Filter Design Toolbox

For Use with MATLAB®

- Computation
- Visualization
- Programming

User's Guide

Version 3

The MathWorks

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508-647-7000	Phone
508-647-7001	Fax
The MathWorks, Inc. 3 Apple Hill Drive Natick, MA 01760-2098	Mail

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

Filter Design Toolbox User's Guide

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Filter Design Toolbox Overview

What Is Filter Design Toolbox?

The Filter Design Toolbox is a collection of tools that provides advanced techniques for designing, simulating, and analyzing digital filters. It extends the capabilities of the Signal Processing Toolbox with filter architectures and design methods for complex real-time DSP applications, including adaptive filtering and multirate filtering, as well as filter transformations.

Used with the Fixed-Point Toolbox, the Filter Design Toolbox provides functions that simplify the design of fixed-point filters and the analysis of quantization effects. When used with the Filter Design HDL Coder, the Filter Design Toolbox lets you generate VHDL and Verilog code for fixed-point filters.

Key Features

- Advanced FIR filter design methods, including minimum-order, minimum-phase, constrained-ripple, halfband, Nyquist, interpolated FIR, and nonlinear phase
- Perfect reconstruction and two-channel FIR filter bank design
- Advanced IIR design methods, including arbitrary magnitude, group-delay equalizers, constrained-pole radius, peaking, notching, and comb filters
- Analysis and implementation of digital filters in single-precision floating-point and fixed-point arithmetic
- Support for IIR filters implemented in second-order sections, including design, scaling, and section reordering
- Round-off noise analysis for filters implemented in single-precision floating point or fixed point
- FIR and IIR filter transformations, including lowpass to lowpass, lowpass to highpass, and lowpass to multiband.
- Adaptive filter design, analysis, and implementation, including LMS-based, RLS-based, lattice-based, frequency-domain, fast transversal, and affine projection
- Multirate filter design, analysis, and implementation, including cascaded integrator-comb (CIC) fixed-point multirate filters
- VHDL and Verilog code generation for fixed-point filters

Advanced FIR and IIR Filter Design

The Filter Design Toolbox lets you design optimal FIR and IIR filters from scratch, import previously designed filters, quantize floating-point filters, and analyze quantization effects. You can access all toolbox functions from the Filter Design and Analysis Tool (FDATool) or from the command line

FIR Filters

The toolbox supports the following FIR filter designs:

- Advanced equiripple FIR designs, including minimum-order, constrained-ripple, minimum-phase, adjacent bands, single-point bands, independent approximation errors, extra-ripple, and maximal-ripple designs
- Least Pth-norm FIR designs, providing optimal nonlinear phase designs that can minimize any norm from 2 (minimum error energy) to infinity (minimax/equiripple error)
- Halfband FIR designs, including equiripple, least-squares, and window methods
- Nyquist (Lth-band) filter designs, providing linear phase and minimum-phase designs, as well as equiripple, sloped-stopband, and window methods
- Interpolated FIR (IFIR) designs, enabling you to:
  - Design filters with narrow transition bands using a reduced number of multipliers
  - Choose between simple, intermediate, and advanced optimization
- FIR filter design for perfect reconstruction and two-channel filter banks

IIR Filters

The toolbox supports the following IIR filter designs:

- Allpass IIR filter design with arbitrary group delay, enabling you to equalize the group delays of other IIR filters to obtain an approximate linear phase passband response
- Least Pth-norm IIR design, enabling the design of arbitrary magnitude optimal IIR filters in addition to lowpass, highpass, bandpass, and bandstop designs

- Constrained least Pth-norm IIR design, constraining the maximum radius of the filter poles to improve robustness to quantization effects
- Peak, notch, and comb filters, to eliminate single-tone and periodic interference

Analyze and Implement Fixed-Point and Single-Precision Floating-Point Filters

When used with the Fixed-Point Toolbox, the Filter Design Toolbox offers bit-true, fixed-point implementation of digital filters using more than 15 structures, including FIR, IIR (SOS and non-SOS), and lattice-based filters. Word lengths for different quantities, such as coefficients, products, and accumulators, can be set to arbitrary values. Full-precision modes are available to simulate the non-loss of bits. Other automatic modes available include best-precision and avoid overflow.

The Filter Design Toolbox provides a full suite of analysis tools for fixed-point filters, including magnitude response, impulse response, pole/zero plots, and round-off-noise. The toolbox supports the implementation and analysis of single-precision floating-point filters for the same filter structures as for fixed-point arithmetic.

Implement IIR Filters in Second-Order Sections

The toolbox includes functions for the design of IIR filters directly in second-order section (SOS) form. SOS minimizes round-off problems when converting from transfer functions or pole-zero representations. The toolbox provides scaling of SOS filters in any of the four direct forms to maximize the performance of these filters when implemented in fixed point. You can use the scaling to reorder the second-order sections for further optimization. The toolbox also provides enhanced filter analysis methods specific to SOS filters.

Design, Analyze, and Implement Adaptive and Multirate Filters

The Filter Design Toolbox provides full support for the design of advanced filters. The toolbox contains objects that let you develop adaptive and multirate filters to stringent specifications and apply the filter to data, minimizing errors between the filter output and the desired signal.

Adaptive Filters

Adaptive filters are self-learning filters that achieve a particular effect when no *a priori* signal information is available. The Filter Design Toolbox provides the following techniques for LMS-based adaptive filters:

- RLS-based
- Affine projection
- Fast transversal
- Frequency-domain
- Lattice-based

The toolbox also includes algorithms for analyzing these filters, including tracking of coefficients, learning curves, and convergence.

Multirate Filters

Multirate filtering involves using different sample rates within a system. This approach yields computational efficiencies that are impossible to obtain with a system that operates on a single, fixed-sample rate. The Filter Design Toolbox supports the analysis, design, and implementation of multirate filters. It includes functions for designing:

- Polyphase interpolators
- Polyphase decimators
- Polyphase sample-rate converters
- CIC multirate filters
- Multistage multirate filters

Specialized functions for analyzing polyphase (multirate) filters are available as they are for adaptive filters.

Designers use filtering and its variant, digital filtering, for many tasks:

- To separate signals that have been combined, such as a musical recording and the noise added during the recording process
- To separate signals into their constituent frequencies
- To demodulate signals
- To restore signals that have been degraded by some process, known or unknown

You can use analog filters to accomplish these tasks, but digital filters offer greater flexibility and accuracy than analog filters. In addition, digital signal processing (DSP) depends in large measure on digital filtering to meet the needs of its users.

Analog filters can be cheaper, faster, and have greater dynamic range; digital filters outstrip their analog cousins in flexibility. The ability to create filters that have arbitrary shape frequency response curves, and filters that meet performance constraints, such as bandpass width and transition region width, is well beyond that of analog filters.

Quantization is a natural outgrowth of digital filtering and digital signal processing development. Also, there is a growing need for fixed-point filters that meet power, cost, and size restrictions. When you convert a filter from floating-point to fixed-point, you use quantization to perform the conversion.

As filter designers began to use digital filters in applications where power limitations and size constraints drove the filter design, they moved from double-precision, floating-point filters to fixed-point filters. When you have enough power to run a floating-point digital signal processor, such as on desktop PC or in your car, fixed-point processing and filtering are unnecessary. But, when your filter needs to run in a cellular phone, or you want to run a hearing aid for hours instead of seconds, fixed-point processing can be essential to ensure long battery life and small size.

Filter Design Toolbox provides the functions you need to develop filters that meet the needs of fixed-point algorithms and electronics systems. In addition to offering tools for analyzing the effects of quantization on filter performance and signal processing performance, the toolbox offers filter structures for you to use to develop prototype filter designs. With structures ranging from finite impulse response (FIR) filters to infinite impulse response (IIR) filters, adaptive filters, and multirate filters, you can investigate alternative fixed-point realizations of filters that might meet your goals.

Configuration Information

To determine whether Filter Design Toolbox is installed on your system, type this command at the MATLAB® prompt.

```
ver
```

When you enter this command, MATLAB displays information about the version of MATLAB you are running, including a list of all toolboxes installed on your system and their version numbers.

For information about installing the toolbox, refer to the installation documentation for your platform.

Note For up-to-date information about system requirements, visit the system requirements page, available in the products area at the MathWorks Web site (`www.mathworks.com`).

Using This Guide

All users of the toolbox should read this guide. You should be generally familiar with basic digital signal processing concepts before you use the toolbox and this User's Guide. The quantization portion of this toolbox assumes some familiarity with fixed-point and floating-point arithmetic in the context of digital filtering applications.

New Users of This Toolbox

You can use this toolbox to:

- Design floating-point and fixed-point filters using advanced design methods
- Design adaptive filters
- Design multirate filters
- Transform filters from one frequency response type to another, such as from lowpass to bandstop
- Convert filters to and from coupled-allpass forms
- Convert filters to second-order section form, or design filters in second-order section form

This toolbox relies on object-oriented programming techniques using objects for quantized filtering and analysis. You do not need to be familiar with these techniques to use this toolbox. However, you may want to review the concepts of MATLAB structures and cell arrays, as these are used in the syntax for several toolbox methods. For more information on MATLAB structures and cell arrays, refer to "Programming and Data Types" in your MATLAB documentation.

As a new user of this toolbox, read the entire guide. Of particular interest are:

- Chapter 2, "Designing Advanced Filters" for its background information on the advanced filter design techniques in this toolbox
- Chapter 3, "Working with Fixed-Point Filters" for information on constructing and using quantized filters
- Chapter 7, "Using FDATool with the Filter Design Toolbox" for information about using Filter Design and Analysis Tool to quantize filters and investigate the effects of quantization on filter performance

- “Property Details for Fixed-Point Filters” on page 8-20 for a description of the quantized and fixed-point filter properties
- “Functions—Categorical List” on page 9-2 for a brief description of every function in the toolbox (online only)

Experienced Filter Design Toolbox Users

As an experienced user of this toolbox, you may find the following sections to be useful reference guides for the toolbox:

- “Property Details for Fixed-Point Filters” on page 8-20
- “Functions—Categorical List” on page 9-2 (online only)
- Reference pages for the functions in the toolbox (available online only)

This section contains the following subsections introducing filter design:

- “Filter Design Functions in the Toolbox” on page 1-10
- “Fixed-Point Filters in the Toolbox” on page 1-14
- “Getting Started with the Toolbox” on page 1-17
- “Selected Bibliography” on page 1-27

Filter Design Functions in the Toolbox

In a system that has unlimited power and size, any filter structure that met your performance specifications would do. You would design a floating-point filter whose frequency response achieved your aims and implement that filter in your system.

When you need a fixed-point filter to meet your requirements, the filter structure you choose can depend very much on how quantization affects the performance of the filter. Filter Design Toolbox offers both FIR and IIR filter design tools and structures that let you experiment with multiple filter designs to see how each responds to quantization effects.

Discrete-Time Filter Structures

The following tables detail some of the quantized FIR and IIR filter structures and designs available in the toolbox. For lists of all the architectures and design methods available in the toolbox, refer to the section "Functions—Categorical List" on page 9-2 in this guide.

Table 1-1: Finite Impulse Response Filter Structures

FIR Filter Structures	Description
`dfilt.dfasymfir`	Asymmetrical FIR filter
`dfilt.dffir`	Direct-form FIR filter
`dfilt.dffirt`	Transposed direct-form FIR filter
`dfilt.dfsymfir`	Symmetric FIR filter

Table 1-2: Infinite Impulse Response Filter Structures

IIR Filter Structures	Description
`dfilt.df1`	Direct form I
`dfilt.df1sos`	Direct form I realized using second-order sections
`dfilt.df1t`	Direct form I transposed

Table 1-2: Infinite Impulse Response Filter Structures

IIR Filter Structures (Continued)	Description
`dfilt.df1tsos`	Direct form I transposed, realized using second-order sections
`dfilt.df2`	Direct form II
`dfilt.df2sos`	Direct form II, realized using second-order sections
`dfilt.df2t`	Direct form II transposed
`dfilt.df2tsos`	Direct form II transposed, realized using second-order sections
`dfilt.scalar`	Scalar form

Table 1-3: Lattice Filter Structures

Lattice Structure	Description
`dfilt.latticeallpass`	All pass lattice form
`dfilt.latticear`	Autoregressive (AR) lattice form
`dfilt.latticearma`	Autoregressive, moving average (ARMA) lattice form
`dfilt.latticemamax`	Lattice moving average with maximum phase
`dfilt.latticemamin`	Lattice moving average with minimum phase

Each of the structures supports floating-point or fixed-point realizations, and you use the toolbox functions, `dfilt.`*`structure`*, to create each one. To review schematics of the filter structures available in this toolbox, refer to the reference pages for each filter constructor, such as `dfilt.df1` or `dfilt.latticearma`. Or to review all of the structures available for discrete-time filters, refer to "FilterStructure" on page 8-46.

To run the filter demos

1 Enter `demo` at the MATLAB command line prompt.

   The Help system opens with the MATLAB demos showing for all products.

2 Double-click the entry **Toolboxes** in the left pane. The list of available toolboxes appears in the left pane.

3 Click **Filter Design**.

4 Click **FIR Filter Design** or **IIR Filter Design** in the right pane. Or choose from the other demonstration categories like adaptive filters.

Adaptive Filters

The following table list a few of the adaptive filter algorithms and filters available in the toolbox. For complete lists of all the adaptive filters available in the toolbox, refer to the section "Adaptive Filter Constructors" on page 9-2 in this guide.

Filter Function	Description
`adaptfilt.lms`	Use the LMS FIR adaptive filter algorithm
`adaptfilt.nlms`	Use the normalized LMS FIR adaptive filter algorithm
`adaptfilt.rls`	Use the recursive-least squares (RLS) adaptation algorithm
`adaptfilt.sd`	Use the sign-data LMS FIR adaptive filter algorithm

Multirate Filters

The following table details some of the more common multirate filters available in the toolbox. For lists of all the multirate filters available in the toolbox, refer to the section "Multirate Filter Constructors" on page 9-11 in this guide.

Filter Function	Description
`mfilt.cicdecim`	Construct a cascaded integrator-comb decimation filter object
`mfilt.cicinterp`	Construct a cascaded integrator-comb interpolation filter object
`mfilt.firfracdecim`	Construct a direct-form FIR polyphase fractional decimation filter object
`mfilt.linearinterp`	Construct an FIR linear interpolation filter object that applies linear interpolation between input samples

Fixed-Point Filters in the Toolbox

Designing floating-point filters solves only part of the filter design problem. In most cases, floating-point filter realizations are not appropriate for digital signal processing applications. Many real-world DSP systems require that their filters use minimum power, generate minimum heat, and do not induce computational overload in their processors. Meeting these constraints often means using fixed-point filters. Unfortunately, converting a floating-point filter to fixed-point realization (called quantizing) can result in lost filter performance and accuracy. To simulate and determine the effects of quantization, and allow you to investigate how switching from floating-point to fixed-point arithmetic affects the performance of your filter, the toolbox includes quantization functions. You use the toolbox construction and analysis functions for constructing, applying, and analyzing floating-point and fixed-point filters.

Quantized Filters

Quantization, or the effect of word length on filter performance, can lead to erroneous behavior in filter designs. Finite word lengths can change the frequency response of a filter from its desired performance. To help you investigate quantization effects that occur during filtering, the toolbox provides two ways to construct a quantized filter:

- Use the function `dfilt.`*`structure`* to create a default filter with a selected filter structure.
- Use `fdesign` and specify a filter magnitude response and use it to design a filter.

In both techniques, your filters have the same properties although the implemented structures may be different.

To help you create and analyze quantized filters, Filter Design Toolbox is object-oriented. You encapsulate the parameters needed to specify your quantized filter under one variable name in a quantized filter object. To specify the parameters associated with a quantized filter, you set the property values for its associated named properties. These properties are assigned to the quantized filter object that represents your quantized filter.

You can design a wide range of fixed-point and custom floating-point filters in Filter Design Toolbox. You use the double-precision filters you design in Signal

Processing Toolbox and Filter Design Toolbox as reference filters to create quantized filters in this toolbox. To develop a quantized filter, use either toolbox to create a double-precision filter that meets your requirements, then use the quantization functions in this toolbox to convert the double-precision filter to a quantized filter.

Refer to Table 1-4 for a list of some of the filter design methods in this toolbox.

Table 1-4: Filter Design Methods in the Toolbox—FIR and IIR

Filter Function	Filter Description
`firlpnorm`	Design minimax solution FIR filters using the least-pth algorithm
`firgr`	Use the generalized Parks-McClellan exchange algorithm to design optimal solution FIR filters with arbitrary response curves
`fircband`	Design constrained-band Remez FIR filters
`firceqrip`	Design constrained, equiripple, finite impulse response (FIR) filters
`firequint`	Design equiripple FIR interpolation filters
`firgr`	Design FIR filters using generalized Remez techniques
`firhalfband`	Design half-band FIR filters
`firnyquist`	Design equiripple Nyquist FIR filters
`firpr2chfb`	Design FIR perfect reconstruction two-channel filter banks
`ifir`	Design interpolated FIR filters
`iircomb`	Design IIR comb notching or peaking discrete-time filter
`iirgrpdelay`	Design optimal solution IIR filters where you specify the group delay in the passband frequencies

Table 1-4: Filter Design Methods in the Toolbox—FIR and IIR (Continued)

Filter Function	Filter Description
`iirlpnorm`	Design minimax solution IIR filters using the least-pth algorithm
`iirlpnormc`	Design minimax solution IIR filters using the least-pth algorithm. In addition, restrict the filter poles and zeros to lie within a fixed radius around the origin of the *z*-plane
`iirpeak`	Design second-order IIR peaking (or resonator) filters

You can construct these filters as single-precision floating-point, double-precision floating-point or fixed-point filters. For more information about all of the design methods in the toolbox, refer to "Discrete-Time Filter Constructors" on page 9-7.

Getting Started with the Toolbox

This section provides an example to get you started using Filter Design Toolbox. You can run the code in this example from the Help browser (select the code, right-click the selection, and choose **Evaluate Selection** from the context menu) or you can enter the code on the command line. This exercise also introduces Filter Design and Analysis Tool (FDATool). You use it to design and analyze filters, and to quantize filters.

As you follow the example, you are introduced to some of the basic tasks of designing a filter and using FDATool. You will engage some of the quantization capabilities of the toolbox, and a few of the filter analyses provided as well.

Before you begin this example, start MATLAB and verify that you have installed Signal Processing and Filter Design Toolboxes (type `ver` at the command prompt). You should see Filter Design Toolbox, version 3.0 and Signal Processing Toolbox, version 6.2, among others, in the list of installed products.

Example—Creating a Fixed-Point IIR Filter

Example Background. To introduce you to designing fixed-point filters in the toolbox, this example starts by using Filter Design and Analysis Tool (FDATool) to design a straightforward IIR filter. In this case, we use the Chebyshev I filter design method to begin.

During the example, you have the chance to export filters to your MATLAB workspace, filter some data with the filter, and use the scaling features in FDATool to improve the filter performance. One of the most salient points in this example is that, while second-order section (SOS) filters are generally good starting points for fixed-point filter design, you might find that without scaling your filter, the SOS implementation may not meet your needs, as this example shows. to

To create a fixed-point filter in FDATool

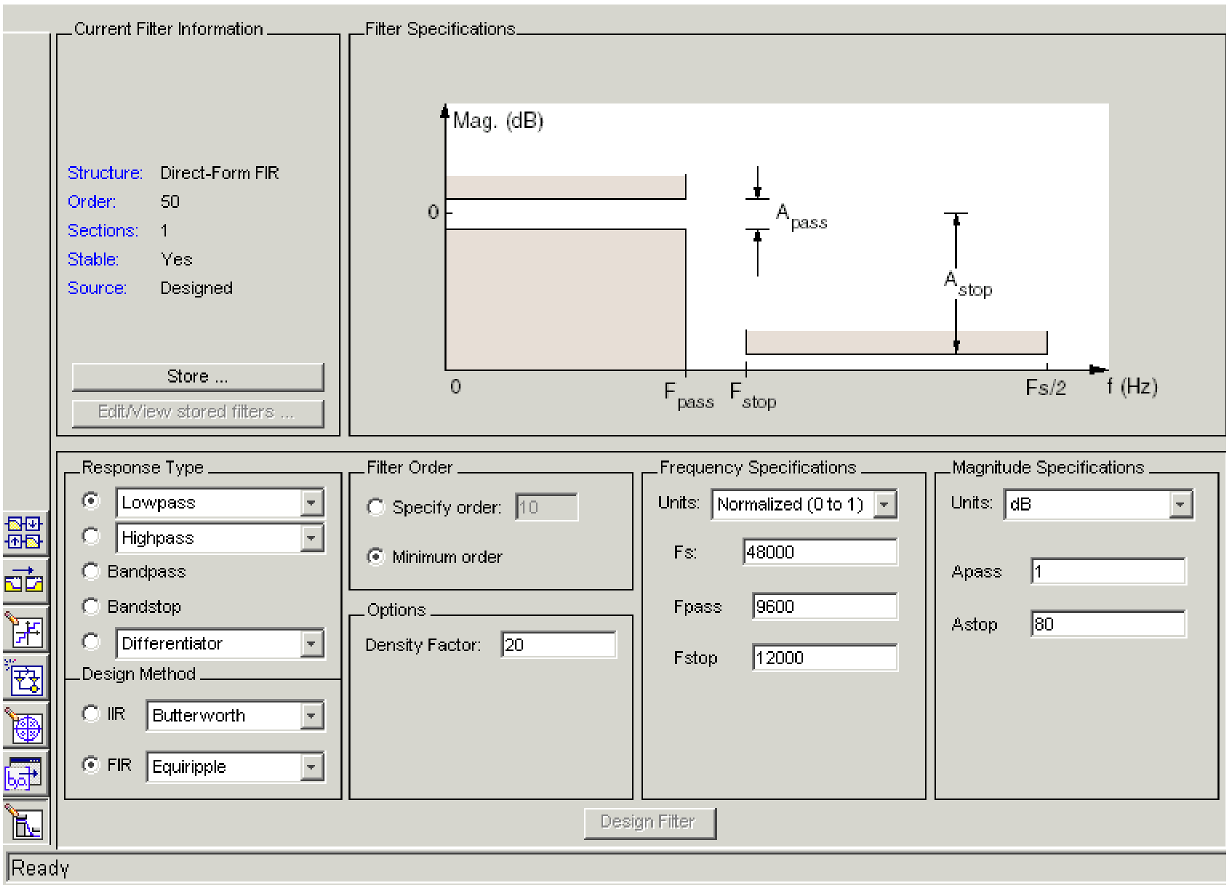
Filter Design and Analysis Tool (FDATool) is one tool this toolbox provides to help you design and analyze filters. From the various design panels in the tool, such as the design filter panel or the multirate filter design panel, you can design FIR and IIR filters, import or export filters, analyze filters, and more.

As an introduction the using the toolbox, this tutorial takes you through designing, quantizing, and scaling a filter in FDATool.

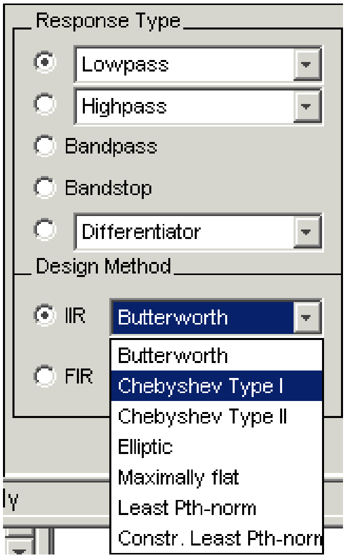
1 Open Filter Design and Analysis Tool by entering

```
fdatool
```

at the MATLAB command prompt. FDATool opens to show you the following dialog.

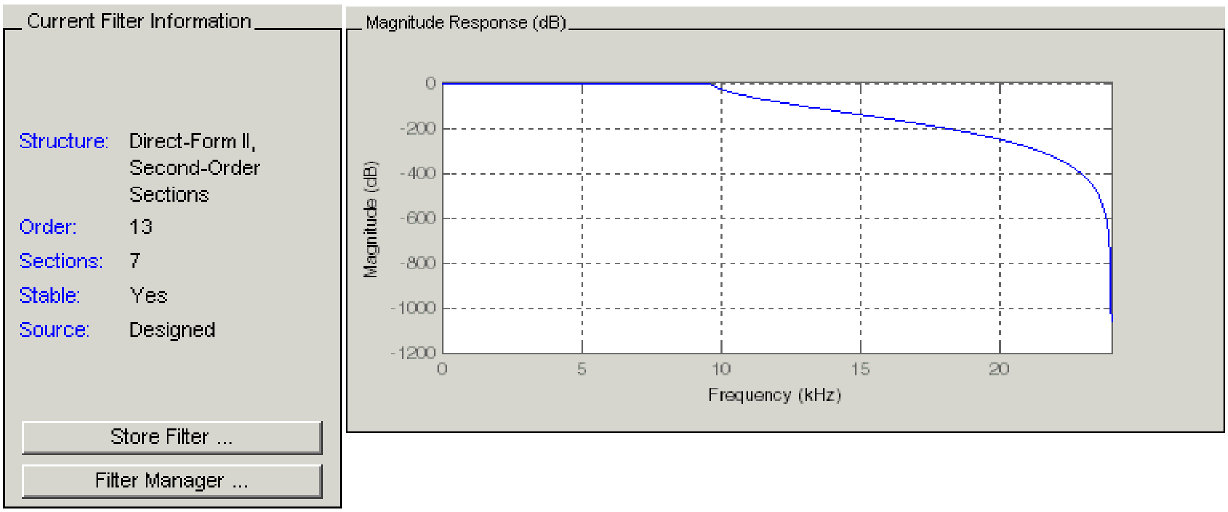


2 Under **Design Method** in the bottom pane, select `Chebyshev Type I` from the **IIR** list and click **Design Filter**.



FDATool designs a double-precision lowpass filter using the Chebyshev I design method and displays the filter magnitude response in the FDATool analysis area. Your new filter is a double-precision filter that uses seven second-order sections. In the Current Filter Information area in FDATool, reproduced in the next figure, you see your filter described by various filter parameters including the filter order (13) and the structure (direct-form II using second-order sections).

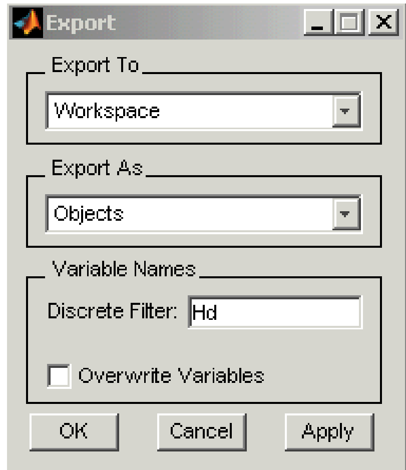
In the figure, next to the current filter information, the curve presents the filter magnitude response. As intended, a lowpass filter with the end of the passband at about 9600 Hz.



Now export this filter to your workspace so you can use it to filter some data.

3 On the FDATool menu bar, select **File—>Export** to open the **Export** dialog.

4 To export the filter to your workspace as a filter object, select `Workspace` from **Export To** and select `Objects` for **Export As**. The export dialog looks like this after you make your selections.



5 Click **OK** to export the filter with the variable name shown in the **Export** dialog. When you return to your workspace in MATLAB, you see the new object. In this case, the new object is named `Hd`.

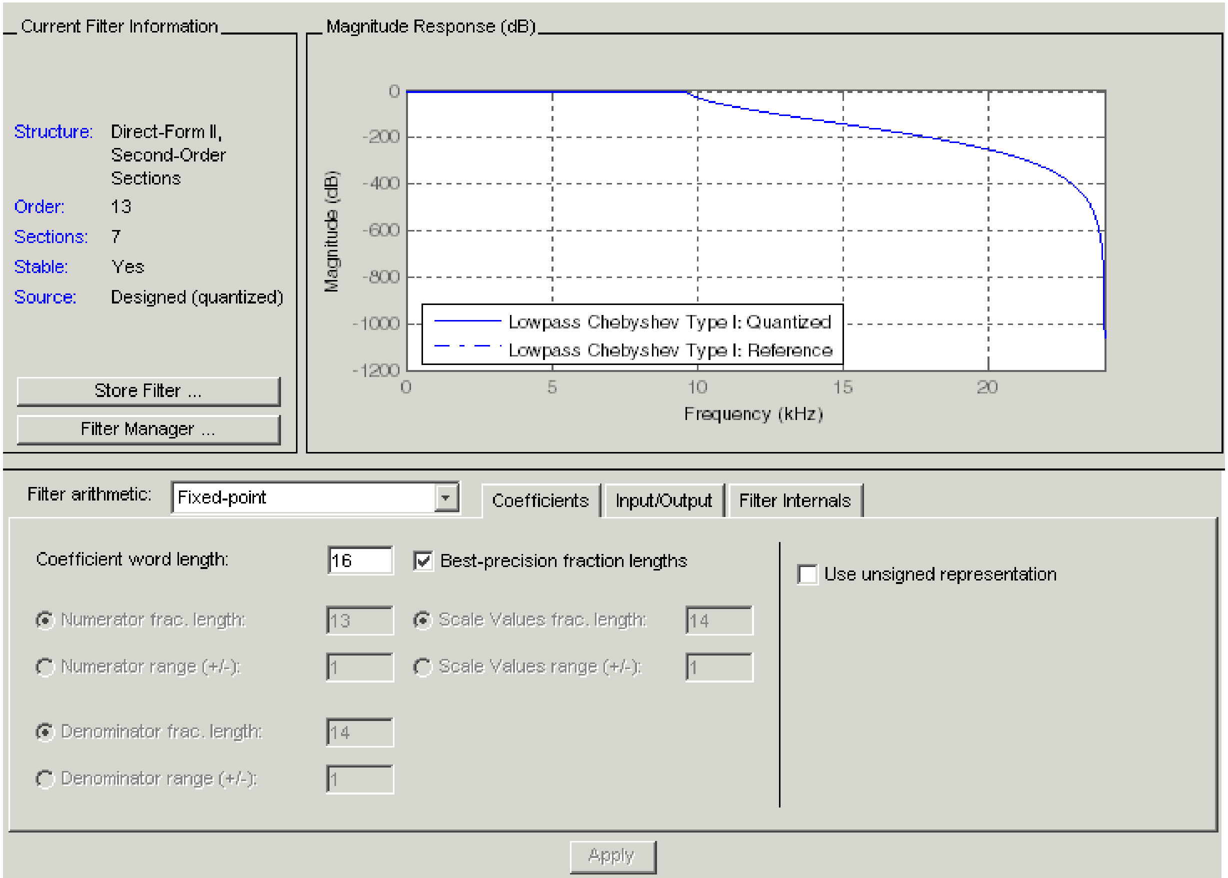
6 In MATLAB, create a vector of random data (with values between 0 and 1) and filter the data with `Hd`.

```
x=rand(1000,1);
y = filter(Hd,x);
```

Now `y` contains the data filtered by running `x` through the filter `Hd`.

7 Back in FDATool, click  on the side bar to switch FDATool to quantization mode.

8 With the quantization pane displayed in FDATool, switch **Filter arithmetic** to `fixed-point`. Now you see the quantization pane in FDATool, as shown in this figure.



