer_delay` function 3-7
  with the Buffer block 2-81
Reciprocal Condition block 2-1006
rectangular windows 2-1412
recursive least-squares (RLS) algorithm 2-1026
registers
  first-input, first-output (FIFO) 2-983
  last-input, first-output (LIFO) 2-1133
Remez exchange algorithm 2-7
Repeat block 2-1014
resampling
  by inserting zeros 2-1274
  Downsample block 2-388
  FIR Decimation block 2-472
  FIR Interpolation block 2-490
  FIR Rate Conversion block 2-510
  procedure 2-510
  Repeat block 2-1014
RLS Adaptive Filter block 2-1021
RLS Filter block 2-1026
RMS block 2-1033
root-mean-square (RMS)
  computing 2-1033

S

Sample and Hold block 2-1041
sample rates
  changing 2-388
Scalar Quantizer block 2-1044
Scalar Quantizer Decoder block 2-1054
Scalar Quantizer Design block 2-1060
Scalar Quantizer Encoder block 2-1069
scalars
  converting to vectors 2-279
  creating from vectors 2-1252
  exporting 2-1218
  importing 2-568
  importing from the workspace 2-1085
selecting
  elements of a vector 2-1299
Selector block 2-1079
Shift Register block
  initial state of 2-282
Short-Time FFT block 2-1080
short-time, fast Fourier transform (ST-FFT)
  method 2-1080
Signal From Workspace block 2-1085
  compared to Simulink To Workspace
    block 2-1085
signal operations
  blocks for 1-14
Signal To Workspace block 2-1091
signals
  control 2-1210
  random 2-994
  Triggered Signal From Workspace
    block 2-1214
Sine Wave block 2-1099
Singular Value Decomposition block 2-1111
Sort block 2-1114
sound
  From Audio Device block 2-546
  From Wave Device block 2-562
  From Wave File block 2-568
  To Audio Device block 2-1183
  To Wave Device block 2-1195
  To Wave File block 2-1204
spectral analysis 2-96
  Burg method 2-96
  covariance method 2-218
  magnitude FFT method 2-772

modified covariance method 2-865
short-time FFT method 2-1080
Yule-Walker method 2-1432
See also power spectrum estimation
Spectrum Scope block 2-1120
ST-FFT method 2-1080
Stack block 2-1133
stack events 2-1134
standard deviation
computing 2-1143
Standard Deviation block 2-1143
state-space forms 2-3
statistical operations
blocks for 1-17
statistics
RMS 2-1033
standard deviation 2-1143
variance 2-1304
Submatrix block 2-1156
SVD Solver block 2-1166
swept cosine 2-116
swept-frequency cosine 2-108
switching
between two inputs 2-881

T

Taylor windows 2-1413
Time Scope block 2-1169
To Audio Device block 2-1183
To Multimedia File block 2-1190
To Wave Device block 2-1195
To Wave File block 2-1204
Toeplitz block 2-1179
transforms
blocks for 1-18
discrete cosine 2-256
discrete wavelet 2-402
Fourier 2-446
inverse Fourier 2-613
Transpose block 2-1207
transposing
matrices 2-1207
trends
removing 2-285
triangular windows 2-1413
Triggered Delay Line block 2-1210
Triggered Shift Register block
initial state of 2-1211
Triggered Signal From Workspace block 2-1214
Triggered To Workspace block 2-1218
triggering, for
N-Sample Enable block 2-876
N-Sample Switch block 2-881
Sample and Hold block 2-1041
Triggered Shift Register block 2-1210
Triggered Signal From Workspace block 2-1214
Triggered Signal To Workspace block 2-1218
Two-Channel Analysis Subband Filter block 2-1221
Two-Channel Synthesis Subband Filter block 2-1236

U

Unbuffer block 2-1252
initial state of 2-1253
unbuffering
with the Buffer block 2-81
Uniform Decoder block 2-1256
Uniform Encoder block 2-1260
Unwrap block 2-1265
unwrapping radian phase angles 2-1265
Upsample block 2-1274
upsampling
by inserting zeros 2-1274
FIR Interpolation block 2-490
FIR Rate Conversion block 2-510
Repeat block 2-1014

utility functions
  dsplib 3-3
  dspstartup 3-4
  rebuffer_delay 3-7

V

Variable Fractional Delay block 2-1282
  initial conditions for 2-1283
Variable Integer Delay block 2-1289
  initial conditions for 2-1291
Variable Selector block 2-1299
variance 2-1304
  tracking 2-1304
Variance block 2-1304
Vector Quantizer Decoder block 2-1322
Vector Quantizer Design 2-1329
Vector Quantizer Encoder block 2-1340
Vector Scope block 2-1351
vectors
  converting to scalars 2-1252
  creating from scalars 2-1210
  exporting 2-1218
  importing 2-568
  importing from the workspace 2-1085

W

Waterfall block 2-1374
Window Function block 2-1411
windows
  applying 2-1411
  Bartlett 2-1412
  Blackman 2-1412
  Chebyshev 2-1412
  Hamming 2-1413
  Hann 2-1413
  Kaiser 2-1413
  rectangular 2-1412
  Taylor 2-1413
  triangular 2-1413

Y

Yule-Walker AR Estimator block 2-1425
Yule-Walker Method block 2-1432

Z

Zero Crossing block 2-1435
zero-padding
  Pad block 2-943
zeros
  Counter block 2-207
  Discrete Impulse block 2-381
  inserting 2-490
  N-Sample Enable block 2-876
  Signal From Workspace block 2-1085
  Triggered Signal From Workspace block 2-1215