DSP Blockset

For Use with SIMULINK®

Modeling

Simulation

Implementation

The MATH WORKS Inc.

User's Guide

Version 3

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DSP Blockset User's Guide

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1

Introduction

Welcome to the DSP Blockset

Welcome to the DSP Blockset, the premier tool for digital signal processing (DSP) algorithm simulation and code generation.

The DSP Blockset brings the full power of Simulink® to DSP system design and prototyping by providing key DSP algorithms and components in Simulink's adaptable block format. From buffers to linear algebra solvers, from dyadic filter banks to parametric estimators, the blockset gives you all the core components to rapidly and efficiently assemble complex DSP systems.

Use the DSP Blockset and Simulink to develop your DSP concepts, and to efficiently revise and test until the design is production-ready. Use the DSP Blockset together with the Real-Time Workshop® (RTW) to automatically generate code for real-time execution on DSP hardware.

Above all, we hope you enjoy using the DSP Blockset, and we look forward to hearing your comments and suggestions.

`support@mathworks.com`	Technical support
`suggest@mathworks.com`	Product enhancement suggestions
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What Is the DSP Blockset?

The DSP Blockset is a collection of block libraries for use with the Simulink dynamic system simulation environment.

The DSP Blockset libraries are designed specifically for digital signal processing (DSP) applications, and include key operations such as classical, multirate, and adaptive filtering, matrix manipulation and linear algebra, statistics, time-frequency transforms, and more.

Key Features

The DSP Blockset extends the Simulink environment by providing core components and algorithms for DSP systems. You can use blocks from the DSP Blockset in the same way that you would use any other Simulink blocks, combining them with blocks from other libraries to create sophisticated DSP systems.

A few of the important features are:

- Frame-based operations
- Matrix support
- Adaptive and multirate filtering
- Statistical operations
- Linear algebra
- Parametric estimation
- Real-time code generation capability

Frame-Based Operations

Most real-time DSP systems optimize throughput rates by processing data in "batch" or "frame-based" mode, where each batch or frame is a collection of consecutive signal samples that have been buffered into a single unit. By propagating these multisample frames instead of the individual signal samples, the DSP system can best take advantage of the speed of DSP algorithm execution, while simultaneously reducing the demands placed on the data acquisition (DAQ) hardware.

The DSP Blockset delivers this same high level of performance for both simulation and code generation by incorporating frame-processing capability

into all of its blocks. A completely frame-based model can run several times faster than the same model processing sample-by-sample; even faster if the data source is frame based.

See "Understanding Sample Rates" and "Understanding Samples and Frames" in Chapter 2 for complete information.

Matrix Support

The DSP Blockset supports two-dimensional matrices. Typical uses of the matrix format are:

- *General*

  A matrix can be used in its traditional mathematical capacity, as a simple structured array of numbers. The matrix values might represent the pixel brightnesses from a charge-coupled device (CCD) camera, a collection of measurements from several tests, or any other group of values. Most blocks for general matrix operations are found in the Matrix Functions and Linear Algebra libraries.

- *To store factored submatrices*

  A number of the matrix factorization blocks in the Linear Algebra library store the submatrix factors (i.e., lower and upper submatrices) in a single compound matrix.

- *To represent multichannel frame-based data*

  The standard format for multichannel frame-based data is a matrix containing each channel's data in a separate column. A matrix with three columns, for example, contains three channels of data, one frame per channel. The number of rows in such a matrix is the number of samples in each frame.

See "Understanding Matrices" in Chapter 2 for complete information.

Adaptive and Multirate Filtering

The Adaptive Filters and Multirate Filters libraries provide key tools for the construction of advanced DSP systems. Adaptive filter blocks are parameterized to support the rapid tailoring of DSP algorithms to application-specific environments, and effortless "what if" experimentation. The multirate filtering algorithms employ polyphase implementations for efficient simulation and real-time code execution.

Statistical Operations

Use the blocks in the Statistics library for basic statistical analysis. These blocks calculate measures of central tendency and spread (e.g., mean, standard deviation, and so on), as well as the frequency distribution of input values (histogram).

Linear Algebra

The Linear Algebra library provides a wide variety of matrix factorization methods, and equation solvers based on these methods. The popular Cholesky, LU, LDL, and QR factorizations are all available.

Parametric Estimation

The Parametric Estimation library provides a number of methods for modeling a signal as the output of an AR system. The methods include the Burg AR Estimator, Covariance AR Estimator, Modified Covariance AR Estimator, and Yule-Walker AR Estimator, which allow you to compute the AR system parameters based on forward error minimization, backward error minimization, or both.

Real-Time Code Generation

You can also use the separate Real-Time Workshop product to generate optimized, compact, C code for models containing blocks from the DSP Blockset.

About the DSP Blockset Libraries

Library Terms. The blocks in the DSP Blockset are organized into several libraries, whose contents and structure are shown on the following pages. The Blockset hierarchy has two levels of libraries. At the top level, some libraries (such as DSP Sources) contain DSP blocks, while others (such as Filtering) contain a second level of DSP libraries. This book uses the term “library” to refer to libraries at both levels in the hierarchy. So, for example, the Filtering library contains the Filter Designs library.

True Libraries. The libraries in the DSP Blockset are *true* libraries. This means that when a block is copied from a DSP Blockset library to the model window, a reference to the *source* block (in the library) is created. Any updates made to

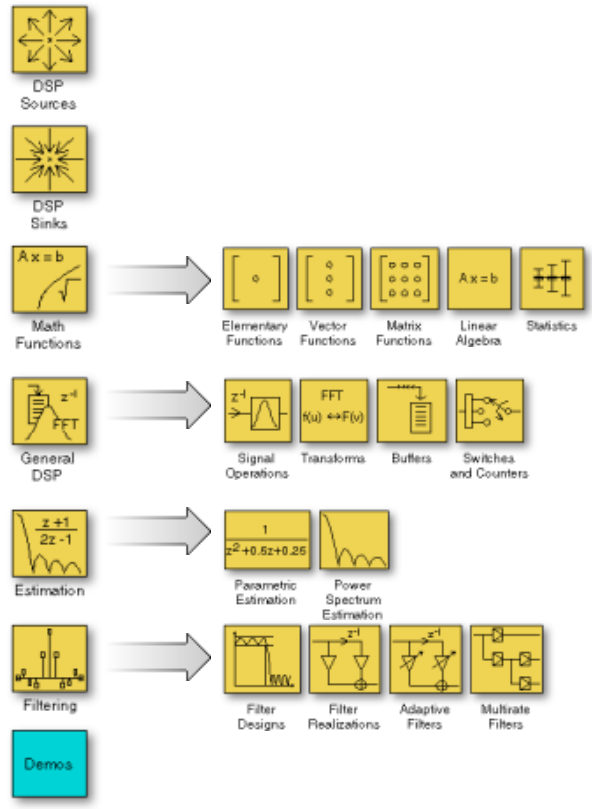
the source block will affect every model block that refers to it. For complete information about true libraries, see *Using Simulink*.

What Is in the DSP Blockset?

The organization of the DSP Blockset is shown in the figure below. Type

```
dsplib
```

in the command window to open the blockset. Each of the libraries in the DSP Blockset contains a collection of related DSP blocks or a group of more specialized libraries. Double-click on any library to display its contents.



DSP Sources Library

The DSP Sources library provides blocks for acquiring data from devices, files, and the workspace, and for generating signals.

DSP Sources	
Chirp	Matrix From Workspace
Constant Diagonal Matrix	N-Sample Enable
Discrete Constant	Signal From Workspace
From Wave Device	Triggered Signal From Workspace
From Wave File	Sine Wave
Matrix Constant	

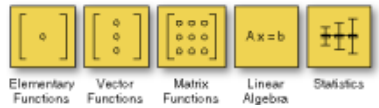
DSP Sinks Library

The DSP Sinks library provides blocks for sending data to devices, files, and the workspace, and for displaying signals on the screen in the time or frequency domain.

DSP Sinks	
Buffered FFT Frame Scope	Time Frame Scope
FFT Frame Scope	To Wave Device
Frequency Frame Scope	To Wave File
Matrix To Workspace	Triggered Matrix To Workspace
Matrix Viewer	Triggered Signal To Workspace
Signal To Workspace	User-Defined Frame Scope

Math Functions Library

The Math Functions library contains five sublibraries covering a wide variety of mathematical operations.



Elementary Functions Library

The Elementary Functions library provides an assortment of general conversion and utility blocks.

Elementary Functions	
Complex Exponential	dB
Contiguous Copy	dB Gain
Convert Complex DSP To Simulink	Inherit Complexity
Convert Complex Simulink To DSP	Variable Selector

Vector Functions Library

The Vector Functions library provides blocks for vector operations such as correlation and convolution.

Vector Functions	
Autocorrelation	Difference
Convolution	Flip
Correlation	Normalization
Cumulative Sum	Unwrap

Matrix Functions Library

The Matrix Functions library provides a variety of matrix-oriented operations, such as transposition, permutation, matrix arithmetic, and scaling.

Matrix Functions	
Constant Diagonal Matrix	Matrix Scaling
Create Diagonal Matrix	Matrix Sum
Extract Diagonal	Matrix To Workspace
Extract Triangular Matrix	Permute Matrix
Matrix 1-Norm	Reshape
Matrix Constant	Submatrix
Matrix From Workspace	Toeplitz

Matrix Functions (Continued)	
Matrix Multiplication	Transpose
Matrix Product	

Linear Algebra Library

The Linear Algebra library provides blocks for matrix factorization and linear equation solution.

Linear Algebra	
Backward Substitution	Levinson Solver
Cholesky Factorization	LU Factorization
Cholesky Solver	LU Solver
Forward Substitution	QR Factorization
LDL Factorization	QR Solver
LDL Solver	Reciprocal Condition

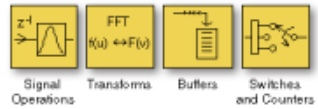
Statistics Library

The Statistics library provides blocks for sorting data, and for computing static and cumulative statistical measures of central tendency, spread, and frequency distribution.

Statistics	
Histogram	RMS
Maximum	Sort
Mean	Standard Deviation
Median	Variance
Minimum	

General DSP Library

The General DSP library contains four sublibraries covering a wide variety of signal processing operations.



Signal Operations Library

The Signal Operations library provides blocks for common signal processing operations such as windowing, resampling, zero padding, and delay.

Signal Operations	
Analytic Signal	Upsample
Detrend	Variable Fractional Delay
Downsample	Variable Integer Delay
Integer Delay	Window Function
LPC	Zero Pad
Repeat	

Transforms Library

The Transforms library provides blocks for FFT, DCT, and cepstrum operations.

Transforms	
Complex Cepstrum	IDCT
DCT	IFFT
FFT	Real Cepstrum

Buffers Library

The Buffers library provides blocks for rebuffering signals to different frame sizes, and for FIFO/LIFO storage.

Buffers	
Buffer	Shift Register
Partial Unbuffer	Stack
Queue	Triggered Shift Register
Rebuffer	Unbuffer

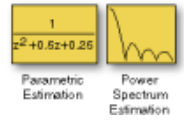
Switches and Counters Library

The Switches and Counters library provides blocks for switching signal sources, and for detecting and counting signal transitions.

Switches and Counters	
Commutator	Multiphase Clock
Counter	N-Sample Enable
Distributor	N-Sample Switch
Edge Detector	Sample and Hold
Event-Count Comparator	

Estimation Library

The Estimation library contains two sublibraries providing blocks for parametric estimation and power spectrum estimation.



Parametric Estimation Library

The Parametric Estimation library provides blocks for estimating the parameters of the autoregressive system generating the input.

Parametric Estimation	
Burg AR Estimator	Modified Covariance AR Estimator
Covariance AR Estimator	Yule-Walker AR Estimator

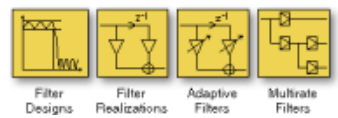
Power Spectrum Estimation Library

The Power Spectrum Estimation library provides blocks for estimating the power spectrum of the input by several different methods.

Power Spectrum Estimation	
Burg Method	Modified Covariance Method
Covariance Method	Short-Time FFT
Magnitude FFT	Yule-Walker Method

Filtering Library

The Filtering library contains four sublibraries providing blocks for a wide range of filtering applications.



Filter Designs Library

The Filter Designs library provides blocks for a variety of analog and digital FIR and IIR filter designs. Included are Butterworth, Chebyshev, elliptic,

differentiator, and Hilbert designs, in lowpass, highpass, bandpass, bandstop, and multiband configurations.

Filter Designs	
Analog Filter Design	Least Squares FIR Filter Design
Digital FIR Filter Design	Remez FIR Filter Design
Digital IIR Filter Design	Yule-Walker IIR Filter Design

Filter Realizations Library

The Filter Realizations library provides blocks and a Filter Realization Wizard for implementing a number of standard filter architectures, including time- and frequency-domain structures.

Filter Realizations	
Biquadratic Filter	Overlap-Save FFT Filter
Direct-Form II Transpose Filter	Time-Varying Direct-Form II Transpose Filter
Filter Realization Wizard	Time-Varying Lattice Filter
Overlap-Add FFT Filter	

Adaptive Filters Library

The Adaptive Filters library provides blocks for the adaptive determination of unknown filter coefficients, which is important in a wide variety of DSP operations, such as equalization and prediction.

Adaptive Filters	
Kalman Adaptive Filter	RLS Adaptive Filter
LMS Adaptive Filter	

Multirate Filters Library

The Multirate Filters library provides blocks for the efficient implementation of rate-conversion algorithms, as well as dyadic synthesis and analysis.

Multirate Filters	
Dyadic Analysis Filter Bank	FIR Interpolation
Dyadic Synthesis Filter Bank	FIR Rate Conversion
FIR Decimation	

Demos Library

The Demos library calls up the MATLAB® Demos window with the DSP Blockset demos selected. Double-click on a demo in the list to open that model, and select **Start** from the **Simulation** menu to run it.

Installation

The DSP Blockset follows the same installation procedure as the MATLAB toolboxes. See the *MATLAB Installation Guide for PC* or the *MATLAB Installation Guide for UNIX*.

Getting Started with the DSP Blockset

To open the DSP Blockset, type

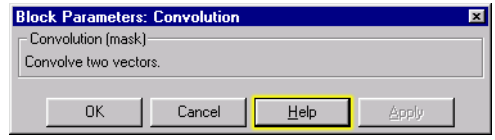
```
dsplib
```

at the MATLAB command line or right-click the **DSP Blockset** listing in the Simulink Library Browser. Double-click on any library in the DSP Blockset to view its contents, and double-click on a block to access its parameter dialog box.

How to Get Help Online

There are a number of easy ways to get help on the DSP Blockset while you're working at the computer.

- **Block Help**: Press the **Help** button in any block dialog box to view the online reference documentation for that block.



- **Simulink Library Browser**: Right-click on a block name to access the help for that block.
- **Help Desk**: Select **Help Desk** from the MATLAB **Help** menu, or type `helpdesk` or `doc` at the command line, to access the Help Desk facility.
- **Release Information**: Type `info dspblks` at the MATLAB command line to view information related to the version of the DSP Blockset that you're using, and to find out about recent changes to the blockset.

How to Use This Guide

This book contains tutorial sections that are designed to help you become familiar with using Simulink and the DSP Blockset, as well as a reference section for finding detailed information on particular blocks in the blockset.

- Read Chapter 2 of this guide, “Simulink and the DSP Blockset,” to get an overview of fundamental Simulink and DSP Blockset concepts. Also see *Using Simulink* for more information on the Simulink environment.
- Read Chapter 3 of this guide, “Using the DSP Blockset,” for details on key operations common to many signal processing tasks, and a discussion of block applications.
- Read Chapter 4, “DSP Block Reference,” for a description of each block’s operation, parameters, and characteristics.
- Read the “DSP Blockset” sections of *Release 11 New Features* and *Known Software and Documentation Problems* to learn about enhancements made to the blockset in the current version.

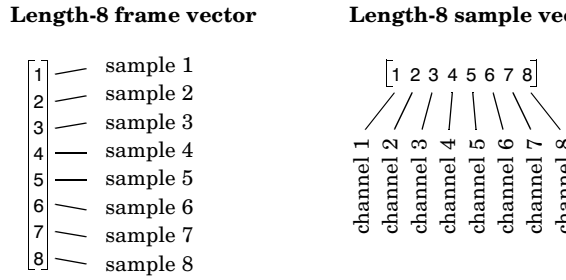
Use this guide in conjunction with the software to learn about the powerful features that the DSP Blockset provides.

Technical Conventions

Vectors. In this book, and in the block dialog boxes, the terms *width* and *length* are used interchangeably to describe the size of a vector, buffer, or frame:

```
u = [1 2 3 4 5]    % a vector of length N=5
u = 1:25           % a buffer of width 25
u = [1:64]'        % a length-64 frame
```

When a vector represents a frame of data (frame vector), it is usually shown as a *column vector* with the most recent sample in the last position, `u(end)`. When a vector represents data from multiple channels (sample vector), it is usually shown as a *row vector* with the first channel in the leftmost position, `u(1)`.



See “Understanding Matrices” and “Understanding Samples and Frames” in Chapter 2 for more information about these different representations.

Matrices. Matrix dimensions are described in terms of the *number of rows* and the *number of columns* of the matrix:

```
u = [1 2 3;4 5 6]       % a 2-by-3 matrix
```

The *number of channels* in a frame-based matrix is the number of columns.

Arrays. The *number of pages* of a three-dimensional array (in the MATLAB workspace) refers to the size of its third dimension:

```
A(:,:,1) = [1 2 3;4 5 6]   % the first page of a 3-page array
A(:,:,2) = [7 8 9;0 1 2]   % the second page
A(:,:,3) = [3 4 5;6 7 8]   % the last page
```

Important sampling-related notational conventions are listed in “Understanding Sample Rates” in Chapter 2.

Typographical Conventions

To Indicate	This Manual Uses	Example
Example code	`Monospace type`	To assign the value 5 to `A`, enter `A = 5`

MATLAB output	`Monospace type`	MATLAB responds with
		`A =`
		`5`

Function names	`Monospace type`	The `cos` function finds the cosine of each array element.
New terms	*Italics*	An *array* is an ordered collection of information.
Keys	**Boldface** with an initial capital letter	Press the **Return** key.
Menu names, items, and GUI controls	**Boldface** with an initial capital letter	Choose the **File** menu.
Mathematical expressions	Variables in *italics*. Functions, operators, and constants in standard type.	This vector represents the polynomial $p = x^2 + 2x + 3$

Related Products and Documentation

The DSP Blockset requires:

- MATLAB
- Simulink
- The Signal Processing Toolbox

In addition to these requirements, the Real-Time Workshop is an optional software component that you can use to generate code from Simulink models.

What Is MATLAB?

MATLAB is a powerful collection of tools for algorithm expression, computation, and visualization. It provides much of the control and flexibility of a traditional high-level programming language. Unlike such languages, however, MATLAB is compact and easy to learn, letting you express algorithms in concise, readable code. In addition, MATLAB provides an extensive set of ready-to-use functions including mathematical and matrix operations, graphics, color and sound control, and low-level file I/O. MATLAB is readily extensible – you can use the MATLAB language to easily create functions that operate as part of the MATLAB environment.

The DSP Blockset uses MATLAB as the computational engine for most of the block algorithms, and provides a number of blocks for exchanging data with the MATLAB workspace. Additionally, MATLAB offers powerful capabilities, such as advanced data manipulation and analysis, that you can use to complement and enhance the features in the DSP Blockset.

Note The DSP Blockset requires version 5.3 or later of MATLAB.

The *Using MATLAB* book describes the MATLAB language, including how to enter and manipulate data and how to use MATLAB's extensive collection of functions. It also explains how to create your own functions and scripts. The online *MATLAB Function Reference* provides reference descriptions of the supplied MATLAB functions and commands.

What Is Simulink?

Simulink is a dynamic system simulation environment. It allows you to represent systems as block diagrams that you build using your mouse to connect blocks and your keyboard to edit block parameters. The DSP Blockset is part of this environment – many blocks are actually masked Simulink block diagrams.

Note The DSP Blockset requires version 3.0 or later of Simulink.

The *Using Simulink* book describes how to work with Simulink. It explains how to manipulate Simulink blocks, access block parameters, and connect blocks to build models. It also provides reference descriptions of each block in the standard Simulink libraries.

Chapter 2 of this book, "Simulink and the DSP Blockset," gives a brief tutorial on working in the Simulink environment, and shows how to configure a simple DSP model for simulation.

What Is the Signal Processing Toolbox?

The Signal Processing Toolbox is a collection of tools built on the MATLAB numeric computing environment. The toolbox supports a wide range of signal processing operations, from waveform generation to filter design and implementation, parametric modeling, and spectral analysis.

Many of the DSP block algorithms (for example, all of the filter design blocks) are implemented with functions from the Signal Processing Toolbox. You can find out which toolbox functions are used by a particular block by reading the description of the block in Chapter 4, "DSP Block Reference."

Note The DSP Blockset requires version 4.2 or later of the Signal Processing Toolbox.

The *Signal Processing Toolbox User's Guide* describes the toolbox in detail. It discusses how to use the toolbox functions to address a variety of signal processing tasks, and provides tutorial information, examples, and individual function reference pages.

What Is the Real-Time Workshop?

The Real-Time Workshop, for use with MATLAB and Simulink, produces code directly from Simulink models and automatically builds programs that can be run in a variety of environments. With the Real-Time Workshop, you can run your Simulink model in real-time on a remote processor, or as a high-speed stand-alone simulation on your host machine or on an external computer. Features include support for multirate systems, as well as *loop-rolling* and S-function *inlining,* which allow you to optimize your code for size and efficiency.

The Real-Time Workshop enables you to use the DSP Blockset for *rapid prototyping* of real-time DSP systems, which can substantially shorten development cycles and reduce costs. All of the blocks in the DSP Blockset are fully qualified for code generation with the Real-Time Workshop.

See the *Real-Time Workshop User's Guide* for complete details on code generation.

2

Simulink and the DSP Blockset

Overview

This chapter will help you get started building DSP models with Simulink and the DSP Blockset. It first provides a brief overview of the Simulink environment, together with guidance on how to tailor Simulink for DSP system simulation. The chapter goes on to cover a number of topics that are especially important in DSP simulations, such as sample rates and frame-based processing. Finally, the last section offers a number of tips for increasing performance in both simulation and real-time code execution.

Introduction to the Simulink Environment

Simulink is a program for simulating dynamic systems. It provides a model-building and simulation "foundation" on which you can build digital signal processing applications. All of the blocks in the DSP Blockset are designed for use together with the blocks in the Simulink libraries.

Starting Simulink

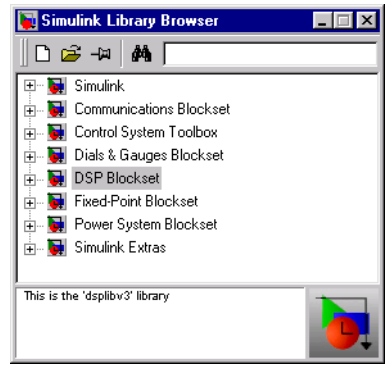
To start Simulink, click the  icon in the MATLAB toolbar, or type

```
simulink
```

at the command line.

Simulink on PC Platforms

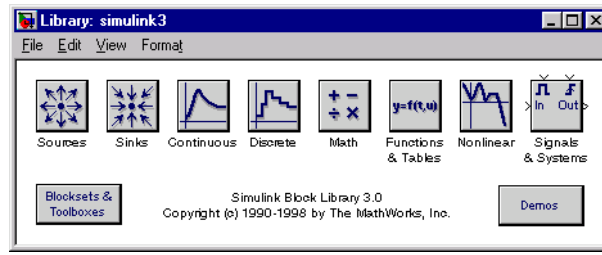
On PC platforms, the Simulink Library Browser opens when you launch Simulink. The Library Browser contains a list of all of the blocksets that you currently have installed.



The first item in the list is the Simulink blockset itself. Right-click the **Simulink** name to open the blockset window, or click the ⊞ symbol to the left of the name to expand the hierarchical list and display the Simulink libraries within the browser.

Simulink on UNIX Platforms

On UNIX platforms, the Simulink window (below) opens immediately when you launch Simulink.



The Simulink Libraries

The eight libraries in the Simulink window contain all of the basic elements you need to construct a model. You should look here for basic math operations, switches, connectors, simulation control elements, and other items that do not have a specific DSP orientation.

To create a new model, select **New** from the Simulink **File** menu. Then simply drag a block from one of the Simulink libraries into the new model window to begin building a system.

Using Simulink

If you have never used Simulink before, take some time to get acquainted with its features. The keys to building models with Simulink are model definition and model simulation.

Model Definition

Simulink is a *model definition* environment. You define a model by creating a block diagram that represents the computations and data flow of your system or application. Try building a simple model:

1. Select **New** from the Simulink **File** menu. A new block diagram window appears on your screen.

2. Double-click on the Sources, Sinks, and Discrete icons in the Simulink window. The Sources, Sinks, and Discrete libraries appear on your screen.

   Alternatively, you can view the libraries' contents by clicking the ⊞ symbol in the Library Browser to expand the hierarchical list.

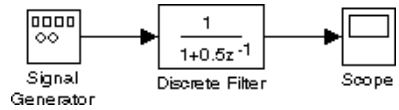
3 Drag the Signal Generator block from the Sources library into the new block diagram window.

4 Drag the Discrete Filter block from the Discrete library into the new block diagram window.

5 Drag the Scope block from the Sinks library into the new block diagram window.

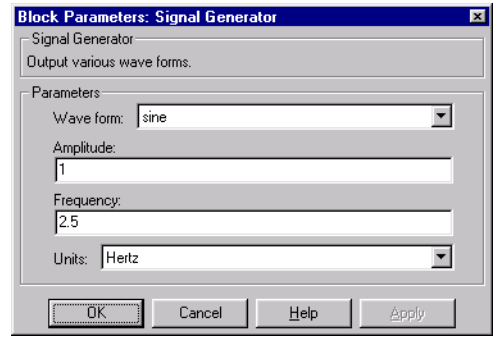
6 Connect the blocks.

Position the cursor near the output port of the Signal Generator block. Hold down the mouse button (the left button for a multibutton mouse) and drag the line that appears until it touches the input port of the Discrete Filter block. Using the same technique, connect the output of the Discrete Filter block to the input port of the Scope block. Your model looks like this:



7 Set the block parameters:

a Double-click on the Signal Generator block. A window appears that lets you set the block's parameters. *Parameters* are defining values that tell the block how to operate. For this example, select **sine** from the **Wave form** pop-up menu. Set the **Frequency** to `2.5` by typing in the text field.



Close the window by clicking on the **OK** button or by pressing **Enter** on the keyboard.

b Double-click on the Discrete Filter block. Set the **Denominator** field to `[1 -0.9]`, or enter the name of a MATLAB workspace variable that contains this vector. Set the **Sample time** parameter to `0.01`. Close the dialog box by clicking on the **OK** button or by pressing **Enter** on the keyboard.

c Double-click on the Scope block. Right-click on the vertical axis and select **Axis properties** from the pop-up menu. Set **Y max** to `7` and **Y min** to `-7`. Click **OK**.

8 In the model window, select **Parameters** from the **Simulation** menu. Set **Stop time** to `4` and click **OK**.

Model Simulation

Simulink is also a *model simulation* environment. You can run the simulation block diagram that you built to see how the system behaves. To do this:

1 Double-click on the Scope block if the Scope window is not already open on your screen. A scope window appears.

2 Select **Start** from the **Simulation** menu for the block diagram window.

When you select **Start**, the simulation progresses according to an underlying integration routine for continuous blocks, or a sample rate for discrete blocks. For this model, and for most models constructed with the DSP Blockset, the blocks process their inputs at a discrete sample interval.

3 Experiment with the block parameters. To change block parameters during the simulation:

a Select **Parameters** from the **Simulation** menu and change the **Stop time** to `inf`. The `inf` setting instructs Simulink to run the model for as long as the computer's memory allows.

b Double-click on the Signal Generator block to open it.

c Select **Start** from the **Simulation** menu to start the simulation.

d Change the frequency of the sine wave in the Signal Generator block. Try typing `1`, `0.1`, and `0.01` in the **Frequency** field, pressing **Apply** after entering each new value. Observe the changes on the scope.

Many blocks have parameters that you cannot change while a simulation is running. There are some parameters, however, that you can *tune* without terminating the simulation. In Chapter 4, "DSP Block Reference," these parameters are designated by a Ⓣ icon, indicating that they are *tunable* while the simulation runs.

4 Select **Stop** from the **Simulation** menu to stop the simulation.

Running a Simulation from an M-File. You can also modify and run a Simulink simulation from within a MATLAB M-file. By doing this, you can automate the variation of model parameters to explore a large number of simulation conditions rapidly and efficiently. For information on how to do this, see "Increasing Performance" at the end of this chapter, and "Running a Simulation from the Command Line" in Chapter 4 of *Using Simulink*.

Learning More About Simulink

Here are a few more suggestions to help you get started with Simulink:

- Look through *Using Simulink* to get complete exposure to all of Simulink's capabilities.
- Open the Simulink library as described at the beginning of this chapter. Build a few simple models using blocks from the Simulink Sources, Linear, Nonlinear, and Sinks libraries. (For example, add two sine waves with different frequencies and view the result on a Scope.)
- Open some of the models in the DSP Blockset Demos library. Most of the advanced demos have blocks that you can double-click to get information about the application. The Demos library also contains easy-to-understand models that demonstrate some of the blockset's elementary math and statistics blocks. In each case, just select **Start** from the **Simulation** menu to run the simulation.

Configuring Simulink for DSP Systems

When you create a new DSP model, you may want to adjust certain Simulink settings to suit your needs. A very typical change, for example, is to adjust the **Stop time** parameter (in the **Simulation Parameters** dialog box) to a different value. Another common change is to specify the **Fixed-step** option in the **Solver options** panel, to reflect the discrete nature of the DSP model.

The DSP Blockset provides an M-file, `dspstartup`, that lets you automate this configuration process so that every new model you create is preconfigured for DSP simulation. The M-file executes the following commands:

```
set_param(0, ...
   'Solver',             'fixedstepdiscrete', ...
   'SolverMode',         'auto', ...
   'StartTime',          '0.0', ...
   'StopTime',           'inf', ...
   'FixedStep',          'auto', ...
   'SaveTime',           'off', ...
   'SaveOutput',         'off', ...
   'AlgebraicLoopMsg',   'error', ...
   'InvariantConstants', 'on', ...
   'RTWOptions',         [get_param(0,'RTWOptions')
                            ' -aRollThreshold=2']);
```

Using dspstartup.m

There are two ways to use the `dspstartup` M-file:

- Run it from the MATLAB command line, by typing `dspstartup`, to preconfigure all of the models that you subsequently create. Existing models are not affected.
- Place a call to `dspstartup` within the `startup.m` file. This is an efficient way to use `dspstartup` if you would like these settings to be in effect every time you start Simulink.

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`, and save the file as `startup.m`. The result should look like something like this:

```
%STARTUP Startup file
%   This file is executed when MATLAB starts up,
%   if it exists anywhere on the path.

dspstartup;
```

The default settings in `dspstartup` will now be in effect every time you launch Simulink.

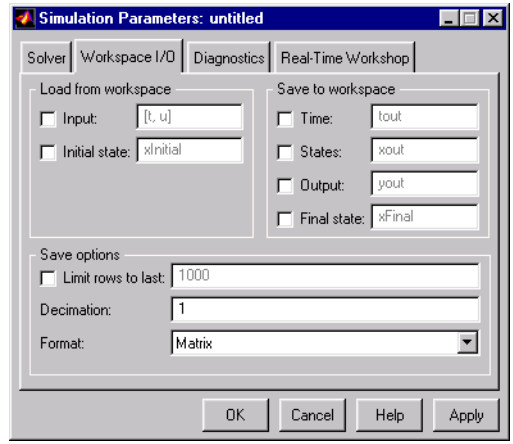
For more information about performing automated tasks at startup, see the documentation for the `startup` command in the online *MATLAB Function Reference*.

Performance-Related Settings

A number of the settings in the `dspstartup` M-file are chosen to improve the performance of the simulation:

- `'SaveTime'` is set to `'off'`

  When `'SaveTime'` is set to `'off'`, Simulink does not save the `tout` time-step vector to the workspace. The time-step record is not usually needed for analyzing discrete-time simulations, and disabling it can save a considerable amount of memory, especially when the simulation runs for an extended period of time. To enable time recording for a particular model, select the **Time** check box in the **Workspace I/O** panel of the **Simulation Parameters** dialog box.



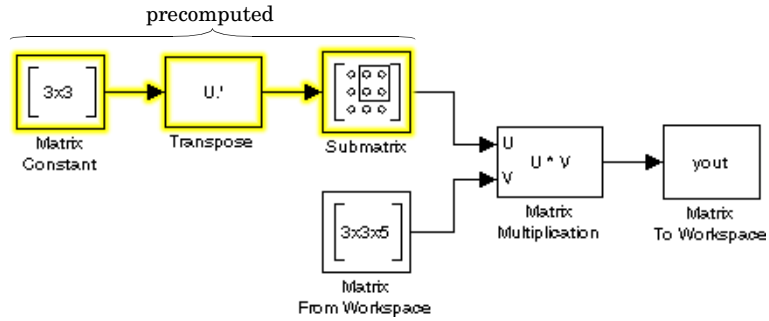
- `'SaveOutput'` is set to `'off'`

  When `'SaveOutput'` is set to `'off'`, Simulink Out blocks in the top level of a model do not generate an output (`yout`) in the workspace. To reenable output recording for a particular model, select the **Output** check box in the **Workspace I/O** panel of the **Simulation Parameters** dialog box (above).

- `'InvariantConstants'` is set to `'on'`

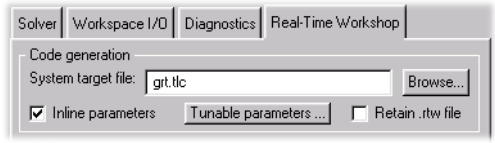
  When `'InvariantConstants'` is set to `'on'`, Simulink precomputes the values of all constant blocks (e.g., Discrete Constant, Constant Diagonal Matrix, Matrix Constant) at the start of the simulation, and does not update them again for the duration of the simulation. Simulink additionally precomputes the outputs of all downstream blocks driven exclusively by constant blocks.

  In the example below, the input to the top port (`U`) of the Matrix Multiplication block is computed only once, at the start of the simulation.



This eliminates the computational overhead of continuously reevaluating these constant branches, which in turn results in faster simulation, and smaller and more efficient generated code.

Note, however, that when `'InvariantConstants'` is set to `'on'`, changes that you make to parameters in a constant block while the simulation is running are not registered by Simulink, and do not affect the simulation. If you would like to adjust the model constants while the simulation is running, you can turn off `'InvariantConstants'` by deselecting the **Inline Parameters** check box in the **Real-Time Workshop** panel of the **Simulation Parameters** dialog box.



- `'RTWOptions'` sets loop-rolling threshold to 2

  By default, the Real-Time Workshop "unrolls" a given loop into inline code when the number of loop iterations is less than five. This avoids the overhead of servicing the loop in cases when inline code can be used with only a modest increase in the file size.

  However, because typical DSP processors offer *zero-overhead* looping, code size is the primary optimization constraint in most designs. It is therefore more efficient to minimize code size by generating a loop for every instance of iteration, regardless of the number of repetitions. This is what the `'RTWOptions'` loop-rolling setting in `dspstartup` accomplishes.

Miscellaneous Settings

The `dspstartup` M-file adjusts several other parameters to make it easier to run DSP simulations. Two of the important settings are:

- `'StopTime'` is set to `'inf'`, which allows the simulation to run until you manually stop it by selecting **Stop** from the **Simulation** menu or by pressing the **Stop Simulation** button on the toolbar. To set a finite stop time, enter a value for the **Stop time** parameter in the **Simulation Parameters** dialog box.

- `'Solver'` is set to `'fixedstepdiscrete'`, which selects the fixed-step solver option instead of Simulink's default variable-step solver. See "Discrete-Time Signals in Simulink" in the next section for more information about the various solver settings.

For complete information on any of these parameters, see *Using Simulink*.

Customizing dspstartup.m

You can edit the `dspstartup` M-file to change any of the settings above or to add your own custom settings. For example, you can change the `'StopTime'` option to a value that is better suited to your particular simulation, or set the `'SaveTime'` option to `'on'` if you typically record the simulation sample times.

Understanding Sample Rates

Sample rates are an important concern in most DSP models, especially in systems incorporating rate conversions. In most cases, when you build a Simulink model you only need to worry about setting sample rates in the source blocks, such as Signal From Workspace; Simulink automatically computes the appropriate sample rates for all downstream blocks.

Nevertheless, if you are designing a DSP system with Simulink for the first time, you may have a number of questions about sampling, especially if your model incorporates frame-based or multirate components.

For example:

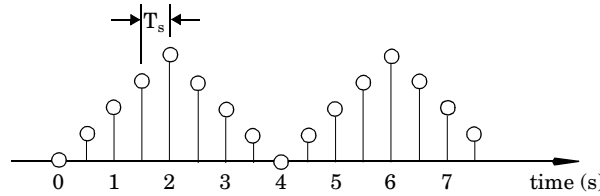
- What is the sample rate of a particular signal in the model?
- Why do some blocks generate an error when the input is not discrete?
- Which DSP blocks change the rate of the input?
- How does sampling differ for sample-based and frame-based sequences?
- How does sampling work for multirate blocks?

The following sections will address these issues, and help you to confidently create sophisticated multirate DSP models.

Discrete-Time Signals

In theory, a discrete-time signal is defined by a sequence of values corresponding to particular instants in time. The time instants where the signal is defined are called the signal's *sample times*; traditionally, a discrete signal is considered to be undefined at points in time between these instants. For a periodically sampled signal, the equal interval between any pair of sample times is the signal's *sample period*, T_s. The *sample rate*, F_s, is the reciprocal of the sample period, or $1/T_s$.

For example, the 7.5-second triangle wave segment below has a sample period of 0.5 sec, and sample times of 0.0, 0.5, 1.0, 1.5, ...,7.5. The sample frequency of the sequence is therefore 1/0.5, or 2 Hz.



Time and Frequency Terminology

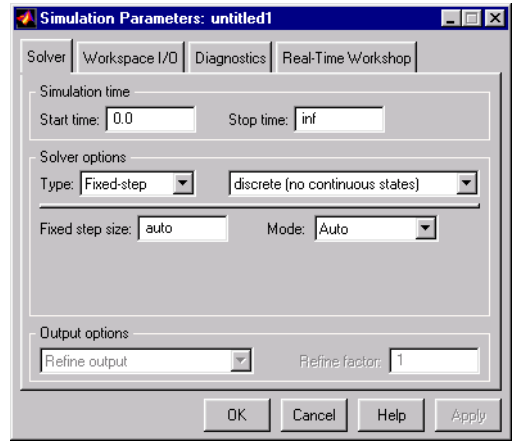
A number of different terms are used in this book to describe the characteristics of discrete-time signals found in Simulink models. These terms, which are listed in the table below, are frequently used in Chapter 4, "DSP Block Reference," to describe the way that various blocks operate on sample-based and frame-based signals. For additional information on frame-based processing, see "Understanding Samples and Frames" in this chapter.

Term	Symbol	Units	Notes
Sample period	T_s, T_{si}, T_{so}, Δt	Seconds	The time interval between consecutive samples in a sequence, as the input to a block (T_{si}) or the output from a block (T_{so}). $T_{si} = T_{fi}$ if the input is sample based. $T_{so} = T_{fo}$ if the output is sample-based.
Frame period, Input period, Output period	T_f, T_{fi}, T_{fo}	Seconds	The time interval between consecutive frames in a sequence, as the input to a block (T_{fi}) or the output from a block (T_{fo}). $T_{fi} = T_{si}$ if the input is sample based. $T_{fo} = T_{so}$ if the output is sample based.
Signal period	T	Seconds	The time elapsed during a single repetition of a periodic signal.
Sample rate, Sample frequency	F_s	Hz (samples per second)	The number of samples per unit time, $F_s = 1/T_s$.
Frequency	f	Hz (cycles per second)	The number of repetitions per unit time of a periodic signal, $f = 1/T$.

Term	Symbol	Units	Notes
Normalized frequency	f_n	Two cycles per sample	Frequency of a periodic signal normalized to the Nyquist frequency, $f_n = \omega_n/\pi = 2f/F_s$.
Angular frequency	ω	Rads per sec	Frequency of a periodic signal in angular units, $\omega = 2\pi f$.
Digital (normalized angular) frequency	ω_n	Rads per sample	Frequency of a periodic signal normalized to the sample frequency, $\omega_n = \omega/F_s = \pi f_n$

Discrete-Time Signals in Simulink

Simulink allows you to select from among several different simulation solver algorithms through the **Solver options** panel in the **Simulation Parameters** dialog box.



Recommended Simulation Settings for DSP. The recommended **Solver options** settings for DSP simulations are:

- **Type = Fixed-step discrete**
- **Fixed step size** = `auto`
- **Mode = Auto**

With these settings, discrete signals in Simulink most accurately model the prototypical discrete signal described in the previous section. In particular,

when these settings are in effect, discrete signals are *undefined* between sample times. Simulink generates an error when operations attempt to reference the undefined region of a signal, for example, when signals with different sample rates are added.

To perform cross-rate operations like the addition of two signals with different sample rates, you must *explicitly* convert the two signals to a common sample rate. There are several blocks provided for precisely this purpose in the Signal Operations and Multirate Filtering libraries. See "Rate Conversion," later in this section, for more information. By requiring explicit rate conversions for cross-rate operations, Simulink helps you to identify sample rate conversion issues early in the design process.

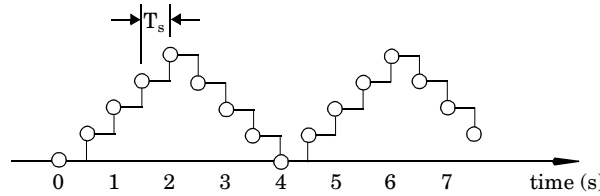
You can automatically set the above solver options for all new models by running the `dspstartup` M-file. See "Configuring Simulink for DSP Systems" earlier in this chapter for more information.

Note In the block dialog boxes, the term *sample time* is often used to refer to the *sample period*, T_s. An example is the **Sample time** parameter in the Signal From Workspace block, which specifies the imported signal's sample period.

Other Simulation Settings. It is worthwhile to know how the other solver options available in Simulink affect discrete signals. In particular, you should be aware of the properties of discrete signals under the following options:

- **Variable-step** (Simulink's default solver)
- **Fixed-step**, **Mode** = **SingleTasking**

In both cases, discrete-time signals *differ* from the prototype described earlier by remaining *defined* between sample times. For example, the representation of the discrete-time triangle wave looks like this:

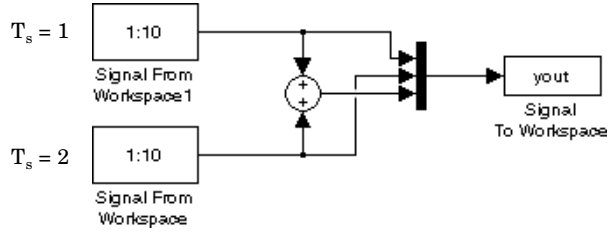


Because Simulink blocks typically apply a zero-order hold (ZOH) to outputs, the value of the signal between two adjacent sample times is the value at the earlier sample time. As an example, the signal's value at t=3.112 seconds is the same as the signal's value at t=3 seconds. In these two modes, a signal's sample times are the instants where the signal is allowed to *change* values, rather than where the signal is defined. Between the sample times, the signal is frozen at its last value.

As a result, in the **Variable step** and **Fixed-step SingleTasking** modes, Simulink permits cross-rate operations such as the addition of two signals of different rates. The next section explains how this works.

More About Variable-Step and Fixed-Step SingleTasking Modes. Because in these modes a discrete-time signal is defined between sample times, if you sample the signal with a rate or phase that is distinct from the signal's own rate and phase, you still measure meaningful values.

Consider the model below, which sums two signals having different sample periods. The fast signal (T_s=1) has sample times 1, 2, 3, ..., and the slow signal (T_s=2) has sample times 1, 3, 5, ...



The output, `yout`, is a matrix containing the fast signal (T_s=1) in the first column, the slow signal (T_s=2) in the second column, and the sum of the two in the third column.

```
yout =

     1     1     2
     2     1     3
     3     2     5
     4     2     6
     5     3     8
     6     3     9
     7     4    11
     8     4    12
     9     5    14
    10     5    15
```

As expected, the slow signal (second column) changes once every two seconds, half as often as the fast signal. Nevertheless, it has a defined value at every moment in-between because of the zero-order hold that the Signal From Workspace block applies to the output. (Simulink implicitly auto-promotes the rate of the slower signal to match the rate of the faster signal before the addition operation is performed.)

In general, for **Variable-step** and **Fixed-step SingleTasking** modes, when you measure the value of a discrete signal in-between sample times, you are observing the value of the signal at the most recent sample time.

Sample Time Offsets. Simulink offers the ability to shift a signal's sample times by an arbitrary value, which is equivalent to shifting the signal's phase by a fractional sample period. However, sample-time offsets are rarely used in DSP systems, and blocks from the DSP Blockset do not support them.

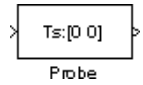
Inspecting Sample Rates

When constructing a frame-based or multirate model, it is often helpful to check the sample rates that Simulink computes for different signals. There are two basic ways to inspect the sample rates in a model:

- By using the Probe block
- By turning on sample rate color coding

Probe Block

Connect Simulink's Probe block to any line to display the period of the signal on that line. The period is displayed in the block icon itself (together with the line width and data type, if desired), making it easy to verify that the sample rates in the model are what you expect them to be. When the line width and data type displays are suppressed (by deselecting the appropriate check boxes in the block dialog box), the Probe block looks like this:



The block displays the label `Ts`, followed by a two-element vector. The first (left) element is the period of the signal being measured. The second (right) is the signal's sample time offset, which is usually `0`, as described in "Sample Time Offsets" above.

For sample-based signals, the value shown in the Probe block icon is the actual sample period of the sequence, T_s. For frame-based signals, the value shown in the Probe block icon is the frame period, T_f. The next section explains the difference.

Note The Probe block *always* displays the generic `Ts` label in the block icon along with the period of the signal. Be aware, however, that when measuring frame-based signals, the value displayed by the Probe block is the frame period, T_f, not the sequence sample period, T_s.

Frame Period vs. Sample Period. It is important to distinguish between the frame period and the sample period of a frame-based signal.

The *input frame period* (T_{fi}) of a frame-based signal is the time interval between consecutive vector or matrix inputs to a block. This interval is what the Probe block displays when you connect it to an input line. Similarly, the *output frame period* (T_{fo}) is the time interval at which the block updates the vector or matrix value at the output port. This interval is what the Probe block displays when you connect it to the output line.

In contrast, the sample period, T_s, is the time interval between individual samples in a frame, which is always shorter than the frame period itself. The

sample period of the sequence contained in consecutive frames is the quotient of the frame period and the frame size, M.

$$T_s = T_f / M$$

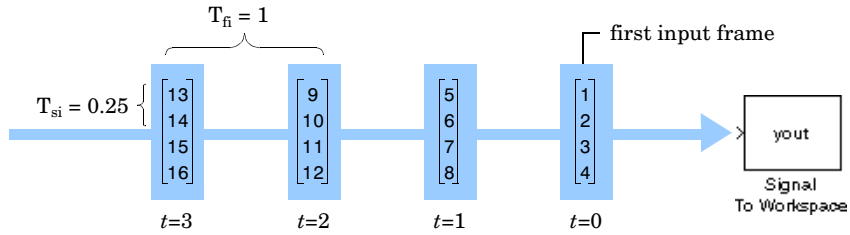
More specifically, the sample periods of inputs and outputs are related to their respective frame periods by

$$T_{si} = T_{fi} / M_i$$

$$T_{so} = T_{fo} / M_o$$

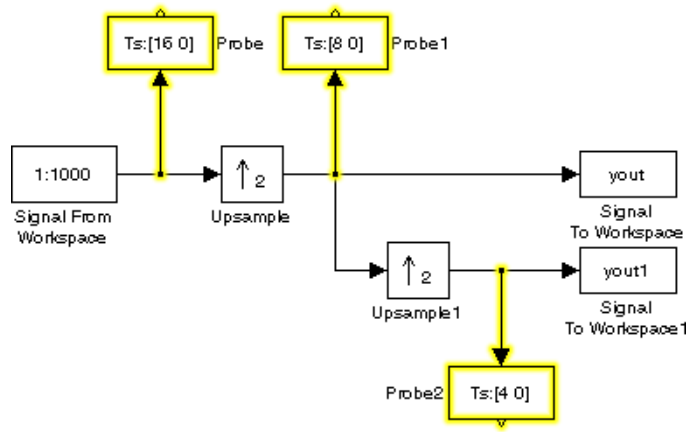
where M_i and M_o are the input and output frame sizes, respectively.

The illustration below shows a frame-based signal with a frame size (M_i) of 4 and a frame period (T_{fi}) of 1. The sample period, T_{si}, is 1/4, or 0.25 seconds. A Probe block connected to this signal would display the frame period, $T_{fi} = 1$.



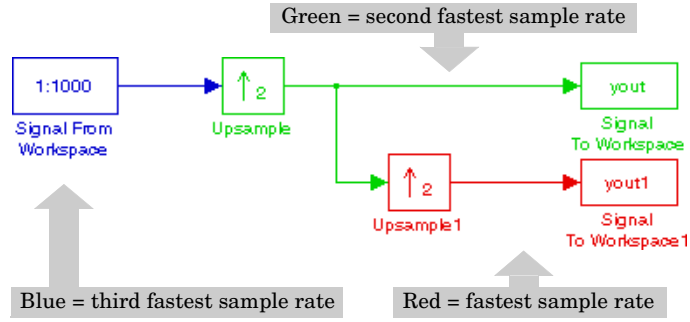
In most cases, the sequence sample rate, T_{si}, is of primary interest, and the frame rate is simply a secondary result of the frame size that you choose for the signal. For a sequence with a particular sample period, a larger frame size corresponds to a slower frame rate, and vice versa.

Probe Block Example. The three Probe blocks in the sample-based model below verify that the scalar signal's sample period is halved with each upsample operation: The output from the Signal From Workspace block has a sample period of 16, the output from the first Upsample block has a sample period of 8, and the output from the second Upsample block has a sample period of 4.



Sample Time Color Coding

Turn on Simulink's sample time color coding option by selecting **Sample time colors** from the **Format** menu. Here's the above model with the Probe blocks removed and sample time color coding turned on.



Types of Sampling

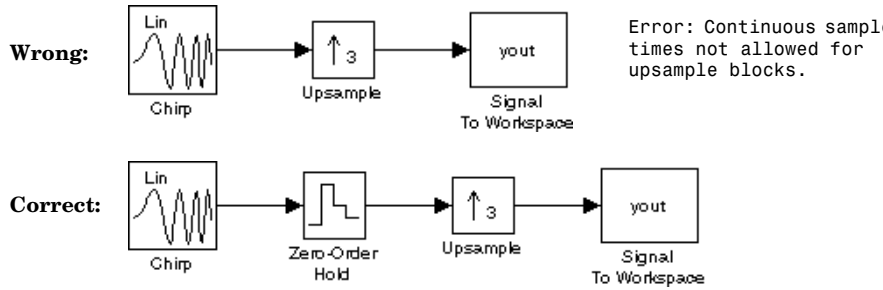
Most signals in a DSP Simulink model are discrete-time signals, and all of the blocks in the DSP Blockset accept discrete-time inputs. However, many blocks can also operate on continuous-time signals, whose values vary continuously with time. Similarly, most blocks generate discrete-time outputs, but some generate continuous-time outputs.

The sampling behavior of a particular block determines which other blocks you can connect as an input or output. The following sections describe the behavior of both source and nonsource blocks in the DSP Blockset. See Chapter 4, "DSP Block Reference," for information about the particular sample characteristics of each block in the blockset.

Source Blocks

Source blocks are those blocks that generate or import data into a model. Many of these blocks have the term "from workspace" or "constant" in the block name (e.g., Signal From Workspace, Matrix Constant), and most appear in the DSP Sources library. See "Working with Sources and Sinks" in Chapter 3 to fully explore the features of these blocks.

Continuous-Time Source Blocks. The sample period for continuous-time source blocks is set internally to zero (which indicates a continuous-time signal). Examples are Chirp and Constant. As shown below, when connecting such blocks to certain nonsource discrete-time blocks, you may need to interpose a Zero-Order Hold block to discretize the signal. Specify the desired sample period for the signal in the **Sample time** parameter of the Zero-Order Hold block.



The Triggered Signal From Workspace block is also considered to be a continuous-time block.

Discrete-Time Source Blocks. Discrete-time source blocks, such as Matrix From Workspace, require a discrete (i.e., nonzero) sample period to specified in the block's **Sample time** parameter. Simulink generates an error if a zero value is specified for the **Sample time** parameter of a discrete-time source block.

Nonsource Blocks

All nonsource blocks in the DSP Blockset accept discrete signals, and inherit the sample period of the input. Others additionally accept continuous-time discrete signals.

Discrete-Time Nonsource Blocks. Many blocks can accept only discrete-time inputs, and generate only discrete-time outputs. Examples are all of the resampling and delay blocks (e.g., Upsample, Integer Delay). These blocks *inherit* the sample period of the driving block (the block supplying the input). This means that the block automatically synchronizes its sampling rate with the driving block. For example, if the driving block's sample period is 0.5 seconds, then the inheriting block that it drives also executes at 0.5 second intervals. Simulink generates an error if a continuous input is connected to a discrete-only block.

Continuous/Discrete Non-Source Blocks. In the continuous/discrete blocks, continuous-time inputs generate continuous-time outputs, and discrete-time inputs generate discrete-time outputs. Examples are all of the blocks in the Vector Functions library (in Math Functions). The nonsource *triggered* blocks (e.g. Triggered Shift Register) are also in this category.

Rate Conversion

In a DSP Blockset model, there are two types of periods that you will commonly be concerned with: frame periods and sample periods. The sample periods of block inputs and outputs are related to their respective frame periods by

$$T_{si} = T_{fi}/M_i$$

$$T_{so} = T_{fo}/M_o$$

where the subscripts i and o indicate *input* and *output*, respectively.

The combined buffering and rate-conversion capabilities of the DSP Blockset generally allow you to independently vary any two of the three parameters (T_{si}, T_{fi}, M_i). In most cases, the sample period and the frame size are the two parameters of primary interest; the frame period is decided by your choices for T_{so} and M_o.

$$T_{fo} = M_o T_{so}$$

There are two common types of operations that impact the frame and sample rates of a signal:

- *Direct rate conversions*

  Direct rate conversions, such as upsampling and downsampling, are a feature of most DSP systems, and can be implemented by altering either the frame rate or the frame size of a signal.

- *Frame rebuffering*

  The principal purpose of frame rebuffering is to alter the frame size of a signal, usually to improve simulation throughput. By redistributing the signal samples to frames of a new size, rebuffering usually changes either the sample or frame rate of the signal.

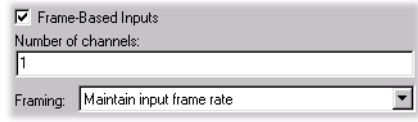
Both operations are discussed in the next sections, along with ways to avoid *unintentional* rate conversions.

Note Technically, when a Simulink model contains signals with various frame periods (rates), the model is called *multirate*. You can find a discussion of multirate models in the "Discrete Time Systems" section in Chapter 9 of *Using Simulink*.

Direct Rate Conversion

Rate conversion blocks accept an input signal at one rate, and output the same signal at a new rate. Several of these blocks contain a **Framing** parameter offering two options for adjusting the rate of the signal:

- **Maintain input frame rate**: Change the frame size ($M_o \neq M_i$) but keep the frame rate constant ($T_{fo} = T_{fi}$)
- **Maintain input frame size**: Change the output frame rate ($T_{fo} \neq T_{fi}$), but keep the frame size constant ($M_o = M_i$)

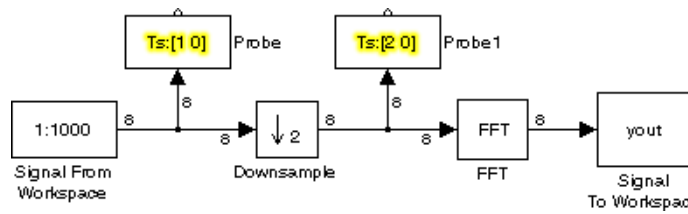


The examples following the list of rate conversion blocks below illustrate both modes.

Rate Conversion Blocks. The following table lists the principal rate conversion blocks in the DSP Blockset. Blocks marked with an asterisk (*) offer the option of changing the rate by either adjusting the frame size or frame rate.

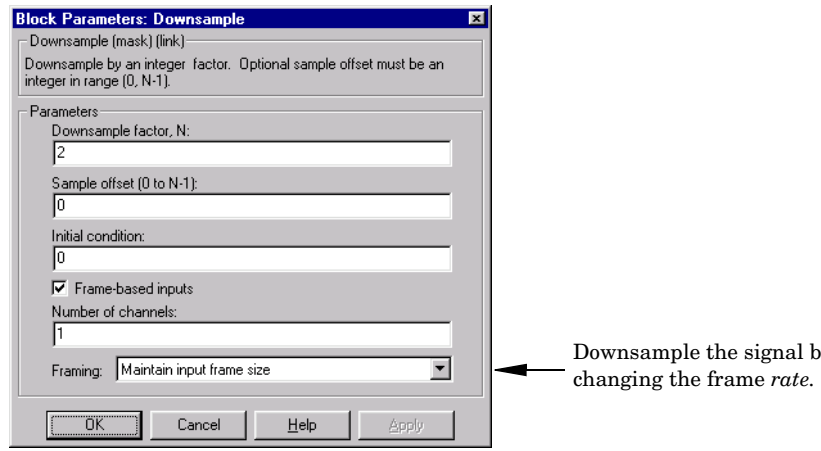
Block	Library
Downsample *	Signal Operations, in General DSP
Dyadic Analysis Filter Bank	Multirate Filters, in Filtering
Dyadic Synthesis Filter Bank	Multirate Filters, in Filtering
FIR Decimation *	Multirate Filters, in Filtering
FIR Interpolation *	Multirate Filters, in Filtering
FIR Rate Conversion	Multirate Filters, in Filtering
Repeat *	Signal Operations, in General DSP
Upsample *	Signal Operations, in General DSP

Example: Rate Conversion by Frame-Rate Adjustment. A common example of direct rate conversion is shown below, where the signal is directly downsampled to half its original rate. The values next to input and output ports are the line widths, displayed by selecting **Vector Line Widths** from the model window's **Format** menu.



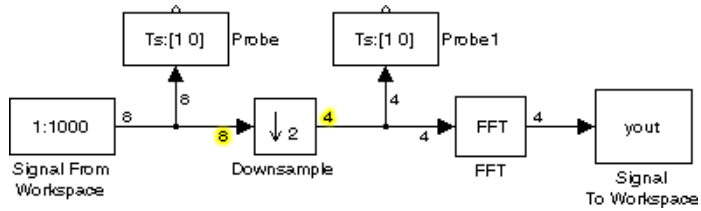
The sample period and frame size of the original signal are set to 0.125 seconds and 8 samples per frame, respectively, by the **Sample time** and **Samples per frame** parameters in the Signal From Workspace block. This results in a frame rate of 1 second (0.125∗8), as shown by the first Probe block.

The Downsample block is configured to downsample the signal by changing the frame rate rather than the frame size. The dialog box with this setting is shown below.

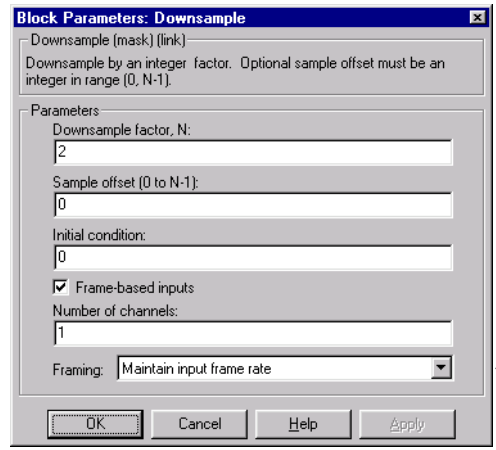


The second Probe block in the model verifies that the output from the Downsample block has a frame period of 2, twice that of the input (half the rate). As a result, the sequence sample period is doubled to 0.25 seconds without any change to the frame size.

Example: Rate Conversion by Frame-Size Adjustment. The same model is shown again below, but this time with the rate conversion implemented by adjusting the frame size, rather than the frame rate.



As before, the frame rate of the original signal is 1 second (0.125∗8), shown by the first Probe block. Now the Downsample block is configured to downsample the signal by changing the frame size rather than the frame rate. The dialog box with this setting is shown below.



Downsample the signal by changing the frame *size*.

The line width display on the Downsample output port verifies that the downsampled output has a frame size of 4, half that of the input. As a result, the sequence sample period is doubled to 0.25 seconds without any change to the frame rate.

Frame Rebuffering

Buffering operations are another common cause of rate changes in DSP models. The purpose of many buffering operations is to adjust the frame size of the signal without altering the sequence sample rate, in essence changing M while holding T_s fixed. Usually, this type of buffering operation results in a change to the signal's frame rate, T_f, according to the relation

$$T_f = MT_s$$

discussed earlier.

However, this is only true when the *original signal is preserved* in the buffering operation, with no samples added or deleted. Buffering operations that generate overlapping frames, or that only partially unbuffer frames, alter the data sequence by adding or deleting samples. In this case the above relation is not valid.

The sections following the list of buffering blocks below discuss both buffering with *preservation* of the signal and buffering with *alteration* of the signal.

Buffering Blocks. The following table lists the principle buffering blocks in the DSP Blockset.

Block	Library
Buffer	Buffers, in General DSP
Partial Unbuffer	Buffers, in General DSP
Rebuffer	Buffers, in General DSP
Shift Register	Buffers, in General DSP
Unbuffer	Buffers, in General DSP
Variable Selector	Elementary Functions, in Math Functions
Zero Pad	Signal Operations, in General DSP

Buffering with Preservation of the Signal. There are various reasons that you may need to rebuffer a signal to a new frame size at some point in a model. For example, your data acquisition hardware may internally buffer the sampled signal to a frame size that is not optimal for the DSP algorithm in the model. In this case, you would want to rebuffer the signal to a frame size more appropriate for the intended operations, but without introducing any change to the data or sample rate.

There are three blocks in the Buffers library that can be used to change a signal's frame size without altering the signal itself:

- Buffer, to buffer scalar samples (frame size = 1) to an arbitrary frame size
- Rebuffer, to redistribute signal samples to a larger or smaller frame size
- Unbuffer, to unbuffer frames to a scalar sequence (frame size = 1)

Rebuffer is the most general of the three blocks, allowing conversions between any two frame sizes. Buffer is specialized for buffering scalar samples. Both Buffer and Rebuffer preserve the signal's data and sample period only when their **Buffer overlap** parameter is set to `0`. The output frame periods, T_{fo}, of the two blocks are

Buffer block:

$$T_{fo} = M_o T_{si}$$

Rebuffer block:

$$T_{fo} = \frac{M_o T_{fi}}{M_i}$$

where T_{si} is the scalar input sample period to the Buffer block, T_{fi} is the input frame period to the Rebuffer block, M_i is the input frame size, and M_o is the output frame size, specified by the **Buffer size** parameter.

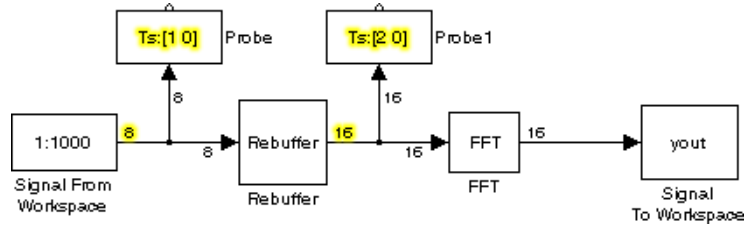
The Unbuffer block is specialized for unbuffering scalar samples, and always preserves the signal's data and sample period.

$$T_{so} = T_{fi}/M_i$$

where T_{fi} and M_i are the period and size, respectively, of the frame-based input.

In all three cases listed above, the sample period of the sequence is preserved in the conversion ($T_{so} = T_{si}$).

Example: Buffering with Preservation of the Signal. In the model below, a signal with a sample period of 0.125 seconds is rebuffered from a frame size of 8 to a frame size of 16. This doubles the frame period from 1 to 2 seconds, but does not change the sample period of the signal ($T_{si} = T_{so} = 0.125$).



Buffering with Alteration of the Signal. Other forms of buffering alter the signal's data or sample period, in addition to adjusting the frame size. There are many instances when this type of buffering is desirable. Examples are creating sliding windows by overlapping consecutive frames of a signal, and selecting a subset of samples from each input frame for processing.

The blocks that alter a signal while adjusting its frame size are listed below. In this list, T_{si} is the input sequence sample period, and T_{fi} and T_{fo} are the input and output frame periods, respectively.

- Buffer and Rebuffer add duplicate samples to a sequence when the **Buffer overlap** parameter, L, is set to a nonzero value. The output frame period is related to the input sample period by

  $$T_{fo} = (M_o - L)T_{si}$$

  where M_o is the output frame size specified by the **Buffer size** parameter. As a result, the new output sample period is

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

- Partial Unbuffer removes samples from the sequence whenever the **First output index** (M_1) and **Last output index** (M_2) parameters do not span the entire input frame. Samples `(1:(M1-1))` are eliminated from the beginning of each frame, and samples `((M2+1):end)` are eliminated from the end of each frame. The output sample period is related to the input sample period by

  $$T_{so} = \frac{M_i T_{si}}{M_2 - M_1 + 1}$$

  where M_i is the input frame size.

- Shift Register adds duplicate samples to the sequence when the **Register size** parameter, M_o, is greater than 1. The output and input frame periods are the same, $T_{fo} = T_{fi} = T_{si}$, and the new output sample period is

  $$T_{so} = \frac{T_{si}}{M_o}$$

- Variable Selector can remove, add, and/or rearrange samples in the input frame. The output and input frame periods are the same, $T_{fo} = T_{fi}$, and the new output sample period is

  $$T_{so} = \frac{M_i T_{si}}{M_o}$$

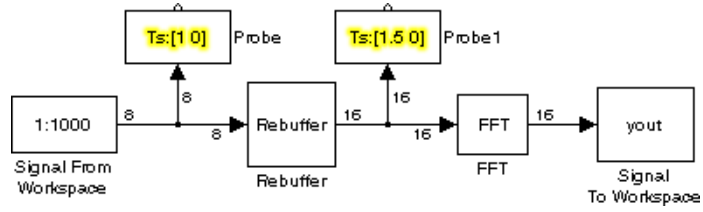
  where M_o is the length of the block's output, determined by the indexing vector in the block dialog box.
- Zero Pad adds samples to the sequence by appending zeros to each frame. The output and input frame periods are the same, $T_{fo} = T_{fi}$, and the new output sample period is

  $$T_{so} = \frac{M_i T_{si}}{M_o}$$

  where M_o is the length of the block's output, determined by the **Output frame size** parameter in the block dialog box.

In all of these cases, the sample period of the output sequence is *not* equal to the sample period of the input sequence.

Example: Buffering with Alteration of the Signal. In the model below, a signal with a sample period of 0.125 seconds is rebuffered from a frame size of 8 to a frame size of 16 with an overlap of 4.



The relation for the output frame period,

$$T_{fo} = (M_o - L)T_{si}$$

indicates that T_{fo} should be (16-4)*0.125, or 1.5 seconds, which is confirmed by the second Probe block. The sample period of the signal at the output of the Rebuffer block is no longer 0.125 seconds, but rather 0.0938 seconds (1.5/16).

Thus, both the signal's data and the signal's sample period have been altered by the rebuffering operation.

Avoiding Unintended Rate Conversions

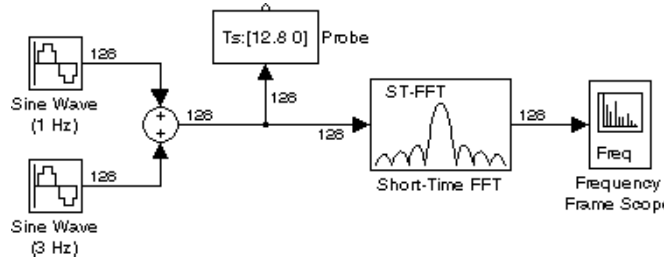
The previous sections discussed a number of the blocks that are responsible for rate conversions. It is important to be aware of where in a model these rate conversions are taking place; in a few cases, *unintentional* rate conversions can produce misleading results. The models below provide an example.

The model plots the periodogram of a signal composed of two sine waves, with frequencies of 1 Hz and 3 Hz. Both Sine Wave blocks have the following parameter settings:

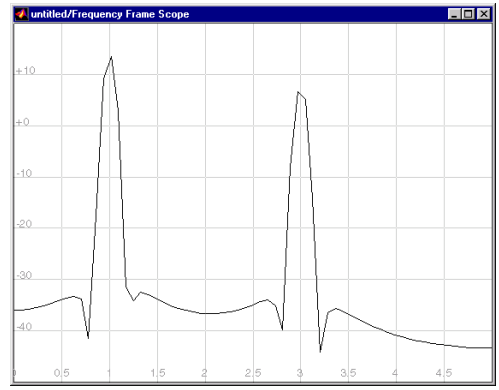
- **Sample time** = `0.1`
- **Samples per frame** = `128`

The Probe block confirms that the frame period is 12.8 seconds (128∗0.1).

No Rate Conversion. In the first case, the Short-Time FFT block uses the default settings for all parameters except for the **FFT length**, which is set to `-1`. This setting instructs the block to use the input frame size (128) as the FFT length (which is also the output size).

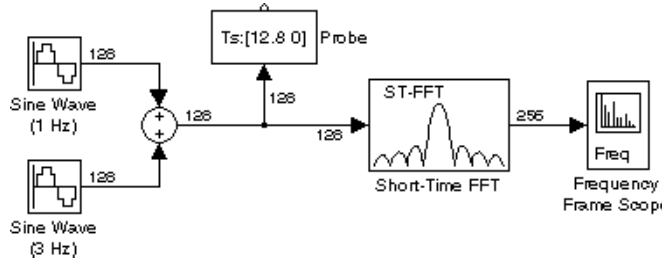


The plot generated by the Frequency Frame Scope is shown below. (The *y*-axis limits have been adjusted to better display the trace: **Minimum Y-limit** = `-50` and **Maximum Y-limit** = `20`).

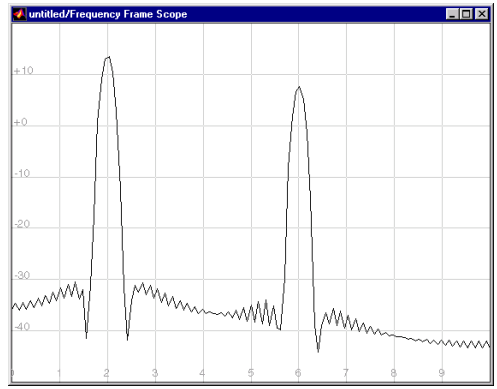


The Frequency Frame Scope uses the input frame size (128) and period (12.8) to deduce the original signal's sample period (0.1), which allows it to *correctly* display the peaks at 1 Hz and 3 Hz.

Unintended Rate Conversion. In the next case, the Short-Time FFT block uses the default settings for all parameters, including the **FFT length** setting of 256. This setting instructs the block to zero-pad the length-128 input frame to a length of 256 before performing the FFT. The line-width display on the new version of the model shows that the output of the Short-Time FFT block is now a length-256 frame.



The plot generated by the Frequency Frame Scope is shown below.



In this case, based on the input frame size (256) and period (12.8), the Frequency Frame Scope calculates the original signal's sample period to be 0.05 seconds (12.8/256), which is *wrong*. As a result, the spectral peaks appear at the incorrect frequencies, 2 Hz and 6 Hz rather than 1 Hz and 3 Hz.

The problem is that the zero-pad operation performed by the Short-Time FFT block halves the sample period of the sequence by appending 128 zeros to each frame. The Frequency Frame Scope, however, needs to know the sample period of the *original* signal. The problem is easily solved by changing the **Sample time of original time series** setting in the Frequency Frame Scope block from `-1` (auto-detect) to the actual sample period of `0.1`. The plot generated with this setting is identical to the first Frequency Frame Scope plot above.

In general, be aware that when you do zero-padding, overlapping buffering, or partial unbuffering, you are changing the sample period of the signal. As long as you keep this in mind, you should be able to anticipate and correct problems like the one above.

Understanding Matrices

One of the most powerful features of the DSP Blockset is its full support of matrix data and matrix operations. Every block in the DSP Blockset that accepts vector inputs also accepts matrix inputs. Additionally, a large number of blocks, including all of those in the Matrix Functions and Linear Algebra libraries, are provided specifically to enable sophisticated matrix-based algorithms.

A DSP Blockset matrix is the traditional rectangular array of M rows and N columns used by MATLAB,

$$\begin{bmatrix} u_{11} & u_{12} & \cdots u_{1N} \\ u_{21} & u_{22} & \cdots u_{2N} \\ \vdots & \vdots & \ddots \vdots \\ u_{M1} & u_{M2} & \cdots u_{MN} \end{bmatrix}$$

and is created and manipulated using the familiar MATLAB notation. Some common examples of MATLAB's matrix notation are shown below.

```
[1 2 3;4 5 6]      % a 2-by-3 matrix

[1 2 3;4 5 6]'     % the transpose, a 3-by-2 matrix

randn(2,3)         % a 2-by-3 matrix with random elements

[1:10;-1:-1:-10]   % a 2-by-10 matrix
```

See Chapter 4 of *Using MATLAB* for a thorough introduction to constructing and indexing matrices.

Matrices provide a versatile and efficient way to organize data, and find a variety of uses in DSP Blockset models. For example, you can use the matrix format to represent the state-space realization of a digital filter, or as a container for a collection of experimental measurements. One illustration of creative matrix usage is provided by the factorization blocks in the Linear Algebra library, which package the factors of the input matrix into a single composite output matrix. The output of the LDL Factorization block, for example, is a composite matrix containing the factorized lower triangle, diagonal, and upper triangle, as shown below.

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}

The matrix format is also ideal for the transmission and storage of multichannel signals, which are a common feature in many DSP systems. To facilitate parallel operations on multiple signal channels, the DSP Blockset recognizes two special classes of *signal-oriented* matrices:

- Sample-based matrices
- Frame-based matrices

What differentiates these matrices from each other is their contents: Sample-based matrices contain sample-based signals, while frame-based matrices contain frame-based signals. The following sections elaborate on this distinction. See "Understanding Samples and Frames," later in this chapter for a full discussion of signal-oriented matrix usage.

Sample-Based Matrices

Matrices provide a flexible alternative to vectors for transmitting multichannel sample-based signals, such as the time-varying array of pixel brightness values shown in the figure below. In sample-based signals, each matrix element represents one sample from a distinct signal channel.

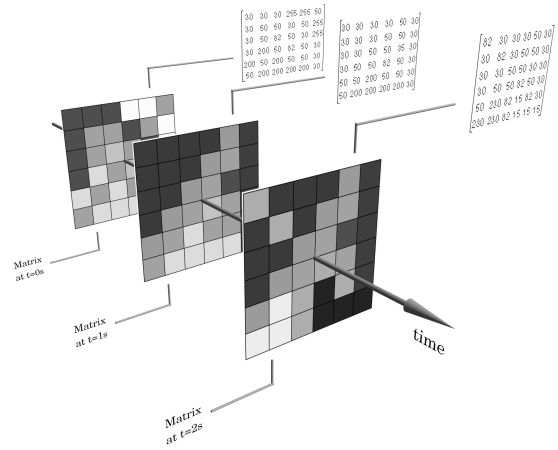
For example, if $u^{t=0}$ is the first matrix in the series, $u^{t=1}$ is the second, $u^{t=2}$ is the third, and so on, then the signal in channel 1 is composed of the following sequence:

$$u_{11}^{t=0}, u_{11}^{t=1}, u_{11}^{t=2}, \ldots$$

Similarly, in the 36-channel signal of the figure below, channel 9 (counting down the columns) contains the following sequence:

$$u_{32}^{t=0}, u_{32}^{t=1}, u_{32}^{t=2}, \ldots$$

In other words, a time-sequence of matrices are *sample-based* when the corresponding time-sequence of each element in the matrix represents the time-sequence of values in an independent signal.

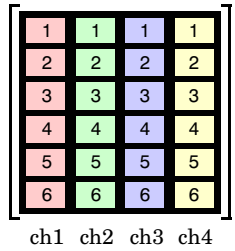


A sequence of sample matrices. Each of the 36 matrix elements represents a single pixel brightness value; each matrix is a snapshot of these 36 independent signals

Frame-Based Matrices

Matrices are also the primary vehicle for transmitting *multichannel* frame-based signals in Simulink. For frame-based signals, the M rows and N columns of the matrix represent, respectively, the M samples per frame and N channels of the multichannel signal.

This is a simple structure, as illustrated below for a sample 6-by-4 frame matrix.



Frame matrix:
4 channels,
1 frame per channel,
6 samples per frame

Consider a sequence of frame matrices, where $u^{t=0}$ is the first matrix in a series, $u^{t=1}$ is the second, $u^{t=2}$ is the third, and so on. The signal in channel 1 is the following sequence:

$$u_{11}^{t=0}, u_{21}^{t=0}, u_{31}^{t=0}, \ldots, u_{M1}^{t=0}, u_{11}^{t=1}, u_{21}^{t=1}, u_{31}^{t=1}, \ldots, u_{M1}^{t=1}, u_{11}^{t=2}, u_{21}^{t=2}, \ldots$$

Similarly, the signal in channel 3 is the following sequence:

$$u_{13}^{t=0}, u_{23}^{t=0}, u_{33}^{t=0}, \ldots, u_{M3}^{t=0}, u_{13}^{t=1}, u_{23}^{t=1}, u_{33}^{t=1}, \ldots, u_{M3}^{t=1}, u_{13}^{t=2}, u_{23}^{t=2}, \ldots$$

Matrices and Signal-Oriented Blocks

Many DSP blocks accept sample-based and frame-based matrices as inputs, or generate matrices of these types as outputs. In particular, any block that offers one or both of the following parameters expects a signal-oriented matrix as an input:

- **Frame-based inputs**
- **Number of channels**

The **Frame-based inputs** check box allows you to select whether the input is sample-based or frame-based. The **Number of channels** parameter should reflect the number of columns in the input matrix.

Some examples of blocks that accept or generate sample-based and frame-based matrices are:

- Signal From Workspace and Sine Wave, in DSP Sources
- Time Frame Scope and Signal To Workspace, in DSP Sinks
- All blocks in Filter Designs
- All blocks in Multirate Filters
- FFT, in Transforms

For more information and examples see “Understanding Samples and Frames” later in this chapter. For information about importing and exporting sample-based or frame-based matrices to the workspace, see “Working with Sources and Sinks” in Chapter 3.

Matrices and Other Blocks

Blocks that do not require sample-based or frame-based inputs typically offer a more generic dialog box parameter, such as one of those below:

- **Matrix size**
- **Number of rows**
- **Number of columns**

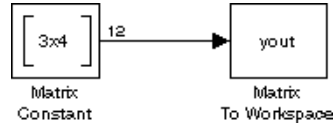
The **Matrix size** parameter is a two-element vector, `[rows columns]`, specifying the size of the input matrix. The **Number of rows** and **Number of columns** parameters are integer values specifying the number of rows and columns in the input matrix.

Specifying Matrix Dimensions

There is one important principle to keep in mind when working with matrices: *The block determines the matrix dimension*.

A matrix signal does not have an intrinsic dimension. The number of rows and columns is determined by the *matrix size* parameter settings of the block that receives the matrix as an input. (Depending on the block, the matrix size parameter may be called **Matrix size**, **Number of rows**, **Number of columns**, or **Number of channels**.)

Consider the simple model below.



When the Matrix Constant block generates the 3-by-4 matrix

$$\begin{bmatrix} 1 & 2 & 3 & 4 \\ 5 & 6 & 7 & 8 \\ 9 & 10 & 11 & 12 \end{bmatrix}$$

Simulink represents the matrix as a simple ordered set of data,

```
[1  5  9  2  6  10  3  7  11  4  8  12]
```

If the **Matrix size** parameter in the Matrix To Workspace block is then set correctly, to `[3 4]`, the constant matrix appears in the workspace with its correct 3-by-4 dimension.

However, if you set the **Matrix size** parameter to an alternate dimension, such as `[4 3]`, the following matrix is output to the workspace:

$$\begin{bmatrix} 1 & 6 & 11 \\ 5 & 10 & 4 \\ 9 & 3 & 8 \\ 2 & 7 & 12 \end{bmatrix}$$

Similarly, a **Matrix size** setting of `[2 6]` generates this output:

$$\begin{bmatrix} 1 & 9 & 6 & 3 & 11 & 8 \\ 5 & 2 & 10 & 7 & 4 & 12 \end{bmatrix}$$

In general, any **Matrix size** setting in the Matrix To Workspace block that corresponds to a matrix with a total of 12 elements is allowable. This includes vector dimensions such as `[1 12]` and `[12 1]`. Only dimensions that do not correspond to a 12-element matrix generate an error.

Note When Simulink's **Vector Line Widths** option is set, the value displayed next to a line carrying a matrix signal is the total number of elements in the matrix, 12 in this example.

Tracking Matrix Sizes

Because the matrix dimension is determined by the parameter setting in the individual receiving block, it is important to track the dimensions of matrix inputs and outputs. Make sure that the block receiving an M-by-N matrix has its matrix size parameter set appropriately. Examples of appropriate settings for an M-by-N matrix input are:

- **Matrix size** = `[M N]`
- **Number of columns** = `N`
- **Number of rows** = `M`
- **Number of channels** = `N`

If the values you specify in these parameters do not agree with the actual input matrix size, errors or unexpected results can occur.

Scalars and Vectors

Almost all matrix-processing blocks (sample-based, frame-based, and general) handle scalars and vectors as special cases of matrices. A scalar input is a 1-by-1 matrix, and a length-N vector input is either a 1-by-N or N-by-1 matrix, depending on the input dimensions you enter in the block's matrix size parameter. As mentioned above, the distinction between a column vector and a row vector is made by the individual *block* based on this parameter setting; Simulink does not have separate data formats for row and column vectors.

Using Matrices with Nonmatrix Blocks

There are a number of blocks in the blockset that are not specifically matrix-oriented, and do not contain any of the parameters discussed in the previous sections (e.g., **Matrix size** or **Number of channels**). Most of these blocks are *element-oriented,* and simply process matrices in the same way that they process scalars and vectors – each element independently. Additionally, there are a few blocks that are heavily *vector-oriented* or *scalar-oriented*, and are not intended to perform matrix operations. All three types are discussed below.

Passing Matrices to Element-Oriented Blocks

Element-oriented blocks, such as some of those in the Elementary Functions library (e.g., Complex Exponential and dB), accept inputs of all sizes, and operate on each input *element* independently. The output from an element-oriented block is always the same size as the input; scalars remain scalars, vectors remain vectors, and matrices remain matrices.

Passing Matrices to Vector-Oriented Blocks

Vector-oriented blocks, such as those in the Vector Functions library and many Simulink libraries, treat *all* inputs as vectors – including matrices. An N-by-M matrix is treated as a vector with N∗M elements. That is, the vector-oriented block treats the matrix input `u` as the vector input `u(:)`.

Passing Matrices to Scalar-Oriented Blocks

Scalar-oriented blocks, such as the Short-Time FFT and adaptive filter blocks, accept only scalar inputs. Matrix inputs cause an error.

Matrix Input and Output

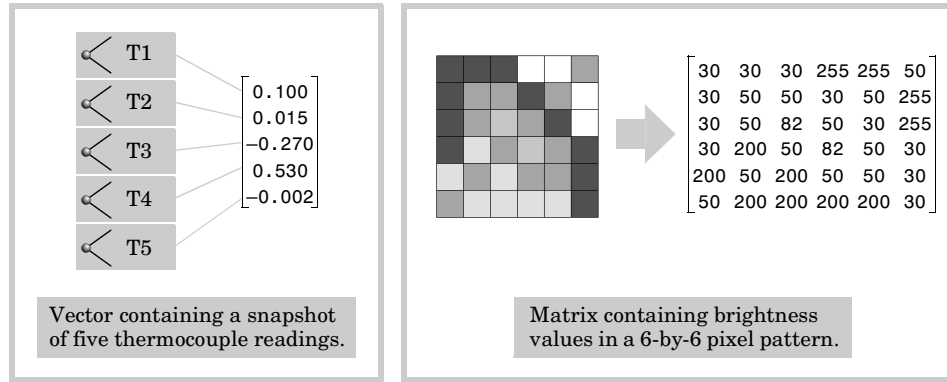
The DSP Sources and DSP Sinks libraries provide a collection of blocks that enable you to exchange matrix data with the workspace, and to display matrices on the screen. The key matrix input/output blocks are:

- Matrix From Workspace
- Matrix To Workspace
- Matrix Viewer
- Triggered Matrix To Workspace

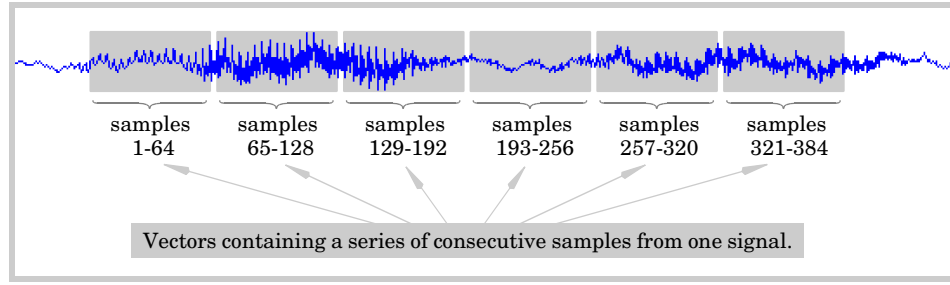
In addition to these, several other sink blocks (e.g., Signal To Workspace and Time Frame Scope) accept the frame-based matrix format. For more information on all the sources and sinks in the blockset, see "Working with Sources and Sinks" in Chapter 3.

Understanding Samples and Frames

In DSP Blockset models, vectors and matrices are often used to group scalar signal samples. For example, a length-N vector may represent a snapshot of the readings from N thermocouples. Similarly, an M-by-N matrix might represent the brightness values of an M-by-N array of pixels at a particular instant in time. In each case, the vector or matrix is used to bundle scalar samples together in a single, convenient unit.



The scalar samples in a vector or matrix may come from multiple independent signals (as shown above) or from a single signal, like the speech segment shown below.



When the scalar samples come from several independent signals, and all correspond to the same instant in time, the vector or matrix is referred to as a *sample vector* or *sample matrix*. When the scalar samples come from a single signal, and correspond to sequential points in time, the vector or matrix is referred to as a *frame vector* or *frame matrix*. A sequence of sample vectors or

sample matrices is called a *sample-based* sequence, while a sequence of frames or frame matrices is called a *frame-based* sequence.

Note The terms *sample vector*, *sample matrix*, *frame vector*, and *frame matrix* do not designate intrinsic Simulink data types like "real" and "complex." Blocks in the DSP Blockset do not *automatically* know whether a particular input is a sample matrix or a frame matrix. Instead, blocks that process sample-based inputs differently than frame-based inputs offer a check box (**Frame-based inputs**) in the parameter dialog box to specify the input type.