In the analysis area, FDATool shows the magnitude responses for two filters—your fixed-point (quantized) filter and the reference filter that accompanies the fixed-point version. Turn on the filter legend (select **View—Legend** from the menu bar) to help you identify which response belongs to each filter.

Zooming in on the curves shows that the two filter responses are very similar. Note that your fixed-point filter used the default settings in the quantization pane—16-bit coefficients and fraction lengths selected to ensure the best precision.

9 Now export the quantized filter to your workspace as an object. Since you are going to use the same variable name `Hd` for the quantized filter in your workspace, select **Overwrite variables** in the **Export** dialog.

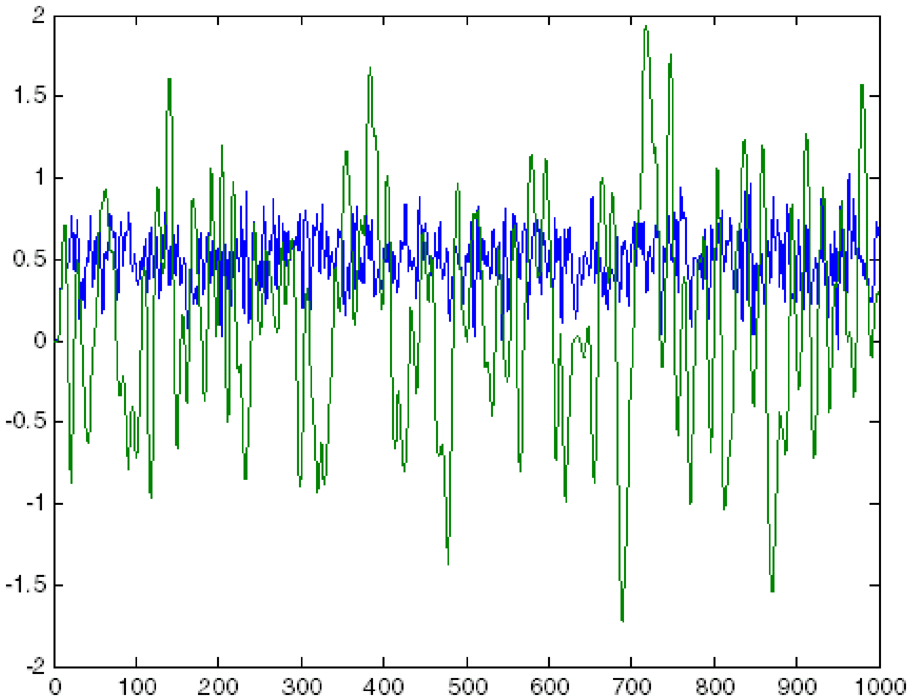
10 Back in MATLAB, perform the filter process again, using the quantized filter `Hd` and the signal `x`.

```
yq = filter(Hd,x);
```

11 This is the important step. Plot `y` and `yq` to see how the filtering process compared between the double-precision filter `Hd` and the fixed-point filter `Hd`.

```
plot([y,yq]) % The results are not close to matching.
```

A look at the plot reveals that the results of filtering the same data (`x`) with each filter were very different. Recall that the magnitude responses seemed to be the same. So quantizing the filter affected the filtering performance in a way that the magnitude response curve does not show. The answer is that the arithmetic performed by the filter after quantization is very different from the double-precision filter before quantization.

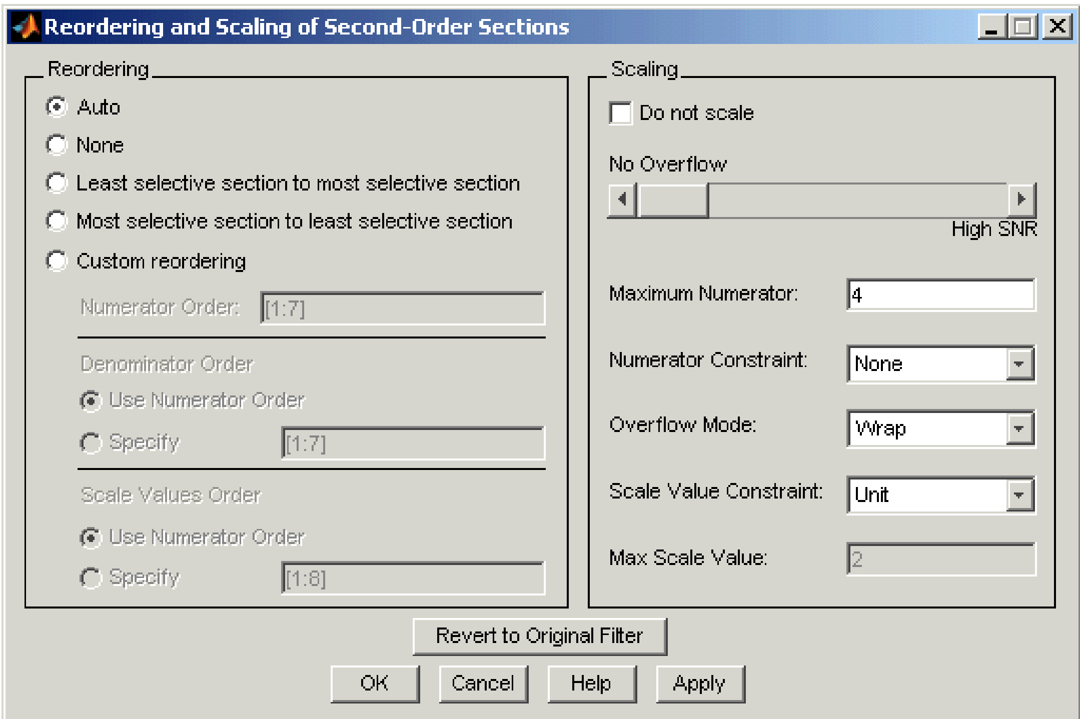


Again, return to FDATool, which should still be open on your desktop. You are going to fix the discrepancy between `y` and `yq` by reordering the sections of the fixed-point filter and scaling the filter to improve the performance after quantization.

1 To access the scaling and SOS filter reordering capability in FDATool, select **Edit—>Reorder and Scale Second-Order Sections** from the menu bar. The **Reorder and Scale Second-Order Sections** dialog opens, shown below. Note the default settings:

   - Automatic reordering is selected.
   - The scaling is set to prevent overflow in the filter.

- There is no maximum allowed value for the numerators (**Numerator Constraint** is `None`).
- **Overflow Mode** is `Wrap` to use wrapping when overflows occur.



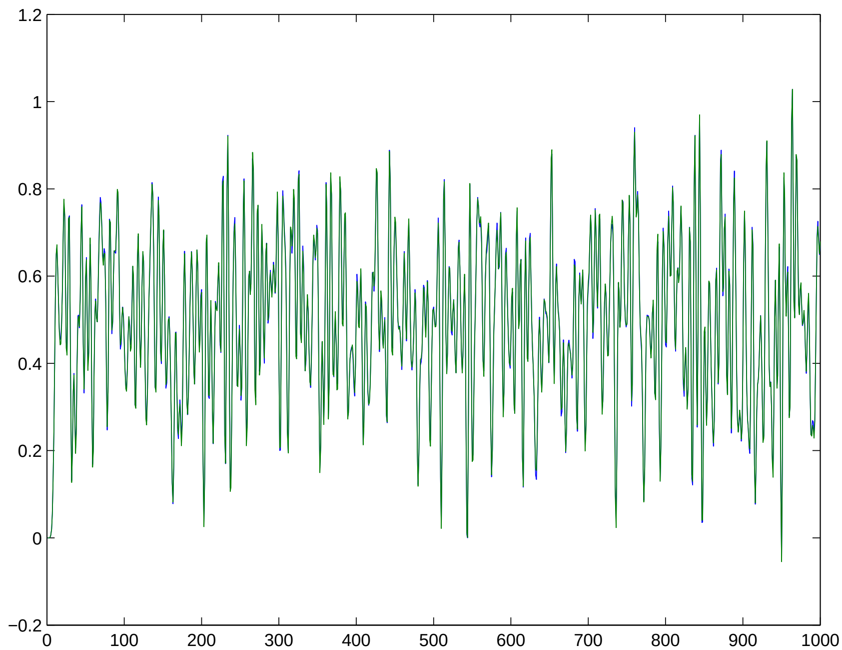
2. Review the default settings and if they match the figure shown, click **OK** to close the dialog and scale and reorder the filter.
3. One more time, export the now-scaled quantized filter to your workspace as `Hd`.
4. Filter the data `x` again, using the latest `Hd` filter—now reordered and scaled.

```
yqs = filter(Hd,x);
```

5. Finally, plot `y` and `yqs` to see if the filtering performance matches now.

```
plot([y,yqs]) % y and yqs are identical.
```

Here is the plot showing the results. Scaling and reordering the fixed-point filter restored the filtering performance to match the double-precision filter performance. The results demonstrate the power of scaling and reordering SOS filters.



Selected Bibliography

For further information about the algorithms and computer models used to design filters and apply quantization in the toolbox, refer to one or more of the following references.

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2

Designing Advanced Filters

The Optimal Filter Design Problem

Filter Design Toolbox provides you with the tools to design optimal filters in the finite impulse response (FIR) and infinite impulse response (IIR) domains.

Often, filter design techniques and algorithms result in filters that are easy to apply and put relatively light demands on computational systems. While these filters are acceptable in many instances, they are not optimal solutions to the filtering needs of some digital signal processing implementations. Suboptimal filter designs can meet the performance specifications for the filter, but generally at the expense of increased filter order. This can result in increased arithmetic computational load for each input sample and lower operating speed than may be possible and necessary.

You use the functions `firlpnorm`, `firgr`, `iirlpnorm`, and `iirlpnormc` to design optimal filters. The following sections review the optimal filter design problem and introduce the filter design functions included in the toolbox:

- "Optimal Filter Design Theory" on page 2-2
- "Optimal Filter Design Solutions" on page 2-5
- "Advanced FIR Filter Designs" on page 2-7
- "Examples—Using firgr and fircband to Design FIR Filters" on page 2-9
- "Advanced IIR Filter Designs" on page 2-42
- "Examples — Using Filter Design Toolbox Functions to Design IIR Filters" on page 2-43

Optimal Filter Design Theory

How do you design a filter that meets your performance needs, such as having the required passbands, stopbands, or transition regions, and is also the optimal solution? (The optimal solution filter minimizes a measure of the error between your desired frequency response and the actual filter response.)

Consider two filter frequency response curves:

- D(ω) — the response of your ideal filter, as defined by your signal processing needs and specifications
- H(ω) — the frequency response of the filter implementation you select

In the following figure you see the response curves for D(ω) and H(ω), both lowpass filters.

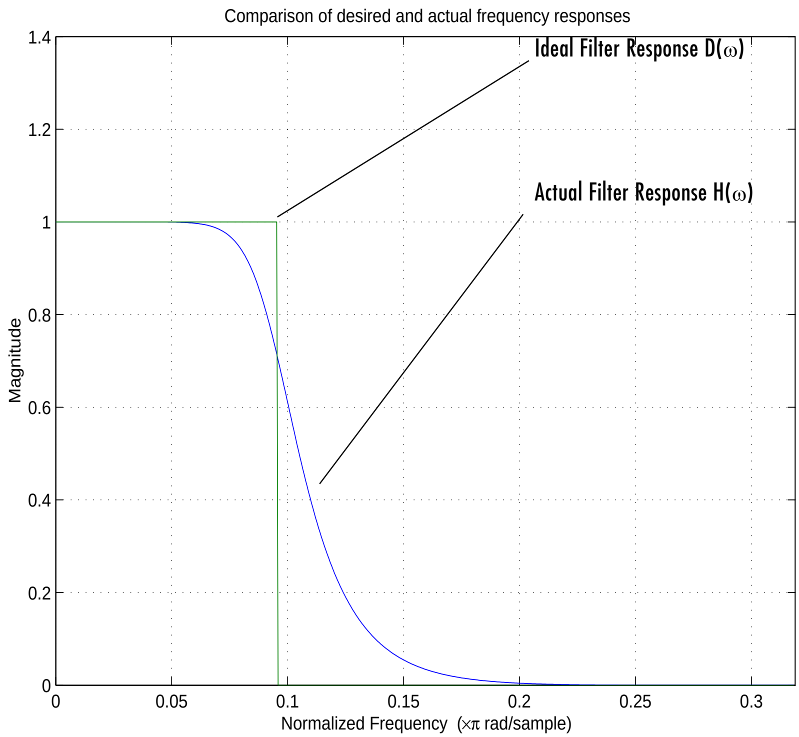


Figure 2-1: Response Curves for Ideal and Actual Lowpass Filters

Optimal filter design theory seeks to make H(ω) match D(ω) as closely as possible by a given measure of closeness.

More precisely, if we define a weighted error

$$E(\omega) = W(\omega)[H(\omega) - D(\omega)]$$

where $E(\omega)$ is the error between the ideal and actual filter response values and $W(\omega)$ is the weighting factor, the optimal filter design problem is to determine an $H(\omega)$ that minimizes some measure or norm of $E(\omega)$ given a particular weighting function $W(\omega)$ and a desired response $D(\omega)$.

$W(\omega)$, the weighting function, lets you determine which portions of the actual filter response curve are most important to your filter performance, whether passband response or attenuation in the stopband.

Usually, developers use the L_p norm to measure the error. This norm is given by

$$\int_{\Omega} [E[\omega]]^p$$

and this is the quantity we minimize.

In practice, the two most commonly used norms are L_2 and L_∞, meaning that p equals 2 and p equals infinity.

Filter designs that minimize the L_∞ are attractive because they lead to equiripple solutions. Their equiripple characteristics tend to produce the lowest order filter that satisfies some prescribed specification.

When p goes to infinity, L_∞ norm simplifies to

$$\max_{\omega \varepsilon \Omega} |E(\omega)|$$

meaning that when p equals ∞, the optimal design minimizes the maximum magnitude of the weighted error. Hence, it yields a *minimax* solution.

Notice that the L_p norm is computed over a region Ω that uses a subset of the positive Nyquist interval $[0,\pi]$. Ω covers the positive Nyquist interval except for certain frequency bands deemed to be “don’t care” regions or transition bands that are not included in the optimization.

Optimal Filter Design Solutions

We have stated that the optimal filter design problem is to find the filter whose magnitude response, $|H(\omega)|$, minimizes

$$\int_{\omega} [W(w)(|H(w)| - D(w))]^p dw$$

for a given Ω, p, $W(\omega)$ and $D(\omega)$. You can use both FIR and IIR filters to meet this requirement.

For the FIR case, with p equal to ∞, and the additional constraint that the filter must have linear phase, you can use a very efficient design method, based on the Parks-McClellan exchange algorithm to determine the optimal solution.

Function `firgr` in the toolbox implements this method. Additionally, `firgr` provides optional calling syntaxes that enable variations and enhancements to the filter design problem.

To design optimal FIR solutions in the general case where p is not necessarily equal to infinity, the toolbox includes the function `firlpnorm`. You may find this useful in cases where minimax solutions lead to abrupt time-domain responses. `firlpnorm` does not use the Remez exchange algorithm and generally takes longer to design a filter than `firgr` and other filter design functions. Moreover, `firlpnorm` is not constrained to linear phase filters.

Note that Signal Processing Toolbox provides the function `firls`, an efficient FIR linear phase solution to the optimal filter design problem in the least-squares sense, that is, when p equals 2.

IIR solutions to the optimal filter design problem are more involved than their FIR counterparts. Filter Design Toolbox offers two functions that design IIR filters that are optimal in the least-p norm sense: `iirlpnorm` and `iirlpnormc`.

`iirplnorm` uses a somewhat faster, unconstrained algorithm, while `iirplnormc` uses a constrained algorithm that designs an optimal filter that meets the specifications while restricting the maximum radius of its poles to a specified value less than one.

Elliptic filters, such as those you use the function `ellip` (in Signal Processing Toolbox) to design, are optimal IIR filters for the case p equals infinity, when the desired magnitude response is piecewise constant, and the filter numerator and denominator orders are the same.

The Parks-McClellan method, which implements the Remez exchange algorithm, produces a filter design that just meets your design requirements, but does not exceed them. In many instances, when you use the window method to design a filter, the result is a filter that performs too well in the stopband. This wastes performance and taxes computational power by using more filter coefficients than necessary. When you use a rectangular window in the window design method, the resulting filter can be shown to be the optimal, unweighted least squares solution to the filter design problem. In summary, the optimal solution is not always a good solution to the filter design problem.

Filters designed using the Parks-McClellan method have equal ripple in their passbands and stopbands. For this reason, they are often called *equiripple* filters. They represent the most efficient filter designs for a given specification, meeting your frequency response specification with the lowest order filter.

Advanced FIR Filter Designs

Filter Design Toolbox includes a function, `firgr`, for designing FIR filters that represent the optimal solutions to filter design requirements. `firgr` provides a minimax filter design algorithm that you use to design the following real FIR filters:

- Types 1 through 4 linear phase
- Minimum phase
- Maximum phase
- Minimum order, even or odd
- Extra-ripple
- Maximal-ripple
- Constrained-ripple
- Single-point band
- Forced gain
- Arbitrarily shaped frequency response

For examples of filters that use `firgr` design features, refer to "firgr Examples" on page 2-8.

`firgr` implements the Shpak-Antoniou algorithm described in "A generalized Remez method for the design of FIR digital filters," D.J. Shpak and A. Antoniou, published in *IEEE Trans. Circuits and Systems*, pp. 161-174, Feb. 1990.

FIR filters, when implemented nonrecursively, do not use feedback in their architectures. This limits the filter design so that you include current inputs to the filter, as opposed to including past outputs (feedback) to calculate the current output of the filter. In this toolbox, you use the function `firgr` to design FIR filters. Among other features, `firgr` lets you:

- Define filters that have arbitrary shape frequency response curves
- Set a range of performance limits for a filter
- Set the weighting for the error between the desired response and the actual response in each band of interest in a filter

`firpm` and `firgr` respond the same way to the same input and output arguments, where the input arguments are valid for both functions. `firgr` extends the Remez algorithm to support the new filter designs by adding new input argument options.

Note To provide improved FIR filter design optimization, `firgr` uses a generalized Remez algorithm that is not identical to the Remez algorithm used by `firpm`. Specifically, `firgr` uses a higher density frequency grid in filter transition regions, such as at the cutoff points. Thus the frequency grid is not constant, but changes density across the frequency spectrum, letting the algorithm more closely optimize filter performance in those areas.

For more straightforward filter designs, `firpm` and `firgr` generate the same filter coefficients and the same design. As the filter gets more complex, such as higher order or more bands or steeper transition regions, the filter designs may diverge. Generally, `firgr` provides better optimized filter designs in these cases.

Using `firgr` to design filters places the following restrictions on your designs:

- Design must be FIR.
- You can select the number of filter coefficients.
- The frequency response curve must be divided into a series of passbands and stopbands separated by transition or “don’t care” bands.
- Within each passband and stopband, you specify your desired amplitude response as a piecewise constant function.
- You cannot constrain the amplitude response in transition bands.

With these considerations in place, `firgr` designs *equiripple*, or *minimax*, filters to meet your specifications.

firgr Examples

Each of these examples uses one or more features provided in the function `firgr`. Review each example to get an overview of the capabilities of the function.

Examples—Using firgr and fircband to Design FIR Filters

`firgr` provides a wide range of new capabilities for FIR filter design. Because of the comprehensive nature of the generalized Remez algorithm, the best way to learn what you can do with the new function is by example. This section presents a series of examples that investigate the filters you can design through `firgr`. You can view these examples as a demonstration program in MATLAB by opening the MATLAB demos and selecting `Filter Design` from `Toolboxes`. Listed there you see a number of demonstration programs. Select `Minimax FIR Filter Design` to see function `firgr` used to create many filters, from a lowpass filter to a constrained stopband design to a minimum phase, lowpass filter with a constrained stopband.

To open the FIR filter design demo.

Follow these steps to open the FIR filter design demo in MATLAB.

1. Start MATLAB.
2. At the MATLAB prompt, enter `demos`.

   The **MATLAB Demo Window** dialog opens.
3. On the left-hand list, double-click `Toolboxes` to expand the directory tree.

   You see a list of the toolbox demonstration programs available in MATLAB.
4. Select `Filter Design`.
5. Select `FIR Filter Design`.
6. From the right-hand pane, select `Minimax FIR Filter Design`.

A few examples include comparisons to other filter designs and some include analysis notes. For details about using function `firgr`, refer to Chapter 9, “Function Reference.” While this set of examples covers some of the options for `firgr`, many options exist that do not appear in these examples. Examples cover common or interesting `firgr` options to demonstrate some of the capabilities.

In each of the examples in this section, we use the output argument `res` to return the structure `res` that contains information about the filter.

Structure res Element	Contents
`res.order`	Filter order.
`res.fgrid`	Vector containing the frequency grid used in the filter design optimization.
`res.H`	Actual frequency response on the grid in `fgrid`.
`res.error`	Error at each point on the frequency grid (desired response- actual response).
`res.des`	Desired response at each point on `fgrid`.
`res.wt`	Weights at each point on `fgrid`.
`res.iextr`	Vector of indices into `fgrid` of extremal frequencies.
`res.fextr`	Vector of extremal frequencies.
`res.iterations`	Number of Remez iterations for the optimization.
`res.evals`	Number of function evaluations for the optimization.
`res.edgeCheck`	Results of the transition-region anomaly check. Computed when the '**`check`**' option is specified. One element returned per band edge. Returned values can be: • `1` = OK • `0` = Probable transition-region anomaly • `-1` = Edge not checked. In the normalized frequency domain, the edges at f=0 and f=1 cannot have anomalies and are not checked.

Example—Designing a Minimax Filter

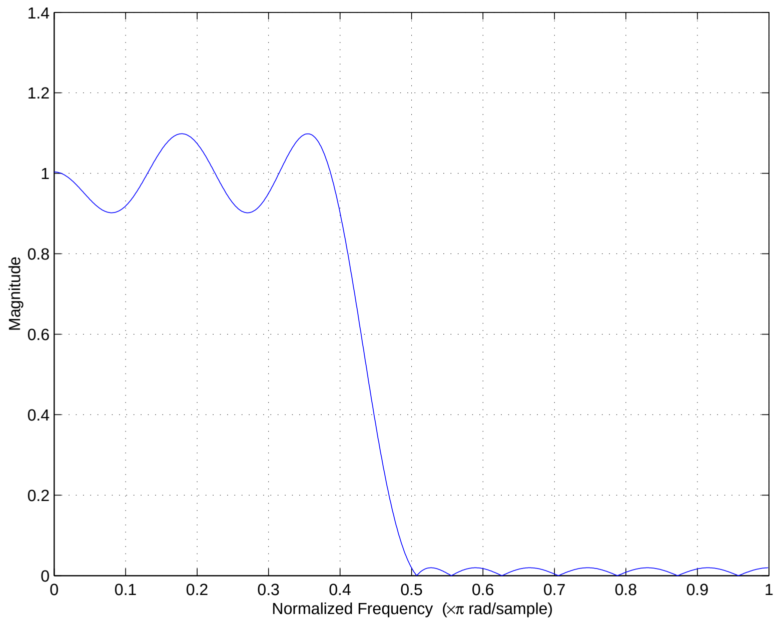
To use `firgr` to design an equiripple or minimax filter, we use the following statement.

```
[b,err,res] = firgr(22,[0 0.4 0.5 1],[1 1 0 0],[1,5]);
```

If you use the same statement, replacing `firgr` with `firpm`, you get the same filter. You can reproduce any filter that `firpm` generates by replacing `firpm` with `firgr` in the statement. `firgr` retains full compatibility with `firpm`.

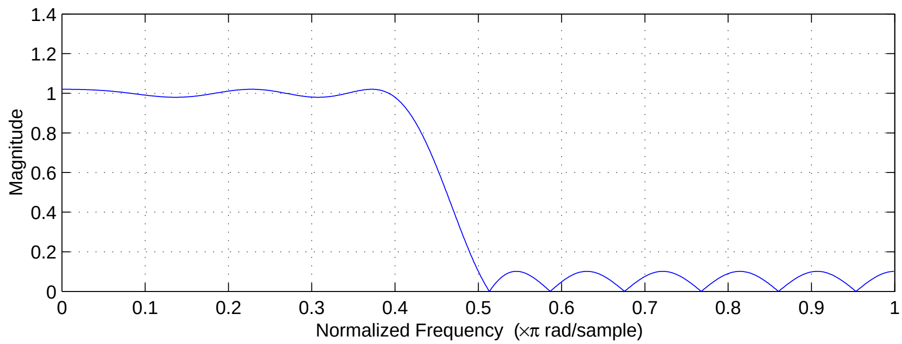
Here's a plot of the magnitude response of the minimax filter as created by `firgr`. The following code creates this figure, after adjusting the *y*-axis limits.

```
fvtool(b);
```



Our filter ends up as a 22nd-order filter with magnitude response that has ripples about 1 in the passband and ripples about 0 in the stopband. Using the weight vector, we chose to emphasize meeting the stopband performance to be five times as important as meeting the passband performance. Hence the reduced ripple in the stopband relative to the passband. In the next figure, we switch the weighting to emphasize the passband, and see that the passband ripple is much smaller than the stopband ripple.

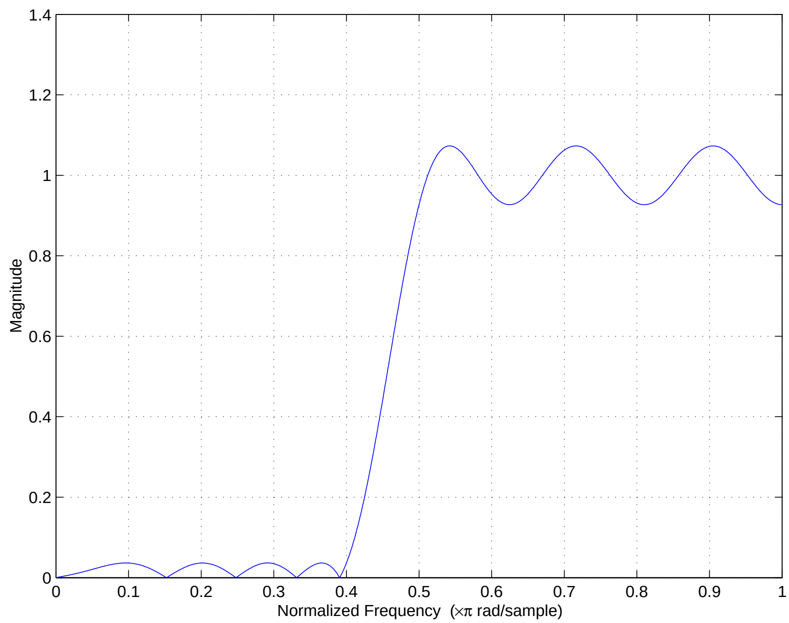
```
[b,err,res] = firgr(22,...,[5,1]);
plot(res.fgrid,abs(res.H))
```



Example—Designing a Minimax Filter, Odd-Order, Antisymmetric

In this example, `firgr` designs a filter that `firpm` cannot. When you evaluate the following code in MATLAB, the result is a minimax FIR filter, this time having odd-order and antisymmetric structure, known as type 4. You can see from the figure that the magnitude response now represents a high pass filter. In this example, we specify the filter as type 4 (`'4'` in the statement) to get the odd-order, antisymmetric design we want.

```
[b,err,res]=firgr(21,[0 0.4 0.5 1], [0 0 1 1],[2 1],'4');
[H,W,S]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot(H,W,S);
```



We have weighted the stopband more heavily than the passband ([2 1]) in the function syntax. The 2 and 1 tell `firgr` that we care about meeting the stopband specification twice as much as the passband specification. Notice that the weighting is relative, not absolute. Our weights say that the stopband is twice as important as the passband. They do not specify the weighting in absolute terms.

Example—Designing a "Least Squares-Like" Filter

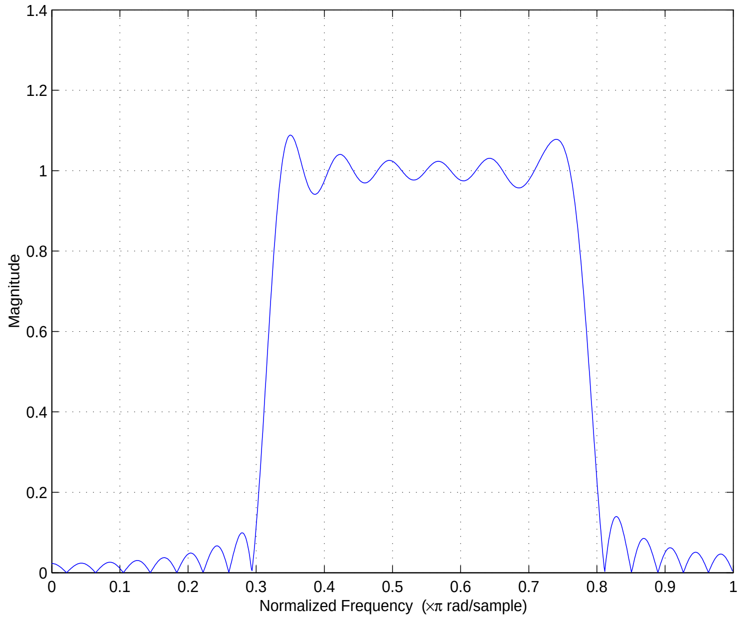
`firgr` lets you design filters that resemble least squares design. In this example, we design a 53rd-order filter and use the user-supplied file `taperedresp.m` to specify a frequency response weighting function to perform the error weighting for the design. So you can reproduce this example, the file `taperedresp.m` is in the *`matlabroot`*`\toolbox\filterdesign\filtdesdemos` folder. `taperedresp.m` contains the following code to specify the weighting.

```
% Example for a user-supplied frequency-response function
% taperedresp.m

function [des,wt] = taperedresp(order, ff, grid, wtx, aa)
nbands = length(ff)/2;
% Create output vectors of the appropriate size
des=grid;
wt=grid;

for i=1:nbands
   k = find(grid >= ff(2*i-1) & grid <= ff(2*i));
   npoints = length(k); t = 0:npoints-1;
   des(k) = linspace(aa(2*i-1), aa(2*i), npoints);
   if i == 1
      wt(k) = wtx(i) * (1.5 + cos((t)*pi/(npoints-1)));
   elseif i == nbands
      wt(k) = wtx(i) * (1.5 + cos(pi+(t)*pi/(npoints-1)));
   else
      wt(k) = wtx(i) * (1.5 - cos((t)*2*pi/(npoints-1)));
   end
end
```

To generate the least-squares-like filter, use the following code.

```
[b,err,res]=firgr(53, [0 0.3 0.33 0.77 0.8 1],...
{'taperedresp',[0 0 1 1 0 0]}, [2 2 1]);
[H,W,S]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot(H,W,S);
```

When you issue these statements at the MATLAB prompt, you get the following plot for the filter magnitude response.