In summary:

- *Sample vector* – contains concurrent samples from multiple signal channels
- *Sample matrix* – contains concurrent samples from multiple signal channels
- *Frame vector* or *frame* – contains consecutive samples from a single signal
- *Frame matrix* – contains consecutive samples from multiple signal channels

The following sections explain how to work with sample-based and frame-based signals in the DSP Blockset.

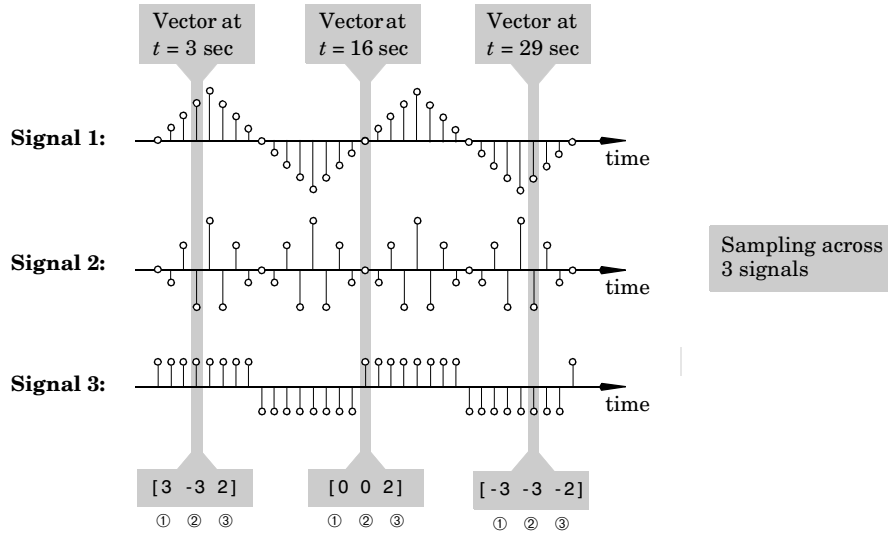
Sample Vectors and Sample Matrices

Both vectors and matrices can be used to represent collections of data points sampled at the same time. This section explains how to create sample vectors and sample matrices in Simulink, and how to import sample-based data from the workspace.

Working with Sample Vectors

A length-N sample vector represents a snapshot of N independent signals at a particular sample instant. The sample vector bundles these N *concurrent* samples into a single unit.

As an illustration, consider the three discrete signals below.



If you simultaneously sample all three signals at t=3, you acquire the following values for each:

- Signal 1: 3
- Signal 2: -3
- Signal 3: 2

Multiplexing these three concurrent samples into a single vector yields the sample vector

```
[3 -3 2]
```

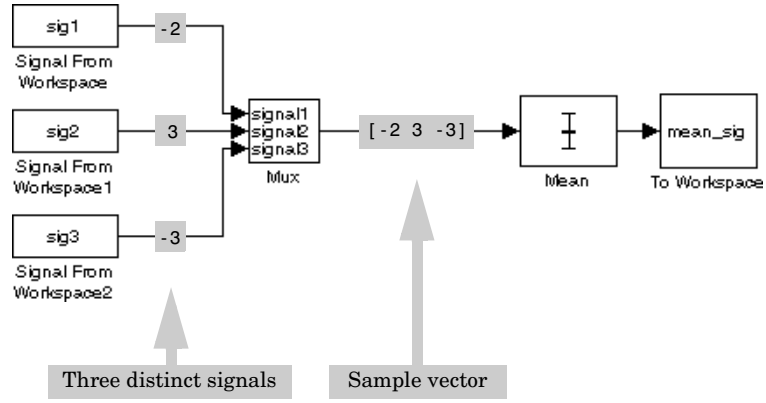
Repeating this at t=16 yields the sample vector

```
[0 0 2]
```

Each sample vector is a *snapshot* of the three signals at a particular moment in time.

Note Sample vectors are represented as row vectors.

Creating Sample Vectors in Simulink. You create a sample vector when you acquire a single scalar sample from each of N distinct signals, and multiplex the N samples together into a vector.

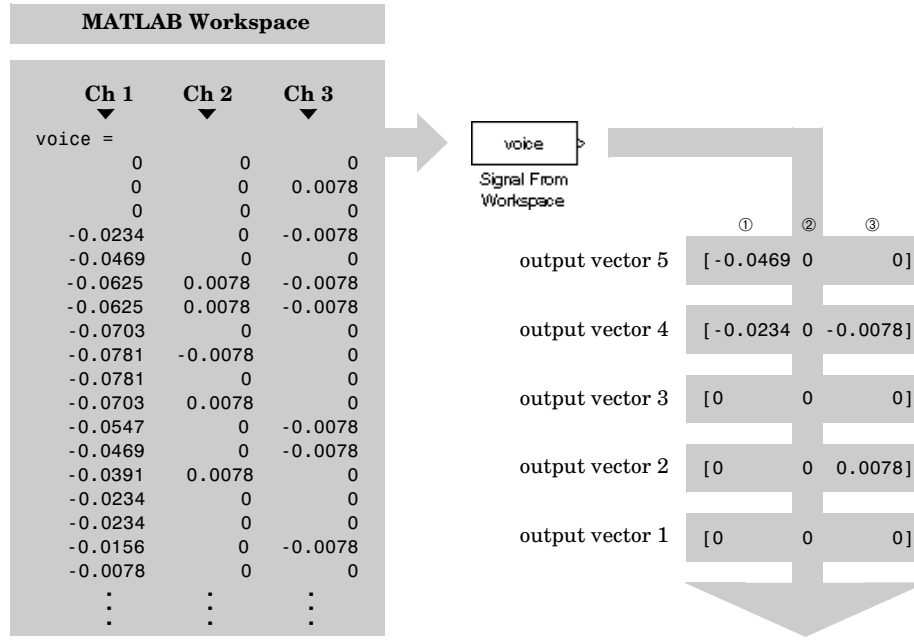


For example, the output from the Simulink Mux block in the figure above is a sample vector containing a single sample from each of the three input signals:

```
[-2 3 -3]
```

All three samples correspond to the same instant in time.

Importing Sample Vectors into Simulink. Often, the data that you acquire (or import) into Simulink from external sources is already in a sample-based format. For example, if you are modeling a communications system, you might import a few seconds of speech data from the MATLAB workspace. The figure below shows an N-by-3 workspace variable, `voice`, that contains three *channels* of voice data. When you import this data into a model using the Signal From Workspace block, the data is output row-wise, as a sequence of sample vectors.

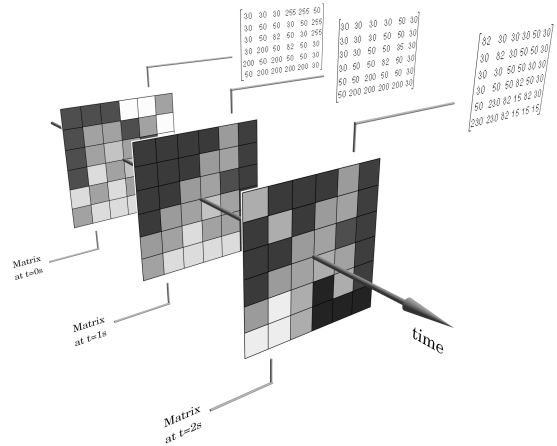


Each sample vector in the output sequence is a snapshot of the three speech signals at a particular sample time. See “Working with Sources and Sinks” in Chapter 3 for additional information about importing data.

Working with Sample Matrices

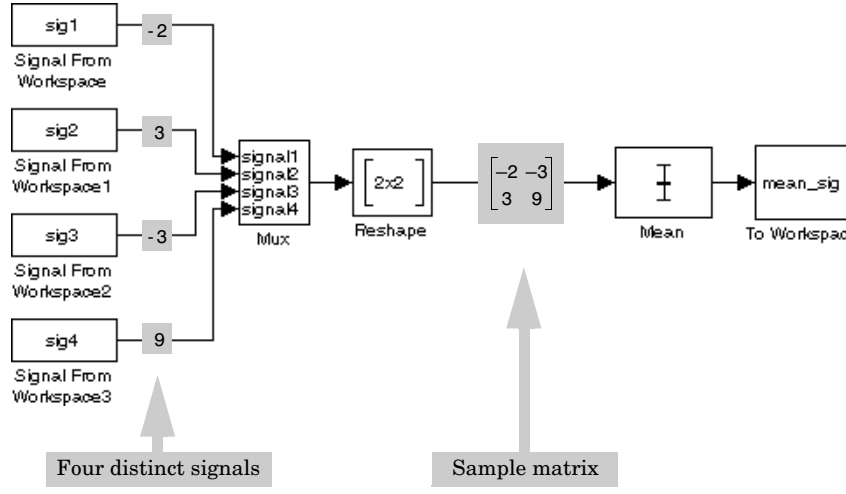
Sample-based data can also be represented by a matrix. (See also “Understanding Matrices” in this chapter.) In this case, each of the elements of the M-by-N matrix represents an independent signal channel. Just like a sample vector, a sample matrix is a snapshot of multiple (M∗N) signals at a single instant in time.

The figure below shows a sequence of three 6-by-6 matrices, each containing the instantaneous values of 36 independent signal channels. In this example, each matrix might be a collection of pixel brightness values; the value of each matrix element represents the corresponding pixel brightness at a particular instant.



A sequence of sample matrices. Each of the 36 matrix elements represents a single pixel brightness value; each matrix is a snapshot of these 36 independent signals

Creating Sample Matrices in Simulink. You create a sample matrix when you concurrently acquire a single scalar sample from each of N∗M distinct signals, and multiplex the N∗M samples together into a matrix.

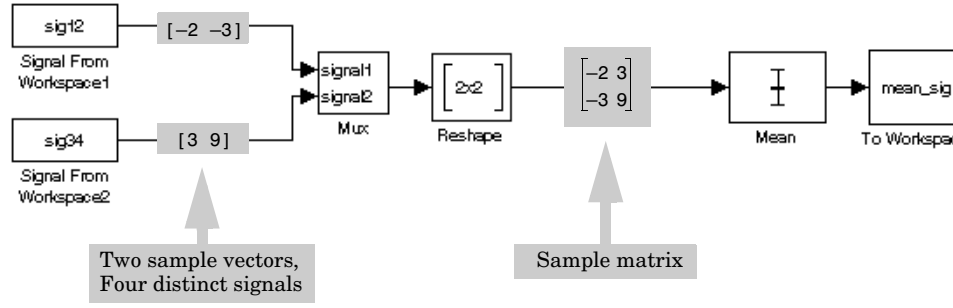


For example, the output from the Reshape block in the figure above is a sample matrix containing a single sample from each of the four input signals,

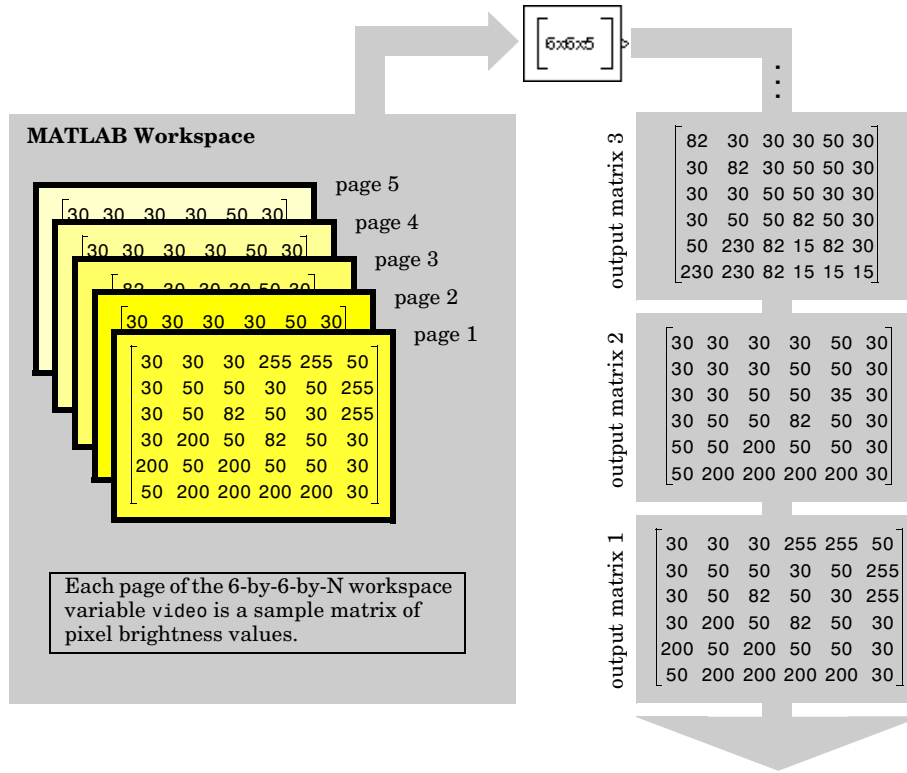
$$\begin{bmatrix} -2 & -3 \\ 3 & 9 \end{bmatrix}$$

All four samples in the matrix correspond to the same instant in time. (The Reshape block used in the model is simply an aid to tracking and verifying matrix sizes, and does not actually rearrange the values in the signal. See "Specifying Matrix Dimensions" in "Understanding Matrices" for additional information about the matrix format.)

You can also create a sample matrix by multiplexing together *concurrent* sample vectors, as shown below. In both cases, the resulting matrix represents a snapshot of concurrent scalar signals.



Importing Sample Matrices into Simulink. Data that you acquire (or import) into Simulink from external sources may already be in sample-based format. For example, you might import a sequence of matrices representing a video segment. The figure below shows a 6-by-6-by-N workspace variable, `video`, that contains 36 (6 times 6) independent *channels* of data. When you import this data into a model using the Matrix From Workspace block, the data is output page-wise, as a sequence of sample matrices.



Each matrix in the output sequence contains a snapshot of the video picture at one sample instant. See "Working with Sources and Sinks" in Chapter 3 for additional information about importing data.

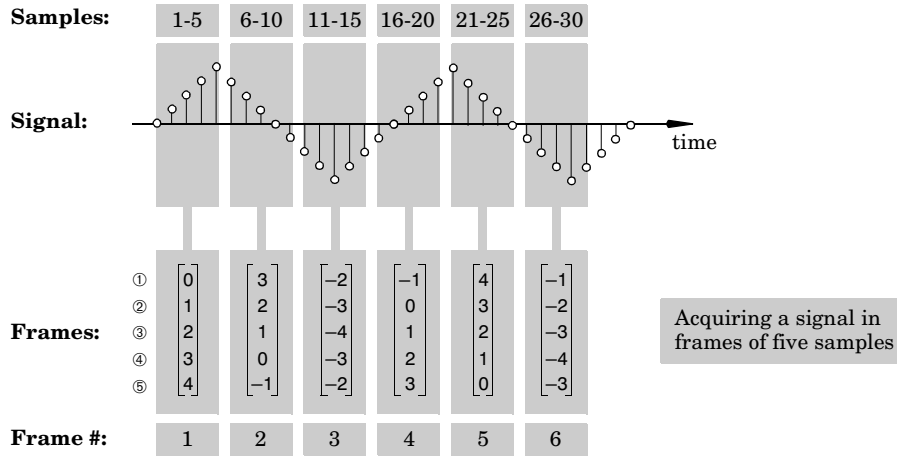
Frames and Frame Matrices

Frame vectors and frame matrices are used to represent collections of data points sampled at consecutive times. This section explains how to create frame vectors and frame matrices in Simulink, and how to import frame-based data from the workspace.

Working with Frame Vectors (Single-Channel Signals)

A length-M frame vector represents the evolution of one signal over M sample times. The frame bundles these M *consecutive* samples into a single unit.

As an illustration, consider the discrete signal below.



Acquiring a signal in frames of five samples

If you acquire five signal samples, beginning with the sixth sample, you see the following sequence:

- Sample 6: 3
- Sample 7: 2
- Sample 8: 1
- Sample 9: 0
- Sample 10: -1

Buffering these five consecutive samples into a single vector yields the *frame vector*

$$\begin{bmatrix} 3 \\ 2 \\ 1 \\ 0 \\ -1 \end{bmatrix}$$

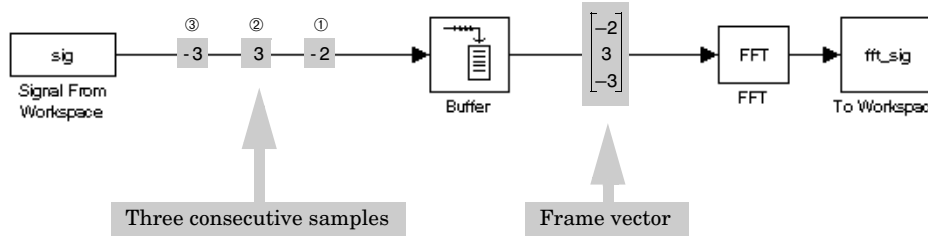
Repeating this for samples 11 to 15 yields the frame vector

$$\begin{bmatrix} -2 \\ -3 \\ -4 \\ -3 \\ -2 \end{bmatrix}$$

Each 5-sample frame represents a contiguous 5-sample segment of the signal.

Note Frame vectors are represented as column vectors.

Creating Frames in Simulink. You create a frame vector when you acquire M consecutive scalar samples from a single signal, and buffer them into a length-M vector.



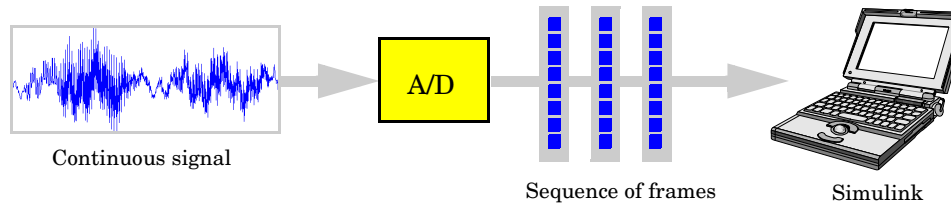
For example, the output from the Buffer block in the figure above is a frame containing three consecutive samples from a single signal,

$$\begin{bmatrix} -2 \\ 3 \\ -3 \end{bmatrix}$$

The three samples correspond to three consecutive sample times. Note that the first frame element, `u(1)`, is always the earliest sample; the last frame element, here `u(3)`, is always the newest sample. In other words, new samples enter the frame vector at the bottom, and are pushed upwards in the frame as later samples are added.

A more efficient way to create the frame-based signal shown above is to set the **Samples per frame** parameter of the Signal From Workspace block to 3. The Signal From Workspace block then performs the buffering internally, and directly generates the frame-based signal; the separate Buffer block is not needed.

Importing Frames into Simulink. Often the data that you acquire (or import) into Simulink from external sources is already in a frame representation. For example, in a real-time system, the analog-to-digital (A/D) converter usually buffers a large segment of the sampled signal before releasing the acquired samples to the host computer (see "Benefits of Frame-Based Processing"). Each such buffer that the A/D hardware propagates to the model is a frame.



Each frame contains a contiguous block of the sampled signal.

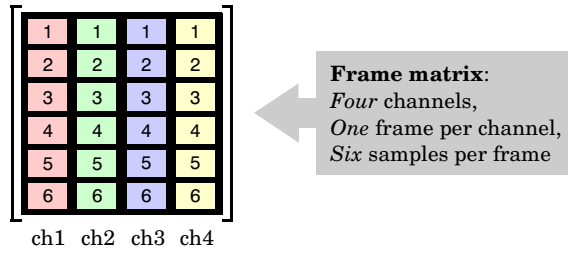
Working with Frame Matrices (Multichannel Signals)

The frame vectors described in the previous section represent consecutive samples from a single signal – one channel of data. You can represent multiple channels of data by using a *frame matrix*.

An M-by-N frame matrix allows you to bundle N channels of frame-based signal data into a single unit. Each of the N matrix columns contains a single frame from one of the signal channels. Each of the M matrix rows therefore contains *one sample* from each of the N signal channels. So:

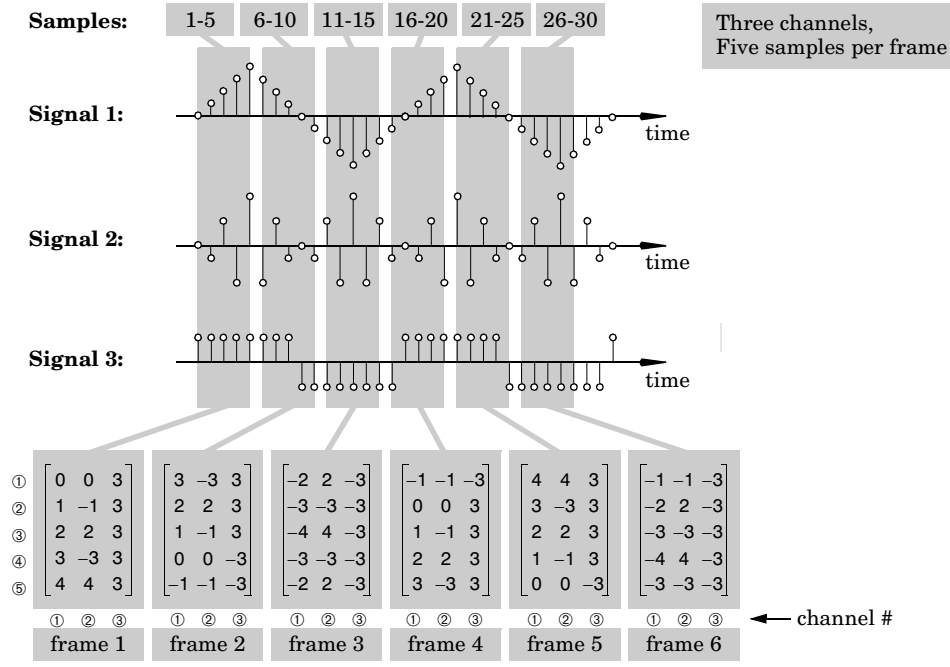
- Each matrix column is a frame vector.
- Each matrix row is a sample vector.

For example, a 6-by-4 frame matrix is structured like this:



There are four channels (columns) in this matrix, each containing six sequential samples (numbered 1 to 6, above). The four samples in each row all correspond to the same sample instant. The first row, `u(1,:)`, contains the earliest set of samples; the last row, `u(6,:)`, contains the newest set of samples. In other words, new samples enter each frame at the bottom, and are pushed upwards in the frame as later samples are added.

As an illustration, consider the three discrete signals below.



If you sample all three signals over five consecutive sample times, beginning with the first sample time, you see the following set of frames:

	Signal 1	**Signal 2**	**Signal 3**
t=0	0	0	3
t=1	1	–1	3
t=2	2	2	3
t=3	3	–3	3
t=4	4	4	3

Multiplexing these three frames into a single matrix yields the *frame matrix*

$$\begin{bmatrix} 0 & 0 & 3 \\ 1 & -1 & 3 \\ 2 & 2 & 3 \\ 3 & -3 & 3 \\ 4 & 4 & 3 \end{bmatrix}$$

Repeating this for samples 6 to 10 yields the frame matrix

$$\begin{bmatrix} 3 & -3 & 3 \\ 2 & 2 & 3 \\ 1 & -1 & 3 \\ 0 & 0 & -3 \\ -1 & -1 & -3 \end{bmatrix}$$

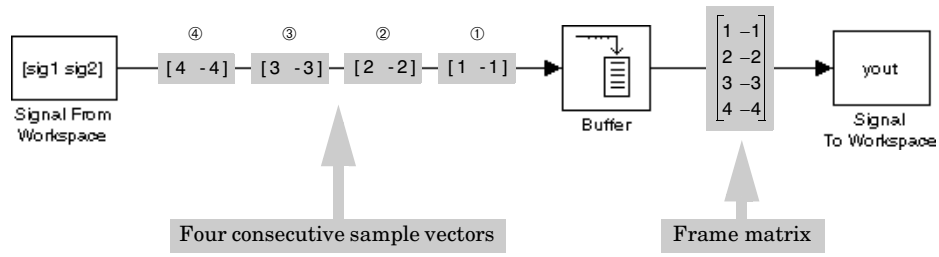
Each frame matrix is a block of consecutive signal samples from three distinct channels.

Creating Frame Matrices in Simulink. There are two complementary ways to create frame matrices in Simulink. To create an M-by-N frame matrix:

- Buffer M consecutive length-N sample vectors into the rows of the matrix

  or
- Multiplex N concurrent length-M frames into the columns of the matrix

The model below illustrates the first approach, which creates a 4-by-2 frame matrix by buffering a series of four consecutive sample vectors (each containing two channels).



The Signal From Workspace block imports the two channel signal `[sig1 sig2]`, where

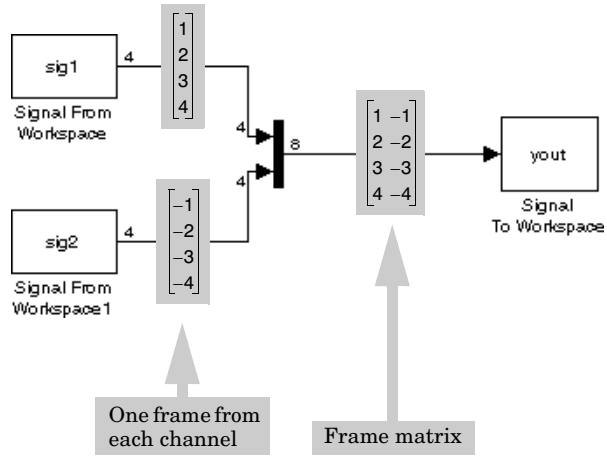
```
sig1 = 1:10'
sig2 = -1:-1:-10'
```

and outputs one 2-channel sample vector at each sample time. The **Frame-based** check box of the Signal To Workspace block is selected, and the **Number of channels** parameter is set to `2`. The **Buffer size** setting in the Buffer block is `4`, so the output is a frame matrix containing the four consecutive sample vectors,

$$\begin{bmatrix} 1 & -1 \\ 2 & -2 \\ 3 & -3 \\ 4 & -4 \end{bmatrix}$$

(The Signal From Workspace block can also import data directly into the frame matrix format, as shown in the next section.)

The second approach is to create the 4-by-2 frame matrix by multiplexing together two 4-sample frames (one from each channel).

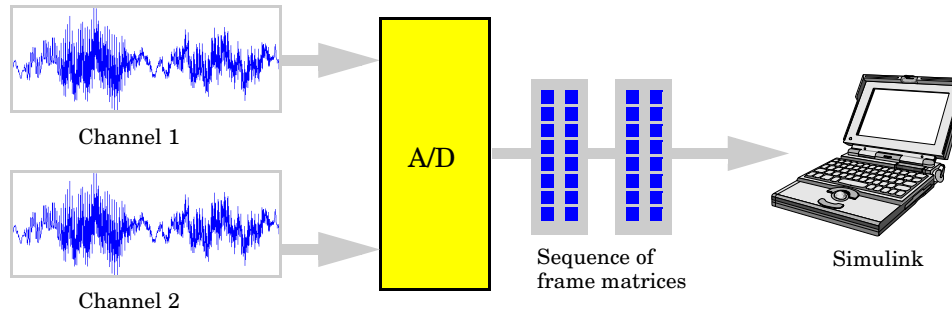


The **Samples per frame** parameter of each Signal From Workspace block is now set to `4`, so that each imports the target signal in frames of four samples. The signals are the same as in the previous example:

```
sig1 = 1:10'
sig2 = -1:-1:-10'
```

The Mux block combines the two frames into a 4-by-2 frame matrix with the same result as the previous example. Note that you could also import both channels of data as a frame matrix using a single Signal From Workspace block.

Importing Frame Matrices into Simulink. Often, the data that you acquire (or import) into Simulink from external sources is already in a frame matrix representation. For example, in a real-time system, a multichannel A/D card usually buffers a segment of the sampled signals before releasing the acquired samples to the host computer (see "Benefits of Frame-Based Processing"). Each such buffer that the A/D hardware propagates to the model is a frame matrix.



Frame matrices can also be imported directly from the workspace using the Signal From Workspace block. See “Working with Sources and Sinks” in Chapter 3 for more information.

Exporting and Displaying Frame Matrices. A number of blocks allow you to output multichannel frame-based signals to the workspace, file, screen, or sound device. All can be found in the DSP Sinks library:

- FFT Frame Scope
- Frequency Frame Scope
- Matrix Viewer
- Signal To Workspace
- Time Frame Scope
- To Wave Device (two channels)
- To Wave File (two channels)
- Triggered Signal To Workspace
- User-Defined Frame Scope

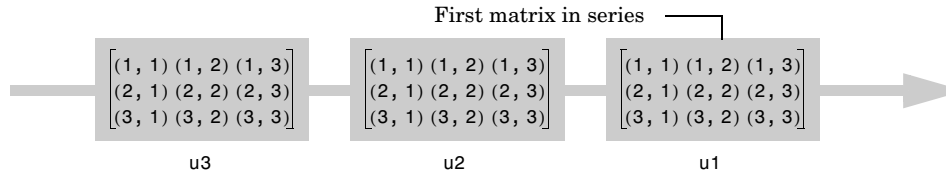
The Matrix Viewer block provides general matrix display capabilities that can be used with all matrices, including multichannel frame matrices. The To Wave Device and To Wave File blocks are limited to one-channel (mono) or two-channel (stereo) inputs, selectable in the **Stereo** check box. The other blocks in the list require you to specify a value for the **Number of channels** in the input matrix.

Understanding Multichannel Signal Processing

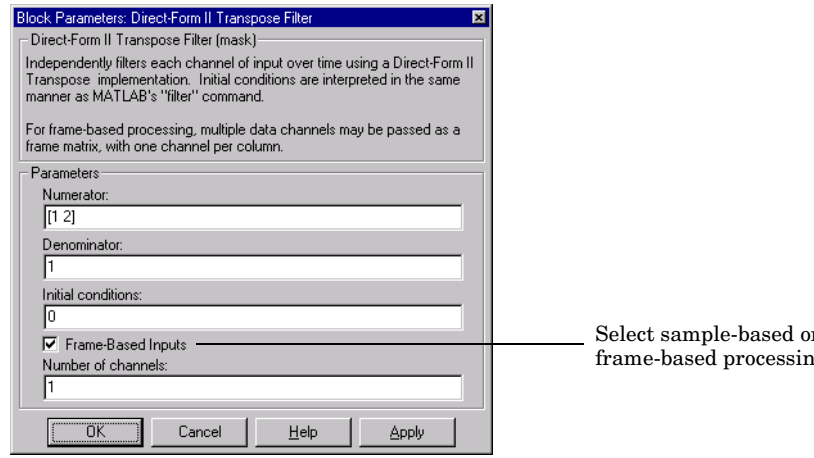
Most signal-oriented blocks in the DSP Blockset are capable of processing both sample matrices and frame matrices.

- *Sample matrices* – M-by-N matrices that represent an array of M*N independent samples
- *Frame matrices* – M-by-N matrices that represent N concurrent frames, each from a distinct signal, and each containing M consecutive samples

Consider a series of matrix inputs, `u1`, `u2`, `u3`, each of size 3-by-3.

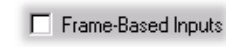


A matrix-oriented block like Direct-Form II Transpose Filter can interpret this input sequence in two different ways, depending on the setting of the **Frame-based inputs** check box.



- *Sample-based processing*

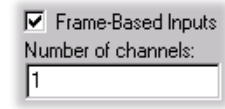
  When the **Frame-Based Inputs** option is not selected, the block views the matrix as a sample matrix, treating



the elements of each matrix input as *concurrent* samples from nine different signals; so, for example, `u1(1,1)` and `u1(2,3)` are samples from two distinct signals that correspond to the same moment in time.

- *Frame-based processing*

  When the **Frame-based inputs** option is selected, the block views the matrix as a frame matrix, treating each column as a frame of consecutive samples from a distinct channel (sampled over the same time period). The **Number of channels** parameter indicates the number of columns in the matrix.

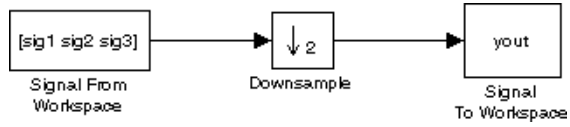


In both sample-based and frame-based modes, vector and scalar inputs are treated as special matrix cases. See “Scalars and Vectors” in “Understanding Matrices” earlier in this chapter. The following sections provide some examples of common sample-based and frame-based block operations. See the “DSP Block Reference” chapter for complete information about any particular block.

Example 1: Sample-Based Operation with Vector Input

The Downsample block can process both sample-based and frame-based signals. To see how sample-based processing works, build the model below using:

- Signal From Workspace block from the DSP Sources library
- Downsample block from the Signal Operations library (in General DSP)
- Signal To Workspace block from the DSP Sinks library



To try the model:

1 Create `sig1`, `sig2`, and `sig3` in the workspace by typing the following commands.

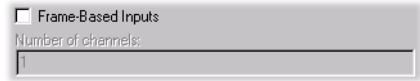
```
sig1 = (1:1000)';
sig2 = zeros(1000,1);
sig3 = (-1:-1:-1000)';
```

2 Double-click the Signal From Workspace block and enter

```
[sig1 sig2 sig3]
```

in the **Signal** parameter. This creates a length-3 sample vector, representing three signal channels. The **Samples per frame** parameter should remain set to 1.

3 Double-click on the Downsample block and enter 2 for the **Downsample factor** parameter. Leave the **Frame-based inputs** parameter unchecked so that the block will operate in sample-based mode.



4 Set the **Stop time** in the **Simulation Parameters** dialog box to 20, and start the simulation by selecting **Start** from the **Simulation** menu.

If you look at output `yout` in the command window, you can see that the block treats each element of the input vector as a separate channel, and downsamples the signal in each channel over time.

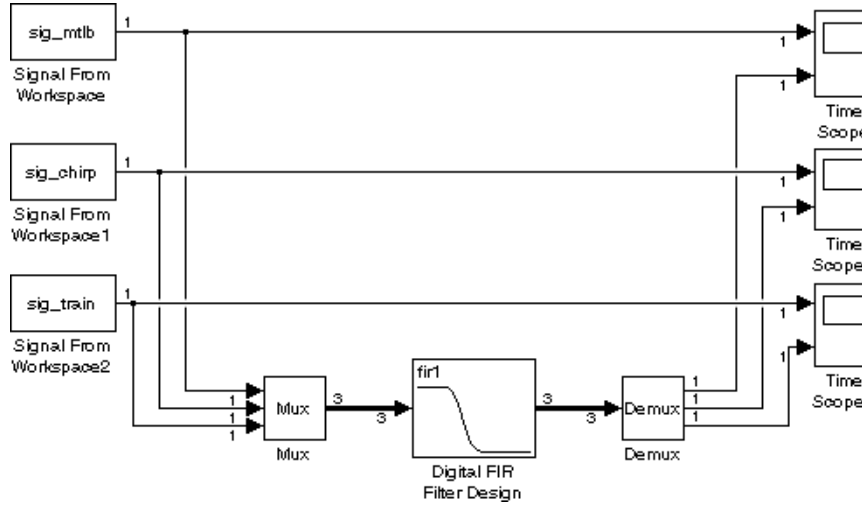
```
yout =
     0     0     0
     1     0    -1
     3     0    -3
     5     0    -5
     7     0    -7
     9     0    -9
    11     0   -11
    13     0   -13
    15     0   -15
    17     0   -17
    19     0   -19
```

Each downsampled output signal is half the length of the corresponding input signal, and there is no interaction between the three channels (i.e., between the three elements in each input vector).

Example 2: Sample-Based Operation with Vector Input

Let's look at how the Digital FIR Filter Design block processes a vector input in sample-based mode. Build the model shown below using:

- Signal From Workspace block from the DSP Sources library
- Mux block from the Simulink Signals & Systems library
- Digital FIR Filter Design block from the Filter Designs library (in Filtering)
- Demux block from the Simulink Signals & Systems library
- Time Scope block from the DSP Sinks library



To try the model:

1 Load three signals into the workspace, and rename them as shown.

```
load mtlb;
sig_mtlb = mtlb;

load chirp;
sig_chirp = y;

load train;
sig_train = y;
```

2. Double-click on each Signal From Workspace block in turn and type in one of the signal names for the **Signal** parameter:
   - `sig_mtlb`
   - `sig_chirp`
   - `sig_train`

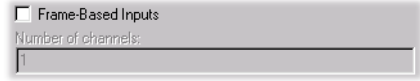
   Enter `1/Fs` for the **Sample time** parameter of each Signal From Workspace block. (`Fs` is the original sample frequency of the `chirp` and `train` signals, and equals 8192 Hz. It is also fairly close to the original sample frequency of the `mtlb` signal, which is 7418 Hz. For convenience, `Fs` is used for all three signals in this example.)

   When the three signals are multiplexed together (using the Simulink Mux block), the result is a 3-element vector, or 3-by-1 matrix.

3. Double-click on each Time Scope block in turn and press the **Properties** button,  ,to set the scope properties:
   - Click the **General** tab, and set the **Number of axes** parameter to `2`.
   - Click the **Data history** tab, and uncheck the **Limit rows to last** parameter.

   Close the **Properties** windows by pressing **OK**, but leave the scope window open.

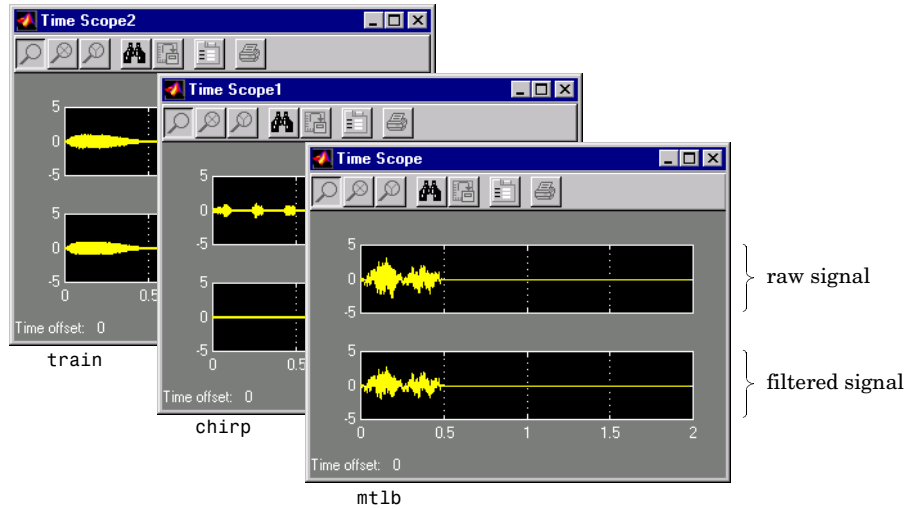
4. Double-click on the Digital FIR Filter Design block, and verify that the **Frame-based inputs** check box is *not* selected.



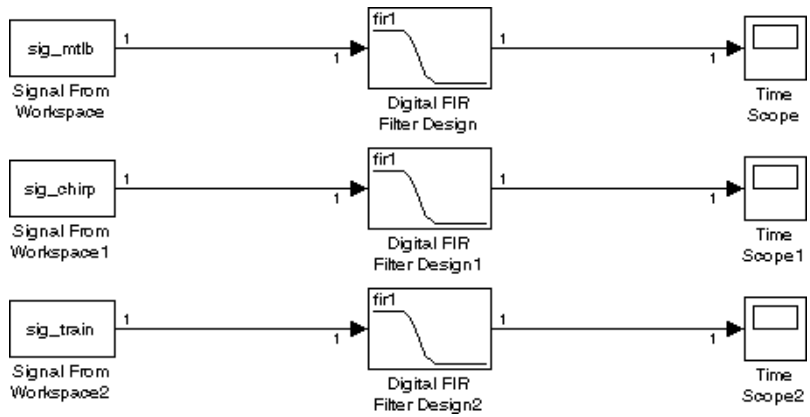
5. Set the **Stop time** to `2` in the **Simulation Parameters** dialog box (available through the **Simulation** menu).

6. Start the simulation by selecting **Start** from the **Simulation** menu.

As the simulation runs, you can see that the Digital FIR Filter Design block is filtering each of the three signals independently; there is no interaction between `sig_mtlb`, `sig_chirp`, and `sig_train` (i.e., between the three elements in each input vector).



The operation is equivalent to independently filtering the three signals by using three distinct blocks, as shown below.

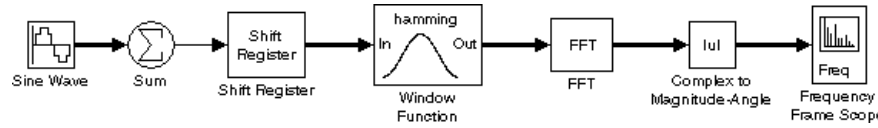


Note What happens if you check **Frame-based inputs** in the Digital FIR Filter Design block dialog box, and set the **Number of channels** parameter to 3 (the number of signals)? The result is the same as filtering the three signals independently. As a general rule, frame-based processing is identical to sample-based processing when the **Number of channels** parameter is equal to the number of individual elements in the input.

Example 3: Frame-Based Operation with Vector Input

The FFT block is another good example of a frame-based block. To see how this block processes a frame vector input, build the model below using:

- Sine Wave block from the DSP Sources library
- Sum block from the Simulink Math library
- Shift Register block from the Buffers library (in General DSP)
- Window Function block from the Signal Operations library (in General DSP)
- FFT block from the Transforms library (in General DSP)
- Complex to Magnitude-Angle block from the Simulink Math library
- Frequency Frame Scope block from the DSP Sinks library



To try the model:

1. Double-click the Sine Wave block enter the following parameter values
   a. **Amplitude** = `[1 2]`
   b. **Frequency** = `[100 25]`
   c. **Sample time** = `0.001`
   d. **Samples per frame** = `1`
2. Double-click on the Sum block and enter `1` for the **List of signs** parameter.

3 Double-click on the Shift Register block and set the **Register size** to `256`.

4 Double-click on the Window Function block and set the **Operation** parameter to **Apply window to input**.

5 Double-click on the Complex to Magnitude-Angle block and set the **Output** parameter to **Magnitude**.

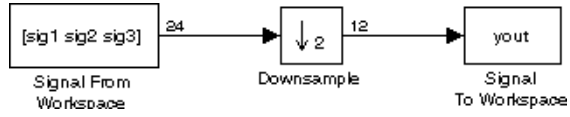
6 Double-click on the Frequency Frame Scope block and set the **Frequency units** to **Hertz**. Set the **Sample time of original time series** to `0.001`. Verify that the **Number of input channels** is set to `1`, as below.

7 Set the **Stop time** in the **Simulation Parameters** dialog box to `inf`, and start the simulation by selecting **Start** from the **Simulation** menu.

The Shift Register block buffers the sinusoidal signal into a frame vector, which is then windowed by the Window Function block. The FFT block operates on the frame-based data as a single unit at each time step. Note that because the Shift Register block implements an overlapping buffer, the sample period of the original signal must be explicitly specified in the **Sample time of original time series** parameter of the Frequency Frame Scope block.

Example 4: Frame-Based Operation with Matrix Input

Use the model from Example 1 to test the Downsample block with a multichannel input in frame-based mode.



If you have already built the previous example's model, skip to step 2 below:

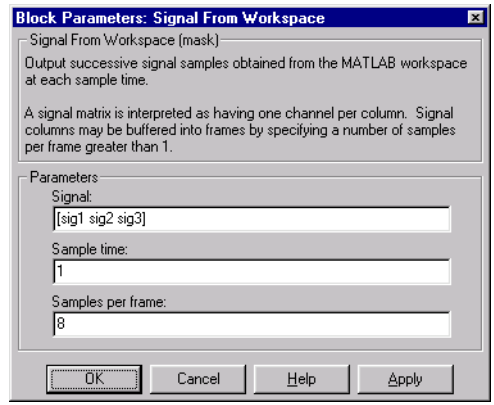
1 Create `sig1`, `sig2`, and `sig3` in the workspace (if you haven't already) by typing the following commands.

```
sig1 = (1:1000)';
sig2 = zeros(1000,1)';
sig3 = (-1:-1:-1000)';
```

2 Double-click on the Signal From Workspace block and enter

```
[sig1 sig2 sig3]
```

for the **Signal** parameter to define the three channel signal. Then set the **Samples per frame** parameter to 8 to establish a frame size of 8 for each channel.



3 Double-click on the Signal To Workspace block and set the **Number of channels** parameter to 3 to match the signal.

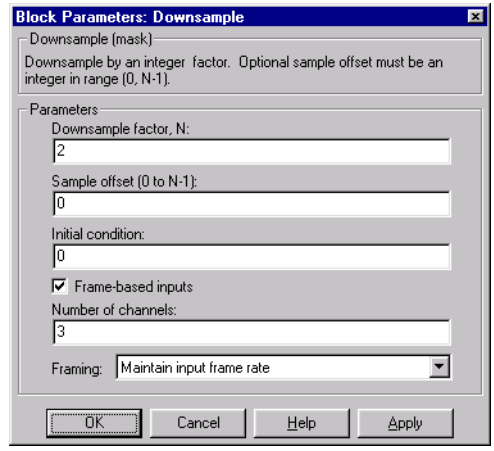
4 Double-click on the Downsample block and make the following settings:

a Enter 2 for the **Downsample factor** parameter.

b Check the **Frame-based inputs** parameter so that the block will operate in frame-based mode.

c Set the **Number of channels** to 3 to match the signal.

d Set the **Framing** parameter to **Maintain input frame rate**. This will allow the block to downsample the signal by adjusting the frame size.



See “Rate Conversion” earlier in this chapter for more information about these parameters.

5 Select **Vector Line Widths** from the **Format** menu to activate the vector line width display.

6 Set the **Stop time** in the **Simulation Parameters** dialog box to 20, and start the simulation by selecting **Start** from the **Simulation** menu

When the simulation begins running, the model’s line widths reveal that the matrix output of the Downsample block contains half the number of elements of the input (12 vs. 24), as expected for decimation by a factor of two. Check the

output, `yout`, to verify that the block independently downsamples each channel over time:

```
yout =

     0     0     0
     0     0     0
     0     0     0
     0     0     0
     1     0    -1
     3     0    -3
     5     0    -5
     7     0    -7
     9     0    -9
    11     0   -11
    13     0   -13
    15     0   -15
```

After the first few samples of delay, you can see that each channel is downsampled by a factor of two; every other sample has been removed. The four-sample delay (zeros) at the start of the sequence reflects the latency inherent in frame-based models. See "Benefits of Frame-Based Processing" for more information.

Benefits of Frame-Based Processing

Frame-based processing is an established method of accelerating both real-time systems and simulations.

Accelerating Real-Time Systems

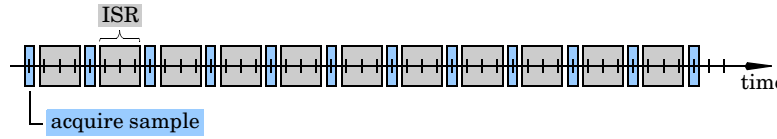
Framed data is a common format in real-time systems, where the data acquisition hardware often operates by accumulating a large number of signal samples at a high rate, and propagating these samples to the real-time system as a block of data. This maximizes the efficiency of the system by distributing fixed process overhead across many samples; the "fast" data acquisition is interrupted by "slow" interrupt processes after each frame is acquired, rather than after each individual sample.

The figure below illustrates how throughput is increased by frame-based data acquisition. The thin blocks each represent the time elapsed during acquisition

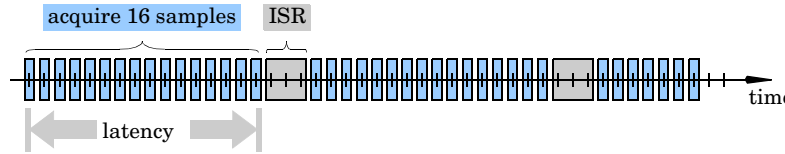
of a sample. The thicker blocks each represent the time elapsed during the interrupt service routine (ISR) that reads the data from the hardware.

In this example, the frame-based operation acquires a frame of 16 samples between each ISR. The frame-based throughput rate is therefore many times higher than the sample-based alternative.

Sample-based operation:



Frame-based operation:



It's important to note that frame-based processing introduces a certain amount of latency into a process due to the inherent lag in buffering the initial frame. In most instances, however, it is possible to select frame sizes that improve throughput without creating unacceptable latencies.

Accelerating Simulations

Simulation also benefits from frame-based processing. In this case, it is the overhead of block-to-block communications that is reduced by propagating frames rather than individual samples.

Increasing Performance

Simulation performance can become a concern in any model of sufficient complexity. In addition to model-specific adjustments, such as altering algorithms or eliminating scope displays, there are a number of general steps you can take to improve the performance of any model.

To begin with, you should familiarize yourself with "Improving Simulation Performance and Accuracy" in Chapter 4 of *Using Simulink*. Several additional options for improving performance are listed below:

- Use frame-based processing wherever possible. It is advantageous for the entire model to be frame-based. See "Understanding Samples and Frames" earlier in this chapter for more information.
- Turn off the Simulink status bar by deselecting the **Status bar** option in the **View** menu. Simulation speed will improve, but the time indicator will not be visible.
- Run simulations from the MATLAB command line by typing

  ```
  sim(gcs)
  ```

  This can greatly increase the simulation speed. However, there are several limitations attached to this method of launching a simulation:
  - You cannot interact with the simulation (to tune parameters, for instance).
  - You must press **Ctrl-C** to stop the simulation.
  - There are no graphics updates in M-file S-functions.
- Use the Real-Time Workshop to generate generic real-time (GRT) code targeted to your host platform, and simulate the model using the generated executable file. See the *Real-Time Workshop User's Guide* for more information.

3

Using the DSP Blockset

Overview

This chapter discusses some of the fundamental classes of blocks in the DSP Blockset. The following topics are covered:

- **Working with Filter Designs**: Highlights the features of the filter design blocks available in the Filter Designs library.
- **Working with Windows**: Explains how to use the Window Function block to generate and apply a variety of windows.
- **Working with Buffers**: Describes the process of converting a signal between sample-based and frame-based representations.
- **Working with Sources and Sinks**: Explains how to import data from the MATLAB workspace during a simulation, how to export simulation results at the end of a simulation, and how to display signals on the screen.
- **Working with Statistical Operations**: Describes the fundamentals of working with the blockset's statistical function blocks.

The discussion and examples included in these sections should help you become familiar and comfortable with the standard operations involved in building and simulating models. See Chapter 2, "Simulink and the DSP Blockset," for more conceptual information on sample rates, matrices, and frame-based processing.

Working with Filter Designs

Filtering is one of the most important operations in signal processing, and is supported in the DSP Blockset with the blocks in the Filter Designs, Filter Realizations, Adaptive Filters, and Multirate Filters libraries (all in the Filters top-level library). As an introduction to filtering, this section discusses the basics of using the blocks in the Filter Designs library to design FIR and IIR filters in a number of configurations.

Filter Designs Blocks

Each block in the Filter Designs library accepts high-level filter specifications as its parameters. Based on these specifications, the block designs the appropriate filter when you close the dialog box, so that the frequency response plot on the icon matches the filter specifications. The block saves the filter coefficients and applies the filter to the input data. In each case, the block's output is the filtered time-sequence.

The Filter Designs library contains six blocks, which can be grouped into three categories:

- *Classical Discrete Time*

  The Digital FIR Filter Design and Digital IIR Filter Design blocks design and implement discrete-time filters with standard band configurations (highpass, lowpass, bandpass, or bandstop). These are classical IIR and linear phase FIR filters, with Butterworth, Chebyshev type I, Chebyshev type II, and elliptic designs.
- *Classical Continuous Time*

  The Analog Filter Design block designs and implements Butterworth, Chebyshev type I, Chebyshev type II, and elliptic filters in standard band configurations.
- *Special Discrete Time*

  The Remez FIR Filter Design, Yule-Walker IIR Filter Design, and Least Squares FIR Filter Design blocks design and implement IIR or FIR filters with arbitrary magnitude responses, including multiband responses.

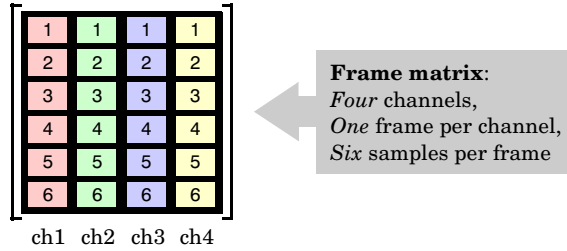
	Analog	Digital FIR	Digital IIR	Remez FIR	Least Squares FIR	Yule-Walker IIR
Bands	Lowpass	Lowpass	Lowpass	Multiband	Multiband	Multiband
	Highpass	Highpass	Highpass			
	Bandpass	Bandpass	Bandpass			
	Bandstop	Bandstop	Bandstop			
Designs	Butterworth		Butterworth	Hilbert Trans	Hilbert Trans	
	Chebyshev I		Chebyshev I	Differentiator	Differentiator	
	Chebyshev II		Chebyshev II			
	Elliptic		Elliptic			

All of the blocks in the Filter Designs library are built on the filter design capabilities of the Signal Processing Toolbox. For details on any of the filter design topics discussed here, see the *Signal Processing Toolbox User's Guide*.

Frame-Based Processing

All of the discrete-input blocks provide frame-based processing capability. This means that the blocks can simultaneously apply the designed filter to multiple channels of a frame-based signal. Multichannel frame-based signals are represented in matrix form, as described in "Understanding Matrices" and "Understanding Samples and Frames" in Chapter 2

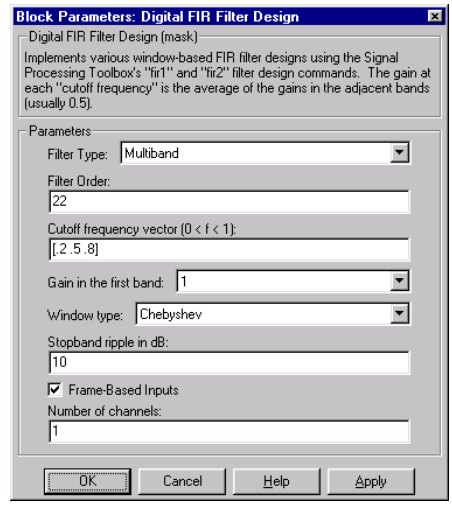
The figure below shows an example of a four-channel frame matrix.

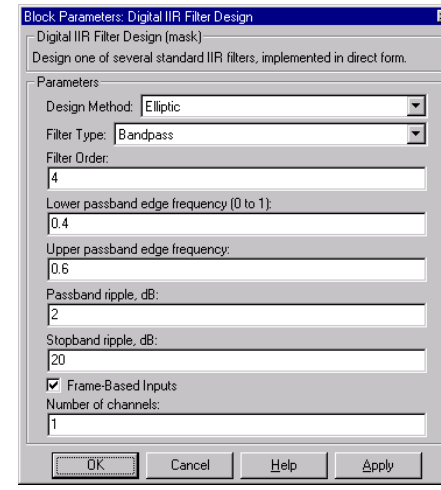


Each column in the above matrix is an independent signal channel containing six sequential samples (numbered 1 to 6 in the figure above). The four samples in each row all correspond to the same sample instant: The first row, `u(1,:)`, contains the earliest set of samples; the last row, `u(6,:)`, contains the newest set of samples. The filter design blocks apply the specified filter independently to the signal data in each channel. The output is a 6-by-4 matrix containing the four independently filtered signal channels.

Classical IIR and FIR Filters, Discrete Time

The Digital FIR Filter Design and Digital IIR Filter Design blocks design and implement discrete-time filters with standard band configurations.





All of the digital filter designs let you specify a filter order. The other available parameters depend on the **Filter type** and band configuration, as shown below, where f_{n0} = cutoff frequency, f_{n1} = lower cutoff frequency, f_{n2} = upper cutoff frequency, R_p = passband ripple in decibels, and R_s = stopband attenuation in decibels.

Configuration	FIR	Butterworth	Chebyshev I	Chebyshev II	Elliptic
Lowpass	f_{n0}	f_{n0}	f_{n0}, R_p	f_{n0}, R_s	f_{n0}, R_p, R_s
Highpass	f_{n0}	f_{n0}	f_{n0}, R_p	f_{n0}, R_s	f_{n0}, R_p, R_s
Bandpass	f_{n1}, f_{n2}	f_{n1}, f_{n2}	f_{n1}, f_{n2}, R_p	f_{n1}, f_{n2}, R_s	f_{n1}, f_{n2}, R_p, R_s
Bandstop	f_{n1}, f_{n2}	f_{n1}, f_{n2}	f_{n1}, f_{n2}, R_p	f_{n1}, f_{n2}, R_s	f_{n1}, f_{n2}, R_p, R_s

For all of the classical discrete-time filter design blocks, frequency parameters use normalized units. The unit frequency is the Nyquist frequency (half the sample frequency), so the band edge frequencies are always in the range `[0 1]`.

The Digital IIR Filter Design block uses a direct-form II transposed representation. It inherits its sample rate from the driving block, and accepts only discrete-time inputs.

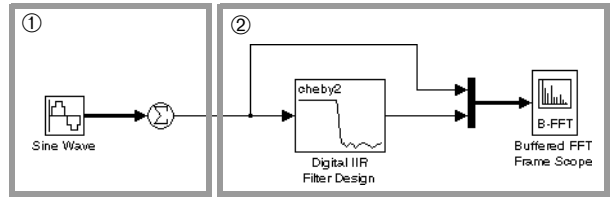
Both of these blocks use Signal Processing Toolbox functions to design the filter:

- The FIR design uses the toolbox function `fir1`.
- The Butterworth design uses the toolbox function `butter`.
- The Chebyshev type I design uses the toolbox function `cheby1`.
- The Chebyshev type II design uses the toolbox function `cheby2`.
- The elliptic design uses the toolbox function `ellip`.

For more information on the filter design algorithms, see the *Signal Processing Toolbox User's Guide*.

Example: Chebyshev Type II Lowpass Filter

Create a model like the one shown below. The model generates a signal composed of two sine waves at different frequencies, and then removes the higher frequency component with a lowpass filter.



① Construct a signal made up of two sinusoids, one at a low frequency and one at a higher frequency.

② Pass the signal through a lowpass filter to attenuate the higher frequency sinusoid. Display the original and filtered signals.

Construct the model using:

- Sine Wave block from the DSP Sources library
- Sum block from the Simulink Linear library
- Digital IIR Filter Design block from the Filter Designs library
- Mux block from the Simulink Signals & Systems library
- Buffered FFT Frame Scope block from the DSP Sinks library

To try the model:

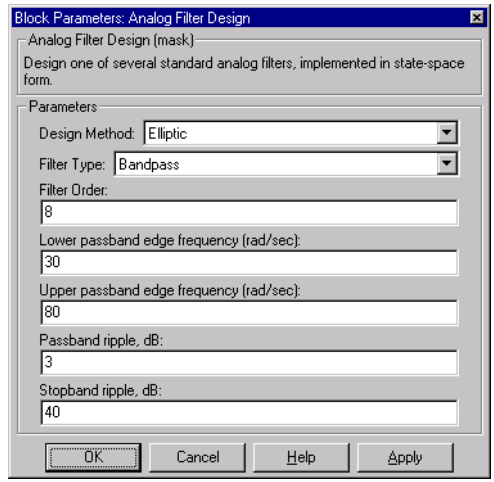
1. Create a signal made up of two sinusoids (at 100 Hz and 400 Hz) by entering a two-element vector in the **Frequency** field of the Sine Wave block:
   a. **Amplitude** = `10`
   b. **Frequency** = `[100 400]`
   c. **Phase** = `0`
   d. **Sample time** = `0.001`
   e. **Samples per frame** = `1`
2. Set the Sum block's **List of signs** parameter to `1`.
3. Verify that the Buffered FFT Frame Scope block is set to inherit the sample period of the input, and to plot two channels of data:
   a. **Sample time of original time series** = `-1`
   b. **Number of input channels** = `2`
4. In the Digital IIR Filter Design dialog, specify a Chebyshev type II lowpass filter with a **Stopband edge frequency** of 250 Hz, to attenuate the sinusoid at 400 Hz but retain the sinusoid at 100 Hz. Note that the Nyquist frequency in this case is 500 Hz (half the sample frequency), so that the normalized **Stopband edge frequency** is `0.5`.
   a. Select **Chebyshev II** from **Design Method**.
   b. Select **Lowpass** from **Filter Type**.
   c. **Filter Order** = `8`
   d. **Stopband edge frequency** = `0.5`
   e. **Stopband ripple** = `20`
5. Set the **Stop time** in the **Parameters** dialog box to `inf`, and start the simulation by selecting **Start** from the **Simulation** menu.
6. As the simulation begins running, right-click in the plot area of the scope, and select **Autoscale** from the pop-up menu. You can also change the styles

and colors of the plotted lines by selecting either **CH 1** or **CH 2** from the right-click pop-up menu.

The Buffered FFT Frame Scope window displays both the original signal and the filtered result, which shows the expected attenuation of the 400 Hz component. Stop the simulation at any time by selecting **Stop** from the **Simulation** menu.

Classical IIR Filters, Continuous Time

The Analog Filter Design block designs and applies continuous-time IIR filters with standard band configurations.



All of the analog filter designs let you specify a filter order. The other available parameters depend on the filter type and band configuration, as shown below, where ω_0 = cutoff frequency, ω_1 = lower cutoff frequency, ω_2 = upper cutoff frequency, R_p = passband ripple in decibels, and R_s = stopband ripple in decibels.

Configuration	Butterworth	Chebyshev I	Chebyshev II	Elliptic
Lowpass	ω_0	ω_0, R_p	ω_0, R_s	ω_0, R_p, R_s
Highpass	ω_0	ω_0, R_p	ω_0, R_s	ω_0, R_p, R_s
Bandpass	ω_1, ω_2	ω_1, ω_2, R_p	ω_1, ω_2, R_s	ω_1, ω_2, R_p, R_s
Bandstop	ω_1, ω_2	ω_1, ω_2, R_p	ω_1, ω_2, R_s	ω_1, ω_2, R_p, R_s

For all of the analog filter designs, frequency parameters are in units of radians per second.

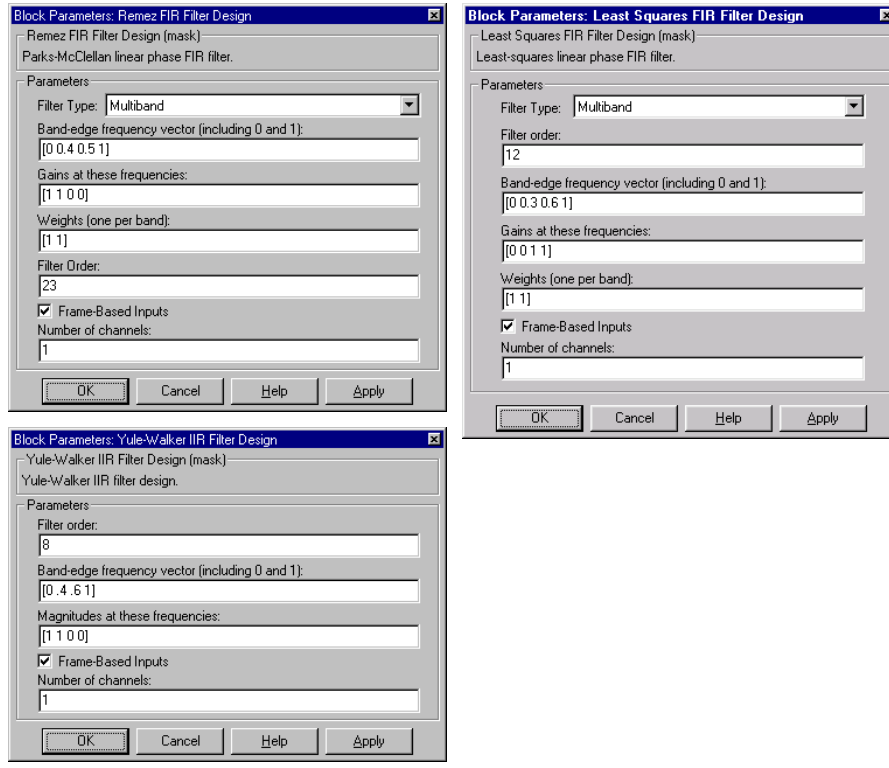
The block uses a state-space filter representation, and applies the filter using the State-Space block in the Simulink Continuous library. All of the design methods use Signal Processing Toolbox functions to design the filter:

- The Butterworth design uses the toolbox function `butter`.
- The Chebyshev type I design uses the toolbox function `cheby1`.
- The Chebyshev type II design uses the toolbox function `cheby2`.
- The elliptic design uses the toolbox function `ellip`.

For more information on the filter design algorithms, see the *Signal Processing Toolbox User's Guide*.

Special IIR and FIR Filters, Discrete-Time

The Remez FIR Filter Design, Yule-Walker IIR Filter Design, and Least Squares FIR Filter Design blocks design and implement IIR or FIR filters with arbitrary magnitude responses, including multiband responses.



All of these blocks automatically apply the designed filter to an input. Each incorporates the Direct-Form II Transpose Filter block from the Filter Realizations library, which yields the same results as the `filter` function in the Signal Processing Toolbox.

Filter Design Characteristics

Yule-Walker IIR Filter Design. The Yule-Walker IIR Filter Design block designs recursive IIR digital filters by fitting a specified frequency response based on arbitrary piecewise linear magnitude responses.

For more on the Yule-Walker algorithm, see the description of the `yulewalk` function in the *Signal Processing Toolbox User's Guide*.

Remez FIR and Least Squares FIR Filter Design. The Remez FIR Filter Design and Least Squares FIR Filter Design blocks design FIR filters using the Parks-McClellan and least-squares techniques, respectively. These techniques reflect two error minimization schemes that provide optimal fits to a desired frequency response, each using a different definition of "optimal fit."

The Remez FIR Filter Design implements the Parks-McClellan algorithm, which uses the Remez exchange algorithm and Chebyshev approximation theory to design filters with optimal fits between the desired and actual frequency responses. The filters are optimal in the sense that they minimize the maximum error between the desired frequency response and the actual frequency response over the designated bands. Filters designed in this way exhibit an equiripple behavior in their frequency response, and hence are also known as *equiripple* filters.

The Least Squares FIR Filter Design block minimizes the integral of the squared error between the desired frequency response and the actual frequency response. This technique provides a better response over most of the passband and stopband than does the Parks-McClellan algorithm. At the band edges, however, the least-squares technique provides a poorer fit than does an equiripple filter designed to fit the same response.

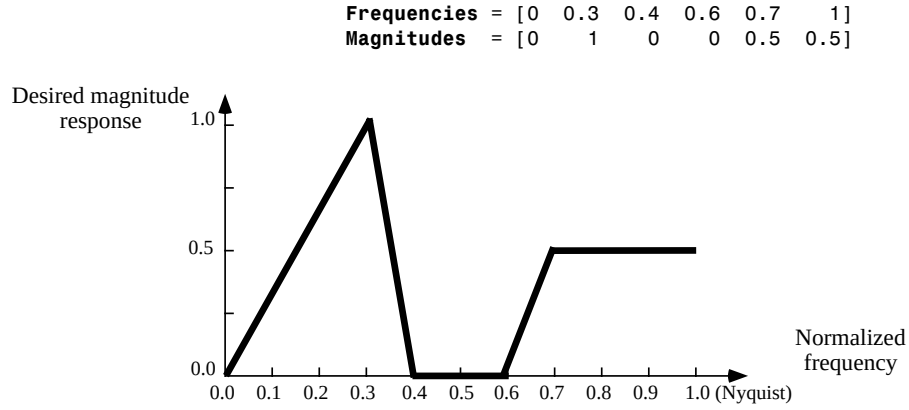
For more on the Parks-McClellan and least squares design techniques, see the descriptions of the `remez` and `firls` functions, respectively, in the *Signal Processing Toolbox User's Guide*.

Frequency and Magnitude Parameters

All of these blocks let you design filters with any magnitude response. The response can include multiple stopbands, passbands, and transition regions. You specify the desired frequency response using the blocks' **Band edge frequency vector** and **Magnitudes at these frequencies** parameters. These parameters specify the range and magnitude, respectively, of the frequency bands that make up the filter's frequency response.

Think of frequency bands as lines over short frequency intervals. The blocks use this scheme to represent any piecewise linear function. A simple bandpass example is

- **Band edge frequency vector** = `[0 0.3 0.4 0.6 0.7 1]`, which specifies the desired frequency points.
- **Magnitudes at these frequencies** = `[0 1 0 0 0.5 0.5]`, which defines the magnitudes corresponding to the frequencies above.



Together, the **Band edge frequency vector** and **Magnitudes at these frequencies** parameters shown define:

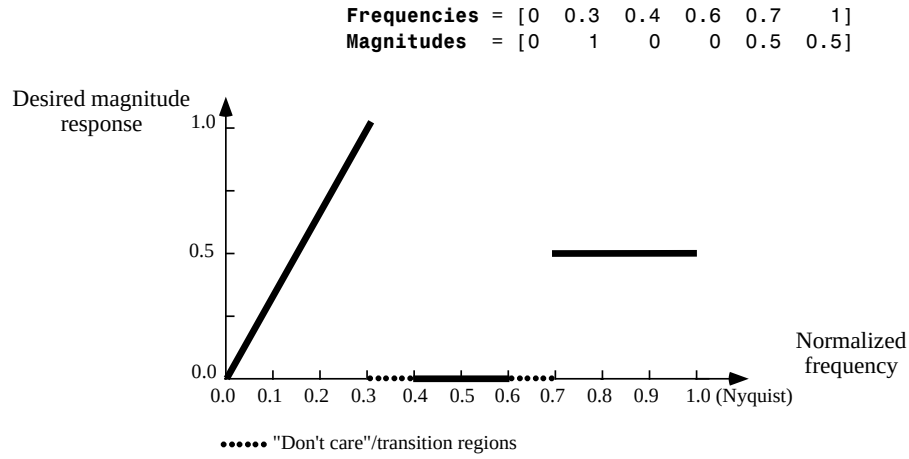
- A stopband, from `0.4` to `0.6`
- A passband from `0.7` to `1`
- Three transition regions: `0` to `0.3`, `0.3` to `0.4`, and `0.6` to `0.7`

The **Band edge frequency vector** contains points in the range `0` to `1`, where `1` corresponds to half the sampling frequency (the Nyquist frequency). The **Magnitudes at these frequencies** vector contains the desired magnitude response at the points in the **Band edge frequency vector**. The two vectors must be the same length.

For the Yule-Walker IIR Filter Design block, the **Band edge frequency vector** and **Magnitudes at these frequencies** parameters must start with `0` and end with `1`, and describe a piecewise linear magnitude response over the entire

frequency range, as shown in the previous figure. In this case, the “transition regions” are linear segments connecting the defined bands.

For the Remez FIR Filter Design and Least Squares FIR Filter Design blocks, the **Band edge frequency vector** and **Gains at these frequencies** vectors describe linear magnitude *segments*, as shown below. The distances between segments represent “don’t care” or transition regions. Both vectors must have even length.



Weight Parameters

The Remez FIR Filter Design and Least Squares FIR Filter Design blocks allow you to weight the error minimization in certain frequency bands by entering a vector for the band **Weights**. The **Weights** parameter is useful when designing a compound filter (for example, a lowpass differentiator). For example, to specify a lowpass filter with a transition region in the normalized frequency range 0.4 to 0.5, and 10 times more error minimization in the stopband than the passband, use:

- **Band edge frequency vector** = `[0 0.4 0.5 1]`
- **Magnitudes at these frequencies** = `[1 1 0 0]`
- **Weights** = `[1 10]`

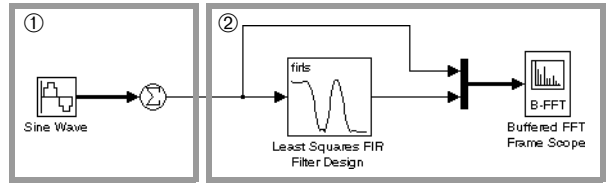
The **Weights** vector is always half the length of the **Band edge frequency vector** and **Magnitudes at these frequencies** vectors; there must be exactly one weight per band.

Differentiator Weights. The differentiator designs use special weighting techniques for nonzero magnitude bands. The Remez FIR Filter Design block assumes that the weight is equal to the inverse of the frequency multiplied by the weight specified in the **Weights** vector. The Least Squares FIR Filter Design block assumes that the weight is equal to the inverse of the frequency squared, multiplied by the weight specified in the **Weights** vector. In each case, the result is a filter with much better fit at low frequencies than at high frequencies. In most cases, however, differentiators have only a single band, so the weight is a scalar value that does not affect the final filter.

Hilbert Transform Weights. The Hilbert transform designs apply a constant weight in each nonzero magnitude band, simply multiplying the error by the specified weight for that band. Similar to the differentiators, Hilbert transformers usually have only a single band, so the weight is a scalar value that does not affect the final filter.

Example: Least Squares Multiband Filter

To try a multiband filter, create a model like the one shown below. You can create this easily by modifying the Chebyshev type II lowpass filter model in the previous example.



① Construct a signal made up of sinusoids at 100, 200, 300, and 400 Hz.

② Pass the signal through a multiband filter to attenuate the sinusoids at 200 and 400 Hz. Display both signals.

Construct the model using:

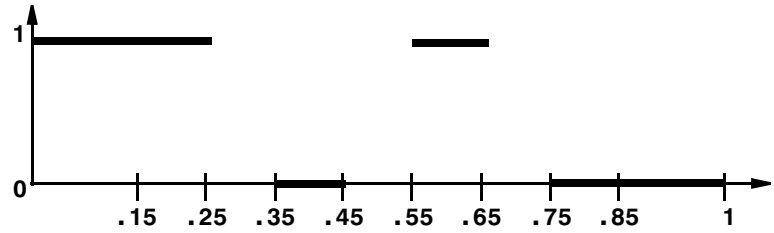
- Sine Wave block from the DSP Sources library
- Sum block from the Simulink Linear library
- Least Squares FIR Filter Design block from the Filter Designs library
- Mux block from the Simulink Signals & Systems library
- Buffered FFT Frame Scope block from the DSP Sinks library

To try the model:

1. Create a signal made up of four sinusoids by entering a four-element vector in the **Frequency** field of the Sine Wave block. The sample rate will be 1 kHz:
   a. **Amplitude** = `10`
   b. **Frequency** = `[100 200 300 400]`
   c. **Phase** = `0`
   d. **Sample time** = `0.001`
2. Set **List of signs** in the Sum block to `1`.
3. Verify that the Buffered FFT Frame Scope block is set to inherit the sample period of the input, and to plot two channels of data:
   a. **Sample time of original time series** = `-1`
   b. **Number of input channels** = `2`
4. Set the Least Squares FIR Filter Design block parameters to attenuate the sinusoids at 200 Hz and 400 Hz with a multiband filter:
   a. **Filter type** = **Multiband**
   b. **Filter order** = `16`
   c. **Band edge frequency vector** =
      `[0 125 175 225 275 325 375 500]/500`

      (For a sample rate of 1 kHz, divide by `500`, the Nyquist frequency.)
   d. **Gains at these frequencies** = `[1 1 0 0 1 1 0 0]`
   e. **Weights** = `[1 1 1 1]`

These **Band edge frequency vector** and **Gains at these frequencies** vectors define a desired magnitude response that looks like this:



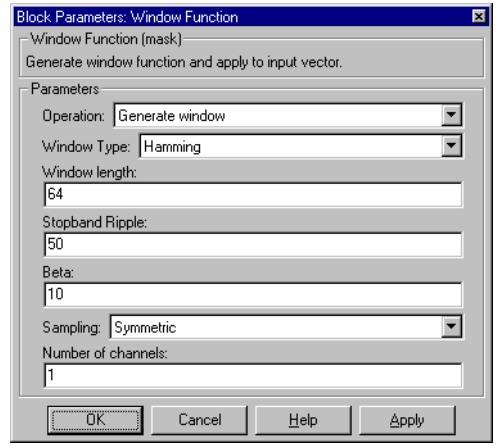
5 Set the **Stop time** in the **Parameters** dialog box to `inf`, and start the simulation by selecting **Start** from the **Simulation** menu.

6 As the simulation begins running, right-click in the plot area of the scope, and select **Autoscale** from the pop-up menu. You can also change the styles and colors of the plotted lines by selecting either **CH 1** or **CH 2** from the right-click pop-up menu.

The Buffered FFT Frame Scope block displays the FFTs of both the original signal, which has four peaks, and the filtered signal, which has two peaks. Stop the simulation at any time by selecting **Stop** from the **Simulation** menu. Try different band configurations to attenuate the peaks at different frequencies.

Working with Windows

Windowing is a common technique used in digital signal processing, and is supported in the DSP Blockset library with the Window Function block in the Signal Operations library.



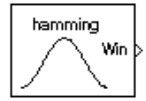
The Window Function block provides three modes of operation:

- Generate and output the coefficients of a selected window.
- Apply an existing window to the input signal.
- Generate the coefficients of a selected window and apply the window to the input signal. Output both the window coefficients and the windowed signal.

The parameters in the block dialog box (above) change to reflect the options available for the particular mode of operation and the selected window. Note that for any combination of these settings, there are always *some* inactive parameters.

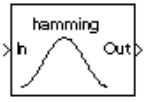
Generating a Window

Open the Window Function block's dialog box, and select **Generate window** from the **Operation** pop-up menu. Note that the block icon now has only a single output port, at which the block generates the specified window vector.



Applying a Window

Open the Window Function block's dialog box, and select **Apply window to input** from the **Operation** pop-up menu. Note that the block icon now has a single input port and a single output port.



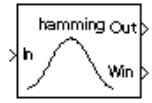
The block multiplies the specified window, `w`, element-wise with an input vector of the same length, `u`:

```
y = w .* u      % equivalent MATLAB code
```

The output from the block is the result of this multiplication.

Generating and Applying a Window

Open the Window Function block's dialog box, and select **Generate and apply window** from the **Operation** pop-up menu. Note that the block icon now has a single input port and two output ports.



The block generates the product of the window and the input, `w.*u`, at the top output (`Out`), and provides the computed window, `w`, at the bottom output (`Win`).

Window Specifications

The type of window to generate and/or apply is specified by the **Window type** parameter. The length of the sampled window, N_w, is specified by the **Window length** parameter, and must be the same size as the input, if there is one.

The **Sampling** parameter determines whether the generalized-cosine windows (**Blackman**, **Hamming**, and **Hanning**) are computed in a *periodic* or a *symmetric* manner. For example, if **Sampling** is set to **Symmetric**, a **Hamming** window of **Window length** N_w is computed as

```
w = hamming(Nw)     % symmetric (aperiodic) window
```

If **Sampling** is set to **Periodic**, the same window is computed as

```
w = hamming(Nw+1)
w = w(1:Nw)         % periodic (asymmetric) window
```

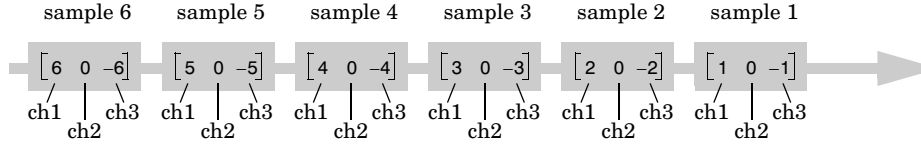
See the reference page for the Window Function block and the *Signal Processing Toolbox User's Guide* for details on all of the blockset's windowing capabilities.

Working with Buffers

The buffering blocks in the Buffering library (in General DSP) are the key to converting between sample-based and frame-based signals, and between different frame sizes. The blocks provide three general classes of buffering operations:

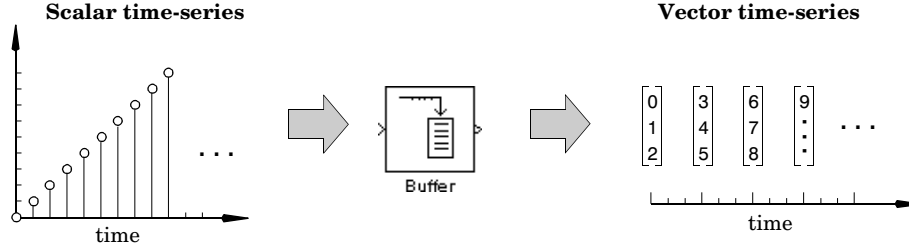
- Buffering
- Rebuffering
- Unbuffering

Buffering. *Buffering* transforms a sample-based sequence of data into a frame-based sequence of data. Each input to a buffering operation is a sample vector, a collection of scalar data points from multiple channels. The three-channel sample-based sequence below provides an example of such a signal.



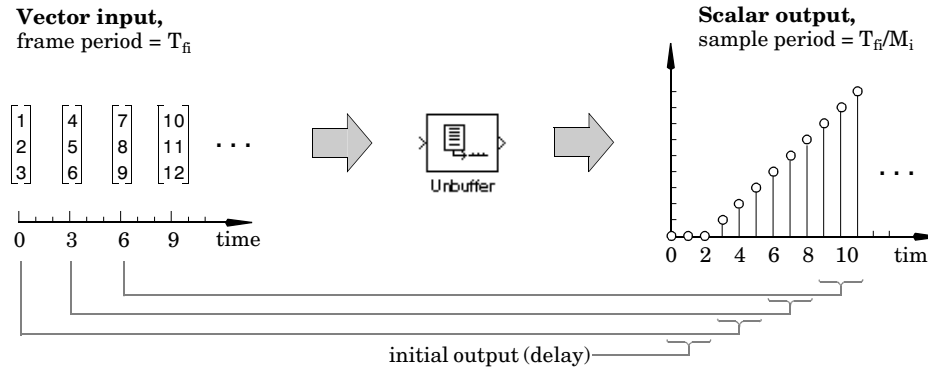
In the simplest (nonoverlapping) case, the buffering operation collects M_o successive vector inputs (each containing N independent samples), and outputs the buffered sequence as an M_o-by-N matrix. This process is repeated continuously to generate a matrix output with a frame period M_o times longer than the input period. The sample period of the sequence, T_s, is not altered by a nonoverlapping buffering operation.

The figure below illustrates a one-channel buffering operation for M_o=3.



Rebuffering. *Rebuffering* converts a signal from one frame size to another, which is a common operation in frame-based models. The input to the rebuffering operation is the standard frame-based M_i-by-N matrix, which contains M_i samples from each of N independent channels. The samples contained in the input matrix are rebuffered to a frame size of M_o, resulting in an M_o-by-N output matrix. In a rebuffering operation with no overlap, the frame period of the output may be longer or shorter than the input, but the sample period of the sequence, T_s, always remains the same.

Unbuffering. *Unbuffering* is the inverse of the sample-to-frame buffering process, and generates a sample-based output from a frame-based input. The sample-based output signal has a sample rate M_i times higher than the frame-based input, where M_i is the input frame size. The figure below shows the one-channel unbuffering operation for M_i=3.



The blocks listed below are the primary buffering blocks in the DSP Blockset.

Block	Library
Buffer	Buffers, in General DSP
Partial Unbuffer	Buffers, in General DSP
Rebuffer	Buffers, in General DSP
Shift Register	Buffers, in General DSP
Unbuffer	Buffers, in General DSP

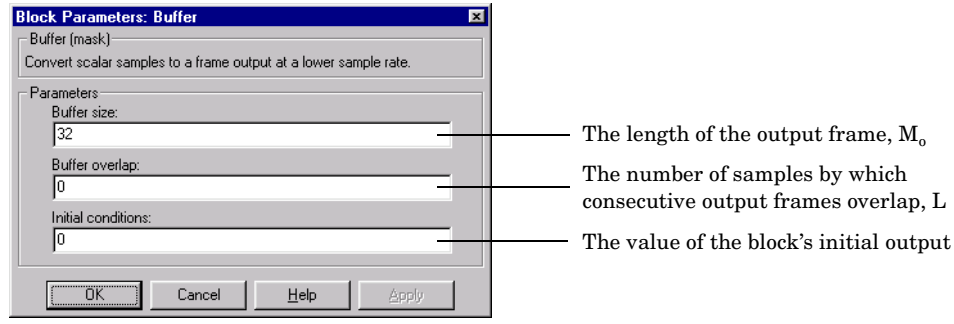
Block	Library
Variable Selector	Elementary Functions, in Math Functions
Zero Pad	Signal Operations, in General DSP

In addition, a number of blocks, including the buffered scopes and spectral estimation blocks, perform internal buffering of inputs, and provide some of the same parameters as the buffering blocks discussed in the next sections.

Note In this book, the terms *length* and *width* are used interchangeably to refer to the number of elements in a buffer, frame, or vector.

Buffering Sample-Based Signals

The Buffer block is the primary block for buffering scalars and sample vectors into frame vectors and frame matrices. The block lets you specify the **Buffer size**, the number of **Buffer overlap** points, and the block's initial output (**Initial condition**).



The sample-based input to the block can be a single channel signal (scalar sequence) or a multichannel signal (vector sequence). In both cases, the Buffer block performs the following operations:

1 Acquire the number of *new* inputs specified by the difference between the **Buffer size** (M_o) and **Buffer overlap** (L). Each new sample enters at the bottom of the buffer, and is pushed upwards as later samples enter. Single

channel (scalar) inputs enter a vector buffer, while multichannel (vector) inputs enter a matrix buffer as *rows*.

2 Propagate the buffered data to the output as a vector or matrix with a longer period (the input sample period multiplied by the number of *new* samples in the buffer). The first element or row in the output corresponds to the earliest input sample.

The output frame period, T_{fo}, is related to the input sample period, T_{si}, by

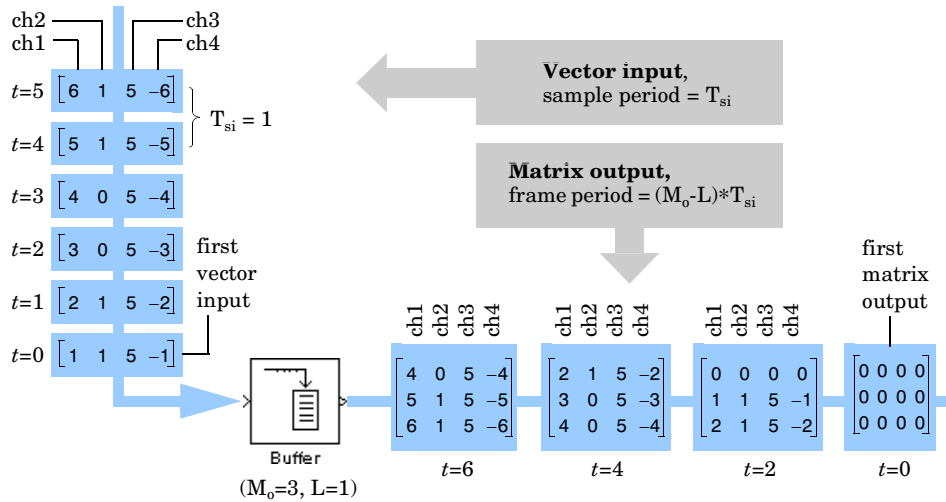
$$T_{fo} = (M_o - L)T_{si}$$

As a result, the new output sample period, T_{so}, is

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

This is equal to the input sample period only when the **Buffer overlap**, L, is zero.

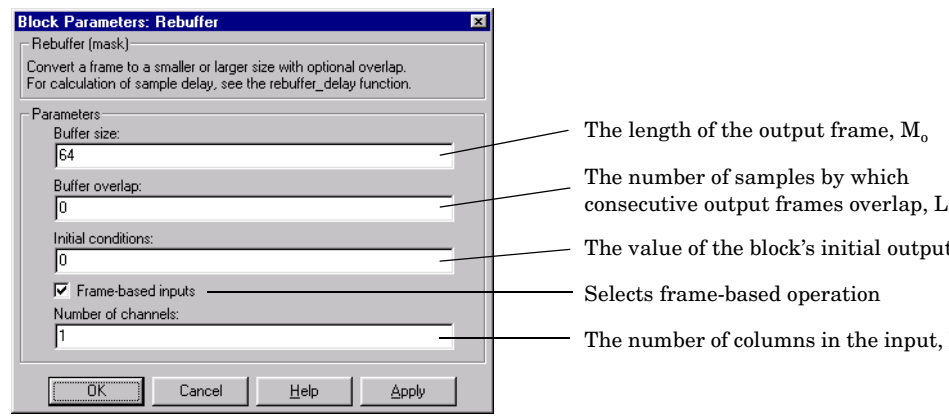
The figure below illustrates overlapping buffering of a four-channel signal. The **Buffer size** is 3 and the **Buffer overlap** is 1. Each signal channel is represented by a column of the output frame matrix. The input sample period (T_{si}) is 1, so the output frame period (T_{fo}) is 2, and the output sample period (T_{so}) is 2/3.



See “Using Overlapping Buffers” for a more thorough discussion of this type of buffering.

Rebuffering Frame-Based Signals

The Rebuffer block is the primary block for converting a frame-based signal to a new frame size. The block lets you specify the **Buffer size**, the number of **Buffer overlap** points, and the block’s initial output (**Initial condition**).



The block's **Frame-based inputs** parameter allows you to toggle the block from sample-based mode to frame-based mode. In sample-based mode (**Frame-based inputs** unchecked), the Rebuffer block operates the same as the buffer block described above.

In frame-based mode (**Frame-based inputs** checked), the input to the block can be a single channel signal (frame vector sequence) or a multichannel signal (frame matrix sequence). In both cases, the block acquires the number of *new samples* (input rows) specified by the difference between the **Buffer size** (M_o) and **Buffer overlap** (L) parameters. Because the block can rebuffer a signal to a larger or smaller frame size, the number of samples acquired from the input can be greater or less than the number of samples in an individual input frame.

The output frame period, T_{fo}, is related to the input sample period, T_{si}, by

$$T_{fo} = (M_o - L)T_{si}$$

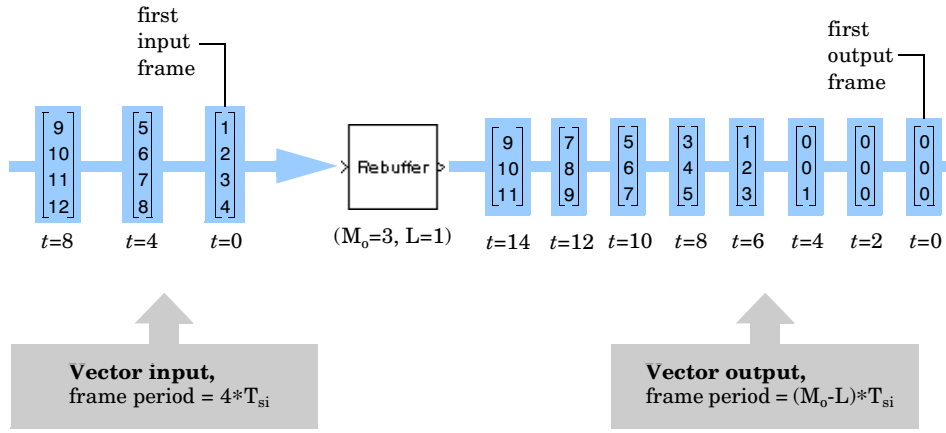
As a result, the new output sample period, T_{so}, is

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

This is equal to the input sample period only when the **Buffer overlap**, L, is zero.

Example: Single-Channel Rebuffering

The figure below illustrates overlapping buffering of a one-channel signal. The **Buffer size** is 3 and the **Buffer overlap** is 1. The input sample period (T_{si}) is 1, so the output frame period (T_{fo}) is 2, and the output sample period (T_{so}) is 2/3.



Computing Rebuffering Delay. Note that the sequence is delayed by eight samples, and the first eight output samples adopt the value specified for the **Initial condition**, which is zero in this example. You can use the `rebuffer_delay` function to determine the length of this initial delay for any combination of frame size and overlap.

For this example, enter

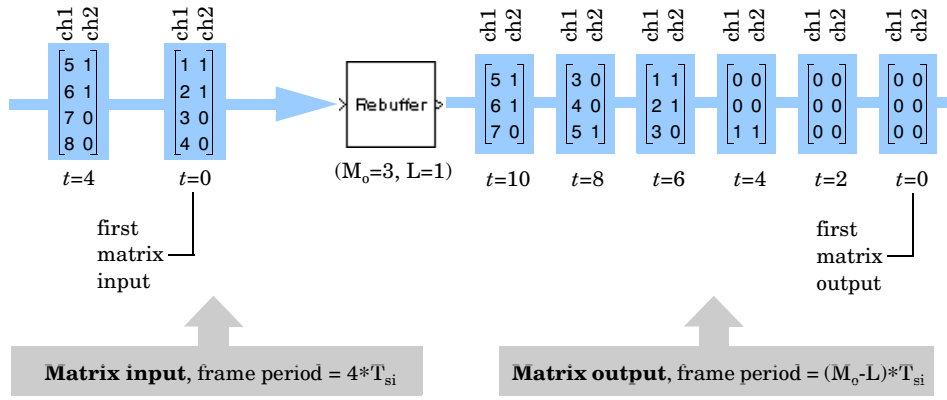
```
d = rebuffer_delay(4,3,1)

d =
     8
```

This agrees with the block's output above.

Example: Multichannel Rebuffering

The next figure illustrates overlapping buffering of a two-channel signal. Again, the **Buffer size** is 3 and the **Buffer overlap** is 1. As in the single-channel case, the input sample period of 1 generates an output frame period of 2 and an output sample period of 2/3.

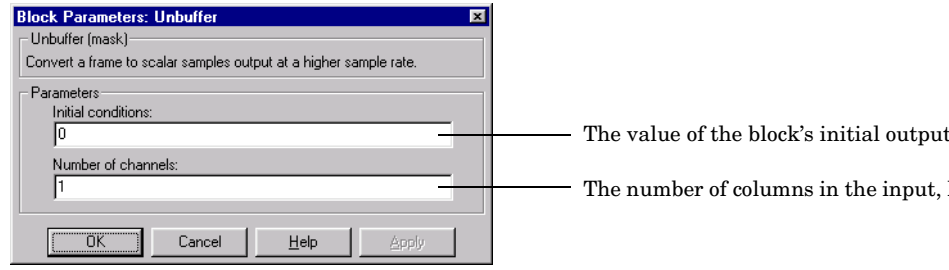


Unbuffering Frame-Based Signals

The Buffers library contains two blocks that unbuffer frame-based sequences into sample-based sequences: Unbuffer and Partial Unbuffer.

The Unbuffer Block

The Unbuffer block lets you specify the block's initial output (**Initial condition**) and the number of channels in the input (number of columns in matrix).



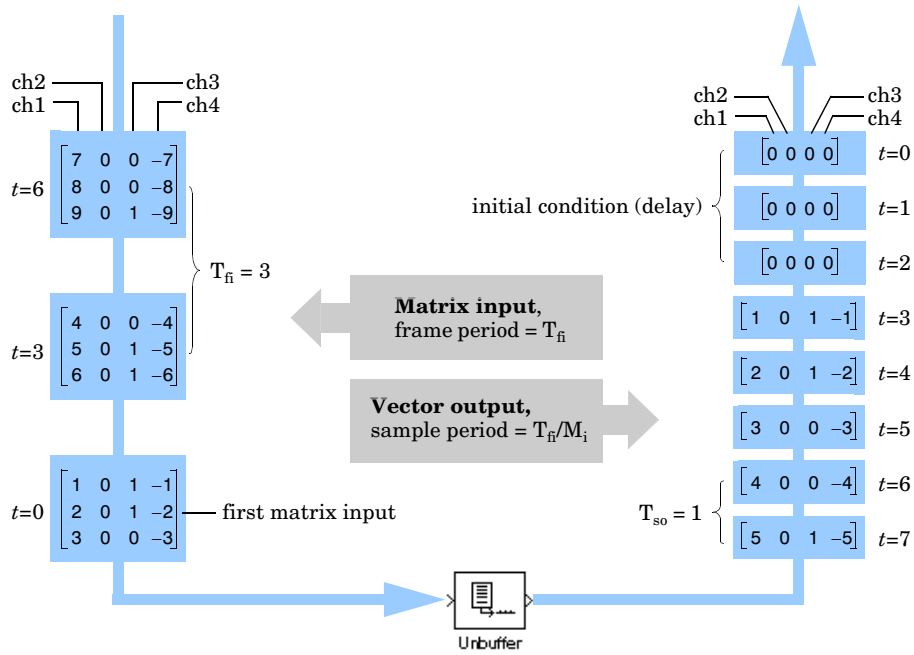
The frame-based input to the block can be a single-channel signal (frame vector sequence) or a multichannel signal (frame matrix sequence). In both cases, the Unbuffer block converts the entire input frame to a sample-based output.

The sample period of the sample-based output, T_{so}, is related to the input frame period, T_{fi}, by the input frame size, M_i.

$$T_{so} = T_{fi}/M_i$$

The Unbuffer block always preserves the signal's sample period ($T_{so} = T_{si}$).

The figure below illustrates the unbuffering of a four-channel signal The **Number of channels** parameter is correspondingly set to 4. Each sample (row) of the input matrix is output separately. The input frame period and frame size are both 3, so the output sample period is 1.



The Unbuffer block delays inputs by one frame length (T_{fi} seconds); see "Initial State of Buffer Blocks" later in this section for more about the unbuffering delay.

The Partial Unbuffer Block

The Partial Unbuffer block unbuffers a selected portion of the input frame into a sample-based output. The **First output index** (M_1) and the **Last output**

index (M_2) parameters determine the range of samples to be unbuffered from the input frame. For a multichannel input `u`, only rows `u(M1:M2,:)` are unbuffered to the output; all other input samples are discarded.

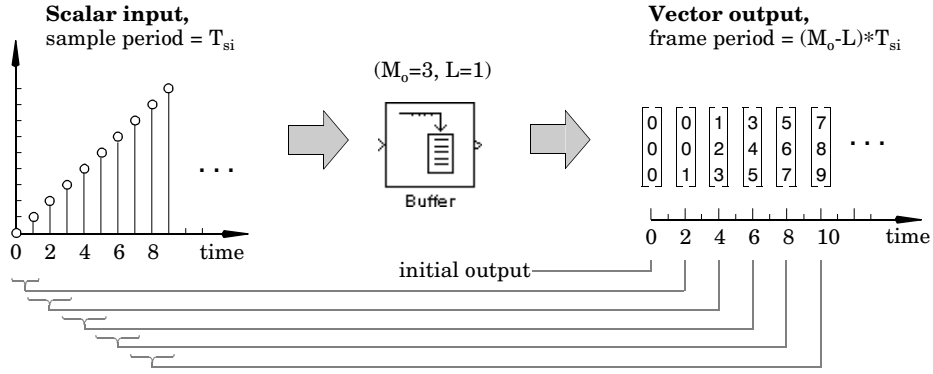
The output sample period is related to the input sample period by

$$T_{so} = \frac{M_i T_{si}}{M_2 - M_1 + 1}$$

where M_i is the input frame size. If the **Last output index** is greater than the length of the input frame ($M_2 > M_i$), or less than or equal to the **First output index** ($M_2 \leq M_1$), the block generates an error.

Using Overlapping Buffers

In some cases it is useful to work with data that represents overlapping sections of the original sample-based or frame-based time-series, as shown below. In estimating the power spectrum of a signal, for example, it is often desirable to compute the FFT of overlapping sections of data. Overlapping buffers are also needed in computing statistics on a sliding window, or for adaptive filtering. Both the Buffer and Rebuffer blocks have a **Buffer overlap** parameter that specifies the number of overlap points, L.



In this case, the frame period for the output vector is $(M_o\text{-}L)*T_{si}$, where T_{si} is the sample period of the input data and M_o is the **Buffer size**.

Note Set the **Buffer overlap** to a negative value to achieve output frame rates *slower* than in the nonoverlapping case. The output frame period is still $T_{si}*(M_o-L)$, but with L<0. Only the M_o newest inputs are included in the output buffer; the previous L inputs are discarded.

When unbuffering overlapping frames using the Partial Unbuffer block, the number of samples to unbuffer should equal the number of *nonoverlapping* samples in the input buffer. (This ensures that the unbuffering process is synchronized with the Buffer block's overlap, and only unique input samples are unbuffered.) That is, for accurate reconstruction of an overlapping signal, M_2-M_1+1 should equal M_o-L.

In this case, the **Last output index** can have any value greater than the **First output index** and less than the input frame length. For example, given a length 256 input frame with 192 points of overlap, possible valid index parameters for Partial Unbuffer would be **First output index** = `1` and **Last output index** = `64`, **First output index** = `2` and **Last output index** = `65`, and so on. To obtain the most recent data points, however, you should set the **First output index** to L+1 and the **Last output index** to M_o (`193` and `256` in this case).

Initial State of Buffer Blocks

The Buffer and Rebuffer Blocks

The Buffer and Rebuffer blocks are initialized to the value specified by the **Initial condition** parameter, which they output at the first simulation step. At the same sample time that the initial buffer is output, the Buffer and Rebuffer blocks read the first input sample into the next buffer. Once a Buffer or Rebuffer block has acquired M_o-L samples (where M_o is the **Buffer size** and L is the **Buffer overlap**), it outputs those samples in the second buffer at time $T_{si}*(M_o-L)$.

Note The first output buffer (at t=0) is specified by the **Initial condition** parameter, whose default is zero. Inputs to the Buffer and Rebuffer blocks only begin appearing in the second output buffer. If the blocks that are generating the input to a Buffer or Rebuffer block also output zeros at the first simulation step, you may see two or more buffers of zeros before the first nonzero output.

For example, consider the scalar input series 1, 2, 3, . . . , with sample period T_s. Given a **Buffer size** of 4, a **Buffer overlap** of 2, and the default **Initial condition** of 0:

Time	Buffer output	Notes
0	0 0 0 0	The first output – all zeros
$2*T_s$	0 0 1 2	
$4*T_s$	1 2 3 4	The first full buffer
$6*T_s$	3 4 5 6	
$8*T_s$	5 6 7 8	

For a **Buffer size** of 5 and a **Buffer overlap** of 3:

Time	Buffer output	Notes
0	0 0 0 0 0	The first output – all zeros
$2*T_s$	0 0 0 1 2	
$4*T_s$	0 1 2 3 4	
$6*T_s$	2 3 4 5 6	The first full buffer

The Unbuffer and Partial Unbuffer Blocks

The Unbuffer and Partial Unbuffer blocks are initialized with the buffer specified by the **Initial condition** parameter, which they begin unbuffering at the first simulation step. (The Partial Unbuffer block unbuffers the initial buffer in the usual manner, beginning with element M_1 and ending with

element M_2, where M_1 is the **First output index** parameter value and M_2 is the **Last output index** parameter value.) At the same sample time that the first value (or M_1) from the initial buffer is output, the Unbuffer and Partial Unbuffer blocks receive and store the first input buffer. When the last value (or M_2) in the initial buffer is output, the Unbuffer and Partial Unbuffer blocks begin unbuffering the first input buffer. Unbuffer and Partial Unbuffer blocks therefore *delay* inputs by one buffer length, or T_{si} seconds (the input sample period).

Note If the blocks that are generating the input to an Unbuffer or Partial Unbuffer block output zeros at the first simulation step, you may see a large number of zeros in the unbuffered output at the start of a simulation.

For example, consider the input series

$$\begin{bmatrix}1\\2\\3\end{bmatrix}, \begin{bmatrix}4\\5\\6\end{bmatrix}, \begin{bmatrix}7\\8\\9\end{bmatrix}, \cdots$$

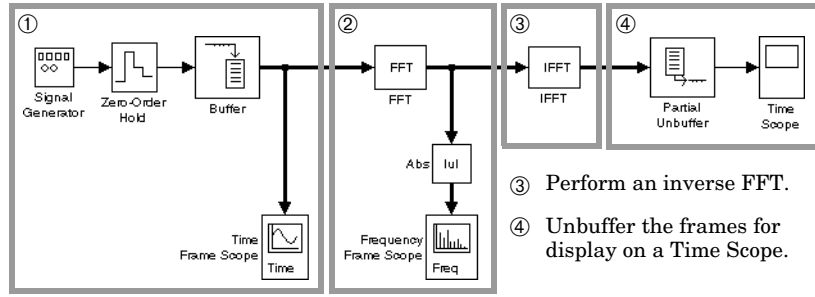
with frame period T_{fi}. The output of the Unbuffer block with zero initial conditions is then

```
[0],[0],[0],[1],[2],[3],[4],[5],[6],[7],[8],[9],...
```

with sample period $T_{fi}/3$.

Example: Using Buffer and Unbuffer

The model below demonstrates how to use the Buffer block to create a frame vector that the FFT block can operate on. At the end of the processing flow, the Partial Unbuffer block converts the frame vectors back to scalar samples for output on a Scope.



③ Perform an inverse FFT.

④ Unbuffer the frames for display on a Time Scope.

① Buffer data to create frames of length 256 with an overlap of 192 samples.

② Perform a length-256 FFT on each frame and plot the magnitude.

Construct the model using:

- Buffer and Partial Unbuffer blocks from the Buffers library (in General DSP)
- FFT and IFFT blocks in the Transforms library (in General DSP)
- Abs block from the Simulink Math library
- Time Frame Scope and Frequency Frame Scope blocks from the DSP Sinks library
- Signal Generator block from the Simulink Sources library
- Zero-Order Hold block from the Simulink Discrete library
- Time Scope from the DSP Sources library

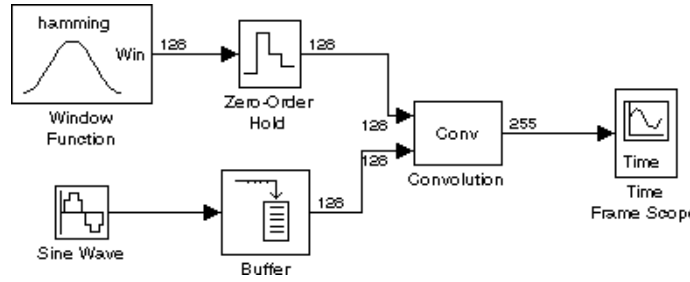
To try the model:

1. Set the input signal parameters in the Signal Generator block:
   a. **Wave form** = **square**
   b. **Amplitude** = `2`
   c. **Frequency** = `3.5`
   d. **Units** = **rad/sec**
2. Set the **Sample time** in the Zero-Order Hold block to `0.001`.
3. Set the Buffer block parameters:
   a. **Buffer size** = `256`
   b. **Buffer overlap** = `192`
4. Set the Partial Unbuffer block parameters:
   a. **Buffer size** = `256`
   b. **First output index** = `193`
   c. **Last output index** = `256`
5. Set the Frequency Frame Scope block parameters:
   a. **Frequency units** = **rads/sec**
   b. **Sample time of vector elements** = `0.001`
6. Set the **Stop time** in the **Parameters** dialog box to `inf`, and start the simulation by selecting **Start** from the **Simulation** menu.

The Time Frame Scope window displays the buffered input, and the Frequency Frame Scope window displays its FFT. Stop the simulation at any time by selecting **Stop** from the **Simulation** menu.

Example: Convolution

The model below convolves a length-128 Hamming window with successive nonoverlapping, 128-sample buffers containing samples generated by the Sine Wave block. The Time Frame Scope window opens automatically and displays the output.



Construct the model using:

- Buffer block from the Buffers library (in General DSP)
- Convolution block from the Vector Functions library (in Math Functions)
- Sine Wave block from the DSP Sources library
- Time Frame Scope block from the DSP Sinks library
- Window Function block from the Signal Operations library (in General DSP)
- Zero-Order Hold block from the Simulink Discrete library

To try the model:

1. Set the Window Function block parameters to generate a length-128 Hamming window.
   a. Select **Generate window** from the **Operation** pop-up menu.
   b. Select **Hamming** from the **Window Type** pop-up menu.
   c. Enter `128` for **Window length**.
2. Set the **Sample time** parameter of the Sine Wave block to `0.001`.
3. Set the **Sample time** parameter of the Zero-Order Hold block to `0.001*128`. This matches the output frame period to the output frame period of the Buffer block.

4 Set the **Buffer size** parameter of the Buffer block to `128`.

5 In the Time Frame Scope block dialog box, click the **Axis properties** check box to expose the axis properties panel. Set the following parameter values:

   a **Minimum Y-limit** = `-75`

   b **Maximum Y-limit** = `75`

6 Set the **Stop time** to `inf` in the **Parameters** dialog box (from the **Simulation** menu) and start the simulation by selecting **Start** from the **Simulation** menu.

   The Time Frame Scope window displays the result of the convolution. For input vectors of length M_u and M_v, the Convolution block outputs a vector of length M_u+M_v-1. Here, each result has length 255.

7 Stop the simulation at any time by selecting **Stop** from the **Simulation** menu.

Try changing some characteristics of the model. For example, use a different windowing function, or change the Window Function block and Buffer block parameters to use frames of length 64.

Working with Sources and Sinks

Two essential features of every Simulink model are data sources and data sinks. These provide the means for initiating signal flow in a system, and for analyzing the system's response during and after the simulation.

The Blockset's DSP Sources library contains blocks for importing data from the workspace, file, and sound device, and for generating constant and time-varying signals. The DSP Sinks library provides blocks for exporting data to the workspace, file, and sound device, and for visualizing time-domain and frequency-domain signals on the screen.

Importing Data from the Workspace

The MATLAB workspace provides a convenient storage facility for data of all kinds. Although Simulink and the DSP Blockset provide a number of basic signal generators (e.g., Sine Wave, Discrete Constant, Signal Generator, Chirp), more complex signals, such as speech or sensor telemetry, are usually read into Simulink from the MATLAB workspace.

A single-channel signal is represented in the workspace as either a row-vector or column-vector. For example, the vectors below

```
u = 1:1000          % a row-vector
u = (1:1000)'       % a column-vector
```

are both one-channel signals.

A multichannel signal is represented as an M-by-N matrix, with each column a separate channel. For example, the commands

```
load mtlb
u = [mtlb flipud(mtlb) zeros(size(mtlb))]
```

create a three-channel signal, the first few samples of which are shown below.

```
      Ch 1       Ch 2        Ch 3
u =
    0.0065   -0.1399          0
   -0.0078   -0.1268          0
   -0.0551   -0.1052          0
   -0.0712   -0.0954          0
   -0.1070   -0.0697          0
   -0.1192   -0.0067          0
   -0.1157    0.0085          0
   -0.1163    0.0253          0
   -0.0922    0.0232          0
   -0.0726    0.0250          0
      :          :            :
```

Channel 1 (the first column) contains the familiar `mtlb` speech signal. Channel 2 contains a flipped copy of the `mtlb` signal, and Channel 3 contains zeros. The three samples in each row are considered to correspond to the same instant in time; that is, each row is a sample vector.

There are three blocks in the DSP Blockset that import data from the MATLAB workspace into a Simulink model:

- Signal From Workspace
- Triggered Signal From Workspace
- Matrix From Workspace

Signal From Workspace

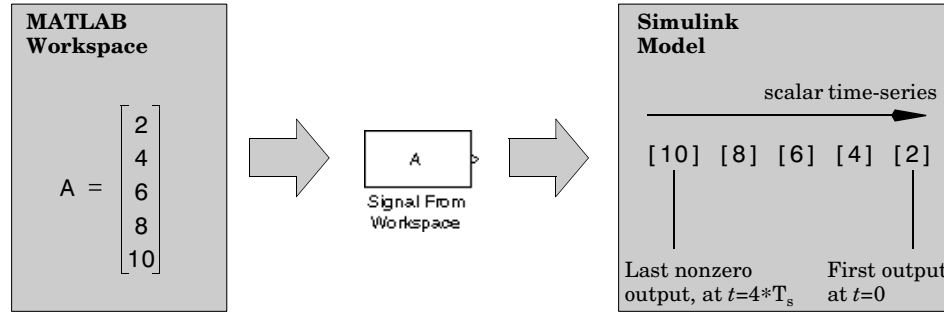
The Signal From Workspace block imports signal data from the workspace to output in the simulation as a sample-based or frame-based sequence. The **Signal** parameter allows you to specify the name of an existing vector or matrix in the workspace, or to directly enter a MATLAB expression defining the signal.

The **Samples per frame** parameter specifies the number of consecutive signal samples from each channel to be included in each output frame.

Workspace Vector. When the signal in the workspace is a vector (one channel), a **Samples per frame** parameter value of 1 creates a scalar output, releasing the

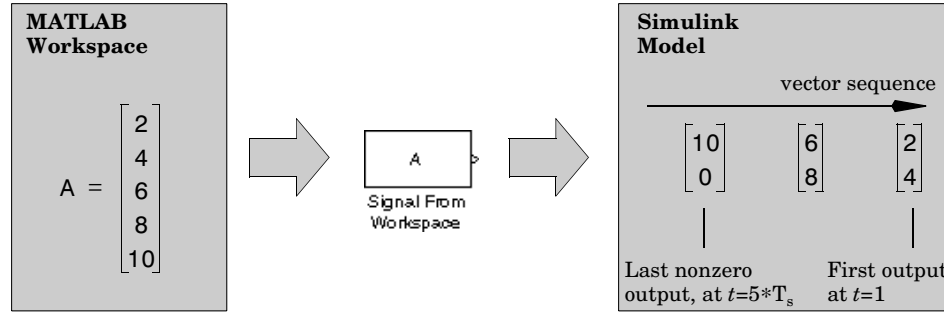
vector elements into the simulation one-by-one. The sample period of the scalar sequence, T_s, is specified by the **Sample time** parameter.

The figure below illustrates this process for a vector, A, in the workspace. The **Sample time** parameter value is 1. After the last vector element is output (if the simulation runs that long), the block outputs zeros for the remainder of the simulation.



For a **Samples per frame** parameter value of M (M>1), the block creates a frame-based output by acquiring M consecutive vector elements before releasing the resulting frame into the simulation. The sample period of the *sequence* contained in the output frames, T_s, is again specified by the **Sample time** parameter. The output frame period, T_{fo}, is $M*T_s$.

The figure below illustrates the same single-channel signal ($T_s = 1$) being output with a frame size of 2.



Workspace Matrix . When the signal in the workspace is a matrix (multiple channels), setting the **Samples per frame** parameter to 1 creates a length-N sample vector output, releasing the matrix *rows* into the simulation

one-by-one. The sample period of the vector sequence, T_{si}, is specified by the **Sample time** parameter.

For a **Samples per frame** parameter value of M (M>1), the block creates a frame-based output by acquiring M consecutive matrix rows before releasing the resulting frame matrix into the simulation. The sample period, T_s, of the multichannel signal contained in the output frame matrix is again specified by the **Sample time** parameter. The output frame matrix period, T_{fo}, is $M*T_s$.

Differences Between Signal From Workspace and Simulink From Workspace. This block differs in three important respects from the From Workspace block in the Simulink Sources library:

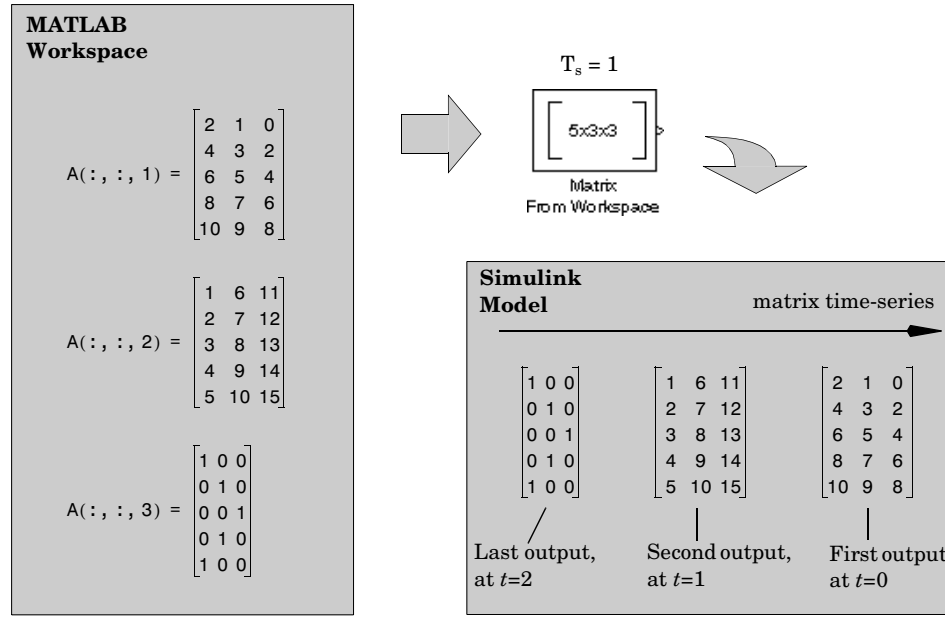
- The Signal From Workspace block does not interpret the first column of the workspace matrix as a time vector. Instead, you specify a fixed period using the **Sample time** parameter in the dialog box.
- After all data elements from the workspace matrix or vector have been output, the Signal From Workspace block outputs zeros, whereas the From Workspace block linearly extrapolates based on the last two samples.
- The Signal From Workspace block uses a zero-order hold instead of linear interpolation to determine signal values at time instants falling between the regular sample intervals.

Triggered Signal From Workspace

The Triggered Signal From Workspace block is identical to the Signal From Workspace block, but only acquires samples from the workspace when triggered by a control signal.

Matrix From Workspace

The Matrix From Workspace block references a three-dimensional array in the workspace, `A`, to generate a matrix output to the system. At each sample time, the block outputs a *page* (a two-dimensional slice) of the three-dimensional array, beginning with the first, `A(:,:,1)`, at t=0. The block continues to output pages of the array until it outputs the last page, `A(:,:,end)`.



Exporting Data to the Workspace

There are four blocks in the DSP Sinks library that export data to the workspace from a Simulink model:

- Signal To Workspace
- Matrix To Workspace
- Triggered Signal To Workspace
- Triggered Matrix To Workspace

The *triggered* sink blocks work the same as their nontriggered counterparts, but only sample the input when a specified trigger signal is received, instead of at a regular sample interval. See Chapter 4, "DSP Block Reference," for detailed information about each of the triggered sink blocks.

Signal To Workspace

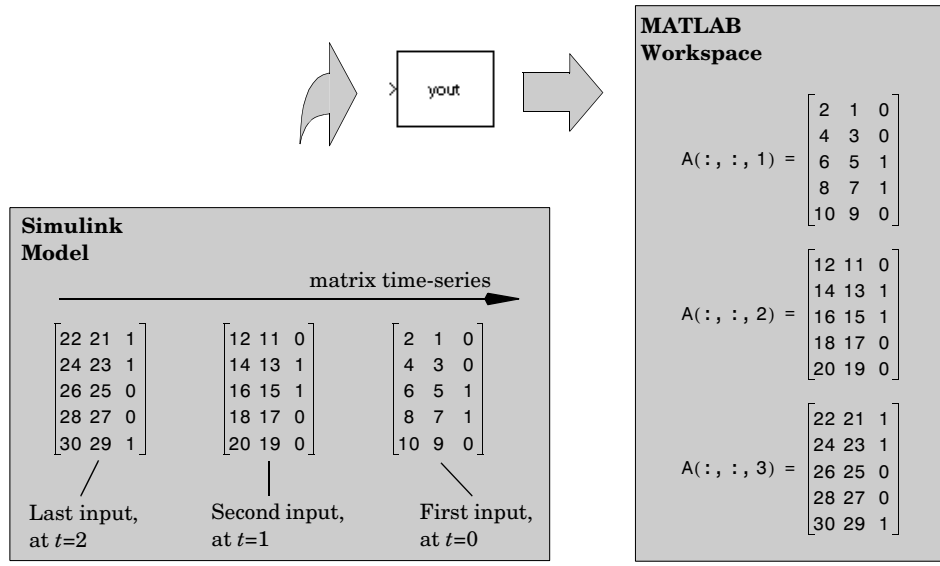
The Signal To Workspace is the counterpart of the Signal From Workspace block as the primary method of exporting sample-based and frame-based sequences to the workspace. The **Variable name** parameter allows you to

specify the name of a workspace vector or matrix in which to store the data. Existing workspace variables with the specified name are overwritten.

The **Frame-based** parameter toggles the block between sample-based and frame-based processing. In sample-based mode (**Frame-based** unchecked), the block treats inputs as sample vectors or sample matrices; each vector or matrix input is written to a unique row of the workspace matrix. In frame-based mode (**Frame-based** checked), the block treats inputs as frame vectors and frame matrices, and appends each successive input to the workspace matrix. In other words, the Signal To Workspace block pastes successive frames together to recreate the original signal.

Matrix To Workspace

The Matrix To Workspace block writes a three-dimensional array, `A`, to the workspace, where `A` contains the acquired samples of a matrix input. At the end of the simulation the block writes every Dth input (a matrix) to a *page* (a two-dimensional slice) of the specified three-dimensional workspace array, where D is specified by the block's **Decimation factor** parameter. The first input is written to the first page of the array, `A(:,:,1)`, and the block continues adding pages to the array until it writes the last input matrix to the last page, `A(:,:,end)`.



Viewing Data with Scopes

The DSP Sinks library provides six special-purpose scope blocks that you can use to view different kinds of signals. Use these to supplement Simulink's standard Scope block in your DSP models:

- Buffered FFT Frame Scope
- FFT Frame Scope
- Frequency Frame Scope
- Matrix Viewer
- Time Frame Scope
- User-Defined Frame Scope

Buffered FFT Frame Scope. The Buffered FFT Frame Scope block is similar to the FFT Frame Scope scope (below) but provides internal buffering of the input, which is assumed to be a sample-based signal. The **Buffer size** and **Buffer overlap** parameters that control the block's buffering operation are the same as those in the Buffer and Rebuffer blocks.

FFT Frame Scope. The FFT Frame Scope block displays the magnitude of the FFT of the input, which is assumed to be a frame-based signal. Each channel of a multichannel (matrix) input is displayed independently.

Frequency Frame Scope. The Frequency Frame Scope block displays a time-varying vector containing frequency-domain data. The block plots the elements of the input vector against frequency, using the input frame period and size, or a manually entered value, to determine the correct frequency placement.

Matrix Viewer. The Matrix Viewer block displays an M-by-N matrix input by mapping the matrix element values to a specified range of colors. The number of input columns must be specified in the **Number of columns** parameter of the **Image properties** panel. The display is updated as each new input is received.

Time Frame Scope. The Time Frame Scope block is the primary tool for displaying time-domain signals. It differs from the Simulink Scope block in that it displays one or more frames of data simultaneously

User-Defined Frame Scope. The User-Defined Frame Scope block displays a sequence of vectors without making any assumption about the source of the data (time-domain or frequency-domain).

Working with Statistical Operations

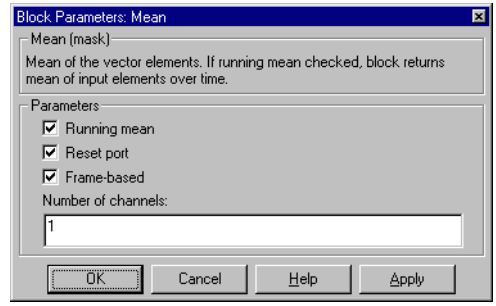
The Statistics library provides fundamental statistical operations such as minimum, maximum, mean, variance, and standard deviation. Most blocks in the Statistics library support two types of operations:

- Basic operations
- Running operations

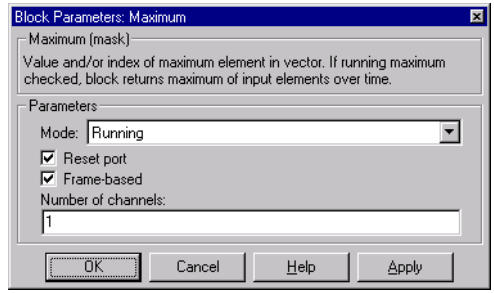
The blocks listed below toggle between basic and running modes using the **Running** check box in the parameter dialog box:

- Histogram
- Mean
- RMS
- Standard Deviation
- Variance

An unchecked box means that the block is operating in basic mode, while a checked box means that the block is operating in running mode.



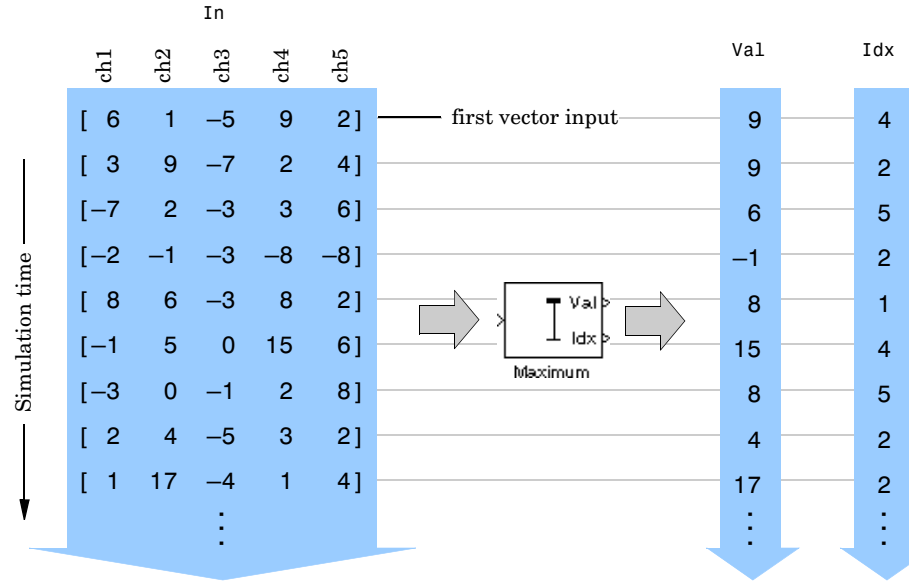
The Maximum and Minimum blocks are slightly different, and provide a **Mode** parameter in the block dialog box to select the type of operation. The **Value and Index**, **Value**, and **Index** options in the **Mode** pop-up menu all specify basic operation, in each case enabling a different set of output ports on the block. The **Running** option in the **Mode** pop-up menu selects running operation.



Basic Operations

A *basic operation* is one that processes a sample-based or frame-based input to produce a scalar result. For example, in basic mode (**Value and Index**) the Maximum block finds the maximum value among all the channels in a sample-based input or among all the samples in a frame-based input. The block provides this maximum value at the top output (`Val`), and provides the index (channel or sample number) of the maximum value at the bottom output (`Idx`). The block repeats this operation for each successive input.

The figure below illustrates how a Maximum block in basic mode operates on a sample vector sequence.



Basic operations process matrix inputs in the same way as vector inputs; the largest single matrix value and its index are output.

Running Operations

A *running operation* is one that processes successive sample-based or frame-based inputs, and computes a result that reflects both present and past inputs. A reset port enables you to restart this tracking at any time. The running statistic is computed for each input channel independently, so the block's output has the same number of channels as the input.

Sample-based inputs:

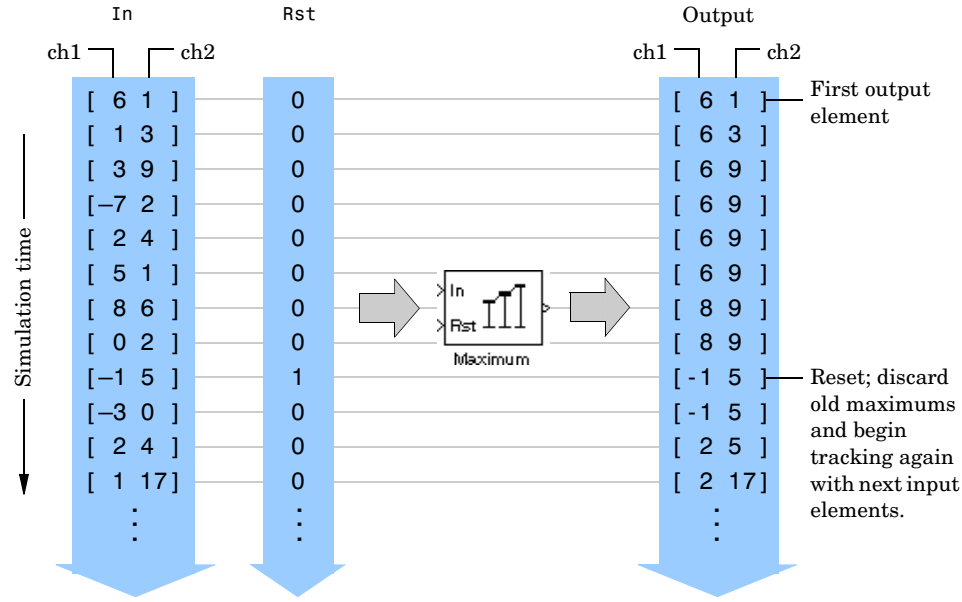
- A length-N sample vector input generates a length-N vector output.
- An M-by-N sample matrix input generates an M-by-N matrix output.

Frame-based inputs:

- A length-M frame vector input generates a scalar output.
- An M-by-N frame matrix input generates a length-N sample vector output.

For example, in running mode (**Running** selected from the **Mode** parameter) the Maximum block outputs a frame-by-frame record of the input's maximum value over time.

The figure below illustrates how a Maximum block in running mode operates on a sample vector sequence.



Demonstration Model: Running Operation

The DSP Blockset includes a demo that illustrates the running mode of a few statistics blocks.

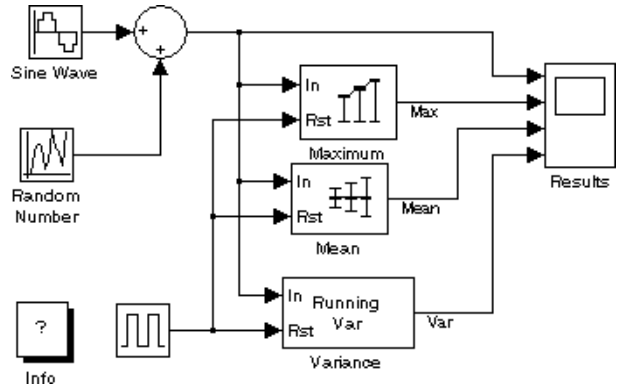
To try the demo model:

1 Double-click on the Demos block in the top-level DSP Blockset library,

or type `demos` at the MATLAB command line.

2 Find the demo called **Statistical Functions** and open it.

The model, shown below, feeds a noisy sine wave into the Maximum, Mean, and Variance blocks. A Discrete Pulse Generator resets the blocks every 100 seconds, and a Scope displays the block outputs.



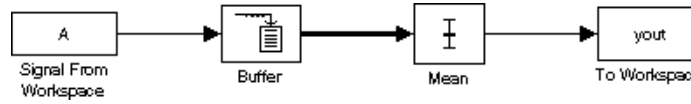
3 Start the simulation by selecting **Start** from the **Simulation** menu.

The Scope displays the output of each function block. Stop the simulation at any time by selecting **Stop** from the **Simulation** menu.

Example: Sliding Windows

You can use the basic statistics operations in conjunction with the Buffer block to implement basic sliding window statistics operations. A *sliding window* is like a stencil that you move along a data stream, exposing only a set number of data points at one time. For example, you may want to process data in 10-point frames, moving the window along by one sample point for each operation.

One way to implement a sliding window is shown below:



The Buffer block's **Buffer size** (M_o) parameter determines the size of the window. The **Buffer overlap** (L) parameter defines the "slide factor" for the window. At each sample instant, the window slides by M_o-L points. The **Buffer**

overlap is often M_0-1 (the same as the Shift Register block), so that a new statistic is computed for every new signal sample.

Construct the model using:

- Signal From Workspace block from the DSP Sources library
- Buffer block from the Buffers library (in General DSP)
- Mean block from the Statistics library (in Math Functions)
- Signal To Workspace block from the DSP Sinks library

To try the model:

1. Create test data. At the MATLAB prompt, enter:

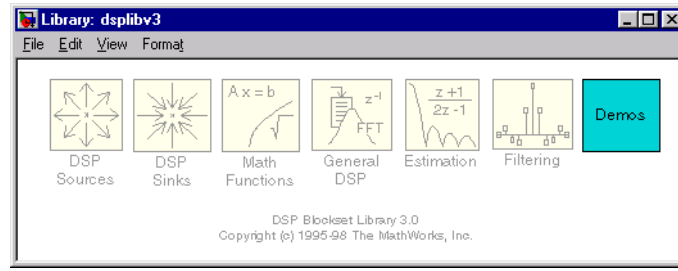
   ```
   A = 1:256;
   ```

2. Set the Signal From Workspace block parameters:
   a. **Signal** = `A`
   b. **Sample time** = `0.1`
3. Set the Buffer block parameters:
   a. **Buffer size** = `128`
   b. **Buffer overlap** = `127`
4. Set the **Maximum number of inputs to record** in the Signal To Workspace block to `1000`.
5. Set the simulation parameters:
   a. Select **Parameters** from the **Simulation** menu.
   b. **Start time** = `0.0`
   c. **Stop time** = `25.6`, the length of `A` multiplied by the sample period.
6. Start the simulation by selecting **Start** from the **Simulation** menu.

The simulation stops automatically when it reaches the specified **Stop Time**. Look at the contents of the `yout` variable. `yout(129)` contains the mean for the first 128 points of the test vector `A`. Note that `yout(1)` is `0`, which was the initial output of the Buffer block.

DSP Blockset Demos

The Demos block in the top-level DSP Blockset library brings up the MATLAB Demos window, which contains a number of advanced DSP Blockset demonstration models.



The library includes the following demo models, along with many others:

- **Linear Prediction** – uses the LMS Adaptive Filter block to adaptively compute the linear prediction coefficients for a noisy input signal
- **LPC Analysis and Synthesis** – uses the Levinson Solver and Time-Varying Lattice Filter for low-bandwidth transmission of speech
- **WWV Digital Receiver** – uses Stateflow® together with a wide variety of DSP Blockset components to implement a time-code receiver
- **Multistage Multirate Filtering** – uses FIR Decimation blocks in multiple stages to perform filtering with very short bandwidths and low computational loads
- **Reverberation** – uses the Integer Delay block to produce the popular audio effect
- **Comparison of Spectral Analysis Techniques** – uses a Frequency Frame Scope to simultaneously display spectral estimates computed by the Short-Time FFT, Burg Method, and Modified Covariance Method blocks

Explore all the demos to see how you can implement both basic and advanced DSP algorithms with the DSP Blockset. You can also use the demos as a base for building your own models. Simply select the section of the demo that you want to build on and copy it into your own model.

4

DSP Block Reference

Using the DSP Block Reference Chapter

This chapter contains complete information on every block in the DSP Blockset in a structured, accessible format. You should turn to this chapter when you need to find information on a particular block. These reference pages are also accessible online via the **Help** button in each block's dialog box, and through the Help Desk.

To learn the basic concepts behind building DSP models with Simulink, see Chapter 2, "Simulink and the DSP Blockset." To find out about using blocks together for common DSP tasks, see Chapter 3, "Using the DSP Blockset."

What Each Block Reference Page Contains

The block reference entries appear in alphabetical order and each contains the following information:

- The block *name*, at the top of the page.
- The *purpose* of the block.
- The *library* or libraries where the block can be found.
- A *description* of the block's use.
- The block's *dialog box* and parameters. Tunable parameters are indicated by a Ⓣ icon. See "About Tunable Parameters" below.
- A *See Also* list of related blocks and functions.

About Tunable Parameters

Block dialog box parameters that can be adjusted while a simulation is running are called *tunable* parameters. On the block reference pages these parameters are indicated by a Ⓣ icon next to the parameter description in the "Dialog Box" section. Parameter descriptions that do not display this icon are not tunable; changing these parameters while the simulation is running generates an error, and suspends the simulation until the error dialog box is dismissed.

Block Library Hierarchy

The DSP Blockset has the following library structure:

- DSP Sources
- DSP Sinks
- Math Functions
  - Elementary Functions
  - Vector Functions
  - Matrix Functions
  - Linear Algebra
  - Statistics
- General DSP
  - Signal Operations
  - Transforms
  - Buffers
  - Switches and Counters
- Estimation
  - Parametric Estimation
  - Power Spectrum Estimation
- Filtering
  - Filter Designs
  - Filter Realizations
  - Adaptive Filters
  - Multirate Filters

Block Library Contents

The DSP blocks in each of these libraries are listed below. Simulink blocks that appear in DSP Blockset libraries (such as Constant) are not included.

DSP Sources	
Chirp	Matrix From Workspace
Constant Diagonal Matrix	N-Sample Enable

DSP Sources (Continued)	
Discrete Constant	Signal From Workspace
From Wave Device	Triggered Signal From Workspace
From Wave File	Sine Wave
Matrix Constant	

DSP Sinks	
Buffered FFT Frame Scope	Time Frame Scope
FFT Frame Scope	To Wave Device
Frequency Frame Scope	To Wave File
Matrix To Workspace	Triggered Matrix To Workspace
Matrix Viewer	Triggered Signal To Workspace
Signal To Workspace	User-Defined Frame Scope

Elementary Functions	
Complex Exponential	dB
Contiguous Copy	dB Gain
Convert Complex DSP To Simulink	Inherit Complexity
Convert Complex Simulink To DSP	Variable Selector

Vector Functions	
Autocorrelation	Difference
Convolution	Flip
Correlation	Normalization
Cumulative Sum	Unwrap

Matrix Functions	
Constant Diagonal Matrix	Matrix Scaling
Create Diagonal Matrix	Matrix Sum
Extract Diagonal	Matrix To Workspace
Extract Triangular Matrix	Permute Matrix
Matrix 1-Norm	Reshape
Matrix Constant	Submatrix
Matrix From Workspace	Toeplitz
Matrix Multiplication	Transpose
Matrix Product	

Linear Algebra	
Backward Substitution	Levinson Solver
Cholesky Factorization	LU Factorization
Cholesky Solver	LU Solver
Forward Substitution	QR Factorization
LDL Factorization	QR Solver
LDL Solver	Reciprocal Condition

Statistics	
Histogram	RMS
Maximum	Sort
Mean	Standard Deviation
Median	Variance
Minimum	

Signal Operations	
Analytic Signal	Upsample
Detrend	Variable Fractional Delay
Downsample	Variable Integer Delay
Integer Delay	Window Function
LPC	Zero Pad
Repeat	

Transforms	
Complex Cepstrum	IDCT
DCT	IFFT
FFT	Real Cepstrum

Buffers	
Buffer	Shift Register
Partial Unbuffer	Stack
Queue	Triggered Shift Register
Rebuffer	Unbuffer

Switches and Counters	
Commutator	Multiphase Clock
Counter	N-Sample Enable
Distributor	N-Sample Switch
Edge Detector	Sample and Hold

Switches and Counters (Continued)	
Event-Count Comparator	

Parametric Estimation	
Burg AR Estimator	Modified Covariance AR Estimator
Covariance AR Estimator	Yule-Walker AR Estimator

Power Spectrum Estimation	
Burg Method	Modified Covariance Method
Covariance Method	Short-Time FFT
Magnitude FFT	Yule-Walker Method

Filter Designs	
Analog Filter Design	Least Squares FIR Filter Design
Digital FIR Filter Design	Remez FIR Filter Design
Digital IIR Filter Design	Yule-Walker IIR Filter Design

Filter Realizations	
Biquadratic Filter	Overlap-Save FFT Filter
Direct-Form II Transpose Filter	Time-Varying Direct-Form II Transpose Filter
Filter Realization Wizard	Time-Varying Lattice Filter
Overlap-Add FFT Filter	

Adaptive Filters	
Kalman Adaptive Filter	RLS Adaptive Filter
LMS Adaptive Filter	

Multirate Filters	
Dyadic Analysis Filter Bank	FIR Interpolation
Dyadic Synthesis Filter Bank	FIR Rate Conversion
FIR Decimation	

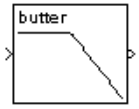
Purpose

Design and implement an analog filter.

Library

Filter Designs, in Filtering

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 block treats each element of the input as a distinct channel to independently filter over time.

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.

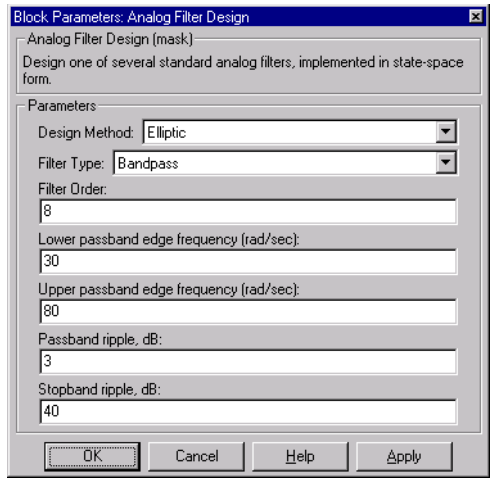
The design and band configuration of the filter are selected from the **Design method** and **Filter type** pop-up menus in the dialog box. For each combination of design method and band configuration, an appropriate set of secondary parameters is displayed.

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 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 rads/sec, 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 Signal Processing Toolbox's filter design commands `buttap`, `cheb1ap`, `cheb2ap`, and `ellipap`, 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 a portion 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**.

Filter type

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

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 rads/sec, for the highpass and lowpass configurations of the Butterworth, Chebyshev type I, and elliptic designs.

Lower passband edge frequency

The lower passband frequency, in rads/sec, for the bandpass and bandstop configurations of the Butterworth, Chebyshev type I, and elliptic designs.

Upper passband edge frequency

The upper passband frequency, in rads/sec, for the bandpass and bandstop configurations of the Butterworth, Chebyshev type I, or elliptic designs.

Stopband edge frequency

The stopband edge frequency, in rads/sec, for the highpass and lowpass band configurations of the Chebyshev type II design.

Lower stopband edge frequency

The lower stopband edge frequency, in rads/sec, for the bandpass and bandstop configurations of the Chebyshev type II design.

Upper stopband edge frequency

The upper stopband edge frequency, in rads/sec, for the bandpass and bandstop filter configurations of the Chebyshev type II design.

Passband ripple

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

Stopband ripple

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

References

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

See Also

Digital FIR Filter Design
Digital IIR Filter Design
`buttap` (Signal Processing Toolbox)
`cheb1ap` (Signal Processing Toolbox)
`cheb2ap` (Signal Processing Toolbox)
`ellipap` (Signal Processing Toolbox)

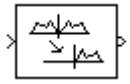
Purpose

Compute the analytic signal of a discrete-time input.

Library

Signal Operations, in General DSP

Description



The Analytic Signal block computes the complex analytic signal corresponding to each channel of the real input. The real part of the output is a replica of the real input; the imaginary part is the Hilbert transform of the input.

$$y(t) = u(t) + j\mathrm{H}\{u(t)\}$$

where H{·} denotes the Hilbert transform. 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 designed using the Remez exchange algorithm. The linear phase filter imposes a delay of one half the **Filter order** on the input samples.

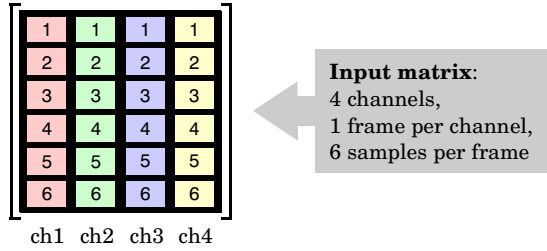
You can choose frame-based or sample-based operation by selecting (or deselecting, respectively) the **Frame-based inputs** check box. In both sample-based and frame-based operation, the output is the same size as the input.

Sample-Based Operation

When the **Frame-based inputs** check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block computes the analytic signal of each of the channel over time.

Frame-Based Operation

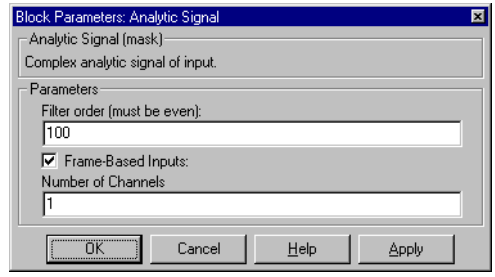
When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 frame matrix:



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix. The block computes the analytic signal of each channel independently.

Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

Dialog Box



Filter order

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

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame based operation, the number of columns (frames) in the input matrix.

See Also

Remez FIR Filter Design

Purpose

Compute the autocorrelation of a vector input.

Library

Vector Functions, in Math Functions

Description

The Autocorrelation block computes the autocorrelation of a vector input. For a length-M input u (indexed from 1 to M), the block output, y, is a vector of length l+1 with elements

$$y(n) = \sum_{k=1}^{M} u(k)u(k+n) \qquad 0 \le n \le l$$

where u is zero when indexed outside of its valid range, and l is specified by the **Maximum positive lag** parameter. A value of `-1` for this parameter indicates that the correlation should be computed for all positive lags (l=M). A matrix input, `u`, is treated as a vector, `u(:)`.

Use the **Scaling** parameter to select from the following correlation scaling options:

- **None**. Outputs the raw autocorrelation with no normalization, as shown for $y(n)$ above. This is the default.
- **Biased**. Outputs the biased estimate of the autocorrelation:

  $$y_{biased}(n) = \frac{y(n)}{M}$$

- **Unbiased**. Outputs the unbiased estimate of the autocorrelation:

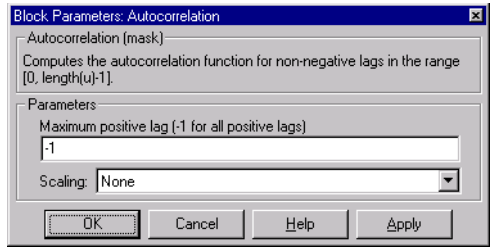
  $$y_{unbiased}(n) = \frac{y(n)}{M - |n|}$$

- **Unity at zero-lag**. Normalizes the estimate of the autocorrelation so that the element at zero lag is identically 1:

  $$y(0) = 1$$

Note If you expect to generate code for the Autocorrelation block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Maximum positive lag

The maximum positive lag for the autocorrelation.

Scaling Ⓣ

The type of scaling for the autocorrelation: **None**, **Biased**, **Unbiased**, or **Unity at zero-lag**. This parameter is not tunable in Simulink's external mode.

See Also

Correlation
`xcorr` (Signal Processing Toolbox)

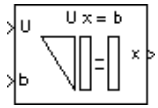
Purpose

Solve the equation U*x*=b for upper triangular matrix U.

Library

Linear Algebra, in Math Functions

Description

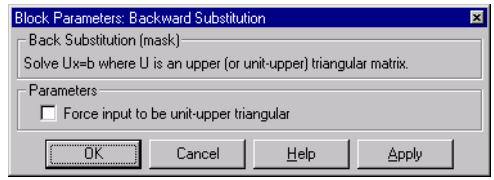


The Backward Substitution block solves the linear system U*x*=b by simple backward substitution of variables, where U is an upper triangular square matrix. The output is the vector solution of the equations, *x*.

The block only uses the elements in the upper triangle of the input matrix; the lower elements are ignored. When **Force input to be unit-upper triangular** is selected, the block replaces the elements on the diagonal of U with ones. 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.

The block may generate `NaN` or `Inf` for underdetermined or inconsistent systems.

Dialog Box



Force input to be unit-upper triangular Ⓣ

Replaces the elements on the diagonal of U with ones when selected.

See Also

Cholesky Solver
Forward Substitution
LDL Solver
Levinson Solver
LU Solver
QR Solver

Biquadratic Filter

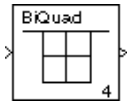
Purpose

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

Library

Filter Realizations, in Filtering

Description



The Biquadratic Filter block applies a cascade of biquadratic filters independently to each input channel. The filter is constructed from L second-order sections, each having a quadratic numerator and denominator. The biquadratic filter is useful for reduced precision implementations because the coefficients are bounded between ±2 for typical minimum-phase designs. This reduces scaling and coefficient sensitivity problems.

The **SOS matrix** parameter specifies the filter coefficients as a second-order section matrix of the type produced by the `tf2sos` and `ss2sos` functions in the Signal Processing Toolbox. This is an L-by-6 matrix,

$$\begin{bmatrix} b_{01} & b_{11} & b_{21} & a_{01} & a_{11} & a_{21} \\ b_{02} & b_{12} & b_{22} & a_{02} & a_{12} & a_{22} \\ \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\ b_{0L} & b_{1L} & b_{2L} & a_{0L} & a_{1L} & a_{2L} \end{bmatrix}$$

whose rows contain the numerator and denominator coefficients b_{ik} and a_{ik} of each second-order section in $H(z)$:

$$H(z) = \prod_{k=1}^{L} H_k(z) = \prod_{k=1}^{L} \frac{b_{0k} + b_{1k}z^{-1} + b_{2k}z^{-2}}{a_{0k} + a_{1k}z^{-1} + a_{2k}z^{-2}}$$

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.

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

- *Scalar* to be repeated for all filter states in all channels. An empty vector, `[]`, is the same as the scalar value `0`.
- *Vector* of length 2∗L to initialize the filter states for all channels. Each pair of elements in the vector specifies z^{-1} and z^{-2} for a particular second-order

section. For example, `ic(1)` and `ic(2)` respectively specify z^{-1} and z^{-2} for $H_1(z)$ in every channel, where `ic` is the initial condition vector.

- *Matrix* of dimension (2∗L)-by-N containing the initial filter states for each of the N channels. Each pair of elements in a column specifies z^{-1} and z^{-2} for a particular second-order section of the corresponding channel. For example, `ic(1,3)` and `ic(2,3)` respectively specify z^{-1} and z^{-2} for $H_1(z)$ in the third channel, where `ic` is the initial condition matrix.

You can choose frame-based or sample-based operation by selecting (or deselecting, respectively) the **Frame-based inputs** check box.

Sample-Based Operation

When the **Frame-based inputs** check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel over time.

Frame-Based Operation

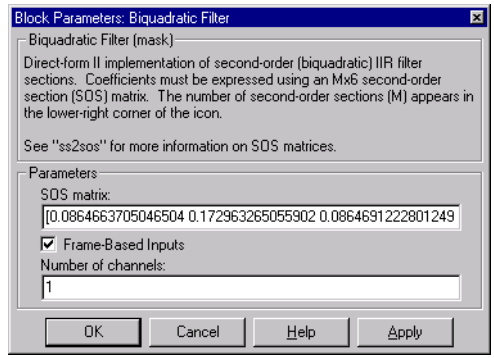
When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals, N, in the matrix. The block independently applies the filter to each channel.

In both sample-based and frame-based modes, the output is the same size as the input.

Note If you expect to generate code for the Biquadratic Filter block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Biquadratic Filter

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`.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (frames) in the input matrix.

See Also

Direct-Form II Transpose Filter
Filter Realization Wizard
Overlap-Add FFT Filter
Overlap-Save FFT Filter
Time-Varying Direct-Form II Transpose Filter
Time-Varying Lattice Filter
`filter` (MATLAB)
`sosfilt` (Signal Processing Toolbox)
`tf2sos` (Signal Processing Toolbox)

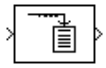
Purpose

Buffer a sequence of scalar inputs into a frame.

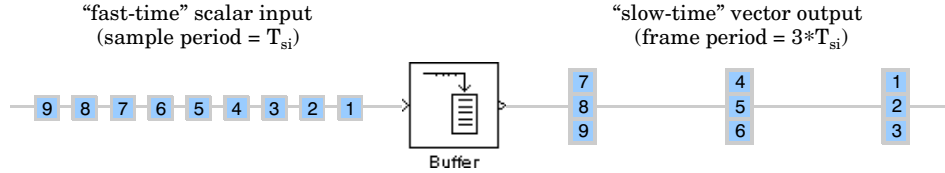
Library

Buffers, in General DSP

Description



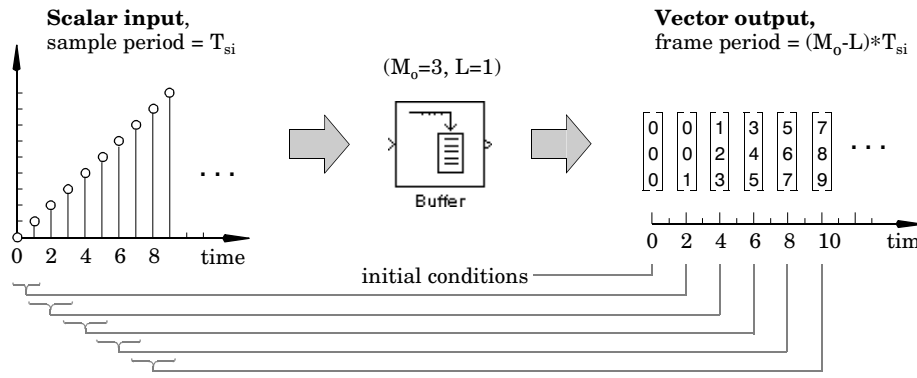
The Buffer block acquires a sequence of input samples into a frame, which is an efficient format for many DSP applications. The frame-based output generally has a slower rate than the input.



To rebuffer frame-based inputs to a larger or smaller frame size, use the Rebuffer block.

The block coordinates the output *frame size* and *frame rate* so that the *sequence sample period* (i.e., the sample-to-sample interval) is the same at both the input and output, T_{so}=T_{si}.

Scalar Inputs. Scalar inputs are buffered into a frame vector (top illustration). The length of the output frame, M_o, is determined by the **Buffer size** parameter. The **Buffer overlap** parameter specifies the number of samples, L, from the previous buffer to include in the current buffer. The number of *new* input samples the block acquires before propagating the buffered data to the output is the difference between the **Buffer size** and **Buffer overlap**, M_o-L.

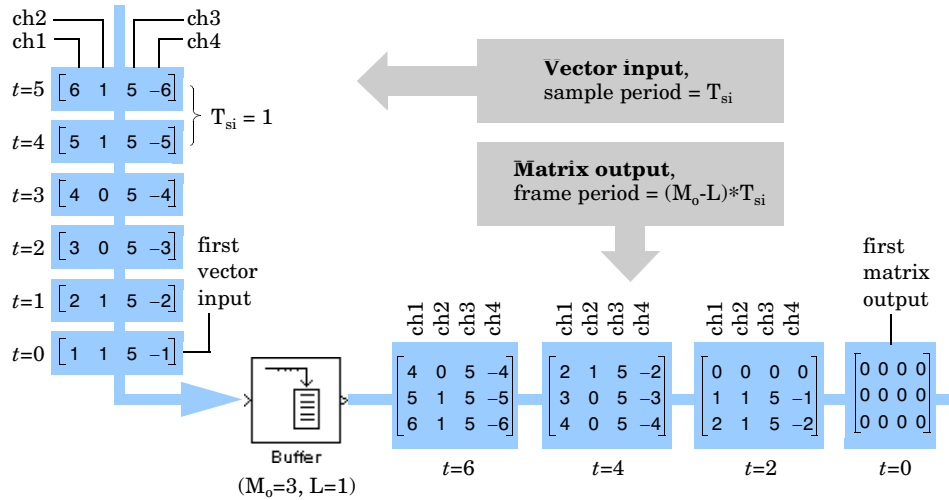


The output frame period is $(M_o\text{-}L)*T_{si}$, which is *equal* to the sequence sample period, T_{si}, when the **Buffer overlap** is M_o-1. For negative **Buffer overlap** values, the block simply discards L input samples after the buffer fills, and outputs the buffer with period $(M_o\text{-}L)*T_{si}$,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

Note To multiplex independent scalar channels into a sample vector, use the Simulink Mux block.

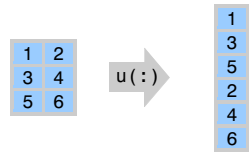
Vector Inputs. Length-N sample vector inputs are buffered into a M_o-by-N matrix, where M_o is specified by the **Buffer size** parameter. Each of the N vector elements represent a distinct channel.

The illustration below shows the block's initial operations on a 4-channel sample vector input with a **Buffer size** of 3 and a **Buffer overlap** of 1.



Note that the input sample vectors do not begin appearing at the output until the second row of the second matrix. The first output matrix (all zeros in this example) reflects the block's initial buffer, while the first row of zeros in the second output is a result of the one-sample overlap between consecutive output frames.

Matrix inputs. An M-by-N matrix input is treated as a single vector with M*N elements. In other words, the matrix input `u` is reshaped to the vector input `u(:)`.

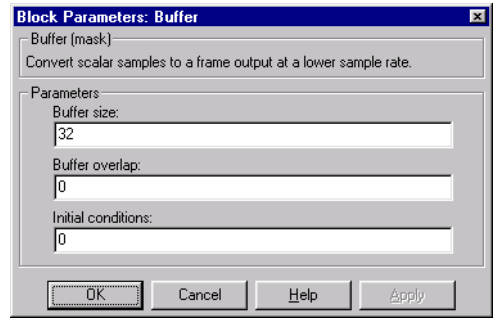


Initial Conditions

The Buffer block's buffer is initialized to the value specified by the **Initial condition** parameter, and the block always outputs this buffer at the first simulation step (t=0). If 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. If 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.

Note If you expect to generate code for the Buffer block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Buffer size

The number of consecutive samples to buffer into the output frame, M_o.

Buffer overlap

The number of samples by which consecutive outputs overlap, L.

Initial conditions

The value of the block's initial output, a scalar, vector, or matrix.

See Also

Distributor
Rebuffer
Shift Register
Unbuffer

Purpose

Compute and display the frequency content of an input sequence.

Library

DSP Sinks

Description

The Buffered FFT Frame Scope block acquires a sequence of input samples into a buffer, and displays the magnitude of the FFT of each full buffer.

The block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block buffers and displays the data in each channel independently.

The number of input samples that the block buffers before computing and displaying the magnitude FFT is specified by the **Buffer size** (M_o) parameter. The **Buffer overlap** (L) parameter specifies 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 **Buffer overlap**, M_o-L.

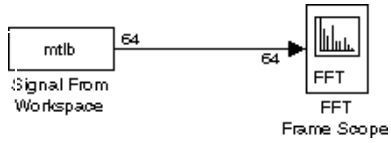
The display update period is $(M_o\text{-}L)*T_s$, where T_s is the input sample period, and is *equal* to the input sample period when the **Buffer overlap** is M_o-1. For negative **Buffer overlap** values, the block simply discards the appropriate number of input samples after the buffer fills, and updates the scope display at a slower rate than the zero-overlap case.

The **FFT length** parameter, N_{fft}, specifies the number of samples on which to perform the FFT. A value of `-1` instructs the block to use the buffer size as the FFT size. Otherwise, the block zero pads or truncates every channel's buffer to N_{fft} before computing the FFT.

In order to correctly scale the frequency axis (i.e., to determine the frequencies against which the transformed input data should be plotted), the block needs to know the actual sample period of the time-domain input. The **Sample time of original time-series** parameter allows you to specify this in two different ways:

Auto-Detect from Input Sample Period. A value of `-1` for this parameter instructs the block to compute the frequency data from the sample period of the input to the block. This parameter setting is appropriate when the input to the block is the original signal, with no samples added or deleted (by insertion of zeros, for example).

The auto-detect mode also makes the assumption that the sample period of the original time-domain signal in the simulation is equal to the period with which the physical signal was originally sampled. For example, the `mtlb` signal imported from the workspace in the model below has an actual sample period of `1/Fs = 1/7418`.



Although the **Sample time** parameter in the Signal From Workspace block can legitimately be set to any value, for the auto-detect mode to compute the correct frequency scaling, the **Sample time** parameter *must* be set to the "real-world" sample period of `1/7418`.

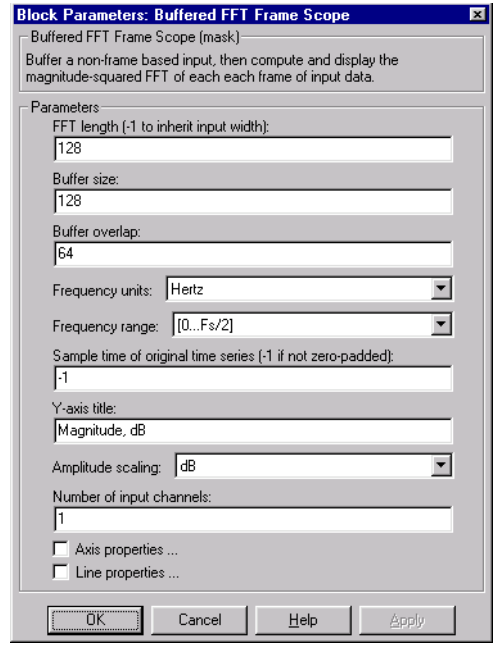
Enter the Appropriate Time-Domain Sample Period. Enter the sample period of the original time-series, T_s. This is necessary only when the input to the block is not the original signal, but a zero-padded or otherwise rate-altered version.

The **Frequency units** parameter specifies whether the frequency axis values should be in units of Hertz or rads/sec, and 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 time-domain signal's actual sample frequency ($F_s/2$ is the Nyquist frequency). If the **Frequency units** parameter specifies **Hertz**, the spacing between frequency points is $1/(N_{fft}T_s)$. For **Frequency units** of **rads/sec**, the spacing between frequency points is $2\pi/(N_{fft}T_s)$.

Note that all of the FFT-based blocks in the DSP Blockset, 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.

For information about the scope window, as well as the **Axis properties** and **Line properties** panels in the dialog box, see the reference page for the Time Frame Scope block.

Dialog Box



FFT length

The number of samples on which to perform the FFT. If the **FFT length** differs from the buffer size, the data is zero-padded or truncated as needed.

Buffer size

The number of signal samples to include in each buffer.

Buffer overlap

The number samples by which consecutive buffers overlap.

Frequency units Ⓣ

The frequency units for the *x*-axis, **Hertz** or **rads/sec**.

Frequency range Ⓣ

The frequency range over which to plot the data, **[0..Fs/2]**, **[-Fs/2..Fs/2]**, or **[0..Fs]**, where F_s is the sample frequency of the original time-domain signal, $1/T_s$.

Sample time of original time series

The sample period, T_s, of the original time-domain signal. Set to `-1` to auto-detect the signal sample period from the sample period of the block input.

Y-Axis title Ⓣ

The text to be displayed to the left of the *y*-axis.

Amplitude scaling Ⓣ

The scaling for the *y*-axis, **dB** or **Magnitude**.

Number of input channels

The number of channels (columns) in the input matrix.

Axis properties Ⓣ

Select to expose the **Axis Properties** panel. See Time Frame Scope for more information.

Line properties Ⓣ

Select to expose the **Line Properties** panel. See Time Frame Scope for more information.

See Also

FFT
FFT Frame Scope
Frequency Frame Scope
Time Frame Scope
User-Defined Frame Scope

Purpose

Compute an estimate of AR model parameters using the Burg method.

Library

Parametric Estimation, in Estimation

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 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.

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

The order, p, of the all-pole model is specified by the **Order** parameter.

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

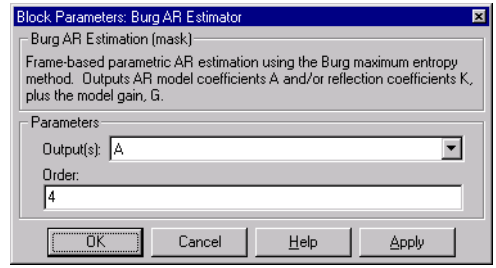
- **A** – The block outputs the normalized estimate of the AR model polynomial coefficients in descending powers of z,

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

- **K** – The block outputs 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`).

Dialog Box



Output(s)

The realization to output, model coefficients or reflection coefficients.

Order

The order of the AR model.

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.

See Also

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

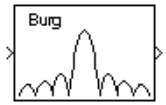
Purpose

Compute a parametric spectral estimate using the Burg method.

Library

Power Spectrum Estimation, in Estimation

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 spectrum is then computed from the FFT of the estimated AR model parameters.

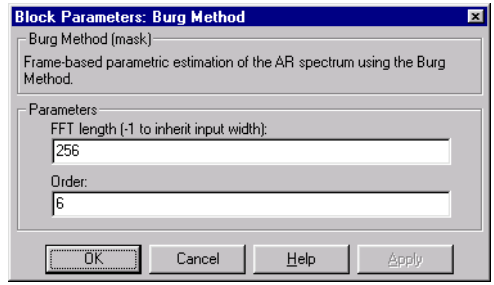
The order of the all-pole model is specified by the **Order** parameter. The Burg Method and Yule-Walker Method blocks return similar results for large frame sizes.

The input is a frame of consecutive time samples; a matrix input, `u`, is also treated as a single frame, `u(:)`. The block's output is the estimate of the signal's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where N_{fft} is specified as a power of 2 by the **FFT Size** parameter and F_s is the signal's sample frequency. A value of `-1` for **FFT size** instructs the block to use the input frame size as the FFT size. Otherwise, the block zero pads or truncates the input to N_{fft} before computing the FFT.

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	
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

Dialog Box



FFT size

The number of samples on which to perform the FFT, N_{fft}. If N_{fft} exceeds the frame size, the data is zero padded as needed.

Order

The order of the AR model.

References

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

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

See Also

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

Chirp

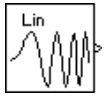
Purpose

Generate a swept-frequency cosine.

Library

DSP Sources

Description



The Chirp block outputs a unity-amplitude swept-frequency cosine (*chirp*) signal. The frequency trajectory of the output signal is defined by two instantaneous frequency values. The instantaneous output frequency is initialized to the **Initial frequency** parameter value at the start of the simulation, and then varies continuously to the **Frequency at target time** parameter value. The transition between these two frequencies takes place over the period of time specified in the **Target time** parameter.

The frequency trajectory established by these three parameters is maintained until the end of the simulation. That is, the frequency of the output (after the target time) continues to change according to the specified trajectory. Note that the output of this block is continuous.

The method that the block uses to transition between the specified instantaneous frequencies is set by the **Frequency sweep** parameter, and can be **Linear**, **Quadratic**, or **Logarithmic**:

- **Linear** uses an instantaneous frequency sweep $f_i(t)$ of

  $$f_i(t) = f_0 + \beta t$$

  where f_0 is the **Initial frequency** parameter value, and

  $$\beta = (f_1 - f_0)/t_1$$

  β ensures that the desired **Frequency at target time** value, f_1, occurs at the specified **Target time**, t_1.
- **Quadratic** uses an instantaneous frequency sweep $f_i(t)$ of

  $$f_i(t) = f_0 + \beta t^2$$

  where

  $$\beta = (f_1 - f_0)/t_1$$
- **Logarithmic** uses an instantaneous frequency sweep $f_i(t)$ of

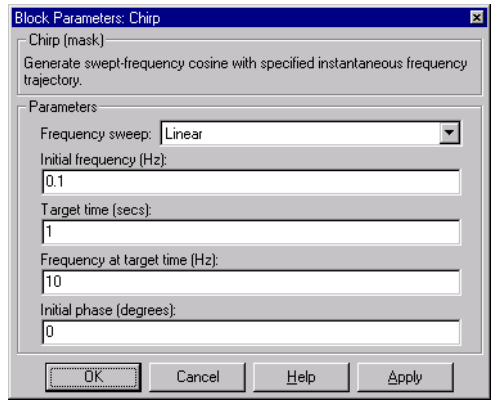
$$f_i(t) = f_0 + 10^{\beta t}$$

where

$$\beta = [\log(f_1 - f_0)]/t_1$$

For the logarithmic sweep, the **Frequency at target time** parameter value must be greater than the **Initial frequency** parameter value.

Dialog Box



Frequency sweep Ⓣ

Function defining the instantaneous frequency trajectory.

Initial frequency Ⓣ

Frequency of output cosine at t=0.

Target time Ⓣ

Time corresponding to the second defining frequency, specified by **Frequency at target time**.

Frequency at target time Ⓣ

Second frequency defining the sweep trajectory.

Initial phase Ⓣ

Phase of the cosine output at t=0.

See Also

Signal From Workspace
Signal Generator (Simulink)
Sine Wave
`chirp` (Signal Processing Toolbox)

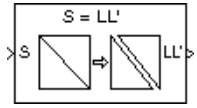
Purpose

Factor a Hermitian positive definite matrix into triangular components.

Library

Linear Algebra, in Math Functions

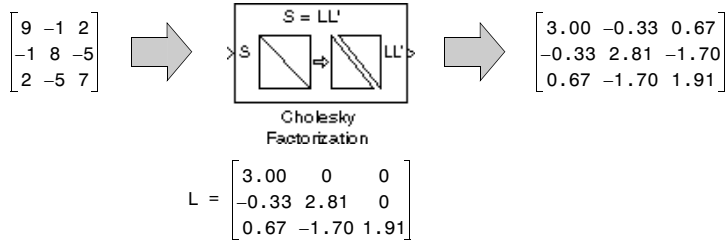
Description



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

$$S = LL^H$$

where L is a lower triangular square matrix with positive diagonal elements and L^H denotes the Hermitian transpose of L. The block's output is a matrix whose lower triangle is L and whose upper triangle is L^H.



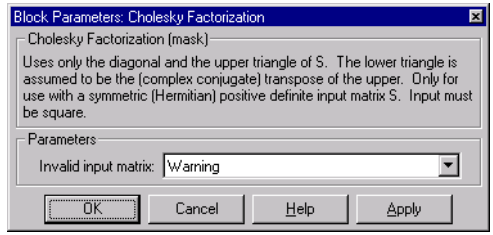
Note that L and L^H share the same diagonal in the output matrix. Cholesky factorization requires half the computation of Gaussian elimination (LU decomposition), and is always stable.

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 **Invalid input matrix** parameter. The following options are available:

- **Ignore** – Proceed with the computation and *do not* issue an alert. The output is not a valid factorization.
- **Warning** – Display a warning message in the MATLAB command window, and continue the simulation.
- **Error** – Display an error dialog box and terminate the simulation.

Cholesky Factorization

Dialog Box



Invalid input matrix Ⓣ

Response to non-positive definite matrix inputs.

References

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

See Also

Backward Substitution
Cholesky Solver
LDL Factorization
LU Factorization
QR Factorization

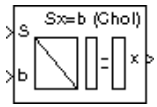
Purpose

Solve the equation S*x*=b for Hermitian positive definite matrix S.

Library

Linear Algebra, in Math Functions

Description



The Cholesky Solver block solves the linear system S*x*=b by applying Cholesky factorization to matrix S (top input), which must be square and Hermitian positive definite. The bottom input is the right-hand-side of the equation, b. The output is the unique solution of the equations, *x*.

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

$$S = LL^H$$

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

The equation S*x*=b then becomes

$$LL^H x = b$$

which is solved for *x* by making the substitution $y = L^H x$, and solving the following two triangular systems by forward and backward substitution, respectively:

$$Ly = b$$

$$L^H x = y$$

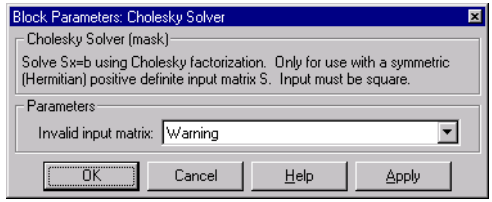
The algorithm requires that the input be square and Hermitian positive definite. When the input is not positive definite, the block reacts with the behavior specified by the **Invalid input matrix** parameter. The following options are available:

- **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.
- **Error** – Display an error dialog box and terminate the simulation.

The block may generate `NaN` or `Inf` for underdetermined or inconsistent (overdetermined) systems.

Cholesky Solver

Dialog Box



Invalid input matrix

Response to non-positive definite matrix inputs.

See Also

Backward Substitution
Cholesky Factorization
LDL Solver
LU Solver
QR Solver

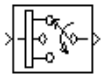
Purpose

Convert a vector input to a scalar output (parallel-to-serial conversion).

Library

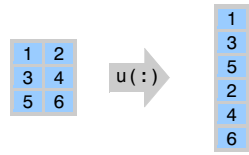
Switches and Counters, in General DSP

Description



The Commutator block sequentially samples the N elements of an input vector (representing N channels), and outputs each sample in succession in a scalar sequence with a sample rate N times faster than the input vector sample rate.

A matrix input, `u`, is reshaped to a vector input, `u(:)`, before sampling.



The Commutator block is initialized with a full buffer of zeros, which it begins to output at the start of the simulation. Inputs to the block are therefore delayed by one input sample period (N elements, or T_{si} seconds).

The Commutator block is functionally equivalent to an Unbuffer block for vector inputs.

Dialog Box

See Also

Distributor
Unbuffer

Complex Cepstrum

Purpose

Compute the complex cepstrum of an input.

Library

Transforms, in General DSP

Description

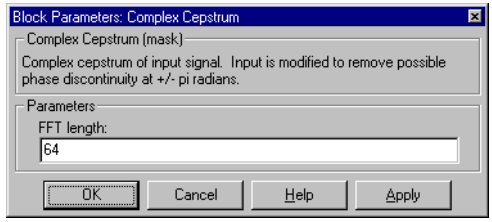
The Complex Cepstrum block computes the complex cepstrum of a real input frame:

```
y = cceps(x)       % equivalent MATLAB code
```

The input is altered, by the application of a linear phase term, to have no phase discontinuity at ±π radians. That is, the input is circularly shifted after zero padding to have zero phase at π radians. The output is real.

Multichannel inputs (i.e., frame matrices) are not accepted.

Dialog Box



FFT length

The number of samples to use in computing the FFT.

See Also

DCT
FFT
Real Cepstrum
`cceps` (Signal Processing Toolbox)

Purpose

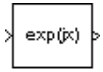
Compute the complex exponential function.

Library

Elementary Functions, in Math Functions

Description

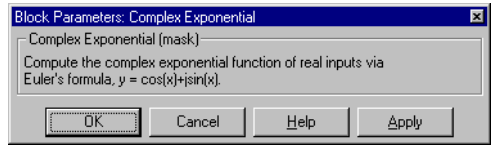
The Complex Exponential block computes the complex exponential function for each element of a *real* input, u.



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

The output is complex, with the same size as the input.

Dialog Box



See Also

Math Function (Simulink)
`exp` (MATLAB)

Constant Diagonal Matrix

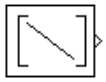
Purpose

Generate a square, diagonal matrix.

Library

Matrix Functions, in Math Functions; DSP Sources

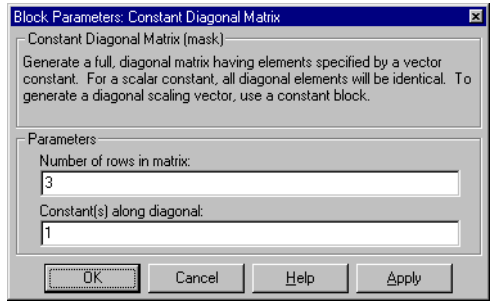
Description



The Constant Diagonal Matrix block outputs a square diagonal matrix constant. The **Number of rows in matrix** parameter sets the matrix size, and 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`.

If the vector specified for the **Constant along diagonal** parameter is larger than the specified number of rows in the matrix, the extra values at the end of the vector are ignored. If the vector contains fewer values than there are diagonal elements, the vector is padded with the correct number of zeros.

Dialog Box



Number of rows in matrix

The number of rows (and columns) in the square matrix output.

Constant(s) along diagonal Ⓣ

The values of the elements along the diagonal, as a scalar or vector.

See Also

Create Diagonal Matrix
Matrix Constant
`diag` (MATLAB)

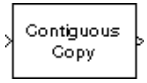
Purpose

Recreate the input in a contiguous block of memory (for code generation).