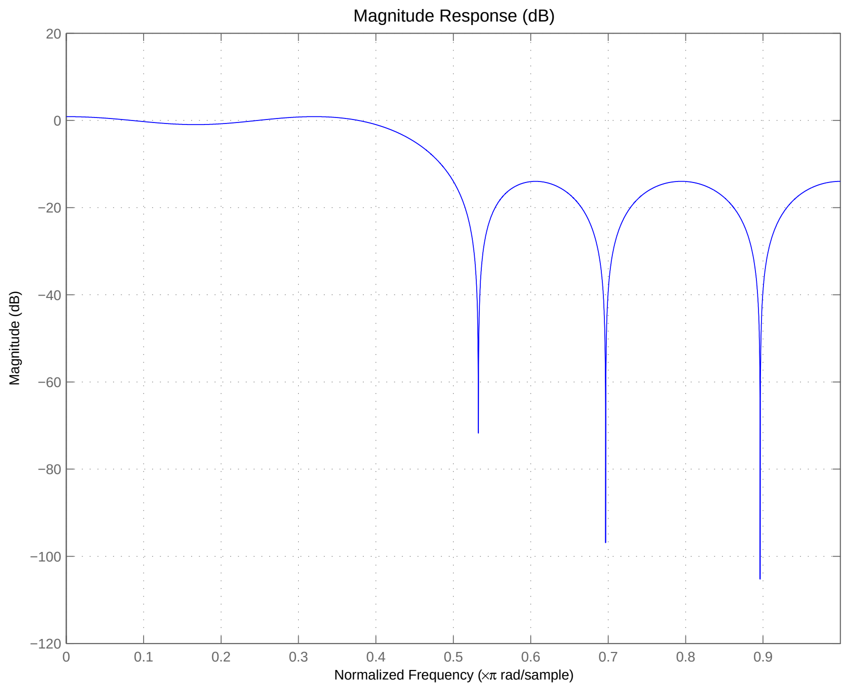


Example—Designing a Constrained Lowpass Filter with fircband

With `fircband`, you can both apply weighting to the passband and apply a limit or *constraint* to the error in the stopband, called constraining. Limiting the stopband error can be useful in circumstances where your filter must meet a specified stopband requirement. To create a lowpass filter with a constrained stopband and weighted passband response, we use `fircband` with the `'w'` optional input argument to weight the passband. The optional input argument `'c'` constrains the filter stopband error not to exceed `0.2`. Note that to use the constraining and weighting options, your filter must have at least one unconstrained band. That is, cell array `c` must contain at least one `'w'` entry. In our example, `c` is `{'w' 'c'}`.

```
b=fircband(12,[0 0.4 0.5 1], [1 1 0 0], [1 0.2],{'w' 'c'});
fvtool(b);
```

The next figure shows the lowpass filter with the constraints applied.



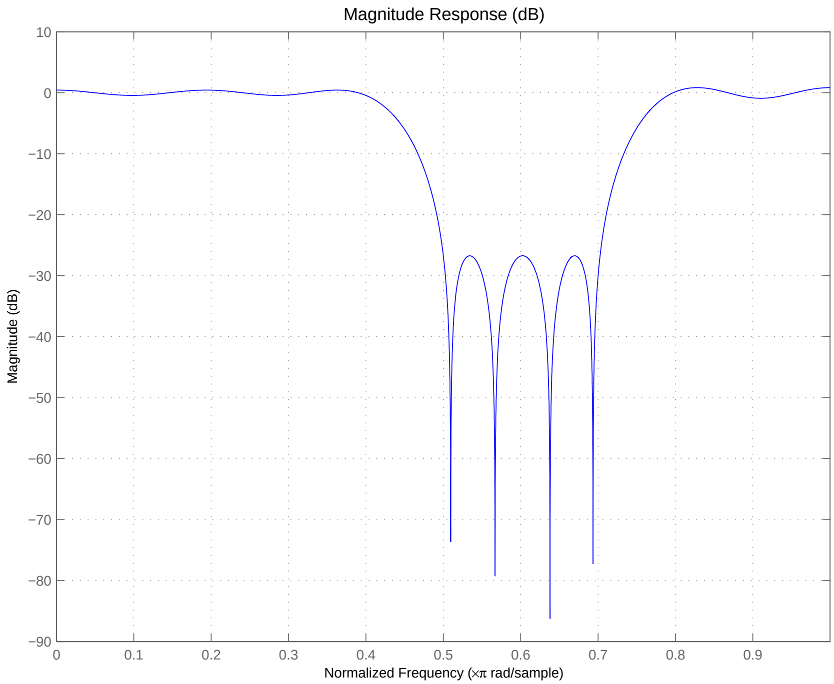
When you use constraining values in your `fircband` filter design, check to see that your filter actually touches the constraining value in the stopband. If it does not, increase the error weighting (`'w'`) for your unconstrained bands. This change causes the constrained errors to approach the constraint value more quickly. Notice that the plot shows our filter just touches the desired constraint of 0.2.

Example—Designing a Constrained Bandstop Filter with fircband

Continuing with the concept of using weighting in `fircband`, we design a bandstop filter whose passband ripple we constrain not to exceed 0.05 and 0.1. In this instance, cell array `c` is `{'c' 'w' 'c'}` to constrain the passbands and we use the optional input vector `W=[0.05 1 0.1]` to constrain the passband ripple not to exceed 0.05 and 0.1.

```
b=fircband(22,[0 0.4 0.5 0.7 0.8 1], [1 1 0 0 1 1],[0.05 1 0.1],...
{'c' 'w' 'c'});
fvtool(b)
```

As expected the magnitude response shows different peak ripple values in the passbands — 0.05 for the low frequency band and 0.1 for the high frequency band.



Example—Designing a Single-Point Band Filter

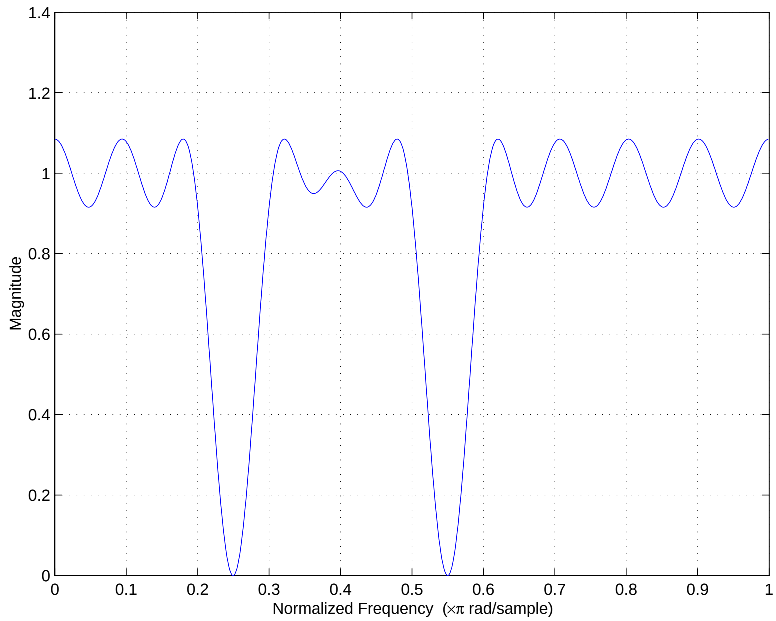
The following statements

```
[b,err,res]=firgr(42,[0 0.2 0.25 0.3 0.5 0.55 0.6 1],...
[1 1 0 1 1 0 1 1], {'n' 'n' 's' 'n' 'n' 's' 'n' 'n'});
[H,W,S]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot(H,W,S);
```

generate an interesting filter that you cannot design when you use functions in Signal Processing Toolbox: a multiple stopband filter where the stop bands are defined by single points. In the `firgr` command in this example, the syntax is `b=firgr(N,F,A,S)`. The input vectors `F`, `A`, and `S`, each containing eight values, define the response curve for the filter.

Input Vector	Use
F=[0 0.2 0.25 0.3 0.5 0.55 0.6 1]	Defines the points of interest in the frequency response. In this case, you are working with frequencies normalized between 0 and 1.
A=[1 1 0 1 1 0 1 1]	Set the gain at each frequency point.
S={'n' 'n' 's' 'n' 'n' 's' 'n' 'n'}	Specifies whether the frequency points represent normal or single-point bands. By comparing the frequency and type vector entries, we see that `F=0.25` and `F=0.55` are single point bands (marked by `s`), and the gain at those points is `0`. The other bands are normal bands (marked with `n`) with gain =1.

From the next figure, you see that the filter has just the response we defined, with zeros at F = 0.25 and F = 0.55.

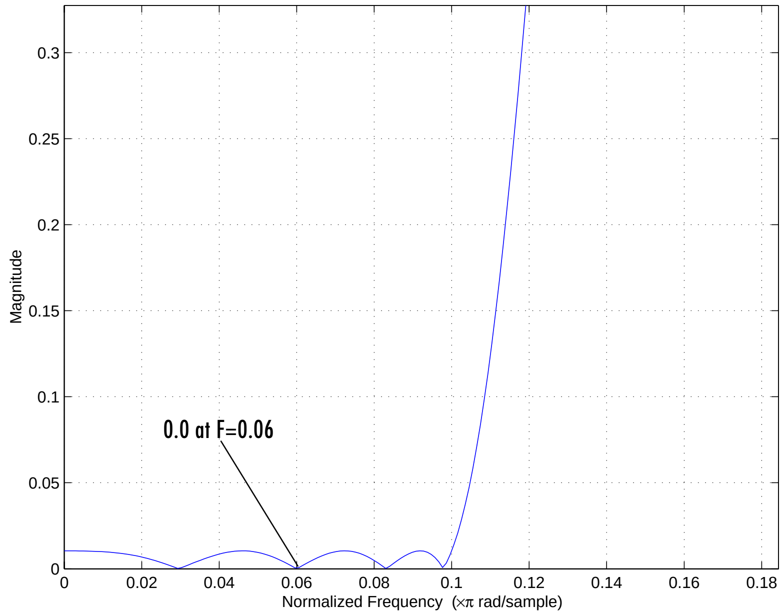


Example—Designing a Filter with a Specified In-band Value

In some filter design tasks, you want a filter whose inband value you determine exactly. For example, you might want a 60 Hz noise rejection filter to have zero gain at F = 0.06 (F = 60 Hz in real frequency). For this example, the sampling frequency is 2 KHz, so 60 Hz is F = 0.06 when we normalize the frequency. We use the following code example to design such a filter.

```
[b,err,res]=firgr(82,[0 0.055 0.06 0.1 0.15 1], [0 0 0 0 1 1],...
{'n' 'i' 'f' 'n' 'n' 'n'});
[H,W,S]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot(H,W,S);
```

At F = 0.06, we require the gain of the filter response to be exactly 0.0. So we force the gain at F = 0.06 to zero by adding the `'f'` input option to the `S` vector. As shown in the plot, the filter response is zero at F = 0.06, and the resulting filter rejects 60 Hz noise quite effectively.



You might have noticed in the `firgr` statement that the `S` vector includes an `'i'` option. Entries in the `S` vector have any of the following values.

Vector Symbol	Meaning
`n`	Represents a normal frequency point
`s`	Represents a single-point band frequency
`f`	Forces the gain at this frequency to a fixed value, as specified in the weighting vector `W`
`i`	Represents an indeterminate frequency point. Usually used when the band should abut the next band

For our noise rejecting filter, the sampling frequency is 2 KHz, so 60 Hz is f=0.06 in normalized frequency.

Example—Designing Extra-Ripple and Maximal-Ripple Filters

Extra-ripple and maximal-ripple filters have some interesting properties:

- They have locally minimum transition region widths
- They tend to converge very quickly

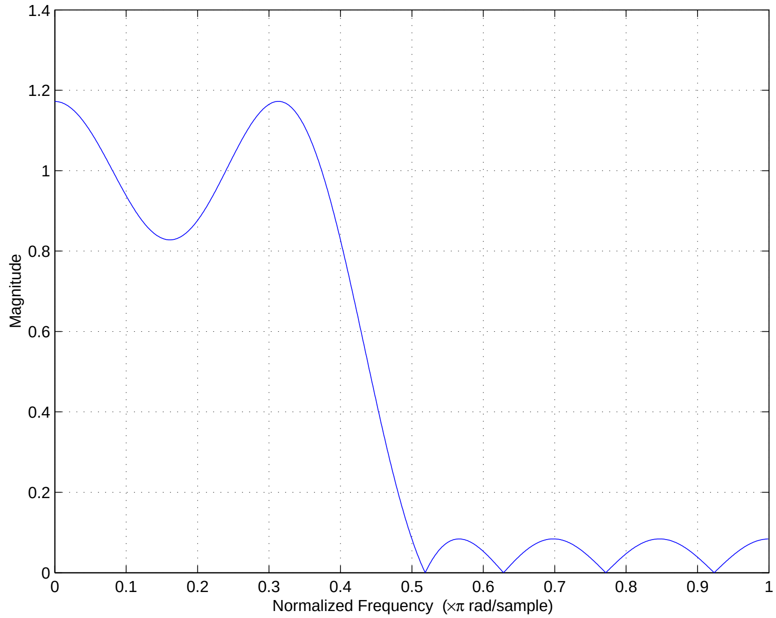
`firgr` lets you use multiple independent approximation errors to directly design extra- and maximal ripple filters. In this example, we use independent errors to design two filters, then we revisit our 60 Hz noise rejection filter to compare these two different approaches to designing the same filter.

Example of an Extra-Ripple Lowpass Filter

The code to design our extra-ripple filter is

```
[b,err,res]=firgr(12,[0 0.4 0.5 1], [1 1 0 0], [1 1],...
{'e1' 'e2'});
[H,W,S]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot(H,W,S);
```

The last entries in the command, `[1 1]` and `{'e1' 'e2'}`, are the vectors `W` and `E` that determine the weights and independent approximation errors for filters with special properties. `'e1'` is applied to the passband and `'e2'` applied to the stopband. Where the `firgr` algorithm usually results in equiripple filters, using the approximations lets `firgr` adjust the ripple in each band separately, as we have done in this design.

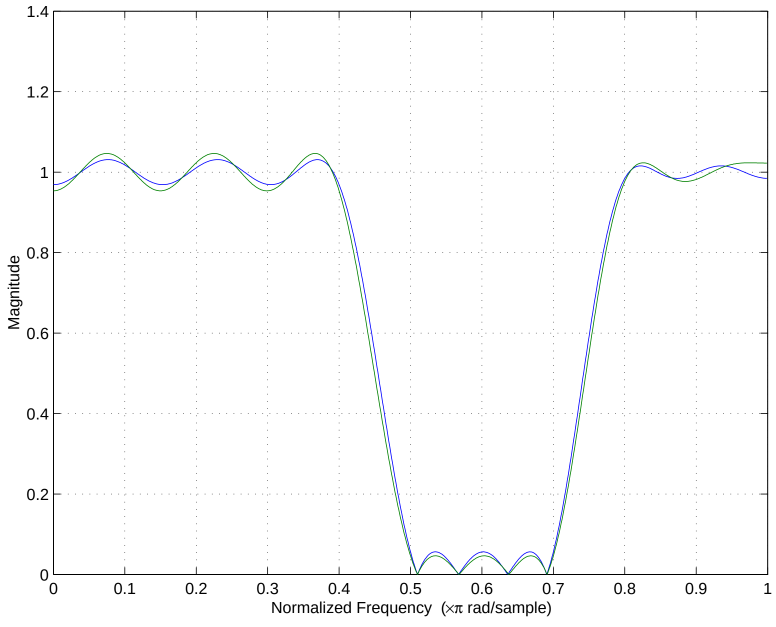


Example of an Extra-Ripple Bandstop Filter With Two Independent Approximation Errors

Now we extend the extra-ripple concept by using two independent error approximations. The two passbands share the first approximation error `'e1'`. The stopband uses `'e2'`. So you can see the effectiveness of this design approach, also create and plot a single approximation error filter for comparison.

```
[b,err,res]=firgr(28,[0 0.4 0.5 0.7 0.8 1], [1 1 0 0 1 1],...
[1 1 2], {'e1' 'e2' 'e1'}); % Extra-rippple filter design
[b2,err2,res2]=firgr(28,[0 0.4 0.5 0.7 0.8 1],...
[1 1 0 0 1 1],[1 1 2]); % Weighted-Chebyshev design
[H,W]=freqz(b,1,1024);[H2,W]=freqz(b2,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot([H H2],W,S);
```

In the figure, the responses are similar for the two designs, but the extra-ripple design shows less ripple in the passbands and slightly more in the stopband. If you evaluate the example code in MATLAB to create the plot, you can select **Zoom in** from the **Tools** menu in the figure window to examine the curves more closely.



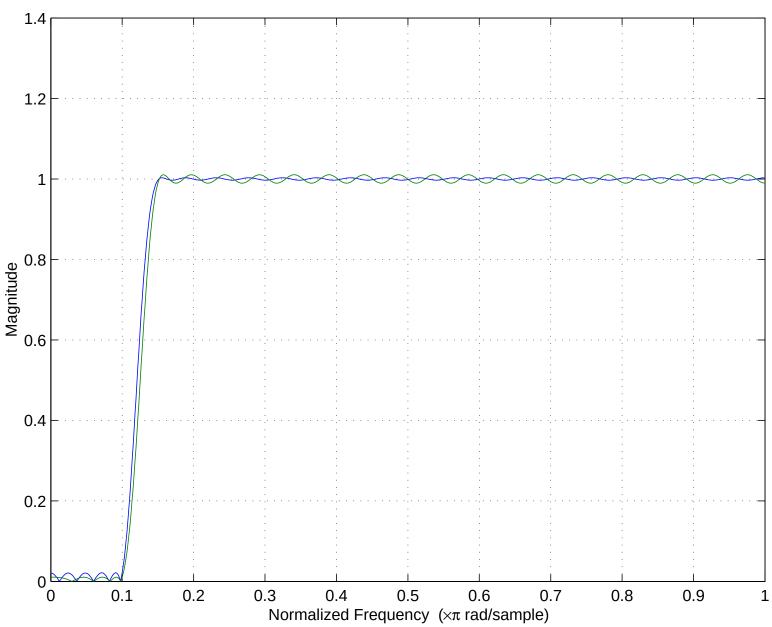
For this design, we let `firgr` use the same error approximation for the passbands and a different one in the stopband. The result is a filter that has minimum total error in the passbands, and minimum error in the stopband.

Example—Comparing Two 60 Hz Noise Rejection Filters

With the extra-ripple filter design technique available in `firgr`, we can use two different design techniques to redo our 60 Hz noise rejection filter. We use three independent error approximations in this design, one for each band, as shown in the following code.

```
[b,err,res]=firgr(82,[0 0.055 0.06 0.1 0.15 1],[0 0 0 0 1 1],...
{'n' 'i' 'f' 'n' 'n' 'n'},[10 1 1],{'e1' 'e2' 'e3'}); % New filter
[b2,err,res]=firgr(82,[0 0.055 0.06 0.1 0.15 1],...
[0 0 0 0 1 1], {'n' 'i' 'f' 'n' 'n' 'n'}); % Original filter
[H,W]=freqz(b,1,1024);
[H2,W]=freqz(b2,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot([H H2],W,S);
```

We have included the second `firgr` statement in this example to reproduce the earlier noise rejection filter for comparison. We plot them on the same figure for easy reference. In the stopband, the original design has lower ripple; the new, independent error design has less ripple in the passband. Also, the new filter has slightly steeper transition region performance.



Using independent approximation errors, as we did in this filter when we specified `'e1'`, `'e2'`, and `'e3'`, can result in better filter performance. The strings `'e1'`, `'e2'`, and so on direct `firgr` to consider the associated band alone, or with other bands that use the same error approximation. By assigning independent errors to each band, we let the generalized Remez algorithm used by `firgr` minimize the error in each band without considering the error in the other bands. If we do not use independent errors, the algorithm minimizes the total error in all bands at once.

At times, you need to use independent approximation errors to get designs that use forced inband values to converge. Error approximations are needed where the polynomial used to approximate the filter becomes undetermined when you try to force the inband values to converge.

Example—Checking for Transition-Region Anomalies

To allow you to check your filter designs for anomalies, `firgr` provides an input option called `'check'`. With the check option included in the command, `firgr` reports anomalies in the response curve for the filter. An anomaly in `firgr` is defined as out-of-the-ordinary response behavior in a transition, or "don't care," region of the filter response.

To demonstrate anomaly checking, we use `firgr` to design a filter with an anomaly, and include the `'check'` optional input argument.

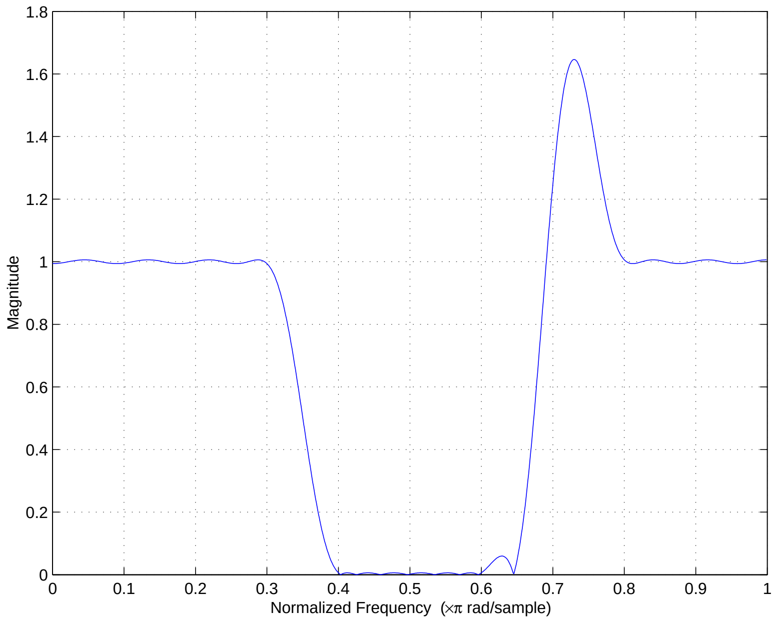
```
[b,err,res]=firgr(44,[0 0.3 0.4 0.6 0.8 1],...
[1 1 0 0 1 1],'check');
[H,W]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot(H,W,S);
```

With the `'check'` option, `firgr` returns the results vector `res.edgeCheck` in the structure `res`. Each zero-valued entry in this vector represents the location of a probable anomaly in the filter response. Entries that are not checked, such as the edges at f=1 and f=0, have -1 entries in `res.edgeCheck`.

To check for anomalies, the following command returns the vector containing the edge check results.

```
res.edgeCheck

ans =

    -1
     1
     1
     0
     0
    -1
```

There are anomalies between the f=0.6 and f=0.8 edges, as shown clearly in the figure. This represents a transition region for our filter. Notice that the edges at f=0 and f=1 were not checked.



In our example, the anomalous behavior happened because of the width of the transition region. When we define a narrower transition band, the anomaly disappears. Generally, reducing the transition region width eliminates anomalies in the filter response.

Example—Using Automatic Minimum Filter Order Determination

Rather than entering the filter order `N` in the `firgr` command, you can let the generalized Remez algorithm determine the minimum order for your filter. You set the specifications for the filter and the generalized Remez algorithm repeatedly designs the filter until the design just meets your specifications.

You have three options for setting the minimum order option for the filter:

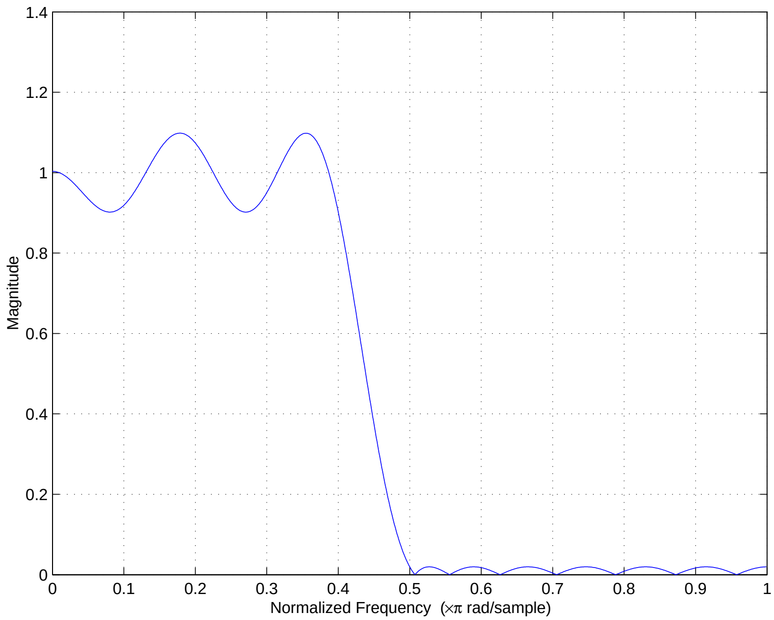
- `'minorder'` directs the Remez algorithm to iterate over the filter design until it finds a design that just fulfills your design specifications and is the lowest possible order. Using this option directs `firgr` to use `firpmord` to get an initial estimate of the filter order.
- `'mineven'` directs the Remez algorithm to iterate over the filter design until it finds a design that just fulfills your design specifications and is the lowest possible even order.
- `'minodd'` directs the Remez algorithm to iterate over the filter design until it finds a design that just fulfills your design specifications and is the lowest possible odd order.

Note When you use the minimum order option `'minorder'`, `firgr` treats the weights in the `W` vector as maximum error values for the associated frequencies in the frequency vector `F`. Also, constraints become absolute limits; `firgr` designs a filter that does not exceed the constraints.

For this example, we let the Remez algorithm find a minimum order filter that implements a lowpass filter with a transition band between f=0.4 and f=0.5.

```
[b,err,res]=firgr('minorder',[0 0.4 0.5 1], [1 1 0 0],...
[0.1 0.02]);
[H,W,S]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot(H,W,S);
```

Our filter, shown in the figure, demonstrates the desired ripple in the passbands and stopbands, 0.1 and 0.02; the transition region meets our specifications; and the filter order (found from `res.order`) is 22.



When you use the minimum order feature, you can specify the initial order (your best guess) in the `firgr` statement. When you estimate the order, `firgr` does not use `firpmord` to make an estimate of the filter order. This is important when `firpmord` does not support your desired filter type, such as differentiators and Hilbert transformers, as well as for filters that use frequency response functions that you supply. For the following filter example, we provide an initial estimate of 18 for the filter order, and we specify that we want our filter to have the minimum even order possible by adding the '`mineven`' option.

```
[b,err,res]=firgr({'mineven',18},[0 0.4 0.5 1], [1 1 0 0],...
[0.1 0.02]);
[H,W,S]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
```

```
freqzplot(H,W,S);
```

Though we provided an initial estimate of 18 for the order, the final order for our filter is again 22. If we had specified 'minodd', the result would be a 23rd-order filter.

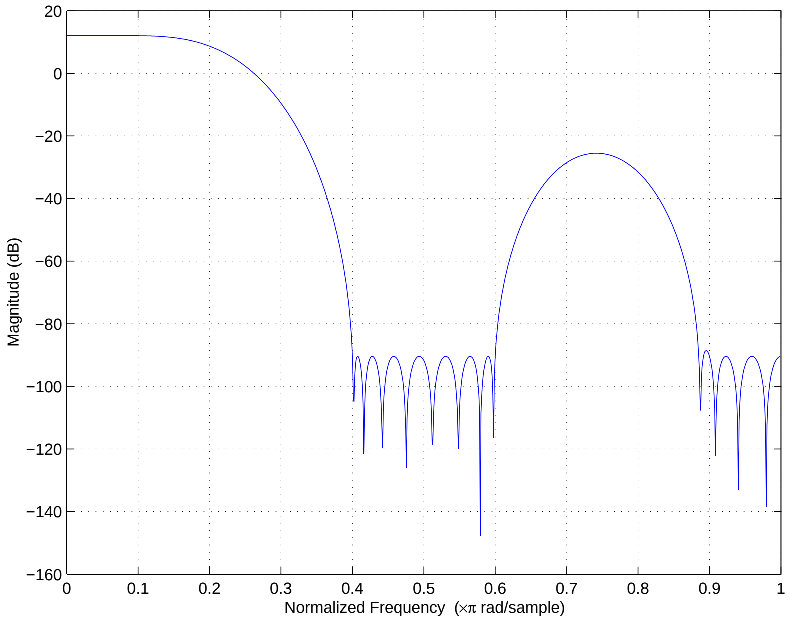
Example — Designing an Interpolation Filter

Now let us design an interpolation filter. These are usually used to upsample a band-limited signal by an integer factor, for example after the signal has been decimated by downsampling. Upsampling is often used while designing multirate filters to reduce the computational load required to use a filter. In Signal Processing Toolbox, you can use the function `intfilt` to design an interpolation filter. While `intfilt` provides a way to design the filter, it does not provide the control that `firgr` offers. Input options for `firgr` let you define the filter response and errors in each passband and stopband, and the weighting of the band responses in the filter design.

```
[b,err,res]=firgr(30,[0 0.1 0.4 0.6 0.9 1], [4 4 0 0 0 0],...
[1 100 100]);
[H,W]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'db';
freqzplot(H,W,S);
```

We specify a 30th-order filter with edges at 0.1, 0.4, 0.6, and 0.9, and weight them as [1 100 100]. The resulting design has stopbands between f=0.4 and f=0.6, and f=0.9 and f=1.0.

The next figure shows a filter designed by `firgr`.



Example—Comparing Filters Designed by firgr and intfilt

Now, to see that `firgr` lets you develop a better interpolation filter than `intfilt`, we compare filters designed by both functions. We need three sets of code to display the filters for our comparison — the first set generates the detail plot of the first stopband, the second set displays the second stopband in detail, and the third plot focuses on the stopband ripple. To keep the frequency response displays consistent, we use the MATLAB `plot` function to ensure that the axes and labels are the same for both filters. `freqzplot` does not provide enough control of the plotting functions.