Library

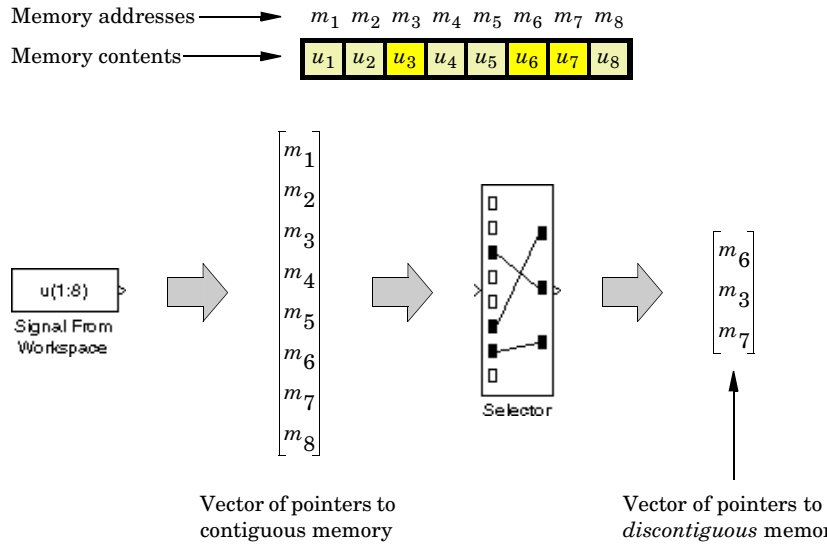
Elementary Functions, in Math Functions

Description

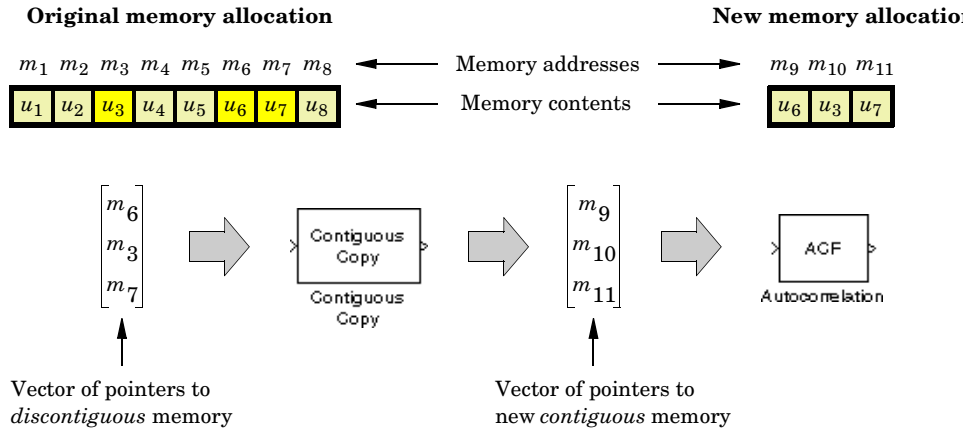


The Contiguous Copy block copies the input to a contiguous block of memory, and passes this new copy to the output. The output is identical to the input, but is guaranteed to reside in a contiguous section of memory.

Because Simulink employs an efficient copy-by-reference method for propagating data in a model, many operations produce outputs with discontiguous memory locations. An example of such an operation is shown below with the Simulink Selector block:



Although this does not present a problem during simulation, a number of blocks in the DSP Blockset require contiguous inputs for code-generation with the Real-Time Workshop (RTW). When these blocks (listed below) 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. The Autocorrelation block is an example of one that requires contiguous inputs for code generation:



Blocks Requiring Contiguous Inputs for Code Generation

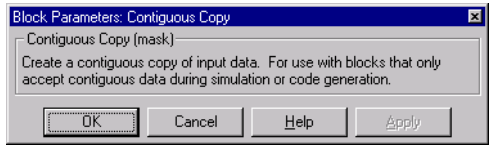
The table below lists the blocks requiring contiguous inputs for code generation. In some cases, a block requires contiguous inputs for one mode of operation, but not for another. This is noted in the right column of the table where appropriate.

Block Library	Block Name	Notes
DSP Sources	*none*	
DSP Sinks	Signal To Workspace	Frame-based operation
Elementary Functions	*none*	
Vector Functions	Autocorrelation	
	Convolution	
	Difference	
Matrix Functions	Create Diagonal Matrix	
	Extract Diagonal	
	Extract Triangular Matrix	
	Matrix Multiplication	most inputs
	Matrix Scaling	
	Permute Matrix	
	Toeplitz	
	Transpose	matrix inputs
Linear Algebra	*none*	

Block Library	Block Name	Notes
Statistics	Histogram	running mode only
	Maximum	running mode only
	Mean	running mode only
	Median	
	Minimum	running mode only
	RMS	running mode only
	Sort	
	Standard Deviation	running mode only
	Variance	running mode only
Signal Operations	Integer Delay	
	Variable Fractional Delay	
	Variable Integer Delay	
	Zero Pad	
Transforms	*none*	
Buffers	Buffer	
	Partial Unbuffer	
	Rebuffer	
	Shift Register	
	Triggered Shift Register	
	Unbuffer	
Switches and Counters	*none*	
Parametric Estimation	*none*	
Power Spectrum Estimation	*none*	
Filter Designs	*none*	
Filter Realizations	Biquadratic Filter	
	Direct-Form II Transpose Filter	
	Time-Varying Direct-Form II Transpose Filter	
	Time-Varying Lattice Filter	
Adaptive Filters	*none*	
Multirate Filters	Dyadic Analysis Filter Bank	
	Dyadic Synthesis Filter Bank	
	FIR Decimation	
	FIR Interpolation	
	FIR Rate Conversion	

Contiguous Copy

Dialog Box



See Also

Convert Complex Simulink To DSP

Purpose

Convert complex data from the DSP Blockset v2.2 format to the Simulink v3 format.

Library

Elementary Functions, in Math Functions

Description

The Convert Complex DSP To Simulink block accepts complex data (scalar, vector, matrix) in the DSP Blockset 2.2 format, and outputs the same data in the Simulink 3 complex format. Only complex data should be supplied to this block.

Blocks provided in Release 11 and later blocksets (e.g., Simulink 3.0, DSP Blockset 3.0, Fixed Point Blockset 2.0) use the Simulink 3 complex format, which is not compatible with the DSP Blockset 2.2 complex format. To add a new block or subsystem (Release 11 and later) to an existing model that uses the DSP Blockset 2.2 complex data format, precede it with the Convert Complex DSP To Simulink block. If the new block or subsystem's output is complex, you should follow it with the complementary Convert Complex Simulink To DSP block (unless the downstream blocks have already been updated to their Release 11+ counterparts).

These convertor blocks are only needed for interfacing v3.0 blocks to the *complex-data* section of a v2.2 or earlier model. Version 3.0 blocks can be added to *real-data* sections of older models without any data format conversion.

Note Within a section of model that uses the v2.2 complex format, you should continue to use the complex port identifier (∗) as a guide to wiring blocks. Outputs ports labeled with the ∗ symbol should only be connected to input ports labeled with the ∗ symbol.

The following figure shows how you can use these two convertor blocks to migrate part of a complex-data model to the v3.0 complex format while letting other components continue to use the v2.2 complex-data format.

Existing (v2.2) complex-data model:

Subsystem B upgraded to v3.0 complex-data format:

(Subsystem A remains a v2.2 implementation)

Subsystem A upgraded to v3.0 complex-data format:

(Subsystem B remains a v2.2 implementation)

Dialog Box

See Also

Convert Complex Simulink To DSP

Purpose

Convert complex data from the Simulink v3 format to the DSP Blockset v2.2 format.

Library

Elementary Functions, in Math Functions

Description

The Convert Complex Simulink To DSP block accepts complex data (scalar, vector, matrix) in the Simulink 3 format, and outputs the same data in the DSP Blockset 2.2 complex format. Only complex data should be supplied to this block.

Blocks provided in Release 11 and later blocksets (e.g., Simulink 3.0, DSP Blockset 3.0, Fixed Point Blockset 2.0) use the Simulink 3 complex format, which is not compatible with the DSP Blockset 2.2 complex format. To add a new block or subsystem (Release 11 and later) to an existing model that uses the DSP Blockset 2.2 complex data format, precede it with the Convert Complex DSP To Simulink block. If the new block's output is complex, you should then follow it with the Convert Complex Simulink To DSP block (unless the downstream blocks have already been updated to their Release 11+ counterparts).

These convertor blocks are only needed for interfacing v3.0 blocks to the *complex-data* section of a v2.2 or earlier model. Version 3.0 blocks can be added to *real-data* sections of older models without any data format conversion.

Note Within a section of model that uses the v2.2 complex format, you should continue to use the complex port identifier (∗) as a guide to wiring blocks. Outputs ports labeled with the ∗ symbol should only be connected to input ports labeled with the ∗ symbol.

The following figure shows how you can use these two convertor blocks to migrate part of a complex-data model to the v3.0 complex format while letting other components continue to use the v2.2 complex-data format.

Existing (v2.2) complex-data model:

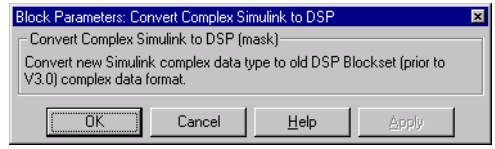
Subsystem B upgraded to v3.0 complex-data format:

(Subsystem A remains a v2.2 implementation)

Subsystem A upgraded to v3.0 complex-data format:

(Subsystem B remains a v2.2 implementation)

Dialog Box



See Also

Convert Complex DSP To Simulink

Purpose

Compute the convolution of two vectors.

Library

Vector Functions, in Math Functions

Description

The Convolution block computes the convolution of two input vectors independently at each time step. For a length-M input vector u (indexed from 1 to M) and a length-N input vector v (indexed from 1 to N), the block output is a vector of length M+N-1 with elements

$$y(n) = \sum_{k=1}^{max(M,N)} u(k)v^*(n-k+1) \qquad 1 \le n \le M+N-1$$

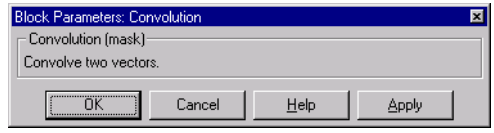
where $*$ denotes conjugation, and u and v are zero when indexed outside of their valid ranges.

When both inputs are real, the output is real as well. When one or both inputs are complex, the output is complex.

A matrix input, `u`, is treated as a vector, `u(:)`.

Note If you expect to generate code for the Convolution block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



See Also

Correlation
`conv` (MATLAB)

Correlation

Purpose

Compute the correlation of two vectors.

Library

Vector Functions, in Math Functions

Description

The Correlation block computes the cross-correlation of the two input vectors independently at each time step. For a length-M input vector u (indexed from 1 to M) and a length-N input vector v (indexed from 1 to N), the block output is a vector of length M+N-1 with elements

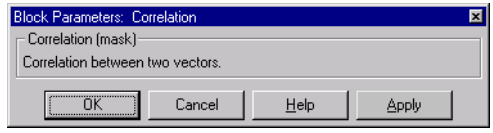
$$y(n) = \sum_{k=1}^{max(M,N)} u(k)v^*(k+n-1) \qquad 1 \le n \le M+N-1$$

where $*$ denotes conjugation, and u and v are zero when indexed outside of their valid ranges.

When both inputs are real, the output is real as well. When one or both inputs are complex, the output is complex.

A matrix input, `u`, is treated as a vector, `u(:)`.

Dialog Box



See Also

Autocorrelation
Convolution
`xcorr` (Signal Processing Toolbox)

Purpose

Count up or down through a specified range of numbers.

Library

Switches and Counters, in General DSP

Description

The Counter block increments or decrements an internal counter each time it receives a trigger event at the top input port (`Clk`). A trigger event at the bottom input port (`Rst`) resets the counter to its initial state. The triggering event for both inputs is specified by the **Count event** pop-up menu, and can be one of the following:

- **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.
- **Nonzero sample** triggers execution of the block at each sample time that the trigger input is not zero.
- **Free running** disables the trigger (`Clk`) port and enables the **Sample time** parameter. The block increments or decrements the counter at a constant interval specified by the **Sample time** parameter.

At the start of the simulation, the block sets the counter to the value specified by the **Initial count** parameter, which can be any value in the range defined by the **Counter size** parameter. The **Counter size** parameter allows you to choose from three standard counter ranges, or specify an arbitrary counter limit:

- **8 bits** specifies a 256-step (2^8) counter with a range of 0 to 255.
- **16 bits** specifies a 65536-step (2^{16}) counter with a range of 0 to 65535.
- **32 bits** specifies a 2^{32}-step 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 upper count limit. The range of the counter is then 0 to the **Maximum count** value.

When the **Count direction** parameter is set to **Up**, a 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 (e.g., 255 for an 8-bit counter). At the next trigger event, the block resets the counter to 0, and resumes incrementing the counter with the subsequent trigger event.

When the **Count direction** parameter is set to **Down**, a 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 trigger event, the block resets the counter to the upper count limit (e.g., 255 for an 8-bit counter), and resumes decrementing the counter with the subsequent trigger event.

Between triggering events the block holds the output at its last value. The block resets the counter to its initial state when the trigger event specified in the **Count event** pop-up menu is received at the optional `Rst` input. When trigger events are received simultaneously at the `Clk` and `Rst` inputs, 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 remove the `Rst` port by deselecting the **Reset input** check box.

You can change the configuration of output ports on the block icon from the **Output** pop-up menu. Three options are available:

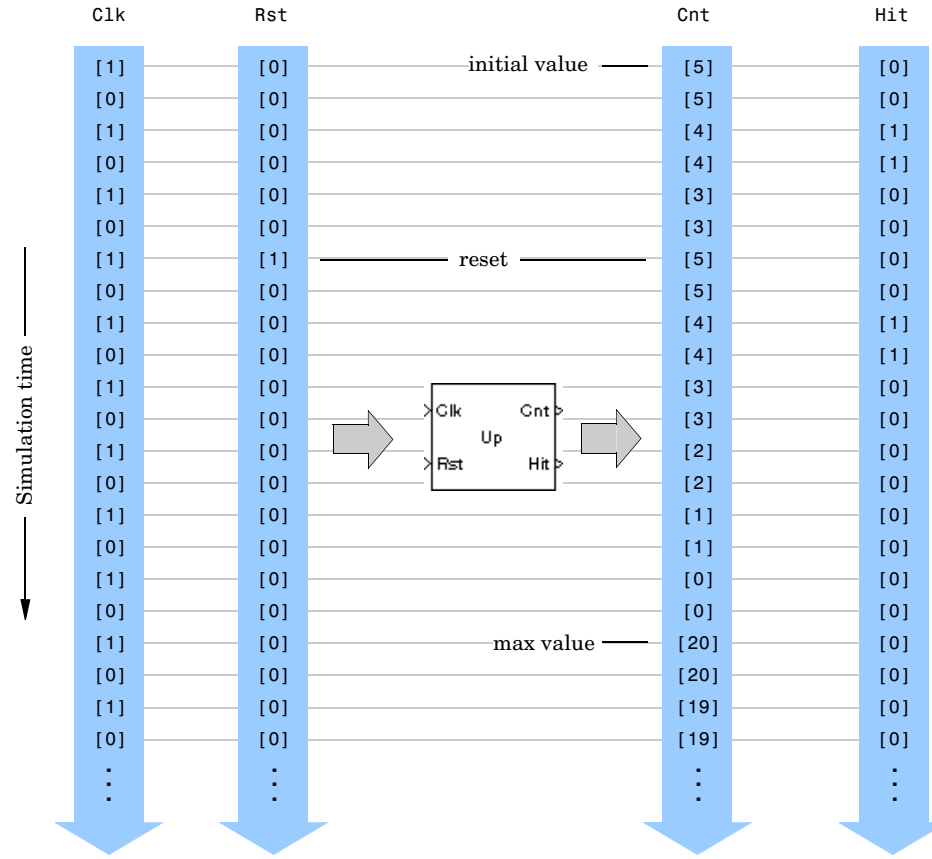
- **Count** configures the block icon to show a `Cnt` port, which outputs the current value of the counter.
- **Hit** configures the block icon to show a `Hit` port. The `Hit` port outputs zeros while the value of the counter does not equal the **Hit value** parameter setting. When the counter value does equal the **Hit value** setting, the block outputs a 1 at the `Hits` port.
- **Count and Hit** configures the block icon with both ports. This is the default.

All of the counter parameter settings must be integer values, and both inputs must be real.

In the figure below, the `Clk` input of the Counter block is driven by Simulink's Discrete Pulse Generator block. The Counter block settings are:

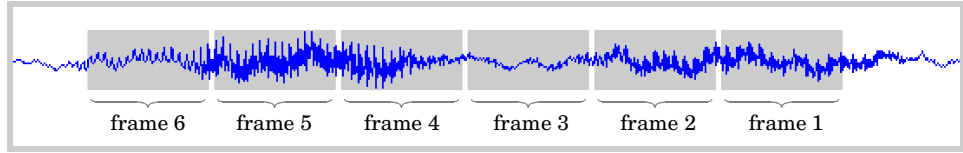
- **Count direction** = **Down**
- **Count event** = **Rising edge**
- **Counter size** = **User defined**
- **Maximum count** = `20`

- **Initial count** = 5
- **Output** = **Count and Hit**
- **Hit value** = 4



Frame-Based Operation

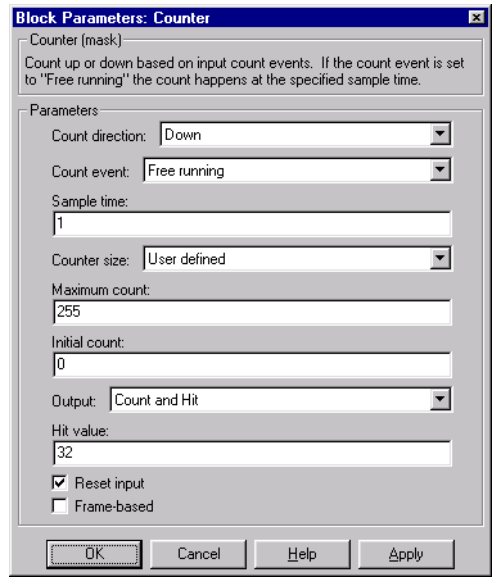
When the **Frame-based inputs** check box is selected, the block interprets each vector input as a frame of data from one channel. Multichannel inputs (i.e., frame matrices) are not accepted.



At each sample time it increments or decrements the counter by the number of trigger events contained in the 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 frame. The output (the block's count) is still a scalar with the same period as the input.

The **Frame-based inputs** check box is disabled along with the `Clk` port in free-running mode.

Dialog Box



Count direction

The counter direction, **Up** or **Down**. This parameter is not tunable in Simulink's 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, respectively). **Free running** disables the `Clk` port, and counts continuously with the period specified by the **Sample time** parameter.

Sample time

The output sample period in free-running mode.

Counter size

The range of integer values the block should count through before recycling to zero.

Maximum count

The counter's maximum value when **Counter size** is set to **User defined**.

Initial count Ⓣ

The counter's initial value at the start of the simulation and after reset. This parameter is not tunable in Simulink's 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 not tunable in Simulink's external mode.

Reset input

Enable the `Rst` input port.

Frame-based

Selects frame-based operation.

See Also

Edge Detector
N-Sample Enable
N-Sample Switch

Covariance AR Estimator

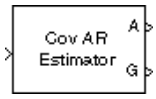
Purpose

Compute an estimate of AR model parameters using the covariance method.

Library

Parametric Estimation, in Estimation

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 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.

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

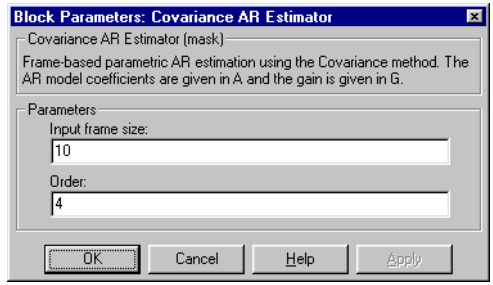
The order, p, of the all-pole model is specified by the **Order** parameter.

The top output, `A`, 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`).

Dialog Box



Input frame size

The number of samples in the input frame.

Order

The order of the AR model.

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.

See Also

Burg AR Estimator
Covariance Method
Modified Covariance AR Estimator
Yule-Walker AR Estimator
`arcov` (Signal Processing Toolbox)

Covariance Method

Purpose

Compute a parametric spectral estimate using the covariance method.

Library

Power Spectrum Estimation, in Estimation

Description

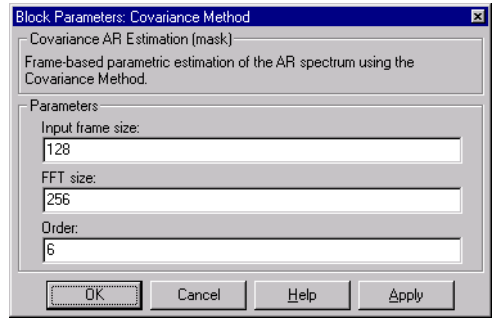
The Covariance Method block estimates the power spectral density (PSD) of the input using the covariance method. This method fits an autoregressive (AR) model to the signal by minimizing the forward prediction error in the least-squares sense. The spectrum is then computed from the FFT of the estimated AR model parameters.

The order of the all-pole model is specified by the **Order** parameter.

The input is a frame of consecutive time samples; a matrix input, `u`, is also treated as a single frame, `u(:)`. The block's output is the estimate of the signal's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where N_{fft} is specified as a power of 2 by the **FFT Size** parameter and F_s is the signal's sample frequency. A value of `-1` for **FFT size** instructs the block to use the input frame size as the FFT size. Otherwise, the block zero pads or truncates the input to N_{fft} before computing the FFT.

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



Input frame size

The number of samples in the input frame.

FFT size

The number of samples on which to perform the FFT, N_{fft}. If N_{fft} exceeds the frame size, the data is zero padded as needed.

Order

The order of the AR model.

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.

See Also

Burg Method
Covariance AR Estimator
Short-Time FFT
Modified Covariance Method
Yule-Walker Method
`pcov` (Signal Processing Toolbox)

Create Diagonal Matrix

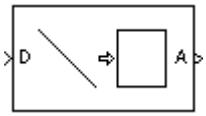
Purpose

Create a matrix from a vector diagonal.

Library

Matrix Functions, in Math Functions

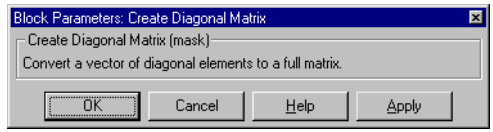
Description



The Create Diagonal Matrix block creates a square matrix from the diagonal specified by the vector input. The elements of the length-M input vector are used to populate the diagonal of an M-by-M matrix output. The elements off the diagonal are zero. A matrix input, `u`, is treated as a vector, `u(:)`.

Note If you expect to generate code for the Create Diagonal Matrix block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



See Also

Constant Diagonal Matrix
Extract Diagonal

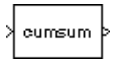
Purpose

Compute the cumulative sum of the elements in a vector.

Library

Vector Functions, in Math Functions

Description

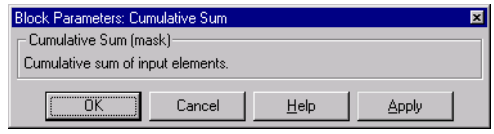


The Cumulative Sum block finds the cumulative sum of all elements in the input vector. For a length-M input vector u (indexed from 1 to M), the output of the Cumulative Sum block is a length-M vector with elements

$$y(n) = \sum_{k=1}^{n} u(k), \qquad 1 \le n \le M$$

A matrix input, `u`, is treated as a vector, `u(:)`.

Dialog Box



See Also

Difference
`cumsum` (MATLAB)

dB

Purpose

Convert magnitude data to decibels (dB).

Library

Elementary Functions, in Math Functions

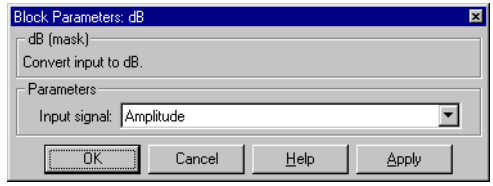
Description

The dB block converts a linear power or amplitude input to dB:

```
y = k * log10(u)   % equivalent MATLAB code
```

where `k` is 10 for power signals and 20 for magnitude signals. Note that the signal should only contain values greater than zero.

Dialog Box



Input signal Ⓣ

The type of input signal: **Power** or **Amplitude**.

See Also

dB Gain

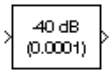
Purpose

Apply a gain specified in dB.

Library

Elementary Functions, in Math Functions

Description



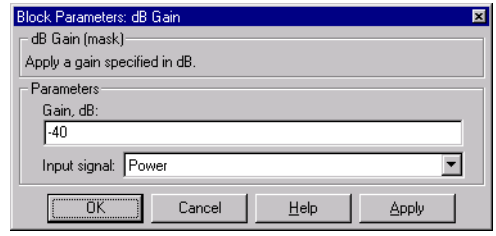
The dB Gain block multiplies the input by a specified constant, variable, or expression. The specified **Gain** value is converted from dB to a linear gain value before being applied to the input signal:

$$g_{lin} = 10^{\left(\frac{g_{dB}}{k}\right)}$$

where g_{lin} is the linear gain value and g_{dB} is the gain in decibels. The value k is set to 10 for power signals (select **Power** from the **Input signal** pop-up menu) and 20 for magnitude signals (select **Amplitude** from the **Input signal** pop-up menu).

The gain must be a real scalar, which you can enter as a numeric value, or as a variable or expression. The value of the equivalent linear gain is displayed below the dB gain value in the block icon.

Dialog Box



Gain (T)

The gain to apply to the input, specified in dB.

Input signal (T)

The type of input signal: **Power** or **Amplitude**.

See Also

dB

DCT

Purpose

Compute the DCT of the input.

Library

Transforms, in General DSP

Description

The DCT block computes the unitary discrete cosine transform (DCT) of the input frame independently at each sample time. For a length-M input u, the DCT is given by

$$U(k) = w(k) \sum_{m=1}^{M} u(m) \cos \frac{\pi(2m-1)(k-1)}{2M}, \qquad k = 1, \ldots, M$$

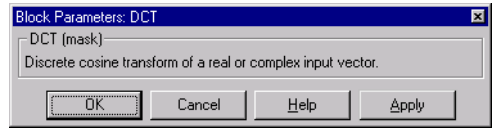
where

$$w(k) = \begin{cases} \frac{1}{\sqrt{M}}, & k = 1 \\ \sqrt{\frac{2}{M}}, & 2 \le k \le M \end{cases}$$

Multichannel inputs (i.e., frame matrices) are not accepted. The output of the block is a vector with the same size, period, and data type (real/complex) as the input vector.

The DCT is closely related to the discrete Fourier transform. You can often reconstruct a sequence very accurately from only a few DCT coefficients, which is useful in applications requiring data compression.

Dialog Box



See Also

Complex Cepstrum
FFT
IDCT
Real Cepstrum
`dct` (Signal Processing Toolbox)

Detrend

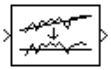
Purpose

Remove a linear trend from a vector.

Library

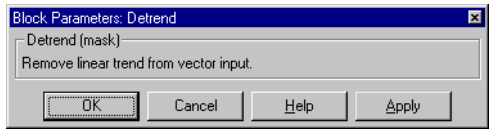
Signal Operations, in General DSP

Description



The Detrend block removes a linear trend from the input vector. It computes the least-squares fit of a straight line to the input vector data, and subtracts the resulting linear function from the vector.

Dialog Box



See Also

Cumulative Sum
Difference
Unwrap
`detrend` (MATLAB)

Purpose

Compute the element-to-element difference along a vector.

Library

Vector Functions, in Math Functions

Description

The Difference block computes the difference between successive vector elements. That is, for an input vector *u* of length N,

```
y = [u(2)-u(1) u(3)-u(2) ... u(N)-u(N-1)]
```

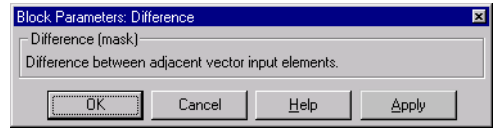
or

```
y = diff(u)        % equivalent MATLAB code
```

The output is a vector of length N-1. A matrix input, `u`, is treated as a vector, `u(:)`.

Note If you expect to generate code for the Difference block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



See Also

Cumulative Sum
`diff` (MATLAB)

Digital FIR Filter Design

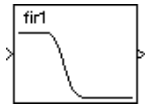
Purpose

Design and implement a variety of FIR filters.

Library

Filter Designs, in Filtering

Description



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 `fir1` function in the Signal Processing Toolbox, 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 in the Filter Realizations library.

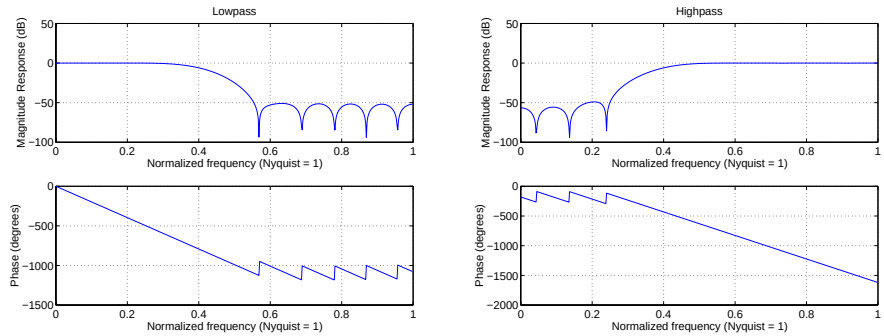
For complete details on the classical FIR filter design algorithm, see the description of the `fir1` and `fir2` functions in the *Signal Processing Toolbox User's Guide*.

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 as appropriate to match the **Filter type** selection.

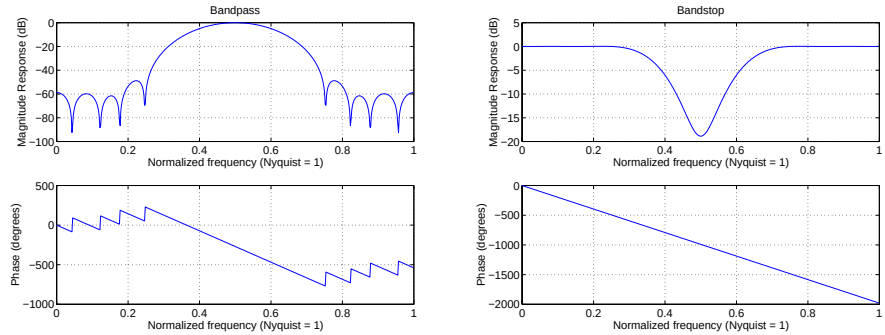
- **Lowpass** and **Highpass**

  In lowpass and highpass configurations, the **Filter order** and **Upper cutoff frequency** parameters specify the filter design. Frequencies are normalized to the Nyquist frequency. The figure below shows the frequency response of the default order-22 filter with cutoff at 0.4.



- **Bandpass** and **Bandstop**

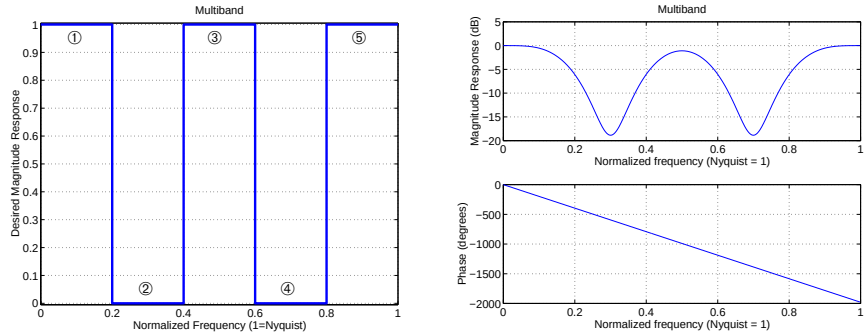
  In bandpass and bandstop configurations, the **Filter order**, **Lower cutoff frequency**, and **Upper cutoff frequency** parameters specify the filter design. Frequencies are normalized to the Nyquist frequency, and the actual filter order is twice the **Filter order** parameter value. The figure below shows the frequency response of the default order-22 filter with lower band edge at 0.4, and upper band edge 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 the Nyquist 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**

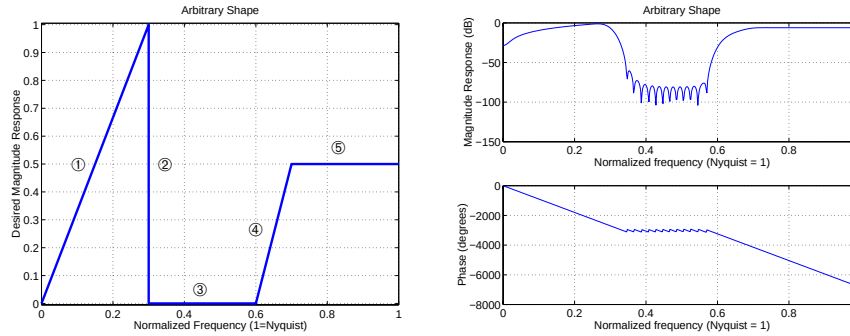
  In the arbitrary shape configuration, the **Filter order**, **Frequency vector**, and **Gains at these frequencies** parameters specify the filter design. The **Frequency vector**, `f`, contains frequency points in the range `0` to `1` (inclusive) in ascending order, where `1` corresponds to the Nyquist frequency. The **Gains at these frequencies** parameter, `m`, 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.)

  The desired magnitude response of the design can be displayed by typing

  ```
  plot(f,m)
  ```

  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.

Frequency = [0.0 0.3 0.3 0.6 0.7 1.0]
Gains = [0.0 1.0 0.0 0.0 0.5 0.5]
Band: ① ② ③ ④ ⑤



Arbitrary-shape filters are designed using the `fir2` function in the Signal Processing Toolbox.

Window Types

The **Window type** parameter allows you to select from a variety of different windows. In the list below, `Nw` is the filter order.

Window Type	Equivalent MATLAB Code
Bartlett	`w = bartlett(Nw)`
Blackman	`w = blackman(Nw)`
Boxcar	`w = boxcar(Nw)`
Chebyshev	`w = chebwin(Nw,r)`
Hamming	`w = hamming(Nw)`
Hanning	`w = hanning(Nw)`
Kaiser	`w = kaiser(Nw,beta)`
Triangular	`w = triang(Nw)`

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

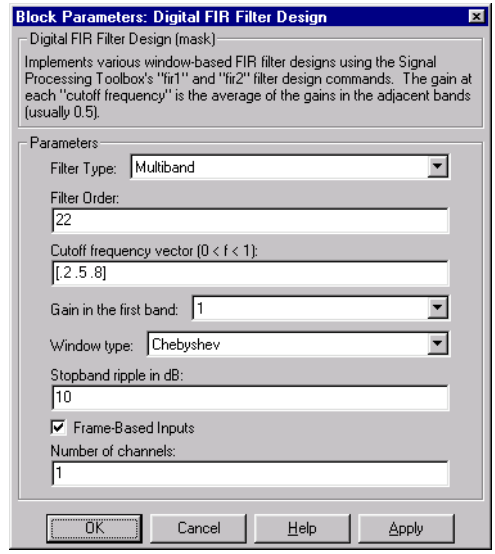
When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel over time.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent channels in the matrix, N, and the block filters each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

In both sample-based and frame-based modes, the output is the same size as the input.

Dialog Box



The parameters displayed in the dialog box vary for different filter types. Not all of the parameters shown above (and 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**.

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.

Upper cutoff frequency

The normalized cutoff frequency for the **Highpass** and **Lowpass** filter configurations. A value of 1 specifies the Nyquist frequency (half the sample frequency).

Lower cutoff frequency

The lower passband or stopband frequency for the **Bandpass** and **Bandstop** filter configurations. A value of 1 specifies the Nyquist frequency (half the sample frequency).

Upper cutoff frequency

The upper passband or stopband frequency for the **Bandpass** and **Bandstop** filters. A value of 1 specifies the Nyquist frequency (half the sample frequency).

Cutoff frequency vector

A vector of ascending frequency points defining the cutoff edges for the **Multiband** filter. A value of 1 specifies the Nyquist frequency (half the sample frequency).

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.

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 the Nyquist frequency (half the sample frequency).

Gains at these frequencies

A vector containing the desired magnitude response for the **Arbitrary shape** filter at the corresponding points in the **Frequency vector**.

Window type

The type of window to apply.

Stopband ripple

The level (dB) of stopband ripple, R_s, for the **Chebyshev** window.

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.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (channels) in the input matrix.

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.

See Also

Digital IIR Filter Design
Least Squares FIR Filter Design
Remez FIR Filter Design
Window Function
Yule-Walker IIR Filter Design
`fir1` (Signal Processing Toolbox)
`fir2` (Signal Processing Toolbox)

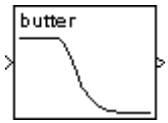
Purpose

Design and implement an IIR filter.

Library

Filter Designs, in Filtering

Description



The Digital IIR Filter Design block designs a discrete-time (digital) IIR filter in a lowpass, highpass, bandpass, or bandstop configuration, and applies the filter to the input using the Direct-Form II Transpose Filter block in the Filter Realizations library. 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.
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 the Nyquist 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}, 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 the Signal Processing Toolbox's filter design commands `butter`, `cheby1`, `cheby2`, and `ellip`.

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

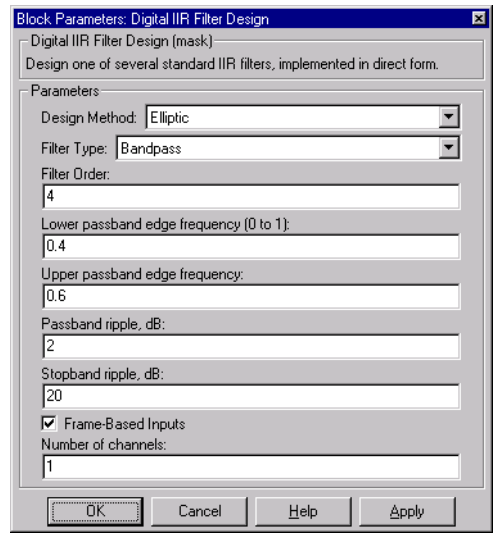
When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel over time.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix. The block filters each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

In both sample-based and frame-based modes, the output is the same size as the input.

Dialog Box



The parameters displayed in the dialog box vary for different design/band combinations. Not all of the parameters shown above (and 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**.

Filter type

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

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.

Lower passband edge frequency

The normalized lower passband frequency for the bandpass and bandstop configurations of the Butterworth, Chebyshev type I, and elliptic designs.

Upper passband edge frequency

The normalized upper passband frequency for the bandpass and bandstop configurations of the Butterworth, Chebyshev type I, or elliptic designs.

Stopband edge frequency

The normalized stopband edge frequency for the highpass and lowpass band configurations of the Chebyshev type II design.

Lower stopband edge frequency

The normalized lower stopband frequency for the bandpass and bandstop configurations of the Chebyshev type II design.

Upper stopband edge frequency

The normalized upper stopband frequency for the bandpass and bandstop filter configurations of the Chebyshev type II design.

Passband ripple

The passband ripple, in dB, for the Chebyshev type I and elliptic designs.

Stopband ripple

The stopband attenuation, in dB, for the Chebyshev type II and elliptic designs.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (channels) in the input matrix.

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.

See Also

Digital FIR Filter Design
Yule-Walker IIR Filter Design
`butter` (Signal Processing Toolbox)
`cheby1` (Signal Processing Toolbox)
`cheby2` (Signal Processing Toolbox)
`ellip` (Signal Processing Toolbox)

Direct-Form II Transpose Filter

Purpose

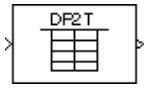
Apply an IIR filter to the input.

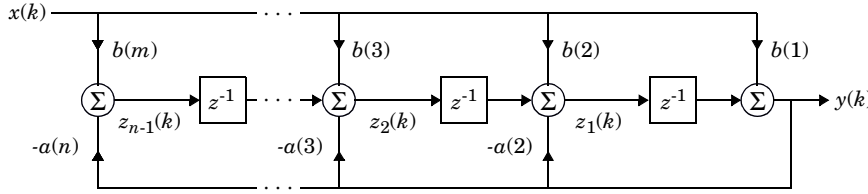
Library

Filter Realizations, in Filtering

Description

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`.

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)]
```

Note that 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.

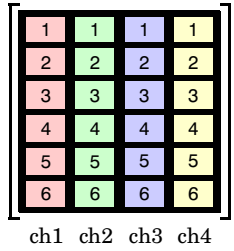
The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel over time.

Frame-Based Operation

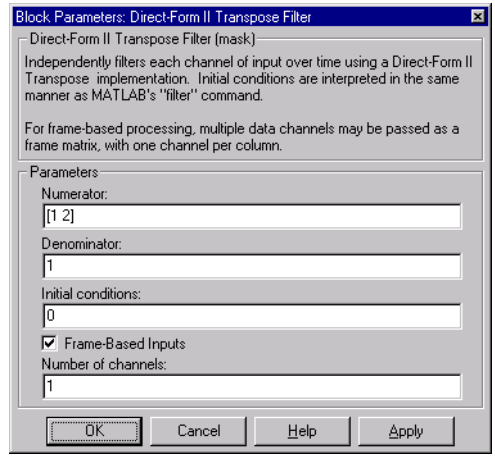
When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block filters each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

Note If you expect to generate code for the Direct-Form II Transpose Filter block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Numerator

The filter numerator.

Denominator

The filter denominator.

Initial conditions

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

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (channels) in the input 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.

See Also

Biquadratic Filter
Discrete Filter (Simulink)
Filter Realization Wizard
Time-Varying Direct-Form II Transpose Filter
`filter` (MATLAB)

Discrete Constant

Purpose

Generate a constant.

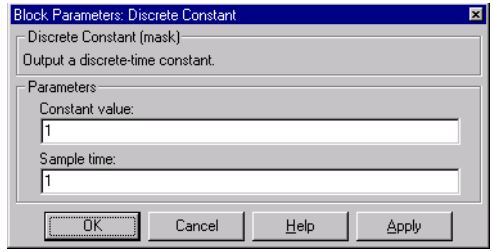
Library

DSP Sources

Description

The Discrete Constant block outputs a constant scalar or vector into a system with a discrete sample period.

Dialog Box



Constant value Ⓣ

The constant to output into the system.

Sample time

The discrete sample period of the output.

See Also

Matrix Constant
Signal From Workspace

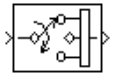
Purpose

Convert a scalar input to a vector output (serial-to-parallel conversion).

Library

Switches and Counters, in General DSP

Description



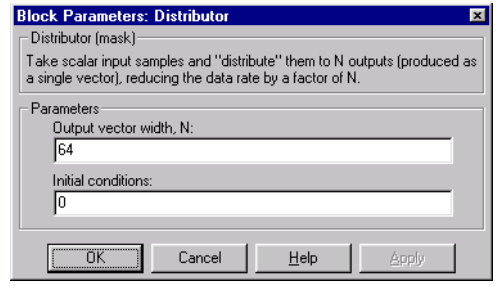
The Distributor block acquires N sequential input samples, and distributes the acquired samples across the N channels of the output. N is specified by the **Output vector width** parameter. The rate of the output is slower than the input rate by a factor of N.

For scalar inputs, the block distributes the acquired scalar samples across the N elements of the vector output. For vector inputs, the block distributes the acquired vector samples across the N *rows* of the matrix output.

The Distributor block's buffer is initialized to the value specified by the **Initial condition** parameter, and the block always outputs this buffer at the first simulation step (t=0). If 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. If 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.

The Distributor block is functionally equivalent to the Buffer block.

Dialog Box



Output vector width

The number of elements in the output vector (number of rows in output matrix).

Initial conditions

The value of the block's initial output, a scalar, vector, or matrix.

Distributor

See Also

Buffer
Commutator

Purpose

Resample the input signal to a lower rate.

Library

Signal Operations, in General DSP

Description

The Downsample block resamples the discrete input at a rate K times slower than the input sample rate by applying a zero-order hold throughout the new sample interval. K is an integer value specified by the **Downsample factor** parameter.

The **Sample offset** parameter delays the output samples by an integer number of sample periods D (D<K), 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 by adjusting the **Sample offset**.

Input Sequence	Sample Offset	Downsampled Output Sequence
`1,2,3,...`	0	`0,1,5,9,13,17,21,25,...`
`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 output sequence above is a result of the zero **Initial condition** parameter setting for this example.

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

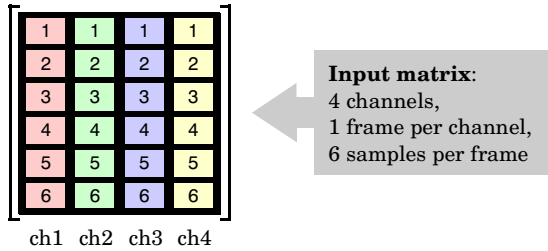
Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block downsamples each channel over time. The output sample period is K times longer than the input sample rate, and the input and output sizes are identical.

In sample-based mode, the **Initial condition** can be a vector containing one value for each channel, or a scalar to be applied to all signal channels. The value specified for the **Initial condition** parameter is output at t=0.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 frame matrix input:



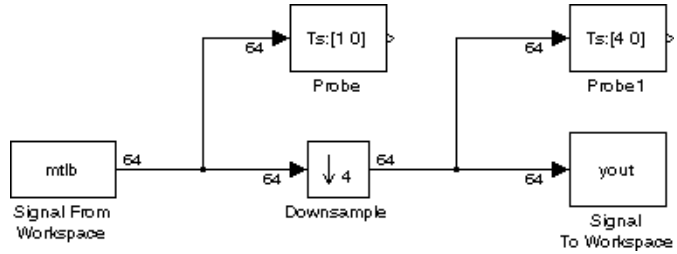
The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

In frame-based operation, the block independently downsamples each channel in the input matrix by skipping K rows after each row that it passes through to the output. The downsample factor must be less than the frame size, K<M. The **Framing** parameter determines how the block adjusts the rate at the output. 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, but the input and output frame sizes are equal.

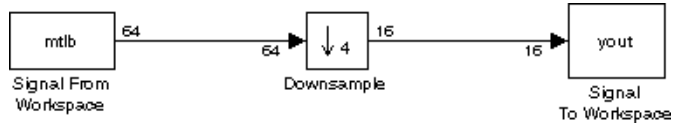
  The example below shows a single-channel input with a frame period of 1 second (**Sample time** = `1/64` and **Samples per frame** = `64` in the Signal From Workspace block) being downsampled by a factor of 4 to a frame period of 4 seconds. The input and output frame sizes are identical.



- **Maintain input frame rate**

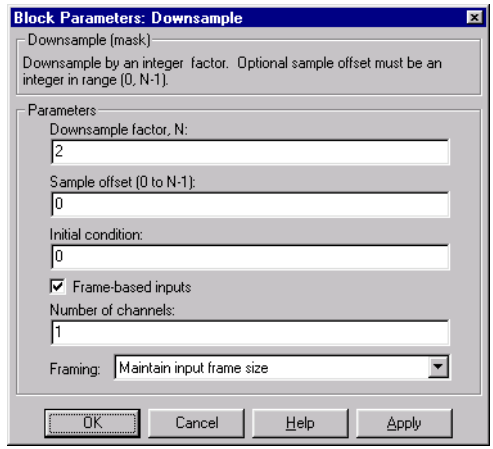
  The block generates the output at the slower (downsampled) rate by using a proportionally smaller frame size than the input. For downsampling by a factor of K, the output frame size is K times smaller than the input frame size, but the input and output frame rates are equal.

  The example 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.



In frame-based mode, the **Initial condition** can be an M-by-N matrix representing the initial input, or a scalar to be repeated across all elements of the M-by-N matrix. The first row of the matrix is added to the output buffer at t=0, the Dth row is added at t=K$*$T$_s$, the (D+K)th row at t=2K$*$T$_s$, and so on, where T$_s$ is the sequence sample period.

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 that the block is initialized with; a scalar or vector in sample-based mode, a scalar or matrix in frame-based mode.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of channels (columns) in the input matrix, N.

Framing

For frame-based operation, the method by which to implement the downsampling: decrease the output sample rate, or decrease the output frame size.

See Also

FIR Decimation
FIR Rate Conversion
Repeat
Upsample

Dyadic Analysis Filter Bank

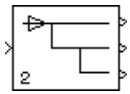
Purpose

Decompose a signal into components of equal or logarithmically decreasing frequency intervals and sample rates.

Library

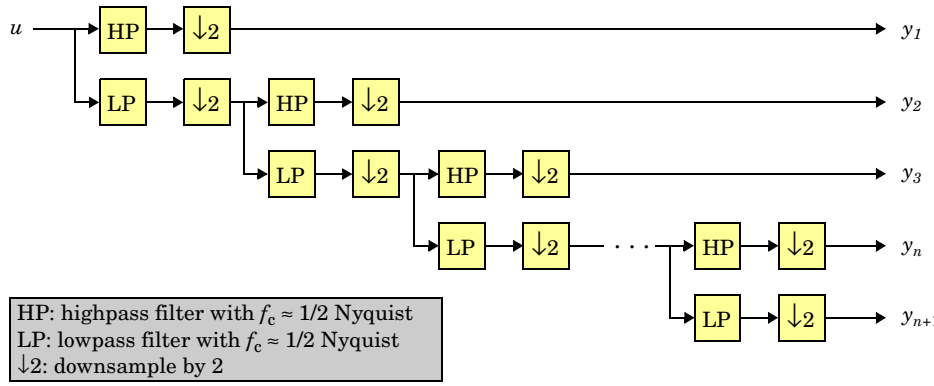
Multirate Filters, in Filtering

Description



The Dyadic Analysis Filter Bank block decomposes a broadband signal into a collection of successively more bandlimited components by repeatedly dividing the frequency range. The typical (asymmetric) *n*-level filter bank structure is shown below:

Asymmetric Filter Bank



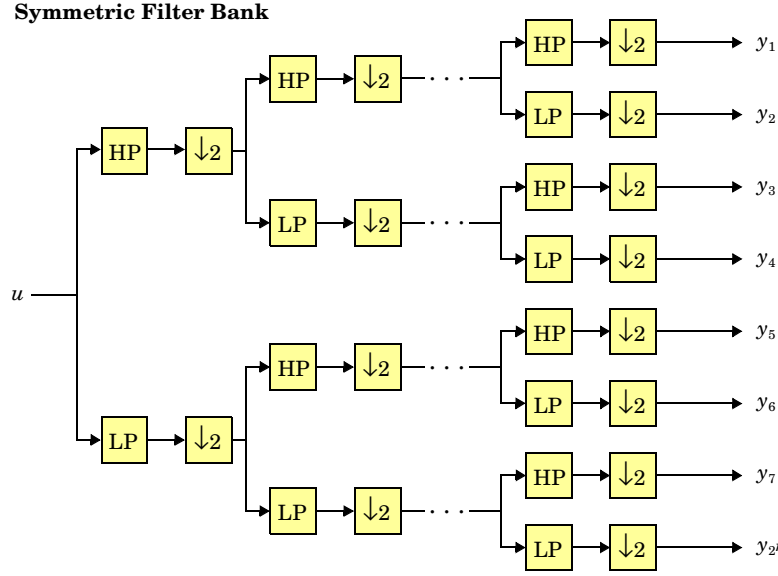
At each level, the *low-frequency* output of the previous level is decomposed into adjacent high- and low-frequency subbands by a highpass (HP) and lowpass (LP) filter pair. Each of the two output subbands is half the bandwidth of the input to that level (hence "dyadic"). The bandlimited output of each filter is maximally decimated by a factor of 2 to preserve the bit rate of the original signal. In wavelet applications (see below) the aliasing introduced by the decimation stage can be exactly canceled in reconstruction.

The **Lowpass FIR filter coefficients** and **Highpass FIR filter coefficients** parameters specify the filter coefficients to be used for every highpass and lowpass (respectively) filter in the structure. The values of these coefficients are typically computed using the wavelet family functions in the Wavelet Toolbox (see the *Wavelet Toolbox User's Guide* for more information).

The **Tree structure** parameter specifies an asymmetric (or wavelet) tree, as shown above, or a symmetric structure, as shown below. Note that the

symmetric structure decomposes both the high- and low-frequency subbands at each level, whereas the asymmetric structure only decomposes the low-frequency bands.

Symmetric Filter Bank



HP: highpass filter with $f_c \approx 1/2$ Nyquist
LP: lowpass filter with $f_c \approx 1/2$ Nyquist
↓2: downsample by 2

The asymmetric structure in the first figure (**Tree structure** set to **Asymmetric**) 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.

$$F_{s,y_{i+1}} = \frac{F_{s,y_i}}{2} \qquad 1 \le i < n$$

$$BW_{y_{i+1}} = \frac{BW_{y_i}}{2} \qquad 1 \le i < n$$

The bottom two outputs share the same sample rate and bandwidth since they originate at the same level,

$$F_{s,y_{n+1}} = F_{s,y_n}$$

and

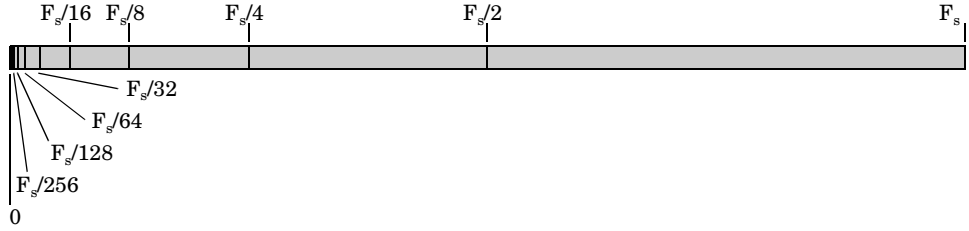
$$BW_{y_{n+1}} = BW_{y_n}$$

Note that in sample-based mode, this change in sample rates is represented by different sample rates at the block outputs. In frame-based mode, the different output sample rates are reflected in the output frame sizes rather than the output frame periods.

When the magnitudes in each of these subband signals are plotted across the full bandwidth of the original signal, the result is a *scalogram*. This is the equivalent of a spectrogram with constant Q, where

$$Q = \frac{f_{y_i}}{BW_{y_i}},$$

and f_{yi} is the midpoint frequency of the band occupied by output y_i. The frequency axis of a scalogram therefore has logarithmic divisions like those shown below:



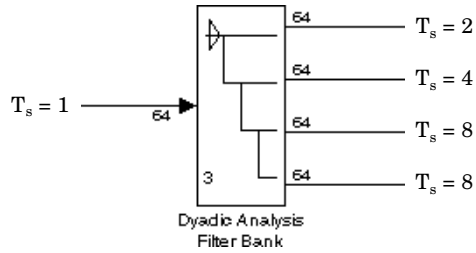
The symmetric structure (**Tree structure** set to **Symmetric**) has 2^n outputs, where n is the number of levels. The sample rate and bandwidth of each output are equal.

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

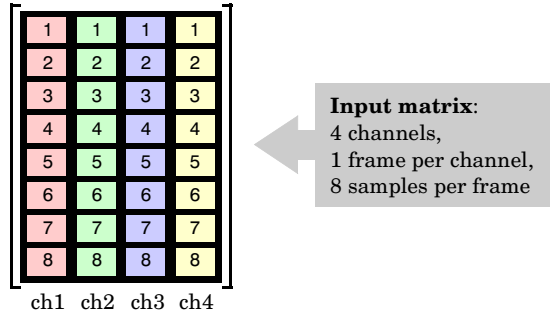
When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel independently over time. The output at each port is the same size as the input, with one channel for each input channel. As described earlier, for the asymmetric tree structure, each output port has a different sample period.

Example:



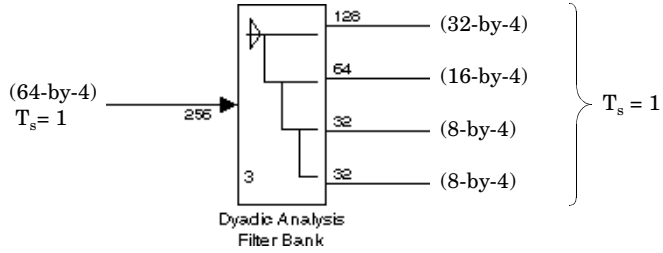
Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input at each port is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal, where M must be a multiple of 2^n, and n is the number of filter bank levels. The illustration below shows a 8-by-4 matrix input, which would be appropriate for a 3-level tree (2^3=8):



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block filters each channel independently over time. The number of columns in each output is therefore the same as the number of columns in the input.

For the asymmetric tree structure, each output port has the *same period* as the input. The reduction in the output sample rates results from the smaller output frame sizes, as shown in the example below.

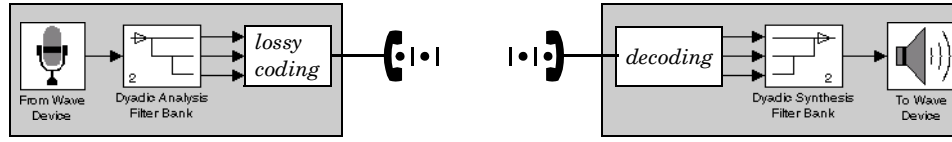


Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

Applications

The primary application for dyadic analysis filter banks is coding for data compression using wavelets.

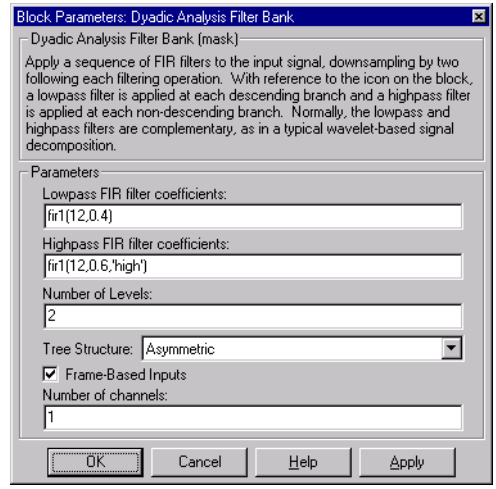
At the transmitting end, the output of the dyadic analysis filter bank is fed to a lossy compression scheme, which typically assigns the number of bits for each filter bank output in proportion to the relative energy in that frequency band. This represents more powerful components of the signal by a greater number of bits than 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.

Note If you expect to generate code for the Dyadic Analysis Filter Bank block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Lowpass FIR filter coefficients

A vector of filter coefficients (descending powers of z) to be shared by all the lowpass filters in the filter bank.

Highpass FIR filter coefficients

A vector of filter coefficients (descending powers of z) to be shared by all the highpass filters in the filter bank.

Number of levels

The number of filter bank levels. An n-level asymmetric structure has $n+1$ outputs; an n-level symmetric structure has 2^n outputs.

Tree structure

The structure of the filter bank, **Asymmetric** (wavelet) or **Symmetric**.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (channels) in the input matrix.

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.

See Also

Dyadic Synthesis Filter Bank

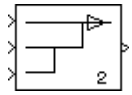
Purpose

Reconstruct a signal from its multirate bandlimited components.

Library

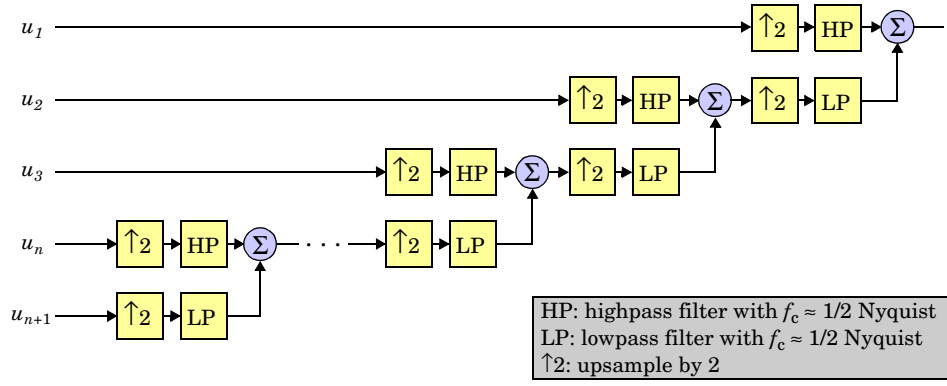
Multirate Filters, in Filtering

Description



The Dyadic Synthesis Filter Bank block typically reconstructs a signal that was decomposed by the Dyadic Analysis Filter Bank 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. The typical (asymmetric) n-level filter bank structure is shown below:

Asymmetric Filter Bank



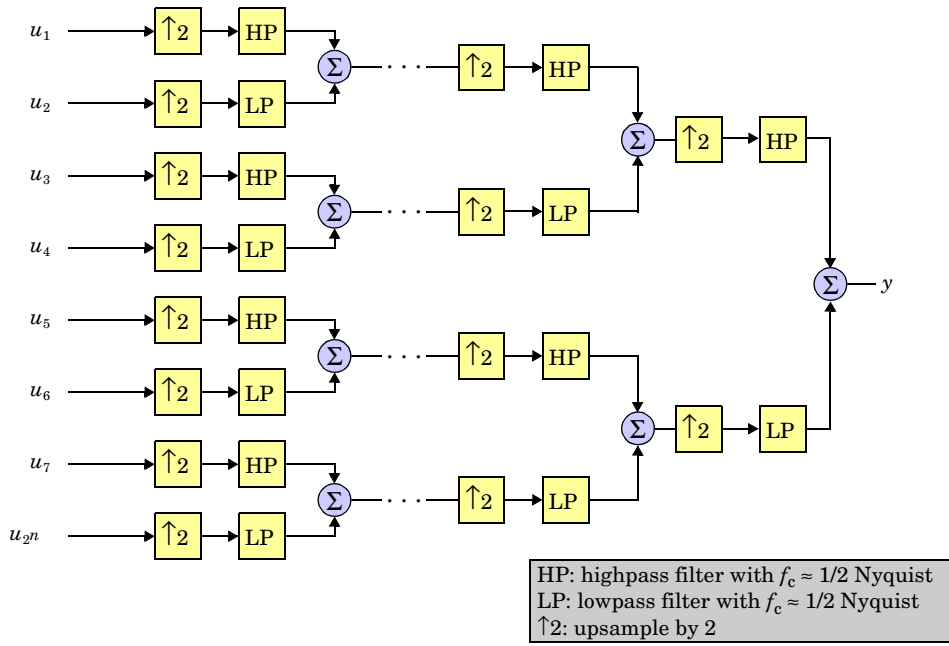
At each level, the two bandlimited inputs (one low-frequency, one high-frequency, both with the same sample rate) are upsampled by a factor of 2 to match the sample rate of the input to the next stage. They are then filtered by a highpass (HP) and lowpass (LP) filter pair with coefficients calculated to cancel (in the subsequent summation) the aliasing introduced in the corresponding dyadic 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 (hence “dyadic”).

The **Lowpass FIR filter coefficients** and **Highpass FIR filter coefficients** parameters specify the filter coefficients to be used for every highpass and lowpass (respectively) filter in the structure. The values of these coefficients are typically computed together with the dyadic analysis coefficients using the

wavelet family functions in the Wavelet Toolbox (see the *Wavelet Toolbox User's Guide* for more information).

The **Tree structure** parameter specifies an asymmetric (or wavelet) tree, as shown above, or a symmetric structure, as shown below. Note that the symmetric structure reconstructs a signal that was symmetrically decomposed by the Dyadic Analysis Filter Bank block (i.e., both the high- and low-frequency subbands were divided at each level). The asymmetric structure reconstructs a signal that was asymmetrically decomposed by the Dyadic Analysis Filter Bank block (i.e., only the low-frequency subbands were divided at each level).

Symmetric Filter Bank



The asymmetric structure in the first figure (**Tree structure** set to **Asymmetric**) 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.

$$F_{s,u_{i+1}} = \frac{F_{s,u_i}}{2} \qquad 1 \le i < n$$

$$BW_{u_{i+1}} = \frac{BW_{u_i}}{2} \qquad 1 \le i < n$$

The bottom two inputs share the same sample rate and bandwidth since they are processed by the same level.

$$F_{s,u_{n+1}} = F_{s,u_n}$$

and

$$BW_{u_{n+1}} = BW_{u_n}$$

Note that in sample-based mode, the different sample rates described above are represented by different port rates at the block inputs. In frame-based mode, the different input sample rates are reflected in the input frame sizes rather than the actual input port periods.

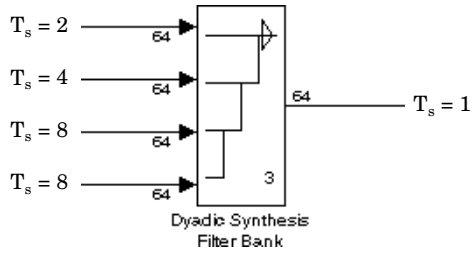
The symmetric structure (**Tree structure** set to **Symmetric**) has 2^n inputs, where n is the number of levels. The sample rate and bandwidth of each input are equal.

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

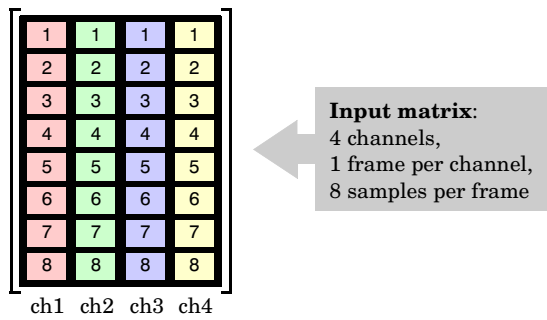
When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel independently over time. The output is the same size as the input at each port, with one channel for each input channel. As described earlier, for the asymmetric tree structure, each input port has a different period.

Example:



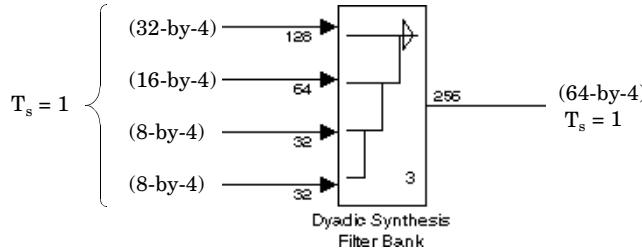
Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input at each port is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 8-by-4 matrix input.



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block filters each channel independently over time. The number of columns in each output is therefore the same as the number of columns in the input.

Note that each input port has the *same sample period* as the output port. The increase in the output sample rate results from the larger output frame size, as shown in the example below.

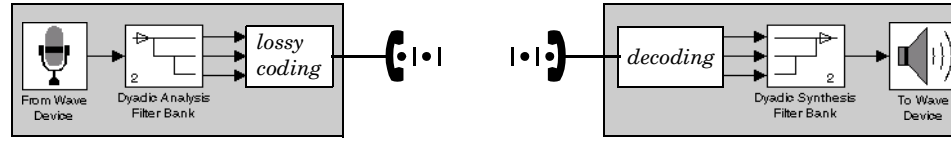


Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

Applications

The primary application for asymmetric dyadic synthesis filter banks is coding for compression using wavelets.

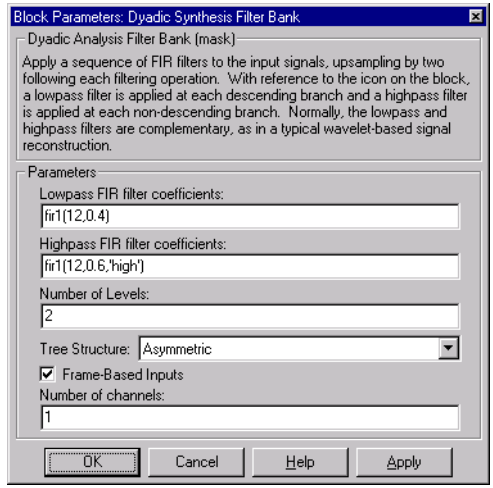
At the transmitting end, the output of a dyadic analysis filter bank is fed to a lossy compression scheme, which typically assigns the number of bits for each filter bank output in proportion to the relative energy in that frequency band. This represents the more powerful components of the signal by a greater number of bits than the less powerful signal components.



At the receiving end, the transmission is decoded and fed to the 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.

Note If you expect to generate code for the Dyadic Synthesis Filter Bank block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Lowpass FIR filter coefficients

A vector of filter coefficients (descending powers of z) to be shared by all the lowpass filters in the filter bank.

Highpass FIR filter coefficients

A vector of filter coefficients (descending powers of z) to be shared by all the highpass filters in the filter bank.

Number of levels

The number of filter bank levels. An n-level asymmetric structure has $n+1$ inputs; an n-level symmetric structure has 2^n inputs.

Tree structure

The structure of the filter bank, **Asymmetric** (wavelet) or **Symmetric**.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (channels) in the input matrix.

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.

See Also

Dyadic Analysis Filter Bank.

Edge Detector

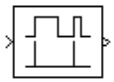
Purpose

Detect a transition of the input from zero to a nonzero value.

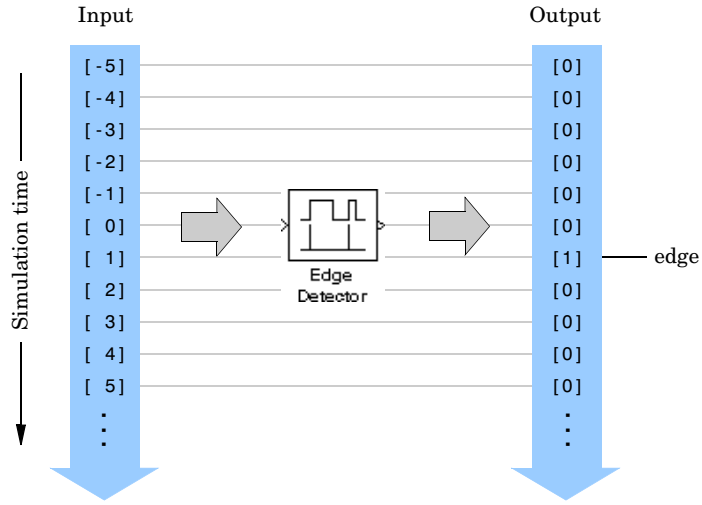
Library

Switches and Counters, in General DSP

Description



The Edge Detector block outputs a unit pulse (1) when the input transitions from zero to a nonzero value. Otherwise, the block outputs zeros at the same rate as the input.



The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation

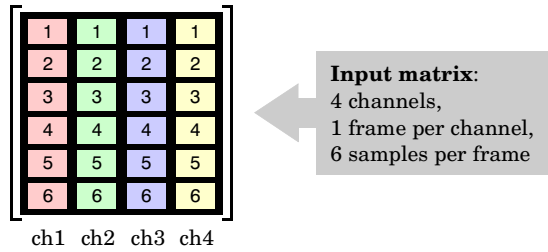
Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a scalar, a 1-by-N sample vector, or M-by-N sample matrix. A scalar sequence represents a single channel, as shown above. For a vector or matrix input, each of the N vector elements or M*N matrix elements is treated as an independent channel, and the block tracks the edges in each channel over time. The input and output sizes are identical.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix

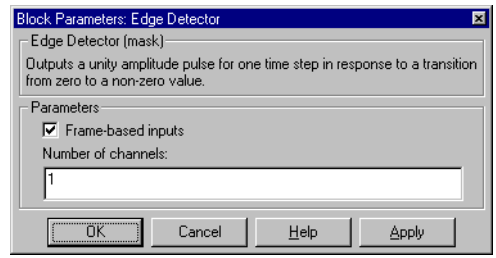
contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block tracks the edges in each channel independently over time. An edge that is split across two consecutive frames (i.e., 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. The input and output sizes are identical.

Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

Dialog Box



Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (channels) in the input matrix.

See Also

Counter
Event-Count Comparator

Event-Count Comparator

Purpose

Detect threshold crossing of accumulated nonzero inputs.

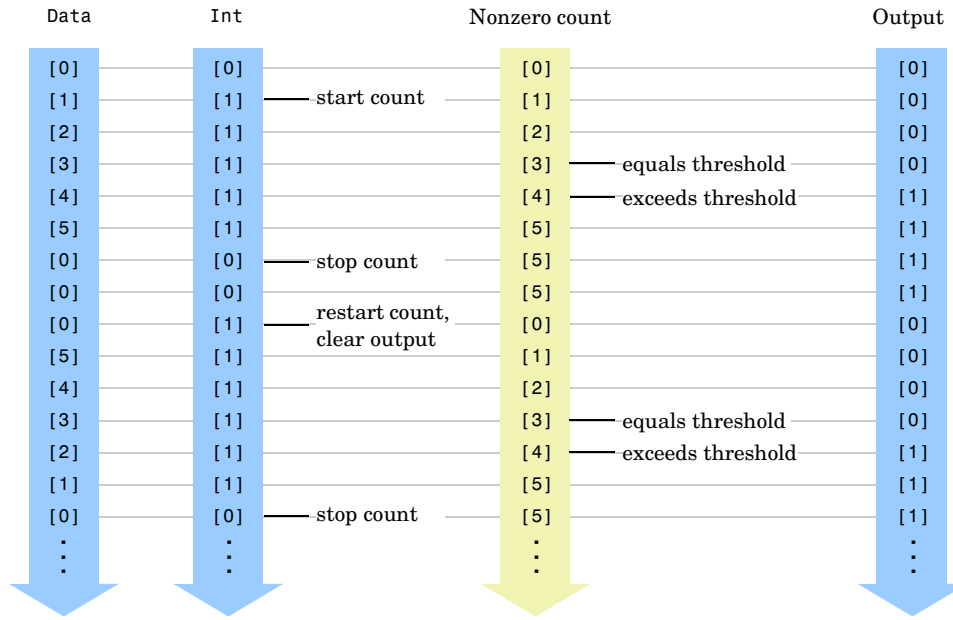
Library

Switches and Counters, in General DSP

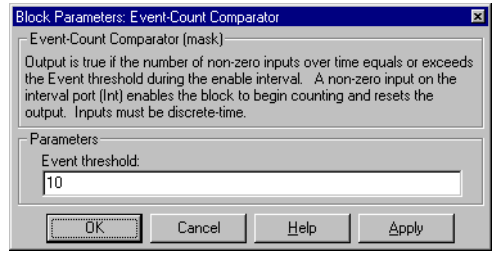
Description

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 (1) signal at the interval (`Int`) port. 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 illustration below shows the block's operation for an **Event threshold** of 3.



Dialog Box



Event threshold

The value against which to compare the number of nonzero inputs.

See Also

Counter
Edge Detector

Extract Diagonal

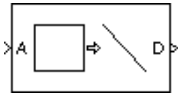
Purpose

Create a vector from the elements of a matrix diagonal.

Library

Matrix Functions, in Math Functions

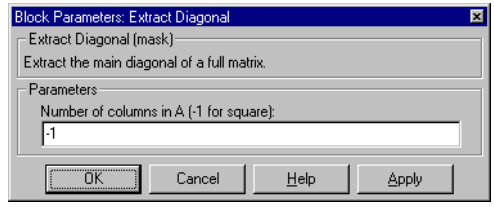
Description



The Extract Diagonal block creates a vector from the diagonal of a matrix input. The diagonal elements of an M-by-N matrix input are used to populate the output vector, which has a length of `min(M,N)`. The **Number of columns in A** parameter specifies the number of columns, N, in the input matrix. A setting of `-1` indicates that the input is square.

Note If you expect to generate code for the Extract Diagonal block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Number of columns in A

The number of columns in the input matrix. A value of `-1` indicates that the input is square.

See Also

Constant Diagonal Matrix
Create Diagonal Matrix
Extract Triangular Matrix

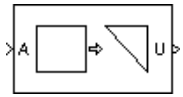
Purpose

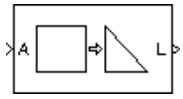
Extract the lower or upper triangle from an input matrix.

Library

Matrix Functions, in Math Functions

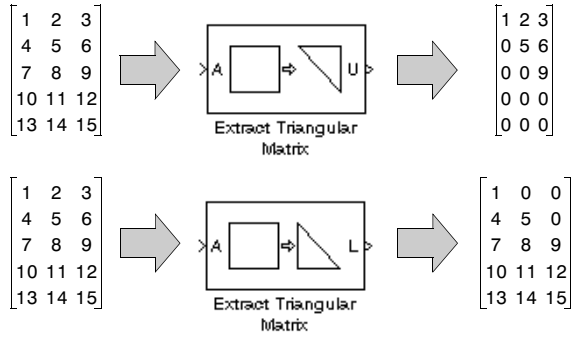
Description





The Extract Triangular Matrix block creates a triangular matrix output from the upper or lower triangular elements of a rectangular input matrix. The **Extract** parameter selects between the two components of the input:

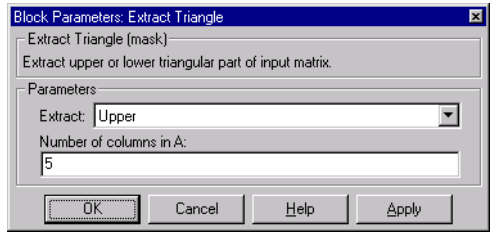
- **Upper** – Copies the upper triangular elements of the input matrix (including those on the diagonal) to an output matrix of the same size. The lower triangular elements of the output are zero.
- **Lower** – Copies the lower triangular elements of the input matrix (including those on the diagonal) to an output matrix of the same size. The upper triangular elements of the output are zero.



Note If you expect to generate code for the Extract Triangular Matrix block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Extract Triangular Matrix

Dialog Box



Extract Ⓣ

The component of the matrix to copy to the output, upper triangle or lower triangle. This parameter is not tunable in Simulink's external mode.

Number of columns in A

The number of columns in the input matrix.

See Also

Backward Substitution
Cholesky Factorization
Constant Diagonal Matrix
Extract Diagonal
Forward Substitution

Purpose

Compute the FFT of the input.

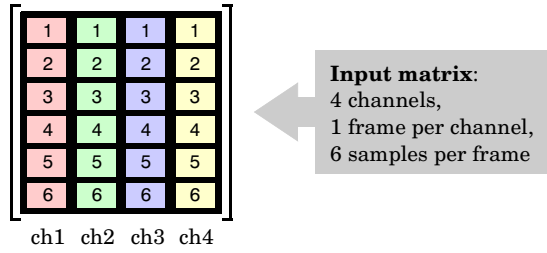
Library

Transforms, in General DSP

Description

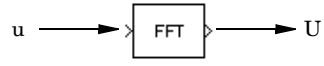
The FFT block computes the fast Fourier transform (FFT) of each input channel independently at each sample time. The block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal.

The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix. The output is complex and has the same dimension and sample rate as the input (i.e., the FFT is computed at M frequency points for each channel).

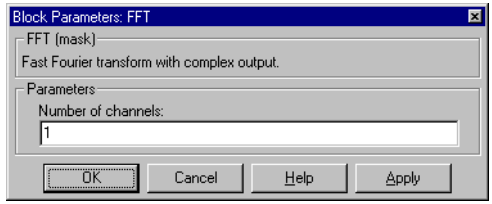
The FFT operation for a single-channel input (**Number of channels** = 1) is shown below.



$$U(k) = \sum_{m=0}^{M-1} u(m)e^{-j2\pi(mk/M)} \qquad k = 0, \ldots, M-1$$

The input frame size, M, must be a power of two. To work with other frame sizes, use the Zero Pad block to pad or truncate the input frame to a power-of-two length.

Dialog Box



Number of channels

The number of columns (frames) in the input matrix.

See Also

Complex Cepstrum
DCT
IFFT
Zero Pad

Purpose

Compute and display the frequency content of a frame-based input.

Library

DSP Sinks

Description

The FFT Frame Scope block displays the magnitude of the FFT of the input, which is assumed to be a frame of sequential time-samples.

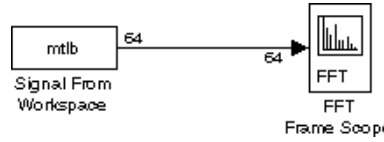
The **FFT length** parameter, N_{fft}, specifies the number of samples on which to perform the FFT. A value of `-1` instructs the block to use the input frame size as the FFT size, in which case you can also set the **Sample time of original time-series** parameter to `-1` (see below). Otherwise, the block zero pads or truncates the input to N_{fft} before computing the FFT.

In order to correctly scale the frequency axis (i.e., to determine the frequencies against which the transformed input data should be plotted), the block needs to know the actual sample period of the time-domain sequence. The **Sample time of original time-series** parameter allows you to specify this in two different ways:

Auto-Detect from Input Sample Period. A value of `-1` for this parameter instructs the block to compute the frequency data from the sample period of the input. This parameter setting is appropriate if the FFT is computed on the same number of points as are contained in the time-domain input (i.e., no zero-padding or truncation). This is only true when the **FFT size** parameter is set to the size of the input (or, equivalently, to `-1`).

The auto-detect mode also makes the following two assumptions:

- The sample period of the original time-domain signal in the simulation is equal to the period at which the physical signal was actually sampled. For example, the `mtlb` signal imported from the workspace in the model below has an actual sample period of `1/Fs = 1/7418`.



Although the **Sample time** parameter in the Signal From Workspace block can legitimately be set to any value, for the auto-detect mode to compute the

correct frequency scaling, the **Sample time** parameter *must* be set to the original sample period of `1/7418`.

- Consecutive frames of the time-domain signal do not overlap each other; that is, a particular sample of the time-domain signal does not appear in more that one sequential frame.

Enter the Appropriate Time-Domain Sample Period. Enter the sample period of the time-series, T_s. This is necessary when the FFT is computed on a different number of points (more or fewer) than are contained in the time-domain input. When the **FFT length** parameter is set to a value other than the frame size of the input, the block either zero-pads or truncates the input before performing the FFT, and cannot automatically compute the original time-domain sample period.

You also need to explicitly specify the time-series' actual sample period when either of the assumptions listed above for the auto-detect mode is not valid.

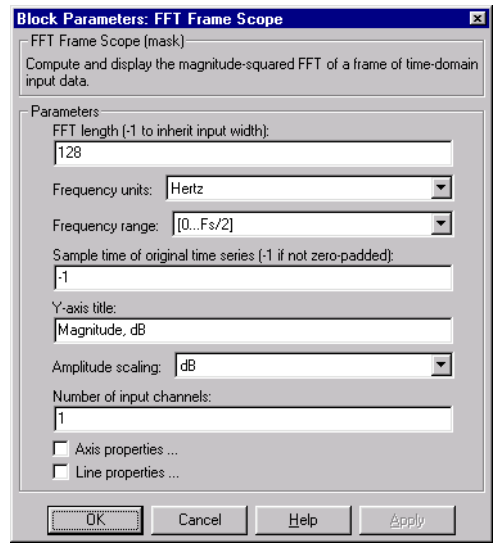
The **Frequency units** parameter specifies whether the frequency axis values should be in units of Hertz or rads/sec, and the **Frequency range** parameter specifies the range of frequencies over which the magnitudes in the input should be plotted. The available options are **[0..Fs/2]**, **[-Fs/2..Fs/2]**, and **[0..Fs]**, where F_s is the time-domain signal's actual sample frequency ($F_s/2$ is the Nyquist frequency). If the **Frequency units** parameter specifies **Hertz**, the spacing between frequency points is $1/(N_{fft}T_s)$. For **Frequency units** of **rads/sec**, the spacing between frequency points is $2\pi/(N_{fft}T_s)$.

Note that all of the FFT-based blocks in the DSP Blockset, 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 scope updates the display for each new input. The input can be an M-by-N frame matrix, where each of the N frames in the matrix contains M sequential time samples from an independent signal channel. The **Number of input channels** parameter specifies the number of independent channels (columns), N, in the matrix. The block overlays a separate FFT plot for each of the N independent channels in the input.

For information about the scope window, as well as the **Axis properties** and **Line properties** panels in the dialog box, see the reference page for the Time Frame Scope block.

Dialog Box



FFT length

The number of samples on which to perform the FFT. If the **FFT length** differs from the size of the input vector, the data is zero-padded or truncated as needed. A value of `-1` sets the FFT length to the input frame size.

Frequency units (T)

The frequency units for the *x*-axis, **Hertz** or **rads/sec**.

Frequency range (T)

The frequency range over which to plot the data, **[0..Fs/2]**, **[-Fs/2..Fs/2]**, or **[0..Fs]**, where F_s is the sample frequency of the original time-domain signal, $1/T_s$.

Sample time of original time series (T)

The sample period, T_s, of the original time-domain signal. Set to `-1` to auto-detect the signal sample period from the frame period and frame size of the block input (use only when the **FFT size** is equal to the size of the input or `-1`).

Y-Axis title

The text to be displayed to the left of the *y*-axis.

Amplitude scaling Ⓣ

The scaling for the *y*-axis, **dB** or **Magnitude**.

Number of input channels

The number of channels (columns) in the input matrix.

Axis properties

Select to expose the **Axis Properties** panel. See Time Frame Scope for more information.

Line properties

Select to expose the **Line Properties** panel. See Time Frame Scope for more information.

See Also

Buffered FFT Frame Scope
FFT
Frequency Frame Scope
Time Frame Scope
User-Defined Frame Scope

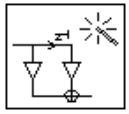
Purpose

Automatically construct filter realizations using Sum, Gain, and Unit Delay blocks.

Library

Filter Realizations, in Filtering

Description



The Filter Realization Wizard is a tool for automatically creating filter realizations with specific architectures. The Wizard's interface allows you to specify the filter's structure and coefficients, the type of data to be filtered, and optimization criteria for the design. The Wizard then builds the specified filter as a subsystem composed of Sum, Gain, and Unit Delay blocks. You can select the name of the subsystem ("Filter" is the default) and whether it is placed in the current model or in a new model.

The **Architecture** panel in the Wizard's interface allows you to select from the following realizations.

Architecture	Parameters
Direct-Form I	Numerator, denominator
Direct-Form II	Numerator, denominator
Lattice (AR)	Lattice coefficients
Lattice (MA)	Lattice coefficients
Lattice (ARMA)	Lattice coefficients, ladder coefficients
Symmetric FIR	Coefficients

The **Optimization** panel in the Wizard's interface lets you choose to optimize for zero and unity gains. Zero-gain optimization removes zero-gain paths from the filter structure, and unity-gain optimization substitutes a wire (short circuit) for unity gains.

Type a name for the new filter block in the **Block Name** text field, and select where the block should be placed from the **Destination** pop-up menu. Within a model, the Filter subsystem operates on a sample-based signal (similar to Simulink's Discrete Filter block), filtering each channel over time. Double-click on the subsystem to open it; you can then modify the gains or the filter structure to suit your needs.

Fixed-Point Options

By default, the filter constructed by the Filter Realization Wizard operates using the Simulink standard double-precision arithmetic. If you have the Fixed-Point Blockset installed on your system, you have the additional option of building the filter to operate using single-precision or fixed-point arithmetic. Select the option you want from the **Data Type** panel.

- **Built-in data types**

  The filter is constructed from the standard Simulink Sum, Gain, and Unit Delay blocks, and operates in any precision supported by Simulink (e.g., double-precision, single-precision, Boolean, etc.). This is the default.

- **Single**

  The filter is constructed from the Fixed-Point Sum, Fixed-Point Gain, and Fixed-Point Unit Delay blocks from the Fixed-Point Blockset. The blocks are configured for single-precision arithmetic.

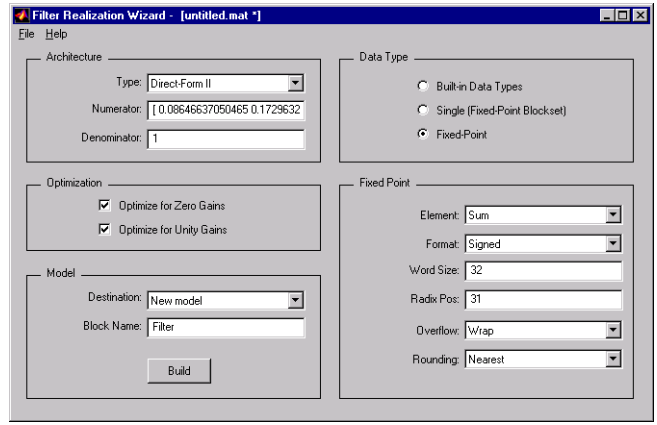
- **Fixed-Point**

  The filter is constructed from the Fixed-Point Sum, Fixed-Point Gain, and Fixed-Point Unit Delay blocks from the Fixed-Point Blockset. The Fixed-Point Sum and Fixed-Point Gain blocks are configured for fixed-point arithmetic using the options specified in the **Fixed-Point** panel of the Filter Realization Wizard. These options include:

  - **Format** (**Signed** or **Unsigned**)
  - **Word size**
  - **Radix pos**
  - **Overflow** (**Wrap** or **Saturate**)
  - **Rounding** (**Zero**, **Nearest**, **Ceiling**, or **Floor**)

For information on these parameters, see the *Fixed-Point Blockset User's Guide*.

Dialog Box



The parameters displayed in the **Architecture** panel vary for different selections in the **Type** menu. Only a portion of the parameters listed below are visible in the wizard at any one time.

Type

The filter architecture: **Direct-Form I**, **Direct-Form II**, **Symmetric FIR**, **Lattice (MA)**, **Lattice (AR)**, **Lattice (ARMA)**.

Numerator

The numerator coefficients for the direct-form I and II structures, specified as a vector or variable name.

Denominator

The denominator coefficients for the direct-form I and II structures, specified as a vector or variable name.

Coefficients

The coefficients for the symmetric FIR structure, specified as a vector or variable name.

Lattice Coefficients

The lattice coefficients for the lattice MA/AR/ARMA structures, specified as a vector or variable name.

Ladder Coefficients

The ladder coefficients for the lattice ARMA structure, specified as a vector or variable name.

Optimize for zero gains

Enables zero-gain optimization (when checked) by removing zero-gain paths from the filter structure.

Optimize for unity gains

Enables unity-gain optimization (when checked) by substituting a wire (short circuit) for unity gains.

Destination

The location where the new filter block should be created.

Block name

The name of the new filter block.

Build

Generate the filter.

Data type

The precision of the data that the filter will process. **Built-in data types**, when selected, configures the block to build the filter using double-precision Simulink blocks. **Single precision** and **Fixed-point** configure the block to build the filter using Fixed-Point Blockset blocks.

Fixed-point

Options for fixed-point filter construction. See the *Fixed-Point Blockset User's Guide*.

Examples

Example 1: Direct Form II

Design an fourth-order, quarter-band, lowpass Butterworth filter:

1. At the MATLAB command line, compute the filter coefficients by entering

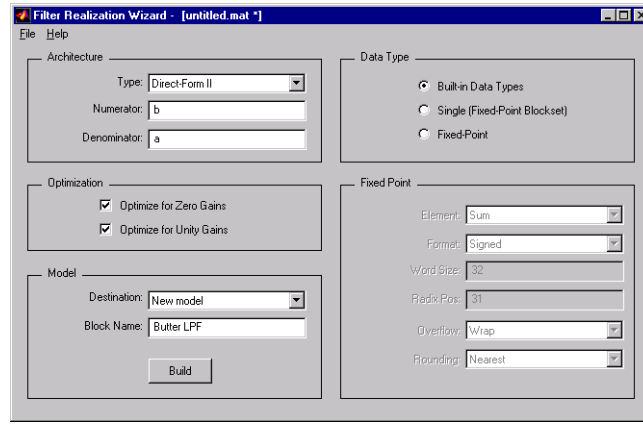
   ```
   [b,a] = butter(4,.25);
   ```

2. Launch the Filter Realization Wizard by double-clicking on the icon in the Filter Realizations library.

3 Configure the Wizard to use `b` and `a` as the numerator and denominator of a Direct-Form II structure:
   - Select **Direct-Form II** from the **Type** menu.
   - Type `b` in the **Numerator** text field.
   - Type `a` in the **Denominator** text field.

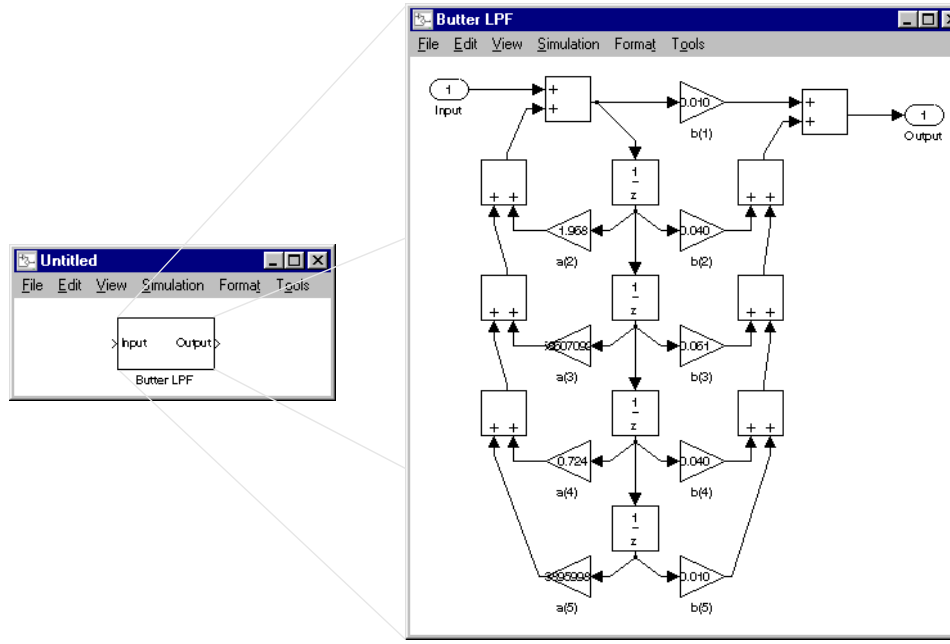
4 Type a name for the new filter subsystem in the **Block Name** field. The example uses `Butter LPF`.

   The GUI with these settings is shown below:



5 Press the **Build** button to create the specified filter subsystem in a new model window.

6 Double-click the new Butter LPF block to see the Direct-Form II filter realization that the Wizard created:



Example 2: Second Order Sections

Design an eighth-order, quarter-band, lowpass Butterworth filter using second-order sections (SOS):

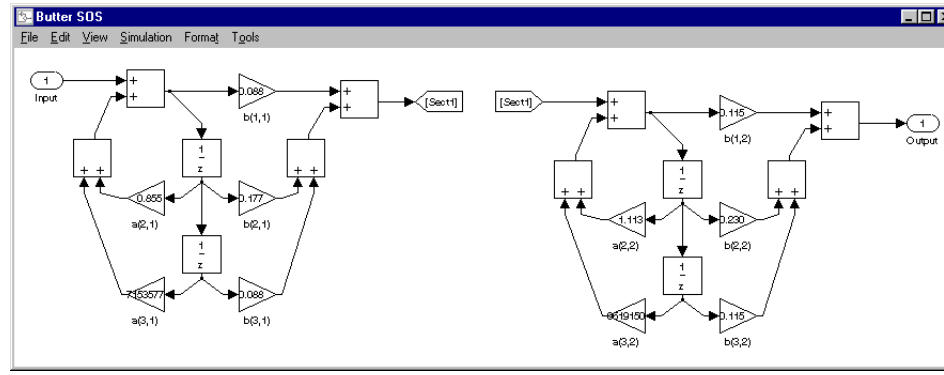
1. At the MATLAB command line, compute the second-order sections by entering

   ```
   [a,b,c,d] = butter(4,.25);
   sos = ss2sos(a,b,c,d);
   ```

2. Configure the Wizard to use `sos` as the numerator of a Direct-Form II structure:
   - Select **Direct-Form II** from the **Type** menu.
   - Type `sos` in the **Numerator** text field.
   - Leave the **Denominator** text field blank.
3. Type a name for the new filter subsystem in the **Block Name** field. The example uses `Butter SOS`.

4 Press the **Build** button to create the specified filter subsystem in a new model window.

5 Double-click the new Butter SOS block to see the Direct-Form II filter realization that the Wizard created:



Note that in a subsystem with the Direct-Form I or II architecture, the filter sections are connected using From and Goto blocks rather than being directly wired together. This makes it easier to recognize and move filter sections in the model window independently of each other.

Example 3: Nth Order Sections

Design a lowpass Butterworth filter using Nth order cascades:

1 At the MATLAB command line, compute the coefficients for the Nth order sections by entering