Code to display the first stopband.

```
[b,err]=firgr(30,[0 0.1 0.4 0.6 0.9 1], [4 4 0 0 0 0],...
[1 100 100]);
b2=intfilt(4, 4, 0.4);
w=linspace(0.4, 0.6)*pi; h=freqz(b,1,w); h2=freqz(b2,1,w);
plot(w/pi,20*log10(abs([h' h2']))); ylabel('Stopband #1 (dB)');
v=axis; v=[0.4 0.6 -100 v(4)]; axis(v);
```

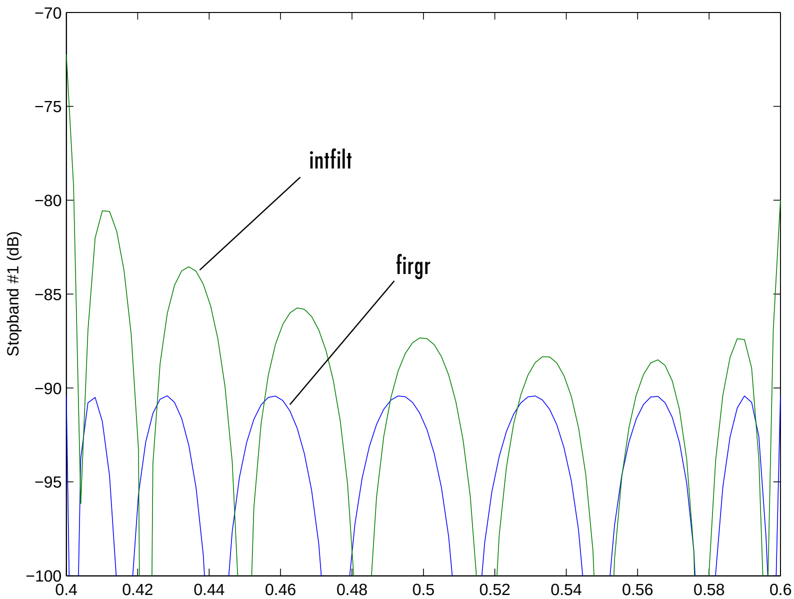
Code set to display the second stopband.

```
[b,err]=firgr(30,[0 0.1 0.4 0.6 0.9 1], [4 4 0 0 0 0],...
[1 100 100]);
b2=intfilt(4, 4, 0.4);
w=linspace(0.9, 1)*pi; h=freqz(b,1,w); h2=freqz(b2,1,w);
plot(w/pi,20*log10(abs([h' h2'])));  ylabel('Stopband #2 (dB)');
v=axis; v=[0.9 1 -100 v(4)]; axis(v);
```

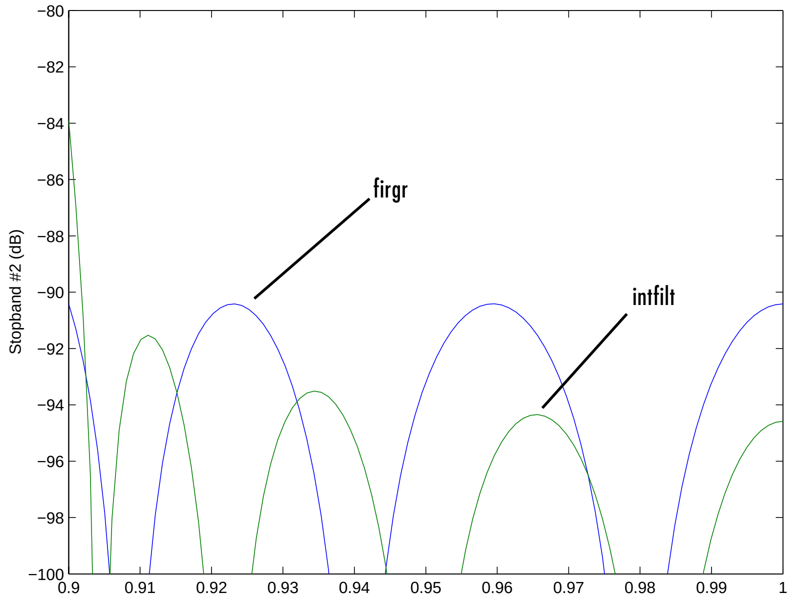
Code set to display the passband ripple.

```
[b,err]=firgr(30,[0 0.1 0.4 0.6 0.9 1], [4 4 0 0 0 0],...
[1 100 100]);
b2=intfilt(4, 4, 0.4);
w=linspace(0, .1)*pi; h=freqz(b,1,w); h2=freqz(b2,1,w);
plot(w/pi,20*log10(abs([h' h2'])));  ylabel('Passband (dB)');
```

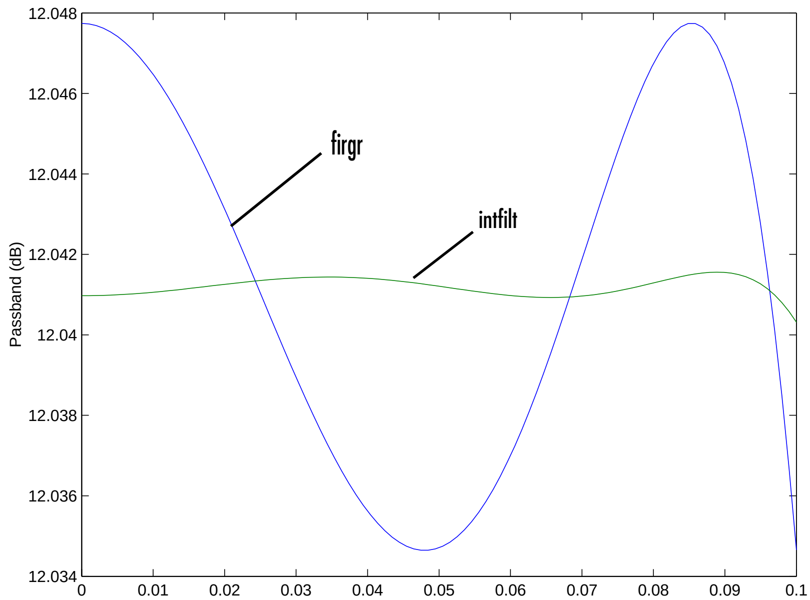
In the next figure, showing the first stopband in detail, you see that using the weighting function in `firgr` improved the minimum stopband attenuation by almost 20 dB over the `intfilt` design.



If we switch to a plot of the second stopband, shown in the next figure, you see that the equiripple attenuation throughout the band is about 6 dB larger for the `firgr`-generated filter than the minimum stopband attenuation of the filter designed by `intfilt`.



Finally, let's look at the passbands of the two filters, shown in the next figure. Here, the ripple in the `firgr`-designed filter is slightly larger than the passband ripple for the `intfilt` design. Still, both are very small, less than 0.014 dB peak-to-peak.



Example—Designing a Minimum Phase Lowpass Filter with a Constrained Stopband

With `firgr` you can determine whether the FIR filter you design is minimum phase, maximum phase, or linear phase. Through this example we show a minimum phase filter and look at the roots of the filter transfer function to see that no roots lie outside the unit circle in the z-plane. First, we create the minimum phase filter by using `firgr` with the '`minphase`' optional input argument.

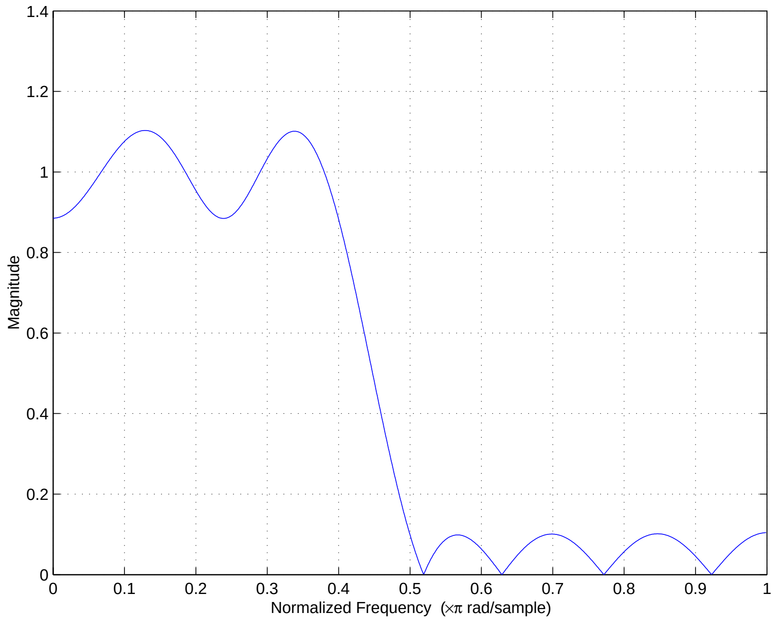
```
[b,err,res]=firgr(12,[0 0.4 0.5 1], [1 1 0 0],[1 0.1],...
{'w' 'c'},{64},'minphase');
```

`firgr` generates a lowpass filter with constrained stopband magnitude equal to 0.1, and the filter is minimum phase as well. We could have specified a maximum phase design by replacing the '`minphase`' option with '`maxphase`'. In the `firgr` statement, you might have noticed the cell array `{64}` entry. The cell array entries define the grid density for points across the frequency spectrum.

Now, plot the filter to view the frequency response.

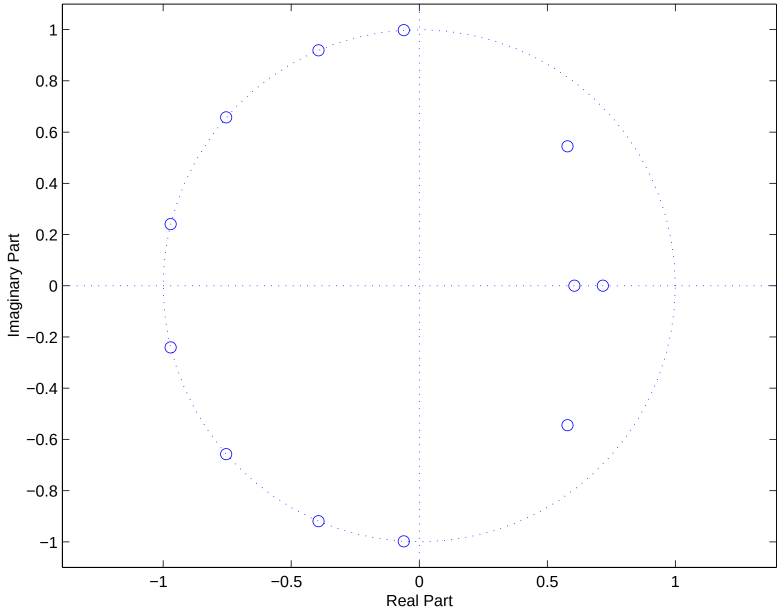
```
[H,W]=freqz(b,1,1024);
S.plot = 'mag'; S.yunits = 'linear';
freqzplot(H,W,S);
```

We have a lowpass filter with stopband ripple not exceeding 0.1, as desired.



In the next figure, viewing our filter roots on the z-plane plot shows us that the roots lie in or on the unit circle. The zeros of a minimum phase delay FIR filter lie on or inside the unit circle. Maximum phase delay filters have zeros that lie on or outside the unit circle.

```
[b,err,res]=firgr(12,[0 0.4 0.5 1], [1 1 0 0],[1 0.1],...
{'w' 'c'},{64},'minphase');
[H,W]=freqz(b,1,1024);
zplane(roots(b));
```



Notice that the filter, with eight zeros on the unit circle, could be very sensitive to quantization. You could use FDATool to investigate the effects of quantizing this filter, and to convert the filter to second order sections or make other changes the reduce the sensitivity to quantization.

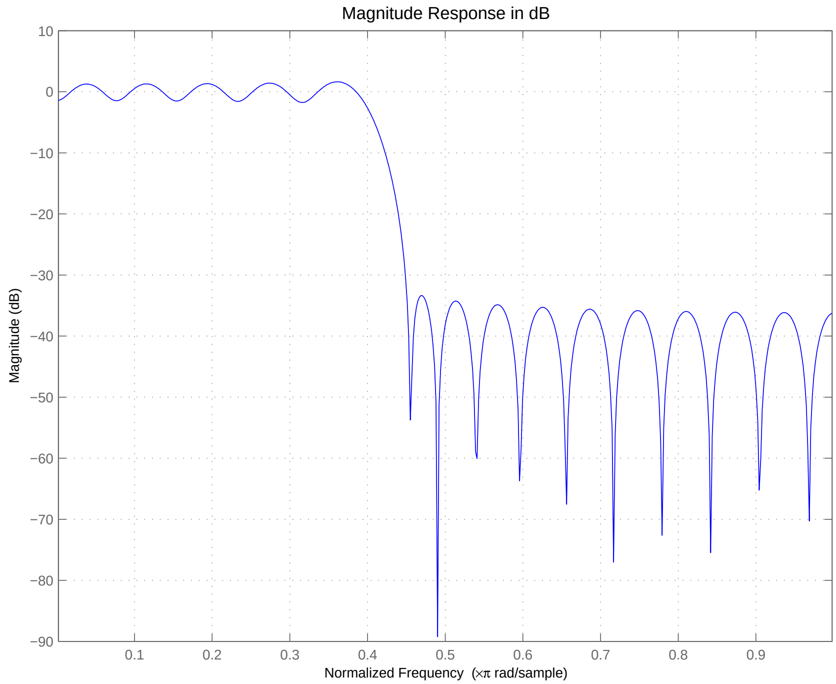
firlpnorm Examples

Review the following examples for an overview of the capabilities of the function—each example uses one or more features provided by `firlpnorm` and the least-Pth unconstrained optimization algorithm. Among the filter designs you can create are filters with arbitrarily defined magnitude response or minimum phase.

Example—Design a Lowpass Filter With pmin = 4 and pmax = 12

With the filter specifications in this example, the result is a quasi-equiripple response lowpass filter. You can see from the plot that follows the code the shape of the magnitude response.

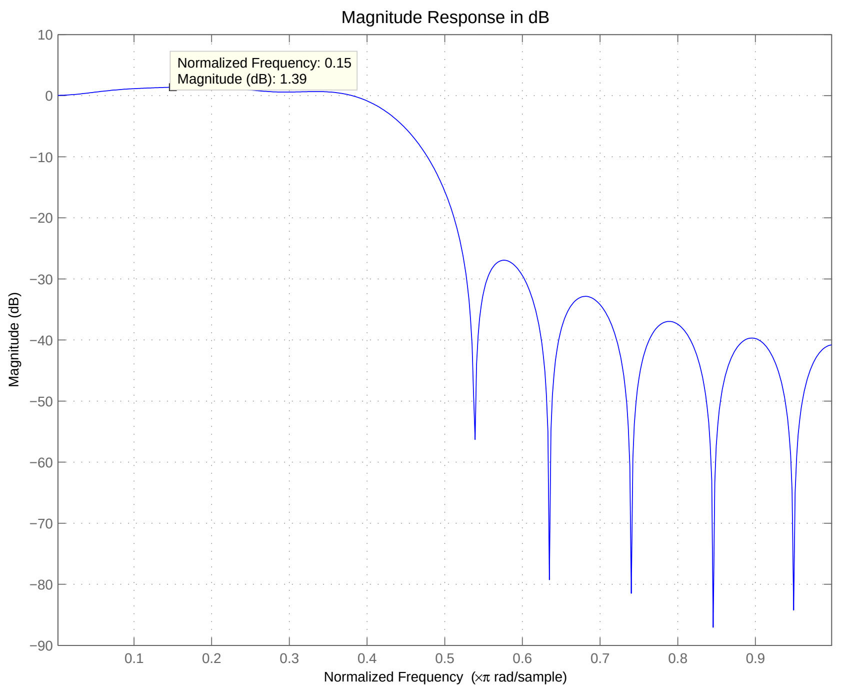
```
b=firlpnorm(30,[0 0.4 0.45 1],[0 0.4 0.45 1],[1 1 0 0],...
[1 1 10 10],[4 12]);
[H,W,S]=freqz(b,1,1024);
S.plot = 'mag';
fvtool(b);
```



Example—Design a Lowpass Least-Squares Filter With a "Peak" In The Passband

Using the appropriate set of input arguments, you can add a slight peak in the passband of the filter. The following code creates a lowpass filter that demonstrates just such tweaking of its passband to add gain. Notice the set of inputs for `a` (the specification of the passband response) `[1 1.2 1 0 0]` in the calling syntax. The `1.2` raises the passband response at the `0.15` normalized frequency point defined in `f`.

```
b=firlpnorm(15, [0 0.15 0.4 0.5 1], [0 0.4 0.5 1],...
[1 1.2 1 0 0],[2 2 2 1 1], [2 2]);
fvtool(b)
```

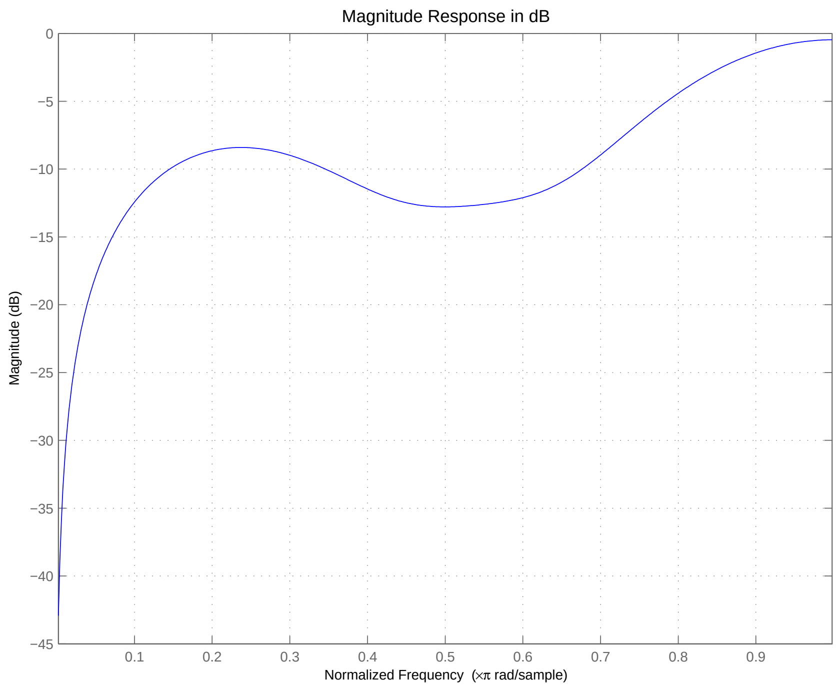


Example—Create a Low-order Filter With One Band

By using the weighting input arguments and the pmin and pmax options, this example creates a low order, n = 5, FIR filter with one band. When you define pmin and pmax as 2 and 16, the optimization algorithm starts at pmin = 2 and continues to optimize in the filter in the pmax sense. By default, pmin and pmax are 2 and 128, achieving the L-infinity or Chebyshev norms.

```
b=firlpnorm(5, [0 .2 .6  1], [0 1], [0 .4 .2  1], [1 1 1 1],...
[2 16]);
fvtool(b)
```

Reviewing the figure from FVTool shows the single band nature of the filter response.

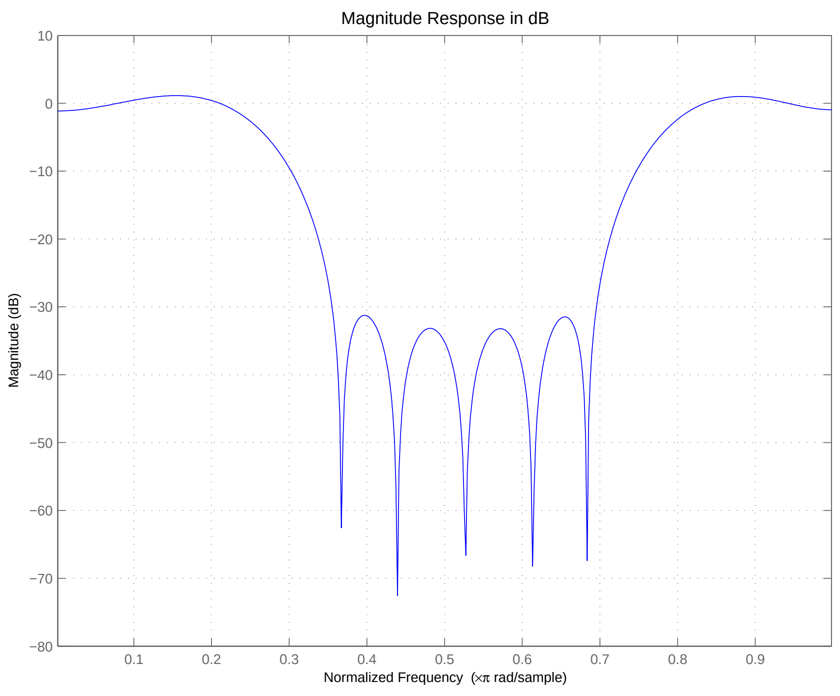


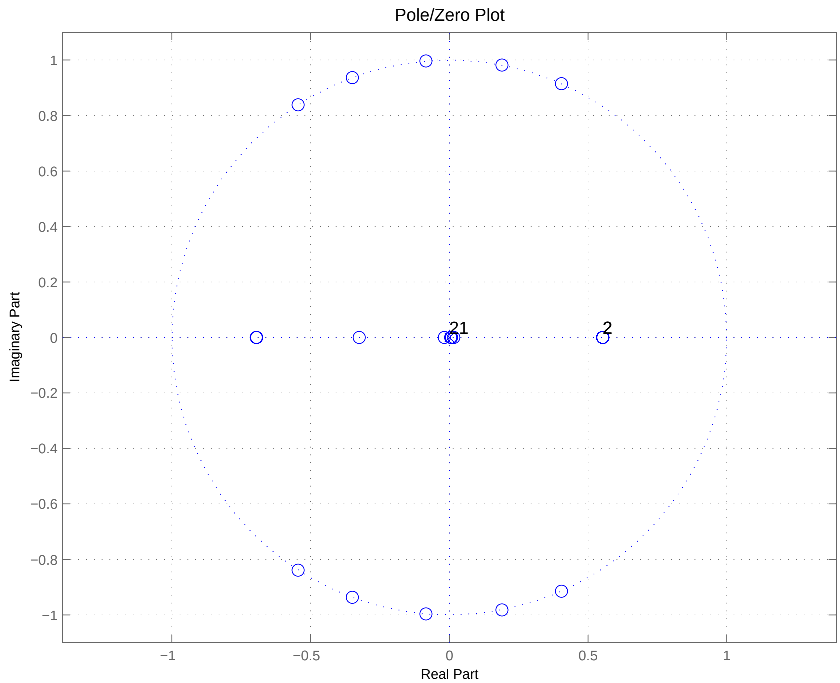
Example—Return a Minimum Phase Bandstop Filter

To generate a minimum phase filter, `firlpnorm` uses the `'minphase'` optional input argument. For this example of creating a bandstop filter, `p = [2 4]` and the filter order is set to `21`. Notice that weight vector `w` emphasizes the error in the stopband region by using `[1 1 5 5 1 1]`. Combined with the a vector of `[1 1 0 0 1 1]`, the result is a bandstop filter, as shown in the figure that follows the code for designing the filter.

```
b=firlpnorm(21, [0 .25 .35 .7 .8 1], [0 .25 .35 .7 .8 1],...
[1 1 0 0 1 1], [1 1 5 5 1 1], [2 4], 'minphase');
fvtool(b)
```

Plotting the zeros on the unit circle shows the minimum phase nature of the filter.




Pole/Zero Plot
Imaginary Part
Real Part
1
0.8
0.6
0.4
0.2
0
−0.2
−0.4
−0.6
−0.8
−1
−1
−0.5
0
0.5
1
21
2

Advanced IIR Filter Designs

Many digital filters use both input values and previous output values from the filter to calculate the current output value. FIR filters can be implemented with feedback, although this is unusual. Cascaded integrated comb filters are one example.

For IIR filters, the transfer function is a ratio of polynomials:

- The numerator of the transfer function. When this expression falls to zero, the value of the transfer function is zero as well. Called a zero of the function.
- The denominator of the transfer function. When this expression goes to zero (division by zero), the value of the transfer function tends to infinity; called a pole of the function or filter.

Filter Design Toolbox introduces three functions: `iirlpnorm`, `iirlpnormc`, and `iirgrpdelay` for designing IIR filters that design optimal solutions to your filter requirements. With these new filter functions, you can design filters to meet your specifications that you could not design using the IIR filter design functions in Signal Processing Toolbox.

Function `iirlpnorm` uses a least-pth norm unconstrained optimization algorithm to design IIR filters that have arbitrary shape magnitude response curves. `iirlpnormc` uses a least-pth norm optimization algorithm as well, only this version is constrained to let you restrict the radius of the poles of the IIR filter.

To let you design allpass IIR filters that meet a prescribed group delay specification, `iirgrpdelay` uses a least-pth constrained optimization algorithm. For basic information about the least-pth algorithms used in the IIR filter design functions, refer to *Digital Filters* by Antoniou [1].

This section uses examples to introduce the IIR filter design functions in the toolbox. As you review these examples, you may notice that the IIR design functions use the same syntax, input, and output arguments. Because the design functions use very similar algorithms, common input and output arguments apply. Arguments are used in the same way, and carry the same defaults and restrictions. That said, if an example of one IIR function uses a syntax that does not appear under another IIR design function, chances are you can use the first syntax in the other design function as well.

Examples — Using Filter Design Toolbox Functions to Design IIR Filters

Filter Design Toolbox provides new capabilities for IIR filter design. Because of the comprehensive nature of the new IIR design functions, learning by example is the best way to discover what you can do with them. This section presents a series of examples that investigate the filters you can implement through IIR filter design in Filter Design Toolbox. You can view these examples as a demonstration program in MATLAB by opening the MATLAB demos and selecting `Filter Design` from `Toolboxes`. Listed there you see a number of demonstration programs. Select one of the following demos to see the IIR filter design functions being used to design a variety of filters:

- Least P-norm Optimal IIR Filter Design demonstrates IIR filter design function `iirlpnorm`. Examples include:
  - "Example — Using iirlpnorm to Design a Lowpass Filter" on page 2-45
  - "Example — Using iirlpnorm to Design a Low Order Filter" on page 2-46
  - "Example — Using iirlpnorm to Design a Bandstop Filter" on page 2-47
  - "Example — Using iirlpnorm to Design a Noise-Shaping Filter" on page 2-49
- Constrained Least P-norm IIR Filter Design demonstrates IIR filter design function `iirlpnormc`. This set of examples includes:
  - "Example — Using iirlpnormc to Design a Lowpass Filter" on page 2-50
  - "Example — Using iirlpnormc to Design a Bandstop Filter with a Constrained Pole Radius" on page 2-52
  - "Example — Using iirlpnormc to Design a High-Order Notch Filter" on page 2-53
  - "Example — Using iirlpnormc to Change an Elliptic Filter to a Constrained Lowpass Filter" on page 2-54
- `IIR Filter Design Given a Prescribed Group Delay` demonstrates IIR filter design function `iirgrpdelay`. These examples include:
  - "Example — Using iirgrpdelay to Design a Filter with a User-Specified Group Delay Contour" on page 2-57
  - "Example — Using iirgrpdelay to Design a Lowpass Elliptic Filter with Equalized Group Delay" on page 2-59

To Open the IIR Filter Design Demos

Follow these steps to open the IIR filter design demos:

1. Start MATLAB.
2. At the MATLAB prompt, enter `demos`.

   The **MATLAB Demo Window** dialog opens.
3. On the list on the left, double-click `Toolboxes` to expand the directory tree.

   You see a list of the toolbox demonstration programs available in MATLAB.
4. Select `Filter Design`.
5. From the list on the right, select one of the following demonstration programs:
   - `Least P-norm Optimal IIR Filter Design`
   - `Constrained Least P-norm IIR Filter Design`
   - `IIR Filter Design Given a Prescribed Group Delay`

A few examples include comparisons to other filter design functions and analysis notes. For details about using the IIR design functions `iirlpnorm`, `iirlpnormc`, and `iirgrpdelay`, refer to the online reference documentation. While this set of examples covers many of the options for the functions, more options exist that do not appear in these examples. Examples cover common or interesting IIR design options to highlight some of the capabilities of the design functions.

In these examples, you can see that `iirlpnorm`, `iirlpnormc`, and `iirgrpdelay` use many of the input arguments used by `firgr`, plus others such as the denominator order. At the most basic level, each IIR filter design function uses the input arguments `N`, `D`, `F`, `Edges`, and `A` — the filter order for the numerator and denominator (so you can specify different order numerators and denominators), the vector containing the filter cutoff frequencies, the band edge frequencies, and the filter response at each frequency point. `F` and `A` must have matching numbers of elements; they can exceed the number of elements in `Edges`. You use this feature to specify a gain contour within a band defined by the entries in `Edges`. Every frequency that appears in `Edges` must also be an element of `F`. Also, the first band edge must equal the first frequency and the last band edge must equal the last frequency in `F`.

iirlpnorm Examples

Each of these examples uses one or more feature provided in the function iirlpnorm. The examples build on one another, although they can be run separately. Review each example to get an overview of the capabilities of the function.

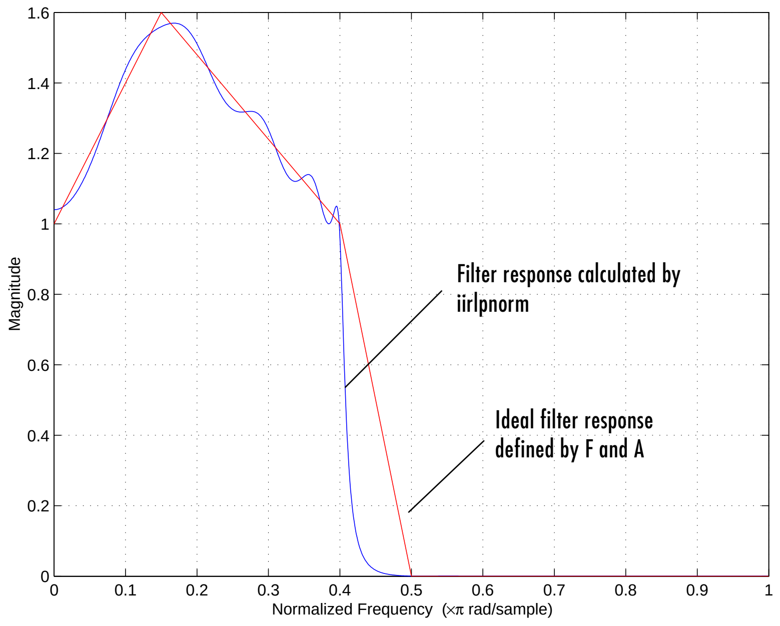
Example — Using iirlpnorm to Design a Lowpass Filter

To design a lowpass filter with maximum gain of 1.6 in the passband, we use the syntax iirlpnorm(n,d,f,edges,a,w). To duplicate the filter in the figure, use this code.

```
[b,a]=iirlpnorm(3, 11, [0 0.15 0.4 0.5 1], [0 0.4 0.5 1],...
[1 1.6 1 0 0], [1 1 1 100 100]);
[h,w,s]=freqz(b,a,1024);
s.plot = 'mag'; s.yunits = 'linear';
freqzplot(h,w,s);
hold on; plot([0 0.15 0.4 0.5 1], [1 1.6 1 0 0], 'r'); hold off;
```

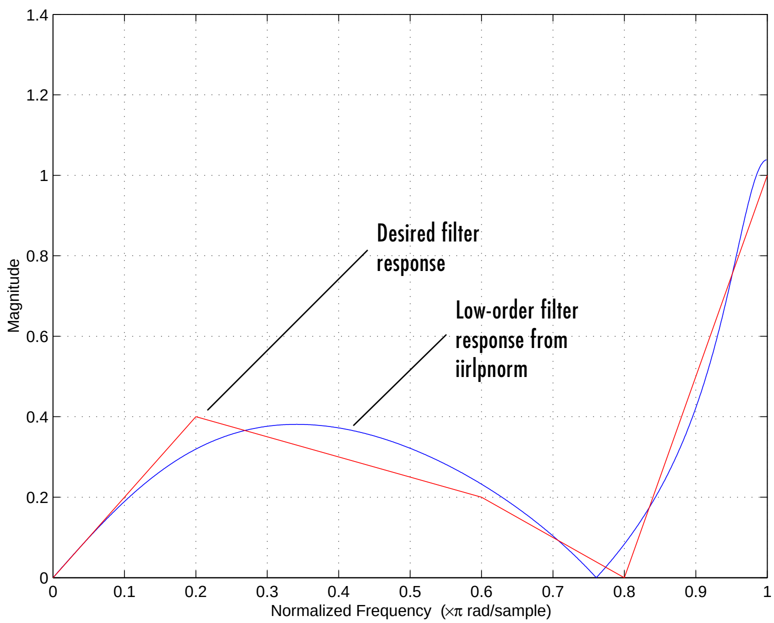
When you look at the magnitude response curve, notice the response reaches 1.6 in the passband.

s



Example — Using iirlpnorm to Design a Low Order Filter

The curves in the next figure show the results of using `iirlpnorm` to design a low-order filter with a single band. For this design, we introduce a new two-element vector `P=[Pmin,Pmax]` that defines the minimum and maximum values of `P` in the least-pth norm algorithm. If you do not specify `P`, the default values are `[2 128]`, resulting in the L_∞ or Chebyshev norm. Specify `Pmin` and `Pmax` to be even numbers. To view the placement of the poles and zeros for your filter before the optimization takes place, replace `[Pmin Pmax]` with the string `'inspect'`. With the option `'inspect'` in use, the algorithm does not optimize the filter design.