```
[b1,a1] = butter(4,.25);
[b2,a2] = butter(3,.25);
```

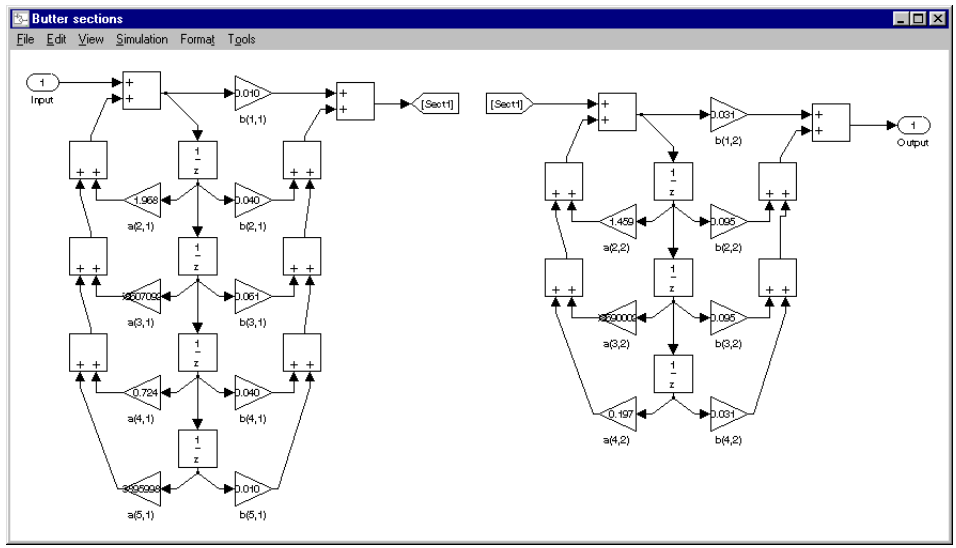
2 Configure the Wizard to use these coefficient vectors as the numerator and denominator of a Direct-Form II structure:

- Select **Direct-Form II** from the **Type** menu.
- Type `{b1,b2}` in the **Numerator** text field. Note that the numerator coefficient vector for each section is entered as an element in a cell array. Since this is a two-section filter, a two-cell array is specified in the

Numerator field. The two filter sections do not need to have the same order.

- Type `{a1,a2}` in the **Denominator** text field. Note that the denominator coefficient vector for each section is also entered as an element in a cell array. Since this is a two-section filter, a two-cell array is specified in the **Denominator** field.

3. Type a name for the new filter subsystem in the **Block Name** field. The example uses `Butter sections`.
4. Press the **Build** button to create the specified filter subsystem in a new model window.
5. Double-click the new Butter sections block to see the Direct-Form II filter realization that the Wizard created:



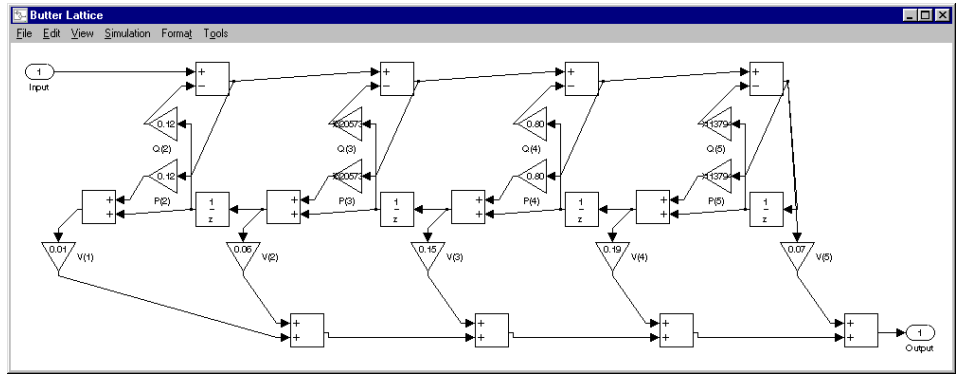
Example 4: ARMA Lattice

Design a fourth-order, quarter-band, lowpass Butterworth filter using an ARMA lattice:

1. At the MATLAB command line, compute the lattice and ladder coefficients (`k` and `v`, respectively) for the ARMA filter:

   ```
   [b,a] = butter(4,.25);
   [k,v] = tf2latc(b,a);
   ```

2. Configure the Wizard to use `k` and `v` as the coefficients of the lattice design:
   - Select **Lattice (ARMA)** from the **Type** menu.
   - Type `k` in the **Lattice Coeffs** text field.
   - Type `v` in the **Ladder Coeffs** text field.
3. Type a name for the new filter subsystem in the **Block Name** field. The example uses `Butter Lattice`.
4. Press the **Build** button to create the specified filter subsystem in a new model window.
5. Double click the new Butter Lattice block to see the ARMA filter realization that the Wizard created:



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.

See Also

Biquadratic Filter
Direct-Form II Transpose Filter
Discrete Filter (Simulink)
Time-Varying Direct-Form II Transpose Filter
Time-Varying Lattice Filter

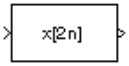
Purpose

Filter and downsample an input signal.

Library

Multirate Filters, in Filtering

Description



The FIR Decimation block resamples the input at an integer rate K times slower than the input sample rate, where K is specified by the **Decimation factor** parameter. This process consists of two steps:

- The block filters the input data with an FIR filter.
- The block downsamples the filtered data to a lower rate.

The FIR Decimation block implements the FIR filtering and downsampling steps together using a polyphase filter structure, which is more efficient than straightforward filter-then-decimate algorithms. The output of the decimator is the first filter phase.

In practice, the filter specified by the **FIR filter coefficients** vector should be a lowpass FIR with normalized cutoff frequency no greater than 1/K. The coefficients in the vector are ordered in descending powers of z.

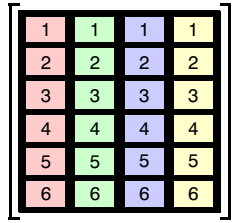
The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block decimates each channel over time.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:



Input matrix:
4 channels,
1 frame per channel,
6 samples per frame

The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block decimates each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

In frame-based operation, the **Framing** parameter determines how the block adjusts the rate at the output. There are two available options:

- **Maintain input frame rate**

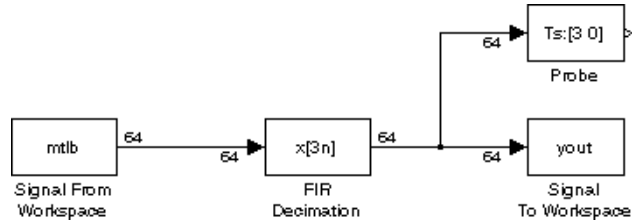
  The block generates the output at the slower (decimated) rate by using a proportionally smaller frame size than the input. For decimation by a factor of K, the output frame size is K times smaller than the input frame size, but the input and output frame rates are equal. The input frame size must be a multiple of the decimation factor.

  The example below 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.

- **Maintain input frame size**

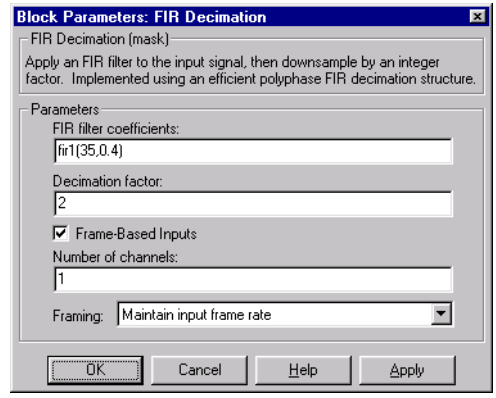
  The block generates the output at the slower (decimated) rate by using a proportionally longer frame period at the output port than at the input port. For decimation by a factor of K, the output frame period is K times longer than the input frame period, but the input and output frame sizes are equal.

  The example below shows a single-channel input (frame size = 64) with a sample period of 1 second being decimated by a factor of 3 to a sample period of 3 seconds. The input and output frame sizes are identical.



Note If you expect to generate code for the FIR Decimation block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



FIR filter coefficients

The FIR filter coefficients, in descending powers of z.

Decimation factor

The integer factor, K, by which to decrease the sample rate of the input sequence.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (channels) in the input matrix, N.

Framing

For frame-based operation, the method by which to implement the decimation; reduce the output frame rate, or reduce the output frame size.

See Also

Downsample
FIR Interpolation
FIR Rate Conversion
`decimate` (Signal Processing Toolbox)

Purpose

Upsample and filter an input signal.

Library

Multirate Filters, in Filtering

Description

The FIR Interpolation block resamples the input at an integer rate L times faster than the input sample rate, where 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 an FIR filter.

The FIR Interpolation block implements the upsampling and FIR filtering steps together using a polyphase filter structure, which is more efficient than straightforward upsample-then-filter algorithms.

In practice, the filter specified by the **FIR filter coefficients** vector (in descending powers of z) should be a lowpass FIR with a normalized cutoff frequency no greater than 1/L. The coefficients are scaled by L.

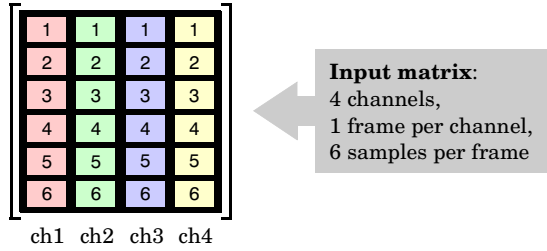
The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block interpolates each channel over time.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns, N) in the matrix, and the block interpolates each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

In frame-based operation, the **Framing** parameter determines how the block adjusts the rate at the output. There are two available options:

- **Maintain input frame rate**

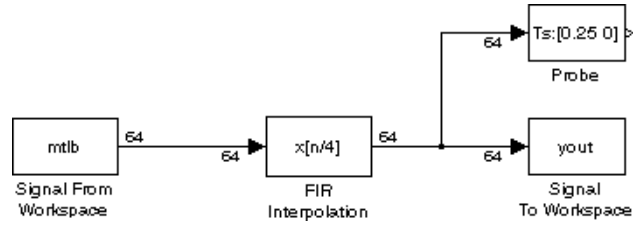
  The block generates the output at the faster (interpolated) rate by using a proportionally larger frame size than the input. For interpolation by a factor of L, the output frame size is L times larger than the input frame size, but the input and output frame rates are equal.

  The example below shows a single-channel input of frame size 16 being upsampled by a factor of 4 to a frame size of 64. The block's input and output frame rates are identical.

- **Maintain input frame size**

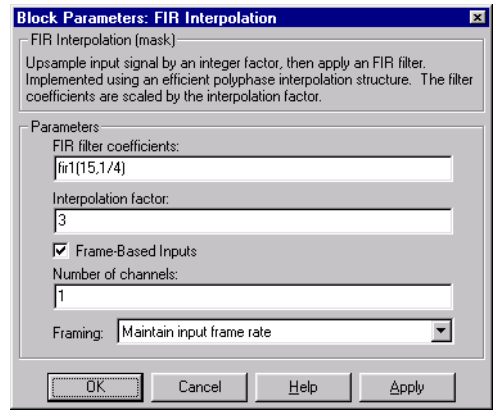
  The block generates the output at the faster (interpolated) rate by using a proportionally shorter frame period at the output port than at the input port. For interpolation by a factor of L, the output frame period is L times shorter than the input frame period, but the input and output frame sizes are equal.

  The example below shows a single-channel input (frame size = 64) with a frame period of 1 second being upsampled by a factor of 4 to a frame period of 0.25 seconds. The input and output frame sizes are identical.



Note If you expect to generate code for the FIR Interpolation block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



FIR filter coefficients

The FIR filter coefficients, in descending powers of z.

Interpolation factor

The integer factor, L, by which to increase the sample rate of the input sequence.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (channels) in the input matrix, N.

Framing

For frame-based operation, the method by which to implement the interpolation: increase the output frame rate, or increase the output frame size.

See Also

FIR Decimation
FIR Rate Conversion
Upsample
`interp` (Signal Processing Toolbox)

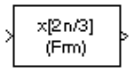
Purpose

Upsample, filter, and downsample an input signal.

Library

Multirate Filters, in Filtering

Description



The FIR Rate Conversion block resamples the input to a period K/L times the input sample period, where K is specified by the **Decimation factor** parameter and L is specified by the **Interpolation factor** parameter. The resampling process consists of the following steps:

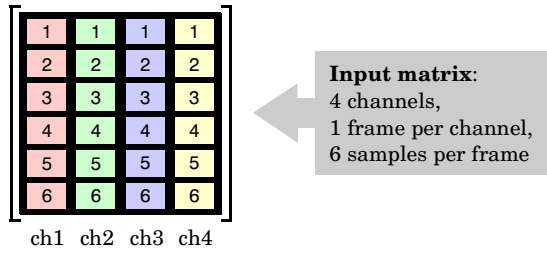
- The block upsamples the input to a higher rate by inserting L-1 zeros between input samples.
- The upsampled data is passed through an FIR filter.
- The block downsamples the filtered data by a factor of K.

K and L must be *relatively prime* integers; that is, the ratio K/L cannot be reducible. The FIR Rate Conversion block implements the three steps together using a polyphase filter structure, which is more efficient than straightforward upsample-filter-decimate algorithms. The output of the interpolator is the first filter phase, while the output of the decimator is the last filter phase. When both K and L are greater than 1, the output is the last decimation phase from the first interpolation phase.

The filter specified by the **FIR filter coefficients** vector should be a lowpass FIR with a normalized cutoff frequency no greater than `min(1/L,1/K)`. The coefficients in the vector are ordered in descending powers of z.

Frame-Based Operation

This block *always* operates in frame-based mode, and expects an M_i-by-N frame matrix input. Each of the N frames in the matrix contains M_i sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:

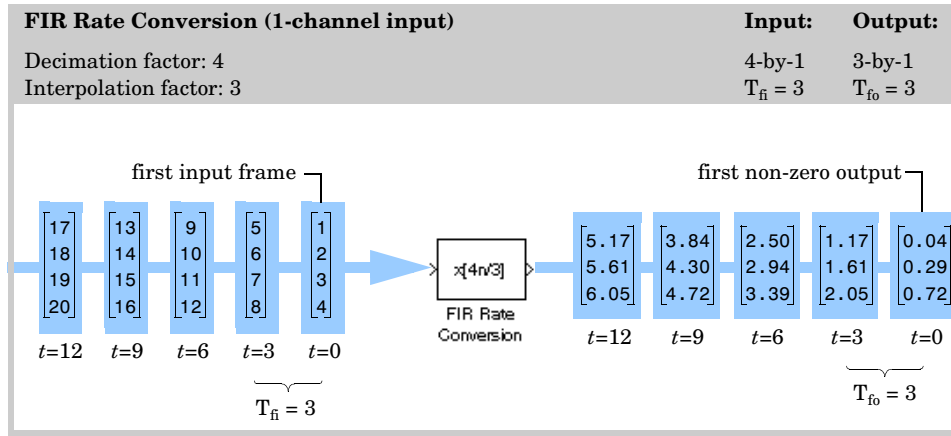


The **Number of channels** parameter specifies the number of independent channels (columns, N) in the matrix, and the block resamples each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

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 ($T_{fo}=T_{fi}$).



Diagnostics

An error is generated if the relation between K and L shown above is not satisfied.

```
(Input port width)/(Output port width) must equal the
(Decimation factor)/(Interpolation factor).
```

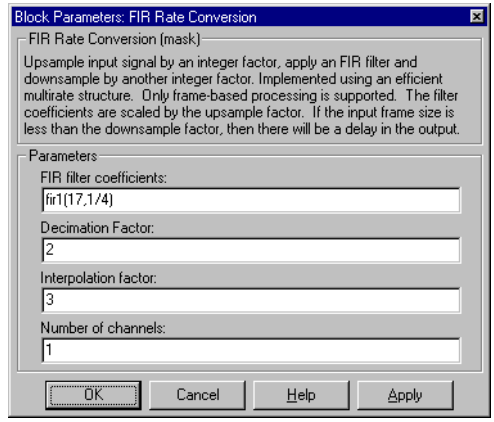
A warning is generated if L and K are not relatively prime; that is, if the ratio L/K can be reduced to a ratio of smaller integers.

```
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.

Note If you expect to generate code for the FIR Rate Conversion block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



FIR filter coefficients

The FIR filter coefficients, in descending powers of z.

Decimation factor

The integer factor, K, by which to downsample the signal after filtering.

Interpolation factor

The integer factor, L, by which to upsample the signal before filtering.

Number of channels

The number of columns (channels) in the input matrix, N.

References

Fliege, N. J. *Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets*. West Sussex, England: John Wiley & Sons, 1994.

See Also

Downsample
FIR Decimation
FIR Interpolation
Upsample
`upfirdn` (Signal Processing Toolbox)

Purpose

Reverse the elements in a vector.

Library

Vector Functions, in Math Functions

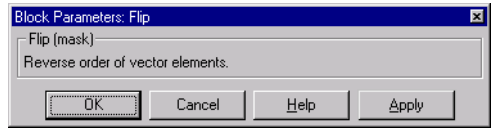
Description

The Flip block reverses the input so that the first element of the input vector is the last element of the output vector, and vice versa:

```
y = flipud(u(:))    % equivalent MATLAB code
```

A matrix input, `u`, is treated as a vector, `u(:)`.

Dialog Box



See Also

Selector (Simulink)
Transpose
Variable Selector
`flipud` (MATLAB)
`fliplr` (MATLAB)

Forward Substitution

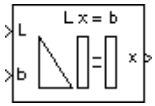
Purpose

Solve the equation L*x*=b for lower triangular matrix L.

Library

Linear Algebra, in Math Functions

Description

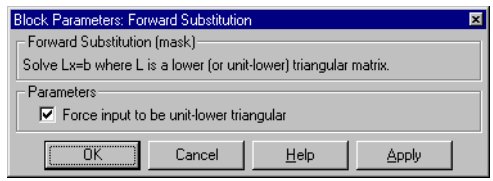


The Forward Substitution block solves the linear system L*x*=b by simple forward substitution of variables, where L is a lower triangular square matrix. The output is the vector solution of the equations, *x*.

The block only uses the elements in the lower triangle of the input; the upper elements are ignored. When **Force input to be unit-lower triangular** is selected, the block replaces the elements on the diagonal of L with ones. 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.

The block may generate `NaN` or `Inf` for underdetermined or inconsistent systems.

Dialog Box



Force input to be unit-lower triangular (T)

Replaces the elements on the diagonal of L with ones when selected.

See Also

Backward Substitution
Cholesky Solver
LDL Solver
Levinson Solver
LU Solver
QR Solver

Purpose

Display frequency-domain frame-based data.

Library

DSP Sinks

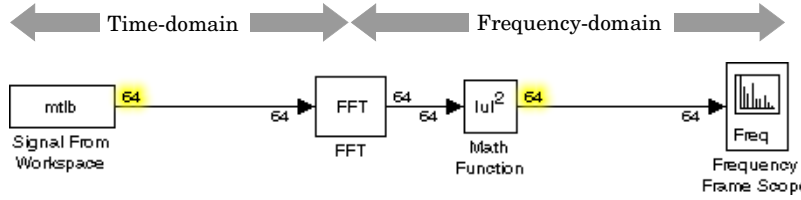
Description

The Frequency Frame Scope block is similar to the Time Frame Scope, but plots frequency-domain data instead of time-domain data. For a complete discussion of this block's axis properties, line properties, scope window, and frame-based operation, see the Time Frame Scope block reference.

The block assumes that the each length-M input frame is a vector of magnitude data corresponding to M ascending frequencies. That is, each data point in the input frame, u, is assumed to correspond to a unique frequency value, $u=u(f)$, where $f_{i+1}>f_i$.

In order to correctly scale the frequency axis (i.e., to determine the frequencies that the data in the input frame should be plotted against), the block needs to know the sample period of the *original time-domain sequence* represented by the frequency-domain data. The **Sample time of original time-series** parameter allows you to specify this in two different ways:

Auto-Detect from Input Sample Period. A value of -1 for this parameter instructs the block to reconstruct the frequency data from the frame-period of the frequency-domain input. This parameter setting is appropriate when each frame of frequency-domain data shares the same length as the frame of time-domain data that it was generated from.



Time-domain frame size = Frequency-domain frame size

This is the case when the FFT is computed on the same number of points as are contained in the time-domain input (as shown above). Blocks that use the FFT internally, for example those in the Power Spectrum Estimation library, usually provide an **FFT size** parameter to specify the number of frequency points in the output. When this parameter is set to the size of the time-domain

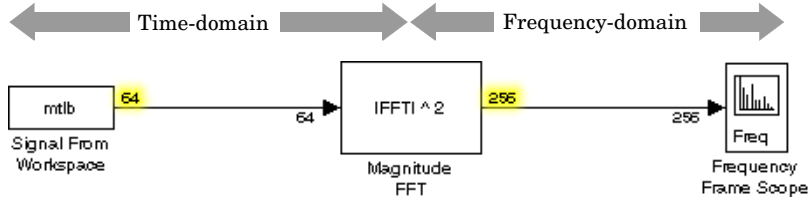
input, the frequency-domain output and time-domain input have the same frame size.

When the frequency-domain input to the Frequency Frame Scope block is related to the original time-domain signal in this way, the block can compute the original time-domain sample period from the frequency-domain input frame size and frame period.

Note that the auto-detect mode makes the following two assumptions:

- The sample period of the time-domain signal in the simulation is equal to the period with which the physical signal was originally sampled. For example, the `mtlb` signal imported from the workspace in the model above has an actual sample period of `1/Fs = 1/7418`. Although the **Sample time** parameter in the Signal From Workspace block can legitimately be set to any value, for the auto-detect mode to compute the correct frequency scaling, the **Sample time** parameter must be set to the original sample period of `1/7418`.
- 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.

Enter the Appropriate Time-Domain Sample Period. Enter the sample period of the original time-series. This is necessary when the frame size of frequency-domain data is *not* identical to the frame size of the time-domain data it was generated from.



Time-domain frame size ≠ Frequency-domain frame size

This is the case when the FFT is computed on a different number of samples (more or fewer) than are contained in the time-domain input. When the **FFT size** parameter of FFT-based blocks is set to a value other than the size of the input frame, the block either zero-pads or truncates the input before performing the FFT, and the frequency-domain output and time-domain input have different frame sizes.

When the time-domain signal is zero-padded or truncated before transformation to the frequency domain, the Frequency Frame Scope block *cannot* automatically compute the original time-domain sample period. You should specify the original sample period explicitly in the **Sample time of original time-series** parameter.

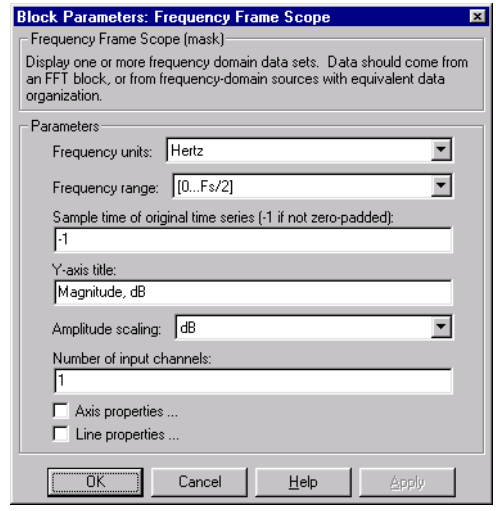
You also need to explicitly specify the time-series' actual sample period when either of the assumptions listed above for the auto-detect mode are not valid.

The **Frequency units** parameter specifies whether the frequency axis values should be in units of Hertz or rads/sec, and 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 ($F_s/2$ is the Nyquist frequency). All of the FFT-based blocks in the DSP Blockset, including those in the Power Spectrum Estimation library, compute the FFT at frequencies in the range $[0,F_s)$.

If the **Frequency units** parameter specifies **Hertz**, the spacing between frequency points is $1/(M*T_s)$. For **Frequency units** of **rads/sec**, the spacing between frequency points is $2\pi/(M*T_s)$. The **Amplitude scaling** parameter allows you to select magnitude or dB scaling along the *y*-axis.

The scope updates the display for each new input frame.

Dialog Box



Frequency units Ⓣ

The frequency units for the *x*-axis, **Hertz** or **rads/sec**.

Frequency range Ⓣ

The frequency range over which to plot the data, **[0..Fs/2]**, **[-Fs/2..Fs/2]**, or **[0..Fs]**, where F_s is the sample frequency of the original time-domain signal, $1/T_s$.

Sample time of original time series Ⓣ

The sample period of the original time-domain signal, T_s. Set to `-1` to auto-detect the time-domain sample period from the frame period and frame size of the frequency-domain input (use only if the time-domain data was not zero-padded prior to the FFT).

Y-Axis title

The text to be displayed to the left of the *y*-axis.

Amplitude scaling Ⓣ

The scaling for the *y*-axis, **dB** or **Magnitude**.

Number of input channels

The number of channels (columns) in the input matrix.

Axis properties

Select to expose the **Axis Properties** panel. See Time Frame Scope for more information.

Line properties

Select to expose the **Line Properties** panel. See Time Frame Scope for more information.

See Also

FFT Frame Scope
Time Frame Scope
User-Defined Frame Scope

Purpose

Read audio data from a standard audio device in real-time (Windows 95/98/NT only).

Library

DSP Sources

Description

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, deselect **Use default audio device**, and enter the desired device identification number in the **Audio device ID** parameter. The device ID is an integer value that the block associates with the sound device. A 3-device system, for example, has device ID numbers of 1, 2, and 3.

The output from the block, *y*, is a vector containing a length-M frame of audio data from a mono signal, or an M-by-2 matrix containing one frame of audio data from each channel of a stereo signal. If the audio source contains two channels, the **Stereo** check box should be selected. The frame size, M, is specified by the **Samples per frame** parameter.

The amplitude of the input from the sound device should be in the range ±1. Values outside this range are clipped to the nearest allowable value. If the audio signal is saturating at ±1, you can reduce the microphone gain from the **Multimedia Properties** window (available through the Windows 95/98/NT **Control Panel**). The audio data is processed in uncompressed PCM (pulse code modulation) 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. Two 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

The 16-bit sample width setting requires more memory but yields better fidelity. The output from the block is independent of the **Sample Width (bits)** setting, and is always double precision.

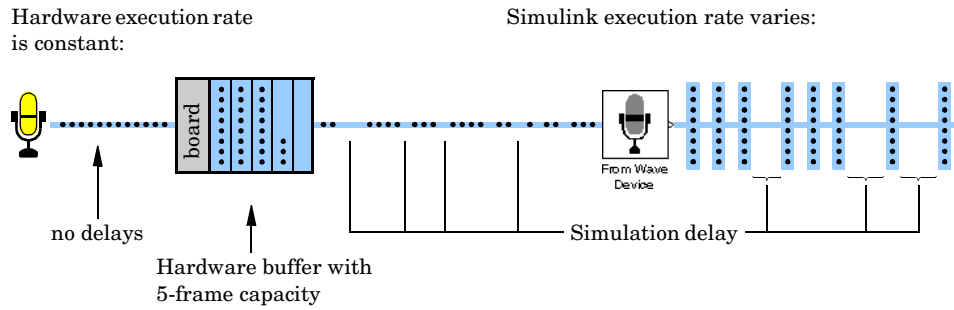
Buffering

Since the audio device accepts real-time audio input, Simulink 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 in principle 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 on, first off) and collecting them in length-M frames for output. As the audio device continues to append inputs to the *front* of the buffer, the From Wave Device block continues to pull inputs off the back 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 8kHz, this small buffer could hold approximately 0.005 seconds of data.)



If 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 hardware throughput rate is higher than the simulation throughput rate, and the buffer tends to fill over the duration of the simulation.

If the buffer size is too small in relation to the simulation throughput rate, the buffer may fill before the entire length of signal is processed. This usually results in a device error or undesired device output. When the device fails to process the entire signal length because the buffer prematurely fills, 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 may increase model latency.

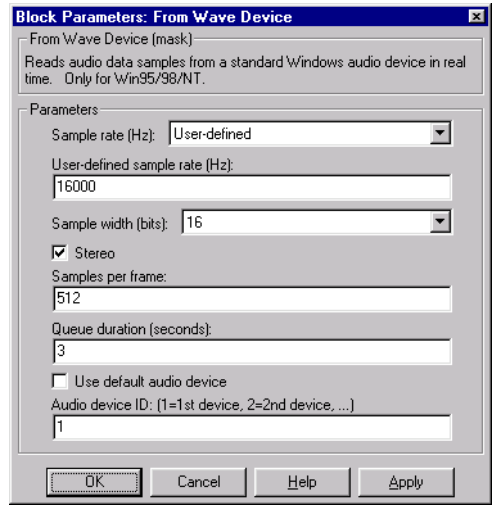
- *Increase the simulation throughput rate*

  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 scalar 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 RTW. 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 "Increasing Performance" in Chapter 2 of this book, and "Improving Simulation Performance and Accuracy" in Chapter 4 of *Using Simulink*, 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 checked, mono (one-channel) inputs when unchecked.

Samples per frame

The number of audio samples in each successive output frame.

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. Deselect to enable the **Audio device ID** parameter and manually enter a device ID number.

Audio device ID

The number of the audio device from which to read the audio output. In a system with several audio devices installed, a value of `1` selects the first audio card, a value of `2` selects the second audio card, and so on. Select **Use default audio device** if the system has only a single audio card installed.

See Also

From Wave File
To Wave Device

From Wave File

Purpose

Read audio data from a Microsoft Wave (`.wav`) file (Windows 95/98/NT only).

Library

DSP Sources

Description

The From Wave File block reads audio data from a Microsoft Wave (`.wav`) file and outputs a double-precision signal with amplitudes in the range ±1. The audio data must be in the uncompressed PCM (pulse code modulation) format.

```
y = wavread('filename')    % equivalent MATLAB code
```

The **File name** parameter can specify an absolute or relative path to the file. If the file is on the MATLAB path or in the current directory (the directory returned by typing `pwd`), you need only specify the file's name. You do not need to specify the `.wav` extension in either case.

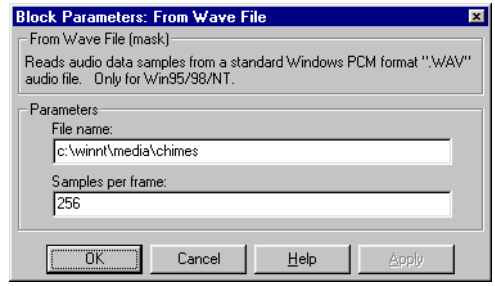
The output from the block, *y*, is a length-M frame of audio data from a mono signal, or an M-by-2 matrix containing one frame of audio data from each channel of a stereo signal. The frame size, M, is specified by the **Samples per frame** parameter. The output frame period, T_{fo}, is

$$T_{fo} = \frac{M}{F_s},$$

where F_s is the data sample rate in Hz.

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 or 16 bits.

Dialog Box



File name

The path and name of the file to read. Paths can be relative or absolute.

Samples per frame

The number of samples in each output frame.

See Also

From Wave Device
Signal From Workspace
To Wave File
`wavread` (MATLAB)

Histogram

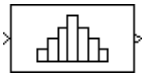
Purpose

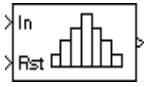
Compute the histogram (frequency distribution) of an input or sequence of inputs.

Library

Statistics, in Math Functions

Description





The Histogram block computes the frequency distribution of the elements in the input vector, or tracks the frequency distribution in a sequence of inputs over a period of time. The **Running histogram** parameter allows you to toggle between basic operation and running operation, which are both described below.

Basic Operation

When the **Running histogram** check box is *not* selected, the block computes the frequency distribution in the input independently at each sample time. The block sorts the elements of the input (by their absolute value) into a number of discrete *bins*, as specified by the **Number of bins** parameter, n.

```
y = hist(u(:),n)   % equivalent MATLAB code
```

The upper-boundary of the highest-valued bin is specified by the **Maximum value of input** parameter (B_M), and the lower-boundary of the lowest-valued bin is specified by the **Minimum value of input** parameter (B_m). Each bin has width

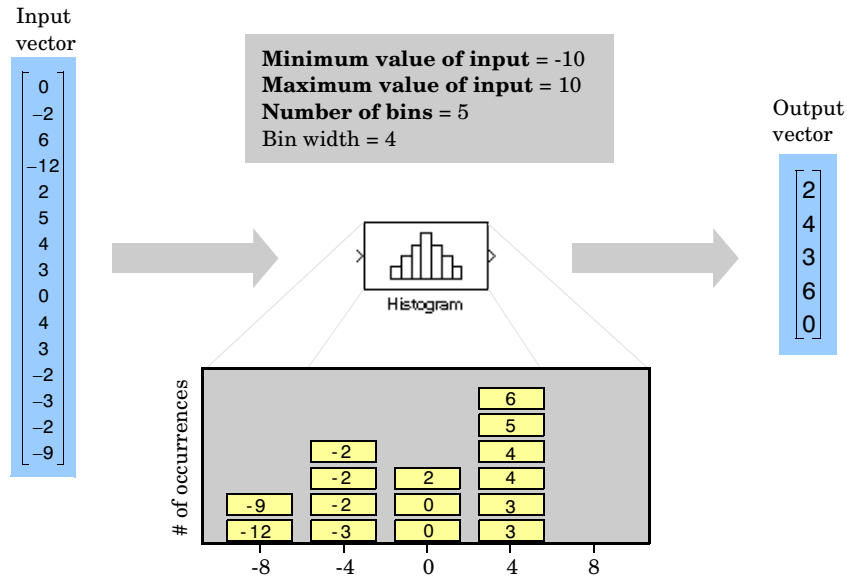
$$\Delta = \frac{B_M - B_m}{n}$$

and the bin centers are located at the following values:

$$B_m + \left(k + \frac{1}{2}\right)\Delta \qquad k = 0, 1, 2, \ldots, n-1$$

Input values that fall on the borders between bins are sorted 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 **Maximum value of input** parameter or less than **Minimum value of input** parameter are sorted into the nearest bin.

At each sample time, the block outputs a length-*n* vector whose elements (one per bin) represent the *frequency of occurrence* of the binned input values. The following example illustrates the block's operation for parameter values of B_m = -10, B_M = 10, and n = 5. The resulting bin width is 4.



When the **Normalized** check box is selected, the block scales the output so that `sum(y)=1`.

Note that a matrix input is sorted as a single vector, `u(:)`.

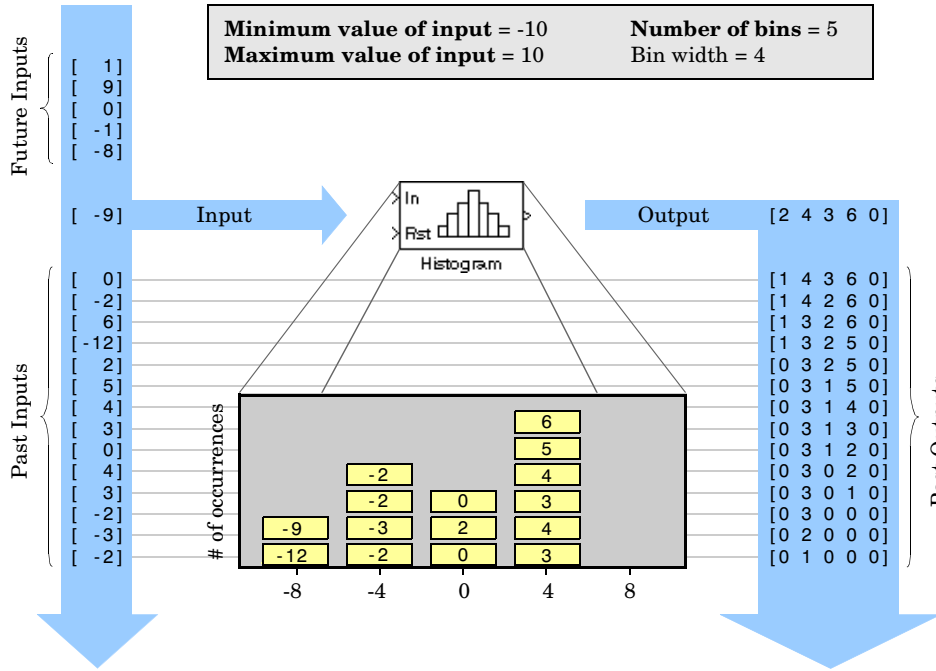
Running Operation

When the **Running histogram** check box is selected, the block tracks the frequency distribution of a sequence of inputs over time. You can choose frame-based or sample-based operation by selecting (or deselecting, respectively) the **Frame-based** check box.

Sample-Based Operation. When the **Frame-based** check box is *not* selected (default), the block accepts a scalar input at the `In` port, and adds each successive scalar input value to the histogram. The block retains past inputs in the histogram as long as the `Rst` input remains zero. When the block receives a nonzero scalar value at the `Rst` port, it resets the histogram by emptying all

the bins. If you do not need to reset the running histogram during the simulation, you can delete the `Rst` port from the block icon by deselecting the **Reset port** check box.

At each sample time the block outputs a vector whose elements (one per bin) represent the *frequency of occurrence* of the binned input values currently in the histogram. The following example illustrates the block's operation for parameter values of B_m = -10, B_M = 10, and n = 5. The resulting bin width is 4.

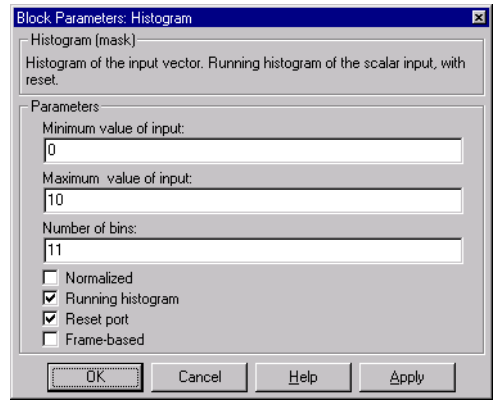


Note that the block has direct feedthrough, so the output histogram is updated with the new input at the same sample time that it is received.

Frame-Based Operation. When the **Frame-based** check box is selected, the block accepts a length-M frame vector containing M sequential time-samples from a single channel. The elements in each successive vector input are added the histogram and retained as long as the `Rst` input remains zero. When the block receives a nonzero value at the `Rst` port, it resets the histogram by emptying all the bins.

Note If you expect to generate code for the Histogram block's running mode using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Minimum value of input

The lower boundary, B_m, of the lowest-valued bin.

Maximum value of input

The upper boundary, B_M, of the highest-valued bin.

Number of bins

The number of bins, n, in the histogram.

Normalized (T)

Normalizes the output vector to `1`. This parameter is not tunable in Simulink's external mode.

Running histogram

Selects running operation.

Reset port

Display `Rst` input port.

Frame-based
Selects frame-based operation.

See Also

Sort
`hist` (MATLAB)

Purpose

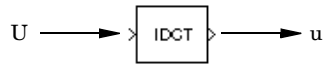
Compute the IDCT of the input.

Library

Transforms, in General DSP

Description

The IDCT block computes the inverse discrete cosine transform (IDCT) of the input frame. For a length-M input U, the IDCT is given by:



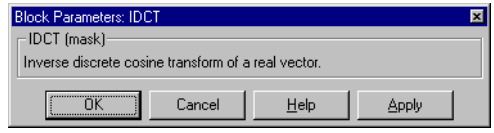
$$u(m) = \sum_{k=1}^{M} w(k)U(k)\cos\frac{\pi(2m-1)(k-1)}{2M}, \qquad m = 1, \ldots, M$$

where

$$w(k) = \begin{cases} \frac{1}{\sqrt{M}}, & k = 1 \\ \sqrt{\frac{2}{M}}, & 2 \le k \le M \end{cases}$$

Multichannel inputs (i.e., frame matrices) are not accepted. The output of the block is a frame with the same size, period, and data type (real/complex) as the input.

Dialog Box



See Also

DCT
IFFT
`idct` (Signal Processing Toolbox)

IFFT

Purpose

Compute the complex-valued IFFT of a complex input.

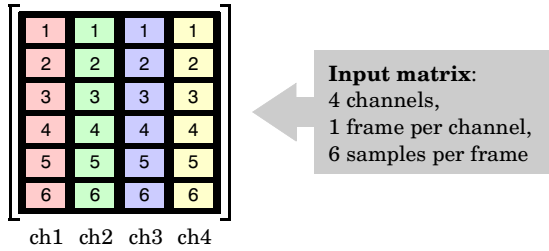
Library

Transforms, in General DSP

Description

The IFFT block computes the inverse fast Fourier transform (IFFT) of each complex input channel independently at each sample time. The block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal.

The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix.

Select **Complex** from the **Output** parameter to output the full complex result of the IFFT.

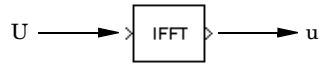
```
U = ifft(u)        % equivalent MATLAB code
```

Select **Real** to output only the real part of the result.

```
U = real(ifft(u))  % equivalent MATLAB code
```

If the input to the block is conjugate symmetric, you should select the **Conjugate symmetric input** check box, which instructs the block to use an appropriate algorithm and generate a purely real output. A common source of conjugate symmetric data is the FFT block, whose output is conjugate symmetric when the input is purely real.

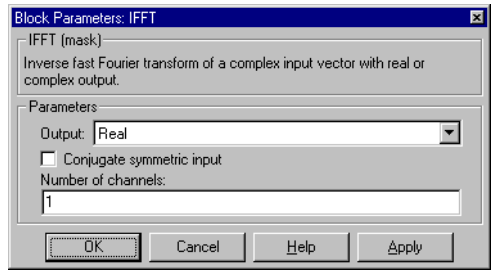
The IFFT operation for a single-channel input (**Number of channels** = 1) is shown below.



$$u(m) = \frac{1}{M}\sum_{k=0}^{M-1} U(k)e^{j2\pi(mk/M)} \qquad m = 0, \ldots, M-1$$

The input frame size, M, must be a power of two. To work with other frame sizes, use the Zero Pad block to pad or truncate the input frame to a power-of-two length.

Dialog Box



Output

The data type of the output, real or complex.

Conjugate symmetric input

Specifies (when checked) that the input is conjugate symmetric.

Number of channels

The number of channels (columns) in the input.

See Also

FFT
IDCT
Zero Pad
`ifft` (MATLAB)

Inherit Complexity

Purpose

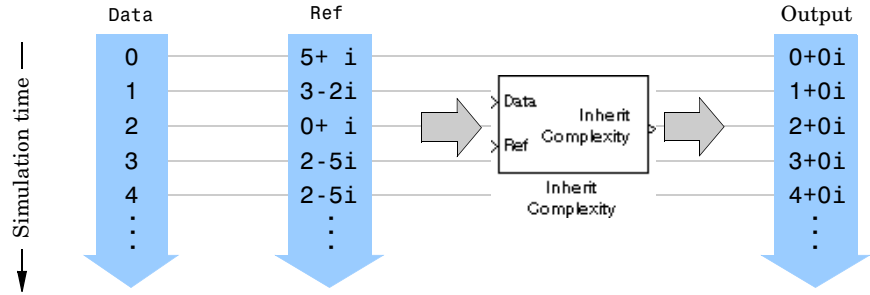
Change the complexity of the input to match that of a reference signal.

Library

Elementary Functions, in Math Functions

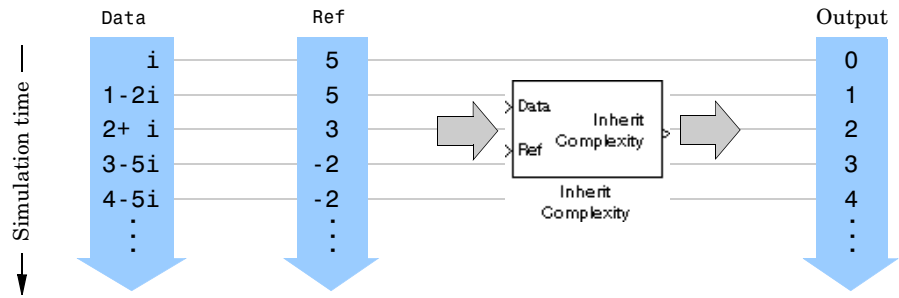
Description

The Inherit Complexity block alters the input data at the `Data` port to match the complexity of the reference input at the `Ref` port. If 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.



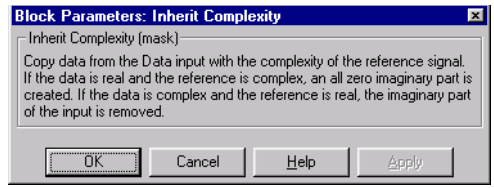
If both the `Data` input and `Ref` input are real, the block propagates the `Data` input with no change.

If the `Data` input is complex, and the `Ref` input is real, the block outputs the real component of the `Data` input.



If both the `Data` input and `Ref` input are complex, the block propagates the `Data` input with no change.

Dialog Box



See Also

Complex to Magnitude-Angle (Simulink)
Complex to Real-Imag (Simulink)
Magnitude-Angle to Complex (Simulink)
Real-Imag to Complex (Simulink)

Integer Delay

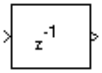
Purpose

Delay an input by an integer number of sample periods.

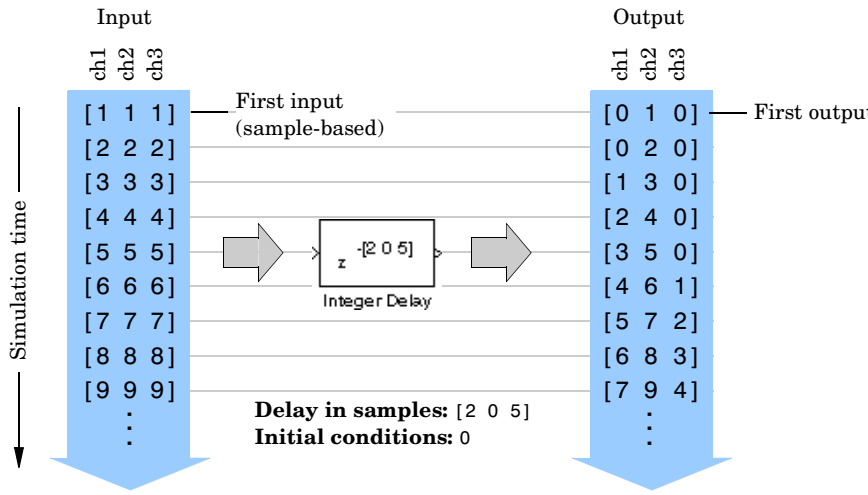
Library

Signal Operations, in General DSP

Description



The Integer Delay block delays a discrete-time input by the number of sample intervals specified in the **Delay in samples** parameter. This can be a scalar value by which to equally delay all N input channels, or a vector containing one delay value for each input channel, `[D(1) D(2) ... D(N)]`. All the samples in channel 1 are uniformly delayed by `D(1)` sample intervals, all the samples in channel 2 are uniformly delayed by `D(2)` sample intervals, and so on. Noninteger delay values are rounded to the nearest integer.



The **Frame-based inputs** check box allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

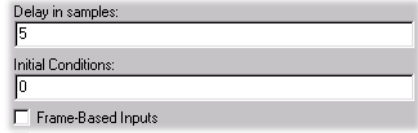
When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block delays each channel as specified by the **Delay in samples** parameter.

The **Initial conditions** parameter specifies the output of the block during the initial delay. Both fixed and time-varying initial conditions can be specified in

a variety of ways to suit the dimensions of the input. 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. For a fixed initial condition, 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 sample-based mode can be specified in one of the following ways:

- *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,

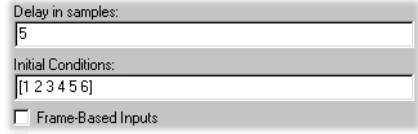
  

  the block outputs a sequence of five M-by-N zero-matrices at the start of the simulation. A scalar initial condition can be used with inputs of any dimension.

- *Vector* containing M∗N elements from which to construct an M-by-N *matrix* to be repeated at the output for each sample time of the initial delay. M and N are the number of rows and columns, respectively, in the input matrix. The initial condition vector, `ic`, is reshaped columnwise to match the input matrix dimensions.

  ```
  y = reshape(ic,M,N)    % equivalent MATLAB code
  ```

  For a 2-by-3 input, and the parameters below,

  the block outputs the matrix

$$\begin{bmatrix} 1 & 3 & 5 \\ 2 & 4 & 6 \end{bmatrix}$$

for the first five sample times. An initial condition of length M∗N can be used with inputs of any dimension, and can be specified as either a row or column vector.

- *Matrix* of dimension M-by-N to be repeated at the output for each sample time of the initial delay, where M and N are the number of rows and columns, respectively, in the input matrix. For a 2-by-3 input, and the parameters below,

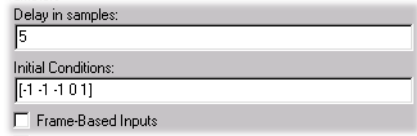
the block outputs the matrix

$$\begin{bmatrix} 1 & 2 & 3 \\ 4 & 5 & 6 \end{bmatrix}$$

for the first five sample times. For cases where M=N=1 or M=1, the initial condition setting reduces to a scalar or a vector, described above.

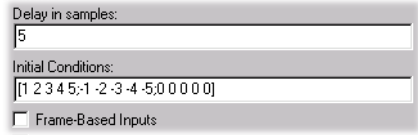
Time-Varying Initial Conditions. The following settings specify *time-varying* initial conditions. For a time-varying initial condition, the values specified in the **Initial conditions** parameter are output in sequence during the initial delay. This allows you to specify a unique output value for each sample of the initial delay. 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 value specified for the **Delay in samples** parameter. 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,



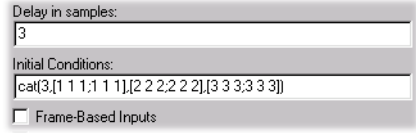
the block outputs values `-1`, `-1`, `-1`, `0`, `1` in sequence over the first five sample times. A length-D vector initial condition can only be used with scalar inputs.

- *Matrix* of dimension N-by-D, where N is the length of the input *sample vector*, and D is the value specified for the **Delay in samples** parameter (the *maximum* value if **Delay in samples** is a vector). The D columns of the matrix are output in sequence, one at each sample time of the initial delay. For a 1-by-3 input, and the parameters below,



the block outputs the sequence `[1 -1 0]`, `[2 -2 0]`, `[3 -3 0]`, etc., for the first 5 sample times. A matrix initial condition can only be used with vector inputs.

- *Array* of dimension M-by-N-by-D, where D is the value specified for the **Delay in samples** parameter (the *maximum* value if the **Delay in samples** 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

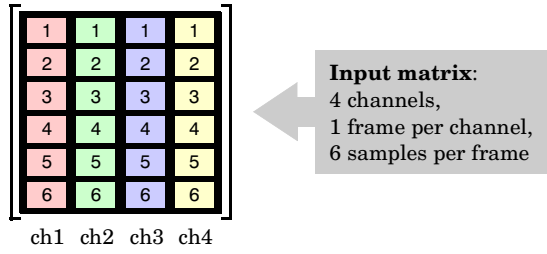
$$\begin{bmatrix} 1 & 1 & 1 \\ 1 & 1 & 1 \end{bmatrix}, \begin{bmatrix} 2 & 2 & 2 \\ 2 & 2 & 2 \end{bmatrix}, \begin{bmatrix} 3 & 3 & 3 \\ 3 & 3 & 3 \end{bmatrix}$$

at the start of the simulation. An array initial condition can only be used with matrix inputs.

In sample-based mode, the Integer Delay block has direct feedthrough (inputs are available immediately at the output) when the delay on any channel is zero. As a result, Simulink may generate an error when the block is used with a zero-delay setting in feedback loops that do not contain at least one block *without* direct feedthrough.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent channel. The illustration below shows a 6-by-4 matrix input:



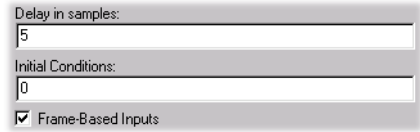
The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block delays each channel as specified by the **Delay in samples** parameter. Frame-based operation provides substantial increases in throughput rates at the expense of greater model latency.

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

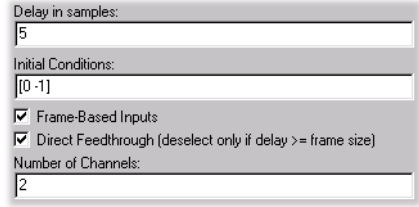
output for each sample time of the initial delay. A fixed initial condition in frame-based mode can be one of the following:

- *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. Note that if the frame size is larger than the delay, all of these zeros are all included in the first output from the block.

- *Vector* containing N samples to be repeated at each sample time of the initial delay, where N is the **Number of channels**. 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 matrices at the start of the simulation.

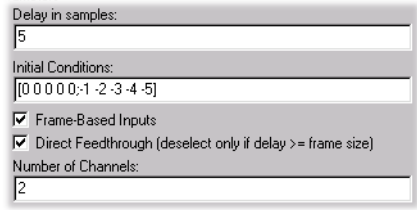
$$\begin{bmatrix} 0 & -1 \\ 0 & -1 \\ 0 & -1 \\ 0 & -1 \end{bmatrix}, \begin{bmatrix} 0 & -1 \\ 1 & 1 \\ 2 & 2 \\ 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 \\ 5 & 5 \\ 6 & 6 \\ 7 & 7 \end{bmatrix}, \ldots$$

  (Note that the first five output samples of channel 1 are zero, and the first five output samples of channel 2 are `-1`, as specified.) If the input frame size is 1, then this operation equivalent to the sample-based operation described above.

Time-Varying Initial Condition. The following setting specifies a *time-varying* initial condition. For a time-varying initial condition, 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 way:

- *Matrix* of dimension N-by-D, where N is the **Number of channels** in the input, and D is the value specified for the **Delay in samples** parameter (the *maximum* value if the **Delay in samples** is a vector). The D columns of the matrix are transposed and output in sequence (as rows), one at each sample time of the initial delay. Note that if the frame size is larger than the delay, all of the columns of the initial condition matrix are included (as rows) in the first matrix output from the block.

  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.

  $$\begin{bmatrix} 0 & -1 \\ 0 & -2 \\ 0 & -3 \\ 0 & -4 \end{bmatrix}, \begin{bmatrix} 0 & -5 \\ 1 & 1 \\ 2 & 2 \\ 3 & 3 \end{bmatrix}, \begin{bmatrix} 4 & 4 \\ 5 & 5 \\ 6 & 6 \\ 7 & 7 \end{bmatrix}, \ldots$$

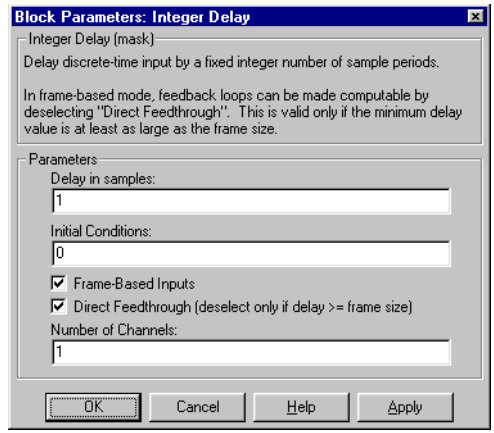
  (Note that the first five output samples of channel 1 are zero, and the first five output samples of channel 2 is the sequence `-1`, `-2`, `-3`, `-4`, `-5`, as specified.) If the input frame size is 1, then this operation equivalent to the sample-based operation described above.

In frame-based mode, the block has direct feedthrough when the **Direct feedthrough** check box is selected. In this case, Simulink may generate an error when the block is used in a feedback loop that does not contain at least one block *without* direct feedthrough.

If you are certain that all delays are greater than or equal to the frame size, *or* that there is no possibility of the block's inclusion in a feedback loop, you can safely deselect the **Direct feedthrough** check box. If you deselect the **Direct feedthrough** check box under other conditions, Simulink may generate an error.

Note If you expect to generate code for the Integer Delay block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Delay in samples

The number of sample periods to delay the input signal.

Initial conditions

The value of the block's output during the initial delay.

Frame-based inputs

Selects frame-based operation.

Direct feedthrough

When selected, specifies that the block has direct feedthrough in frame-based mode.

Number of channels

For frame-based operation, the number of columns (frames) in the input matrix.

See Also

Unit Delay (Simulink)
Variable Fractional Delay
Variable Integer Delay

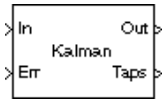
Purpose

Compute filter estimates for an input using the Kalman adaptive filter algorithm.

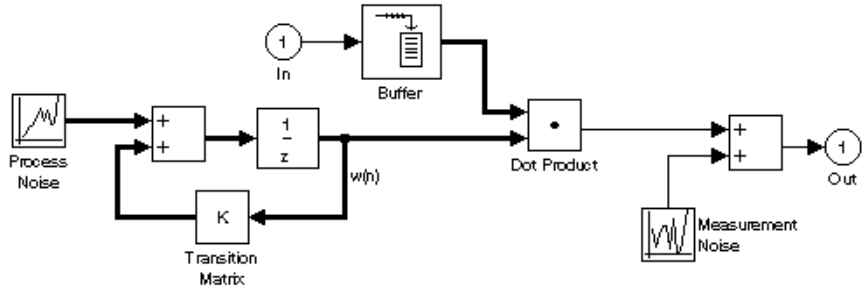
Library

Adaptive Filters, in Filtering

Description



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 particular Kalman filter algorithm is based on the following physical realization of a dynamical system:



The Kalman filter assumes that there are no deterministic changes to the filter taps over time (i.e., 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:

$$g(n) = \frac{K(n-1)u(n)}{u^T(n)K(n-1)u(n) + Q_M}$$

$$y(n) = u^T(n)\hat{w}(n)$$

$$e(n) = d(n) - y(n)$$

$$\hat{w}(n+1) = \hat{w}(n) + e(n)g(n)$$

$$K(n) = K(n-1) - g(n)u^T(n)K(n-1) + Q_P$$

The variables are as follows.

Variable	Description
n	The current algorithm iteration
$u(n)$	The buffered input samples at step n
$K(n)$	The input covariance matrix at step n
$g(n)$	The vector of Kalman gains 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
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 box 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 sometimes called the *random-walk Kalman filter*.

Note that the implementation of the algorithm in the block does not precisely parallel the above equations; symmetry of the input covariance matrix $K(n)$ is exploited to decrease 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.

Block Ports	Corresponding Variables
`In`	u, the scalar input, which is internally buffered into the vector $u(n)$ used by the algorithm
`Out`	$y(n)$, the filtered scalar output

Block Ports	Corresponding Variables
`Err`	$e(n)$, the scalar estimation error
`Taps`	$\hat{w}(n)$, the vector of filter-tap estimates

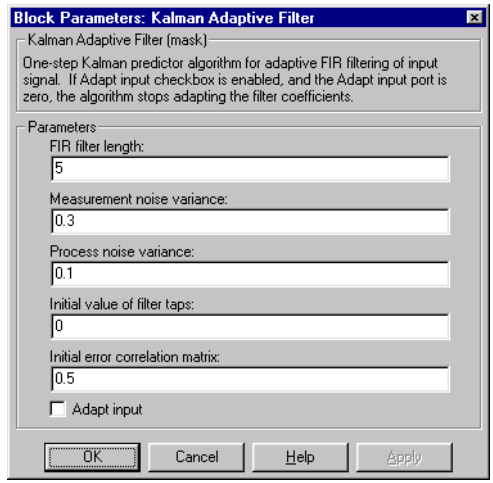
An optional `Adapt` input port is added when the **Adapt input** check box is selected in the dialog box. When this port is enabled, the block continuously adapts the filter coefficients while the `Adapt` input is nonzero. A zero-valued input to the `Adapt` port causes the block to stop adapting, and to hold the filter coefficients at their current values until the next nonzero `Adapt` input.

The **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** is specified by a scalar to be repeated for the diagonal elements of the matrix. 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.

Kalman Adaptive Filter

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.

Process noise variance Ⓣ

The value to appear along the diagonal of the process noise correlation matrix.

Initial value of filter taps

The initial FIR filter coefficients.

Initial error correlation matrix

The initial value of the error correlation matrix.

Adapt input

Enables the `Adapt` port.

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.

See Also

LMS Adaptive Filter
RLS Adaptive Filter

LDL Factorization

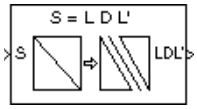
Purpose

Factor a Hermitian positive definite matrix into lower, upper, and diagonal components.

Library

Linear Algebra, in Math Functions

Description



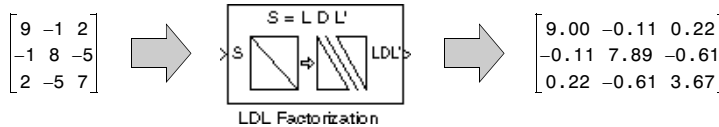
The LDL Factorization block uniquely factors the Hermitian positive definite input matrix S as

$$S = LDL^H$$

where L is a lower triangular square matrix with unity diagonal elements, D is a diagonal matrix, and L^H denotes the Hermitian transpose of L. The block's output is a composite matrix with lower triangle L, diagonal D and upper triangle L^H. The format is shown below for a 5-by-5 matrix.

$$\begin{bmatrix} d_{11} & u_{12} & u_{13} & u_{14} & u_{15} \\ l_{21} & d_{22} & u_{23} & u_{24} & u_{25} \\ l_{31} & l_{32} & d_{33} & u_{34} & u_{35} \\ l_{41} & l_{42} & l_{43} & d_{44} & u_{45} \\ l_{51} & l_{52} & l_{53} & l_{54} & d_{55} \end{bmatrix} \qquad u_{i,j} = l_{j,i}^*$$

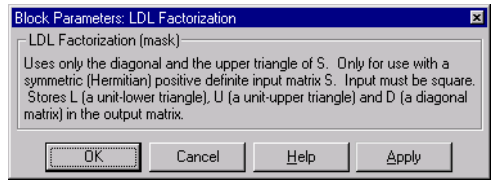
Example:



$$L = \begin{bmatrix} 1 & 0 & 0 \\ -0.11 & 1 & 0 \\ 0.22 & -0.61 & 1 \end{bmatrix} \quad D = \begin{bmatrix} 9.00 & 0 & 0 \\ 0 & 7.89 & 0 \\ 0 & 0 & 3.67 \end{bmatrix} \quad L^H = \begin{bmatrix} 1 & -0.11 & 0.22 \\ 0 & 1 & -0.61 \\ 0 & 0 & 1 \end{bmatrix}$$

LDL factorization requires half the computation of Gaussian elimination (LU decomposition), and is always stable. It is more efficient that Cholesky factorization because it avoids computing the square roots of the diagonal elements.

Dialog Box



References

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

See Also

Cholesky Factorization
LDL Solver
LU Factorization
QR Factorization

LDL Solver

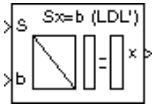
Purpose

Solve the equation S*x*=b for Hermitian positive definite matrix S.

Library

Linear Algebra, in Math Functions

Description



The LDL Solver block solves the linear system S*x*=b by applying LDL factorization to matrix S (top input), which must be square and Hermitian positive definite. The bottom input is the right-hand-side of the equation, b. The output is the unique solution of the equations, *x*.

LDL Factorization uniquely factors the Hermitian positive definite input matrix S as

$$S = LDL^H$$

where L is a lower triangular square matrix with unity diagonal elements, D is a diagonal matrix, and L^H denotes the Hermitian transpose of L.

The equation

$$LDL^H x = b$$

is solved for *x* by the following steps:

1 Substitute

$$y = DL^H x$$

2 Substitute

$$z = L^H x$$

3 Solve one diagonal and two triangular systems:

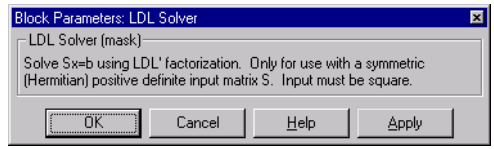
$$Ly = b$$

$$Dz = y$$

$$L^H x = z$$

The block may generate `NaN` or `Inf` for underdetermined or inconsistent (overdetermined) systems.

Dialog Box



See Also

Backward Substitution
Cholesky Solver
LDL Factorization
Levinson Solver
LU Solver
QR Solver

Least Squares FIR Filter Design

Purpose

Design and implement a least-squares FIR filter.

Library

Filter Designs, in Filtering

Description

The Least Squares FIR Filter Design block designs an FIR filter and applies it to the input using the Direct-Form II Transpose Filter block in the Filter Realizations library. The filter design uses the `firls` function in the Signal Processing Toolbox to minimize the integral of the squared error between the desired frequency response and the actual frequency response.

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 rads/sec, 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 sampling frequency (the Nyquist 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**.

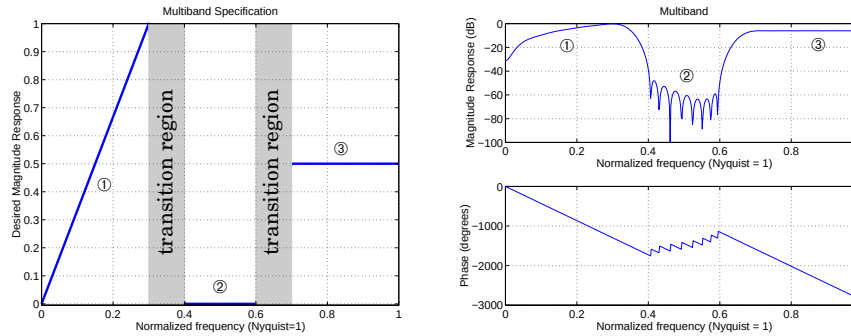
Each odd-indexed frequency-amplitude pair defines the left endpoint of a line segment representing the desired magnitude response in that frequency band. The corresponding even-indexed frequency-amplitude pair defines the right endpoint. Between the frequency bands specified by these end-points, there 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.

Band edge frequency = [0 0.3 0.4 0.6 0.7 1]

Gains = [0 1 0 0 0.5 0.5]

Band: ① ② ③



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 the "Working with Filter Designs" section of Chapter 3. For more on the FIR filter algorithm, see the description of the `firls` function in the *Signal Processing Toolbox User's Guide*.

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector

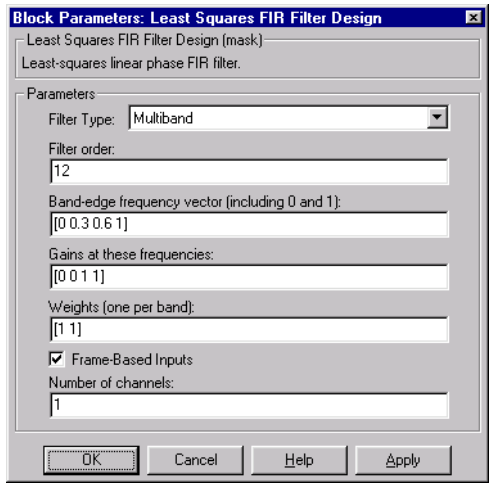
elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel over time.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block filters each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

In both sample-based and frame-based operation, the output is the same size as the input.

Dialog Box



Filter type

The filter type.

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 the Nyquist frequency. This vector must have even length.

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**.

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.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (frames) in the input matrix.

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 the Nyquist 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]`

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.

See Also

Digital FIR Filter Design
Remez FIR Filter Design
Yule-Walker IIR Filter Design
`firls` (Signal Processing Toolbox)

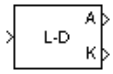
Purpose

Solve a linear system of equations using Levinson-Durbin recursion.

Library

Linear Algebra, in Math Functions

Description



The Levinson Solver block solves the nth-order system of linear equations

$$Ra = b$$

for the particular case where R is a symmetric, 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 algorithm requires $O(n^2)$ flops, and is thus much more efficient for large n than standard Gaussian elimination, which requires $O(n^3)$ flops. The input to the block is the vector $r = [r(1)\ r(2)\ \ldots\ r(n+1)]$, whose elements appear in the matrix R above.

The **Output(s)** parameter allows you to select between two representations of the solution:

- **A** – The output, $a = [1\ a(1)\ a(2)\ \ldots\ a(n+1)]$, is the solution to the Levinson-Durbin equation. The elements of this vector can also be viewed as the coefficients of an nth-order autoregressive (AR) process (see below).
- **K** – The output, κ, contains a vector of reflection coefficients, which are useful for realizing a lattice representation of the AR process.
- **A and K** – The block outputs both representation.

When the **Special-case handling of zero-input** check box is selected (default), an input vector whose $r(1)$ element is zero generates a zero-valued output. When the check box is *not* selected, an input vector with $r(1)=0$ generates `NaN`s in the output. In general, an input vector 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 (or all-pole IIR filter) with assumed 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 equation of the type shown above, which is most efficiently solved by Levinson-Durbin recursion. In this case, the input vector r represents the autocorrelation sequence, with $r(1)$ being the zero-lag value. The output vector a 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:

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

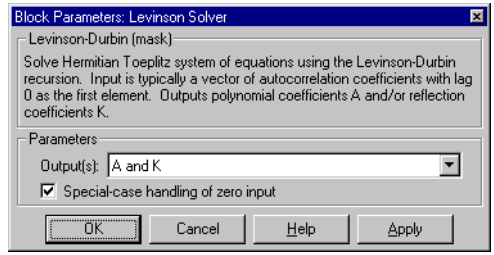
The output vector κ contains the corresponding reflection coefficients, $\kappa(1)$ to $\kappa(n+1)$, for the lattice realization of this IIR filter. The Yule-Walker AR Estimator block implements this autocorrelation-based method for AR model estimation, while the Yule-Walker Method block extends the 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 vector r represents the signal's autocorrelation sequence, with $r(1)$ being the zero-lag value, and output vector a contains the coefficients of the predictive MA process (in descending powers of z):

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

Again, the output vector κ contains the corresponding reflection coefficients, $\kappa(1)$ to $\kappa(n+1)$, for the lattice realization of this FIR filter. The LPC block in the Signal Operations library implements this autocorrelation-based prediction method.

Dialog Box



Output(s)

The representation to output, solution to Ra=b (model coefficients) or reflection coefficients.

Special-case handling of zero input

If selected, output a zero-vector for inputs having $r(1)=0$. If not selected, output `NaN`.

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.

See Also

Cholesky Solver
LDL Solver
LPC
LU Solver
QR Solver
Yule-Walker AR Estimator
Yule-Walker Method
`levinson` (Signal Processing Toolbox)

LMS Adaptive Filter

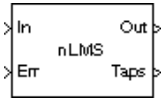
Purpose

Compute filter estimates for an input using the normalized LMS adaptive filter algorithm.

Library

Adaptive Filters, in Filtering

Description



The LMS Adaptive Filter block implements an adaptive FIR filter using the stochastic gradient algorithm known as the normalized Least Mean-Square (LMS) algorithm:

$$y(n) = \hat{w}^T(n-1)u(n)$$
$$e(n) = d(n) - y(n)$$
$$\hat{w}(n) = \hat{w}(n-1) + \frac{u(n)}{a + u^T(n)u(n)}\mu e(n)$$

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 unit-less adaptation constant

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, deselect the **Use normalization** check box in the parameter dialog box. The block then computes the filter-tap estimate as

$$\hat{w}(n) = \hat{w}(n-1) + u(n)\mu e(n)$$

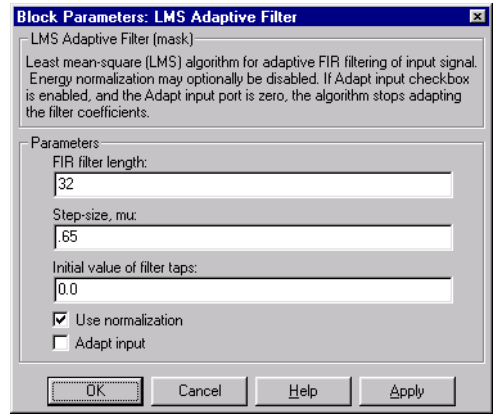
The block icon has port labels corresponding to the inputs and outputs of the LMS algorithm.

Block Ports	Corresponding Variables
`In`	u, the scalar input, which is internally buffered into the vector $u(n)$ used by the algorithm
`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 the **Adapt input** check box is selected 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 **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, and specifies how quickly the filter forgets past sample information. Typically, for convergence in the mean square, 0<μ<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.

Dialog Box



FIR filter length

The length of the FIR filter.

Step-size Ⓣ

The step size, usually in the range (0,2).

Initial value of filter taps

The initial FIR filter coefficients.

Use normalization

Select or deselect normalization.

Adapt input

Enables the `Adapt` port.

References

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

See Also

Kalman Adaptive Filter
RLS Adaptive Filter

Purpose

Determine the coefficients of a one-step forward linear predictor.

Library

Signal Operations, in General DSP

Description

The LPC block determines the coefficients of a *one-step forward linear predictor* 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.

At the output port, the LPC block provides the coefficients of an nth-order moving average (MA) linear process that predicts the next value in the time-series u, contained in the length-L input frame.

$$\hat{u}(L+1) = -u(L) - a(2)u(L-1) - a(3)u(L-2) - \cdots - a(n+1)u(L-n)$$

where $\hat{u}(L+1)$ is the estimate of the next sequence value, and n is the **Prediction order**. The filter coefficients above are output in vector form, $a = [1\ a(2)\ \ldots\ a(n+1)]$. If a value of `-1` is specified for the **Prediction order** parameter, the block uses `length(u)-1` for n.

A matrix input, `u`, is treated as a vector frame, `u(:)`.

Algorithm

The LPC block computes the least-squares solution to

$$\min_{a \in \Re^{n+1}} ||Xa - b||$$

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

$$X = \begin{bmatrix} x(1) & 0 & \cdots & 0 \\ x(2) & x(1) & \ddots & \vdots \\ \vdots & x(2) & \ddots & 0 \\ x(L) & \vdots & \ddots & x(1) \\ 0 & x(L) & \ddots & x(2) \\ \vdots & \ddots & \ddots & \vdots \\ 0 & \cdots & 0 & x(L) \end{bmatrix}, \quad a = \begin{bmatrix} 1 \\ a(2) \\ \vdots \\ a(n+1) \end{bmatrix}, \quad b = \begin{bmatrix} 1 \\ 0 \\ \vdots \\ 0 \end{bmatrix}$$

Solving the least-squares problem via the normal equations

$$X^*Xa = X^*b$$

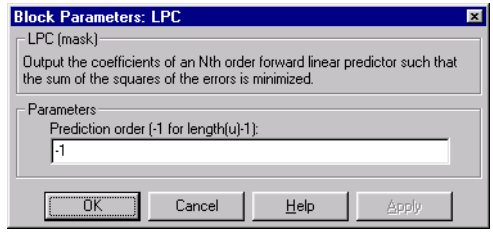
leads to the system of equations

$$\begin{bmatrix} r(1) & r(2)^* & \cdots & r(n)^* \\ r(2) & r(1) & \ddots & \vdots \\ \vdots & \ddots & \ddots & r(2)^* \\ r(n) & \cdots & r(2) & r(1) \end{bmatrix} \begin{bmatrix} a(2) \\ a(3) \\ \vdots \\ a(n+1) \end{bmatrix} = \begin{bmatrix} -r(2) \\ -r(3) \\ \vdots \\ -r(n+1) \end{bmatrix}$$

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



Prediction order

The prediction order, n.

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.

See Also

Levinson Solver
Yule-Walker Method
`lpc` (Signal Processing Toolbox)

LU Factorization

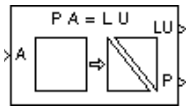
Purpose

Factor a square matrix into lower and upper triangular components.

Library

Linear Algebra, in Math Functions

Description



The LU Factorization block factors a row permutation of the square input matrix A as

$$A_p = LU$$

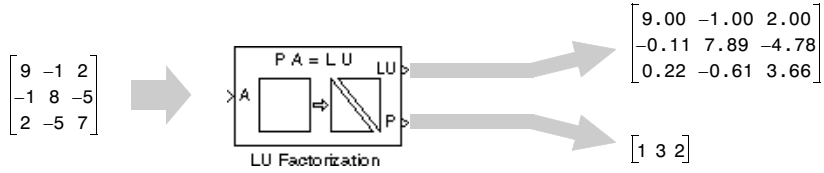
where L is a lower-triangular square matrix with unity diagonal elements, and U is an upper-triangular square matrix. The row-pivoted matrix A_p contains the rows of A permuted as indicated by the permutation index vector P.

```
Ap = A(P,:)          % equivalent MATLAB code
```

Example:

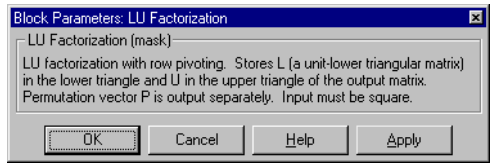
$$A = \begin{bmatrix} 9 & -1 & 2 \\ -1 & 8 & -5 \\ 2 & -5 & 7 \end{bmatrix} \qquad P = \begin{bmatrix} 1 & 3 & 2 \end{bmatrix} \qquad A_p = \begin{bmatrix} 9 & -1 & 2 \\ 2 & -5 & 7 \\ -1 & 8 & -5 \end{bmatrix}$$

The block's top output (LU) is a composite matrix whose lower sub-triangle forms L and whose upper triangle is U.



$$L = \begin{bmatrix} 1 & 0 & 0 \\ -0.11 & 1 & 0 \\ 0.22 & -0.61 & 1 \end{bmatrix} \qquad U = \begin{bmatrix} 9.00 & -1.00 & 2.00 \\ 0 & 7.89 & -4.78 \\ 0 & 0 & 3.66 \end{bmatrix}$$

Dialog Box



References

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

See Also

Backward Substitution
Cholesky Solver
LDL Factorization
LU Solver
Permute Matrix
QR Factorization
`lu` (MATLAB)

LU Solver

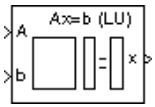
Purpose

Solve the equation A*x*=b for square matrix A.

Library

Linear Algebra, in Math Functions

Description



The LU Solver block solves the linear system A*x*=b by applying LU factorization to matrix A (top input), which must be square. The bottom input is the right-hand-side of the equation, b. The output is the unique solution of the equations, *x*.

LU factorization 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:

$$Ly = b_p$$

$$Ux = y$$

The block may generate `NaN` or `Inf` for underdetermined or inconsistent (overdetermined) systems.

Dialog Box

See Also

Backward Substitution
Cholesky Solver
LDL Solver
Levinson Solver
LU Factorization
QR Solver

Magnitude FFT

Purpose

Compute a nonparametric estimate of the spectrum using the periodogram method.

Library

Power Spectrum Estimation, in Estimation

Description

The Magnitude FFT block computes a nonparametric estimate of the spectrum using the periodogram method. For input u,

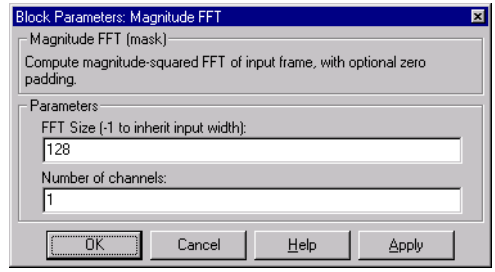
```
y = abs(fft(u,nfft)).^2    % equivalent MATLAB code
```

where N_{fft} is specified as a power of 2 by the **FFT size** parameter. A value of `-1` for **FFT size** instructs the block to use the input frame size as the FFT size. Otherwise, the block zero pads or truncates the input to N_{fft}.

The input is an M-by-N frame matrix, with each of the N frames containing M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals, N, in the matrix.

The block computes a separate periodogram for each of the N independent channels in the input, generating an N_{fft}-by-N matrix output. 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.

Dialog Box



FFT size

The number of data points on which to perform the FFT, N_{fft}. If N_{fft} exceeds the input frame size, the frame is zero-padded as needed.

Number of channels

The number of channels (columns) in the input matrix, N.

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.

See Also

Burg Method
FFT Frame Scope
Short-Time FFT
Yule-Walker Method
`pwelch` (Signal Processing Toolbox)

Matrix 1-Norm

Purpose

Compute the 1-norm of a square matrix.

Library

Linear Algebra, in Math Functions

Description

The Matrix 1-Norm block computes the 1-norm, or maximum column-sum, of an M-by-N input matrix, A:

$$\|A\|_1 = \max_{1 \le j \le N} \sum_{i=1}^{M} |a_{ij}|$$

or

```
y = max(sum(abs(A)))   % equivalent MATLAB code
```

$$\begin{bmatrix} a_{11} & a_{12} & a_{13} \\ a_{21} & a_{22} & a_{23} \\ a_{31} & a_{32} & a_{33} \end{bmatrix}$$

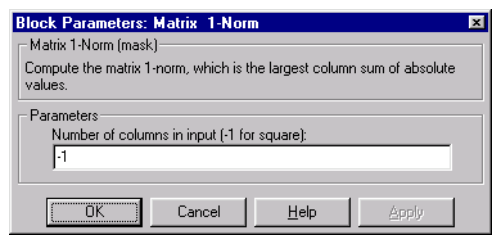
$$\|A\|_1 = \max(A_1, A_2, A_3)$$

$$|a_{13}| + |a_{23}| + |a_{33}| = A_3$$

$$|a_{12}| + |a_{22}| + |a_{32}| = A_2$$

$$|a_{11}| + |a_{21}| + |a_{31}| = A_1$$

Dialog Box



Number of columns in input

The number of columns in the input matrix. A value of `-1` indicates that the input is a square matrix.

References

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

See Also

Normalization
Reciprocal Condition
`norm` (MATLAB)

Matrix Constant

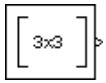
Purpose

Generate a constant matrix.

Library

Matrix Functions, in Math Functions; DSP Sources

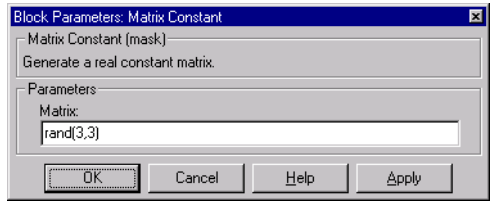
Description



The Matrix Constant block outputs a constant matrix into the system. If a scalar or vector is specified (as 1-by-1, M-by-1, or 1-by-N matrix), the output is the indicated scalar or vector constant.

Because the output of the Matrix Constant block is continuous, a Zero-Order Hold block should be inserted between this block and blocks that require discrete inputs.

Dialog Box



Matrix (T)

The matrix to output. The values of the matrix elements can be changed while the simulation is running, but the dimensions of the matrix must remain the same.

See Also

Constant (Simulink)
Constant Diagonal Matrix
Matrix From Workspace

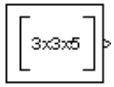
Purpose

Read a time-sequence of matrices from the workspace.

Library

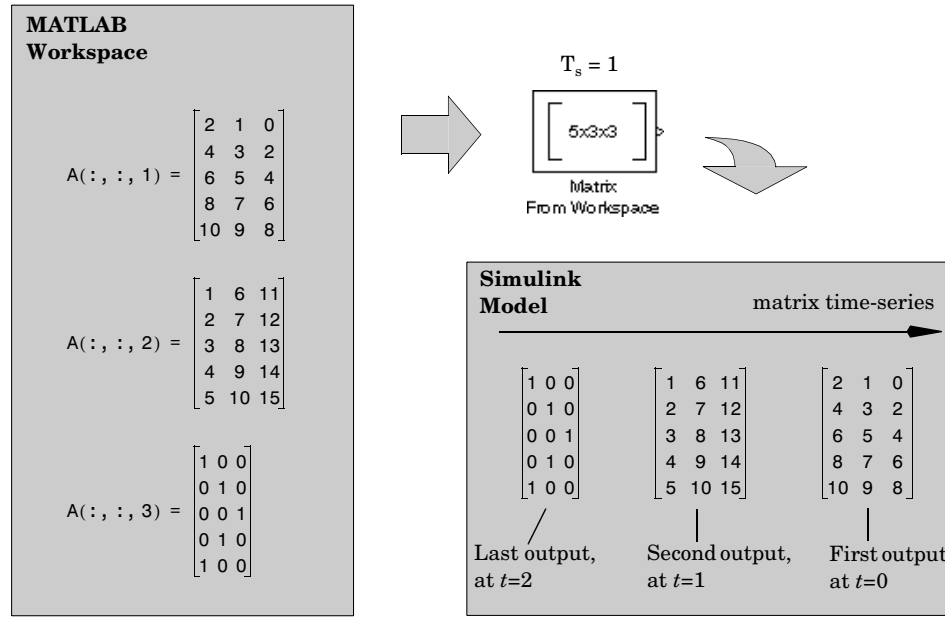
Matrix Functions, in Math Functions; DSP Sources

Description



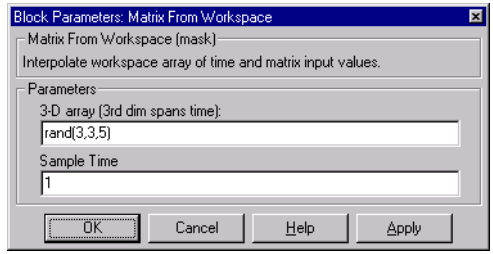
The Matrix From Workspace block references a three-dimensional array in the workspace, `A`, to generate a matrix output to the system. At each sample time the block outputs a *page* (a two-dimensional slice) of the three-dimensional array, beginning with the first, `A(:,:,1)`. The block continues to output pages of the array until it outputs the last page, `A(:,:,end)`. The output sample period, T_s, is specified by the **Sample time** parameter.

As with Simulink's From Workspace block, the value of the output between specified sample times is linearly interpolated from the nearest two values. The value of the output preceding the first specified sample time and following the last specified sample time is extrapolated from the first or last two points.



If `size(A,1)` or `size(A,2)` equals 1, the block's output is a vector. If both equal 1, the output is a scalar.

Dialog Box



3-D array

The workspace array or MATLAB expression that defines the matrix time-series to output.

Sample time

The sample period at which the array pages are output, T_s.

See Also

Matrix Constant
Matrix To Workspace
Signal From Workspace

Purpose

Multiply two input matrices.

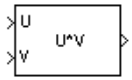
Library

Matrix Functions, in Math Functions; DSP Sources

Description

The Matrix Multiplication block multiplies two matrix inputs:

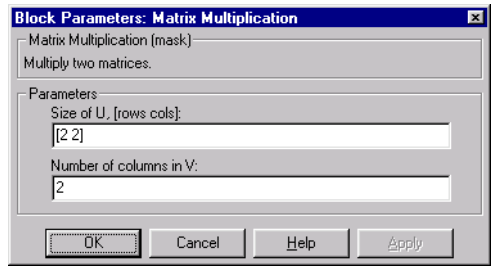
```
y = u * v      % equivalent MATLAB code
```



The size of the output is `[size(u,1),size(v,2)]`. The input matrices must have sizes compatible for matrix multiplication; that is, `size(u,2)` must equal `size(v,1)`.

Note If you expect to generate code for the Matrix Multiplication block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Size of U

The size of the premultiplying matrix, `u`, in the format `[rows columns]`.

Number of columns in V

The number of columns in the postmultiplying matrix, `v`.

See Also

Dot Product (Simulink)
Matrix Product
Matrix Scaling
Product (Simulink)

Matrix Product

Purpose

Multiply the elements of a matrix along rows or columns.

Library

Matrix Functions, in Math Functions

Description

The Matrix Product block multiplies the elements of an M-by-N input matrix u along either the rows or columns.

When the **Multiply along** parameter is set to **Rows**, the block multiplies across the elements of each row and outputs the resulting M-by-1 vector.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix} \Rightarrow \begin{bmatrix} u_{11} \times u_{12} \times u_{13} \\ u_{21} \times u_{22} \times u_{23} \\ u_{31} \times u_{32} \times u_{33} \end{bmatrix}$$

This is equivalent to

```
y = prod(u,2)      % equivalent MATLAB code
```

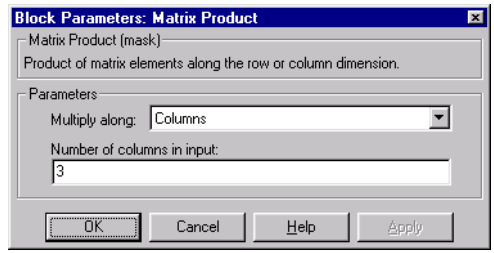
When the **Multiply along** parameter is set to **Columns**, the block multiplies down the elements of each column and outputs the resulting 1-by-N vector.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix} \Rightarrow \begin{bmatrix} u_{11} & u_{12} & u_{13} \\ \times & \times & \times \\ u_{21} & u_{22} & u_{23} \\ \times & \times & \times \\ u_{31} & u_{32} & u_{33} \end{bmatrix}$$

This is equivalent to

```
y = prod(u)        % equivalent MATLAB code
```

Dialog Box



Multiply along

The dimension of the matrix along which to multiply, row or column.

Number of columns in input

The number of columns in the input matrix.

See Also

Matrix Multiplication
Matrix Sum
`prod` (MATLAB)

Matrix Scaling

Purpose

Scale the rows or columns of a matrix by a specified vector.

Library

Matrix Functions, in Math Functions

Description

The Matrix Scaling block scales the rows or columns of the input matrix A by the input vector D.

When the **Mode** parameter is set to **Scale rows**, the block multiplies each element of vector D across the corresponding *row* of matrix A.

$$\begin{bmatrix} d_1 \\ d_2 \\ d_3 \end{bmatrix} \times \begin{bmatrix} a_{11} & a_{12} & a_{13} \\ a_{21} & a_{22} & a_{23} \\ a_{31} & a_{32} & a_{33} \end{bmatrix} \Rightarrow \begin{bmatrix} d_1 a_{11} & d_1 a_{12} & d_1 a_{13} \\ d_2 a_{21} & d_2 a_{22} & d_2 a_{23} \\ d_3 a_{31} & d_3 a_{32} & d_3 a_{33} \end{bmatrix}$$

This is equivalent to premultiplying A by a diagonal matrix with diagonal D.

```
y = diag(D)*A      % equivalent MATLAB code
```

When the **Mode** parameter is set to **Scale columns**, the block multiplies each element of vector D across the corresponding *column* of matrix A.

$$\begin{bmatrix} d_1 & d_2 & d_3 \end{bmatrix} \times \begin{bmatrix} a_{11} & a_{12} & a_{13} \\ a_{21} & a_{22} & a_{23} \\ a_{31} & a_{32} & a_{33} \end{bmatrix} \Rightarrow \begin{bmatrix} d_1 a_{11} & d_2 a_{12} & d_3 a_{13} \\ d_1 a_{21} & d_2 a_{22} & d_3 a_{23} \\ d_1 a_{31} & d_2 a_{32} & d_3 a_{33} \end{bmatrix}$$

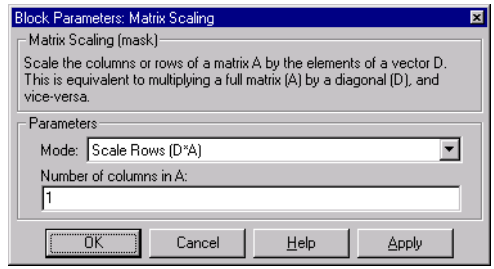
This is equivalent to postmultiplying A by a diagonal matrix with diagonal D.

```
y = A*diag(D)      % equivalent MATLAB code
```

The length of vector D must be consistent with the intended matrix operation (i.e., D must have the same length as the dimension of A being scaled). The output is the same size as the input matrix A.

Note If you expect to generate code for the Matrix Scaling block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Mode

The mode of operation, row scaling or column scaling.

Number of columns in A

The number of columns in the input matrix.

See Also

Matrix Multiplication

Matrix Sum

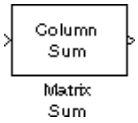
Purpose

Sum the elements of a matrix along rows or columns.

Library

Matrix Functions, in Math Functions

Description



The Matrix Sum block sums the elements of an M-by-N input matrix u along either the rows or columns.

When the **Sum along** parameter is set to **Rows**, the block sums across the elements of each row and outputs the resulting M-by-1 vector.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix} \Rightarrow \begin{bmatrix} u_{11}+u_{12}+u_{13} \\ u_{21}+u_{22}+u_{23} \\ u_{31}+u_{32}+u_{33} \end{bmatrix}$$

This is equivalent to

```
y = sum(u,2)        % equivalent MATLAB code
```

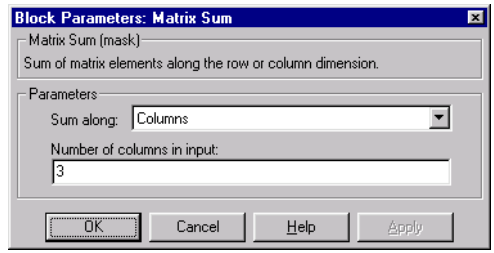
When the **Sum along** parameter is set to **Columns**, the block sums down the elements of each column and outputs the resulting 1-by-N vector.

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \\ u_{31} & u_{32} & u_{33} \end{bmatrix} \Rightarrow \begin{bmatrix} u_{11}+u_{21}+u_{31} & u_{12}+u_{22}+u_{32} & u_{13}+u_{23}+u_{33} \end{bmatrix}$$

This is equivalent to

```
y = sum(u)          % equivalent MATLAB code
```

Dialog Box



Sum along

The dimension of the matrix to sum along, row or column.

Number of columns in input

The number of columns in the input matrix.

See Also

Matrix Product
Matrix Multiplication
`sum` (MATLAB)

Matrix To Workspace

Purpose

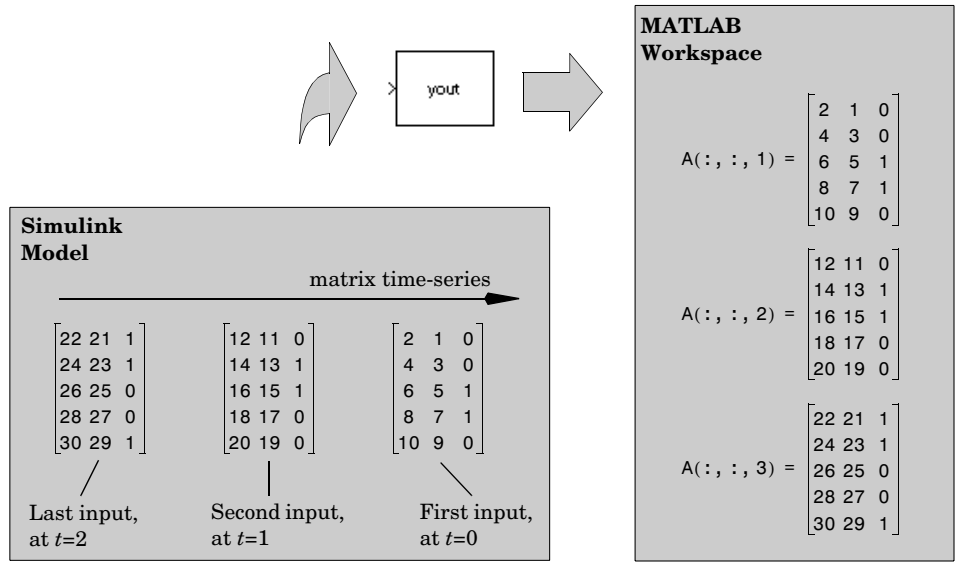
Send a time-sequence of matrices to the workspace.

Library

Matrix Functions, in Math Functions; DSP Sinks

Description

The Matrix To Workspace block writes a three-dimensional array, `A`, to the workspace, where `A` contains the acquired samples of a matrix input. At the end of the simulation the block writes every Dth input sample (a matrix) to a *page* (a two-dimensional slice) of the specified three-dimensional array, where D is specified by the block's **Decimation factor** parameter. The block writes the first input to the first page of the array, `A(:,:,1)`, and continues adding pages until it writes the last input matrix to the last array page, `A(:,:,end)`.

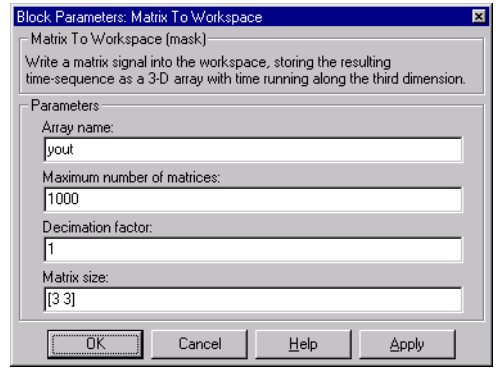


To save a record of the actual sample times corresponding to each page of the three-dimensional array, check the **Time** box in the **Save to workspace** panel of the **Simulation Parameters** dialog box. You can access these parameters by selecting **Parameters** from the **Simulation** menu, and clicking on the **Workspace I/O** tab.

Use the Signal To Workspace block to write frame-matrix inputs to the workspace.

Note The Matrix To Workspace block does not support real-time data logging when used with the Real-Time Workshop. You should not use this block if you expect to generate real-time code from your model.

Dialog Box



Array name

The name of the three-dimensional array to create in the workspace.

Maximum number of matrices

The maximum number of pages, P, in the array created in the workspace. If the block samples more than P times, it stores only the last P matrices.

Decimation factor

The factor by which to reduce the input sample rate.

Matrix size

The dimensions of the input matrix.

See Also

Matrix From Workspace
Signal To Workspace
Triggered Matrix To Workspace

Matrix Viewer

Purpose

Display a matrix as a color image.

Library

DSP Sinks

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 number of input columns must be specified in the **Number of columns** parameter of the **Image properties** panel (select the **Image properties** check box to expose the panel). The display is updated as each new input is received.

Image Properties

Click on the **Image properties** check box to expose the image property parameters, which control the colormap and display.

The mapping of matrix element values to colors is specified by the **Colormap matrix**, **Minimum input**, and **Maximum input** 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 `ColorSpec` in the online *MATLAB Function Reference* for complete information about defining RGB triples.

MATLAB provides a number of functions for generating predefined colormaps. Examples are `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, if 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 is used to represent the value specified by the **Minimum input** parameter, and the color in the last row is used to represent the value specified by the **Maximum input** parameter. Values between the minimum and maximum are quantized and mapped to the intermediate rows of the colormap matrix.

The documentation for MATLAB's `colormap` function provides complete information about specifying colormap matrices, and includes a complete list of the available colormap functions.

Axis Properties

Click on the **Axis properties** check box to expose 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 **Upper left corner** is specified, 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 **Lower left corner** is specified, 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. Menus and axis titles are not displayed. This option can also be selected from the right-click pop-up menu in the figure window.

When **Axis zoom** is deselected, the axis labels and titles are displayed in a gray border surrounding the image axes, and the window's menus (including **Axes**) and toolbar are visible. The Plot Editor tools allow you to annotate and customize the image display. Select **Help Plot Editor** from the figure's **Help** menu for more information about using these tools. For information on printing or saving the image, or on the other options found in the generic figure menus (**File**, **Edit**, **Window**, **Help**), see *Using MATLAB Graphics*.

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 0 to N (number of columns), and the *y*-axis is numbered from 0 to M (number of rows).

In addition to the standard MATLAB figure window menus (**File**, **Edit**, **Window**, **Help**), the Matrix Viewer window has an **Axes** menu containing the following items:

- **Refresh** erases all data on the scope display, except for the most recent image.
- **Autoscale** recomputes the **Minimum input** and **Maximum input** parameter values to best 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** and **Maximum input** parameters, where you can make further adjustments to them manually.
- **Axis zoom**, when selected, causes the image to completely fill the containing figure window. Menus and axis titles are not displayed. When **Axis zoom** is deselected, the axis labels and titles are displayed in a gray border surrounding the scope axes, and the window's menus (including **Axes**) and toolbar are visible. This option can also be set in the **Axis properties** panel of the parameter dialog box.
- **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** field to reflect the figure window's current position and size. To make the scope window open at a particular location on the screen when the simulation runs, simply drag the window to the desired location, resize it as needed, and select **Save Position**. Note that the parameter dialog box must be closed when you select **Save Position** in order for the **Figure position** parameter to be updated.

Many of these options can also be accessed by right-clicking with the mouse anywhere on the displayed image. The right-click menu is very helpful when the scope is in zoomed mode and the **Axes** menu is not visible.

Dialog Box

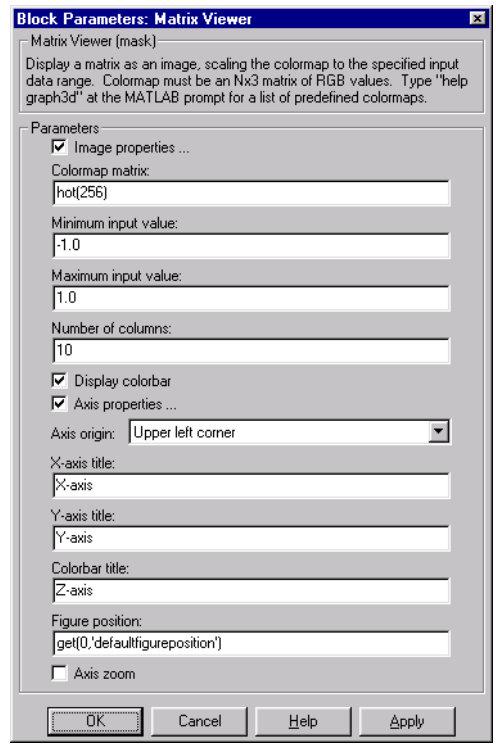


Image properties (T)

Select to expose the image property parameters.

Colormap matrix (T)

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 `colormap` function for a list of colormap-generating functions.

Minimum input value (T)

The input value to be mapped to the color defined in the first row of the colormap matrix. Select **Autoscale** from the right-click pop-up menu to set this parameter to the minimum value observed in a series of 10 consecutive matrix inputs.

Maximum input value Ⓣ

The input value to be mapped to the color defined in the last row of the colormap matrix. Select **Autoscale** from the right-click pop-up menu to set this parameter to the maximum value observed in a series of 10 consecutive matrix inputs.

Number of columns

The number of columns in the matrix input.

Display colorbar Ⓣ

Select to display a bar with the selected colormap to the right of the image axes.

Axis properties Ⓣ

Select to expose the axis property parameters.

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.

X-axis title Ⓣ

The text to be displayed below the *x*-axis.

Y-axis title Ⓣ

The text to be displayed to the left of the *y*-axis.

Colorbar title Ⓣ

The text to be displayed to the right of the color bar, if **Display colorbar** is currently selected.

Figure position Ⓣ

A 4-element vector of the form `[left bottom width height]` specifying the position of the figure window. `(0,0)` is the lower-left corner of the display.

Axis zoom Ⓣ

Resizes the image to fill the figure window.

Examples

See the demo `dspstfft.mdl` for an example of using the Matrix Viewer block to create a moving spectrogram (time-frequency plot) of a speech signal by updating just one column of the input matrix at each sample time.

See Also

Buffered FFT Frame Scope
Frequency Frame Scope
Time Frame Scope
User-Defined Frame Scope
`ColorSpec` (MATLAB)
`colormap` (MATLAB)

Maximum

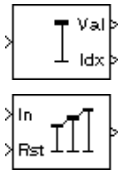
Purpose

Find the maximum value of an input or sequence of inputs.

Library

Statistics, in Math Functions

Description



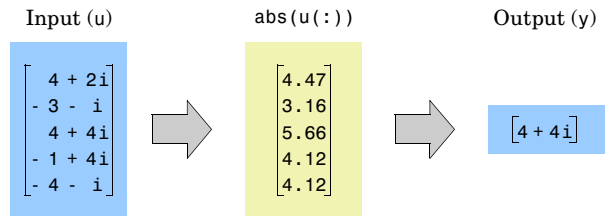
The Maximum block identifies the value and position of the largest element in the input, or tracks the maximum value in a sequence of inputs over a period of time. The **Mode** parameter specifies the block's mode of operation and can be set to **Value**, **Index**, **Value and Index**, or **Running**. These settings are described below.

Value

When **Mode** is set to **Value**, the block computes the maximum value of the input vector.

```
[y,i] = max(u(:))      % equivalent MATLAB code
```

The block output, *y*, is the maximum value of the input at each sample time. For complex inputs the block uses the magnitude of the input, `abs(u(:))`, to identify the maximum. The output is the corresponding complex value from the input.



Index

When **Mode** is set to **Index**, the block performs the computation shown above, and outputs the index, *i*, corresponding to the position of the maximum value in the input vector. The index is an integer in the range `[1 length(u(:))]`.

If there are duplicates of the maximum value in the input, the index corresponds to the first occurrence. For example, if the vector input is `[3 2 1 2 3]`, the index of the maximum value is `1`, not `5`.

Value and Index

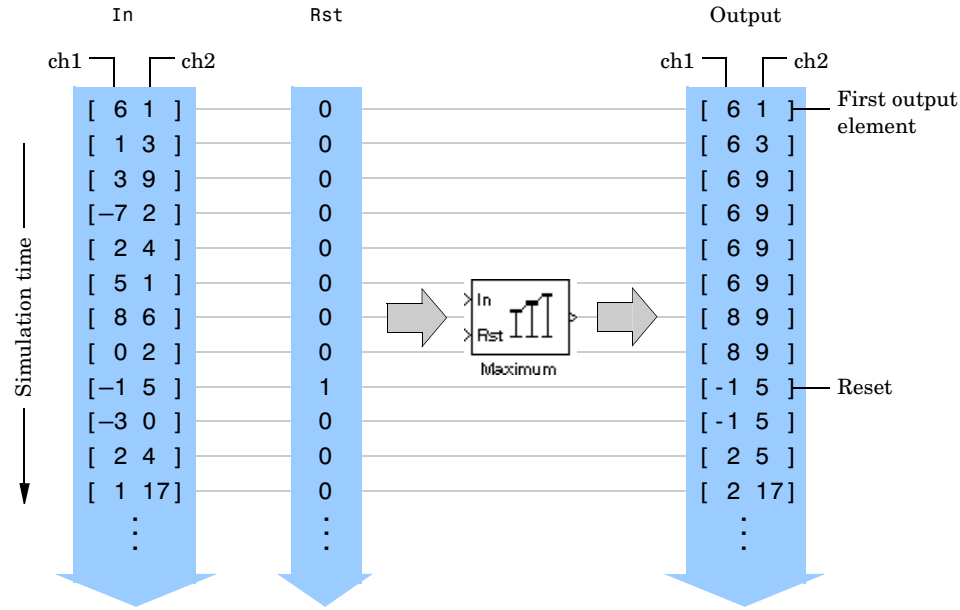
When **Mode** is set to **Value and Index**, the block outputs both the value, y, and the index, i.

In all three of the above modes, a matrix input, `u`, is treated as a vector, `u(:)`.

Running

When **Mode** is set to **Running**, the block tracks the maximum value in a sequence of inputs over time. You can choose frame-based or sample-based operation by selecting or deselecting the **Frame-based** check box.

Sample-Based Operation. When the **Frame-based** check box is *not* selected (default), the block assumes that the input at the `In` port is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M*N matrix elements) is treated as an independent channel, and the block tracks the maximum value in each of the channels over time.

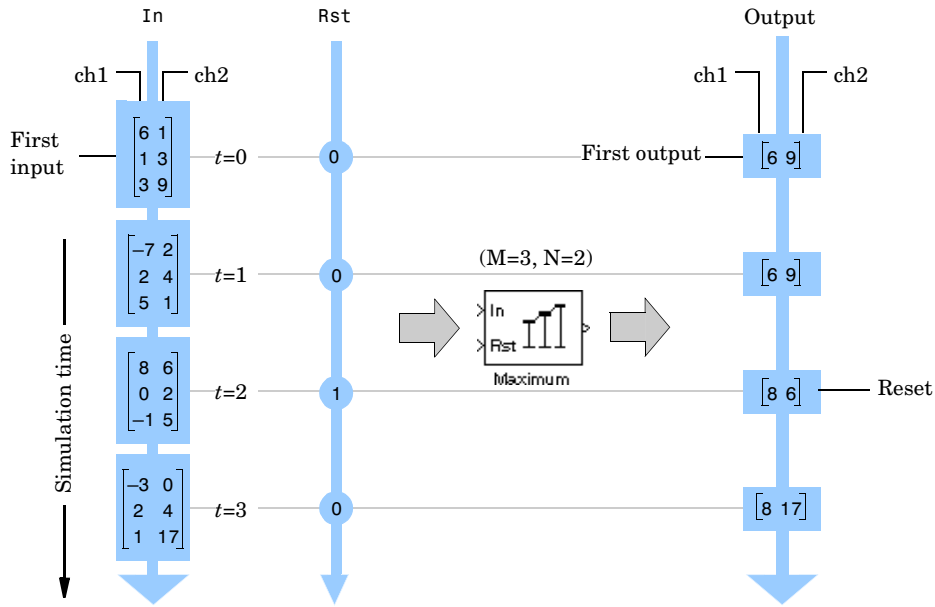


The block resets the running maximum when the scalar input at the optional `Rst` port is nonzero. The output is the same size as the input, and contains the maximum for each input channel since the last reset.

If you do not need to reset the running maximum during the simulation, you can delete the `Rst` port from the block icon by deselecting the **Reset port** check box.

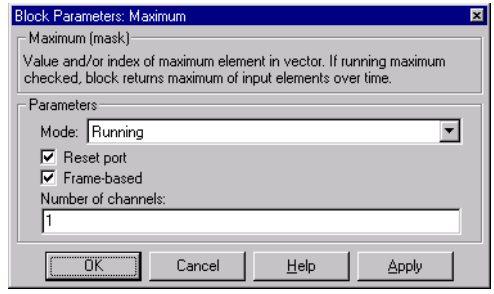
Frame-Based Operation. When the **Frame-based** check box is selected, the block assumes that the input at the `In` port is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals, N, in the matrix.

The block tracks the maximum value in each of the N independent channels over time, and resets the running maximum when the input at the `Rst` port is nonzero. The output is a sample vector of length N which contains the maximum for each input channel since the last reset.



Note If you expect to generate code for the Maximum block's running mode using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Mode

The block's mode of operation: Output the maximum value of each input, the index of the maximum value, both the value and the index, or track the maximum value of the input sequence over time.

Reset port

Enable `Rst` input port.

Frame-based

Selects frame-based operation.

Number of channels

For frame based operation, the number of channels (columns) in the input matrix, N.

See Also

Mean
Minimum
`max` (MATLAB)

Mean

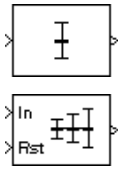
Purpose

Find the mean value of an input or sequence of inputs.

Library

Statistics, in Math Functions

Description

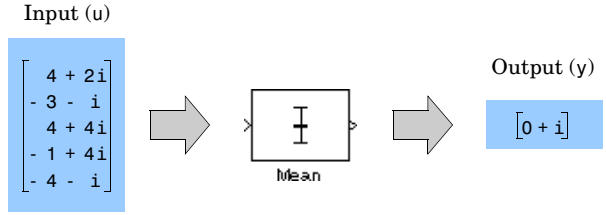


The Mean block computes the mean of the input vector, or tracks the mean of a sequence of inputs over a period of time. The **Running mean** parameter allows you to select between basic operation and running operation, which are both described below.

Basic Operation

When the **Running mean** check box is *not* selected, the block computes the mean value of the input vector at each sample time.

```
y = mean(u(:))          % equivalent MATLAB code
```



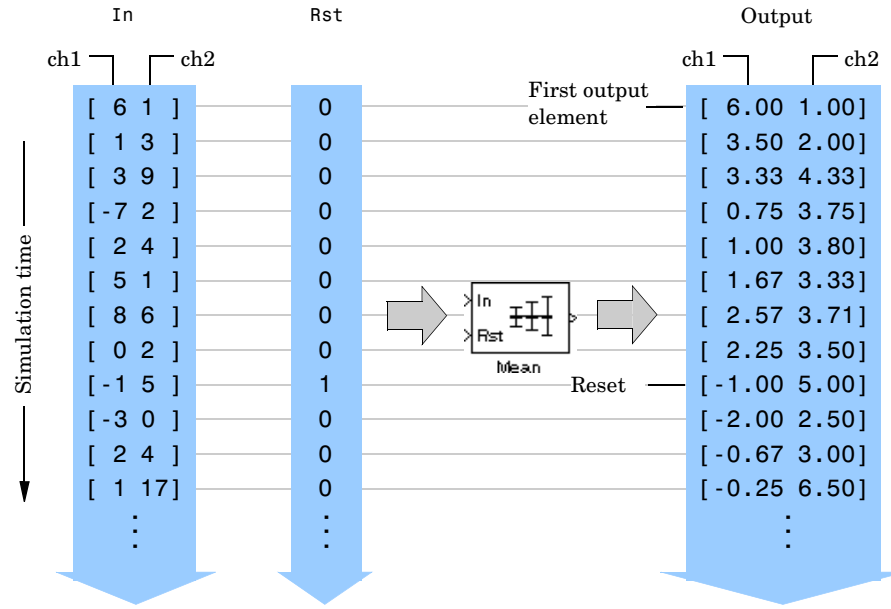
A matrix input, `u`, is treated as a vector, `u(:)`.

Running Operation

When the **Running mean** check box is selected, the block tracks the mean of a sequence of inputs over time. You can choose frame-based or sample-based operation by selecting or deselecting the **Frame-based** check box.

Sample-Based Operation. When the **Frame-based** check box is *not* selected (default), the block assumes that the input at the `In` port is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block tracks the mean of each of the channels over time.

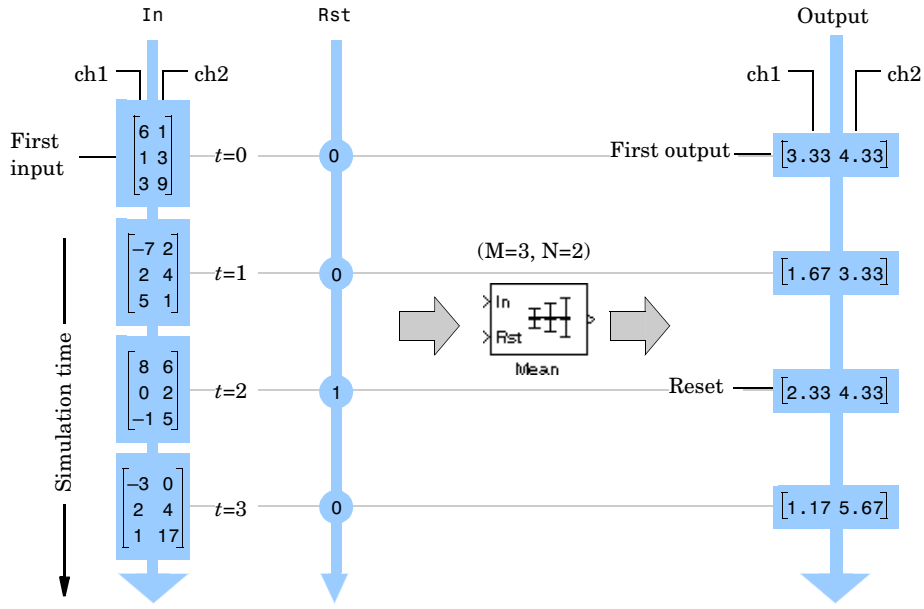
The block resets the running mean when the scalar input at the optional `Rst` port is nonzero. The output is the same size as the input, and contains the mean for each input channel since the last reset.



If you do not need to reset the running mean during the simulation, you can delete the `Rst` port from the block icon by deselecting the **Reset port** check box.

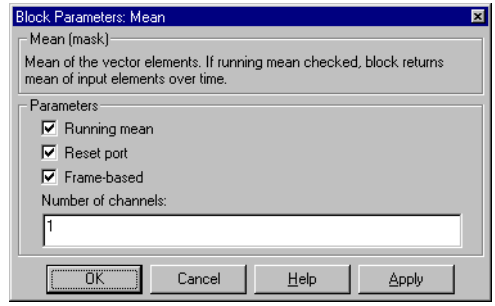
Frame-Based Operation. When the **Frame-based** check box is selected, the block assumes that the input at the `In` port is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals, N, in the matrix.

The block tracks the mean of each of the N independent channels over time, and resets the running mean when the input at the `Rst` port is nonzero. The output is a sample vector of length N which contains the mean for each input channel since the last reset.



Note If you expect to generate code for the Mean block's running mode using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Running mean

Selects running operation.

Reset port

Enable `Rst` input port.

Frame-based

Selects frame-based operation.

Number of channels

For frame based operation, the number of channels (columns) in the input matrix, N.

See Also

Maximum
Minimum
Standard Deviation
`mean` (MATLAB)

Median

Purpose

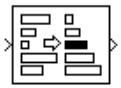
Find the median value of a vector input.

Library

Statistics, in Math Functions

Description

The Median block outputs the median value of the elements in a vector input.



```
y = median(u)  % equivalent MATLAB code for vector input
```

For odd-length inputs, the block sorts the input elements by value, and outputs the central element of the sorted vector,

```
y = s((n+1)/2)
```

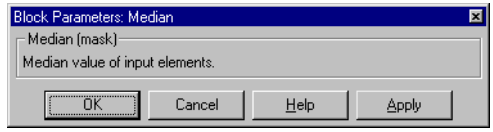
where `s` is the sorted vector and `n=length(u)`. For even-length inputs, the block sorts the input elements by value, and outputs the average of the two central elements in the sorted vector.

```
y = mean([s(n/2) s(n/2+1)])
```

A matrix input, `u`, is treated as a vector, `u(:)`.

Note If you expect to generate code for the Median block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



See Also

Maximum
Mean
Minimum
Standard Deviation
Variance
`median` (MATLAB)

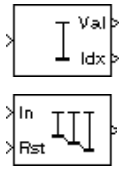
Purpose

Find the minimum value of an input or sequence of inputs.

Library

Statistics, in Math Functions

Description



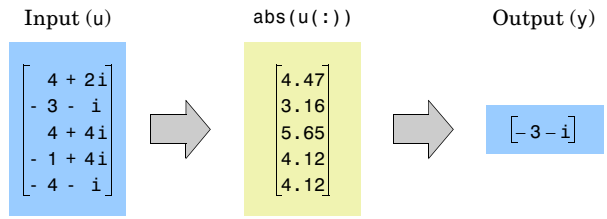
The Minimum block identifies the value and position of the smallest element in the input, or tracks the minimum value in a sequence of inputs over a period of time. The **Mode** parameter specifies the block's mode of operation and can be set to **Value**, **Index**, **Value and Index**, or **Running**. These settings are described below.

Value

When **Mode** is set to **Value**, the block computes the minimum value of the input vector.

```
[y,i] = min(u(:))    % equivalent MATLAB code
```

The block output, *y*, is the minimum value of the input at each sample time. For complex inputs the block uses the magnitude of the input, `abs(u(:))`, to identify the minimum. The output is the corresponding complex value from the input.



Index

When **Mode** is set to **Index**, the block performs the computation shown above, and outputs the index, *i*, corresponding to the position of the minimum value in the input vector. The index is an integer in the range `[1 length(u(:))]`.

If there are duplicates of the minimum value in the input, the index corresponds to the first occurrence. For example, if the vector input is `[1 2 3 2 1]`, the index of the minimum value is `1`, not `5`.

Value and Index

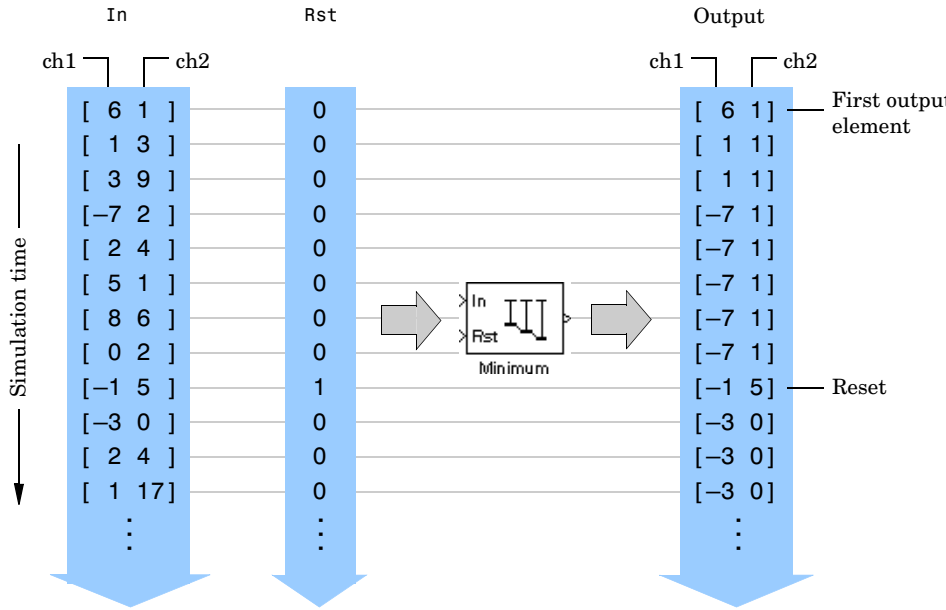
When **Mode** is set to **Value and Index**, the block outputs both the value, *y*, and the index, *i*.

In all three of the above modes, a matrix input, `u`, is treated as a vector, `u(:)`.

Running

When **Mode** is set to **Running**, the block tracks the minimum value in a sequence of inputs over time. You can choose frame-based or sample-based operation by selecting (or deselecting, respectively) the **Frame-based** check box.

Sample-Based Operation. When the **Frame-based** check box is *not* selected (default), the block assumes that the input at the `In` port is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block tracks the minimum value in each of the channels over time.

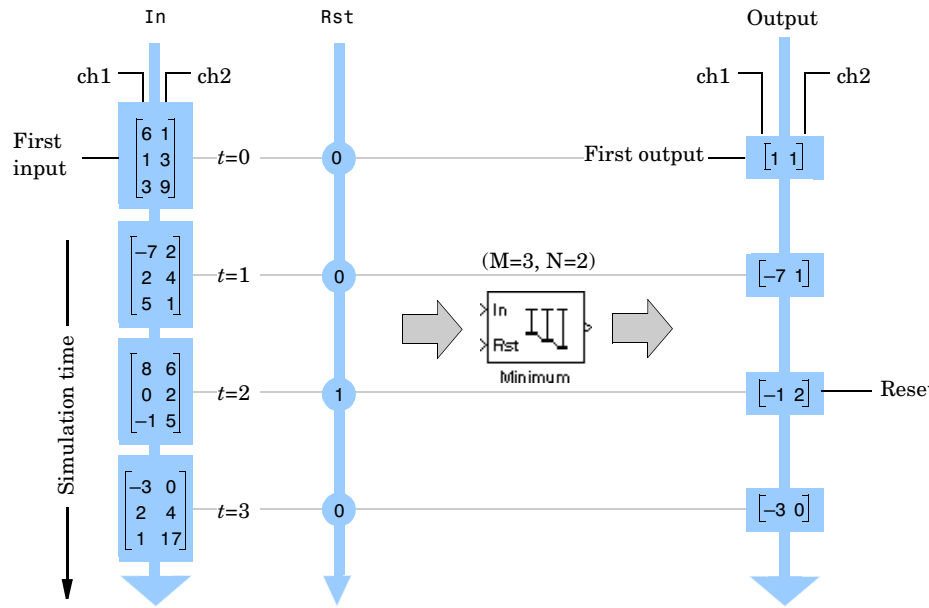


The block resets the running minimum when the scalar input at the optional `Rst` port is nonzero. The block's output is the same size as the input, and contains the minimum for each input channel since the last reset.

If you do not need to reset the running minimum during the simulation, you can delete the `Rst` port from the block icon by deselecting the **Reset input** check box.

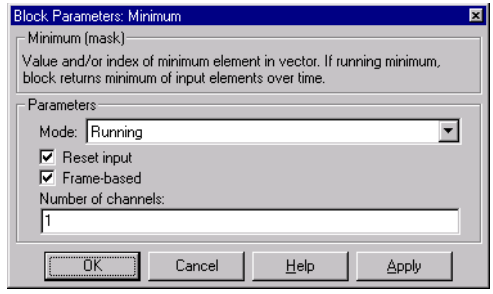
Frame-Based Operation. When the **Frame-based** check box is selected, the block assumes that the input at the `In` port is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals, N, in the matrix.

The block tracks the minimum value in each of the N independent channels over time, and resets the running minimum when the input at the `Rst` port is nonzero. The output is a sample vector of length N which contains the minimum for each input channel since the last reset.



Note If you expect to generate code for the Minimum block's running mode using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Mode

The block's mode of operation: Output the minimum value of each input, the index of the minimum value, both the value and the index, or track the minimum value of the input sequence over time.

Reset input

Enable `Rst` input port.

Frame-based

Selects frame-based operation.

Number of channels

For frame based operation, the number of channels (columns) in the input matrix, N.

See Also

Maximum
Mean
Histogram
`min` (MATLAB)

Purpose

Compute an estimate of AR model parameters using the modified covariance method.

Library

Parametric Estimation, in Estimation

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.

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

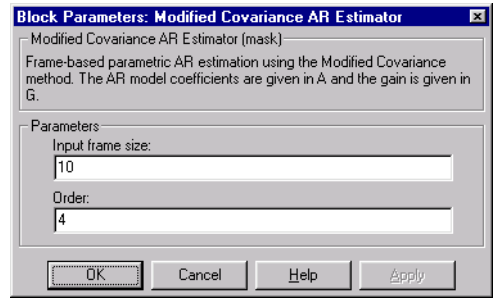
The order, p, of the all-pole model is specified by the **Order** parameter.

The top output, `A`, 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`).

Dialog Box



Input frame size

The number of samples in the input frame.

Order

The order of the AR model.

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.

See Also

Burg AR Estimator
Covariance AR Estimator
Modified Covariance Method
Yule-Walker AR Estimator
`armcov` (Signal Processing Toolbox)

Purpose

Compute a parametric spectral estimate using the modified covariance method.

Library

Power Spectrum Estimation, in Estimation

Description

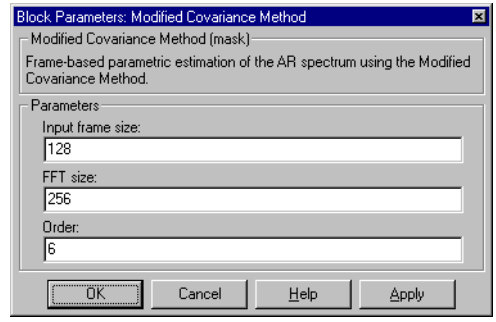
The Modified Covariance Method block estimates the power spectral density (PSD) of the input using the modified covariance method. This method fits an autoregressive (AR) model to the signal by minimizing the forward and backward prediction errors in the least-squares sense. The spectrum is then computed from the FFT of the estimated AR model parameters.

The order of the all-pole model is specified by the **Order** parameter.

The input is a frame of consecutive time samples; a matrix input, `u`, is also treated as a single frame, `u(:)`. The block's output is the estimate of the signal's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where N_{fft} is specified as a power of 2 by the **FFT Size** parameter and F_s is the signal's sample frequency. A value of `-1` for **FFT size** instructs the block to use the input frame size as the FFT size. Otherwise, the block zero pads or truncates the input to N_{fft} before computing the FFT.

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



Input frame size

The number of samples in the input frame.

FFT size

The number of samples on which to perform the FFT, N_{fft}. If N_{fft} exceeds the frame size, the data is zero padded as needed.

Order

The order of the AR model.

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.

See Also

Burg Method
Covariance Method
Modified Covariance AR Estimator
Short-Time FFT
Yule-Walker Method
`pmcov` (Signal Processing Toolbox)

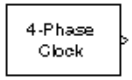
Purpose

Generate multiple binary clock signals.

Library

Switches and Counters, in General DSP

Description



The Multiphase Clock block generates a vector of N clock signals, where N is specified by the **Number of phases** parameter. All phases share the same frequency, *f*, specified in Hertz by the **Clock frequency** parameter.

The output 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 following the preceding signal's activation by 1/(N∗*f*) seconds, the *phase interval*. The output sample period is therefore 1/(N∗*f*) seconds.

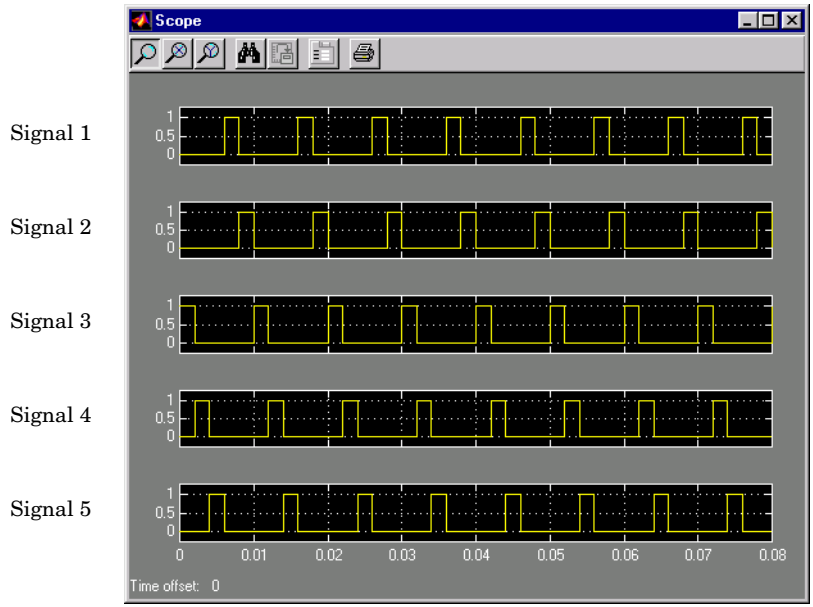
For example, if **Starting phase** is set to `3` for a 100 Hz five-phase output `y`, the output signals become active at the following times.

Signal	Output	Becomes Active at Time:
3	`y(3)`	*t* = 0.000, 0.010, 0.020, 0.030, ...
4	`y(4)`	*t* = 0.002, 0.012, 0.022, 0.032, ...
5	`y(5)`	*t* = 0.004, 0.014, 0.024, 0.034, ...
1	`y(1)`	*t* = 0.006, 0.016, 0.026, 0.036, ...
2	`y(2)`	*t* = 0.008, 0.018, 0.028, 0.038, ...

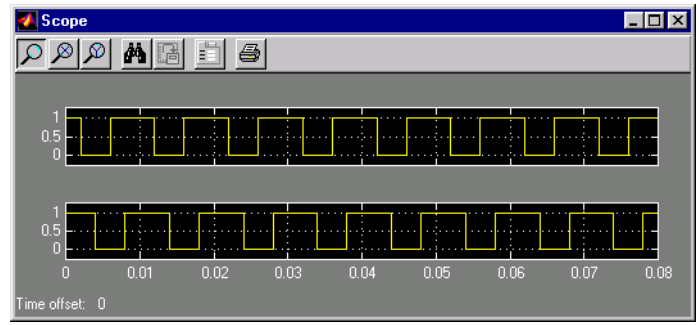
The first Scope window below shows the Multiphase Clock block's output for these settings. Note that the first active level appears at *t*=0 on `y(3)`, the second active level appears at *t*=0.002 on `y(4)`, the third active level appears at *t*=0.004 on `y(5)`, the fourth active level appears at *t*=0.006 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.

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. (The Scope window immediately below

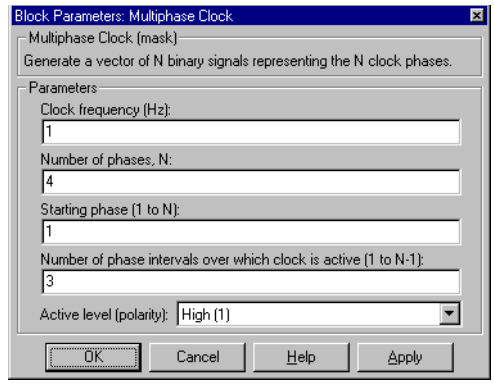
illustrates an **Active level** setting of **High** and a **Number of phase intervals over which clock is active** setting of 1.)



The Scope below shows Signals 1 and 2 with an active-level (high) duration of three phase intervals (60% duty cycle), corresponding to a setting of 3 for **Number of phase intervals over which clock is active**.



Dialog Box



Clock frequency

The frequency of all output clock signals.

Number of phases

The number of different phases 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).

See Also

Clock (Simulink)
Counter
Discrete Pulse Generator (Simulink)
Event-Count Comparator

N-Sample Enable

Purpose

Output ones or zeros for a specified number of sample times.

Library

Switches and Counters, in General DSP; DSP Sources

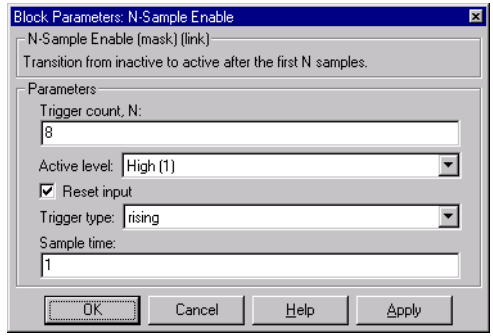
Description

The N-Sample Enable block outputs the *inactive* value (zero or one, whichever is *not* selected in the **Active** parameter pop-up) 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 (one or zero, whichever is selected in the **Active level** parameter pop-up) until a reset event or the end of the simulation.

The **Reset input** check box enables the `Trig` input port. At any time during the count, a trigger event at the input port resets the counter to its initial state. The triggering event is specified by the **Trigger type** pop-up menu, and can be one of the following:

- **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.

Dialog Box



Trigger count (T)

The number of samples for which the block outputs the active value.

Active level (T)

The value to output after the first `N` sample times, `0` or `1`.

Reset input

Enables the `Trig` input port.

Trigger type

The type of event that triggers a reset.

Sample time

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).

See Also

Counter
N-Sample Switch

N-Sample Switch

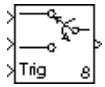
Purpose

Switch between two inputs after a specified number of sample periods.

Library

Switches and Counters, in General DSP

Description

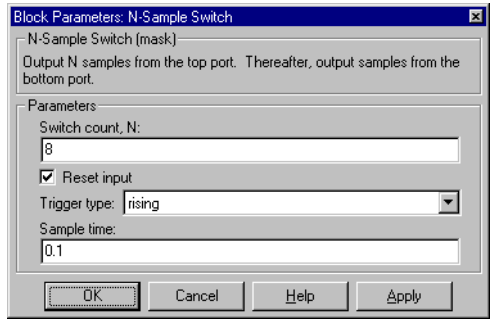


The N-Sample Switch block outputs the signal connected to the top input for the first N sample times after the simulation begins or the block is reset. Beginning at sample time N+1, the block outputs the signal connected to the bottom input until the next reset or the end of the simulation. The two inputs must be the same size except when one is a scalar, in which case the block expands the scalar to the size of the other input.

The block resets the count to zero when a trigger signal is received at the optional `Trig` port at any time during the simulation. Enable the `Trig` port by selecting the **Reset input** check box. The triggering event is specified by the **Trigger type** pop-up menu, and can be one of the following:

- **Rising edge** resets the block when the discrete trigger input rises from a negative value to zero or a positive value, or from zero to a positive value.
- **Falling edge** resets the block when the discrete trigger input falls from a positive value to zero or a negative value, or from zero to a negative value.
- **Either edge** resets the block when either a rising or falling edge (as described above) occurs.

Dialog Box



Switch count

The number of sample periods for which the output is connected to the top input before switching to the bottom input.

Reset input

Enables the `Trig` input port when selected.

Trigger type

The type of event at the `Trig` port that resets the block's counter.

Sample time

The sample period, T_s, for the block's counter. The block switches inputs at t=T_s*(N+1).

See Also

N-Sample Enable

Normalization

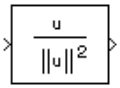
Purpose

Normalize an input vector by the 2-norm.

Library

Vector Functions, in Math Functions

Description



The Normalization block normalizes the input using the vector 2-norm or squared 2-norm, as specified by the **Norm** parameter. When **2-norm** is selected, the output is given by

$$y = \frac{u}{||u|| + b}$$

where u is the length-N input, b is specified by the **Normalization bias** parameter, and

$$||u|| = \sqrt{|u_1|^2 + |u_2|^2 + \cdots + |u_N|^2}$$

Equivalently,

```
y = u ./ (norm(u) + b)      % equivalent MATLAB code
```

The normalization bias, b, is typically chosen to be a small positive constant (e.g., 1e-10) that prevents potential division by zero.

When **Squared 2-norm** is selected, the output is given by

$$y = \frac{u}{||u||^2 + b}$$

where

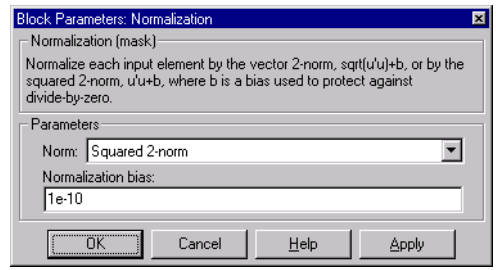
$$||u||^2 = |u_1|^2 + |u_2|^2 + \cdots + |u_N|^2$$

Equivalently,

```
y = u ./ (norm(u).^2 + b)   % equivalent MATLAB code
```

In both cases the output is the same size as the input. A matrix input, `u`, is treated as a vector, `u(:)`.

Dialog Box



Norm (T)

The type of normalization to apply, **2-norm** or **Squared 2-norm**. This parameter is not tunable in Simulink's external mode.

Normalization bias (T)

The value to add in the denominator (to avoid division by zero). This parameter is not tunable in Simulink's external mode.

See Also

Matrix Scaling
Reciprocal Condition
`norm` (MATLAB)

Overlap-Add FFT Filter

Purpose

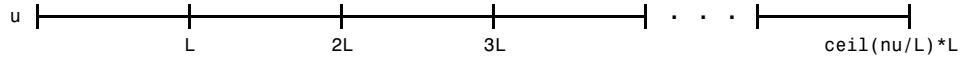
Implement the overlap-add method of frequency-domain filtering.

Library

Filter Realizations, in Filtering

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. The block breaks the input sequence u, of length nu, into length-L nonoverlapping data sections,



which it linearly convolves with the filter's FIR coefficients,

$$H(z) = \frac{B(z)}{1} = b_1 + b_2 z^{-1} + \ldots + b_{n+1} z^{-n}$$

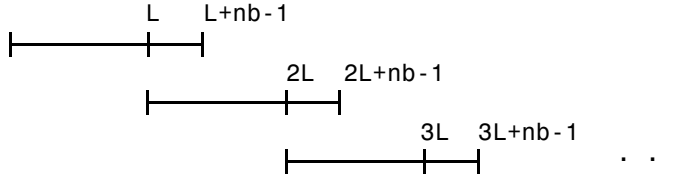
The numerator coefficients of the transfer function above are specified as a vector by the **FIR coefficients** parameter:

```
B = [b(1) b(2) ... b(n+1)]
```

The block's overlap-add operation is equivalent to

```
y = ifft(fft(u(i:i+L-1),nfft) .* fft(B,nfft))
```

where N_{fft} is specified by the **FFT size** parameter. The block overlaps successive output sections by nb-1 points, where nb is the length of the filter (`length(B)`), and sums them:

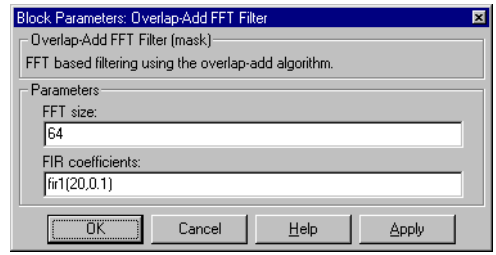


The first L samples of each summation are output in sequence. The block chooses the parameter L based on the length of the filter and the FFT size:

```
L = nfft-nb+1
```

Note that N_{fft} must be greater (and is typically *much* greater) than `length(B)`.

Dialog Box



FFT size

The size of the FFT, which must be greater than the length of the specified FIR filter.

FIR coefficients

The filter numerator coefficients.

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.

See Also

Direct-Form II Transpose Filter
Overlap-Save FFT Filter

Overlap-Save FFT Filter

Purpose

Implement the overlap-save method of frequency-domain filtering.

Library

Filter Realizations, in Filtering

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. The overlapping input sections are circularly convolved with the FIR filter coefficients,

$$H(z) = \frac{B(z)}{1} = b_1 + b_2 z^{-1} + \ldots + b_{n+1} z^{-n}$$

which are specified as a vector by the **FIR coefficients** parameter:

```
B = [b(1) b(2) ... b(n+1)]
```

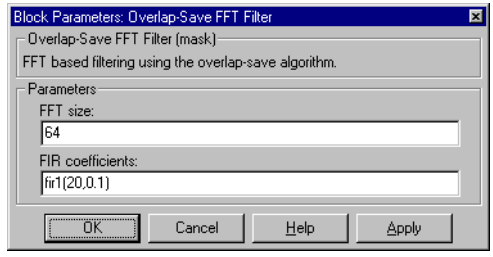
The block's overlap-save operation is equivalent to

```
y = ifft(fft(u(i:i+(L-1)),nfft) .* fft(B,nfft))
```

where u is the input and y is the output.

The circular convolution of each section is computed by multiplying the FFT of the input and the filter coefficients, and computing the inverse FFT of the product. For filter length nb (the number of FIR coefficients) and FFT size N_{fft}, the first $nb-1$ points of the circular convolution are invalid, and are discarded. The remaining $(N_{fft}-nb+1)$ points, which are equivalent to the linear convolution, are output in serial fashion by an Unbuffer block.

Dialog Box



FFT size

The size of the FFT, which must be greater than the length of the specified FIR filter.

FIR coefficients

The filter numerator coefficients.

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.

See Also

Direct-Form II Transpose Filter
Overlap-Add FFT Filter

Partial Unbuffer

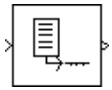
Purpose

Unbuffer a portion of an input frame to a sequence of scalar outputs.

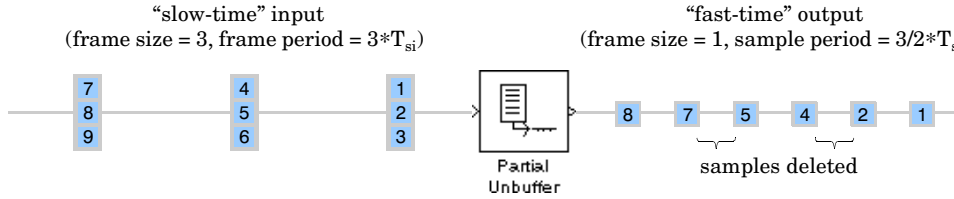
Library

Buffers, in General DSP

Description



The Partial Unbuffer block unbuffers a selected portion of the input frame into a sequence of scalar outputs. Multichannel (frame matrix) inputs are unbuffered into sample vectors, with the designated matrix *rows* being output in sequence. The sample-based output generally has a faster rate than the frame-based input.

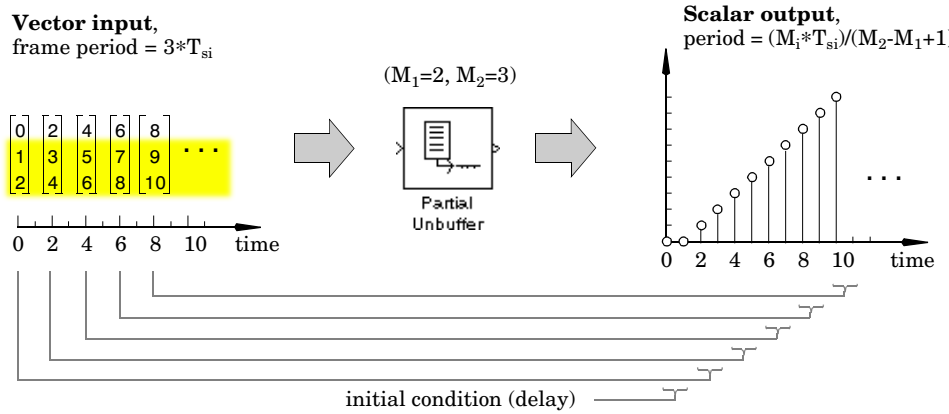


Because the block unbuffers only a portion of the samples in each input frame, the *sequence sample period* (i.e., the sample-to-sample interval) at the output is longer than that at the input, $T_{so}>T_{si}$.

Vector Inputs. Vector inputs are unbuffered to a scalar sequence. The scalar output sample period for an input of frame size M_i is a factor of $M_i/(M_2-M_1+1)$ longer than the input sample period, or

$$T_{so} = \frac{M_i T_{si}}{M_2 - M_1 + 1}$$

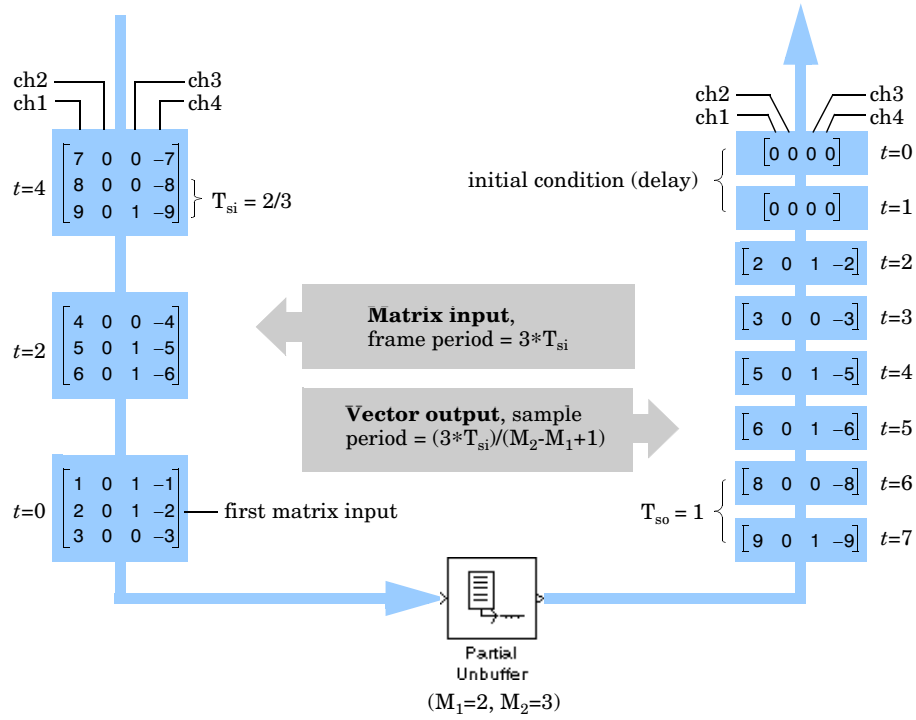
where M_1 is specified by the **First output index** parameter and M_2 and is specified by the **Last output index** parameter.



If all samples in the input frame are to be unbuffered (M_1=1 and M_2=M_i), you can use the Unbuffer block instead. To rebuffer the input samples to a larger or smaller frame size, use the Rebuffer block.

The **Number of channels** parameter, N, should typically be 1 for vector inputs, indicating that the input represents a single channel.

Matrix Inputs. The block's operation for vector inputs extends naturally to matrix inputs. A M_i-by-N matrix input represents a collection of N frames, each containing M_i sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent elements (columns, N) in the matrix. Matrix inputs are unbuffered *row-wise* so that each matrix row in the selected range becomes an independent time-sample in the vector output.



Initial Conditions

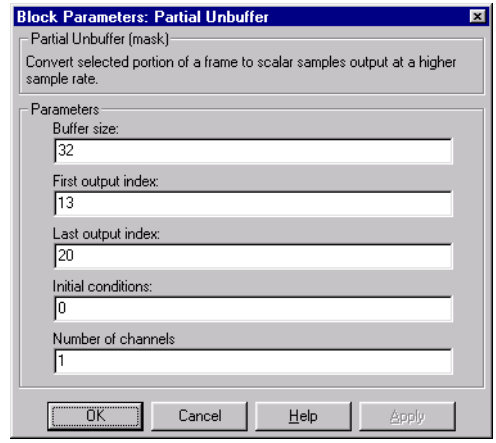
The Partial Unbuffer block's buffer is initialized with the value specified by the **Initial conditions** parameter, and the block begins unbuffering this frame (with element M_1) at the start of the simulation. Inputs to the block are therefore delayed by one partial-buffer length (M_2-M_1+1 elements, or M_i*T_{si} seconds).

If the block's output is a scalar (single channel), the **Initial condition** can be a scalar to be repeated at the output for the first M_2-M_1+1 sample times, or a vector containing M_2-M_1+1 samples to be output in sequence. If the block's output is a vector (N channels), the **Initial condition** can be a scalar value to be repeated across all elements of the initial output(s), a vector containing M_2-M_1+1 samples to be output in sequence for all channels, or an N-column matrix containing M_2-M_1+1 rows to be output in sequence.

The block generates an error if the **Last output index** is greater than the input frame size ($M_2 > M_i$), or if the **First output index** equals or exceeds the **Last output index** ($M_1 \geq M_2$).

Note If you expect to generate code for the Partial Unbuffer block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Buffer size

The number of elements in the input frame (number of rows in matrix), M_i.

First output index

The index, M_1, of the first input sample (matrix row) to propagate to the output.

Last output index

The index, M_2, of the last input sample (matrix row) to propagate to the output.

Initial conditions

The value of the block's initial output, a scalar, vector, or matrix.

Number of channels

The number of columns in the input, N. Use 1 for a vector input containing consecutive time-samples.

See Also

Buffer
Rebuffer
Unbuffer

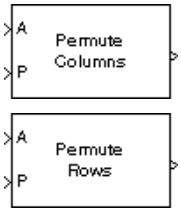
Purpose

Reorder the rows or columns of a matrix.

Library

Matrix Functions, in Math Functions

Description

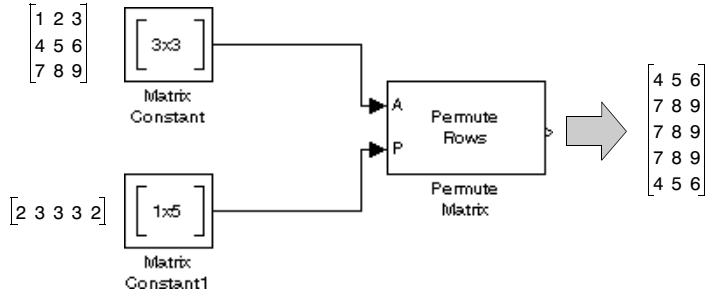


The Permute Matrix block reorders the rows or columns of the input matrix (A) as indicated by the indexing vector (P).

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. Vector P indicates where each row from A should be placed in the output matrix. The number of rows in the output matrix is determined by the number of elements in P.

```
% equivalent MATLAB code
y = [A(P(1),:) ; A(P(2),:) ; A(P(3),:) ; ... ; A(P(end),:)]
```

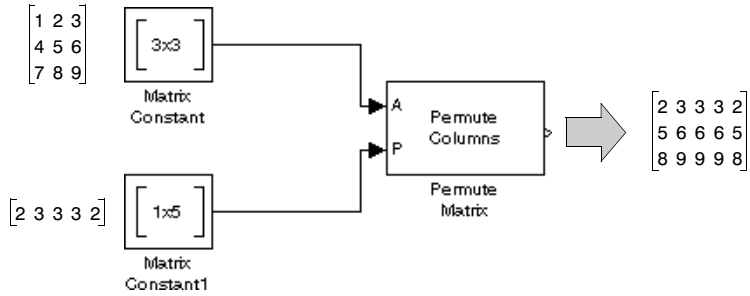
As shown below, rows of A can appear any number of times in the output, or not at all.



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. Vector P indicates where each column from A should be placed in the output matrix. The number of columns in the output matrix is determined by the number of elements in P.

```
% equivalent MATLAB code
y = [a(:,p(1)) a(:,p(2)) a(:,p(3)) ... a(:,p(end))]
```

Columns of A can appear any number of times in the output, or not at all.

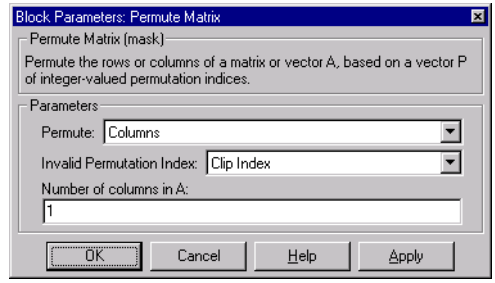


When an element in the vector P references a non-existent row or column of matrix A, the block reacts with the behavior specified by the **Invalid permutation index** parameter. The following options are available:

- **Clip index** – Clip the index to the nearest valid value. Do not issue an alert. Example: For a 3-by-7 input matrix, a column index of 9 is clipped to 7; a row 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.

Note If you expect to generate code for the Permute Matrix block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Permute

Method of constructing the output matrix; by permuting rows or columns of the input.

Invalid permutation index Ⓣ

Response to an invalid index value. This parameter is not tunable in Simulink's external mode.

Number of columns in A

Number of columns in the input matrix.

See Also

Submatrix
Transpose
Variable Selector
`permute` (MATLAB)

QR Factorization

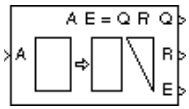
Purpose

Factor a rectangular matrix into unitary and upper triangular components.

Library

Linear Algebra, in Math Functions

Description



The QR Factorization block uses 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. The column-pivoted matrix A_e contains the columns of A permuted as indicated by the length-N permutation vector E.

```
Ae = A(:,E)        % equivalent MATLAB code
```

Example:

$$A = \begin{bmatrix} 9 & -1 & 2 \\ -1 & 8 & -5 \\ 2 & -5 & 7 \end{bmatrix} \qquad E = \begin{bmatrix} 1 & 3 & 2 \end{bmatrix} \qquad A_e = \begin{bmatrix} 9 & 2 & -1 \\ -1 & -5 & 8 \\ 2 & 7 & -5 \end{bmatrix}$$

The column permutation vector E is selected to ensure that the diagonal elements of matrix R are arranged in order of decreasing magnitude.

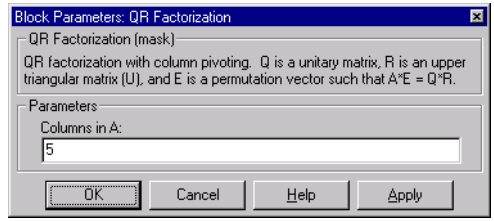
$$|r_{i+1,j+1}| > |r_{i,j}| \qquad i = j$$

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.

Dialog Box



Columns in A

The number of columns in input matrix A.

References

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

See Also

Cholesky Factorization
LU Factorization
QR Solver
`qr` (MATLAB)

QR Solver

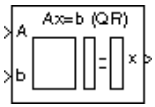
Purpose

Find a minimum-norm-residual solution to the equation A*x*=b.

Library

Linear Algebra, in Math Functions

Description



The QR Solver block solves the linear system A*x*=b by applying QR factorization to the input matrix A, which can be over- or under-determined (i.e., rectangular). The bottom input is the right-hand-side of the equation, b.

The output, *x*, is a solution to the equations that minimizes the 2-norm of the residual b-A*x*. Note that *x* itself is not guaranteed to be the minimum-norm solution.

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

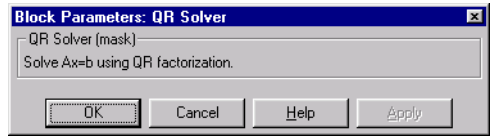
$$QRx = b_e$$

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*:

$$Rx = y$$

The block may generate `NaN` or `Inf` for inconsistent (overdetermined) systems.

Dialog Box



See Also

Levinson Solver
LU Solver
QR Factorization

Queue

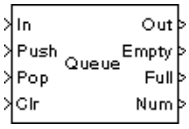
Purpose

Store inputs in a FIFO register.

Library

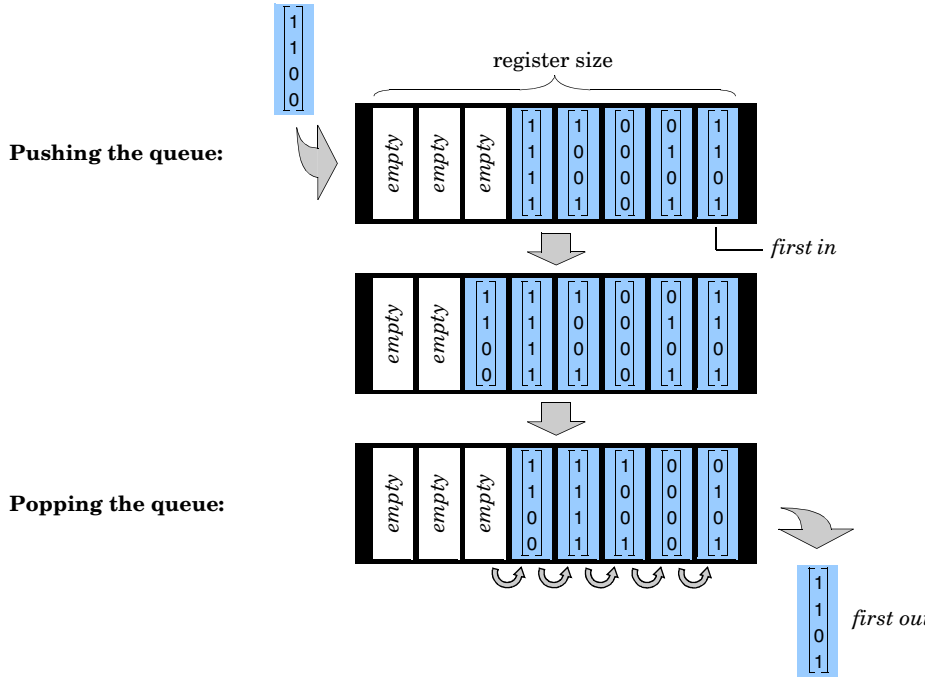
Buffers, in General DSP

Description



The Queue block stores a sequence of input samples in a FIFO (first in, first out) 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 `Clr` port (enabled by the **Clear input** check box) empties the queue contents. If **Clear output port on reset** is selected, then a trigger event at the `Clr` 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 if **Clear output port on reset** is selected.

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. `Clr`
2. `Push`
3. `Pop`

The triggering event for the `Push`, `Pop`, and `Clr` ports is specified by the **Trigger type** pop-up menu, and can be one of the following:

- **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 **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.

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 **Output number of register entries** option.

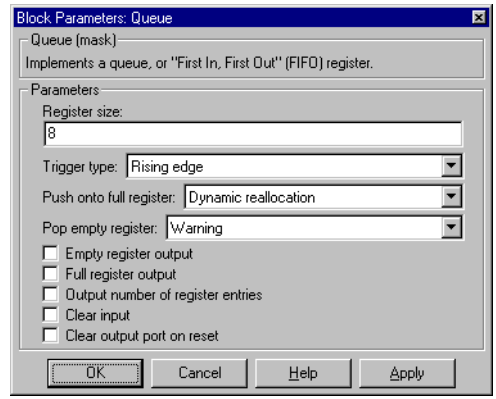
The table below illustrates the Queue block's operation for a **Register size** of `4`, **Trigger type** of **Either edge**, and **Clear output port on reset** enabled. Because the block triggers on both rising and falling edges in this example,

each transition from `1` to `0` or `0` to `1` in the `Push`, `Pop`, and `Clr` 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	Clr	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
9	0	0	0	*top* *bottom*	5	1	0	0
10	1	0	0	*top* 10 *bottom*	5	0	0	1
11	0	0	0	*top* 11 10 *bottom*	5	0	0	2
12	1	0	1	*top* 12 *bottom*	0	0	0	1

Note that at the last step shown, the `Push` and `Clr` ports are triggered simultaneously. The `Clr` trigger takes precedence, and the queue is first cleared and then pushed.

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.

Push onto full register

Response to a trigger received at the `Push` port when the register is full.

Pop empty register Ⓣ

Response to a trigger received at the `Pop` port when the register is empty.

Empty register output

Enable the `Empty` output port, which is high (`1`) when the queue is empty, and low (`0`) otherwise.

Full register output

Enable the `Full` output port, which is high (`1`) when the queue is full, and low (`0`) otherwise. The `Full` port remains low when **Dynamic reallocation** is selected from the **Push onto full register** parameter.

Output number of register entries

Enable the `Num` output port, which tracks the number of entries currently on the queue.

Clear input

Enable the `Clr` input port, which empties the queue when the trigger specified by the **Trigger type** is received.

Clear output port on reset (T)

Reset the `Out` port to zero (in addition to clearing the queue) when a trigger is received at the `Clr` input port.

See Also

Stack
Rebuffer
Shift Register

Purpose

Compute the real cepstrum of an input.

Library

Transforms, in General DSP

Description

The Real Cepstrum block outputs the real cepstrum of the input frame.

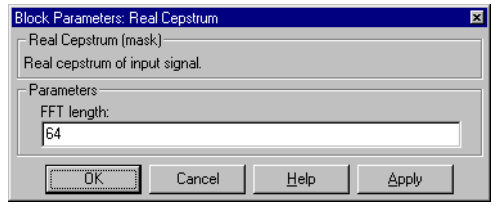

```
y = real(ifft(log(abs(fft(u)))))    % equivalent MATLAB code
```

or, more compactly,

```
y = rceps(u)
```

Multichannel inputs (i.e., frame matrices) are not accepted.

Dialog Box



FFT length

The number of samples to use in computing the FFT.

See Also

Complex Cepstrum
DCT
FFT
`rceps` (Signal Processing Toolbox)

Rebuffer

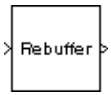
Purpose

Rebuffer the input sequence to a smaller or larger frame size.

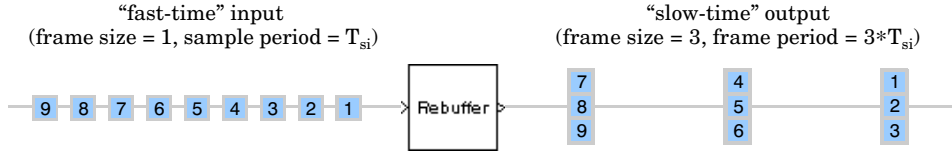
Library

Buffers, in General DSP

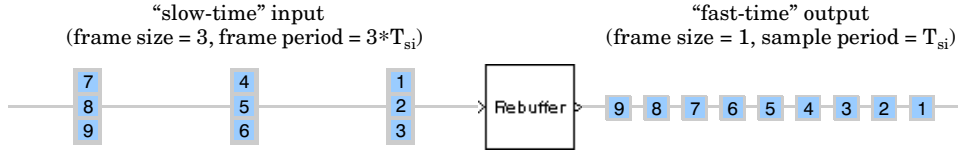
Description



The Rebuffer block redistributes the input samples to a new frame size, larger or smaller than the input frame size. Rebuffering to a larger frame size yields an output with a slower frame rate than the input.



Rebuffering to a smaller frame size yields an output with a faster frame rate than the input.



The block coordinates the output *frame size* and *frame rate* of nonoverlapping buffers so that the *sequence sample period* (i.e., the sample-to-sample interval) is the same at both the input and output, $T_{so}=T_{si}$.

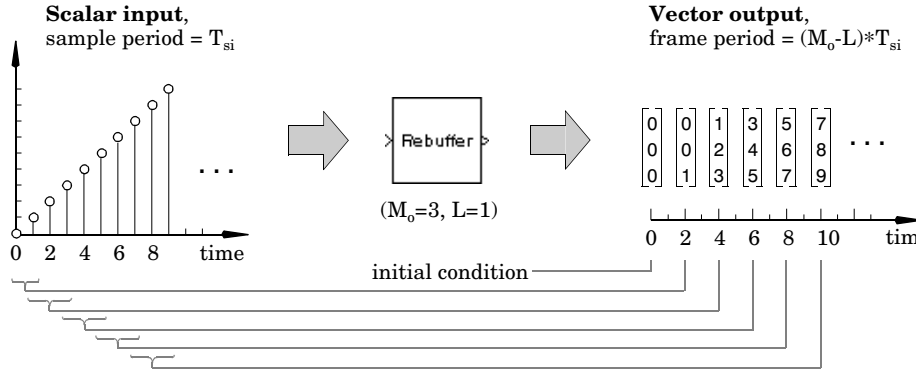
The **Frame-based inputs** check box allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

For sample-based operation (default), the **Frame-based inputs** check box in the parameter dialog box is *not* selected, and the Rebuffer block creates frame-based outputs from the sample-based inputs.

Scalar Inputs. Scalar inputs are buffered into a vector frame (top illustration, initial delay not shown). The size of the output frame, M_o, is determined by the **Buffer size** parameter. The **Buffer overlap** parameter specifies the number of samples, L, from the previous buffer to include in the current buffer. The number of *new* input samples the block acquires before propagating the

buffered data to the output is the difference between the **Buffer size** and **Buffer overlap**, M_o-L.

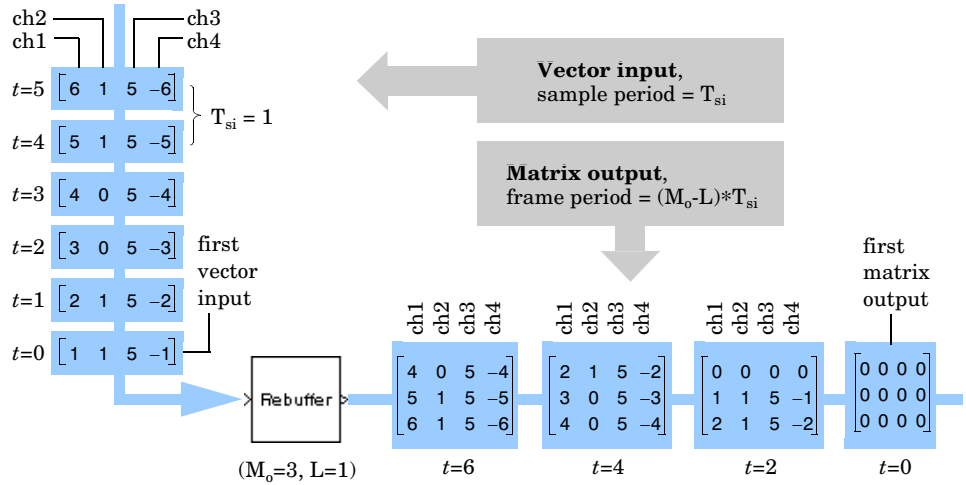


The output frame period is (M_o-L)*T_{si}, which is *equal* to the sequence sample period, T_{si}, when the **Buffer overlap** is M_o-1. For negative **Buffer overlap** values, the block simply discards L input samples after the buffer fills, and outputs the buffer with period (M_o-L)*T_{si}, which is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

Note To multiplex independent scalar channels into a multichannel vector signal, use the Simulink Mux block.

Vector Inputs. In sample-based operation, a sequence of length-N vector inputs is buffered into a M_o-by-N matrix, where M_o is specified by the **Buffer size** parameter. Each of the N input vector elements represents a distinct channel.

The illustration below shows the block operating on a 4-channel vector input with a **Buffer size** of 3 and a **Buffer overlap** of 1.



The output frame period of $(M_o\text{-}L)*T_{si}$ is the same as that for scalar inputs, described above.

Note that the input sample vectors do not begin appearing at the output until the second row of the second matrix. The first output matrix (all zeros in this example) reflects the block's initial buffer, while the first row of zeros in the second output is a result of the one-sample overlap between consecutive output frames.