We specified a lowpass filter with third-order numerator and denominator, and used the P vector to limit the optimization range, by using the function syntax `iirlpnorm(n,d,f,edges,a,w,p)`.

```
[b,a]=iirlpnorm(3, 3, [0 .2 .6 .8 1], [0 1], [0 .4 .2 0 1],...
[1 1 1 1 1], [2 64]);
[h,w,s]=freqz(b,a,1024);
s.plot = 'mag'; s.yunits = 'linear';
freqzplot(h,w,s);
hold on; plot([0 .2 .6 .8 1], [0 .4 .2 0 1], 'r'); hold off;
```

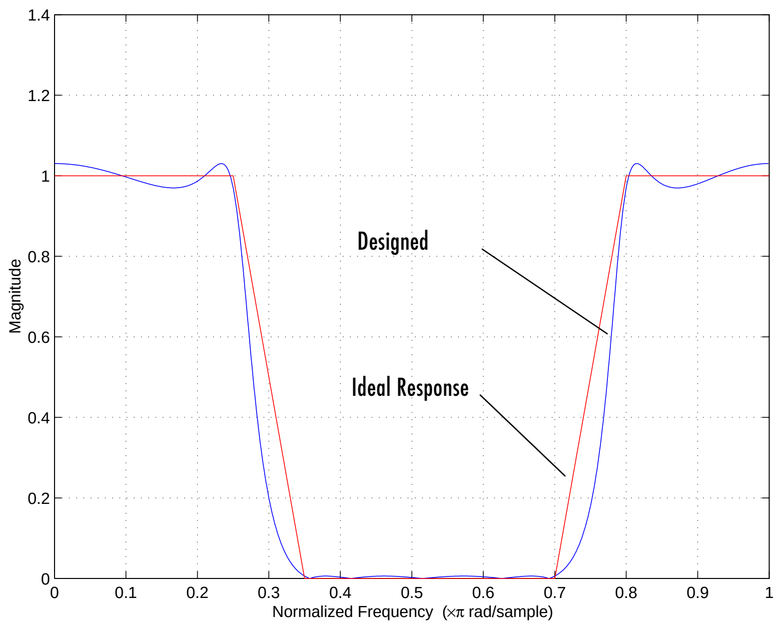
Setting `W=[1 1 1 1 1]` is the same as not setting weight values.

Example — Using iirlpnorm to Design a Bandstop Filter

Designing IIR bandstop filters is straightforward. Enter the frequency, magnitude, edges, and weight vectors using the syntax `iirlpnorm(n,d,f,edges,a,w)` as shown here. To ensure that the stopband rejects undesired frequencies aggressively, we weight the magnitude response in the stopband more heavily by entering the weight vector [1 1 5 5 1 1], telling the optimization algorithm that meeting the inband response specification is five times as important as meeting the out-of-band response.

```
[b,a]=iirlpnorm(10, 7, [0 .25 .35 .7 .8 1],...
[0 .25 .35 .7 .8 1], [1 1 0 0 1 1], [1 1 5 5 1 1]);
[h,w,s]=freqz(b,a,1024);
s.plot = 'mag'; s.yunits = 'linear';
freqzplot(h,w,s);
hold on; plot([0 .25 .35 .7 .8 1], [1 1 0 0 1 1], 'r'); hold off;
```

As you can see from the following figure, the filter meets our design needs quite closely.

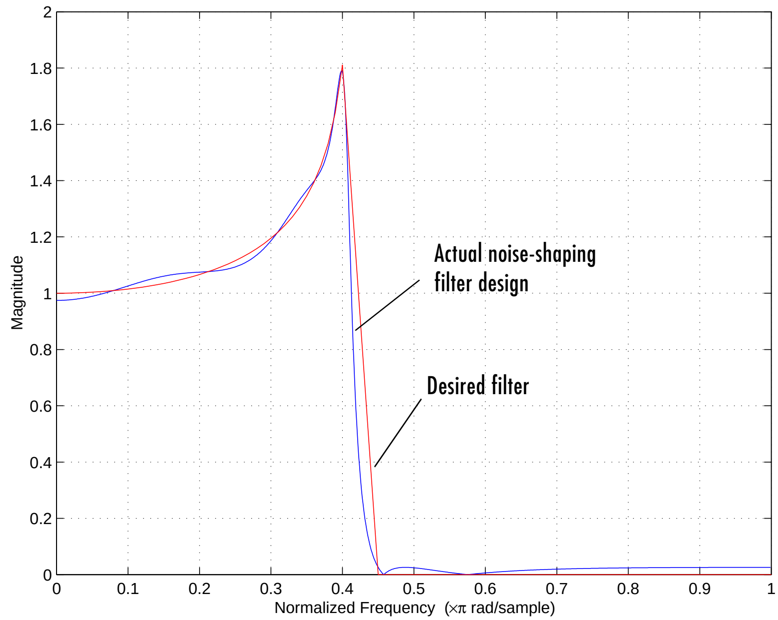


Example — Using iirlpnorm to Design a Noise-Shaping Filter

In this example, we create a lowpass filter with a rising magnitude in the passband. Communications designers use the filter when they simulate the effects of motion between a transmitter and receiver, such as you find in cellular telephone networks. Here, we use `iirlpnorm` to design the same filter. Because of the complex shape of the passband, we define the vectors `f`, `a`, `w`, and `edges` in the workspace, then use the vector names in the `iirlpnorm` statement.

```
f = 0:0.01:0.4;
a = 1.0 ./ (1 - (f./0.42).^2).^0.25;
f = [f 0.45 1];
a = [a 0 0];
edges = [0 0.4 0.45 1];
w = ones(1, length(a));
[b,a]=iirlpnorm(4, 6, f, edges, a, w);
[h,w,s]=freqz(b,a,1024);
s.plot = 'mag'; s.yunits = 'linear';
freqzplot(h,w,s);
hold on; plot(F,A, 'r'); hold off;
```

When you compare the figure below to the filter design in “Getting Started with the Toolbox” on page 1-17, you see they match very well.



iirlpnormc Examples

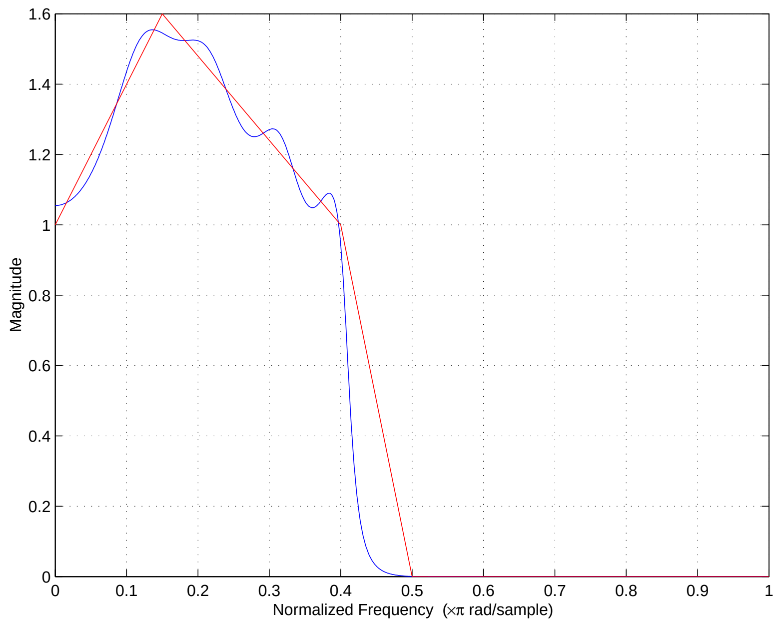
Each of these examples uses one or more feature provided in the function `iirlpnormc`. Review each example to get an overview of the capabilities of the function.

Example — Using iirlpnormc to Design a Lowpass Filter

Just as you use `iirlpnorm` to design lowpass filters, you can use `iirlpnormc` to design them as well. `iirlpnormc` lets you limit the radius of the filter poles when you specify the filter in the function. By restricting the poles to be less than a certain distance from the origin of the unit circle in the z-plane, the filter remains stable, while possibly improving the robustness of the filter to quantization effects. In this lowpass filter example, we restrict the pole radius not to exceed 0.95, using the function syntax `iirlpnormc(n,d,f,edges,a,w,radius)`.

```
[b,a]=iirlpnormc(3, 11, [0 0.15 0.4 0.5 1], [0 0.4 0.5 1],...
[1 1.6 1 0 0], [1 1 1 100 100], 0.95);
[h,w,s]=freqz(b,a,1024);
s.plot = 'mag'; s.yunits = 'linear';
freqzplot(h,w,s);
hold on; plot([0 0.15 0.4 0.5 1], [1 1.6 1 0 0], 'r'); hold off;
```

`radius` takes values between 0 and 1.



Compared to the unconstrained `iirlpnorm` lowpass filter example (refer to "iirlpnorm Examples" on page 2-45), you see that the filter performance is about the same, although the ripple in the passband is slightly greater, and the transition somewhat sharper. The difference between these two designs is the constraint applied to the poles when you use `iirlpnormc` with a radius value. Both filters demonstrate peaks in their passband.

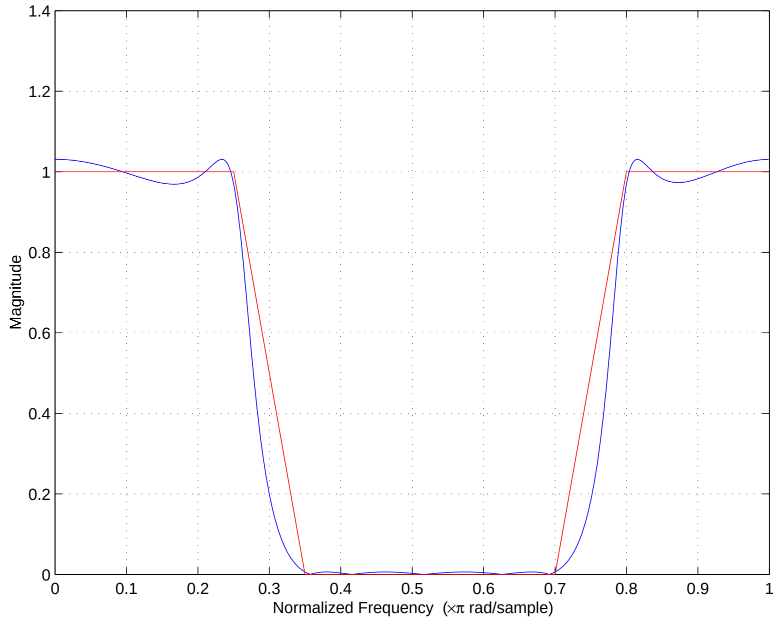
Example — Using iirlpnormc to Design a Bandstop Filter with a Constrained Pole Radius

Here we use `iirlpnormc` to design a bandstop filter. Notice that we specify different orders for the numerator (`n=10`) and denominator (`d=7`) and the frequency and edges vectors are the same. With `radius=.91`, none of the 11 filter poles lies farther than 0.91 away from the origin, as you can see in the zero-pole plot.

```
f = [0 .25 .35 .7 .8 1];
[b,a]=iirlpnormc(10, 7, f, f, [1 1 0 0 1 1], [1 1 5 5 1 1], .91);
[h,w,s]=freqz(b,a,1024);
s.plot = 'mag'; s.yunits = 'linear';
freqzplot(h,w,s);
hold on; plot([0 .25 .35 .7 .8 1], [1 1 0 0 1 1], 'r'); hold off;
```

To generate the zero-pole plot, use `zplane(b,a)` at the MATLAB prompt.

When we plot the magnitude response curve, the emphasis we placed on reducing the error in the stopband is clear — note the close match between the desired and calculated responses. (We weighted the magnitude response `w=[1 1 5 5 1 1]` to minimize the error in the vicinity of the stopband frequency points.)

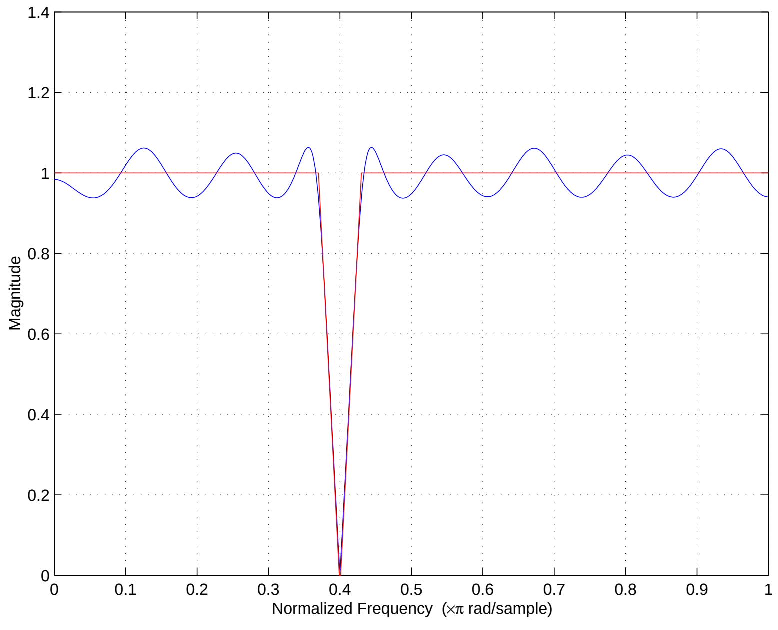


Example — Using iirlpnormc to Design a High-Order Notch Filter

To create an optimized design for an IIR high-order notch filter, use `iirlpnormc` to design the filter. The following code results in the optimal solution to creating a filter with different numerator and denominator orders, and with a maximum pole radius of `0.92`.

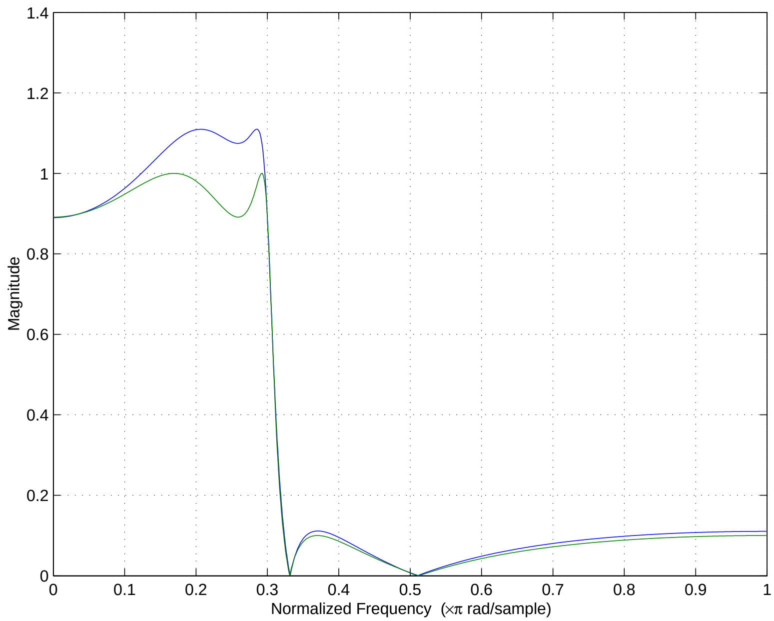
```
f = [0 0.37 0.399 0.401 0.43 1];
[b,a]=iirlpnormc(2, 17, f, f, [1 1 0 0 1 1], [1 1 2 2 1 1], 0.92);
[h,w,s]=freqz(b,a,1024);
s.plot = 'mag'; s.yunits = 'linear';
freqzplot(h,w,s);
hold on;
plot([0 0.37 0.399 0.401 0.43 1], [1 1 0 0 1 1], 'r'); hold off;
```

Note the frequency vector entries `0.37`, `0.399`, `0.401`, and `0.43`. These represent the cutoff points for the filter stopband, a fairly narrow filter. Looking at the filter response plot, you see it is similar to the single-point filter example we designed with the `firgr` function (refer to "Example—Designing a Single-Point Band Filter" on page 2-18). This filter has two pairs of constrained poles.



Example — Using iirlpnormc to Change an Elliptic Filter to a Constrained Lowpass Filter

Using an elliptic filter design as the initial conditions, with a maximum pole radius of 0.96, we reduce the pole radius to 0.95 when we use `iirlpnormc` to create an optimal filter solution. The result is a filter with the same band edge frequencies, and a gain in the passband greater than one. The following code uses the function `ellip` from Signal Processing Toolbox to create an elliptical filter. Then we use the function `iirlpnormc` with the syntax `iirlpnormc(n,d,f,edges,a,w,radius,p, dens,initnum,initden)`. `initnum` and `initden` are the initial estimates of the filter numerator and denominator coefficients. We use `be` and `ae` from our elliptic filter as the vectors `Initnum` and `initden`.




```
[be,ae]=ellip(4,1,20,0.3);
f = [0 0.3 0.323 1];
[b,a]=iirlpnormc(4, 4, f, f, [1 1 0 0], [1 1 1 1], .95,...
[128 128], 20, be, ae);
[h,w,s]=freqz(b,a,1024);
he=freqz(be,ae,1024);
s.plot = 'mag'; s.yunits = 'linear';
freqzplot([h he],w,s);
```

A few points to think about when you use `iirlpnormc`. These hints can help you converge on a good filter design:

- `iirlpnormc` implements a weighted, least-pth optimization algorithm.
- Check the location and radii of the designed filter poles and zeros.
- If the zeros are on the unit circle and the poles are well inside the circle, try increasing the numerator order `N`, or reducing the error weighting (`W`) in the stopband.

- If several poles have large radii, and the zeros are well inside the unit circle, try increasing D, the denominator order, or reducing the error weighting in the passband.
- As you reduce the pole radius, you may need to increase the denominator order.

iirgrpdelay Examples

Filter Design Toolbox provides a new filter design function iirgrpdelay for designing allpass IIR filters that have group delay characteristics that meet your needs. When you cascade these allpass filters with other IIR filters, they act as compensating elements. They produce equalized or specified group delay across the combined filter frequency response while maintaining the IIR filter pass and stop bands. For more information about group delay in filters, refer to “Signal Processing Basics” in *Signal Processing Toolbox User’s Guide*.

Note iirgrpdelay creates allpass filters you use to compensate for the phase changes caused by other filters. You cannot use iirgrpdelay to create filters that both filter input signals and compensate for phase changes in output signals.

In this section, we introduce the function iirgrpdelay through a series of examples. Each of these examples uses one or more feature provided in the function. The examples build on one another, although they can be run separately. By reviewing each example you get an overview of the capabilities of the design function.

In much the same way that you use other IIR filter design functions to create filters with arbitrary magnitude response curves, you use iirgrpdelay to create filters that have arbitrary group delay curves in the filter passband and stopband. (In most cases, specifying the group delay in the stopband is not useful; the filter rejects those frequencies by design. Nonetheless, you can specify the group delay for frequencies that fall within filter stopbands.)

To specify a filter that approximates a given relative group delay, use iirgrpdelay with the following input argument syntax

```
iirgrpdelay(N,F,Edges,Gd)
```

where N is the filter order, F is a vector containing frequencies between 0 and 1, Gd is a vector whose elements are the desired group delay at the frequencies specified in F, and Edges specifies the band edges. Filter order N must be an even number, and the vectors F and Gd must have the same number of elements. To let you specify the shape of the group delay within a band or bands, vectors F and Gd can contain more elements than Edges.

Considering the following ideas can help you design your group delay compensator:

- After you use iirgrpdelay to design a filter, use freqz, grpdelay, and zplane to check your design for undesirable features.
- Remember that allpass filters have positive group delay. You cannot develop allpass filters that have negative group delay characteristics.
- For some difficult filter optimization problems, use the iirgrpdelay syntax

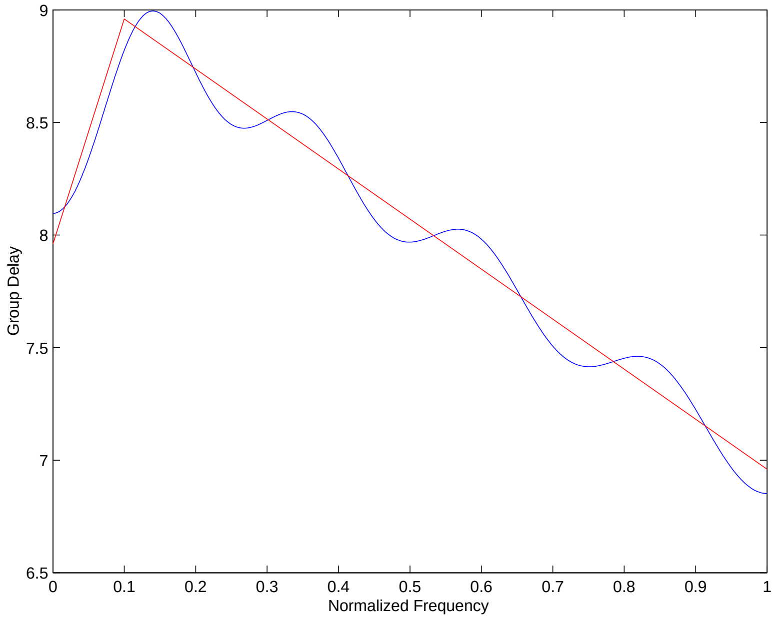
  ```
  iirgrpdelay(n,d,edgees,a,w,radius,p,dens,initden)
  ```

  where initden is a vector containing your estimates of the transfer function coefficients for the denominator. You can use the Pole-Zero editor in Signal Processing Toolbox to generate values for initden.
- If the poles and zeros of your filter design cluster together, you may need to increase the filter order or relax the pole radius restriction (if you used one).

Example — Using iirgrpdelay to Design a Filter with a User-Specified Group Delay Contour

To show the ability to create an arbitrary shape group delay contour in the passband of an IIR filter, we use iirgrpdelay and specify the group delay we desire. Notice that we also specify the maximum pole radius of 0.99. We plot the ideal group delay contour on the figure as well to compare the desired result to the designed filter.

```
[b,a,tau] = iirgrpdelay(8, [0 0.1 1], [0 1], [2 3 1],...
[1 1 1], 0.99);
[G,F] = grpdelay(b,a, 0:0.001:1, 2);
plot(F, G); hold on; plot([0 0.1 1], [2 3 1]+tau, 'r'); hold off;
```



The straight lines represent the desired group delay contour, the wavy line the designed contour. The desired group delay, `[2 3 1]`, is relative. Note that the actual group delay approximates `[8 9 7]`. If we increase the filter order, to 10 for example, the approximation improves, but the absolute group delay increases.

One of the output arguments for `iirgrpdelay` is `tau`, the resulting group delay offset. In all cases, filters created by `iirgrpdelay` have a group delay that approximates (`gd` + `tau`) where `gd` is the specified relative group delay of the filter.

When you look at the zero-pole plot for our filter (use the function `zplane`), you can see that the poles stay well within the radius constraint. Optimizing the filter may not result in poles that are near the constraint. Pole constraints come into play only when needed to limit the optimization. In this example, our design did not require the constraint to stay within the bounds of the unit circle.

You can verify that this is an allpass filter by plotting the magnitude response curve for the design. Use `freqz(b,a)` to plot the curve.

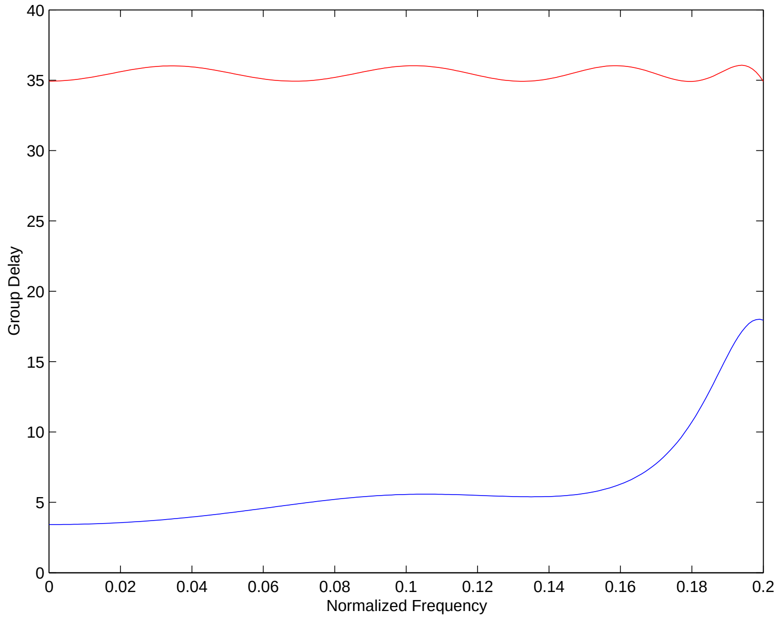
In general, you determine the contour to use for the group delay equalization of an IIR filter by subtracting the filter group delay from the filter maximum group delay. In the next example, we use this process to create our lowpass filter.

Example — Using iirgrpdelay to Design a Lowpass Elliptic Filter with Equalized Group Delay

The following code designs a pair of filters that together create a lowpass filter with equalized group delay.

```
[be,ae] = ellip(4,1,40,0.2); % Lowpass filter
f = 0:0.001:0.2;
g = grpdelay(be,ae,f,2);
g1 = max(g)-g;
[b,a,tau] = iirgrpdelay(8, f, [0 0.2], g1); % Phase compensator
gd = grpdelay(b,a,f,2);
plot(f, g); hold on; plot(f, g+gd, 'r'); hold off;
```

Cascading the filters is the same as adding the group delay for each filter frequency-point by frequency-point (`g+gd` in the `plot` function input arguments). In the figure, the lower curve is the group delay for the elliptic filter. The compensated, or equalized, group delay is the upper curve — an essentially flat group delay across the passband from 0 to 0.2. Since this example used the lowpass elliptic filter from our earlier `iirlpnorm` examples, you can see that combining these filters results in a lowpass filter with equalized group delay. Note that the group delay of the combination is twice the maximum group delay of the reference filter. When you use an allpass filter to equalize the group delay of a reference filter, the final group delay is the sum of the group delays of the reference and allpass filters.

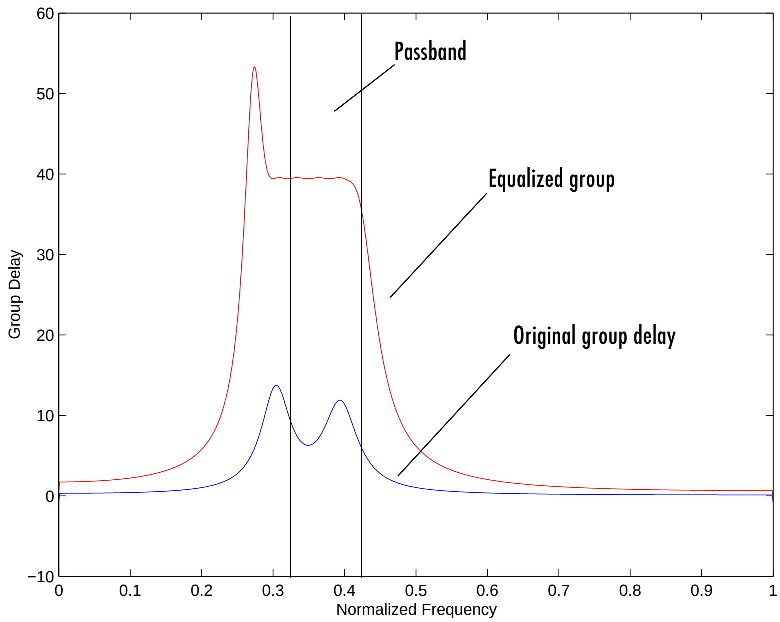


To determine the group delay contour necessary to compensate for the phase effects of our elliptic filter, we use the elliptic filter group delay as a reference.

In the example, we used `grpdelay` to return vector `g` containing the group delay value at many frequencies across the elliptic filter passband. After determining the maximum group delay in the elliptic filter passband (returned by `max(g)` in the example code), we subtract each individual group delay from the maximum group delay `(g1=max(g)-g)`. The result is vector `g1` containing values that define a curve that is the mirror image of the group delay contour of our elliptic filter. Then we use `g1` as the input group delay values to `iirgrpdelay`, and the resulting allpass filter has a group delay contour that equalizes the group delay of our lowpass elliptic filter, as shown in the figure.

Example — Demonstrating Passband Equalization for a Bandpass Chebyshev Filter

You can use `iirgrpdelay` to create filters that compensate for the group delay of many kinds of filters. In this example, we create an allpass filter that equalizes the group delay of a bandpass filter. In the figure, the lower curve is the group delay of the bandpass filter and the upper curve is the equalized group delay for the combination of the bandpass filter and the allpass filter. Group delay variation across the passband is less than 0.2.