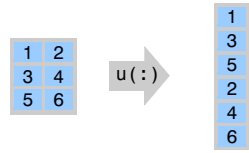
You can use the `rebuffer_delay` function with a frame size of 1 to precisely compute the delay (in samples) for sample-based signals. For the above example,

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

d =
     4
```

This agrees with the four samples of delay (zeros) shown in the figure above.

Matrix Inputs. In sample-based operation, an M-by-N matrix input is treated as a single vector with M*N elements. In other words, the matrix input `u` is reshaped to the vector input `u(:)`.



Frame-Based Operation

For frame-based operation, the **Frame-based inputs** check box in the parameter dialog box is selected and the Rebuffer block redistributes the samples in the input frame to an output frame of different size and rate.

Scalar Inputs. Scalar input processing is the same for frame-based operation as for sample-based operation. The **Number of channels** parameter must be set to 1.

Vector Inputs. In frame-based operation, a length-M_i vector input represents a single frame of data (M_i sequential samples) from one channel. The **Number of channels** parameter should correspondingly be set to 1. The **Buffer size** specifies the size of the output frame, i.e., the number of consecutive samples from the input frame to rebuffer into the output frame. This value, M_o, can be greater or less than the number of samples in the input frame.

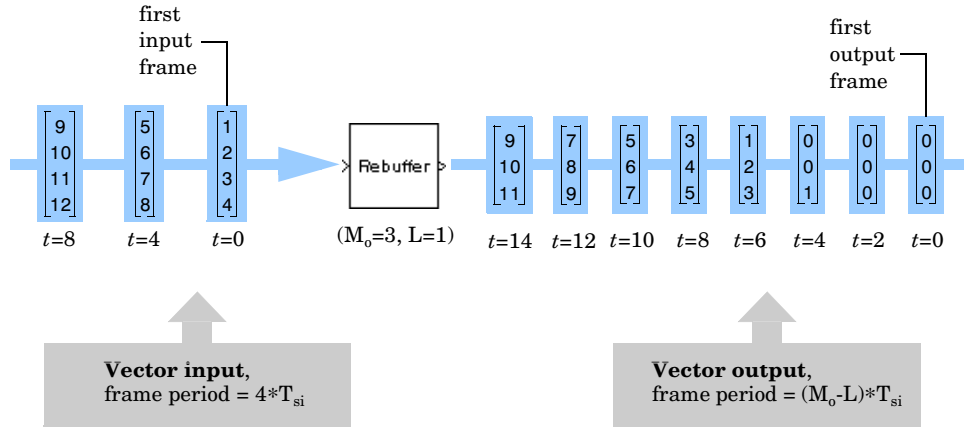
The **Buffer overlap** parameter specifies the number of samples, L, from the previous buffer to include in the current buffer. The number of *new* samples the block acquires before propagating the buffered data to the output is the difference between the **Buffer size** and **Buffer overlap**, M_o-L.

The input frame period is M_i*T_{si}, where T_{si} is the sequence 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.

The illustration below shows the block rebuffering a 1-channel input with a **Buffer size** of 3 and a **Buffer overlap** of 1.

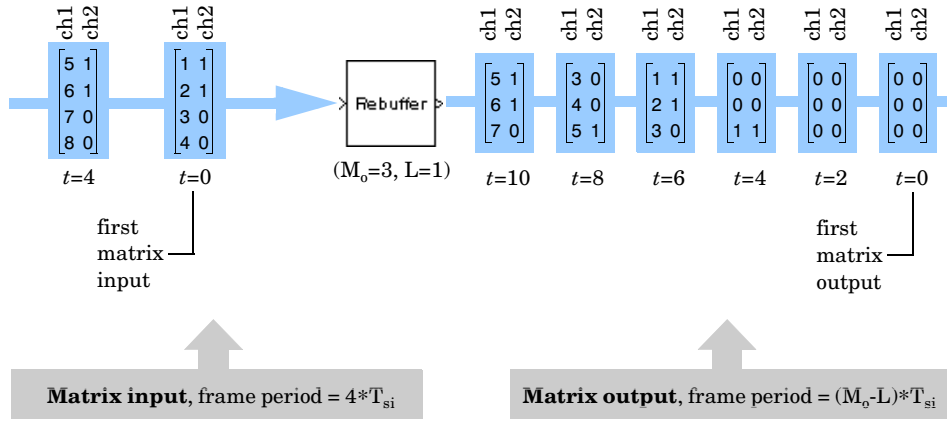


Note that the sequence is delayed by eight samples, and the first eight output samples adopt the value specified for the **Initial condition**, which is zero in this example. Use the `rebuffer_delay` function to determine the length of this initial delay for any combination of frame size and overlap.

Matrix Inputs. The block's operation for frame-based vector inputs extends naturally to frame-based matrix inputs. In frame-based operation, an M_i-by-N matrix input is treated as a collection of N frames, each containing M_i sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals (columns, N) in the matrix.

The **Buffer size** and **Buffer overlap** parameters work the same for matrix inputs as for vector inputs, described above. The number of columns in the output is the same as the input (i.e., each of the N channels is preserved in the output).

The figure below shows the block buffering a 2-channel input with a **Buffer size** of 3 and a **Buffer overlap** of 1.



Initial Conditions

The Rebuffer block's buffer is initialized to the value specified by the **Initial condition** parameter, and the block repeats this buffer at the output for the first D steps of the simulation (from t=0 to t=D-1), where D is the initial delay. For sample-based operation, D equals M_o-L. For frame-based operation, use the `rebuffer_delay` function to compute the exact delay (in samples) that the Rebuffer block introduces for a given combination of buffer size and buffer overlap.

For general frame-to-frame rebuffering, the **Initial condition** must be a scalar value, which is then repeated across all elements of the initial output(s). In the special case where the *input* is a N-channel sample vector (and the block's output is therefore an M_o-by-N frame matrix), the **Initial condition** can be:

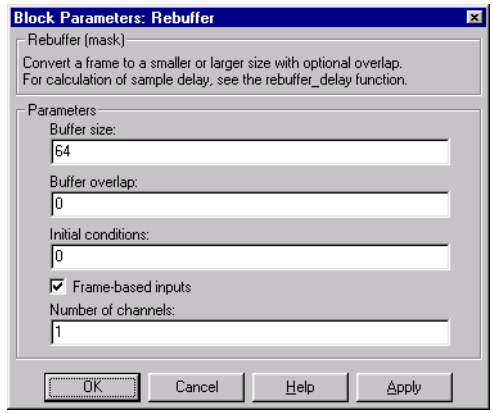
- An M_o-by-N matrix
- A length-M_o vector to be repeated across all columns of the initial output(s)
- A scalar to be repeated across all elements of the initial output(s)

In the special case where the *output* is a sample vector (the result of unbuffering an M_i-by-N frame matrix or a length-M_i frame vector), the **Initial condition** 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 If you expect to generate code for the Rebuffer block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Buffer size

The number of consecutive samples, M_0, from each channel to buffer into the output frame.

Buffer overlap

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

Initial conditions

The value of the block's initial output, a scalar, vector, or matrix.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame based operation, the number of channels (columns, N) in the input matrix.

See Also

Buffer
Distributor
Partial Unbuffer
Shift Register
Unbuffer
`rebuffer_delay`

Reciprocal Condition

Purpose

Compute the reciprocal condition of a square matrix in the 1-norm.

Library

Linear Algebra, in Math Functions

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 output ($0 \leq y < 1$).

The matrix 1-norm, $\|A\|_1$, is the maximum column-sum in the M-by-M matrix A.

$$\|A\|_1 = \max_{1 \leq j \leq M} \sum_{i=1}^{M} |a_{ij}|$$

or, for a 3-by-3 matrix:

$$\begin{bmatrix} a_{11} & a_{12} & a_{13} \\ a_{21} & a_{22} & a_{23} \\ a_{31} & a_{32} & a_{33} \end{bmatrix}$$

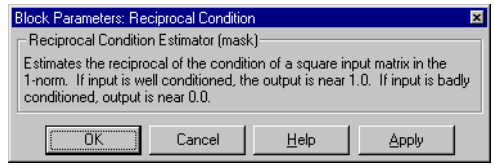
$$\|A\|_1 = \max(A_1, A_2, A_3)$$

$$|a_{13}| + |a_{23}| + |a_{33}| = A_3$$

$$|a_{12}| + |a_{22}| + |a_{32}| = A_2$$

$$|a_{11}| + |a_{21}| + |a_{31}| = A_1$$

Dialog Box



References

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

See Also

Matrix 1-Norm
Normalization
`rcond` (MATLAB)

Remez FIR Filter Design

Purpose

Design and apply an equiripple FIR filter.

Library

Filter Designs, in Filtering

Description

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 `remez` function in the Signal Processing Toolbox, 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 **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 rads/sec, 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 sampling frequency (the Nyquist 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**.

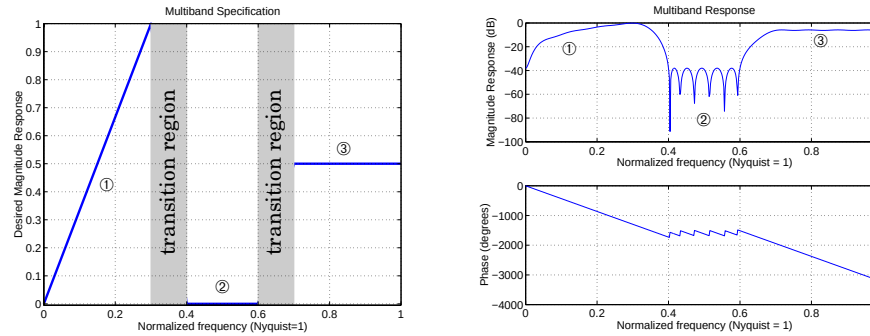
Each odd-indexed frequency-amplitude pair defines the left endpoint of a line segment representing the desired magnitude response in that frequency band. The corresponding even-indexed frequency-amplitude pair defines the right endpoint. Between the frequency bands specified by these end-points, there 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.

Band edge frequency = [0 0.3 0.4 0.6 0.7 1]

Gains = [0 1 0 0 0.5 0.5]

Band: ① ② ③



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.

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

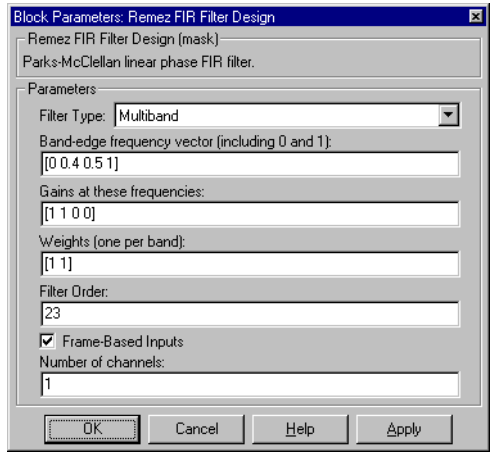
When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel over time.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block filters each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

In both sample-based and frame-based operation, the output is the same size as the input.

Dialog Box



Filter type

The filter type.

Band-edge frequency vector

A vector of frequency points, in ascending order, in the range `0` to `1`. The value `1` corresponds to the Nyquist frequency. This vector must have even length.

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.

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.

Filter order

The filter order.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (frames) in the input matrix.

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 the Nyquist 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`

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.

See Also

Digital FIR Filter Design
Least Squares FIR Filter Design
Yule-Walker IIR Filter Design
`firls` (Signal Processing Toolbox)
`remez` (Signal Processing Toolbox)

Purpose

Repeat the input sample a specified number of times.

Library

Signal Operations, in General DSP

Description

The Repeat block upsamples the input signal by repeating each consecutive input sample L times at the output, where L is specified by the **Repetition count** parameter. The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

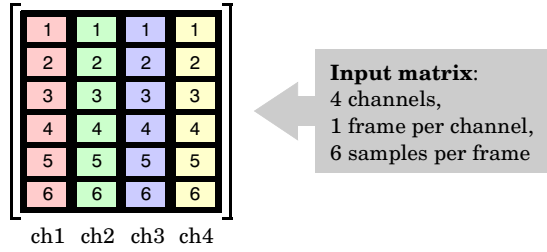
Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block repeats the value in each channel L times at the output. The output sample rate is L times higher than the input sample rate, and the input and output sizes are identical.

In sample-based mode, the **Initial condition** parameter specifies the value of the first L output samples, and can be a vector containing one value for each channel, or a scalar to be applied to all signal channels.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix. Frame-based operation provides

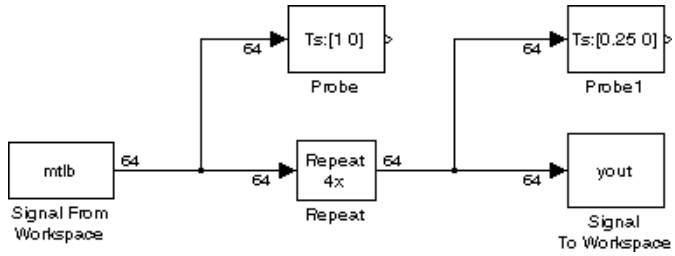
substantial increases in throughput rates, at the expense of greater model latency.

In frame-based operation, the block upsamples each channel independently by repeating each row in the input matrix L times at the output. The **Framing** 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 sample period, but the input and output frame sizes are equal.

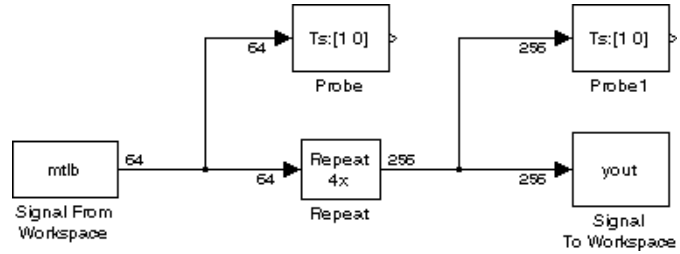
  The example below shows a single-channel input with a frame period of 1 second (**Sample time** = `1/64` and **Samples per frame** = `64` in the Signal From Workspace block) being upsampled through 4-times repetition to a frame period of 0.25 seconds. The input and output frame sizes are identical.



- **Maintain input frame rate**

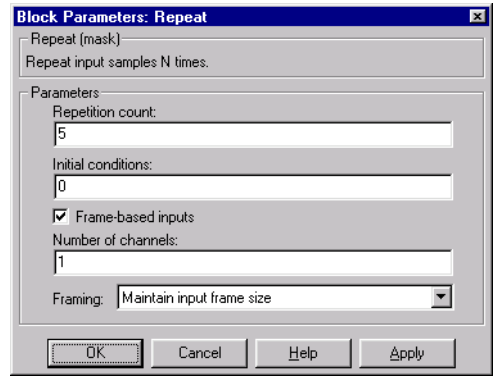
  The block generates the output at the faster (upsampled) rate by using a proportionally larger frame size than the input. For L repetitions of the input, the output frame size is L times larger than the input frame size, but the input and output frame rates are equal.

  The example below shows a single-channel input of frame size 64 being upsampled through 4-times repetition to a frame size of 256. The input and output rates are identical.



In frame-based mode, the **Initial condition** can be an M-by-N matrix representing the initial input, or a scalar to be repeated across all elements of the M-by-N matrix. The first row of the matrix is repeated at the output for the first L sample times, the second row is repeated for the next L sample times, and so on.

Dialog Box



Repetition count

The 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 that the block is initialized with; a scalar or vector in sample-based mode, or a scalar or matrix in frame-based mode. This value (first row in frame-based mode) is repeated at the output for the first L sample times.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (channels) in the input matrix.

Framing

For frame-based operation, the method by which to implement the repetition (upsampling): increase the output sample rate, or increase the output frame size.

See Also

FIR Interpolation
Upsample
Zero Pad

Purpose

Resize a matrix input.

Library

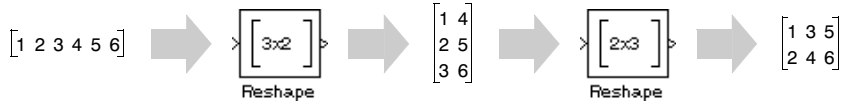
Matrix Functions, in Math Functions

Description

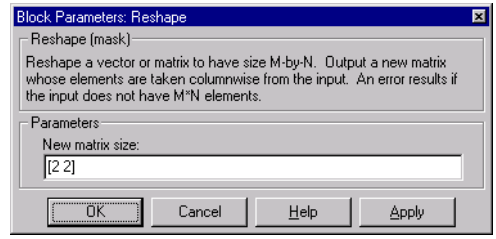
The Reshape block accepts a matrix input and outputs the data reshaped to the specified dimensions.


```
y = reshape(u,M,N)      % equivalent MATLAB code
```

where u is the real or complex input, and M and N are, respectively, the desired number of rows and columns in the output. Elements are read from the input matrix and redistributed to the output matrix in column-wise order.



Dialog Box



New matrix size

The dimensions of the output matrix in the format `[rows columns]`. The number of elements in the output must be the same as in the input.

See Also

Selector (Simulink)
Submatrix
Transpose
`reshape` (MATLAB)

RLS Adaptive Filter

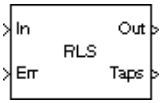
Purpose

Compute filter estimates for an input using the RLS adaptive filter algorithm.

Library

Adaptive Filters, in Filtering

Description



The RLS Adaptive Filter block recursively computes the least-squares estimate (RLS) of the FIR filter coefficients based on an externally generated error signal.

The corresponding RLS filter is expressed in matrix form as

$$
\begin{aligned}
k(n) &= \frac{\lambda^{-1}P(n-1)u(n)}{1+\lambda^{-1}u^{T}(n)P(n-1)u(n)} \\
y(n) &= \hat{w}^{T}(n-1)u(n) \\
e(n) &= d(n)-y(n) \\
\hat{w}(n) &= \hat{w}(n-1)+k(n)e(n) \\
P(n) &= \lambda^{-1}(I(n-1)-\lambda^{-1}k(n)u^{T}(n))P(n-1)
\end{aligned}
$$

where λ^{-1} denotes the inverse exponential weighting. 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
$y(n)$	The filtered output at step n
$e(n)$	The estimation error at step n
$d(n)$	The desired response at step n
λ	The memory weighting factor

The block icon has port labels corresponding to the inputs and outputs of the RLS algorithm.

Block Ports	Corresponding Variables
`In`	u, the scalar input, which is internally buffered into the vector $u(n)$ used by the algorithm
`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 the **Adapt input** check box is selected 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.

Note that the implementation of the algorithm in the block does not precisely parallel the above equations; symmetry of the inverse correlation matrix $P(n)$ is exploited to decrease the total number of computations by a factor of two.

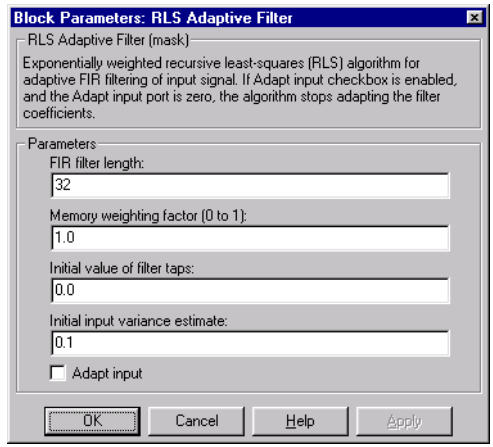
The **FIR filter length** parameter specifies the length of the filter that the RLS algorithm estimates. The **Memory weighting factor** corresponds to λ in the equations, and specifies how quickly the filter "forgets" past sample information. Setting λ=1 specifies an infinite memory; typically, $0.95 \le \lambda \le 1$.

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 $\hat{\sigma}^2$ is specified by the **Initial input variance estimate** parameter.

Dialog Box



FIR filter length

The length of the FIR filter.

Memory weighting factor (T)

The exponential weighting factor, in the range `[0,1]`. A value of `1` specifies an infinite memory.

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

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.

See Also

Kalman Adaptive Filter
LMS Adaptive Filter

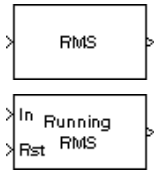
Purpose

Compute the root-mean-square (RMS) value of an input or sequence of inputs.

Library

Statistics, in Math Functions

Description



The RMS block computes the root-mean-square value of the elements in the input vector, or tracks the RMS value of a sequence of inputs over a period of time. The **Running RMS** parameter allows you to select between basic operation and running operation, which are described below.

Basic Operation

When the **Running RMS** check box is *not* selected, the block computes the RMS value of the input vector at each sample time.

```
y = sqrt(sum(u(:).^2)/length(u(:)))     % equivalent MATLAB code
```

This implements the following mathematical formula.

$$y = \sqrt{\frac{\sum_{i=1}^{n} u_i^2}{n}}$$

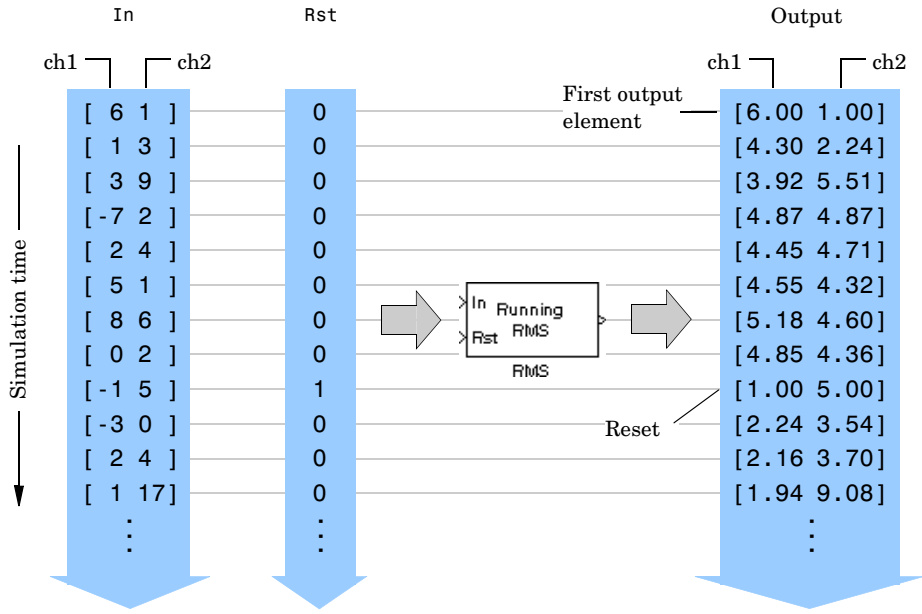
A matrix input is treated as a vector, `u(:)`.

Running Operation

When the **Running RMS** check box is selected, the block tracks the RMS value of a sequence of inputs over time. You can choose frame-based or sample-based operation by selecting (or deselecting, respectively) the **Frame-based** check box.

Sample-Based Operation. When the **Frame-based** check box is *not* selected (default), the block assumes that the input at the `In` port is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M*N matrix elements) is treated as an independent channel, and the block tracks the RMS value of each of the channels over time.

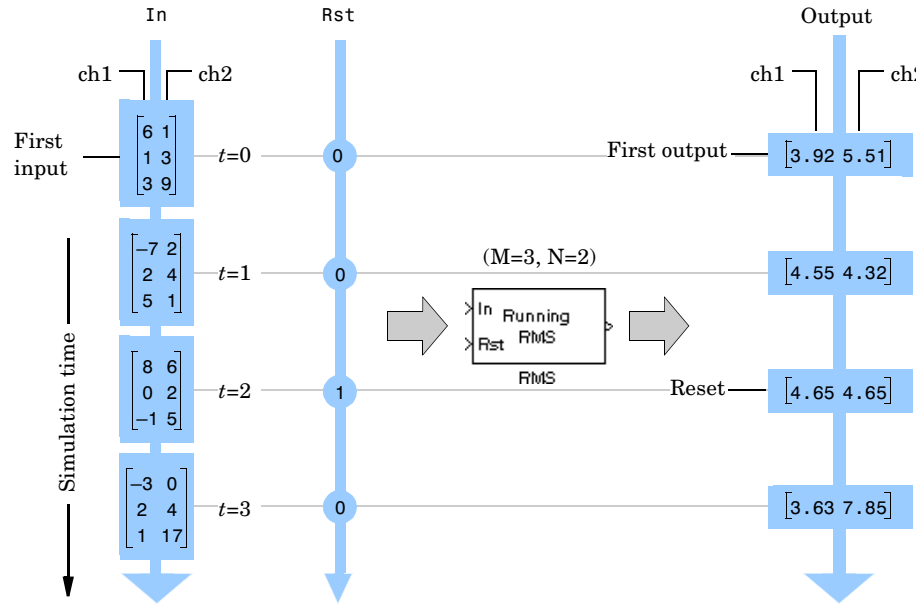
The block resets the running RMS value when the scalar input at the optional `Rst` port is nonzero. The output is the same size as the input, and contains the RMS value for each input channel since the last reset.



If you do not need to reset the running RMS value during the simulation, you can delete the `Rst` port from the block icon by deselecting the **Reset port** check box.

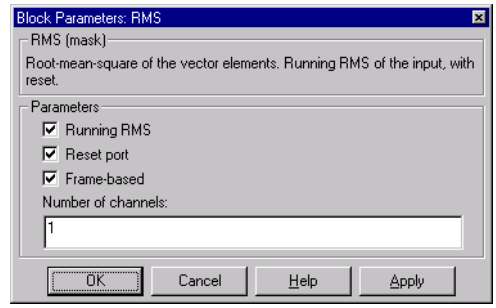
Frame-Based Operation. When the **Frame-based** check box is selected, the block assumes that the input at the `In` port is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals, N, in the matrix.

The block tracks the RMS value of each of the N independent channels over time, and resets the running RMS value when the input at the `Rst` port is nonzero. The output is a sample vector of length N which contains the RMS value for each input channel since the last reset.



Note If you expect to generate code for the RMS block's running mode using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Running RMS

Selects running operation.

Reset port

Enable `Rst` input port.

Frame-based

Selects frame-based operation.

Number of channels

For frame based operation, the number of channels (columns) in the input matrix, N.

See Also

Mean
Variance

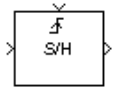
Purpose

Sample and hold an input signal.

Library

Switches and Counters, in General DSP

Description



The Sample and Hold block acquires a new sample (scalar, vector, or matrix) from the signal input whenever it is triggered by the control signal at the trigger input ( ). The block holds the output at the sampled input value until the next triggering event occurs.

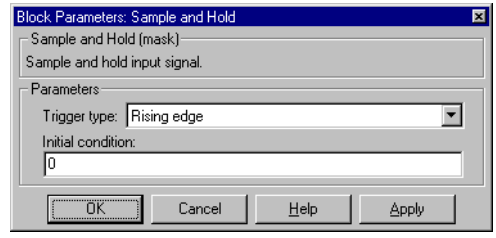
The triggering event at the top port is specified by the **Trigger type** pop-up menu, and can be one of the following:

- **Rising edge** triggers the block to sample the signal input when the control signal rises from zero to a positive value.
- **Falling edge** triggers the block to sample the signal input when the control signal falls from zero to a negative value.
- **Either edge** triggers the block to sample the signal input when the control signal either rises from zero to a positive value or falls from zero to a negative value.

The block's output prior to the first triggering event is specified by the **Initial condition** parameter. If the input 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 output. If the block's input is an M-by-N matrix, the **Initial condition** can be a scalar to be repeated across all M∗N elements of the output matrix, or a vector, `ic`, of length M∗N, to be reshaped to matrix dimensions.

```
y = reshape(ic,M,N)    % equivalent MATLAB code
```

Dialog Box



Trigger type

The type of event that triggers the block's to sample the input signal.

Initial condition

The block output prior to the first triggering event.

See Also

Downsample
N-Sample Switch

Purpose

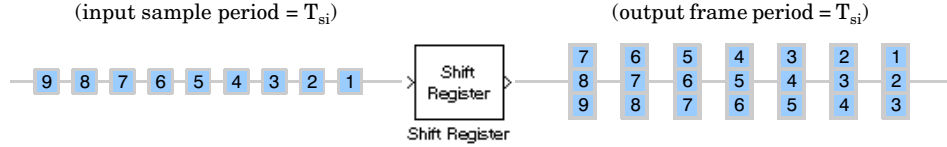
Buffer a sequence of inputs into a frame-based output with the same rate.

Library

Buffers, in General DSP

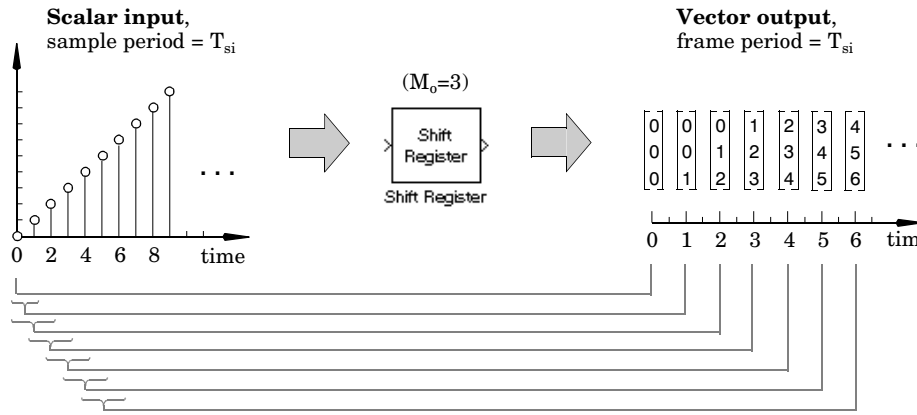
Description

The Shift Register block acquires a sequence of input samples into a frame. Each frame in the output sequence differs from the preceding output frame by only one sample, so the output frame period is the same as the input sample period.

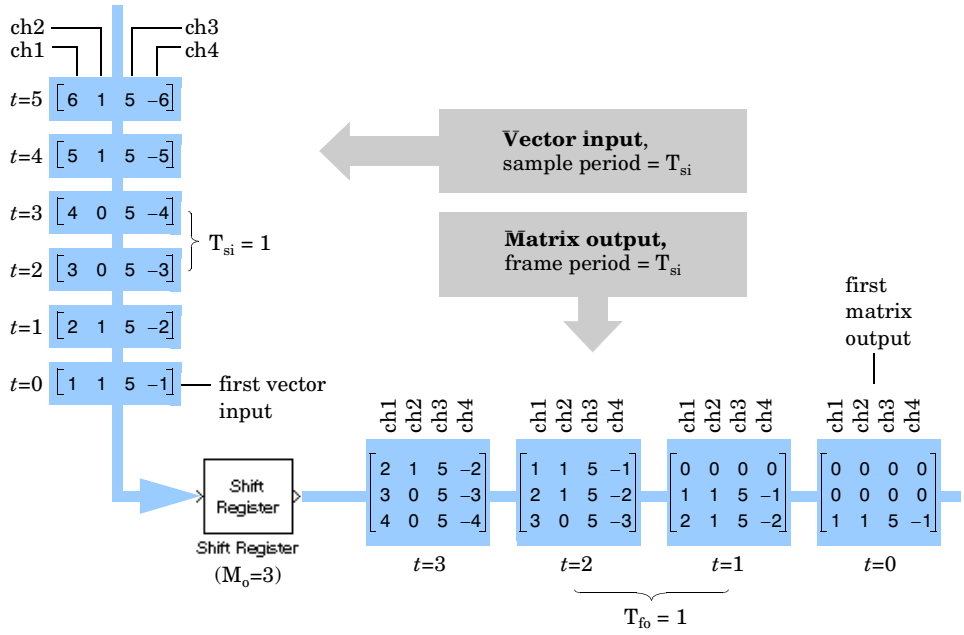


Scalar Inputs. Scalar inputs are buffered into a frame vector. The length of the output frame, M_o, is determined by the **Register size** parameter. At each sample time one new input sample is added to the output frame, so each output overlaps the previous output by M_o-1 samples, and the output frame period is the same as the input sample period (T_{fo}=T_{si}).

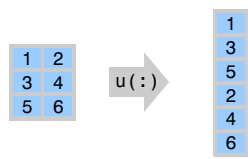
Note that this block is similar to a Buffer block with **Buffer size** equal to M_o and **Buffer overlap** equal to M_o-1, except that the Shift Register block supports *direct feedthrough* (the current input appears in the output frame at the same simulation time step).



Vector Inputs. Length-N vector inputs are buffered into a M_o-by-N matrix, where M_o is specified by the **Register size** parameter. Each of the N vector elements is treated as a distinct channel. The illustration below shows the operation for a four-channel input.



Matrix Inputs. An M-by-N matrix input is treated as a single vector with M∗N elements (channels). In other words, the matrix input `u` is reshaped to the vector input `u(:)`.



Initial Conditions

The Shift Register block's buffer is initialized to the value specified by the **Initial condition** parameter. The block outputs this buffer, with the addition of the first input sample, at the first simulation step (t=0). If 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. If 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.

Note If you expect to generate code for the Shift Register block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box

Register 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.

See Also

Buffer
Triggered Shift Register
Unbuffer

Short-Time FFT

Purpose

Compute a nonparametric estimate of the spectrum using the short-time, fast Fourier transform (ST-FFT) method.

Library

Power Spectrum Estimation, in Estimation

Description

The Short-Time FFT 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.

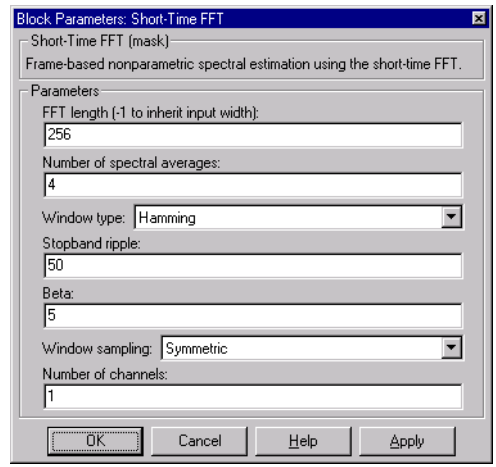
The block accepts an M-by-N frame matrix input, where each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent channels, N, in the matrix. A value of 1 for **Number of channels** specifies a single channel (vector) input.

The block computes a separate estimate for each of the N independent channels in the input, generating an N_{fft}-by-N matrix output, where N_{fft} is specified as a power of 2 by the **FFT Size** parameter. A value of `-1` for **FFT size** instructs the block to use the input frame size as the FFT size. Otherwise, the block zero pads or truncates 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 **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 reference page for the Window Function block for more details on these four parameters.

Dialog Box



FFT length

The number of data points, N_{fft}, on which to perform the FFT. If N_{fft} exceeds the input frame size, the frame is zero-padded as needed.

Number of spectral averages

The number of spectra to average; setting this parameter to `1` effectively disables averaging.

Window type (T)

The type of window to apply. (See the Window Function block reference.)

Stopband ripple (T)

The level (dB) of stopband ripple, R_s, for the Chebyshev window. Disabled for other **Window type** selections.

Beta (T)

The β parameter for the Kaiser window. Disabled for other **Window type** selections. Increasing **Beta** widens the mainlobe and decreases the amplitude of the window sidelobes in the window's frequency magnitude response.

Window sampling (T)

The window sampling, symmetric or periodic.

Number of channels

The number of channels (columns) in the input, N.

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.

See Also

Burg Method
FFT Frame Scope
Magnitude FFT
Window Function
Yule-Walker Method
`pwelch` (Signal Processing Toolbox)

Purpose

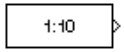
Acquire a signal from the workspace, and output at a constant sample rate.

Library

DSP Sources

Description

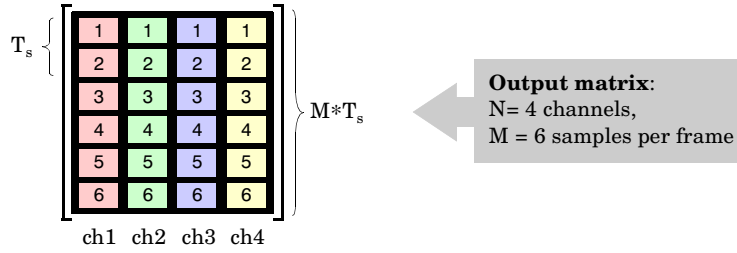
The Signal From Workspace block references a specified vector or matrix in the MATLAB workspace to generate scalar, vector, or matrix output.



The **Sample time** parameter value, T_s, specifies the sample period of the elements in the workspace vector, or the sample period of the *rows* in a workspace matrix. Each element of a workspace vector, and each *row* of a workspace matrix is considered to be an individual sample. Matrix columns represent independent channels.

The block acquires the number of input samples specified by the **Samples per frame** parameter value, M, and outputs this frame with a period of $M*T_s$. When the **Samples per frame** parameter is set to 1 (default), the block successively outputs the individual input samples at the sequence sample period, T_s.

For a general W-by-N workspace matrix, the output size is M-by-N, with each column representing a distinct signal channel. A 6-by-4 output matrix is illustrated below.

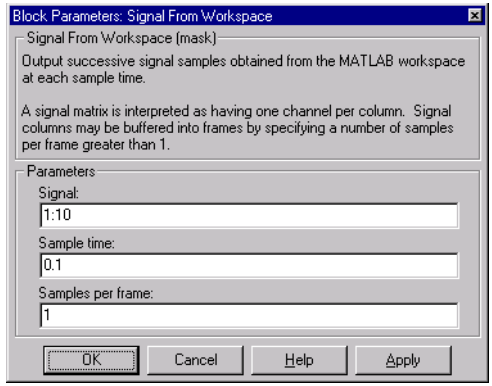


Note, however, that a 1-by-N matrix (row vector) is treated as a N-by-1 matrix (column vector), and acquired element-wise rather than row-wise.

Unlike Simulink's From Workspace block, the Signal From Workspace block holds the output value constant between successive output frames (i.e., no linear interpolation takes place), and does not extrapolate past the end of the indicated signal samples. The block outputs zeros (or zero-vectors or zero-matrices, as appropriate) when it has used all of the input samples. Additionally, the initial value is always output at the first simulation step.

Signal From Workspace

Dialog Box



Signal

The name of the workspace vector or matrix from which to acquire data, or a valid MATLAB expression.

Sample time

The sample period, T_s, of the input vector elements or matrix rows. The output frame period is $M*T_s$.

Samples per frame

The number of input samples (vector elements or matrix rows) to acquire into each output frame, M.

See Also

From Wave Device
From Wave File
Matrix From Workspace
Signal To Workspace
Sine Wave
Triggered Signal From Workspace

Purpose

Write the time-sequence of an input to the workspace.

Library

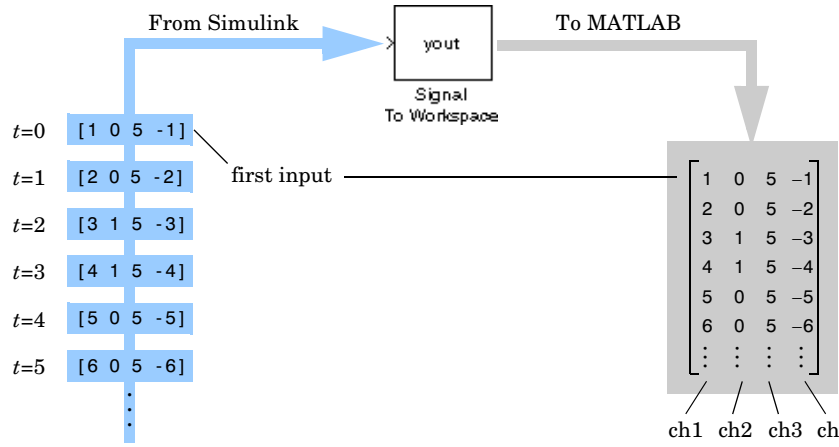
DSP Sinks

Description

The Signal To Workspace block creates a matrix variable in the workspace, where it stores the output at the end of a simulation. If the output variable name specified in the block's **Variable name** parameter already exists in the workspace, the original workspace variable is overwritten.

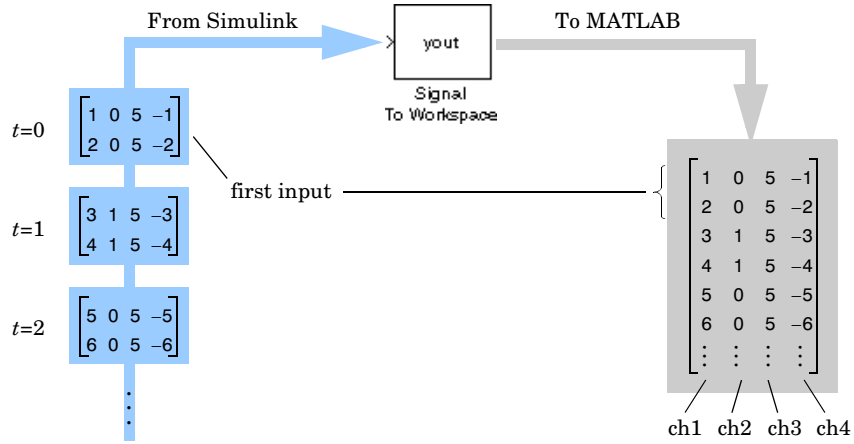
Sample-Based Operation

In sample-based mode (**Frame-based** check box *not* selected), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block writes each input sample to a row of the workspace matrix (beginning with the first row). A matrix input, `u`, is written to the workspace matrix as the row vector `u(:)'`.



Frame-Based Operation

In frame-based mode (**Frame-based** check box selected), the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent channels (columns, N) in the matrix. The block writes each M-by-N input to M rows of the workspace matrix.



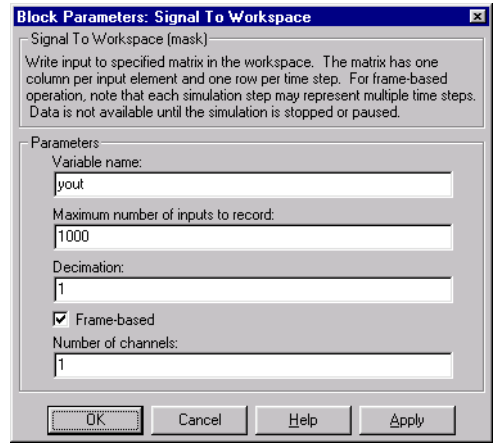
The maximum size of this workspace variable is limited to M*P rows by the **Maximum number of inputs to record** (P) parameter. (If the simulation progresses long enough for the block to sample more than P times, it stores only the last P inputs.) The **Decimation factor**, D, allows you to store only every Dth input.

To save a record of the sample time corresponding to each sample value, check the **Time** box in the **Save to workspace** parameters list of the **Simulation Parameters** dialog. You can access these parameters by selecting **Parameters** from the **Simulation** menu, and clicking on the **Workspace I/O** tab.

Note The Signal To Workspace block does not support real-time data logging with the Real-Time Workshop when a value of `inf` is specified for either the **Maximum number of inputs to record** parameter or the simulation **Stop time** parameter (in the **Simulation Parameters** dialog box).

If you expect to generate code for the block's frame-based mode, you should ensure that inputs to the block are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Variable name

The name of the workspace matrix in which to store the data. An existing workspace variable with the same name is overwritten

Maximum number of inputs to record

The maximum number of samples (or frames) to be saved, P. The default is 1000 samples (frames).

Decimation

The decimation factor. The default is 1.

Frame-based

Selects frame-based operation.

Number of channels

For frame-based operation, the number of channels (columns) in the input matrix, N.

See Also

Matrix To Workspace
Signal From Workspace
Triggered Signal To Workspace

Sine Wave

Purpose

Generate samples of one or more sine waves over time.

Library

DSP Sources

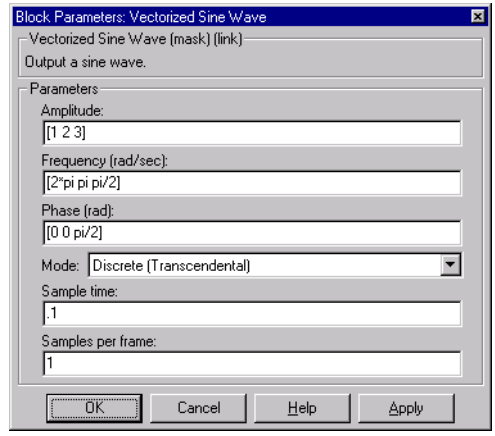
Description

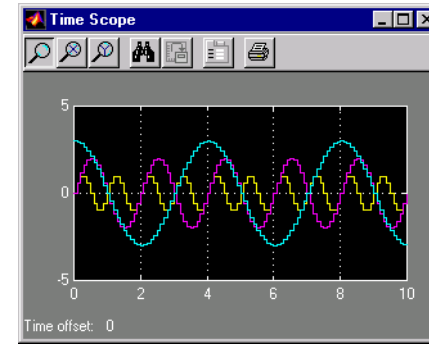
The Sine Wave block generates one or more sinusoidal signals with independent amplitude, frequency, and phase characteristics.

The **Amplitude**, **Frequency**, and **Phase** parameters specify the characteristics (A, ω, and ϕ) of the generated sine waves. Each parameter value can be a scalar or a length-N vector, where N is the number of sine waves to output. A particular sine wave in the output is defined by the corresponding elements of the A, ω, and ϕ vectors. For example, A_5, ω_5, and ϕ_5 define the characteristics of the fifth sinusoid in the output, y_5. If a scalar value is specified for one of these parameters, the value is applied to each output sinusoid.

The figure below shows the block dialog configured to generate 3 sinusoidal signals:

1. $A = 1$, $\omega = 2\pi$, $\phi = 0$
2. $A = 2$, $\omega = \pi$, $\phi = 0$
3. $A = 3$, $\omega = \pi/2$, $\phi = \pi/2$





The **Sample mode** parameter specifies the block's sampling property, **Continuous** or **Discrete**:

- **Continuous**

  In continuous mode, each sinusoid in the output, y_i, is computed as a continuous function

  $$y_i = A_i \sin(\omega_i t + \phi_i)$$

  and the block's output is continuous. In this mode, the block's operation is the same as that of the Simulink Sine Wave block when that block's **Sample time** is set to `0`. It 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.

- **Discrete**

  In discrete mode, the block's discrete-time output can be generated by directly evaluating the trigonometric function, or by a differential method. The two options are explained below.

Discrete Computational Methods

When **Discrete** is selected from the **Sample mode** parameter, the **Computation method** parameter provides two options for generating the discrete sinusoid, **Trigonometric functions** and **Differential method**.

- **Trigonometric functions**

  Each sinusoid in the output, y_i, is computed by sampling the continuous function

  $$y_i = A_i \sin(\omega_i t + \phi_i)$$

  with a period of T_s, where T_s is specified by the **Sample time** parameter. This mode of operation is a more efficient (but otherwise identical) implementation of a Simulink Sine Wave block with **Sample time** set to `0` followed by a Zero-Order Hold block with **Sample time** set to T_s. It shares the same benefits and liabilities as the **Continuous** sample mode, described above.

- **Differential method**

  The differential method uses an incremental (differential) algorithm rather than one based on absolute time. The algorithm computes the output

samples based on the values computed at the previous sample time and precomputed update terms, 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 each sinusoid in the output, y_i, can therefore be written in matrix form as

$$\begin{bmatrix} \sin\{\omega_i(t + T_s) + \phi_i\} \\ \cos\{\omega_i(t + T_s) + \phi_i\} \end{bmatrix} = \begin{bmatrix} \cos(\omega_i T_s) & \sin(\omega_i T_s) \\ -\sin(\omega_i T_s) & \cos(\omega_i T_s) \end{bmatrix} \begin{bmatrix} \sin(\omega_i t + \phi_i) \\ \cos(\omega_i t + \phi_i) \end{bmatrix}$$

where T_s is specified by 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 $A_i\sin[\omega_i(t+T_s)+\phi_i]$ is then computed from the values of $\sin(\omega_i t+\phi_i)$ and $\cos(\omega_i t+\phi_i)$ by a simple matrix multiplication at each time step.

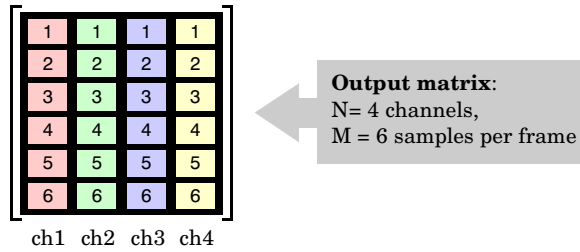
This mode of operation is the same as that of the Simulink Sine Wave block with **Sample time** set to a positive number (discrete). It offers reduced computational effort, but is subject to drift over time due to the cumulative quantization errors. Because the method is not contingent on an absolute time, there is no danger of discontinuity during extended operations (when an absolute time variable might overflow). This is therefore the recommended method to use when running long simulations and real-time systems.

Frame-Based Operation

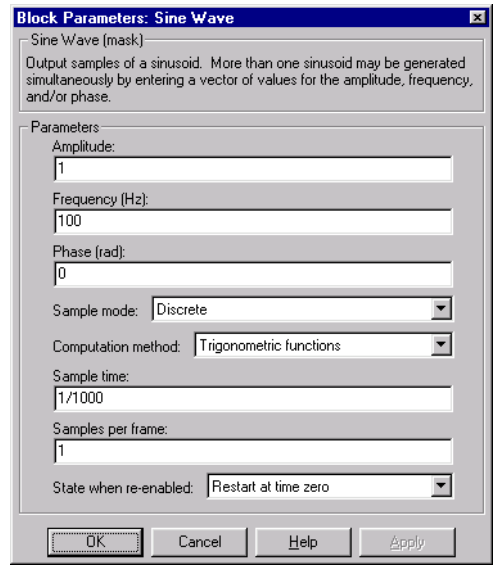
In both discrete modes (differential or transcendental), the block can optionally buffer the output samples into frames.

In these modes the block computes and buffers the number of samples (for each individual sine) specified by the **Samples per frame** parameter value, M, and outputs this frame of samples with a frame period of $M*T_s$ (where T_s is specified by the **Sample time** parameter). When the **Samples per frame** value is 1 (default), the block successively outputs the individual samples with a sample period of T_s.

For the general case of N independent sinusoidal outputs, the output is an M-by-N matrix with each column representing a distinct signal channel.



Dialog Box



Amplitude

A vector containing the individual amplitudes of the sine waves in the output (one amplitude for each sine), or a scalar to be applied to all. The vector length must be the same as that specified for the **Frequency** and **Phase** parameters. The amplitude values can be altered while a simulation is running, but the vector length must remain the same.

Frequency

A vector containing the frequencies of the sine waves in the output (one frequency for each sine, in rads/sec), or a scalar to be applied to all. The

vector length must be the same as that specified for the **Amplitude** and **Phase** parameters. The frequency values can be altered while a simulation is running, but the vector length must remain the same. (The **Frequency** parameter is not tunable in Simulink's external mode when using the differential method.)

Phase Ⓣ

A vector containing the phase offsets of the sine waves in the output (one phase offset for each sine, in radians), or a scalar to be applied to all. The vector length must be the same as that specified for the **Amplitude** and **Frequency** parameters. The phase values can be altered while a simulation is running, but the vector length must remain the same. (The **Frequency** parameter is not tunable in Simulink's external mode when using the differential method.)

Sample mode

The block's sampling behavior, **Continuous** or **Discrete**.

Computation method

The method by which discrete-time sinusoids are generated.

Sample time

The period with which the sine wave is sampled, T_s. The block's output frame period is $M*T_s$, where M is specified by the **Samples per frame** parameter.

Samples per frame

The number of consecutive samples from each sinusoid to buffer into the output frame, M.

State when reenabled

The behavior of the block when a disabled subsystem that contains it is reenabled. 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**).

See Also

Chirp
Signal From Workspace
Signal Generator (Simulink)
Sine Wave (Simulink)
`sin` (MATLAB)

Sort

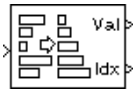
Purpose

Sort the elements in a vector by value.

Library

Statistics, in Math Functions

Description



The Sort block sorts the elements in a real input vector by value using a *Quicksort* algorithm. The output vector, y, contains the input values arranged in order of ascending or descending magnitudes, as specified by the **Sort order** parameter.

```
[y,i] = sort(u(:))          % equivalent MATLAB code (ascending)
[y,i] = flipud(sort(u(:)))  % equivalent MATLAB code (descending)
```

The **Mode** parameter specifies the block's output, and can be set to **Value**, **Index**, or **Value and Index**:

- **Value** – The block outputs only the sorted vector, y.
- **Index** – The block outputs the index vector, i, that permutes the input to the desired sorting order:

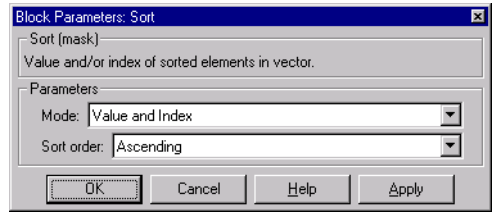
  ```
  y = u(i)
  ```

- **Value and Index** – The block generates both of the above outputs.

Note that a matrix input is sorted as a single vector, `u(:)`, rather than column by column.

Note If you expect to generate code for the Sort block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Mode

The block's mode of operation: Output the sorted vector, the index vector, or both.

Sort order Ⓣ

The order in which to sort the input values, **Descending** or **Ascending**. This parameter is not tunable in Simulink's external mode.

See Also

Histogram
Median
`sort` (MATLAB)

Stack

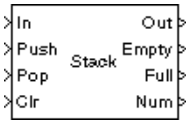
Purpose

Store inputs into a LIFO register.

Library

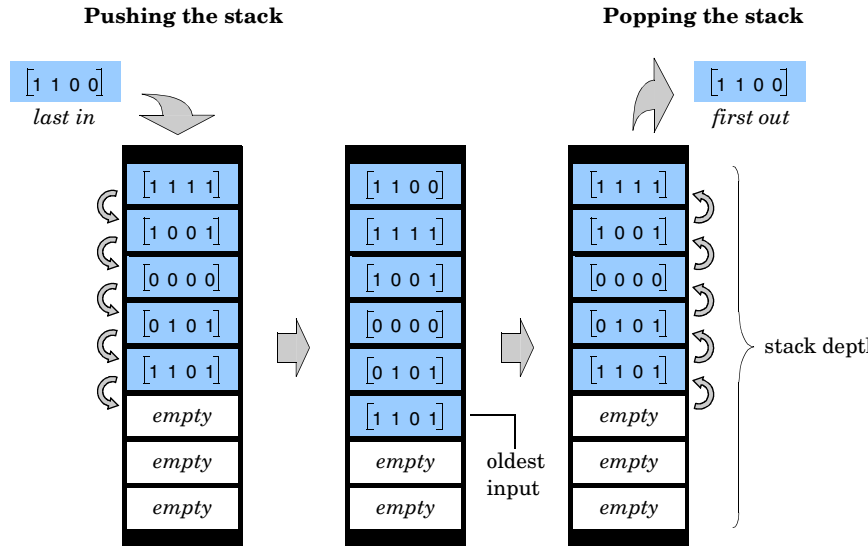
Buffers, in General DSP

Description



The Stack block stores a sequence of input samples in a LIFO (last in, first out) 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 `Clr` port (enabled by the **Clear input** check box) empties the stack contents. If **Clear output port on reset** is selected, then a trigger event at the `Clr` 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 re-enabled; the `Out` port value is only reset to zero in this case if **Clear output port on reset** is selected.

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. `Clr`
2. `Push`
3. `Pop`

The triggering event for the `Push`, `Pop`, and `Clr` ports is specified by the **Trigger type** pop-up menu, and can be one of the following:

- **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 **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.

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 **Output number of stack entries** option.

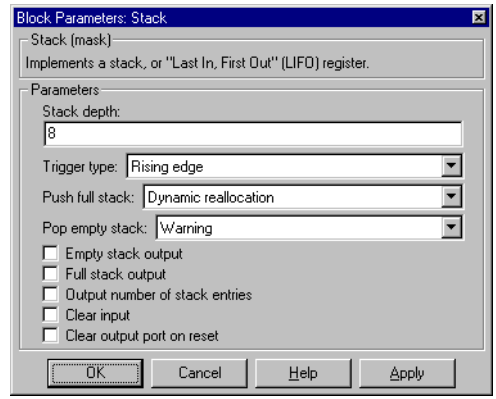
The table below illustrates the Stack block's operation for a **Stack depth** of `4`, **Trigger type** of **Either edge**, and **Clear output port on reset** enabled. Because the block triggers on both rising and falling edges in this example, each transition from `1` to `0` or `0` to `1` in the `Push`, `Pop`, and `Clr` 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	Clr	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 `Clr` ports are triggered simultaneously. The `Clr` trigger takes precedence, and the stack is first cleared and then pushed.

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.

Push full stack

Response to a trigger received at the `Push` port when the register is full.

Pop empty stack Ⓣ

Response to a trigger received at the `Pop` port when the register is empty.

Empty stack output

Enable the `Empty` output port, which is high (`1`) when the stack is empty, and low (`0`) otherwise.

Full stack output

Enable the `Full` output port, which is high (`1`) when the stack is full, and low (`0`) otherwise. The `Full` port remains low when **Dynamic reallocation** is selected from the **Push full stack** parameter.

Output number of stack entries

Enable the `Num` output port, which tracks the number of entries currently on the stack.

Clear input

Enable the `Clr` input port, which empties the stack when the trigger specified by the **Trigger type** is received.

Clear output port on reset (T)

Reset the `Out` port to zero (in addition to clearing the stack) when a trigger is received at the `Clr` input port.

See Also

Queue
Rebuffer
Shift Register

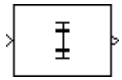
Purpose

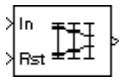
Find the standard deviation of an input or sequence of inputs.

Library

Statistics, in Math Functions

Description





The Standard Deviation block computes the standard deviation of the input vector, or tracks the standard deviation of a sequence of inputs over a period of time. The **Running standard deviation** parameter allows you to select between basic operation and running operation, which are described below.

Basic Operation

When the **Running standard deviation** check box is *not* selected, the block computes the standard deviation of the input vector at each sample time.

```
y = std(u(:))      % equivalent MATLAB code
```

This implements the mathematical formula

$$y = \sqrt{\frac{\sum_{i=1}^{n} |u_i - \mu_x|^2}{n-1}}$$

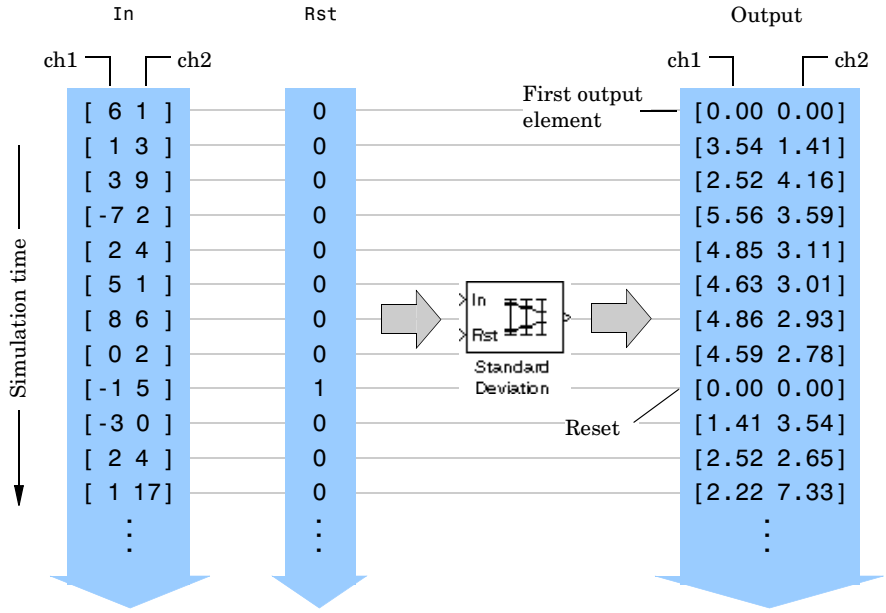
where μ_x is the mean of the input vector. The block outputs `0` for a scalar input, and treats a matrix input as a vector, `u(:)`

Running Operation

When the **Running standard deviation** check box is selected, the block tracks the standard deviation of a sequence of inputs over time. You can choose frame-based or sample-based operation by selecting (or deselecting, respectively) the **Frame-based** check box.

Sample-Based Operation. When the **Frame-based** check box is *not* selected (default), the block assumes that the input at the `In` port is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block tracks the standard deviation of each of the channels over time.

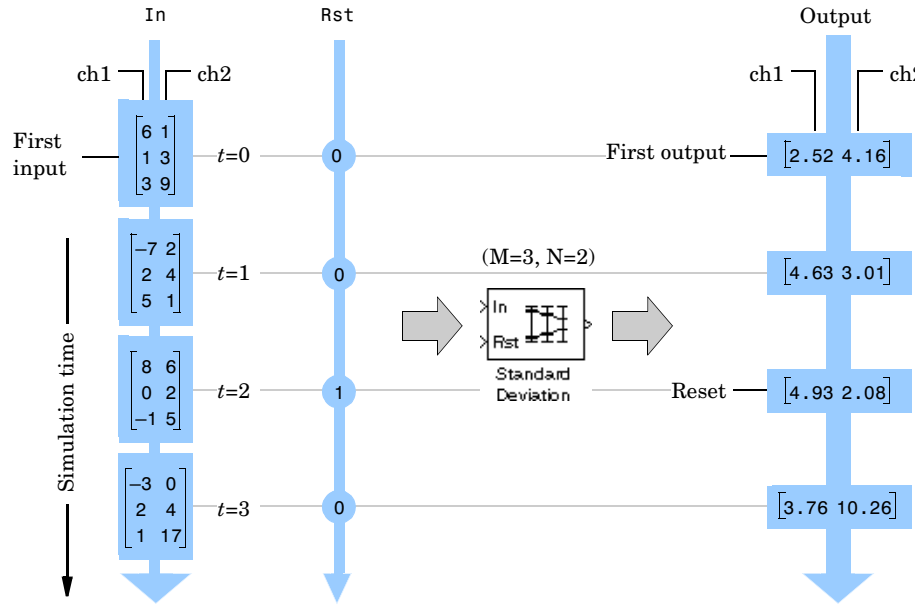
The block resets the running standard deviation when the scalar input at the optional `Rst` port is nonzero. The output is the same size as the input, and contains the standard deviation for each input channel since the last reset.



If you do not need to reset the running standard deviation during the simulation, you can delete the `Rst` port from the block icon by deselecting the **Reset port** check box.

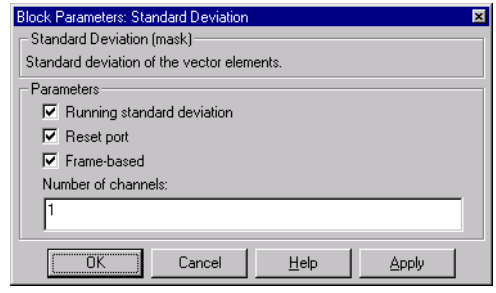
Frame-Based Operation. When the **Frame-based** check box is selected, the block assumes that the input at the `In` port is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals, N, in the matrix.

The block tracks the standard deviation of each of the N independent channels over time, and resets the running standard deviation when the input at the `Rst` port is nonzero. The output is a sample vector of length N which contains the standard deviation for each input channel since the last reset.



Note If you expect to generate code for the Standard Deviation block's running mode using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Running standard deviation

Selects running operation.

Reset port

Enable `Rst` input port.

Frame-based

Selects frame-based operation.

Number of channels

For frame based operation, the number of channels (columns) in the input matrix, N.

See Also

Mean
RMS
Variance
`std` (MATLAB)

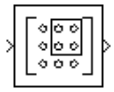
Purpose

Select a subset of elements (submatrix) in a matrix.

Library

Matrix Functions, in Math Functions

Description



The Submatrix block outputs a matrix that is a subset of the input matrix. The submatrix is specified by the **Index expression** parameter, a cell array containing a comma-separated MATLAB indexing expression, `expr`. For input `u` and output `y`, the indexing operation is equivalent to:

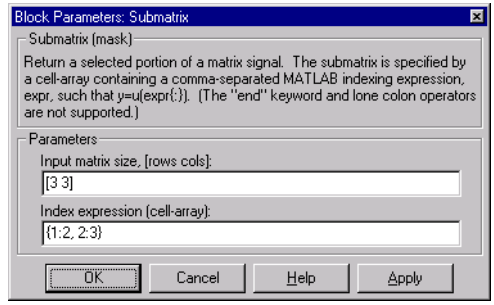
```
y = u(expr{:})      % equivalent MATLAB code
```

The `end` keyword and lone colon (`:`) operator are not supported in the indexing expression.

Some examples of valid indexing expressions are:

Index Expression	Equivalent MATLAB Code
`{1:2,2:3}`	`y = u(1:2,2:3)`
`{a,b:c}`	`y = u(a,b:c)`
`{[1 3:5],5}`	`y = u([1 3:5],5)`

Dialog Box



Input matrix size

The dimensions of the input matrix.

Index expression

The indexing expression, `{rows,columns}`. Note the cell array braces. The indexing expression can be changed while the simulation runs, although the dimensions of the specified submatrix must remain the same.

See Also

Selector (Simulink)
Variable Selector

Purpose

Display frame-based data.

Library

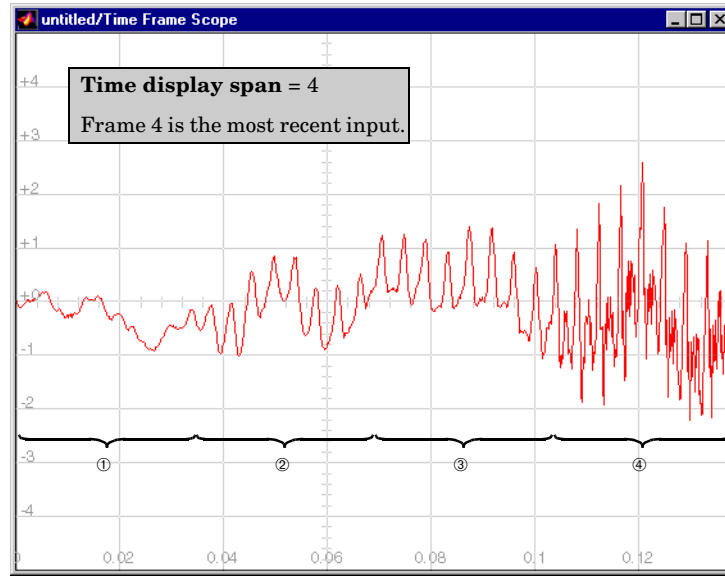
DSP Sinks

Description

The Time Frame Scope block is a comprehensive tool, similar to a digital oscilloscope, for displaying frame-based signals and data. The scope window, axis-property settings, line-property settings, and frame-handling capability are shared with the Frequency Frame Scope and User-Defined Frame Scope blocks.

The block assumes that each length-M input frame represents a block of M consecutive samples from a time-series. That is, each data point in the input frame is assumed to correspond to a unique time value, $u=u(t)$.

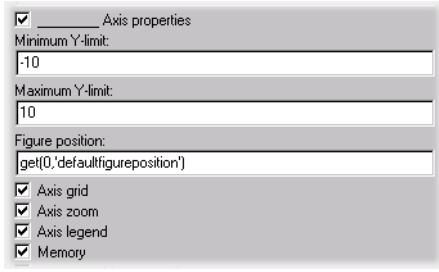
The scope updates the display for each new input frame. At any one time, the number of sequential frames displayed on the scope is specified by the **Time display span** parameter, S. Setting S equal to 1 plots the current input frame's data across the entire width of the scope. Setting S to a larger number allows you to see a broader section of the signal by fitting more frames of data into the display region. A single frame is the smallest unit that can be displayed, so S cannot be less than 1.



The range of the horizontal (time) axis is $[0,S*T_{fi}]$, where T_{fi} is the input frame period, and the spacing between time points is $T_{fi}/(M-1)$.

Axis Properties

All the frame scope blocks offer a similar collection of axis property settings. These can be exposed in the parameter dialog box by selecting the **Axis properties** check box. A complementary set of properties can be accessed under the **Axes** menus in the *unzoomed* scope view, or by right-clicking on the scope window. See the "Scope Window" section for more on these methods.



Minimum Y-limit and **Maximum Y-limit** set the range of the vertical axis. If **Autoscale** is selected from the right-click pop-up menu, the **Minimum Y-limit** and **Maximum Y-limit** values are automatically recalculated to best fit the range of the data on the scope.

The **Figure position** parameter specifies a 4-element vector of the form

```
[left bottom width height]
```

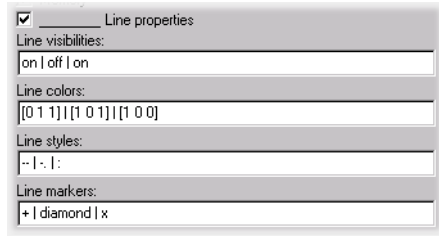
specifying the position of the scope window on the screen, where (0,0) is the lower-left corner of the display. See the MATLAB `figure` command for more information.

The **Axis grid** and **Axis legend** check boxes add or remove a grid and legend from the scope window. Click and drag on the legend to reposition it in the scope window; double click on the line label to edit the text.

The **Memory** parameter, when checked, causes the window to maintain successive displays (infinite persistence). That is, the scope does not erase the display after each frame (or collection of frames) is plotted, but *overlays* successive input frames in the scope display.

Line Properties

All the frame scope variations also offer a similar collection of line property settings. These can be exposed in the parameter dialog box by selecting the **Line properties** check box. These properties can also be accessed under the **Channels** menus in the *unzoomed* scope view, or by right-clicking on the scope window. See the "Scope Window" section for more on these methods.



The **Line properties** setting are typically used to help distinguish between two or more independent channels of data on the scope. See "Frame-Based Operation" for a description of how the block handles multichannel frame-based inputs.

The **Line visibilities** parameter specifies which channels' 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:

```
on | on | off | on | off
①    ②     ③     ④     ⑤
```

Note that the first (leftmost) list item corresponds to the first signal channel (leftmost column of the input matrix).

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 MATLAB's `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]
   ①         ②         ③          ④             ⑤
```

or

```
'k' | 'b' | 'r' | 'g' | [.7529 0 .7529]
 ①     ②     ③     ④           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 first (leftmost) list item, `'k'`, corresponds to the first signal channel (leftmost column of the input matrix). See `ColorSpec` in the online *MATLAB Function Reference* for more information about the color syntax.

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 instead plot each line with a different style, enter

```
- | -- | : | -. | -
①    ②    ③    ④    ⑤
```

These settings plot the signal channels with the following styles:

Line Style	Appearance
Solid	
Dashed	
Dotted	
Dash-dot	
Solid	

Note that the first (leftmost) list item, `'-'`, corresponds to the first signal channel (leftmost column of the input matrix). See `LineStyle` property of the `line` function (in the *MATLAB Function Reference*) for more information about the style syntax. To specify a marker for the individual sample points, use the **Line markers** parameter, described below.

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 (i.e., the individual sample points are not marked on the scope). To instead plot each line with a different marker style, you could enter

```
* | . | x | s | d
① ② ③ ④ ⑤
```

These settings plot the signal channels with the following styles:

Marker Style	Appearance
Asterisk	
Point	

Marker Style	Appearance
Cross	
Square	
Diamond	

Note that the first (leftmost) list item, `'*'`, corresponds to the first signal channel (leftmost column of the input matrix). See the `Marker` property of the `line` function (in the *MATLAB Function Reference*) for more information about the available markers.

Type the word `stem` instead of one of the basic `Marker` shapes to produce a *stem plot* for the data in a particular channel.

Scope Window

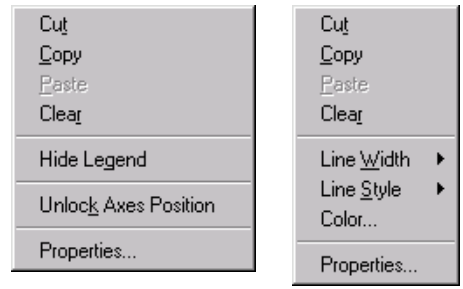
The scope title (in the window title bar) is the same as the block title. The axis scaling is set by parameters in the dialog box.

In addition to the standard MATLAB figure window menus (**File**, **Edit**, **Window**, **Help**), the Time Frame Scope window has an **Axes** and a **Channels** menu containing the following items:

- **Axes** (properties apply to all channels)
  - **Memory**, when selected, causes the window 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. This option can also be set in the **Axis properties** panel of the parameter dialog box.
  - **Refresh** erases all data on the scope display, except for the most recent trace. This command is useful in conjunction with the **Memory** 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 change them by editing those values.
  - **Axis grid** toggles the background grid on and off. This option can also be set in the **Axis properties** panel of the parameter dialog box.

- **Axis zoom**, when selected, causes the scope to completely fill the containing figure window. Menus and axis titles are not displayed, and the numerical axis labels are shown within the axes. This option can also be set in the **Axis properties** panel of the parameter dialog box.

  When **Axis zoom** is deselected, the axis labels and titles are displayed in a gray border surrounding the scope axes, and window's menus (including **Axes** and **Channels**) and toolbar are visible. The Plot Editor tools allow you to annotate and customize the contents of the scope. Select **Help Plot Editor** from the figure's **Help** menu for more information about using these tools. Note that when the Plot Editor is active, the right-click pop-up menu contains options for the Plot Editor rather than for the Scope.



Plot Editor right-click menus

  Deactivate the Plot Editor to access the Scope right-click pop-up menu.

  For information on printing or saving a figure, or on the other options found in the generic figure menus (**File**, **Edit**, etc.), see *Using MATLAB Graphics*.

- **Frame #**, when selected with **Axis zoom** off, displays the number of the current frame in the input sequence, incrementing the count as each new input is received. Counting starts at 1 with the first input frame, and continues until the simulation stops. The frame number is not shown in the zoomed view.
- **Legend**, when selected, adds a legend indicating the line color, style, and marker of each channel's data. Each channel in the legend is labeled with the channel number (`CH 1`, `CH 2`, etc.). Although you can edit the labels by double-clicking on them, the new edits are lost when the simulation runs

again, and the labels revert to the defaults. The **Legend** option can also be set in the **Axis properties** panel of the parameter dialog box.

  - **Save Position** automatically updates the **Figure 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, simply drag the window to the desired location, resize it as needed, and select **Save Position**. Note that the parameter dialog box must be closed when you select **Save Position** in order for the **Figure position** parameter to be updated.

- **Channels** (properties apply to a particular channel)
  - **Visible**, when selected for a particular channels, causes the window to display the channel's data on the scope. When this option is deselected for a particular channel, the channel's data is hidden. Visibility can also be set for each channel in the **Line properties** panel of the parameter dialog box.
  - **Style** lets you choose from several line styles with which to display the channel's data. Line style can also be set for each channel in the **Line properties** panel of the parameter dialog box.
  - **Marker** lets you select a marker with which to display the individual data points in the channel. Marker style can also be set for each channel in the **Line properties** panel of the parameter dialog box.
  - **Color** lets you choose a color for the channel's line on the scope display. Colors can also be set for each channel in the **Line properties** panel of the parameter dialog box.

  See "Frame-Based Operation" for a description of how the block handles multichannel frame-based inputs.

Many of these options can also be accessed by right-clicking with the mouse anywhere on the scope display. The menu that pops up contains a combination of the options available in both the **Axes** and **Channels** menus. The right-click menu is very helpful when the scope is in zoomed mode, when the **Axes** and **Channels** menus are not visible.

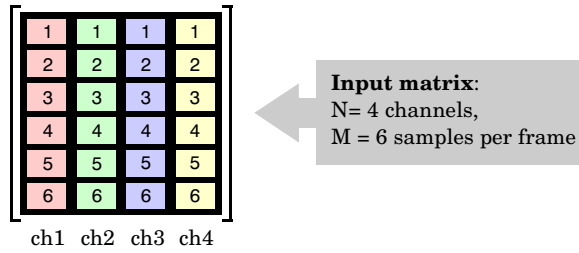
Frame-Based Operation

Inputs to the frame scope blocks are loosely considered to be *frames,* even though they need not contain consecutive time samples. For example, valid inputs include vectors of power spectral density data and histogram data. Indeed, the input can even be a sequence of sample vectors to plot against a

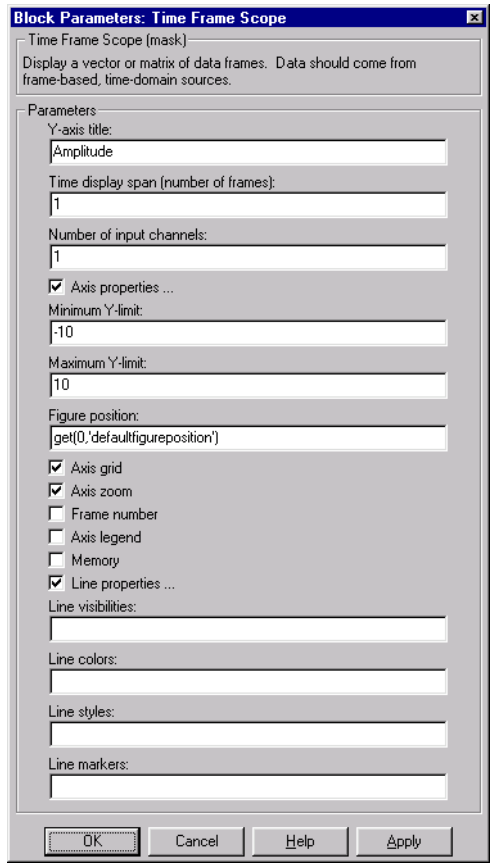
common axis. The blocks are referred to as frame-based because they process matrix inputs in the usual frame-based fashion.

The block assumes that an M-by-N matrix input contains a collection of N frames, where each frame contains M samples (time-domain, frequency-domain, etc.) from an independent signal. The **Number of channels** parameter specifies the number of independent channels (columns, N) in the matrix, and the block plots each channel separately. Different colors, marker and styles can be specified for the different channels. See above.

The illustration below shows a 6-by-4 frame matrix input.



Dialog Box



Y-Axis title

The text to be displayed to the left of the *y*-axis.

Time display span

The number of consecutive frames to display (horizontally) on the scope at any one time.

Number of input channels

The number of channels (columns) in the input matrix.

Axis properties Ⓣ

Select to expose the **Axis Properties** panel.

Minimum Y-limit

The minimum value of the *y*-axis.

Maximum Y-limit

The maximum value of the *y*-axis

Figure position

A 4-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.

Axis grid Ⓣ

Toggles the scope grid on and off.

Axis zoom Ⓣ

Resizes the scope to fill the window.

Frame number Ⓣ

Displays the number of the current frame in the input sequence, when selected with **Axis zoom** off. The frame number is not shown in the zoomed view.

Axis legend Ⓣ

Toggles the legend on and off.

Memory Ⓣ

Causes the window 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

Line properties Ⓣ

Select to expose the **Line Properties** panel.

Line visibilities Ⓣ

The visibility of the various channels' scope traces, `on` or `off`. Channels are separated by a pipe (`|`) symbol.

Line colors Ⓣ

The colors of the various channels' scope traces, in one of the `ColorSpec` formats. Channels are separated by a pipe (`|`) symbol.

Line styles Ⓣ

The line styles of the various channels' scope traces. Channels are separated by a pipe (|) symbol.

Line markers Ⓣ

The line markers of the various channels' scope traces. Channels are separated by a pipe (|) symbol

See Also

FFT Frame Scope
Frequency Frame Scope
Matrix Viewer
User-Defined Frame Scope

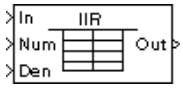
Purpose

Apply a variable IIR filter to the input.

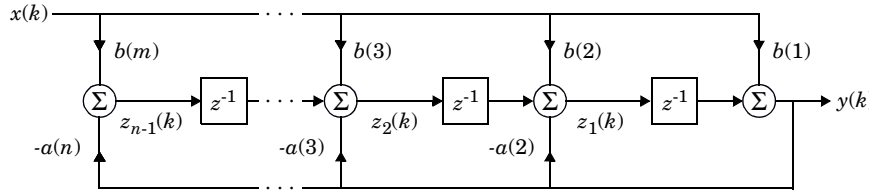
Library

Filter Realizations, in Filtering

Description



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 transposed direct-form II IIR filter to the top input (`In`), which must be a discrete-time signal.



This is a canonical form that has the minimum number of delay elements. The filter order is `max(m,n)-1`.

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.

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.

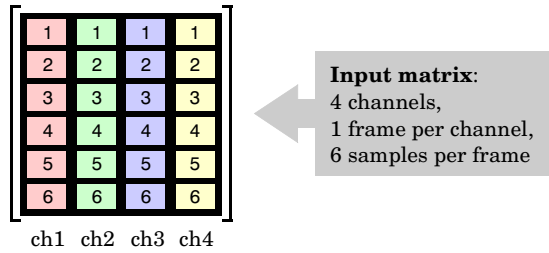
The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel over time.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block filters each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

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**

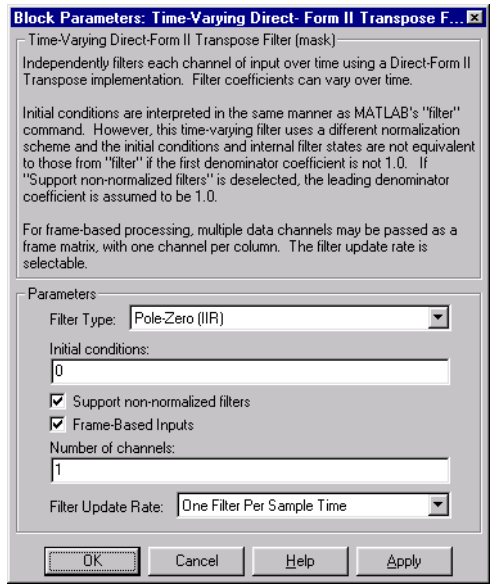
  The block updates the filter coefficients (from inputs `Num` and `Den`) for each individual scalar sample in the framed input. This means that each output sample could potentially be computed by a different filter (assuming that `Num` and `Den` inputs are updated frequently enough).

- **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.

Note If you expect to generate code for the Time-Varying Direct-Form II Transpose Filter block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Filter Type

The type of filter to apply: ARMA, MA, or AR. The `Num` and `Den` input ports are enabled or disabled appropriately.

Initial conditions

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

Support non-normalized filters

Normalizes the filter by a_1 when selected.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of channels (columns) in the input matrix.

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.

See Also

Discrete Filter (Simulink)
Filter Realization Wizard
Direct-Form II Transpose Filter
Time-Varying Lattice Filter
`filter` (MATLAB)

Time-Varying Lattice Filter

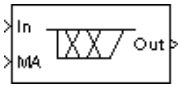
Purpose

Apply a variable lattice filter to the input.

Library

Filter Realizations, in Filtering

Description



The Time-Varying Lattice Filter block applies a moving average (MA) or autoregressive (AR) lattice filter to the top input (`In`), which must be a discrete-time signal. The filter reflection coefficients are specified by the input to the `MA` or `AR` port, and can vary with time.

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.

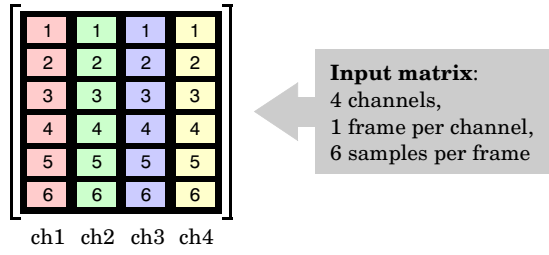
The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel over time.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix, and the block filters each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

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:

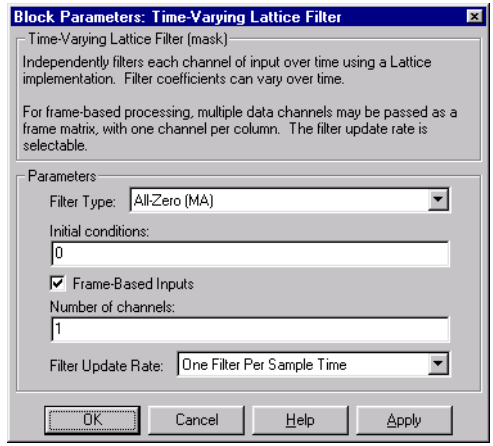
- **One filter per sample time**

  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.

Note If you expect to generate code for the Time-Varying Lattice Filter block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

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.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of channels (columns) in the input matrix.

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.

See Also

Discrete Filter (Simulink)
Direct-Form II Transpose Filter
Filter Realization Wizard
Time-Varying Direct-Form II Transpose Filter
`filter` (MATLAB)

Toeplitz

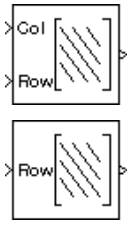
Purpose

Generate a matrix with Toeplitz symmetry.

Library

Matrix Functions, in Math Functions

Description



The Toeplitz block generates a Toeplitz matrix from vectors defining the first column and first row. The top input (`Col`) is the vector to be placed in the first column of the matrix. The bottom input (`Row`) is the vector to be placed in the first row of the matrix.

```
y = toeplitz(Col,Row)  % equivalent MATLAB code
```

Other elements, $r_{i,j}$, obey the relationship

$$r_{i,j} = r_{(i-1),(j-1)}$$

The r_{11} element is inherited from `Col`, and the two vectors must be the same length. For example, the following inputs

```
Col = [1 2 3 4 5]
Row = [7 7 3 3 2]
```

produce the Toeplitz matrix

$$\begin{bmatrix} 1 & 7 & 3 & 3 & 2 \\ 2 & 1 & 7 & 3 & 3 \\ 3 & 2 & 1 & 7 & 3 \\ 4 & 3 & 2 & 1 & 7 \\ 5 & 4 & 3 & 2 & 1 \end{bmatrix}$$

When the **Symmetric** check box is selected, the block generates a symmetric (Hermitian) Toeplitz matrix from a single vector input (`Row`) defining both the first row and first column of the matrix:

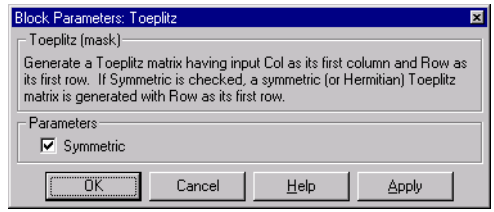
```
y = toeplitz(Row)      % equivalent MATLAB code
```

For example, the Toeplitz matrix generated from the input vector `[1 2 3 4]` is

$$\begin{bmatrix} 1 & 2 & 3 & 4 \\ 2 & 1 & 2 & 3 \\ 3 & 2 & 1 & 2 \\ 4 & 3 & 2 & 1 \end{bmatrix}$$

Note If you expect to generate code for the Toeplitz block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Symmetric

Selects a symmetric Toeplitz matrix output.

See Also

Constant Diagonal Matrix
Matrix Constant
`toeplitz` (MATLAB)

To Wave Device

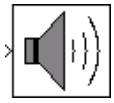
Purpose

Send audio data to a standard audio device in real-time (Windows 95/98/NT only).

Library

DSP Sinks

Description



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. (Models that contain both this block and the From Wave Device block require a *duplex-capable* sound card.) The data is sent to the hardware in uncompressed PCM (pulse code modulation) format, and should typically be sampled at one of the standard Windows audio device rates: 8000, 11025, 22050, or 44100 Hz. Some hardware may support other rates in addition to these.

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 target. In cases when the default sound device is *not* the desired output device, deselect **Use default audio device**, and enter the desired device identification number in the **Audio device ID** parameter. The device ID is an integer value that the block associates with the sound device. A 3-device system, for example, has device ID numbers of `1`, `2`, and `3`.

The input to the block, u, can be a vector containing a frame of audio data from a mono signal, or a 2-column matrix containing one frame of audio data from each channel of a stereo signal. (The **Stereo** check box should be selected in this case.)