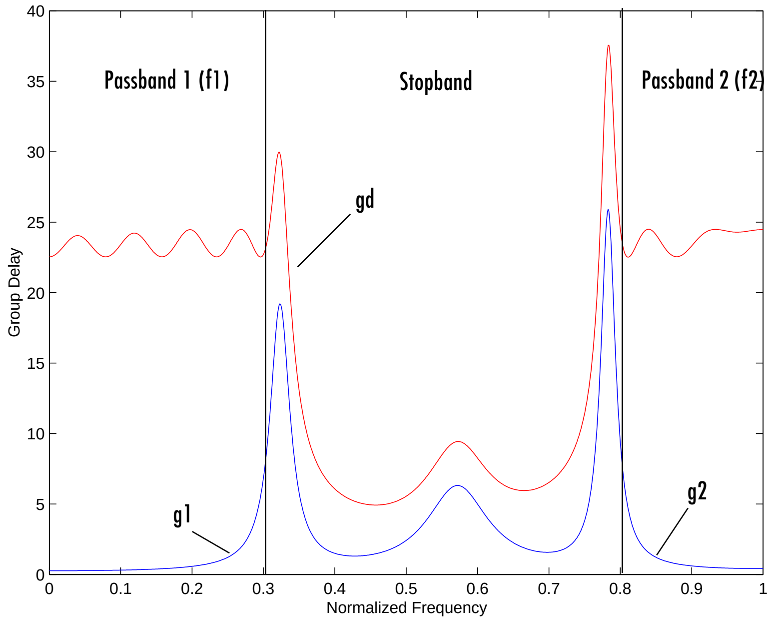

```
[bc,ac] = cheby1(2,1,[0.3 0.4]); % Bandpass filter design
f = 0.3:0.001:0.4;
g = grpdelay(bc,ac,f,2);
g1 = max(g)-g;
wt = ones(1, length(f));
[b,a,tau] = iirgrpdelay(8, f, [0.3 0.4], g1, wt, 0.95);
f = 0:0.001:1;
g = grpdelay(bc,ac,f,2);
gd = grpdelay(b,a,f,2);
plot(f, g); hold on; plot(f, g+gd, 'r'); hold off;
```

Example — Demonstrating Passband Equalization for a Bandstop Chebyshev Filter

Our final example shows how to equalize the group delay in the passband of a bandstop filter. Since this filter has two passbands, we equalize the group delay in each band according to the needs of each band. Vectors g1 and g2 in the example code contain the group delays within each passband of the bandpass filter. We ignore the stopband group delay for this case. To determine the group delay contour across both passbands, we concatenate g1 and g2 (using the command g = [g1; g2]), then use the vector g as the basis for the group delay input argument gx to iirgrpdelay.

```
[bc,ac] = cheby2(3,1,[0.3 0.8], 'stop'); % Bandstop filter
f1 = 0.0:0.001:0.3;
g1 = grpdelay(bc,ac,f1,2);
f2 = 0.8:0.001:1;
g2 = grpdelay(bc,ac,f2,2);
f = [f1 f2]; g = [g1; g2]; % Concatenate the passband group delays
gx = max(g)-g;
wt = ones(1, length(f));
[b,a,tau] = iirgrpdelay(14, f, [0 0.3 0.8 1], gx, wt, 0.95);
f = 0:0.001:1;
g = grpdelay(bc,ac,f,2);
gd = grpdelay(b,a,f,2);
plot(f, g); hold on; plot(f, g+gd, 'r'); hold off;
```

The figure shows that our approach works. You see that the group delay in the passbands is well-equalized (illustrated by the upper curve; the lower curve presents the nonequalized group delay). The stop band is unaffected, and the overall equalized group delay variation in the passbands is close to a constant.


Passband 1 (f1)
Stopband
Passband 2 (f2)
gd
g1
g2
Group Delay
Normalized Frequency
0
5
10
15
20
25
30
35
40
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1

Robust Filter Architectures

We have been considering FIR and IIR filters whose transfer function is represented by constant coefficients and where the input signals and coefficients can be any double-precision value from $-\infty$ to $+\infty$. These systems are in the discrete time domain, with infinite precision values for the dependent variable, often magnitude.

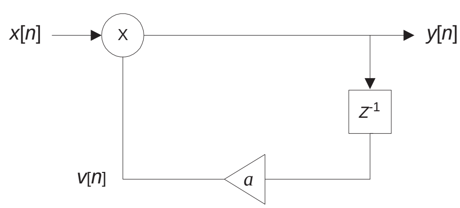
When you represent filters in software, or in general purpose or special purpose computing hardware, the inputs to the filters and the filter coefficients can be represented only by discrete values. The process of converting the infinite precision variables to discrete values is called *quantization* and represents a source of error when you implement digital filters.

Converting to the discrete domain produces three sources of errors:

- Error caused by the discrete representation of infinitely precise information, such as filter transfer function coefficients or signal amplitude values. Real systems create error when they quantize amplitude values.
- Analog-to-digital conversion error in the input signal.
- Arithmetic round off errors caused by the limited word length available to represent the data in the arithmetic process.

Transfer Function Coefficient Quantization Error

To illustrate the effects of converting from continuous to discrete representations, and to show error sources resulting from quantization, consider the following first-order IIR filter.



The constant coefficient difference equation that defines this filter is

$$y[n] = \alpha y[n-1] + x[n]$$

where $y[n]$ and $x[n]$ are the output and input signal variables. In transfer function form, the following equation describes our IIR filter.

$$H(z) = \frac{1}{1-\alpha z^{-1}} = \frac{z}{z-\alpha}$$

When you implement this filter form in hardware, the filter coefficient α assumes discrete values that approximate the design value. Therefore, the actual transfer function that you implement is

$$\hat{H}(z) = \frac{z}{z-\hat{\alpha}}$$

where $\hat{H}$ and $\hat{\alpha}$ are the close approximations to the original H and α in the filter design. Notice that this transfer function differs from the theoretical function $H(z)$. As a result, the actual filter response can differ substantially from the ideal response.

The main effect of transfer function coefficient quantization is to move the poles and zeros to different locations in the z-plane, away from their desired, or designed locations (the locations for the ideal, nonquantized coefficient filter). Moving the poles or zeros can have two effects:

- Changing the frequency response of the quantized filter so it is not the same as the ideal or designed filter.
- Moving poles from inside to outside the unit circle, causing the quantized IIR filter to be unstable. Applies only to IIR filters.

Input Sampling Error (A/D Error)

Given the difference equation for our IIR filter, from earlier

$$y[n] = \alpha y[n-1] + x[n]$$

where $x[n]$ is the sampled output from an analog to digital converter. Sampling the continuous signal $x_a(t)$ results in $x[n]$. Then the sampled input to the filter from the A/D convertor, $\hat{x}[n]$, is

$$\hat{x}[n] = x[n] + e[n]$$

and $e[n]$ is the error in the A/D conversion process. Our discrete input to the filter no longer matches the continuous signal $x_a(t)$. Discrete-time input $x_a(t)$ does not match $x[n]$ because analog-to-digital conversion made the input discrete in time. Similarly, quantized input $\hat{x}[n]$ does not match $x[n]$ because it has been convert to discrete data in amplitude.

Arithmetic Quantization Error

Quantization in arithmetic operations causes another error. For our first-order filter example, the output from our multiplier $v[n]$ is generated by multiplying the signal, $y[n\text{-}1]$ with the transfer function coefficients, α

$$v[n] = \alpha y[n-1]$$

and storing the result. When we quantize the result to fit it into a storage register, we generate a quantized value $\hat{v}[n]$ that we write as

$$v[n] = v[n] + e_{\alpha}[n]$$

where $e_{\alpha}[n]$ is the error sequence resulting from the product quantization process.

Limitcycles and Arithmetic Quantization

There is another source of errors in digital filter implementation, caused by the nonlinearity of quantized arithmetic operations. These errors are apparent in an effect called limit cycling that occurs at the filter output. Limit cycles usually appear when there is no input to the filter, or the input to the filter is constant or sinusoidal.

Low Sensitivity Filter Architectures

Quantizing filter coefficients can have serious effects on the performance of digital filters. As a result of coefficient quantization, the frequency response of the filter with quantized coefficients can be significantly different from the desired filter without quantized coefficients. In some cases, the performance of the quantized filter can make it unsuitable for your application.

Low sensitivity filter architectures, or robust architectures as they are sometimes called, are interesting because they can reduce the effects of coefficient quantization. By being inherently less sensitive to coefficient quantization, these filter architectures withstand the quantization process and result in filters that retain the performance of the original filter.

Approaches to Designing Low Sensitivity Filters

Consider either of two approaches to designing low sensitivity filters:

- Convert low sensitivity analog filters composed of inductors, capacitors, and resistors to digital architectures by replacing the analog components and connections with their digital equivalents so the digital filter approximates the analog version.
- Develop digital filter implementations that respond directly to the conditions that create low coefficient sensitivity in a digital filter designs.

Filter Design Toolbox uses the latter approach to provide low sensitivity filter architectures.

Generally, filter architecture sensitivity ranges from high for direct forms to very low for coupled allpass forms. For reference, the following list ranks the filter forms in the toolbox by their sensitivity to coefficient quantization, from high sensitivity to low:

1. Direct forms—often very sensitive to quantization
2. Lattice forms—moderately sensitive to quantization
3. Allpass forms—quite robust under quantization

Quantization sensitivity is also a function of the locations of the poles and zeros for a filter, so use this list for guidance only.

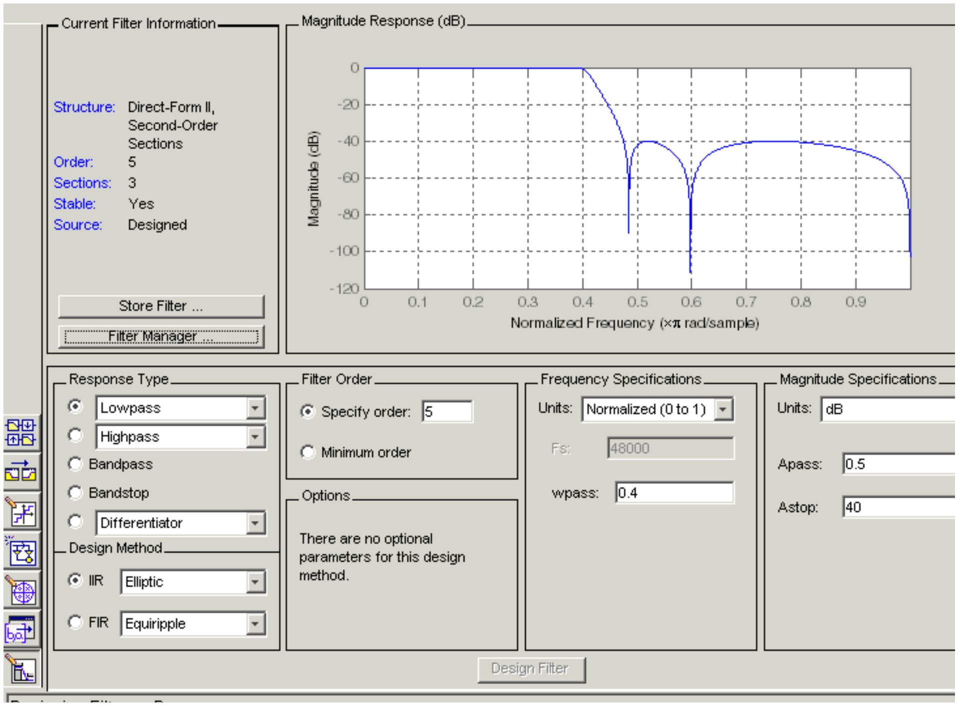
Within the forms

- FIR filters tend to be less sensitive than IIR filters
- For the direct forms, second-order section implementations are often less sensitive to coefficient quantization

Filter Design Example That Includes Quantization

To demonstrate the effects of coefficient quantization on the performance of a filter, this example creates a 5th-order, lowpass elliptic IIR filter. We choose a cutoff frequency of 0.4π radians (normalized frequency from 0 to 1), passband ripple less than 0.5 dB, and stopband attenuation of at least 40 dB. In the figure you see the filter response.

We used the Filter Design and Analysis tool (FDATool) to design the filter. Notice that we used the default filter structure `df2tsos`, or Direct form 2 transposed using second-order sections. When we want to compare the quantized version of the filter to the floating-point filter, FDATool lets us quantize the filter and display the filter response curves together.



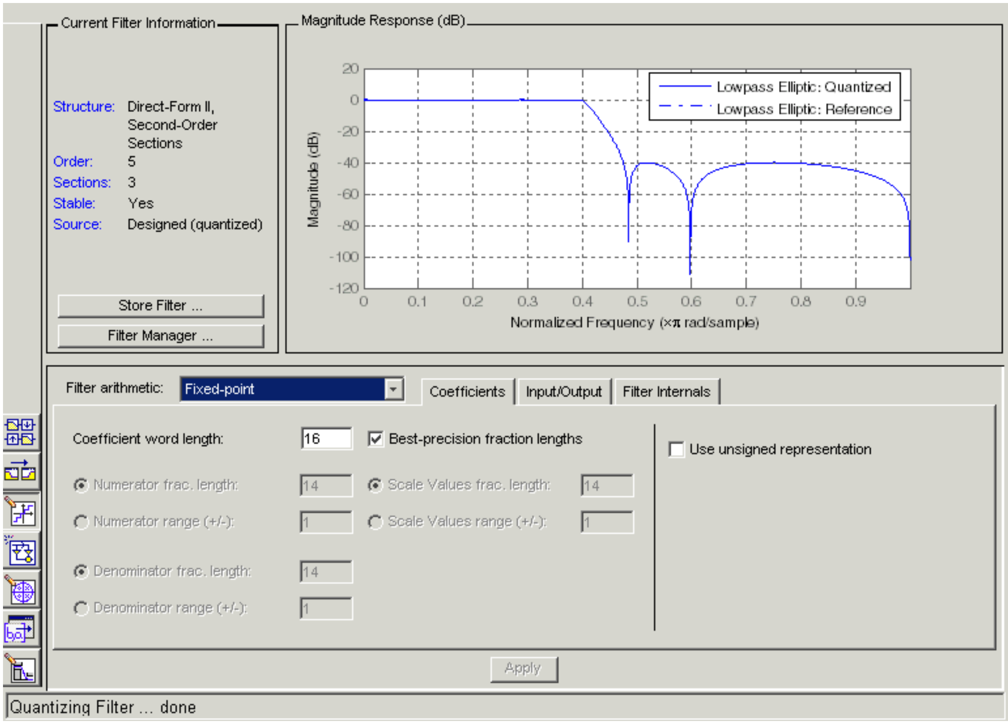
We could have used the function `ellip` from Signal Processing Toolbox to create the filter, and then converted the filter to second-order section form to match the FDATool design.

```
[b,a] = ellip(5,0.5,40,0.4);
```

The results are identical because FDATool uses the same function to design the lowpass filter.

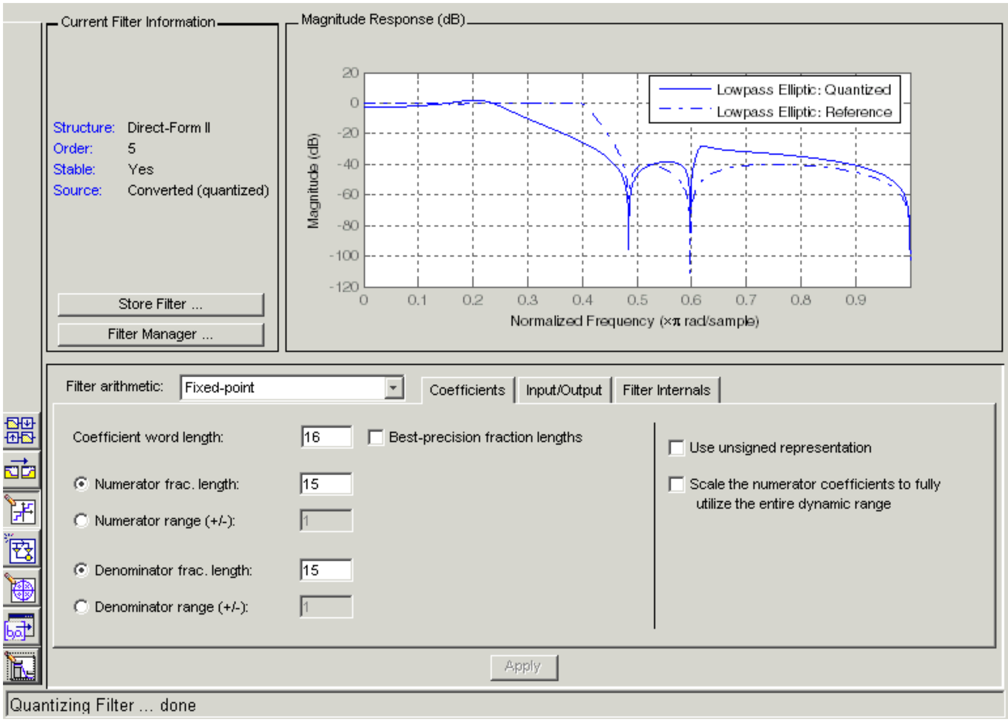
We quantize the filter by selecting **Set quantization parameters**, and then selecting fixed-point for the filter arithmetic. FDATool quantizes our elliptic filter and displays the magnitude response for both the original (or reference) filter and the quantized filter.

For this quantization process we use the default coefficient format settings in FDATool. Later in this example we change the coefficient format to illustrate the effects of changing the word length used to represent the filter coefficients.

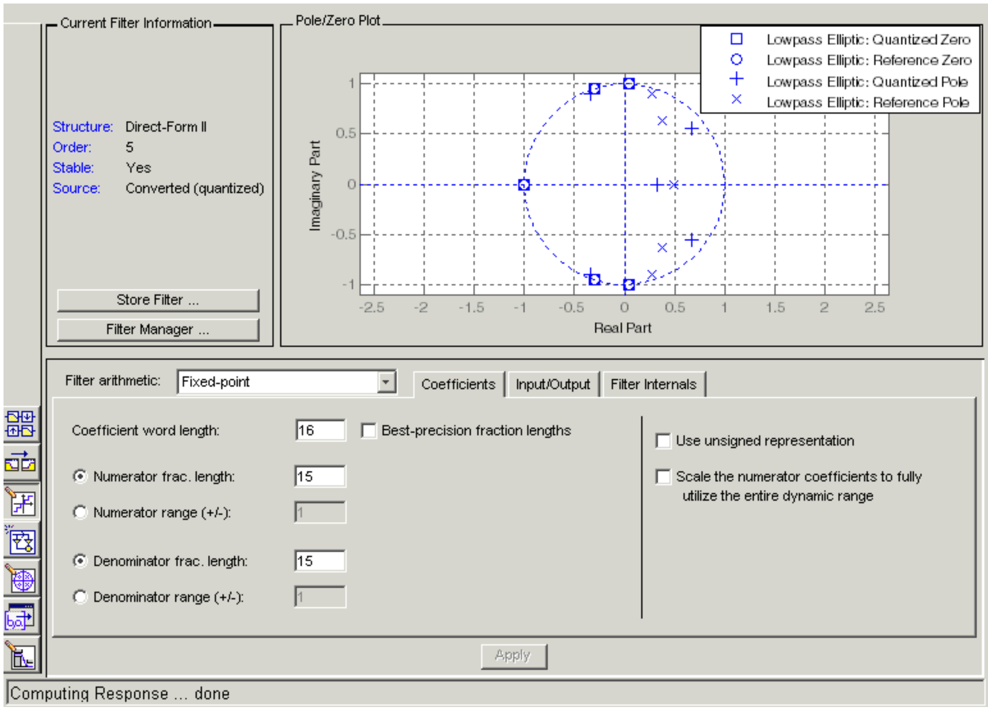


Quantizing the coefficients has not damaged our filter magnitude response, primarily because FDATool implements the filter in second-order section form—resistant to the effects of quantization.

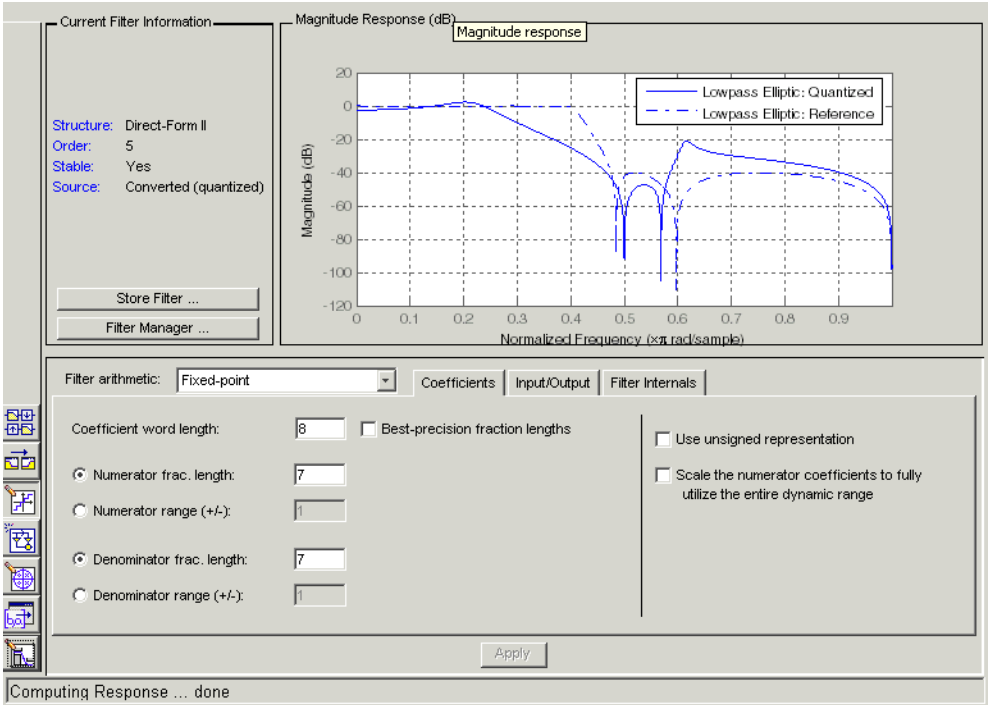
If we convert to single-section representation for demonstration purposes (adjusting the coefficient formats as needed), the single-section quantized filter filter transition band starts much earlier and is much shallower, and the stopband attenuation has been reduced—the filter performance is clearly affected. Here is what you see in FDATool after you convert to a single-section representation for the filter and use 16 bits for the coefficient word lengths and 15 bits for the numerator and denominator fraction lengths.



When we look at the zero-pole plot for the unquantized and quantized versions of our filter, we see that quantization has moved the poles from their designed locations. Coefficient overflow, rather than sensitivity to quantization, caused the terrible quantized response in this filter. Coefficient quantization changes filter coefficients by at most one quantization level. Overflow can change the coefficients by an arbitrarily large amount. In this case, quantization changed the largest magnitude coefficient from 2.49 to saturation at 0.999. You see this from the coefficient view by selecting **Analysis -> Filter Coefficients**. Thus we see how sensitive this direct form IIR filter is to coefficient quantization.



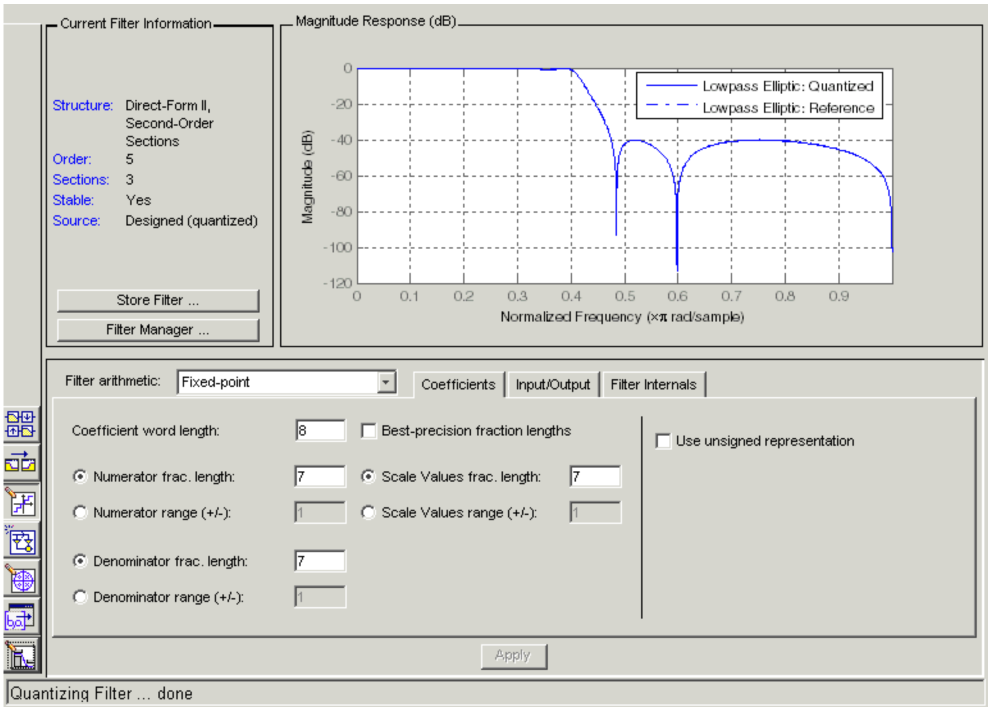
To continue this example, we look at the effects of changing the coefficient format from fixed-point, 16-bits to fixed-point, 8-bits. After we make the desired change, we see the response curves shown in this figure.



When you inspect the entries in the **Set Quantization Parameters** dialog, you see that we changed the coefficient format to [8 7], meaning we are using eight-bit wordlength and seven-bit fraction length to represent each filter coefficient. Changing the coefficient format to 8-bit, fixed point representation causes the effects shown in the figure — the passband rolls off early, the transition is less sharp, and the cutoff frequency lies beyond our 0.4 specification.

In FDATool, select **Analysis->Pole/Zero Plot** to view the poles and zeros for the 8-bit filter plotted on the unit circle. Or you might select **Analysis->Filter Coefficients** to see the coefficient numerical values for the filter.

One more experiment in this example. We try changing the Direct form II transposed (`df2t`) filter structure back to using second-order sections (where we started), which tend to be resistant to quantization effects. As we see in the figure, the elliptic filter that uses second-order sections, even with the 8-bit coefficient format, performs identically to our reference filter. One more change—we set the scale values fraction length to 7 bits. This restores the gain for the quantized filter to match our original. As you see here, the filter magnitude responses appear the same.



We changed the scale values by resetting the Scale Value frac. length option to 7 as shown. Without making that change, the quantized filter magnitude response has the same shape, but insufficient gain to match the original.

Selected Bibliography

[1] Antoniou, A., *Digital Filters: Analysis, Design, and Applications*, Second Edition, McGraw-Hill, Inc., 1993, 330–360.

[2] Mitra, S. K., *Digital Signal Processing: A Computer-Based Approach*, McGraw-Hill, Inc., 1998, 573–584.

3

Working with Fixed-Point Filters

In the Filter Design Toolbox you can implement and analyze single-input single-output filters either as fixed-point filters, or as single-precision or double-precision floating-point filters. Both the single-precision floating-point and fixed-point filters are referred to as *quantized filters*.

You can create a quantized filter from a reference filter, that is, a filter whose coefficients and arithmetic operations you want to quantize in some fashion.

When you apply a quantized filter to data, not only are the filter coefficients quantized to your specification, but so are:

- The data that you filter, both input and output
- The results of any arithmetic operations that occur during filtering

Refer to “Bibliography” for a list of relevant references on quantized filtering.

This chapter covers what you need to know to construct and use quantized filters:

- Getting Started with Fixed-Point Filters
- Constructing quantized and fixed-point filters
- Fixed-point filter properties
- Filtering data with fixed-point filters
- Transformation functions for fixed-point filter coefficients
- Working with Fixed-Point Direct-Form FIR Filters

The filters you create in this toolbox are objects with properties. You can find most of the basic information you need to know about setting and retrieving property values in your MATLAB documentation and by reading about the `set` and `get` functions.

Getting Started with Fixed-Point Filters

As filter designers begin to use digital filters in applications where power limitations and size constraints drive the filter design, they move from double-precision, floating-point filters to fixed-point filters. This tutorial shows you how to analyze the quantization effects introduced by such a conversion using discrete-time filter objects (`dfilt` objects).

Through the course of this exercise, we cover the following processes:

- "Designing Double Precision Floating-Point Filter Coefficients" on page 3-3
- "Converting the Filter to Fixed-Point" on page 3-3
- "Quantizing Filter Coefficients with Automatic Scaling" on page 3-5
- "Scaling Filter Coefficients Manually" on page 3-6
- "Specifying Arithmetic Rules" on page 3-8

Because each section builds on the contents and filters from preceding sections, progressing through the tutorial from the start is most effective. Otherwise, code examples that depend on earlier tutorial sections may not work properly.

Designing Double Precision Floating-Point Filter Coefficients

To begin this tutorial, you design a lowpass FIR filter using the window method.

```
b = fir1(101, .45, 'low', kaiser(102, 7.8573));
```

Converting the Filter to Fixed-Point

The toolbox adds an `Arithmetic` property to `dfilt` objects that provides a way to analyze the filter not only in double-precision floating-point arithmetic but also in single-precision floating-point and fixed-point arithmetic.

```
hd = dfilt.dffir(b)

hd =

     FilterStructure: 'Direct-Form FIR'
          Arithmetic: 'double'
           Numerator: [1x102 double]
```

```
      PersistentMemory: false
```

When you install the Filter Design Toolbox, it adds an additional property value for the `Arithmetic` property of `dfilt` objects—`single`, indicating a filter that performs single-precision floating-point arithmetic. With the Fixed-Point Toolbox installed as well, you can set the `Arithmetic` property of the filter to `fixed` to turn quantization on and construct filters that perform fixed-point arithmetic.

Additional filter properties are added dynamically to the `dfilt` object when you set the `Arithmetic` property. This next display of the filter object enhances the readability of the list of properties by grouping them together in a logical manner.

Notice that only writable properties show in the listing. Also, some filter properties, such as `CoeffAutoScale`, control the display of other properties. For example, `CoeffAutoScale` controls the display of `NumFracLength` and whether you can write (change) the property value for `NumFracLength`.

```
hd
hd =

       FilterStructure: 'Direct-Form FIR'
            Arithmetic: 'fixed'
             Numerator: [1x102 double]
      PersistentMemory: false

       CoeffWordLength: 16
        CoeffAutoScale: true
                Signed: true

       InputWordLength: 16
       InputFracLength: 15


       FilterInternals: 'FullPrecision'
```

In contrast to the property display that the filter handle `hd` generates, the `get` function always returns the complete collection of properties and property values for the filter, whether or not you can change the property.

```
get(hd)
         PersistentMemory: 0
```

```
NumSamplesProcessed: 0
    FilterStructure: 'Direct-Form FIR'
             States: [101x1 embedded.fi]
          Numerator: [1x102 double]
         Arithmetic: 'fixed'
    CoeffWordLength: 16
     CoeffAutoScale: 1
             Signed: 1
          RoundMode: 'convergent'
       OverflowMode: 'wrap'

    InputWordLength: 16
    InputFracLength: 15

      NumFracLength: 16
    FilterInternals: 'FullPrecision'
   OutputWordLength: 39

   OutputFracLength: 31
  ProductWordLength: 32
  ProductFracLength: 31
    AccumWordLength: 39
    AccumFracLength: 31
```

Quantizing Filter Coefficients with Automatic Scaling

To determine the number of bits the filter is using to represent the filter coefficients, you look at the value of the `CoeffWordlength` property.

```
hd.CoeffWordLength
ans =

    16
```

To determine how the coefficients are being scaled, look at the `NumFracLength` property.

```
hd.NumFracLength
ans =

    16
```

This tells you that the filter coefficients are 16 bits long (the word length), and the least-significant bit (LSB) is weighed by 2^{-16} (the fraction length). The section "Notes About Fraction Length, Word Length, and Precision" on page 3-28 provides more information about interpreting the fraction length in the data format.

16 bits is just the default number the filters use for coefficients word length. To understand the scaling, you can look at the `CoeffAutoScale` setting.

```
hd.CoeffAutoScale % Returns a logical true = 1


ans =

     1
```

When the `CoeffAutoScale` property is true, the filter adjusts the fraction length of the coefficients to avoid overflow each time you change the coefficient word length. Verify this automatic scaling by changing the number of bits used to quantize the coefficients from 16 bits to 24 bits.

```
hd.CoeffWordLength = 24;
hd.NumFracLength
ans =

    24
```

The 2^{-24} weight has been computed automatically to represent the coefficients with the best precision possible while using the round-to-nearest setting (property value `round`) for the filter property `RoundMode`. "RoundMode" on page 8-89 provides further information about `RoundMode`.

Scaling Filter Coefficients Manually

Setting the `CoeffAutoScale` property to `false` turns the `NumFracLength` property writable and visible in the display.

```
h1 = copy(hd); % Keep a copy of the original object for...
               % latter comparison
h1.CoeffAutoScale = false
h1 =

         FilterStructure: 'Direct-Form FIR'
```

```
           Arithmetic: 'fixed'
            Numerator: [1x102 double]
     PersistentMemory: false
               States: [1x1 embedded.fi]
  NumSamplesProcessed: 0

      CoeffWordLength: 24
       CoeffAutoScale: false
        NumFracLength: 24
               Signed: true

      InputWordLength: 16
      InputFracLength: 15

     OutputWordLength: 16
           OutputMode: 'AvoidOverflow'

          ProductMode: 'FullPrecision'

            AccumMode: 'KeepMSB'
      AccumWordLength: 40
        CastBeforeSum: true

            RoundMode: 'convergent'
         OverflowMode: 'wrap'
```

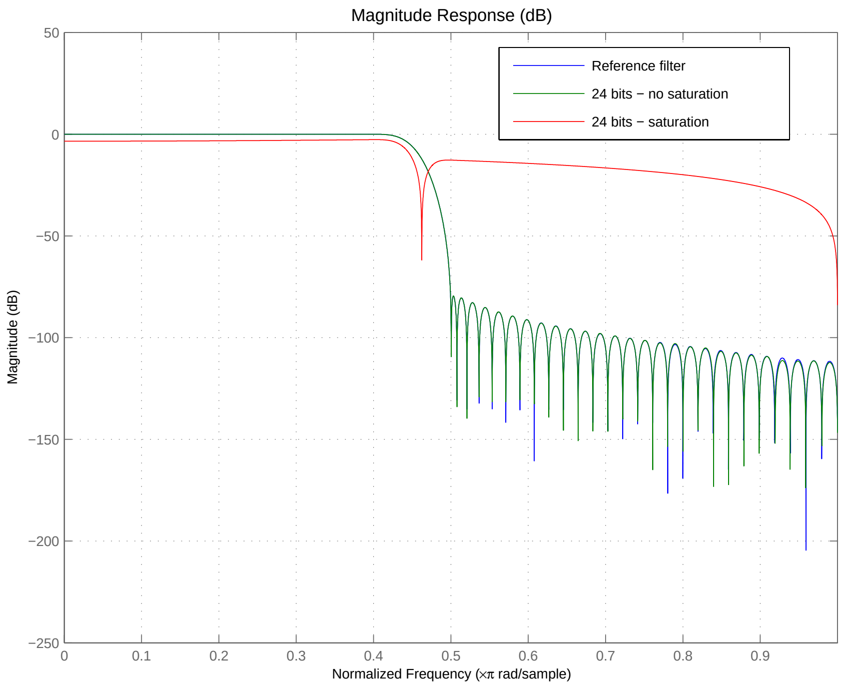
The quantized coefficients are always rounded to the nearest value and saturated when overflow occurs. Because the scaling process chose the fraction length to avoid overflow, increasing the fraction length saturates the quantized coefficients, introducing severe distortion in the magnitude response of the filter. Try increasing the numerator fraction length to 25 bits.

```
h1.NumFracLength = 25;
```

This is more clear when you plot the magnitude response to show the effect of saturating the coefficients. Here is the code to show the response.

```
href = reffilter(hd); % Get the reference double-precision...
                      % floating-point filter.
hfvt = fvtool(href,hd,h1);
set(hfvt,'ShowReference','off'); % Reference already displayed
```

```
legend(hfvt, 'Reference filter', '24 bits - no saturation',...
'24 bits - saturation')
set(hfvt, 'Color', [1 1 1])
```



Saturating the coefficients compromises the filter cutoff performance considerably.

Specifying Arithmetic Rules

Once you quantize the coefficients, you need to pay attention to the filter internal settings that govern how arithmetic is done inside the filter. For the remainder of this tutorial, you use a classic 16-bit word length filter.

```
hd.CoeffWordLength = 16;
```

Two properties—ProductMode and AccumMode—help you simulate different filter arithmetic scenarios in the multipliers and adders. Setting these properties to specify full precision (set the property values to FullPrecision) allows you to determine the minimum resources required to avoid losing any precision during filtering.

```
hd.ProductMode = 'FullPrecision'; % (default)
[hd.ProductWordLength hd.ProductFracLength]
ans =

    32    31
hd.AccumMode = 'FullPrecision';
[hd.AccumWordLength hd.AccumFracLength]
ans =

    39    31
```

Given an input format of [16 15] and coefficients format of [16 16]—the current settings for hd—these responses tell you that you need a product register twice the size of the coefficients (or twice the size of the input) and an accumulator register with seven guard bits to allow for bit growth during the accumulation process. They also tell you the position of the binary point in those registers—the AccumFraclength and ProductFracLength property values.

Starting from this ideal scenario that allows your filter to perform most accurately, you can introduce constraints on the product or the accumulator register or both. The KeepMSB option for the fraction length properties sets the fraction lengths automatically to avoid overflows while the KeepLSB option sets the fraction lengths automatically to avoid underflows. Finally, the SpecifyPrecision option allows full control of the settings. You need to run your filter to see the effect of these settings on the output.

For further discussion about product and accumulator settings, refer to the tutorial “Working with Fixed-Point Direct-Form FIR Filters” on page 3-33.

Constructing Fixed-Point Filters

You construct filters in the Filter Design Toolbox by either:

- Using the appropriate filter constructor function `dfilt.`*`structure`*, where *`structure`* is the filter topology to implement
- Using an `fdesign.response` object combined with a filter design method such as `butter`
- Using FDATool design features
- Copying an existing filter

All filter characteristics are stored as properties that you can set or retrieve. These filter characteristics include:

- Filter structure.
- Reference filter coefficients.
- Filter topology (single section or cascaded second-order sections).
- Fixed-point filter data format parameters such as:
  - Quantization parameters (word lengths, fraction lengths, and precisions).
  - Data type (signed or unsigned fixed-point; double-precision or single-precision floating-point; and signed or unsigned integers).
  - Rounding method used in quantization.
  - Overflow method used in quantization.
- Scaling factors for each section of a second-order section filter.

You can specify quantized filter properties by creating a quantized filter with default property values, and then changing some or all of these property values later

Defining Quantized and Fixed-Point Filters

With the `dfilt` objects in this toolbox you can create quantized and fixed-point filter objects that you use to filter signals or data. We distinguish between fixed-point and quantized filters only very rarely—mostly we use the term interchangeably. There is a difference between them that is worth noting and recalling as you work with the filter objects in this toolbox.

We use quantized to mean using limited precision arithmetic, either fixed-point or floating-point. Underlying all the filters in this toolbox, including the floating-point filters, is quantized arithmetic. With this in mind, we approximate ideal arithmetic (arithmetic with infinite precision) using double-precision, floating-point arithmetic and we refer to floating-point filters as nonquantized, or reference, filters.

Roughly explained, quantizing is the act of reducing the precision with which you represent numeric quantities.

Fixed-point arithmetic is a subset of quantized arithmetic, and fixed-point filters are thus a subset of quantized filters. In fixed-point arithmetic, the word length and fraction length you use limit the precision of your results, and arithmetic operations occur without moving the binary, or radix, point. Hence the name fixed-point or fixed binary-point arithmetic.

In summary, quantized filters use limited precision arithmetic and data representations, and fixed-point filters use limited precision representations and fixed-point arithmetic where the binary point location does not change.

Constructors for Fixed-Point Filters

The most direct way to create a fixed-point arithmetic filter (`dfilt` object) is to create one with the default properties. Fixed-point filter object construction requires two steps:

- Create a default double-precision filter `hd` by entering

  ```
  hd = dfilt.structure
  ```

- Change the `Arithmetic` property value for your filter object `hd` from `double` to `fixed`.

  ```
  set(hd,'arithmetic','fixed')
  ```

MATLAB displays a listing of all of the properties of the filter `hd` you just created, along with the associated property values. All property values are set to defaults when you construct a fixed-point filter this way.

Constructing a Quantized Filter from a Design Object

In general you construct quantized filters by constructing default filters or filters with specified filter coefficients. You begin with a set of unquantized filter coefficients to implement in a quantized filter. For this example you start with a design object that defines the type of filter to design.

```
d = fdesign.lowpass('n,fp,fst,ap',3,0.5,0.6,3);
```

To implement d as a quantized filter, use one of the design methods in the toolbox:

```
hd=ellip(d)

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [2x6 double]
             ScaleValues: [3x1 double]
        PersistentMemory: false
                  States: [2x2 double]
     NumSamplesProcessed: 0

set(hd,'arithmetic','fixed'); % Convert to quantized filter.
```

Since filters designed with a second-order section topology are more robust against quantization errors than those composed of higher order transfer functions, ellip constructs the dfilt object as an SOS filter.

Constructing a Fixed-Point Filter in Second-Order Sections

By default, many of the filter design functions in the toolbox return filters that use second-order sections. In most cases, this is a desirable feature when you are using fixed-point arithmetic because SOS filters tend to resist errors from quantization.

```
hs = fdesign.bandpass(.3, .4, .6, .7, 80, .5, 60);
designmethods(hs)


Design Methods for class fdesign.bandpass:
```

```
butter
cheby1
cheby2
ellip

hd=butter(hs)

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [13x6 double]
             ScaleValues: [14x1 double]
        PersistentMemory: false
                  States: [2x13 double]
     NumSamplesProcessed: 0
```

Copying Filters to Inherit Properties

If you already have a quantized filter `hd` with the property values set the way you want them, you can create a new quantized filter `hd2` with the same property values as `hd` by entering

```
hd2 = copy(hd)
```

This function is convenient to use when you are changing a small number of properties on a set of filters.

For example, create a 16-bit precision filter `hd` from an FIR reference filter with

```
b = fir1(80,0.5,kaiser(81,8)); % Reference filter
hd = dfilt.dffir(b)
hd2 = hd;
```

`hd2` inherits the property values for `hd`, but is an independent entity that you can change in any way you choose without affecting `hd`.

Fixed-Point Filter Structures

When you construct filter objects, the `FilterStructure` property value is returned containing one of the strings shown in the following table. Property

FilterStructure indicates the filter architecture and comes from the constructor you use to create the filter.

After you create a filter object you cannot change the FilterStructure property value. To make filters that use different structures you construct new filters using the appropriate object constructors. In some instances, function convert allows you to change the structure of an existing filter object.

You specify the filter structure by selecting the appropriate dfilt.*structure* method to construct your filter. For information about setting properties for fixed-point filter objects, refer to the reference information for dfilt in your Signal Processing Toolbox documentation and in this user's guide, and get and set in your MATLAB documentation.

The figures included in the reference page for each filter structure, such as dfilt.dfasymfir, act as aids to help you determine how to enter your filter coefficients for each filter structure and how the filter performs quantizations in the filter signal flow. Each reference page also contains an example for constructing a filter of the given structure.

Filter Constructor Name	FilterStructure Property String and Filter Type
dfilt.dfasymfir	Antisymmetric finite impulse response (FIR).
dfilt.df1	Direct form I.
dfilt.df1sos	Direct form I filter implemented using second-order sections.
dfilt.df1t	Direct form I transposed.
dfilt.df2	Direct form II.
dfilt.df2sos	Direct form II filter implemented using second order sections.
dfilt.df2t	Direct form II transposed.
dfilt.dffir	Direct form FIR.
dfilt.dffirt	Direct form FIR transposed.
dfilt.latticear	Lattice autoregressive (AR).

Filter Constructor Name	FilterStructure Property String and Filter Type
`dfilt.latticemamin`	Lattice moving average (MA) minimum phase.
`dfilt.latticemamax`	Lattice moving average (MA) maximum phase.
`dfilt.latticearma`	Lattice ARMA.
`dfilt.dfsymfir`	Symmetric FIR. Even and odd forms.
`dfilt.scalar`	Scalar

Fixed-Point Arithmetic Filter Structure Diagrams

To help you understand where quantizations occur in filter structures like those provided in the toolbox, Figure 3-1 presents the structure for a direct form 2 filter, including the quantizations that the quantized filter incorporates. You see that one or more quantizations accompany each filter element, such as a delay, coefficient, or summation element. The input to or output from each element reflects the result of applying the associated quantization.

Wherever a particular filter element appears in a filter structure, recall the quantization that accompanies the element as it appears in this figure. For example, a product quantization, either numerator or denominator, follows every coefficient element and a sum quantization, also either numerator or denominator, follows each sum element. In this figure, we show the structure for a direct-form II IIR filter, with the `arithmetic` property value set to '`fixed`'.

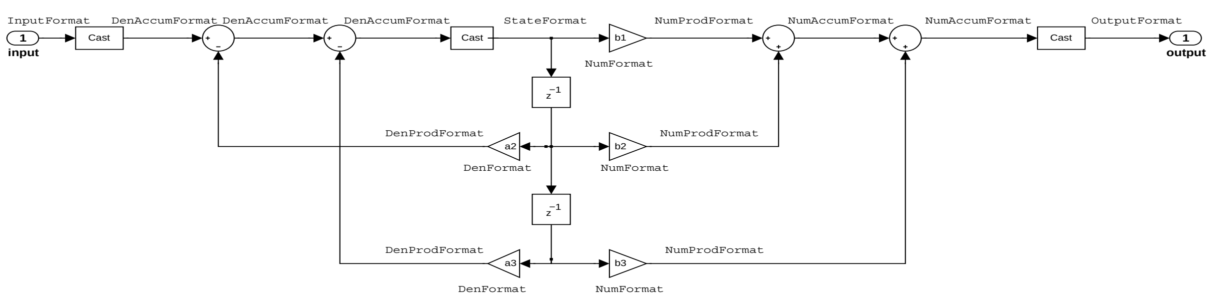


Figure 3-1: df2 IIR filter structure including the formatting quantizers, with Arithmetic property value 'fixed'

To set the `Arithmetic` property of a filter to `fixed` to create a fixed-point filter, your filter must have the leading denominator coefficient a(1) equal to zero.

Fixed-Point Arithmetic Filter Structures

You choose among several filter structures when you create quantized filters. You can also specify filters with single or multiple cascaded sections of the same type. Because quantization is a nonlinear process, different quantized filter structures produce different results.

About the Filter Structure Diagrams

In the diagrams that appear on each filter structure reference page, you see the active operators that define the filter, such as sums and gains, and the quantizers that control the processing in the filter. Notice also that the coefficients are labeled in the figure. This tells you the order in which the filter processes the coefficients.

While the meaning of the block elements is straightforward, the labels for the quantizers that form part of the filter, is less clear. Each figure includes text in the form *labelformat* that represents the existence of a quantization operation at that point in the structure. The *format* stands for word length and fraction length applied at that point in the filter flow, and the *label* specifies the data that the quantization process affects.

For example, in the `dfilt.df2` filter shown in Figure 3-1, the labels InputFormat and OutputFormat are the quantizations applied to the filter input and output data at the labeled location in the filter. InputFormat refers to the `InputWordLength and InputFracLength` filter properties and OutputFormat refers to the `OutputWordLength` and `OutputFracLength` filter properties. Property names like `CoeffWordLength` and `DenFracLength` define the properties that control filter operations with coefficients or denominator coefficients at that point in the structure and are properties of the filter.

Data Type Handling in Discrete-Time Filters

In this section you learn how discrete-time filters (`dfilt` objects) handle different data types in significant filtering areas:

- Different data types as input data to your filter
- Different data types to represent your filter coefficients
- Different data types used to represent the states of your filter
- Filter reference coefficients

How these varied filter areas respond is driven primarily by the value you select for the `Arithmetic` property of the filter object. The next sections cover each of the areas noted above and how each responds when you set the value for the `Arithmetic` property.

Property `Arithmetic` accepts one of three valid entries:

- `Double`
- `Single`
- `Fixed`

Each option affects how the filter handles the states, coefficients, input and output data, and filter arithmetic. And what you use as input to the filter object.

Filter Input Signals, Coefficients, and States

Filter object properties and their values directly affect how and in what form your filter works with input data, the filter coefficients, and the states of the filter.

fixed-point filters use fixed-point objects in many cases to handle fixed-point values such as coefficients, input, or filter states. The Fixed-Point Toolbox documentation provides details about the fixed-point, or `fi`, object that dfilt objects use.

Arithmetic Property Setting and Filter Input Data Types

The rules governing the input data that work with your filter object depend mostly on the setting of the `Arithmetic` property.

The arithmetic property setting controls the quantization of input to the filter, but all arithmetic property settings—double, single, fixed—support the same input data types:

- double-precision floating-point
- single-precision floating-point
- int*
- uint*
- `fi` objects

Each possible `Arithmetic` property value refines how the filter accepts input data. When you specify one of the following values for `Arithmetic`, here is what happens.

- `Arithmetic` set to `double`

  Setting the property value to `double` means the filter casts the input data to double-precision format. Also, the filter states and output are double data type as well. This is the default value for the filter `Arithmetic` property and the setting used when you create your filter. The resulting filter is considered double-precision, floating-point.
- `Arithmetic` set to `single`

  Selecting `single` means the filter casts the input data to single-precision format. Both the states and the output from the filter are in single data type. This is a quantized filter that uses single-precision floating-point data formats.
- `Arithmetic` set to `fixed`

  Setting `Arithmetic` to `fixed` means the filter casts the input data to fixed-point (`fi`) objects to use fixed-point formats defined by the filter properties [`InputWordLength` `InputFracLength`]; adds properties to the filter object for configuring the filter; and switches the filter to using fixed-point arithmetic. The added properties let you determine the data formats (the word length and fraction length) the filter uses for all filter operations and data.

Arithmetic Property Setting and Filter Coefficients Data Types

Changing the arithmetic mode controls the format the filter uses to represent coefficients. Discrete-time filters accepts coefficients in any of the following formats:

- double-precision floating-point
- single-precision floating-point
- int*
- uint*
- `fi` objects

Based on the setting you give for `Arithmetic`, whether `double`, `single`, or `fixed`, the filter handles the coefficients in the following manner:

- `Arithmetic = double`—the filter casts the coefficients to double data type. Reference coefficients for the filter are stored in the data type you provided them. In this case, the quantized and reference coefficients for the filter are identical.
- `Arithmetic = single`—the filter casts the coefficients to singles. Single data type coefficients are unchanged. Reference coefficients for the filter are stored in the data type that you use to provide them.
- `Arithmetic = fixed`—the filter casts the coefficients to fixed-point (`fi`) objects, using the `[InputWordLength InputFracLength]` filter properties to format the coefficients. The fixed-point filter stores the reference coefficients in the data type that you supplied. When you use `reffilter`, you get back a reference filter whose coefficients are double-precision approximations to the actual reference coefficients.

Reference Coefficients for Fixed-Point Filters

Quantized or fixed-point filters in the toolbox have both quantized coefficients (or fixed-point coefficients) that result from changing the `Arithmetic` property to `fixed` or `single`, and reference coefficients. You can access both sets from the command line.

How the toolbox stores the reference coefficients for a filter depends on the data type you use to specify the coefficients—reference coefficients are stored in the data type in which they were specified. Retaining the specified data type prevents the memory necessary for storing the coefficients from growing unnecessarily.

When you view the fixed-point filter coefficients, what you see are the double-precision approximations to the actual fixed-point or quantized coefficients used for filtering. In many cases, the approximation is exact, including when your filter uses single or double arithmetic.

If the `Arithmetic` property value is `fixed`, the approximation is exact whenever we can store the fixed-point values exactly as a double data type value. Otherwise, you see the double data type approximation of the value. We return the double-precision approximations so we can represent the leading denominator coefficient of an IIR filter exactly as a one, even if you are working in a fractional mode, such as Q15.

You use the function `reffilter` to return a filter that has the reference coefficients that accompany any fixed-point filter. For example, when you create a fixed-point direct form filter `hd` with

```
b=fir1(5,0.45);
hd = dfilt.dffir(b);
hd.arithmetic='fixed';
```

which has fixed point coefficients

```
hd.numerator

ans =

   -0.0044    0.0808    0.4235    0.4235    0.0808   -0.0044
```

Now change the word length the filter uses to represent the numerator coefficients.

```
hd.coeffautoScale=false

hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'fixed'
               Numerator: [-0.0044 0.0808 0.4235 0.4235 0.0808 -0.0044]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0
```

```
            CoeffWordLength: 16
             CoeffAutoScale: false
              NumFracLength: 16
                     Signed: true

            InputWordLength: 16
            InputFracLength: 15

           OutputWordLength: 16
                 OutputMode: 'AvoidOverflow'

                ProductMode: 'FullPrecision'

                  AccumMode: 'KeepMSB'
            AccumWordLength: 40
              CastBeforeSum: true

                  RoundMode: 'convergent'
               OverflowMode: 'wrap'
set(hd,'coeffWordLength',14');
hd.numerator

ans =

   -0.0044    0.0808    0.1250    0.1250    0.0808   -0.0044
```

Using `reffilter` returns a filter object with reference coefficients, as follows:

```
hdref=reffilter(hd)


hdref =

            FilterStructure: 'Direct-Form FIR'
                 Arithmetic: 'double'
                  Numerator: [-0.0044 0.0808 0.4235 0.4235 0.0808 -0.0044]
           PersistentMemory: false
                     States: [5x1 double]
        NumSamplesProcessed: 0
```

`hdref` has the original filter coefficients and is a double-precision filter. The reference filter coefficients match the original set of fixed-point coefficients for `hd`, but not the coefficients as represented with 14 bits.

Arithmetic Property Setting and Filter States Data Types

How the filter stores and operates on filter states depends on the setting of the `Arithmetic` property. You can provide the states in any of the following formats:

- double-precision floating-point
- single-precision floating-point
- int*
- uint*
- fixed-point (`fi`) objects

When you set the arithmetic setting you change how the filter responds to the state values.

- `Arithmetic = double`—the filter casts the states to double-precision data type.
- `Arithmetic = single`—the filter casts the filter states to single-precision data type format.
- `Arithmetic = fixed`—the filter casts the states to fixed-point objects, using the [`InputWordLength InputFracLength`] filter properties to format the states
  - fixed-point objects
  - double

  Other data types return an error in MATLAB.

Note that when you set `PersistentMemory` to `true` for your filter, the word length and fraction length settings for the filter states must be the same as the filter input word length and fraction length. If these settings do not match, you receive an error.

Note also that the filter does not store reference values for the states.

Disabling the autoscaling filter properties such as `CoeffAutoScale` and `InputAutoScale`, and `OutputAutoScale` results in making all of the additional fraction length properties available in the filter. To make turning off all of the automatic scaling for a filter easier, use `specifyall`. When you use

```
specifyall(hd)
```

all of the automatic control properties of `hd` are set to `SpecifyPrecision`:

- `AccumMode`
- `ProductMode`
- `OutputMode`

Also, `specifyall` disables the automatic scaling provided by

- `CoeffAutoScale`
- All other *`AutoScale` properties for the filter, since this varies from structure to structure

Now you have access to the fraction length properties for coefficients, the accumulator, products, and output values, which lets you set the precision yourself.

`specifyall` also helps you return your filter to the default automatic modes. Use the syntax

```
specifyall(hd,false)
```

to reset filter `hd` to the default automatic mode settings.

You might want more information about filter states after you read this review. Refer to `filtstates` in your Signal Processing Toolbox documentation for detail about filter states and the `filtstates` object the filters use.

Fixed-Point Filters and Second-Order Sections

Listed within the `dfilt` methods for creating quantized filters you find methods that return second-order section (SOS) versions of the direct form IIR filters—`df1sos`, `df1tsos`, `df2sos`, and `df2tsos`. The following figure shows how the second-order sections go together to form a filter, in this case a direct form II SOS filter. This diagram (or a similar one) appears with each SOS filter structure as well.

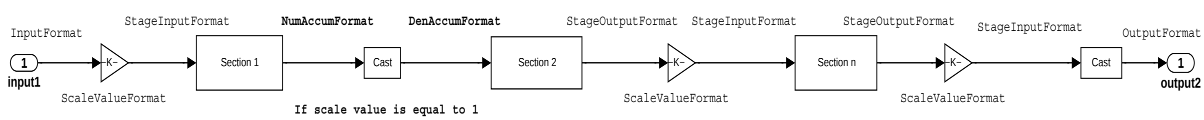


Figure 3-2: Cascaded structure of second-order sections for fixed-point filters. Note the location of the formatting information in the structure.

Combining this figure with the structures and signal flows for each SOS filter lets you work out the details about quantization in the SOS filter. This is not the same as cascading the filters, as the `dfilt.cascade` or `dfilt.parallel` methods in the Signal Processing Toolbox allow you to do with any `dfilt` objects.

The CastBeforeSum Property

Setting the `CastBeforeSum` property determines how the filter handles the input values to sum operations in the filter. After you set your filter `Arithmetic` property value to `fixed`, you have the option of using `CastBeforeSum` to control the data type of some inputs (addends) to summations in your filter. To determine which addends reflect the `CastBeforeSum` property setting, refer to the reference page for the signal flow diagram for the filter structure.

`CastBeforeSum` specifies whether to cast selected addends to summations in the filter to the output format from the addition operation before performing the addition. When you specify `true` for the property value, the results of the affected sum operations match most closely the results found on most digital signal processors. Performing the cast operation before the summation adds one or two additional quantization operations that can add error sources to your filter results.

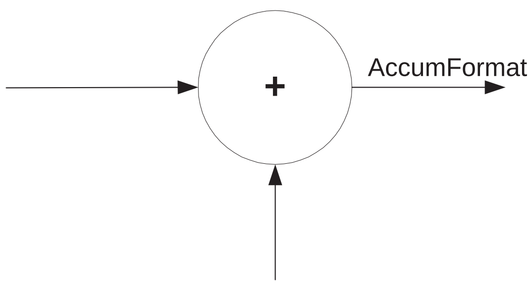
Specifying `CastBeforeSum` to be `false` prevents the addends from being cast to the output format before the addition operation. Choose this setting to get the most accurate results from summations without considering the hardware your filter might use.

Notice that the output format for every sum operation reflects the value of the output property specified in the filter structure diagram. Which input property is referenced by `CastBeforeSum` depends on the structure.

Property Value	Description
`false`	Configures filter summation operations to retain the addends in the format carried from the previous operation.
`true`	Configures filter summation operations to convert the input format of the addends to match the summation output format before performing the summation operation. Usually this generates results from the summation that more closely match those found from digital signal processors

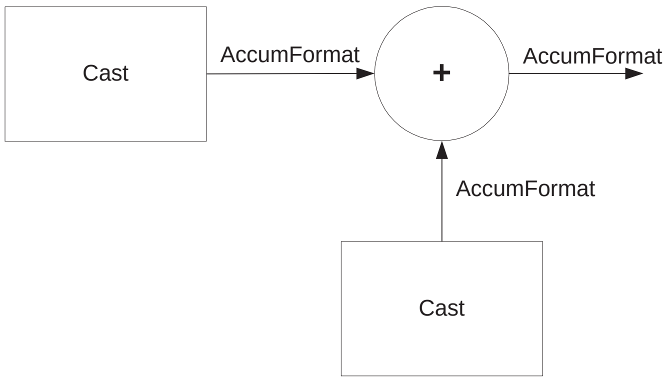
Diagrams of CastBeforeSum Settings

When `CastBeforeSum` is `false`, sum elements in filter signal flow diagrams look like this:



showing that the input data to the sum operations (the addends) retain their format word length and fraction length from previous operations. The addition process uses the existing input formats and then casts the output to the format defined by `AccumFormat`. Thus the output data has the word length and fraction length defined by `AccumWordLength` and `AccumFracLength` or `AccumMode` and `AccumWordLength`.

When `CastBeforeSum` is `true`, sum elements in filter signal flow diagrams look like this:



showing that the input data gets recast to the accumulator format word length and fraction length (AccumFormat) before the sum operation occurs. The data output by the addition operation has the word length and fraction length defined by `AccumWordLength` and `AccumFracLength` and `AccumMode`.

Introduction to Fixed-Point Arithmetic

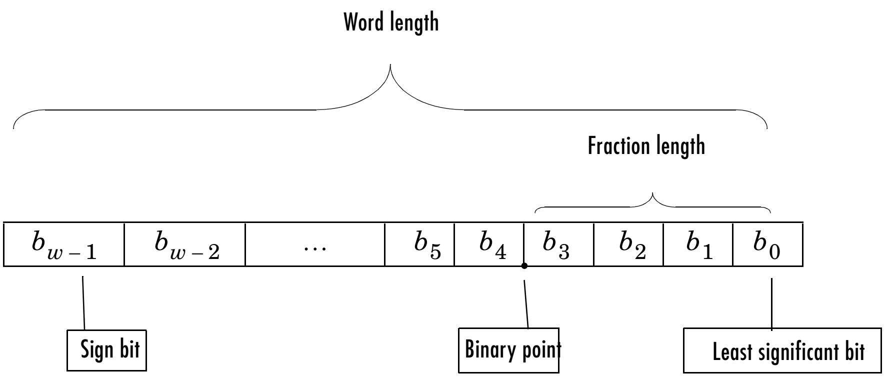
You can specify how numbers are quantized using fixed-point arithmetic in this toolbox with two quantities:

- Word length in bits
- Fraction length in bits

This toolbox does bit-true fixed-point arithmetic for all word lengths. It properly handles overflows and the results are bit-true results when the numbers are scaled properly. For example, (small numbers + small numbers) work correctly and (large numbers + large numbers) are right as well.

Fraction lengths used to represent numeric values can be any positive or negative integer, including larger than the associated word length for the value.

A general representation for a signed two's-complement binary fixed-point number is



where:

- b_i are the binary digits (bits, zeros or ones).
- The word length in bits is given by w.
- The most significant bit (MSB) is the leftmost bit. It is represented by the location of $b_{w\text{-}1}$. In Filter Design Toolbox, this number represents the sign bit; a 1 indicates the number is negative, and a 0 indicates it is nonnegative.

- The least significant bit (LSB) is the rightmost bit, represented by the location of b_0.
- The binary point is shown four places to the left of the LSB for this example.
- The fraction length *f* is the distance from the LSB to the binary point.

Binary Point Interpretation

Where you place the binary point determines how fixed-point numbers are interpreted in two's complement arithmetic. For example, the five bit binary number:

- 10110 represents the integer $-2^4+2^2+2 = -10$.
- 10.110 represents $-2+2^{-1}+2^{-2} = -1.25$.
- 1.0110 represents $-2^{-0}+2^{-2}+2^{-3} = -0.625$.

Notes About Fraction Length, Word Length, and Precision

Word length and fraction length combine to make the format for a fixed-point number, where word length is the number of bits used to represent the value and fraction length specifies, in bits, the location of the binary point in the fixed-point representation. Therein lies a problem—fraction length, which you specify in bits, can be larger than the word length, or a negative number of bits. This section explains how that idea works and how you might use it.

Unfortunately fraction length is somewhat misnamed (although it continues to be used in this User's Guide and elsewhere for historical reasons).