```
sound(u,Fs,bits)       % equivalent MATLAB code
```

The amplitude of the input should be in the range ±1. Values outside this range are clipped to the nearest allowable value.

The **Sample Width (bits)** parameter specifies the number of bits used to represent the signal samples sent to the audio device. Two 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

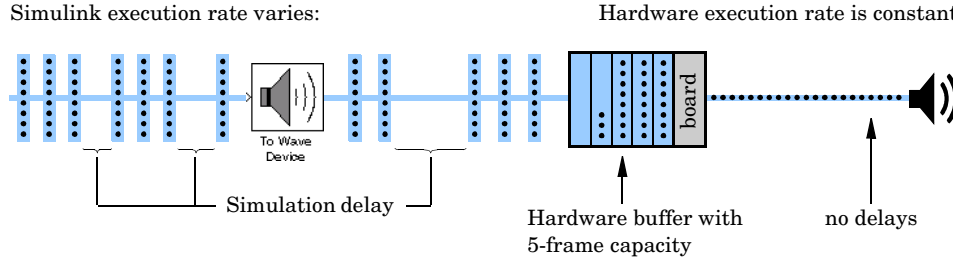
The 16-bit sample width setting requires more memory but yields better fidelity for double-precision inputs.

Buffering

Because the audio device generates real-time audio output, Simulink must maintain a continuous flow of data to the device throughout the 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 from within Simulink rather than as generated code. (Simulink 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 is accessible to the hardware on demand.

At the start of the simulation, the To Wave Device block writes T_d seconds worth of signal data to the device (hardware) buffer, where T_d is specified by the **Initial output delay** parameter. When this initial data is loaded into the buffer, the audio device begins processing the buffered data, and continues at a constant rate until the buffer empties. The size of the buffer, T_b, is specified by the **Queue duration** parameter. As the audio device reads data from the *front* of the buffer, the To Wave Device block continues appending inputs to the back of the buffer at the rate they are received.

The following figure shows an audio signal with 8 samples per frame. The buffer of the sound board has a five-frame capacity, not fully used at the instant shown. (If the signal sample rate was 8kHz, for instance, this small buffer could hold approximately 0.005 seconds of data.)



If the simulation throughput rate is higher than the hardware throughput rate, the buffer remains at a constant level throughout the simulation. If necessary, the To Wave Device block buffers inputs until space becomes available in the hardware buffer (data is not thrown away). More typically, the hardware throughput rate is higher than the simulation throughput rate, and the buffer tends to empty over the duration of the simulation.

Under normal operation, an empty buffer indicates that the simulation is finished, and the entire length of the audio signal has been processed. However, if the buffer size is too small in relation to the simulation throughput rate, the buffer may also empty before the entire length of signal is processed. This usually results in a device error or undesired device output.

When the device fails to process the entire signal length because the buffer prematurely empties, you can choose to either increase the buffer size or the simulation throughput rate.

- *Increase the buffer size*. The **Queue duration** parameter specifies the length of signal, T_b (in real-time seconds), to buffer to the audio device during the simulation. The number of frames buffered is approximately

  $$\frac{T_b F_s}{N}$$

  where F_s is the sample rate of the signal and N is the number of samples per frame. The optimal buffer size for a given signal depends on the signal length, the frame size, and the speed of the simulation. The maximum number of frames that can be buffered is 1024.
- *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 scalar 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. (Note that increasing the

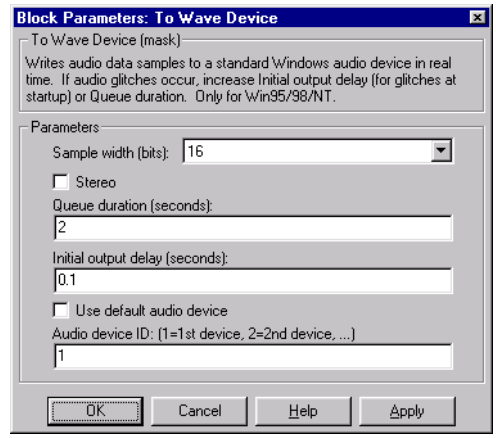
audio signal frame size does not affect the number of samples buffered to the hardware since the **Queue duration** is specified in seconds.)

- Generate executable code with RTW. Native code runs much faster than Simulink, and should provide rates adequate for real-time audio processing.

Audio problems at startup can often be corrected by entering a larger value for the **Initial output delay** parameter, which allows a greater portion of the signal to be preloaded into the hardware buffer. A value of 0 for the **Initial output delay** parameter specifies the smallest possible initial delay, which is one frame.

More general ways to improve throughput rates include simplifying the model, and running the simulation on a faster PC processor. See *Using Simulink* and "Increasing Performance" in Chapter 2 of this book for other ideas on improving simulation performance.

Dialog Box



Sample width (bits)

The number of bits used to represent each signal sample.

Stereo

Specifies stereo (two-channel) inputs when checked, mono (one-channel) inputs when unchecked.

Queue duration (seconds)

The length of signal (in seconds) to buffer to the hardware at the start of the simulation.

Initial output delay (seconds)

The amount of time by which to delay the initial output to the audio device. A value of `0` specifies the smallest possible initial delay, a single frame.

Use default audio device

Directs audio output to the system's default audio device when selected. Deselect to enable the **Audio device ID** parameter and manually enter a device ID number.

Audio device ID

The number of the audio device to receive the audio output. In a system with several audio devices installed, a value of `1` selects the first audio card, a value of `2` selects the second audio card, and so on. Select **Use default audio device** if the system has only a single audio card installed.

See Also

From Wave Device
To Wave File
`sound` (MATLAB)

Purpose

Write audio data to file in the Microsoft Wave (`.wav`) format (Windows 95/98/NT only).

Library

DSP Sinks

Description

The To Wave File block writes audio data to a Microsoft Wave (`.wav`) file in the uncompressed PCM (pulse code modulation) 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 be a vector containing a frame of audio data from a mono signal, or a 2-column matrix containing one frame of audio data from each channel of a stereo signal. (The **Stereo** check box should be selected in this case.)

```
wavwrite(u,Fs,bits,'filename') % equivalent MATLAB code
```

The amplitude of the input should be in the range ±1. Values outside this range are clipped to the nearest allowable value.

The **Sample rate** parameter specifies the value, F_s, to record in the Wave file as the data's sample rate; a value of `-1` instructs the block to use the sample rate of the input as this value.

The **Sample Width (bits)** parameter specifies the number of bits used to represent the signal samples in the file. Two settings are available:

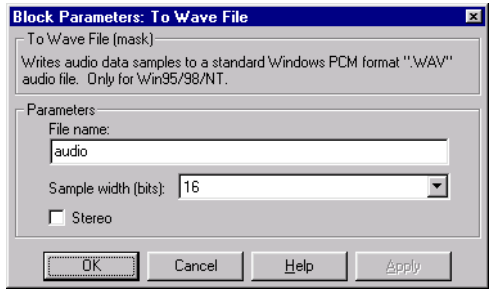
- **8** – allocates 8 bits to each sample, allowing a resolution of 256 levels
- **16** – allocates 16 bits to each sample, allowing a resolution of 65536 levels

The 16-bit sample width setting requires more memory but yields better fidelity for double-precision inputs.

The **File name** parameter can specify an absolute or relative path to the file. You do not need to specify the `.wav` extension.

To Wave File

Dialog Box



File name

The path and name of the file to write. Paths can be relative or absolute.

Sample width (bits)

The number of bits used to represent each signal sample.

Stereo

Specifies stereo (two-channel) inputs when checked, mono (one-channel) inputs when unchecked.

See Also

From Wave File
Signal To Workspace
To Wave Device
`wavwrite` (MATLAB)

Purpose

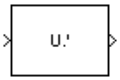
Compute the transpose of a matrix.

Library

Matrix Functions, in Math Functions

Description

The Matrix Transpose block transposes the M-by-N input matrix such that the output matrix has size N-by-M.



When the **Hermitian** check box is selected, the block performs the Hermitian (conjugate) transpose:

```
y = u'          % equivalent MATLAB code
```

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \end{bmatrix} \xrightarrow{\text{u'}} \begin{bmatrix} u_{11}^* & u_{21}^* \\ u_{12}^* & u_{22}^* \\ u_{13}^* & u_{23}^* \end{bmatrix}$$

When the **Hermitian** check box is not selected, the block performs the non-conjugate transpose:

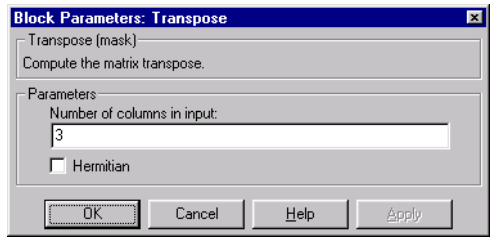
```
y = u.'         % equivalent MATLAB code
```

$$\begin{bmatrix} u_{11} & u_{12} & u_{13} \\ u_{21} & u_{22} & u_{23} \end{bmatrix} \xrightarrow{\text{u.'}} \begin{bmatrix} u_{11} & u_{21} \\ u_{12} & u_{22} \\ u_{13} & u_{23} \end{bmatrix}$$

Note If you expect to generate code for the Matrix Transpose block using the Real-Time Workshop, you should ensure that matrix inputs are contiguous in memory. See the Contiguous Copy block for more information.

Transpose

Dialog Box



Number of columns in input

The number of columns in the input matrix.

Hermitian Ⓣ

Selects the non-conjugate transpose when checked. This parameter is not tunable in Simulink's external mode.

See Also

Permute Matrix
Reshape
Submatrix

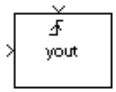
Purpose

Write the input matrix to the workspace when triggered.

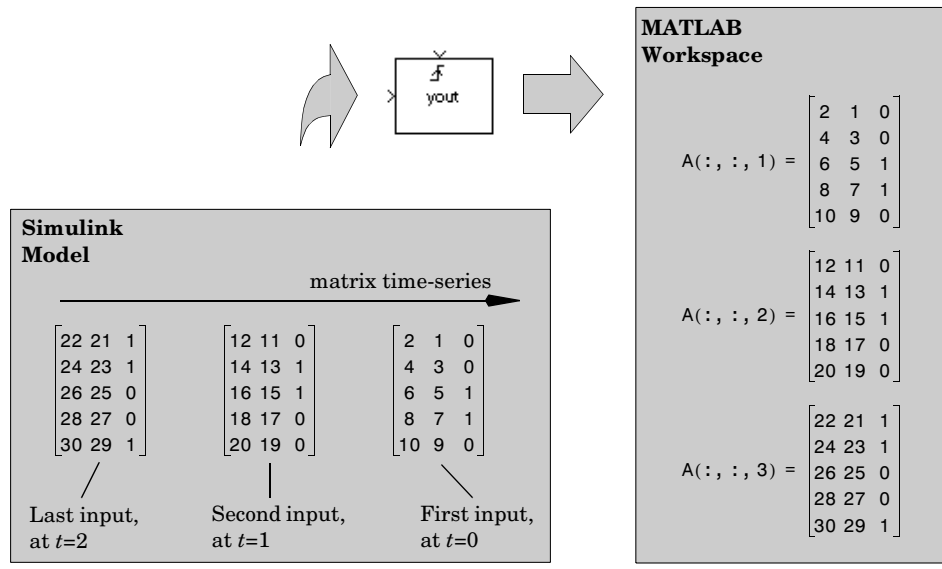
Library

DSP Sinks

Description



The Triggered Matrix To Workspace block writes a three-dimensional array, `A`, to the workspace, where `A` contains the acquired samples of a matrix input. At the end of the simulation the block writes every Dth input sample (a matrix) to a *page* (a two-dimensional slice) of the specified three-dimensional array, where D is specified by the block's **Decimation factor**. The block writes the first input to the first page of the array, `A(:,:,1)`, and continues adding pages until it writes the last input matrix to the last array page, `A(:,:,end)`.



The block acquires and buffers a single sample from input 1 whenever it is triggered by the control signal at input 2. At all other times, the block ignores input 1. The triggering event at input 2 is specified by the **Trigger type** pop-up menu, and can be one of the following:

- **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.

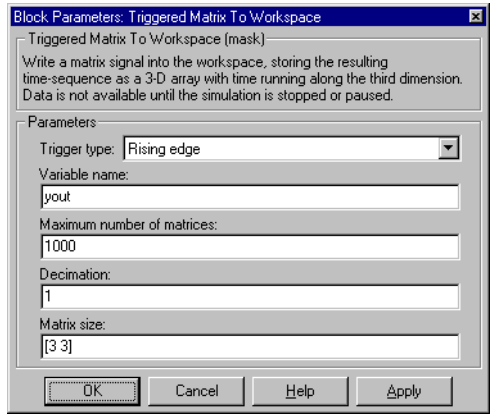
- **Falling edge** triggers execution of the block when the trigger input falls from a positive value to zero or a negative value, or from zero to a negative value.
- **Either edge** triggers execution of the block when either a rising or falling edge (as described above) occurs.

To save a record of the simulation sample times corresponding to each page of the three-dimensional array, check the **Time** box in the **Save to workspace** panel of the **Simulation Parameters** dialog. You can access these parameters by selecting **Parameters** from the **Simulation** menu, and clicking on the **Workspace I/O** tab.

The nontriggered version of this block is Matrix To Workspace.

Note The Triggered Matrix To Workspace block does not support real-time data logging when used with the Real-Time Workshop. You should not use this block if you expect to generate real-time code from your model.

Dialog Box



Trigger type

The type of event that triggers the block's execution.

Variable name

The name of the array to create in the workspace.

Maximum number of matrices

The maximum number of array pages to create in the workspace, P. If the block samples more than P times, it stores only the last P matrices.

Decimation

The factor by which to decimate the input samples, D.

Matrix size

The dimensions of the input matrix, in the form `[rows columns]`.

See Also

Matrix From Workspace
Matrix To Workspace
Triggered Signal To Workspace

Triggered Shift Register

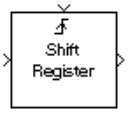
Purpose

Buffer a sequence of inputs into a frame-based output.

Library

Buffers, in General DSP

Description



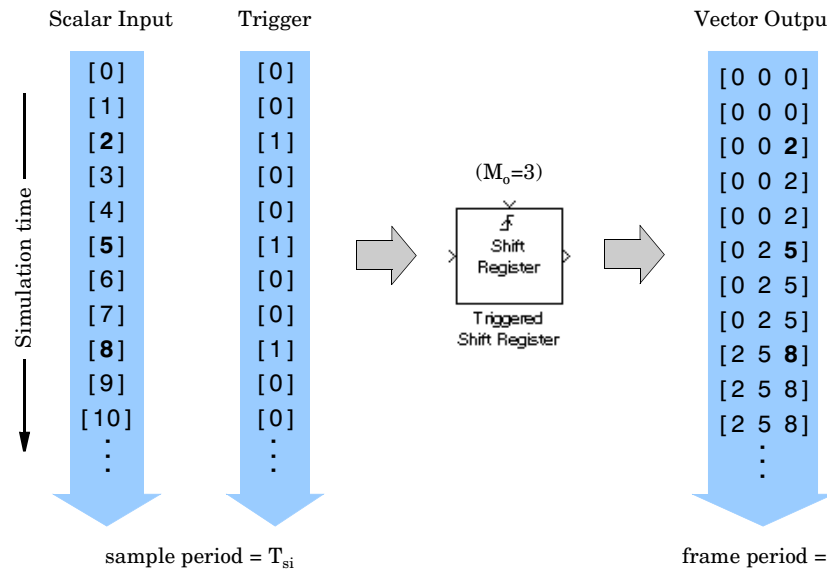
The Triggered Shift Register block acquires a collection of M_o input samples into a frame, where M_o is specified by the **Register size** parameter. The block buffers a single sample from input 1 whenever it is triggered by the control signal at input 2 (⨍). The newly acquired input sample is appended to the output frame (in the same simulation step) 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.

The triggering event at input 2 is specified by the **Trigger type** pop-up menu, and can be one of the following:

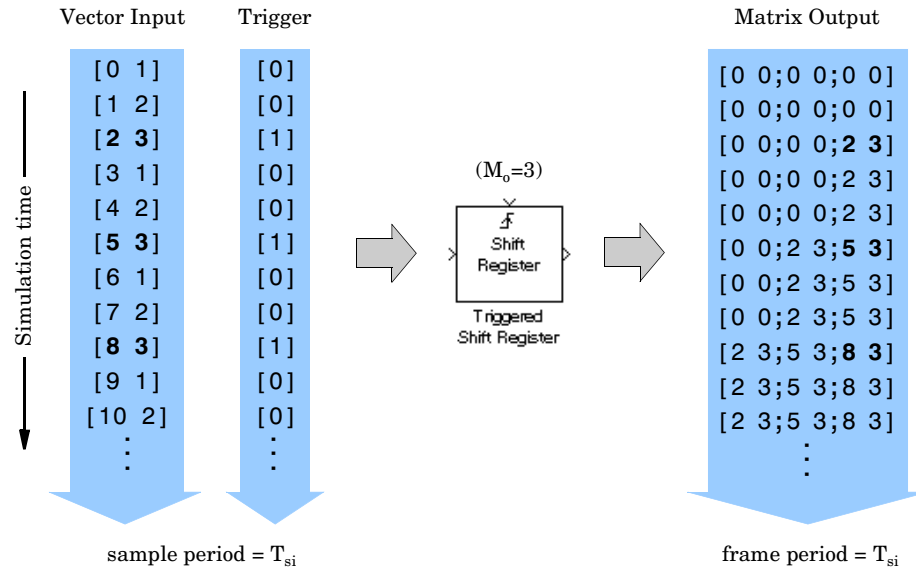
- **Rising edge** triggers execution of the block when the trigger input rises from a negative value to zero or a positive value, or from zero to a positive value.
- **Falling edge** triggers execution of the block when the trigger input falls from a positive value to zero or a negative value, or from zero to a negative value.
- **Either edge** triggers execution of the block when either a rising or falling edge (as described above) occurs.

Note that the Triggered Shift Register block has *direct feedthrough,* 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, $T_{fo}=T_{si}$.

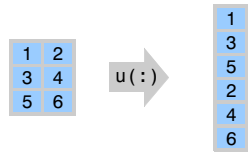
Scalar Inputs. Scalar inputs are buffered into a frame vector of length M_o, where M_o is specified by the **Register size** parameter.



Vector Inputs. Length-N sample vector inputs are buffered into a M_0-by-N matrix, where M_0 is specified by the **Register size** parameter.



Matrix Inputs. An M-by-N matrix input is treated as a single vector with M∗N elements (channels). In other words, the matrix input `u` is reshaped to the vector input `u(:)`.

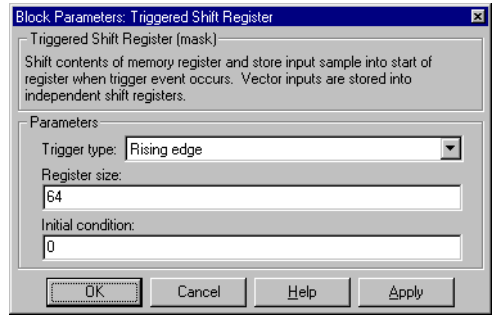


Initial Conditions

The Triggered Shift Register 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). If 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. If 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.

Note If you expect to generate code for the Triggered Shift Register block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Trigger type

The type of event that triggers the block's execution.

Register 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.

See Also

Buffer
Shift Register
Unbuffer

Triggered Signal From Workspace

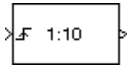
Purpose

Acquire and output a workspace signal when triggered.

Library

DSP Sources

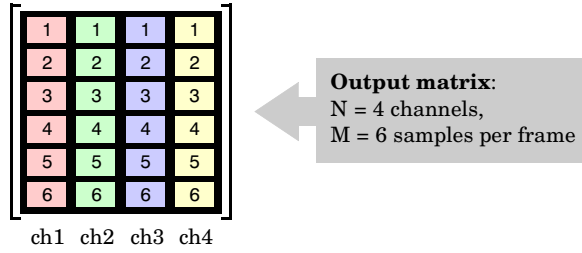
Description



The Triggered Signal From Workspace block references a specified vector or matrix in the MATLAB workspace to generate scalar, vector, or matrix output. Each element of a workspace vector, and each *row* of a workspace matrix is considered to be individual sample. (Matrix columns represent independent channels.)

The block acquires and outputs a frame of M new samples from the specified workspace variable each time it is triggered by the control signal at the input port. The output frame size, M, is specified in the block dialog by the **Samples per frame** parameter. When the **Samples per frame** parameter is set to `1`, the block acquires and outputs a single sample (a vector element or matrix row) each time it is triggered.

For a W-by-N workspace matrix, the output size is M-by-N, with each column representing a distinct signal channel. For example, if the **Signal** parameter references a W-by-4 workspace matrix and the **Samples per frame** parameter is set to `6`, the block generates a sequence of 6-by-4 matrices, as illustrated below. Each matrix contains six consecutive samples from four distinct channels.



Note, however, that a 1-by-N matrix (row vector) is treated as a N-by-1 matrix (column vector), and acquired element-wise rather than row-wise.

Trigger Event

The triggering event at the input is specified by the **Trigger type** pop-up menu, and can be one of the following:

- **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

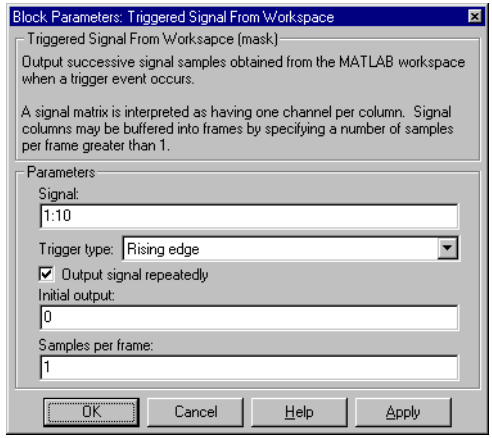
The **Initial output** parameter specifies the output of the block from the start of the simulation until the first trigger arrives. Subsequently, the block holds the output value constant (at its last value) between trigger events. For vector signals, the **Initial output** parameter value can be a vector of length M or a scalar to copy across the M elements of the initial frame. For matrix signals, the **Initial output** parameter value can be a vector of length N to copy across the M rows of the initial output, or a scalar to copy across the M∗N elements of the initial output.

When the block has output all of the signal samples, it can start again with the beginning of the signal or simply output zeros (or zero-vectors or zero-matrices, as appropriate) until the end of the simulation.

- If the **Output signal repeatedly** check box is selected, when the block outputs the last frame of the signal, it returns to the beginning of the signal to repeat the process. If there are not enough samples at the end of the signal to fill the final frame, the last frame is zero-padded as necessary. This ensures that the block's output for each cycle is identical (e.g., the third frame of one cycle contains the same data as the third frame of any other cycle).
- If the **Output signal repeatedly** check box is not selected, when the block outputs the last frame of the signal, it continues outputting frames of zeros for the duration of the simulation.

Triggered Signal From Workspace

Dialog Box



Signal

The name of the workspace vector or matrix from which to acquire data, or a valid MATLAB expression.

Trigger type

The type of event that triggers the block's execution. This parameter is not tunable in Simulink's external mode.

Output signal repeatedly

Specifies continuous (repeating) operation or one-time operation.

Initial output

The value to output until the first trigger event is received. This parameter is not tunable in Simulink's external mode.

Samples per frame

The number of input samples (vector elements or matrix rows) to acquire into each output frame, M.

See Also

From Wave Device
From Wave File
Matrix From Workspace
Sine Wave
Signal From Workspace
Triggered Signal To Workspace

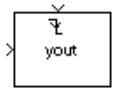
Purpose

Write the input sample to the workspace when triggered.

Library

DSP Sinks

Description



The Triggered Signal To Workspace block creates a matrix variable in the workspace, where it stores the acquired inputs at the end of a simulation. Each row of the workspace matrix represents an input sample, with the most recent sample occupying the last row. The maximum size of this variable is limited to the size specified by the **Maximum number of rows** (P) parameter. (If the simulation progresses long enough for the block to trigger more than P times, it stores only the last P samples.) The **Decimation factor**, D, allows you to store only every Dth sample.

The block acquires and buffers a single sample from input 1 whenever it is triggered by the control signal at input 2. At all other times, the block ignores input 1. The triggering event at input 2 is specified by the **Trigger type** pop-up menu, and can be one of the following:

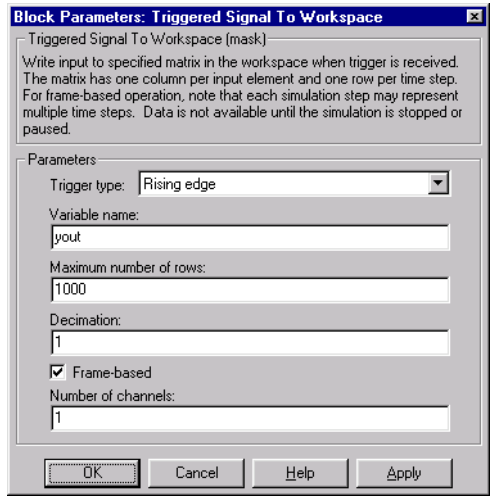
- **Rising edge** triggers execution of the block when the trigger input rises from a negative value to zero or a positive value, or from zero to a positive value.
- **Falling edge** triggers execution of the block when the trigger input falls from a positive value to zero or a negative value, or from 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, check the **Time** box in the **Save to workspace** parameters list of the **Simulation Parameters** dialog. You can access these parameters by selecting **Parameters** from the **Simulation** menu, and clicking on the **Workspace I/O** tab.

The nontriggered version of this block is Signal To Workspace.

Triggered Signal To Workspace

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. The default is `1000` rows.

Decimation

The decimation factor, D. The default is `1`.

Frame-based

Selects frame-based operation.

Number of channels

For frame based operation, the number of channels (columns) in the input matrix, N.

See Also

Signal From Workspace
Signal To Workspace
Triggered Matrix To Workspace

Purpose

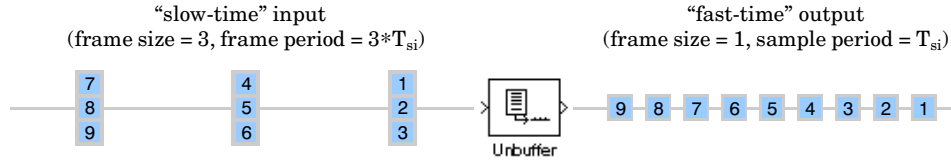
Unbuffer a frame input to a sequence of scalar outputs.

Library

Buffers, in General DSP

Description

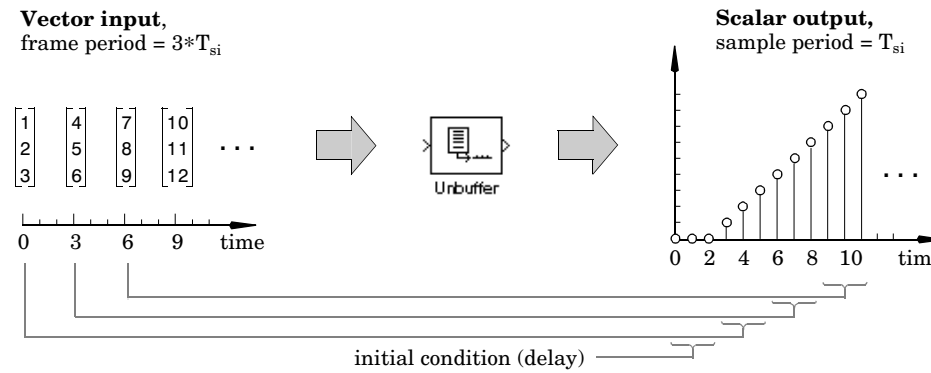
The Unbuffer block unbuffers the input frame into a sequence of scalar outputs. Multichannel (matrix) inputs are unbuffered into sample vectors, with each matrix *row* being output in sequence. The sample-based output generally has a faster rate than the frame-based input.



To rebuffer frame-based inputs to a larger or smaller frame size, use the Rebuffer block.

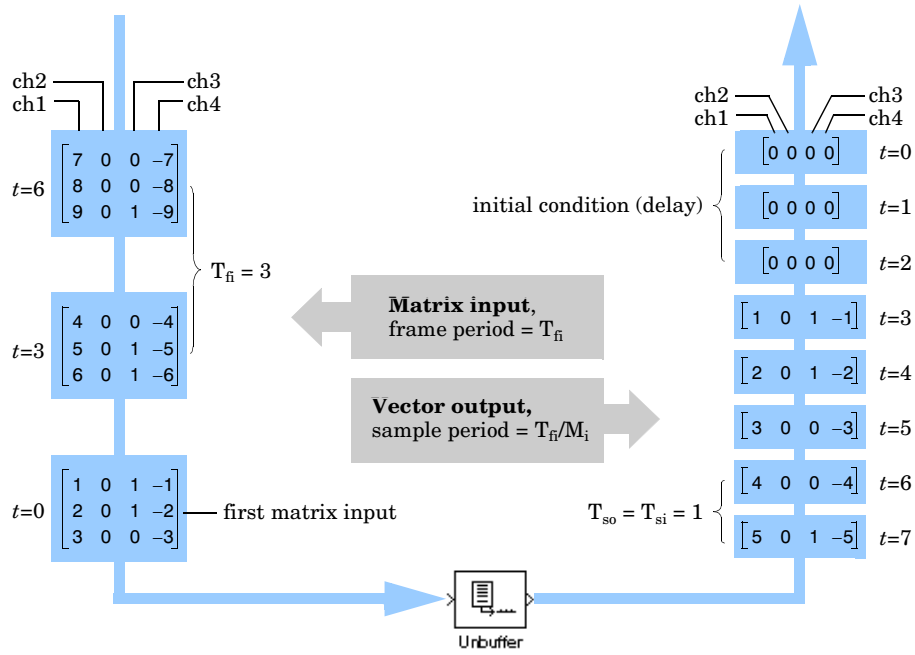
The block adjusts the output rate so that the *sequence sample period* (i.e., the sample-to-sample interval) is the same at both the input and output, $T_{so}=T_{si}$. Therefore, the output sample period for an input of frame size M_i and frame period T_{fi} is T_{fi}/M_i, which represents a *rate* M_i times higher than the input frame rate.

Vector Inputs. Vector inputs are unbuffered to a scalar sequence.



The **Number of channels** parameter, N, should typically be 1 for vector inputs, indicating that the input represents a single channel.

Matrix Inputs. The block's operation for vector inputs extends naturally to matrix inputs. A M_i-by-N matrix input represents N frames, each containing M_i sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals (columns, N) in the matrix. Matrix inputs are unbuffered *row-wise* so that each matrix row becomes an independent time-sample in the vector output.



Initial Conditions

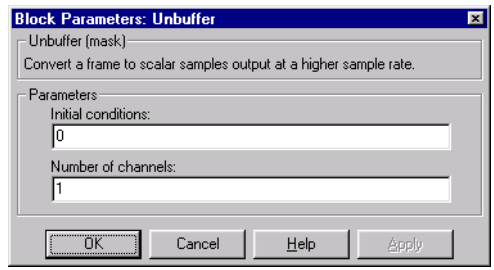
The Unbuffer block's buffer is initialized with the value specified by the **Initial condition** parameter, and the block begins unbuffering this frame at the start of the simulation. Inputs to the block are therefore delayed by one buffer length (M_i samples, or T_{fi} seconds).

If the block's output is a scalar (single channel), the **Initial condition** can be a scalar to be repeated at the output for the first M_i sample times, or a vector containing M_i samples to be output sequentially for the first M_i sample times. If the block's output is a vector (N channels), the **Initial condition** can be a scalar value to be repeated across all elements of the initial output(s), a vector

containing M_i samples to be repeated for all channels and output sequentially for the first M_i sample times, or an M_i-by-N matrix whose rows are output in sequence for the first M_i sample times.

Note If you expect to generate code for the Unbuffer block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Initial conditions

The value of the block's initial output, a scalar, vector, or matrix.

Number of channels

The number of columns in the input, N. Use `1` for a vector input containing consecutive time-samples.

See Also

Buffer
Partial Unbuffer
Rebuffer

Unwrap

Purpose

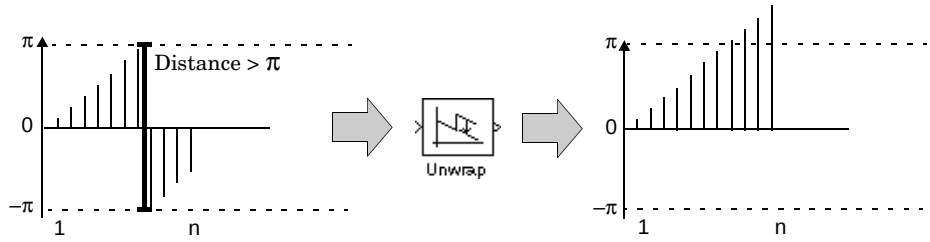
Unwrap a vector of radian phase angles.

Library

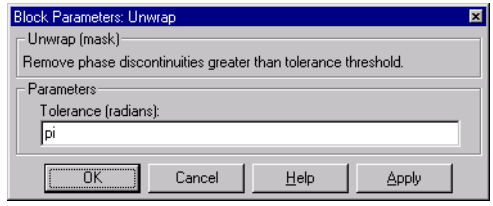
Vector Functions, in Math Functions

Description

The Unwrap block unwraps radian phases in the input vector by replacing absolute jumps greater than the specified **Tolerance** with their 2π complement.



Dialog Box



Tolerance Ⓣ

The jump size. The default is set to π to avoid altering legitimate signal features. To jump more readily, set the **Tolerance** to a value slightly less than π. This parameter is not tunable in Simulink's external mode.

See Also

`unwrap` (MATLAB)

Purpose

Resample an input at a higher rate by inserting zeros.

Library

Signal Operations, in General DSP

Description

The Upsample block resamples the discrete input at a rate L times faster than the input sample rate by inserting L-1 zeros between consecutive samples, where L is the integer **Upsample factor**. The **Sample offset** delays the output samples by an integer number of sample periods D (D<L), so that any of the L possible output phases can be selected.

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

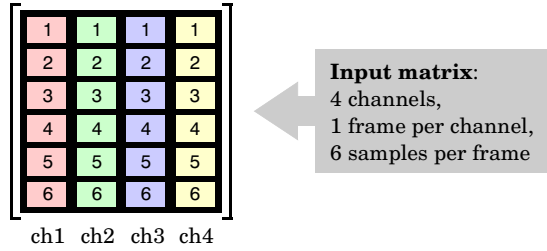
Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block upsamples each channel over time. The output sample rate is L times higher than the input sample rate, and the input and output sizes are identical.

In sample-based mode, the **Initial condition** can be a vector of length N (length M∗N for a matrix input) containing one value for each channel, or a scalar to be applied to all signal channels. This value is output at t=D.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:



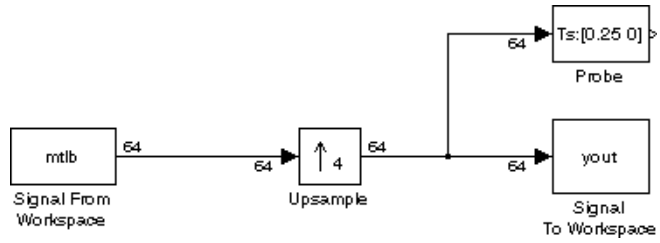
The **Number of channels** parameter specifies the number of independent channels (columns, N) in the matrix. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

In frame-based operation, the block upsamples each channel independently by inserting L rows of zeros between each row in the original input matrix. The **Framing** parameter determines how the block adjusts the rate at the output. 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, but the input and output frame sizes are equal.

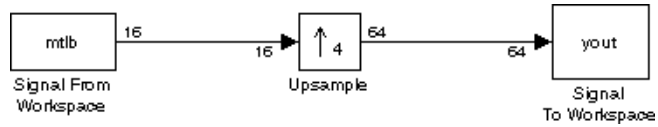
  The example below shows a single-channel input with a frame period of 1 second (**Sample time** = `1/64` and **Samples per frame** = `64` in the Signal From Workspace block) being upsampled by a factor of 4 to a frame period of 0.25 seconds. The input and output frame sizes are identical.



- **Maintain input frame rate**

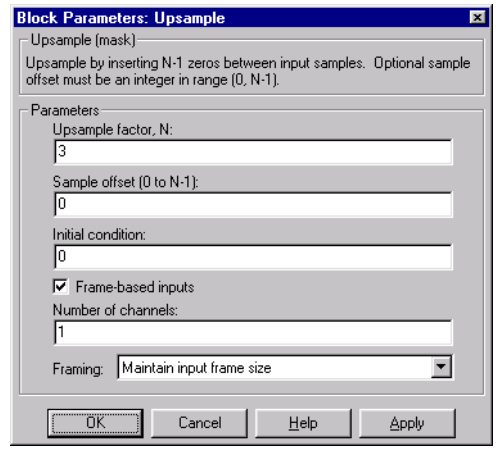
  The block generates the output at the faster (upsampled) rate by using a proportionally larger frame size than the input. For upsampling by a factor of L, the output frame size is L times larger than the input frame size, but the input and output frame rates are equal.

  The example 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 rates are identical.



In frame-based mode, the **Initial condition** can be an M-by-N matrix representing the initial input, or a scalar to be repeated across all elements of the M-by-N matrix. The first row of the matrix is output at *t*=D, where D is the **Sample offset**.

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 that the block is initialized with; a scalar or vector in sample-based mode, or a scalar or matrix in frame-based mode. This value (first row in frame-based mode) is output at *t*=D.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of columns (frames) in the input matrix.

Framing

For frame-based operation, the method by which to implement the upsampling: increase the output sample rate, or increase the output frame size.

See Also

Downsample
FIR Interpolation
FIR Rate Conversion
Repeat

Purpose

Display frame-based data.

Library

DSP Sinks

Description

The User-Defined Frame Scope block is similar to the Time Frame Scope, but is not limited to plotting time-domain data. For a complete discussion of this block's axis properties, line properties, scope window, and frame-based operation, see the Time Frame Scope block reference.

The block does not make any assumptions about the nature of the data in the input frame. In particular, it does not assume that it is time-domain or frequency-domain data. The dialog box parameters give you complete freedom to plot the data in the most appropriate manner.

The scope updates the display for each new input frame. At any one time, the number of sequential frames displayed on the scope specified by the **Horizontal display span** parameter, S. Setting S=1 plots the current input frame's data across the entire width of the scope. Setting S to a larger number allows you to see a broader section of the input by fitting more frames of data into the display region. A single frame is the smallest unit that can be displayed, so S cannot be less than 1.

In order to correctly scale the horizontal axis, the block needs to know the spacing of the data in the input. This is specified by the **Increment per sample in input frame** parameter, I_s, which represents the numerical interval between adjacent *x*-axis points corresponding to the input data. For example, an input signal sampled at 500 Hz has an increment per sample of 0.002 seconds. The actual units of this interval (seconds, meters, Volts, etc.) are not needed for axis scaling.

A value of `-1` for this parameter instructs the block to compute the horizontal interval between samples in the input frame from the frame period of the input. For example, if the input frame period is 1, and there are 64 samples per input frame, the interval between samples is computed to be 1/64. Allowing the block to auto-compute the interval this way is usually only valid if:

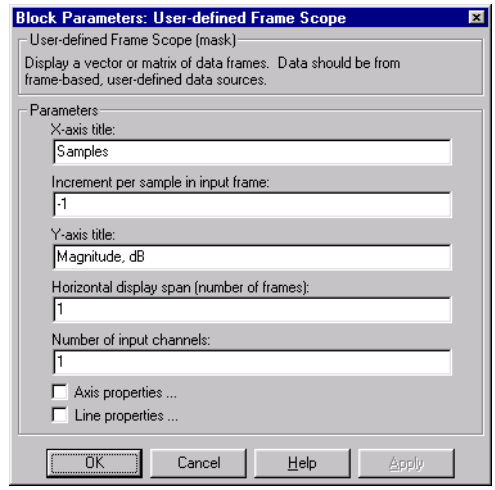
- The input is a nonoverlapping time-series (i.e., the *x*-axis on the scope represents time)

  and

- 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 other cases, the frame rate and frame size do not provide enough information for the block to correctly scale the *x*-axis, and you should specify the appropriate value for the **Increment per sample in input frame** parameter. The range of the horizontal axis is $[0,M*I_s*S]$, where M is the number of samples in each consecutive input frame.

Dialog Box



X-Axis title Ⓣ

The text to be displayed below the *x*-axis.

Increment per sample in input frame Ⓣ

The numerical interval, I_s, between adjacent *x*-axis points corresponding to the input data.

Y-Axis title Ⓣ

The text to be displayed to the left of the *y*-axis.

Horizontal display span

The number of consecutive input frames, S, to display (horizontally) on the scope at any one time.

Number of input channels

The number of channels (columns) in the input matrix, N.

Axis properties Ⓣ

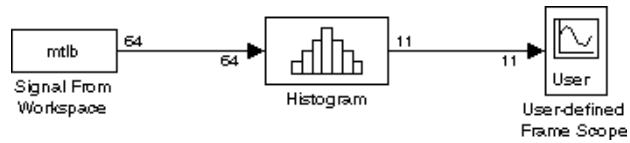
Select to expose the **Axis Properties** panel. See Time Frame Scope for more information.

Line properties Ⓣ

Select to expose the **Line Properties** panel. See Time Frame Scope for more information.

Examples

One possible application of the User-Defined Frame Scope is plotting the histogram distribution of a signal. For example, the simple model below lets you view a histogram computed over sections of the `mtlb` signal.



The block parameters are set as follows:

- Signal From Workspace
  - **Signal** = `mtlb`
  - **Sample time** = `0.1`
  - **Samples per frame** = `64`
- Histogram
  - **Minimum value of input** = `-3`
  - **Maximum value of input** = `3.1`
  - **Number of bins** = `11`
  - **Normalized** ☑
  - **Running histogram** ☐

- User-Defined Frame Scope
  - **X-axis value** = `Value`
  - **Increment per sample in input frame** = `(3.1+3)/11`
  - **Y-axis title** = `Normalized # of Occurrences`
  - **Horizontal display span** = `1`
  - **Number of input channels** = `1`
  - **Memory** ☑

The value of the **Increment per sample in input frame** parameter is the distance between histogram bins,

$$\Delta = \frac{B_M - B_m}{n}$$

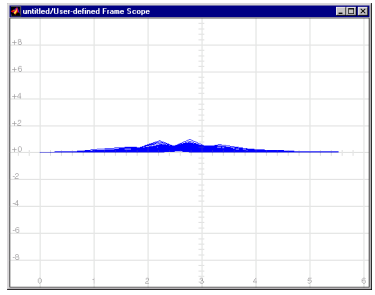
where B_M is the **Maximum value of input**, B_m is the **Minimum value of input**, and n is the **Number of bins**. See Histogram for more about these parameters.

To run the simulation, load the signal into the workspace by typing

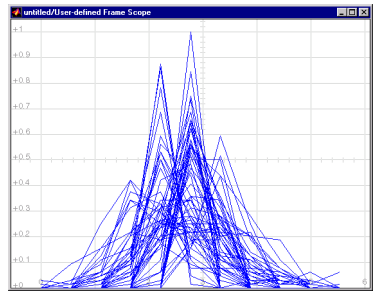
```
load mtlb
```

and set the **Stop time** parameter in the **Simulation Parameters** dialog box to `1000` (select **Parameters** from the **Simulation** menu).

Start the simulation. After a moment you should see the scope window below appear and update until the end of the simulation.



Right-click with the mouse and select **Autoscale** to better fit the data to the scope window. Run the simulation again. You should see the scope below. Deselect **Axis zoom** from the right-click menu to see the axis titles.



Note that although the spacing of the bins is correct, the bin values are not. This is because the first bin is placed at zero by default. The actual bin centers are located at

$$B_m + \left(k + \frac{1}{2}\right)\Delta\ , \qquad k\ =\ 0, 1, 2, \ldots, n-1$$

or

```
[-2.72,-2.17,-1.61,-1.06,-0.50,0.05,0.60,1.16,1.71,2.27,2.82]
```

See Also

FFT Frame Scope
Frequency Frame Scope
Time Frame Scope

Variable Fractional Delay

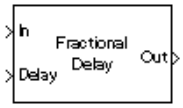
Purpose

Delay an input by a time-varying fractional number of sample periods.

Library

Signal Operations, in General DSP

Description



The Variable Fractional Delay block delays the discrete-time input by a variable, possibly noninteger, number of sample intervals. This block differs from the Variable Integer Delay block in the following two ways.

Variable Fractional Delay	Variable Integer Delay
Allows noninteger delays	Does not allow noninteger delays
Accepts only a vector delay, containing a unique delay value for each sample in the input frame (in frame-based mode)	Accepts a scalar delay by which to uniformly delay all samples in the input frame, or a vector containing a unique delay for each sample in an input frame (in frame-based mode)

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block applies the delay at the `Delay` port identically to each channel.

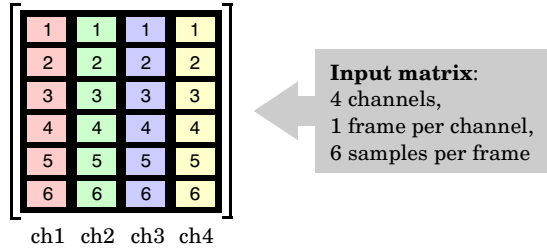
Like the Variable Integer Delay block, the Variable Fractional Delay block stores the D+1 most recent samples received at the `In` port for each channel, where D is the **Maximum delay in samples**. The input to the `Delay` port, v, is a floating-point value in the range $0 \le v \le D$ that specifies the number of sample intervals to delay every channel of the current input. At each sample time the block computes the value of the output based on the stored samples in memory most closely indexed by v, and the interpolation method specified by the **Mode** parameter. The available methods for computing the output are **Linear Interpolation** and **FIR Interpolation**, which are described below.

See the Variable Integer Delay block reference for a 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 adjacent physical samples in memory.

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 are specified in the same manner as for the Variable Integer Delay block. See the section on sample-based initial conditions there for complete information.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent channel. The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns, N) in the matrix.

In frame-based mode, the input at the `Delay` port must be a length-M vector, `v = [v(1) v(2) ... v(M)]`, containing one delay for each sample in the input frame(s). The earliest sample in each input 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 vector `v` is applied identically to every channel of a multichannel input.

See the Variable Integer Delay block reference for a discussion of how frame-based inputs 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 adjacent physical samples in memory.

Frame-based operation provides substantial increases in throughput rates at the expense of greater model latency.

The **Initial conditions** specifies the values in the block's memory at the start of the simulation. Both fixed and time-varying initial conditions are specified in the same manner as for the Variable Integer Delay block. See the section on frame-based initial conditions there for complete information.

Linear Interpolation Mode

The delay value specified at the `Delay` port is used as an index into the block's memory. 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)`. 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 delay for a scalar input, the block's output, `y`, is

```
vi = floor(v)                  % vi = integer delay
vf = v-vi                      % vf = fractional delay
y = (1-vf)*U(vi) + vf*U(vi+1)
```

Delay values less than 0 are clipped to 0, and delay values greater than the **Maximum delay in samples** are clipped to the **Maximum delay in samples**. Note that 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 2∗P to the stored samples on either side of the desired delay, where P is the **Interpolation filter half-length**. For periodic signals, a larger **Interpolation filter half length** (i.e., a higher order filter) yields a better estimate of the sample at the specified delay. A value between `4` and `6` for this parameter (i.e. a 7th to 11th order filter) is usually adequate.

A vector of 2∗P filter tap weights is precomputed at the start of the simulation for each of Q-1 discrete points between input samples, where Q is specified by 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 the **Interpolation points per input sample** parameter 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.

Note that increasing the **Interpolation filter half length** 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** 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 half the Nyquist frequency, you can specify a value of `0.5` (half Nyquist) for the **Normalized input bandwidth** to constrain the frequency content of the output to that range.

(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 is used to improve the stopband in critical regions, and to relax the stopband requirements in frequency regions where there is no signal energy.)

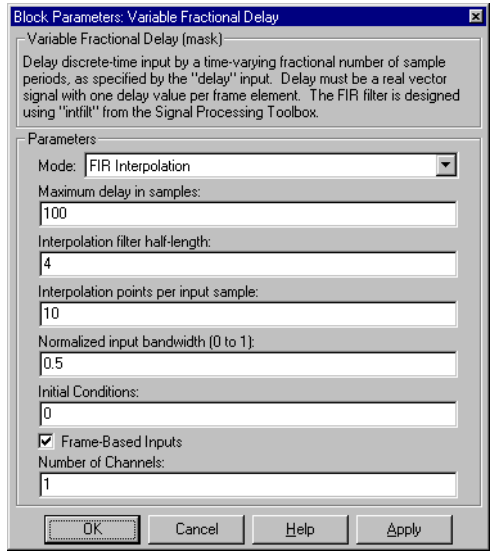
For delay values less that P/2-1 (where P is the **Interpolation filter half-length**), the output is computed using linear interpolation. Delay values greater than the **Maximum delay in samples** are clipped to the **Maximum delay in samples**.

The block uses the `intfilt` function in the Signal Processing Toolbox to compute the FIR filters.

Note When the Variable Fractional Delay block is used in a feedback loop, at least one block *without* direct feedthrough (e.g., an Integer Delay block with **Direct feedthrough** deselected) 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.

If you expect to generate code for the Variable Fractional Delay block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Mode

The method by which to interpolate between two adjacent samples in memory to obtain a value for the sample indexed by the input at the `Delay` port.

Maximum delay in samples

The maximum delay that the block can produce. 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 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 the Nyquist frequency.

Initial conditions

The values with which the block's memory is initialized.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of channels (columns) in the input matrix.

See Also

Integer Delay
Unit Delay (Simulink)
Variable Integer Delay

Variable Integer Delay

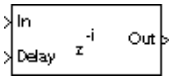
Purpose

Delay the input by a time-varying integer number of sample periods.

Library

Signal Operations, in General DSP

Description



The Variable Integer Delay block delays the discrete-time input at the top (`In`) port by the integer number of sample intervals specified by the input to the bottom (`Delay`) port. Both ports must have the same rate. The delay for a sample-based input sequence is a scalar value by which to uniformly delay every channel. The delay for a frame-based input sequence can be a scalar value by which to uniformly delay every sample in every channel, or a vector containing one delay value for each sample in the input frame.

The delay values should be in the range of 0 to D, where D is the **Maximum delay in samples**. 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 Integer Delay block in the following three ways.

Variable Integer Delay	Integer Delay
Delay is provided as an input to the `Delay` port	Delay is specified as a parameter setting in the dialog box
A unique delay can be applied to each consecutive *sample* in an input frame (in frame-based mode)	Every *sample* in an input frame is always delayed by an equal amount
The same delay is always applied to every input channel	A unique delay can be applied to each input channel

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector

elements (or M*N matrix elements) is treated as an independent channel, and the block applies the delay at the `Delay` port to each channel.

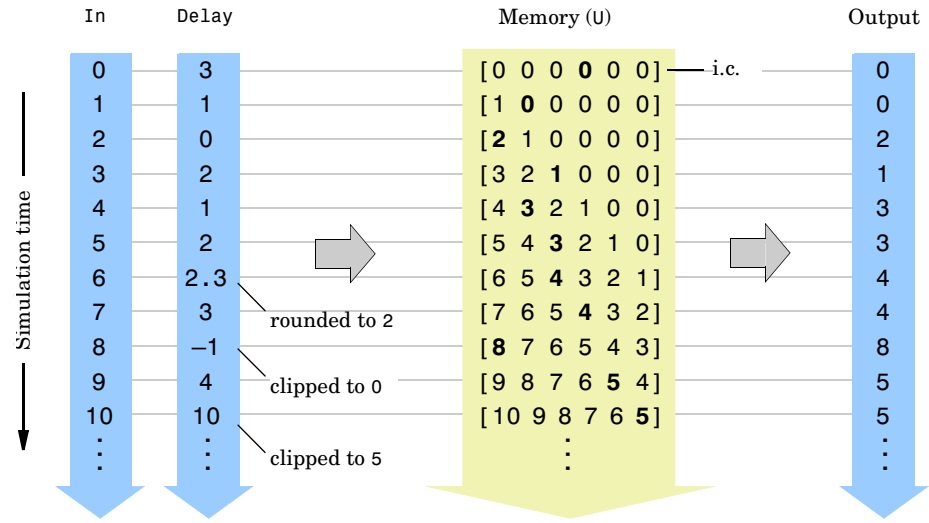
The Variable Integer Delay block stores the D+1 most recent samples received at the `In` port for each channel. At each sample time the block outputs the stored sample(s) indexed by the input to the `Delay` port.

For example, if 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. If we call the current input sample `U(1)`, the previous input sample `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. Note that 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 figure below shows the block output for a scalar ramp sequence at the `In` port, a **Maximum delay in samples** of 5, an **Initial conditions** of 0, and a variety of different delays at the `Delay` port.

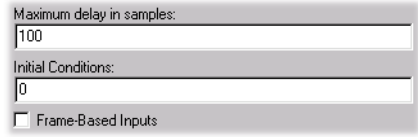


Note that the current input at each time-step is immediately stored in memory as `U(1)`. This allows the current input to be available at the output for a delay of 0 (`v=0`).

The **Initial conditions** parameter specifies the values in the block's memory at the start of the simulation. Unlike the Integer 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 shown below 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 below,

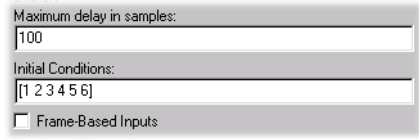
  

  the block initializes 100 M-by-N matrices in memory with zeros. A scalar initial condition can be used with input sequences of any dimension.

- *Vector* containing M∗N elements from which to construct an M-by-N *matrix* to initialize every sample in memory. M and N are the number of rows and columns, respectively, in the input matrix sequence. The initial condition vector, `ic`, is reshaped columnwise to match the input matrix dimensions.

  ```
  y = reshape(ic,M,N)    % equivalent MATLAB code
  ```

  For a 2-by-3 input and the parameters below,

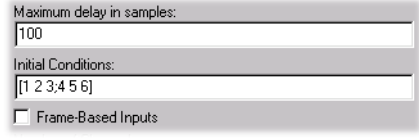


the block initializes 100 2-by-3 matrices in memory with

$$\begin{bmatrix} 1 & 3 & 5 \\ 2 & 4 & 6 \end{bmatrix}$$

An initial condition of length M∗N can be used with input sequences of any dimension, and can be specified as either a row or column vector.

- *Matrix* of dimension M-by-N with which to initialize every sample in memory, where M and N are the number of rows and columns, respectively, in the input matrix sequence. For a 2-by-3 input and the parameters below,



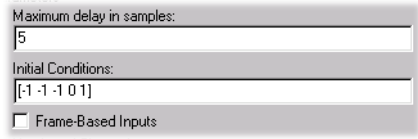
the block initializes 100 2-by-3 matrices in memory with

$$\begin{bmatrix} 1 & 2 & 3 \\ 4 & 5 & 6 \end{bmatrix}$$

For cases where M=N=1 or M=1, the initial condition setting reduces to a scalar or a vector, described above.

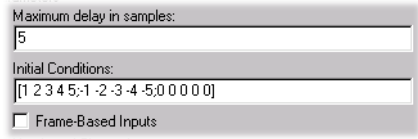
Time-Varying Initial Conditions. The following settings specify *time-varying* initial conditions. For a time-varying initial condition, the block initializes each of D samples in memory to one of the values entered in the **Initial conditions** parameter. This allows you to specify a unique output value 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 in samples**. For a scalar input and the parameters shown below,



the block initializes `U(2:6)` with values `[-1, -1, -1, 0, 1]`. A length-D vector initial condition can only be used with scalar inputs.

- *Matrix* of dimension M-by-D with which to initialize memory samples `U(2:D+1)`, where M is the length of the input *vector*, and D is the **Maximum delay in samples**. For a 1-by-3 input and the parameters below,

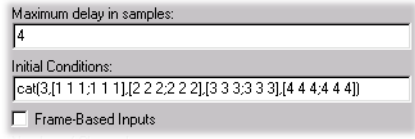


the block initializes memory locations `U(2:6)` with values

```
U(2) = [1 -1 0]
U(3) = [2 -2 0]
U(4) = [3 -3 0]
U(5) = [4 -4 0]
U(6) = [5 -5 0]
```

A matrix initial condition can only be used with vector inputs.

- *Array* of dimension M-by-N-by-D with which to initialize memory samples `U(2:D+1)`, where D is the **Maximum delay in samples** and M and N are the number of rows and columns, respectively, in the input matrix. For a 2-by-3 input and the parameters below,



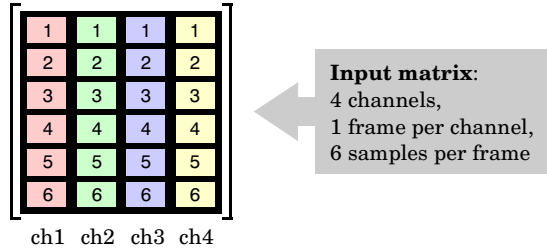
the block initializes memory locations `U(2:5)` with values

$$U(2) = \begin{bmatrix} 1 & 1 & 1 \\ 1 & 1 & 1 \end{bmatrix}, U(3) = \begin{bmatrix} 2 & 2 & 2 \\ 2 & 2 & 2 \end{bmatrix}, U(4) = \begin{bmatrix} 3 & 3 & 3 \\ 3 & 3 & 3 \end{bmatrix}, U(5) = \begin{bmatrix} 4 & 4 & 4 \\ 4 & 4 & 4 \end{bmatrix}$$

An array initial condition can only be used with matrix inputs.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the frames in the matrix contains M sequential time samples from an independent channel. The illustration below shows a 6-by-4 matrix input:



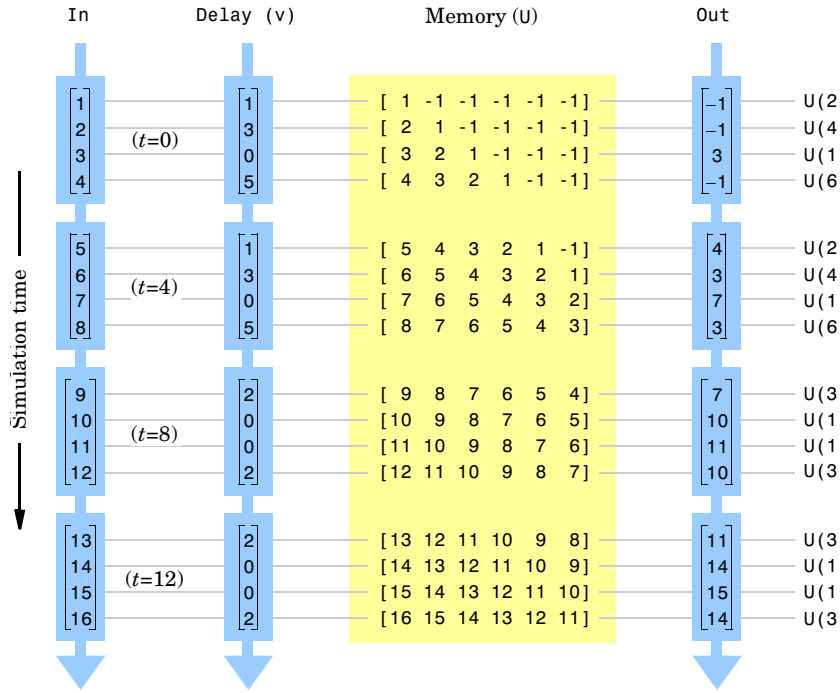
The **Number of channels** parameter specifies the number of independent channels (columns), N, in the matrix.

In frame-based mode, the input at the `Delay` port can be a scalar value by which to uniformly delay every sample in every channel, or length-M vector, `v = [v(1) v(2) ... v(M)]`, containing one delay for each sample in the input frame(s). The set of delays contained in vector `v` is applied identically to every channel of a multichannel 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 in samples** (`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. Note that 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)
```

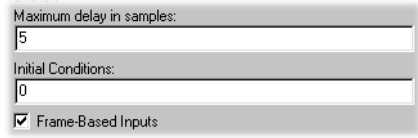


Frame-based operation provides substantial increases in throughput rates at the expense of greater model latency.

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.

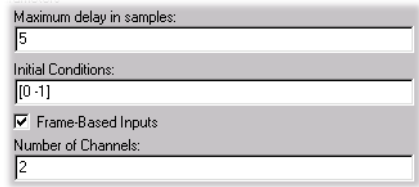
Fixed Initial Conditions. The settings shown below specify *fixed* initial conditions. For a fixed initial condition, the block initializes each of D samples in memory to the value entered in the **Initial conditions** parameter. A fixed initial condition in frame-based mode can be one of the following:

- *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 the six samples in memory with zeros.

- *Vector* of length N with which to initialize every sample in memory, where N is the **Number of channels**. For a two-channel input with a frame size of 4 and the parameter settings below,

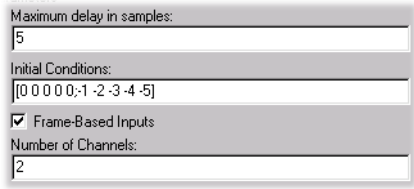


  the block the block initializes each of five samples in memory to `[0 -1]`. If the input frame size is specified to be 1, then this operation is equivalent to the sample-based operation described above.

Time-Varying Initial Condition. 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 way:

- *Matrix* of dimension M-by-D, where M is the **Number of channels** in the input, and D is the value specified for the **Maximum delay in samples** parameter (the *maximum* value if the **Maximum delay in samples** is a vector). The block's memory, `U(2:D+1)`, is initialized with the D columns of the matrix.

  For a two-channel input with a frame size of 4 and the parameter settings below,



the block initializes memory locations `U(2:6)` with values

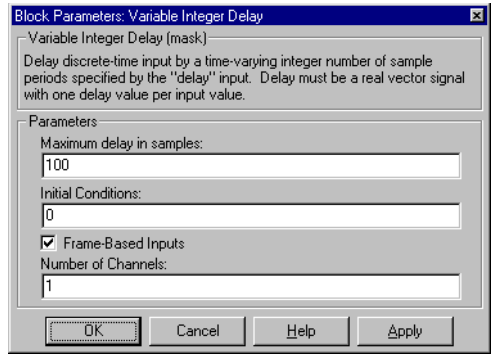
```
U(2) = [0 -1]
U(3) = [0 -2]
U(4) = [0 -3]
U(5) = [0 -4]
U(6) = [0 -5]
```

If the input frame size is specified to be 1, then this operation equivalent to the sample-based operation described above.

Note When the Variable Integer Delay block is used in a feedback loop, at least one block *without* direct feedthrough (e.g., an Integer Delay block with **Direct feedthrough** deselected) 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.

If you expect to generate code for the Variable Integer Delay block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Maximum delay in samples

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.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of channels (columns) in the input matrix.

See Also

Integer Delay
Variable Fractional Delay

Variable Selector

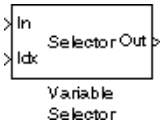
Purpose

Select a subset of elements in a vector.

Library

Elementary Functions, in Math Functions

Description



The Variable Selector block outputs a subset of the elements in the input vector.

When the **Selector mode** parameter is set to **Variable**, the vector at the bottom input (`Idx`) selects the elements of the vector at the top input (`In`) to pass through to the output. The elements of the indexing vector can change at each sample time, but the vector must remain the same length throughout the simulation.

When the **Selector mode** parameter is set to **Fixed**, the vector specified in the **Elements** parameter selects the elements of the top input vector (`In`) 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 indexing vector must remain the same length. The `Idx` port is not present in **Fixed** mode.

For both variable and fixed indexing vectors, the selection operation is equivalent to

```
y = u(idx)     % equivalent MATLAB code
```

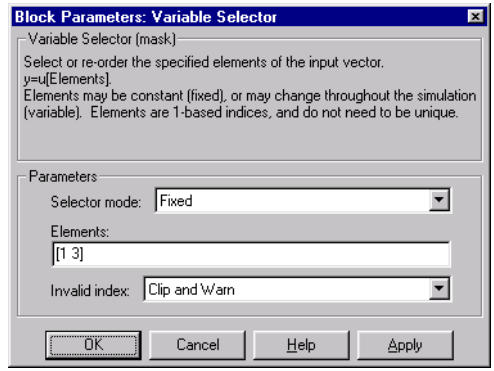
where `u(1)` is the first input element and `idx` is the indexing vector. The output is therefore the same length as the indexing vector. Input elements can appear any number of times in the output, or not at all.

When an element in the indexing vector references a non-existent element 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. Do not issue an alert. Example: For a 64-element 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.

Note The Variable Selector block always copies the selected input elements to a contiguous block of memory (unlike the Simulink Selector block).

Dialog Box



Selector mode

The type of indexing operation to perform, **Variable** of **Fixed**. Variable indexing uses the input at the `Idx` port to select elements of the input at the `In` port. Fixed indexing uses the **Elements** parameter value to select elements of the input at the `In` port.

Elements Ⓣ

A vector containing the indices of the input elements that will appear in the output vector.

Invalid index Ⓣ

Response to an invalid index value.

See Also

Permute Matrix
Selector (Simulink)
Submatrix

Variance

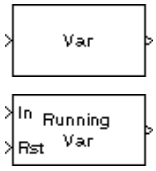
Purpose

Find the variance of an input or sequence of inputs.

Library

Statistics, in Math Functions

Description



The Variance block computes the variance of the input vector, or tracks the variance of a sequence of inputs over a period of time. The **Running variance** parameter allows you to select between basic operation and running operation, which are described below.

Basic Operation

When the **Running variance** check box is *not* selected, the block computes the variance of the input vector at each sample time.

```
y = var(u(:))      % equivalent MATLAB code
```

This implements the mathematical formula

$$y = \frac{\sum_{i=1}^{n} |u_i - \mu_x|^2}{n-1}$$

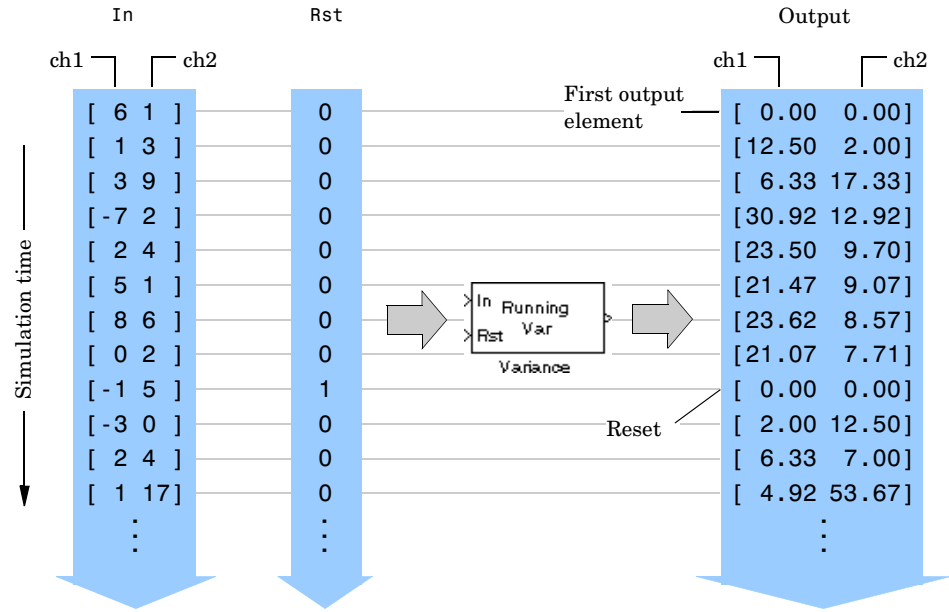
where μ_x is the mean of the input vector. The block outputs `0` for a scalar input, and treats a matrix input as a vector, `u(:)`

Running Operation

When the **Running variance** check box is selected, the block tracks the variance of a sequence of inputs over time. You can choose frame-based or sample-based operation by selecting (or deselecting, respectively) the **Frame-based** check box.

Sample-Based Operation. When the **Frame-based** check box is *not* selected (default), the block assumes that the input at the `In` port is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block tracks the variance of each of the channels over time.

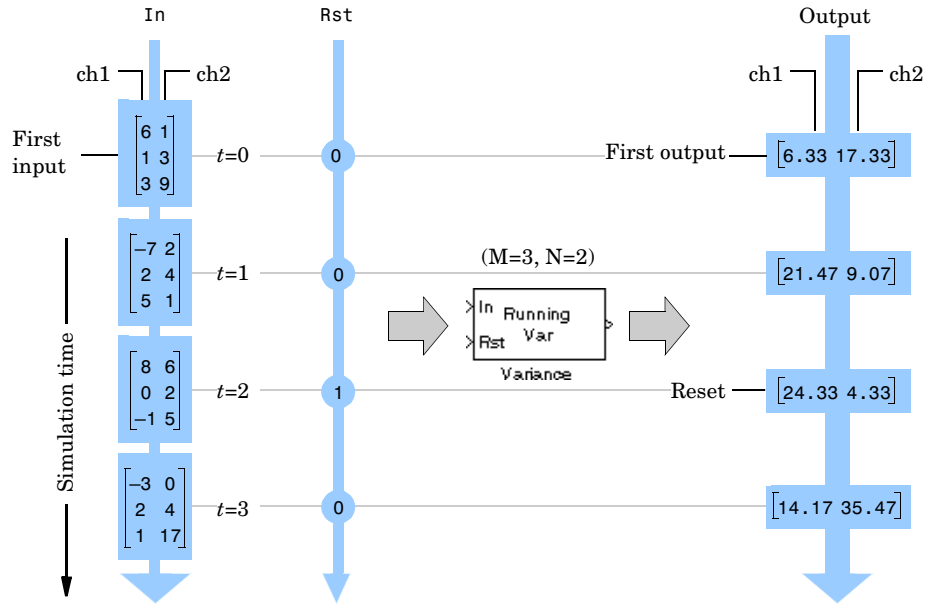
The block resets the running variance when the scalar input at the optional `Rst` port is nonzero. The output is the same size as the input, and contains the variance for each input channel since the last reset.



If you do not need to reset the running variance during the simulation, you can delete the `Rst` port from the block icon by deselecting the **Reset port** check box.

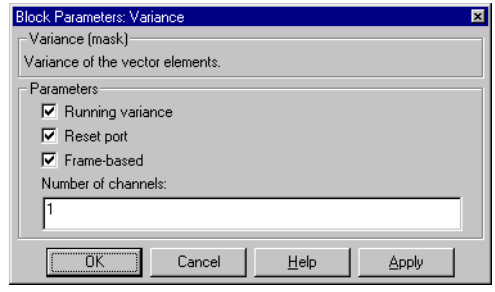
Frame-Based Operation. When the **Frame-based** check box is selected, the block assumes that the input at the `In` port is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent signals, N, in the matrix.

The block tracks the variance of each of the N independent channels over time, and resets the running variance when the input at the `Rst` port is nonzero. The output is a sample vector of length N which contains the variance for each input channel since the last reset.



Note If you expect to generate code for the Variance block's running mode using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Running variance

Selects running operation.

Reset port

Display `Rst` input port.

Frame-based

Selects frame-based operation.

Number of channels

For frame based operation, the number of channels (columns) in the input matrix, N.

See Also

Mean
Standard Deviation
`var` (MATLAB)

Window Function

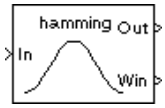
Purpose

Compute a specified window, and apply it to an input signal.

Library

Signal Operations, in General DSP; DSP Sources

Description



The Window Function block has three modes of operation, which are selected by the **Operation** parameter. In each mode of operation, the block computes samples of the window specified by the **Window type** parameter. The length of the sampled window, N_w, is determined by the size of the input, or specified by the **Window length** parameter if there is no input.

Operation	Description
Apply window to input	Multiplies the sampled window, `w`, element-wise with the input vector, `u`: `y = w .* u` The output is the result of this multiplication.
Generate window	Outputs the sampled window, `w`. There is no input.
Generate and apply window	Outputs the product of the window and the input, `w.*u`, in output 1, and outputs the sampled window, `w`, in output 2.

The **Sampling** parameter determines whether the window samples are computed in a *periodic* or a *symmetric* manner. For example, if **Sampling** is set to **Symmetric**, a **Hamming** window of **Window length** N_w is computed as

```
w = hamming(Nw)    % symmetric (aperiodic) window
```

If **Sampling** is set to **Periodic**, the same window is computed as

```
w = hamming(Nw+1)  % periodic (asymmetric) window
w = w(1:Nw)
```

The periodic window option only applies to the generalized-cosine windows (**Blackman**, **Hamming**, and **Hanning**).

The available window types are shown below. The **Beta** and **Stopband ripple** parameters specify the characteristics of the **Chebyshev** and **Kaiser** windows, and are only available when the respective window designs are selected.

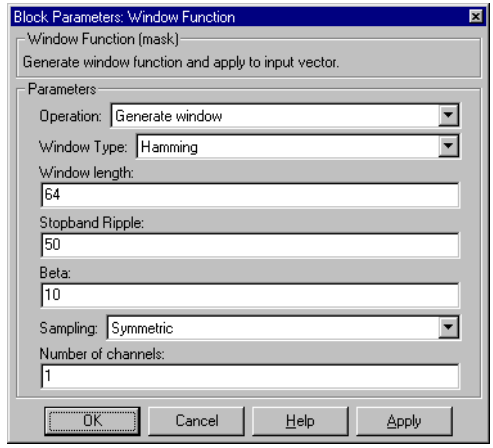
Window Type	Description
Bartlett	Applies a Bartlett window to the real input vector: `w = bartlett(Nw)`
Blackman	Applies a Blackman window to the real input vector: `w = blackman(Nw)`
Boxcar	Applies a Boxcar window to the real input vector: `w = boxcar(Nw)`
Chebyshev	Applies a Chebyshev window to the real input vector: `w = chebwin(Nw,R)`
Hamming	Applies a Hamming window to the real input vector: `w = hamming(Nw)`
Hanning	Applies a Hanning window to the real input vector: `w = hanning(Nw)`
Kaiser	Applies a Kaiser window to the real input vector: `w = kaiser(Nw,beta)`
Triang	Applies a triangular window to the real input vector: `w = triang(Nw)`

Frame-Based Operation

In the cases when the window is being applied to an input (**Generate and apply window** or **Apply window to input** selected from the **Operation** menu), the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent channels (columns, N) in the matrix. The block computes a

window to match the input frame size (N_w=M), and applies the window to each channel independently. The output is the same size as the input. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

Dialog Box



The parameters displayed in the dialog box vary for different operation/window type combinations. Not all of the parameters listed below are active in the dialog box at any one time.

Operation

The block's operation: generate a window, apply a window to an input, or both.

Window type Ⓣ

The type of window to apply.

Window length

The length of the window to apply, N_w.

Stopband ripple Ⓣ

The level (dB) of stopband ripple, R_s, for the Chebyshev window.

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.

Sampling Ⓣ

The window sampling, symmetric or periodic.

Number of channels

For frame-based operation, the number of channels (columns) in the input matrix, N.

See Also

FFT
`bartlett` (Signal Processing Toolbox)
`blackman` (Signal Processing Toolbox)
`boxcar` (Signal Processing Toolbox)
`chebwin` (Signal Processing Toolbox)
`hamming` (Signal Processing Toolbox)
`hanning` (Signal Processing Toolbox)
`kaiser` (Signal Processing Toolbox)
`triang` (Signal Processing Toolbox)

Yule-Walker AR Estimator

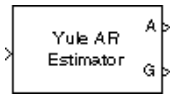
Purpose

Compute an estimate of AR model parameters using the Yule-Walker method.

Library

Parametric Estimation, in Estimation

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. 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.

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

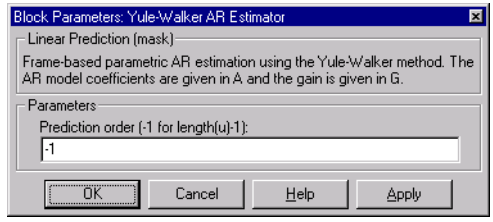
The order, p, of the all-pole model is specified by the **Order** parameter. The Yule-Walker AR Estimator and Burg AR Estimator blocks return similar results for large frame sizes.

The top output, `A`, 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`).

Dialog Box



Prediction order

The order of the AR model. A value of `-1` specifies a model order one less than the input frame size.

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.

See Also

Burg AR Estimator
Covariance AR Estimator
Modified Covariance AR Estimator
Yule-Walker Method
`aryule` (Signal Processing Toolbox)

Yule-Walker IIR Filter Design

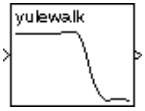
Purpose

Design and apply an IIR filter.

Library

Filter Designs, in Filtering

Description



The Yule-Walker IIR Filter Design block designs a recursive (ARMA) digital filter with arbitrary multiband magnitude response. The filter design, which uses the `yulewalk` function in the Signal Processing Toolbox, performs a least-squares fit to the specified frequency response.

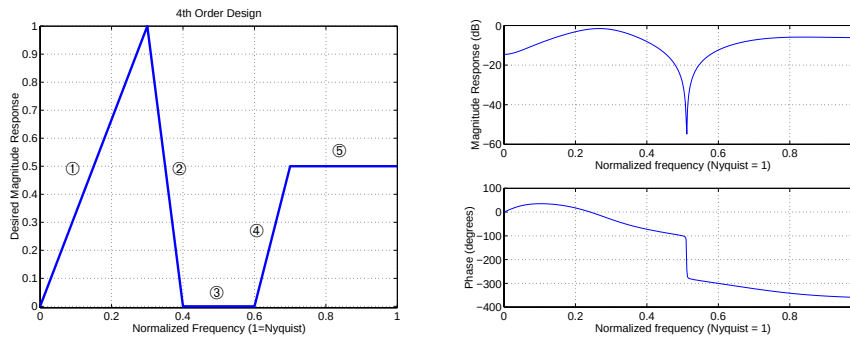
The **Band-edge frequency vector** parameter is a vector of frequency points in the range `0` to `1`, where `1` corresponds to half the sampling frequency (the Nyquist frequency). The first element of this vector must be `0` and the last element `1`, and intermediate points must appear in ascending order. 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 “transition” regions.

Band edge frequency = `[0.0 0.3 0.4 0.6 0.7 1.0]`

Magnitudes = `[0.0 1.0 0.0 0.0 0.5 0.5]`

Band: ① ② ③ ④ ⑤



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 User's Guide*.

The **Frame-based inputs** parameter allows you to choose between sample-based and frame-based operation.

Sample-Based Operation

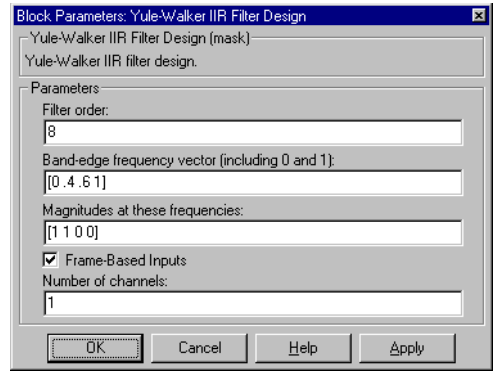
When the check box is *not* selected (default), the block assumes that the input is a 1-by-N sample vector or M-by-N sample matrix. Each of the N vector elements (or M∗N matrix elements) is treated as an independent channel, and the block filters each channel over time.

Frame-Based Operation

When the **Frame-based inputs** check box is selected, the block assumes that the input is an M-by-N frame matrix. Each of the N frames in the matrix contains M sequential time samples from an independent signal. The **Number of channels** parameter specifies the number of independent channels (columns, N) in the matrix, and the block filters each channel independently over time. Frame-based operation provides substantial increases in throughput rates, at the expense of greater model latency.

In both sample-based and frame-based operation, the output is the same size as the input.

Dialog Box



Filter order

The order of the filter.

Band-edge frequency vector

A vector of frequency points. The value `1` corresponds to the Nyquist frequency. The first element of this vector must be `0` and the last element `1`.

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**.

Frame-based inputs

Selects frame-based operation.

Number of channels

For frame-based operation, the number of channels (columns) in the input 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.

See Also

Digital IIR Filter Design
Least Squares FIR Filter Design
Remez FIR Filter Design
`yulewalk` (Signal Processing Toolbox)

Purpose

Compute a parametric estimate of the spectrum using the Yule-Walker AR method.

Library

Power Spectrum Estimation, in Estimation

Description

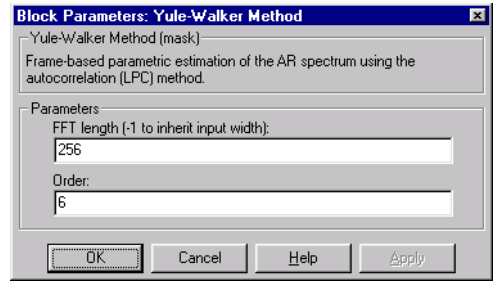
The Yule-Walker Method block estimates the power spectral density (PSD) of the input using the Yule-Walker AR method. This method, also called the autocorrelation method, fits an autoregressive (AR) model to the windowed input data by minimizing the forward prediction error in the least-squares sense. This formulation leads to the Yule-Walker equations, which are solved by Levinson-Durbin recursion. The spectrum is then computed from the FFT of the estimated AR model parameters.

The order of the all-pole model is specified by the **Order** parameter. The Yule-Walker Method and Burg Method blocks return similar results for large buffer lengths.

The input is a frame of consecutive time samples; a matrix input, `u`, is also treated as a single frame, `u(:)`. The block's output is the estimate of the signal's power spectral density at N_{fft} equally spaced frequency points in the range $[0,F_s)$, where N_{fft} is specified as a power of 2 by the **FFT Size** parameter and F_s is the signal's sample frequency. A value of `-1` for **FFT size** instructs the block to use the input frame size as the FFT size. Otherwise, the block zero pads or truncates the input to N_{fft} before computing the FFT.

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



FFT length

The number of data points on which to perform the FFT, N_{fft}. If N_{fft} exceeds the size of the input frame, the data is zero padded as needed.

Order

The order of the AR model.

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.

See Also

Burg Method
Covariance Method
Levinson Solver
LPC
Short-Time FFT
Yule-Walker AR Estimator
`pyulear` (Signal Processing Toolbox)

Purpose

Increase the input frame size by appending zeros.

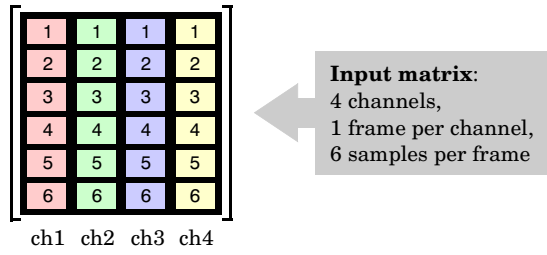
Library

Signal Operations, in General DSP

Description

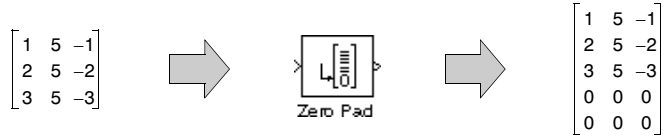
The Zero Pad block increases the frame size of the input from M_i to M_o by appending an equal number of zeros to each channel. The output frame size, M_o, is specified by the **Output frame size** parameter. If the **Output frame size** value is smaller than the input frame size ($M_o<M_i$), and the **Allow truncation of input** check box is enabled, the block truncates elements at the end of each input frame as necessary. If the **Allow truncation of input** check box is disabled, the input is not altered.

The Zero Pad block assumes that the input is an M_i-by-N frame matrix. Each of the N frames in the matrix contains M_i sequential time samples from an independent signal. The illustration below shows a 6-by-4 matrix input:



The **Number of channels** parameter specifies the number of independent channels (columns, N) in the matrix. A value of 1 indicates that the input is a vector (i.e., a single frame).

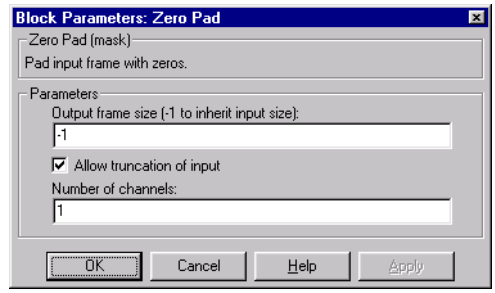
The figure below illustrates how zero-padding increases the number of matrix rows from M_i to M_o by appending zeros at the end of each frame.



Output frame size = 2
Number of channels = 3

Note If you expect to generate code for the Zero Pad block using the Real-Time Workshop, you should ensure that inputs are contiguous in memory. See the Contiguous Copy block for more information.

Dialog Box



Output frame size

The desired output frame length, M_o. A value of `-1` effectively disables zero-padding by passing the input through unchanged.

Allow truncation of input

When selected (default), the block truncates the input to **Output frame size** if it is larger than this size. When not selected, the block passes such an input through unchanged.

Number of channels

For frame based operation, the number of columns (channels) in the input matrix, N.

See Also

Repeat
Upsample

5

DSP Function Reference

DSP Blockset Utility Functions

In addition to the blocks contained in the DSP Blockset libraries, a number of utility functions and scripts are provided in the `toolbox\dspblks\dspblks` directory. The key functions are listed below and described on the following pages.

- `dsp_links`
- `dsplib`
- `dspstartup`
- `liblinks`
- `rebuffer_delay`

Purpose

Display library link information for blocks linked to the DSP Blockset.

Syntax

```
dsp_links
dsplinks(sys)
dsplinks(sys,mode)
```

Description

`dsp_links` displays library link information for blocks linked to the DSP Blockset. For each block in the current model, `dsp_links` replaces the block name with the full pathname to the block's library link in the DSP Blockset. Block's linked to the v3.0 DSP Blockset blocks are highlighted in blue while blocks linked to v2.2 DSP Blockset blocks are highlighted in red. Blocks at all levels of the model are analyzed.

A summary report indicating the number of blocks linked to each blockset version is also displayed in the MATLAB command window. The highlighting and link display is disabled when the model is executed or saved, or when `dsp_links` is executed a second time from the MATLAB command line.

`dsp_links(sys)` toggles the display of block links in system `sys`. If `sys` is the current model (`gcs`), this is the same as the plain `dsp_links` syntax.

`dsp_links(sys,mode)` directly sets the link display state, where *`mode`* can be `'on'`, `'off'`, or `'toggle'`. The default is `'toggle'`.

See Also

`liblinks`

dsplib

Purpose

Open the main DSP Blockset library.

Syntax

```
dsplib
dsplib ver
```

Description

`dsplib` opens the current version of the main DSP Blockset library.

`dsplib ver` opens version *ver* of the DSP Blockset library, where *ver* can be `2` or `3`.

When you launch an older version of the DSP Blockset, MATLAB displays a message reminding you that a newer version exists.

Purpose

Configure the Simulink environment for DSP systems.

Syntax

```
dspstartup
```

Description

`dspstartup` configures a number of Simulink environment parameters with settings appropriate for a typical DSP project. When the Simulink environment has successfully been configured, the function displays the following message in the command window.

```
Changed default Simulink settings for DSP 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 documentation for the `startup` command in the online *MATLAB Function Reference*.

The `dspstartup.m` script sets the following Simulink environment parameters. See Appendix A, "Model and Block Parameters," in *Using Simulink* for complete information about a particular setting.

Parameter	Setting
`Solver`	`fixedstepdiscrete`
`SolverMode`	`auto`
`StartTime`	`0.0`
`StopTime`	`inf`

Parameter	Setting
FixedStep	auto
SaveTime	off
SaveOutput	off
AlgebraicLoopMsg	error
InvariantConstants	on
RTWOptions	[get_param(0,'RTWOptions')', -aRollThreshold=2']

See Also

startup

Purpose

Display library link information for blocks linked to the DSP Blockset.

Syntax

```
liblinks
liblinks(sys)
liblinks(sys,mode,lib)
liblinks(sys,mode,lib,clrs)
blks = liblinks(...)
```

Description

Please see the command line help for `liblinks`. Type

```
help liblinks
```

in the MATLAB command window.

See Also

`dsp_links`

rebuffer_delay

Purpose

Compute the number of samples of delay introduced by the Rebuffer block.

Syntax

```
d = rebuffer_delay(f,n,m)
```

Description

`d = rebuffer_delay(f,n,m)` returns the delay (in samples) introduced by the Rebuffer block for frame-based operation with input frame size `f`, **Buffer size** `n`, and **Buffer overlap** `m`.

In sample-based operation, the delay is always `n-m` samples.

See Also

Rebuffer block

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