Fraction length defined as the number of fractional bits (bits to the right of the binary point) is true only when the fraction length is positive and less than or equal to the word length. In MATLAB format notation we use [word length fraction length]. For example, for the format [16 16], the second 16 (the fraction length) is the number of fractional bits or bits to the right of the binary point. In this example, all 16 bits are to the right of the binary point.

But it is also possible to have fixed-point formats of [16 18] or [16 -45]. In these cases the fraction length can no longer be the number of bits to the right of the binary point since the format says the word length is 16—there cannot be 18 fraction length bits on the right. And how can there be a negative number of bits for the fraction length, such as [16 -45]?

A better way to think about fixed-point format [word length fraction length] and what it means is that the representation of a fixed-point number is a weighted sum of powers of two driven by the fraction length, or the two's complement representation of the fixed-point number.

Consider the format [B L], where the fraction length L can be positive, negative, 0, greater than B (the word length) or less than B. (B and L are always integers and B is always positive.)

Given a binary string b(1) b(2) b(3) ... b(B), to determine the two's-complement value of the string in the format described by [B L], use the value of the individual bits in the binary string in the following formula, where b(1) is the first binary bit (and most significant bit, MSB), b(2) is the second, and on up to b(B).

The decimal numeric value that those bits represent is given by

```
value =-b(1)*2^(B-L-1)+b(2)*2^(B-L-2)+b(3)*2^(B-L-3)+...+ b(B)*2^(-L)
```

L, the fraction length, represents the negative of the weight of the last, or least significant bit (LSB). L is also the step size or the precision provided by a given fraction length.

Precision

Here is how precision works.

When all of the bits of a binary string are zero except for the LSB (which is therefore equal to one), the value represented by the bit string is given by $2^{(-L)}$. If L is negative, for example L=-16, the value is 2^{16}. The smallest step between numbers that can be represented in a format where L=-16 is given by 1 x 2^{16} (the rightmost term in the formula above), which is 65536. Note the precision does not depend on the word length.

Take a look at another example. When the word length set to 8 bits, the decimal value 12 is represented in binary by 00001100. That 12 is the decimal equivalent of 00001100 tells us we are using [8 0] data format representation—the word length is 8 bits and fraction length 0 bits, and the step size or precision (the smallest difference between two adjacent values in the format [8,0], is 2^0=1.

Suppose you plan to keep only the upper 5 bits and discard the other three. The resulting precision after removing the right-most three bits comes from the weight of the lowest remaining bit, the fifth bit from the left, which is 2^3=8, so the format would be [5,-3].

Note that in this format the step size is 8, I cannot represent numbers that are between multiples of 8.

In MATLAB, with the Fixed-Point Toolbox installed:

```
x=8;
q=quantizer([8,0]); % Word length = 8, fraction length = 0
xq=quantize(q,x);
binxq=num2bin(q,xq);
q1=quantizer([5 -3]); % Word length = 5, fraction length = -3
xq1 = quantize(q1,xq);
binxq1=num2bin(q1,xq1);
binxq

binxq =

00001000

binxq1

binxq1 =

00001
```

But notice that in [5,-3] format, 00001 is the two's complement representation for 8, not for 1; `q = quantizer([8 0])` and `q1 = quantizer([5 -3])` are not the same. They cover the about the same range—`range(q)>range(q1)`—but their quantization step is different—`eps(q)`= 8, and `eps(q1)=1`.

Look at one more example. When you construct a quantizer `q`

```
q = quantizer([a,b])
```

the first element in `[a,b]` is `a`, the word length used for quantization. The second element in the expression, `b`, is related to the quantization step—the numerical difference between the two closest values that the quantizer can represent. This is also related to the weight given to the LSB. Note that `2^(-b) = eps(q)`.

Now construct two quantizers, `q1` and `q2`. Let `q1` use the format [32,0] and let `q2` use the format [16, -16].

```
q1 = quantizer([32,0])
```

```
q2 = quantizer([16,-16])
```

Quantizers q1 and q2 cover the same range, but q2 has less precision. It covers the range in steps of 2^{16}, while q covers the range in steps of 1.

This lost precision is due to (or can be used to model) throwing out 16 least-significant bits.

An important point to understand is that in dfilt objects and filtering you control which bits are carried from the sum and product operations in the filter to the filter output by setting the format for the output from the sum or product operation.

For instance, if you use [16 0] as the output format for a 32-bit result from a sum operation when the original format is [32 0], you take the lower 16 bits from the result. If you use [16 -16], you take the higher 16 bits of the original 32 bits. You could even take 16 bits somewhere in between the 32 bits by choosing something like [16 -8], but you probably do not want to do that.

Dynamic Range and Precision

A fixed-point quantization scheme determines the dynamic range of the numbers that can be applied to it. Numbers outside of this range are always mapped to fixed-point numbers within the range when you quantize them. The precision is the distance between successive numbers occurring within the dynamic range in a fixed-point representation. The dynamic range and precision depend on the word length and the fraction length.

For a signed fixed-point number with word length w and fraction length f, the range is from -2^{w-f-1} to $2^{w-f-1}-2^{-f}$.

For an unsigned fixed-point number with word length w and fraction length f, the range is from 0 to $2^{w-f}-2^{-f}$.

In either case the precision is 2^{-f}.

Overflows and Scaling

When you quantize a number that is outside of the dynamic range for your specified precision, *overflows* occur. Overflows occur more frequently with fixed-point quantization than with floating-point, because the dynamic range of fixed-point numbers is much less than that of floating-point numbers with equivalent word lengths.

Overflows can occur when you create a fixed-point quantized filter from an arbitrary floating-point design. You can normalize your fixed-point filter coefficients and introduce a corresponding scaling factor for filtering to avoid overflows in the coefficients.

In this toolbox you can specify how you want overflows to be handled:

- Saturate on the overflow
- Wrap on the overflow

Working with Fixed-Point Direct-Form FIR Filters

This chapter ends with a tutorial that illustrates various aspects of working with direct-form FIR filters using fixed-point arithmetic.

As you follow this example, you learn about these topics while working with fixed-point filters:

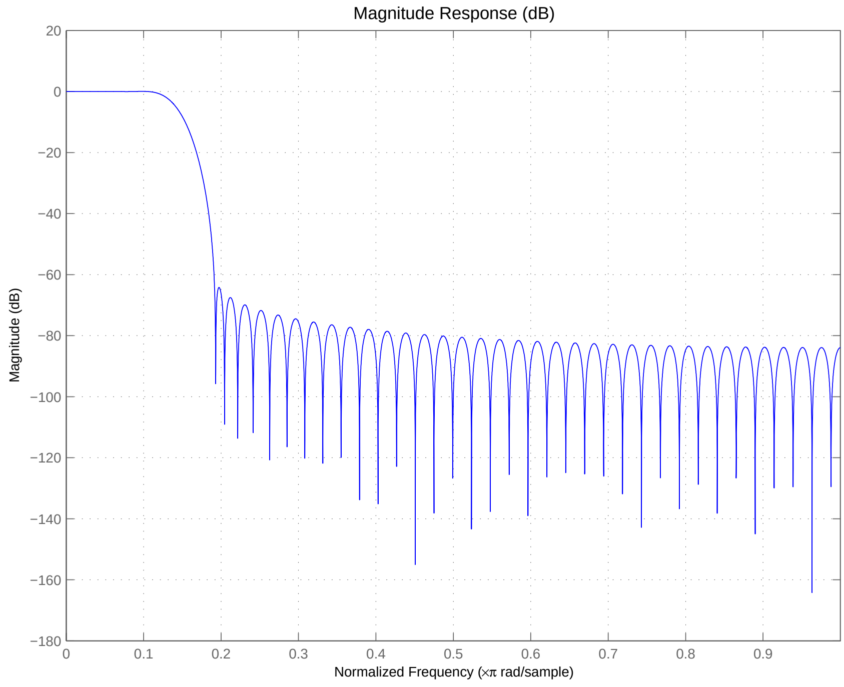
Since each section builds on the contents and filters from preceding sections, progressing through the tutorial from the start is most effective. Otherwise, code examples that depend on earlier tutorial sections may not work properly.

Obtaining the Filter Coefficients

For this tutorial, the FIR filter you use is not critical. Given the importance of direct-form FIR filters, you use the direct-form structure here—it does not even need to have linear phase. This demonstration uses a least-squares design method to obtain the filter coefficients. To display the filter, pass the filter coefficients to the Filter Visualization Tool (FVTool).

```
b = firls(80,[0 0.11 0.19 1],[1 1 0 0],[1 100]);
hfvt = fvtool(b);
```

Here is the magnitude response for b as shown by FVTool.



Creating the Direct-Form FIR Fixed-Point Filter

To create the fixed-point direct-form FIR filter using the coefficients you have requires two steps—choose a dfilt filter construction function and change the arithmetic setting for the filter to fixed-point arithmetic.

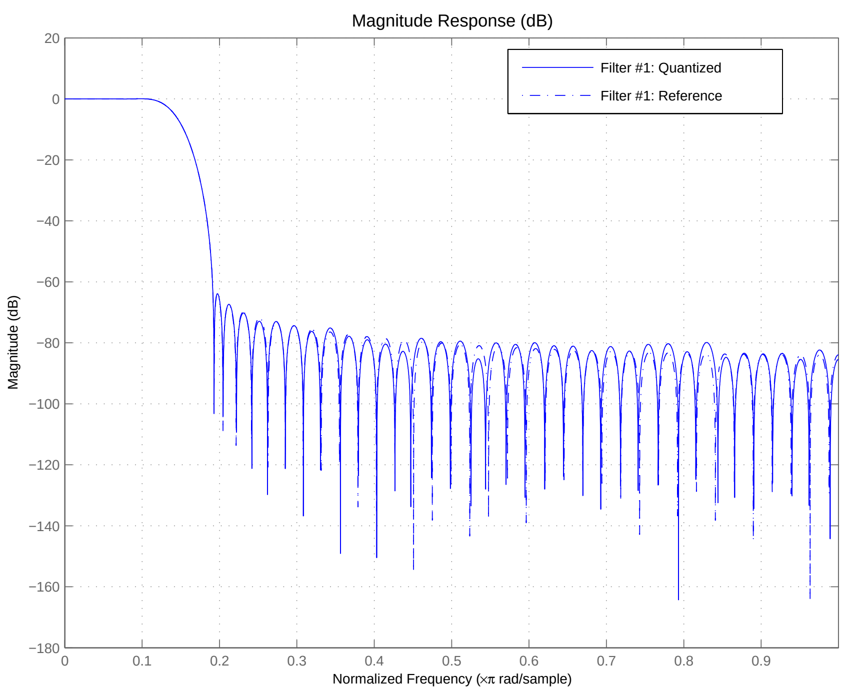
```
hd = dfilt.dffir(b); % Create the direct-form FIR filter.
set(hd,'Arithmetic','fixed');
```

Comparing Quantized Coefficients to Nonquantized Coefficients

There are several parameters for a fixed-point direct-form FIR filter. To start with, it is best to concentrate on the coefficient word length and fraction length (scaling). Use the Filter Visualization Tool to compare the quantized coefficients filter to the nonquantized (reference) coefficient filter.

```
set(hfvt, 'Filters', hd, 'legend','on');
```

FVTool returns the plot of the magnitude responses for both filters—the quantized filter and the corresponding reference filter. This is automatic—when you use FVTool to display a fixed-point filter, it plots both the quantized filter and the reference filter associated with the quantized version.



Determining the Number of Bits being Used

To determine the number of bits being used in the fixed-point filter `hd`, look at the `CoeffWordlength` property value. Check the `CoeffAutoScale` setting to determine how the filter is scaling the coefficients.

```
get(hd,'CoeffWordLength')
get(hd,'NumFracLength')
ans =

    16


ans =

    17
```

These values tells us that `hd` uses 16 bits to represent the coefficients, and the least-significant bit (LSB) is weighted by 2^{-17}. 16 bits is the default coefficient word length the filter uses for coefficients, but the 2^{-17} weight has been computed automatically to represent the coefficients with the best possible precision, given the `CoeffWordLength` value. You control this scaling through the `CoeffAutoScale` property. Set `CoeffAutoScale` to `false` to give yourself manual control of the coefficient scaling. The next command verifies that autoscaling is enabled in filter `hd`.

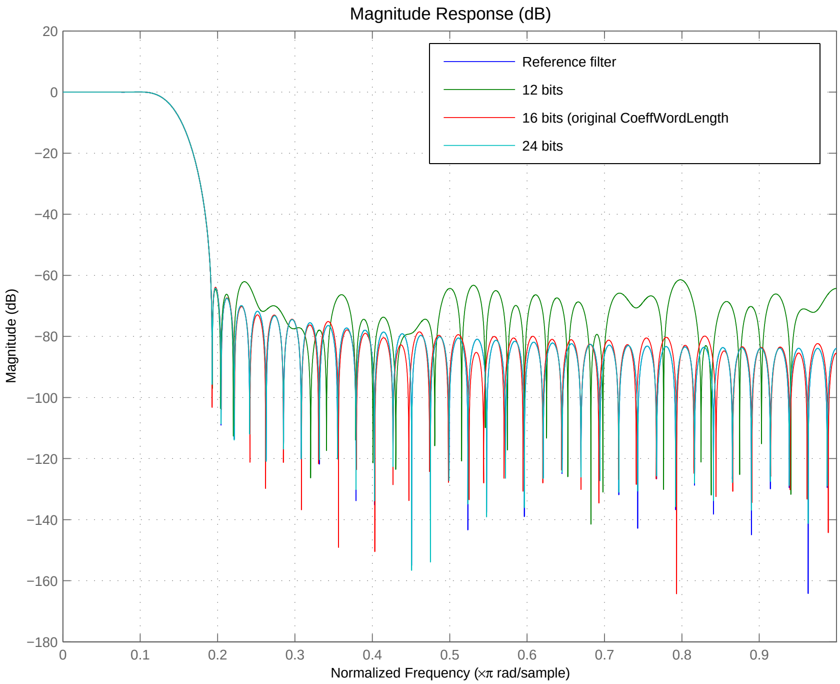
```
get(hd,'CoeffAutoScale') % Returns a logical true

ans =

     1
```

Determining the Proper Coefficient Word Length

Now make several copies of the filter to try different word lengths. Allow the coefficient autoscaling process to determine the best precision in each case. In the figure that follows the code presented here, you see the magnitude responses for the various version of `hd` (`h1`, `h2`, and the reference filter) so you can compare the effects of changing the coefficient word length.

```
h1 = copy(hd);
set(h1,'CoeffWordLength',12); % Use 12 bits.
```

```
h2 = copy(hd);
set(h2,'CoeffWordLength',24); % Use 24 bits.
href = reffilter(hd);
set(hfvt, 'Filters', [href, h1, hd, h2]);
set(hfvt,'ShowReference','off'); % Reference already displayed
                                 % once.
legend(hfvt,'Reference filter','12 bits','16 bits (original...
CoeffWordLength','24 bits');
```



12 bits are clearly not enough to accurately represent this filter. 16 bits may be enough for many applications.

For the remaining sections of this tutorial, you to use 16 bits to represent the filter coefficients.

As a rule-of-thumb, you should expect an attainable attenuation in the stop band of about 5 dB per bit—so 16 bits provides about 80 dB attenuation.

Fixed-Point Filtering

The main purpose of this tutorial is to evaluate the accuracy of the fixed-point filter when compared to a double-precision floating-point version of the same filter. Through the tutorial sections to come, you see that representing the filter coefficients such that the magnitude response of the fixed-point filter is about the same as the double-precision filter is not enough to ensure the performance of the fixed-point filter during filtering.

Generating Random Test Input Data

To evaluate the accuracy of the fixed-point filter, you use some random data to filter and compare against. You create a quantizer, with a range of [-1,1) to generate random, uniformly distributed white-noise data using 16 bits of word length.

```
rand('state',0); % Make results reproducible by initializing the
                 % random generator.
q = quantizer([16,15],'RoundMode','round');
xq = randquant(q,1000,1); % 1000 Data points in the range [-1,1).
xin = fi(xq,true,16,15);
```

Now `xin` is an array of integers with 1000 members, represented as a fixed-point object (a `fi` object).

```
get(xin)
DataTypeMode: 'Fixed-point: binary point scaling'

                DataType: 'Fixed'
                 Scaling: 'BinaryPoint'
                  Signed: 1
              WordLength: 16
          FractionLength: 15
           FixedExponent: -15
                   Slope: 3.0518e-005
    SlopeAdjustmentFactor: 1
                    Bias: 0
               RoundMode: 'round'
            OverflowMode: 'saturate'
```

```
            ProductMode: 'FullPrecision'
      ProductWordLength: 32
   MaxProductWordLength: 128
  ProductFractionLength: 30
                SumMode: 'FullPrecision'
          SumWordLength: 32
       MaxSumWordLength: 128
      SumFractionLength: 30
          CastBeforeSum: 1
```

Your Fixed-Point Toolbox documentation can provide more information about `fi` objects.

Generating a Baseline Output to Compare Against

When you evaluate the accuracy of fixed-point filtering, consider three quantities for the comparison between filtering with the quantized filter and filtering with the reference filter:

1. The ideal output—this quantity is what we would like to achieve. You compute it using the reference coefficients and double-precision floating-point arithmetic.

2. The best-you-can-hope-for output—this is the best you can hope to achieve. This value you compute using the quantized coefficients and double-precision floating-point arithmetic.

3. The output you can actually attain with the quantized filter—this is the output you compute using the quantized coefficients and fixed-point arithmetic.

Clearly you want to compare what you can actually attain (number 3) to the best you can hope for (number 2). To compute this last quantity (the best you can hope for), cast the fixed-point filter to double-precision and filter with double-precision floating-point arithmetic.

```
xdouble = double(xin);
hdouble = double(hd);
ydouble = filter(hdouble,xdouble);
```

Notice that you had to cast the input data xin to double format to use it with the double-precision filter hdouble. Double-precision filters require double-precision input values.

For completeness, here is how you compute the ideal output. Then you see how much quantizing only the filter coefficients affects the output of the filter.

```
yideal = filter(href,xdouble); % Reference filter, double data.
norm(yideal-ydouble)           % Total error.

ans =

  3.4886e-004

norm(yideal-ydouble,inf) % Maximum deviation.

ans =

  3.7219e-005
```

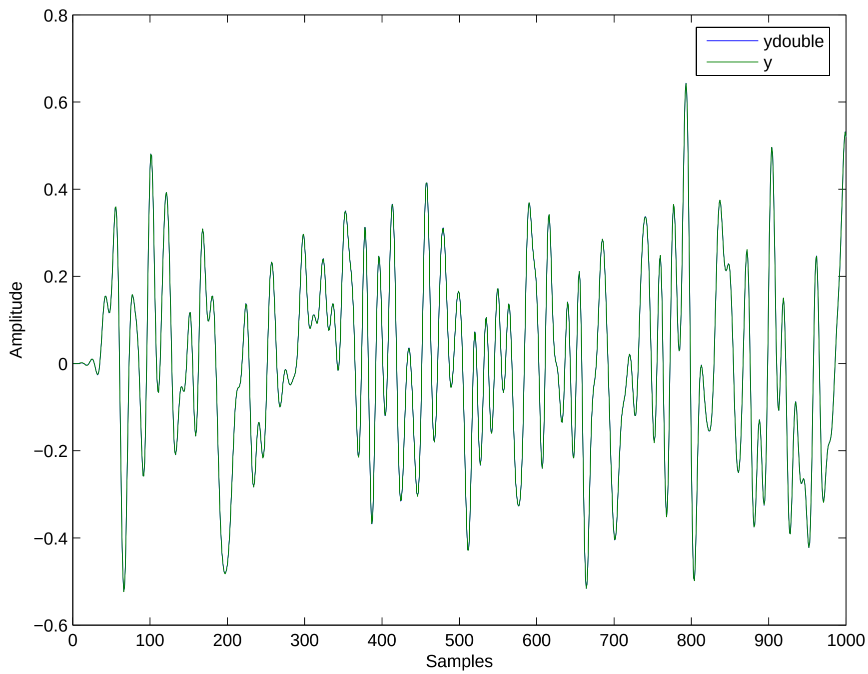
Computing the Fixed-Point Filter Output

Next you perform the actual fixed-point filtering. Again, the best you can hope to achieve is to have an output identical to ydouble.

```
y = filter(hd,xin);
norm(double(y)-ydouble)     % Total error.

ans =

    0.0178

norm(double(y)-ydouble,inf) % Maximum deviation.


ans =


  9.7186e-004
```

Improving the Filtering Results

Initially, the fixed-point filter output seems reasonably close to the best you can hope for performance. Plotting y and ydouble, the next figure, shows there is hardly any visible difference between them. (it is necessary to zoom in to reveal any differences.) After the figure you see the code to generate the plot.




```
plot([ydouble,double(y)])
xlabel('Samples'); ylabel('Amplitude')
legend('ydouble','y')
```

The question is whether you can improve the output accuracy. To address this you first examine the output to determine the range necessary to represent y.

```
norm(double(y),inf) % Maximum absolute value of y.
```

```
ans =

    0.6426
```

Changing the Filter Output Settings

Next, examine the properties that control the output from the filter.

```
get(hd,'OutputWordLength')
get(hd,'OutputFracLength')

ans =

    16


ans =

     9
```

OutputWordLength tells you that 16 bits are being used to represent the output and OutputFracLength tells you that the LSB is weighted by 2^{-9}. These settings, or this format [16 9] can represent the numerical range [-64, 64), which is much too large for the maximum absolute value of the output. The price you pay for this range is reduced precision within the range. Precision has to do with the output mode for the filter:

```
get(hd,'OutputMode')

ans =

AvoidOverflow
```

The AvoidOverflow property setting for OutputMode is a very conservative, essentially worst-case scenario, setting. It is there to avoid overflows regardless of the input signal and regardless of the actual value of the filter coefficients. A more accurate and efficient value is BestPrecision. This adjusts the output fraction length to accommodate each input signal, setting the fraction length for the output to provide the best precision possible on a case-by-case basis.

```
set(hd,'OutputMode', 'BestPrecision');
```

```
y = filter(hd,xin);
get(hd,'OutputFracLength')

ans =

    15

norm(double(y)-ydouble)     % Total error.
norm(double(y)-ydouble,inf) % Maximum deviation.

ans =

  2.7623e-004


ans =

  1.5251e-005
```

With these settings, your total and maximum errors have diminished significantly. You are still using 16 bits to represent the output, but the scaling has been improved to obtain better precision. Now the representable range at the filter output is -1 to 1.

Further Reducing Filter Output Quantization

To isolate any other quantization errors that are being introduced in the filter, you can eliminate quantization error at the output completely by setting the output format to have the same specifications as the accumulator. Think of this as being able to look inside the accumulator:

```
set(hd,'OutputMode','SpecifyPrecision');
set(hd,'OutputWordLength',get(hd,'AccumWordLength'));
set(hd,'OutputFracLength',get(hd,'AccumFracLength'));
y = filter(hd,xin);
norm(double(y)-ydouble)     % Total error.
ans =

     0

norm(double(y)-ydouble,inf) % Maximum deviation.
```

```
ans =

     0
```

The errors are exactly zero, indicating that the accumulator is not adding further quantization to the output. The arithmetic products (multiplies) are set by default to use full precision, so you know that no errors are occurring in multiplication operations.

Usually it is not possible to have a full 40-bit output of the filter, so you must expect some difference between `y` and `ydouble`. Nevertheless, you have verified that in this filtering case, the difference between the ideal filter and the quantized filter is due to output quantization. This is not always the case—in some cases bits get lost in the accumulator. In fact overflow may occur in the accumulator.

The Advantages of Guard Bits

If you compare the arithmetic product word length and fraction length with the accumulator word and fraction lengths by looking at the associated filter properties

```
get(hd,'ProductWordLength')
get(hd,'ProductFracLength')

ans =

    32


ans =

    32

get(hd,'AccumWordLength')
get(hd,'AccumFracLength')
ans =

    40
```

```
ans =

    32
```

you see that the accumulator has 8 extra bits available (AccumWordLength is 40 bits). This is typical of most fixed-point DSP processors. The extra bits are usually referred to as guard bits. They provide a safety valve for intermediate overflows.

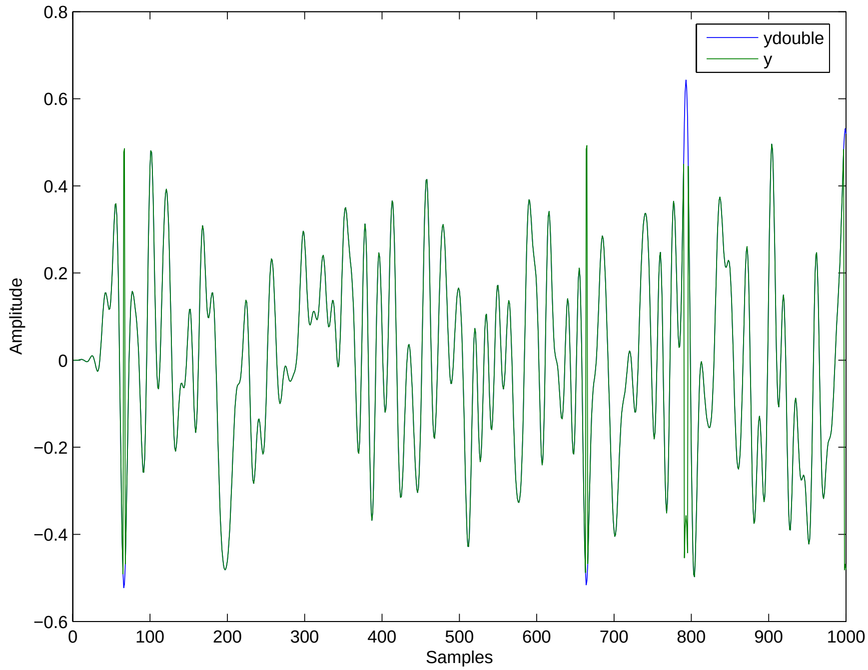
The easiest way of appreciating their value is to remove them and see what happens (adjust the output setting accordingly),

```
set(hd,'AccumMode','SpecifyPrecision');
set(hd,'AccumWordLength',get(hd,'ProductWordLength'));
set(hd,'OutputWordLength',get(hd,'AccumWordLength'));

hd

hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'fixed'
               Numerator: [1x81 double]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 1000

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 32
              OutputMode: 'SpecifyPrecision'
        OutputFracLength: 32

             ProductMode: 'FullPrecision'

               AccumMode: 'SpecifyPrecision'
```

```
          AccumWordLength: 32
          AccumFracLength: 32
            CastBeforeSum: true

                RoundMode: 'convergent'
             OverflowMode: 'wrap'
```

Now the accumulator word length matches the product word length of 32 bits, and the output word length matches the accumulator word length, also 32 bits. Now use hd to filter some data, and plot the results.

```
y = filter(hd,xin);
norm(double(y)-ydouble)     % Total error.

ans =

    3.4641

norm(double(y)-ydouble,inf) % Maximum deviation.
ans =

     1

plot([ydouble,double(y)])
xlabel('Samples'); ylabel('Amplitude')
legend('ydouble','y')
```



The total error is large now because overflow occurred during filtering. The representable range for the accumulator and output formats [32 32] is -0.5 to 0.5. When you look at the plot, one of the amplitudes around sample 800 is larger than 0.5, indicating an overflow. Recall that you have the output settings equal to the accumulator settings. So the overflow is occurring in the accumulator itself—you removed the guard bits by setting the accumulator word length to 32 bits.

Avoiding Overflow Without Guard Bits

It is possible not to have overflow even when guard bits are not available in the accumulator. The easiest way to do this is to use the `KeepMSB` property (keep the MSB) value for the `AccumMode` property for your filter.

```
set(hd,'AccumMode','KeepMSB');
```

```
set(hd,'OutputFracLength',get(hd,'AccumFracLength'));
y = filter(hd,xin);
norm(double(y)-ydouble)     % Total error.
norm(double(y)-ydouble,inf) % Maximum deviation.

ans =

  2.4442e-006

ans =

  2.5332e-007
```

`KeepMSB` is a very conservative accumulator mode for any input data. The error values seem small because there is no output quantization error in this case. the output format is [32 32].

If we use 16 bits for the output word length and set the output mode to maintain the best precision for this word length, the resulting error is much larger—almost two orders of magnitude.

```
set(hd,'OutputWordLength',16);
set(hd,'OutputMode','BestPrecision');
y = filter(hd,xin);
norm(double(y)-ydouble)     % Total error.

ans =

  2.7627e-004

norm(double(y)-ydouble,inf)  % Maximum deviation.

ans =

  1.5400e-005
```

From the earlier plots of `y` and `ydouble`, you might have realized that one extra bit was all that would have been required to avoid overflow in those examples. You can improve the results slightly with this one bit change, but remember that this is specific to the filter coefficients and input signal you are using in this tutorial.

Reducing the accumulator fraction length from 32 bits to 31 bits provides one more bit in the integer part of the accumulator word and reduces the filtering error.

```
set(hd,'AccumMode','SpecifyPrecision');
set(hd,'AccumFracLength',31);
y = filter(hd,xin);
norm(double(y)-ydouble)     % Total error.
norm(double(y)-ydouble,inf) % Maximum deviation.
ans =

  2.7623e-004


ans =

  1.5251e-005
```

The errors are the same as when you used 40 bits for the accumulator and 2^{-32} to scale the least-significant bit. This indicates that the errors in the filtering are due to quantization effects between the accumulator and the output.

4

Designing Adaptive Filters

Over the past three decades, digital signal processors have made great advances in increasing speed and complexity, and reducing power consumption. As a direct result, real-time adaptive filtering is quickly becoming an enabling technology for the future of communications, both wired and wireless. In the following sections, this guide presents an overview of adaptive filtering; discussions of some of the common applications for adaptive filters; and details about the adaptive filters available in the toolbox.

Listed below are the sections that cover adaptive filters in this guide. Within each section, examples and a short discussion of the theory of the filters introduces the adaptive filter concepts.

- "Overview of Adaptive Filters and Applications" on page 4-4—presents a general discussion of adaptive filters and their applications.
  - "System Identification" on page 4-7—talks using adaptive filters to identify the response of an unknown system such as a communications channel or a telephone line.
  - "Inverse System Identification" on page 4-8—talks about using adaptive filters to develop a filter which has a response that is the inverse of an unknown system. You can overcome echoes in modem connections and local telephone lines by inserting an inverse adaptive filter and using it to compensate for the induced noise on the lines.
  - "Noise Cancellation (or Interference Cancellation)" on page 4-9—useful for performing active noise cancellation where the filter adapts in real-time to keep the error small. Compare this to system identification where the filter adapts once and stays fixed thereafter.
  - "Prediction" on page 4-9—describes using adaptive filters to predict a signals future values.
- "Adaptive Filters in the Filter Design Toolbox" on page 4-11 lists the adaptive filters included in the toolbox.
- "Examples of Adaptive Filters That Use LMS Algorithms" on page 4-17 presents a discussion of using LMS techniques to perform the filter adaptation process.
- "Example of Adaptive Filter That Uses RLS Algorithm" on page 4-37 discusses adaptive filters based on the RMS techniques for minimizing the total error between the known and unknown systems.

For more detailed information about adaptive filters and adaptive filter theory, refer to the books listed in "Selected Bibliography" on page 4-48.

Overview of Adaptive Filters and Applications

Adaptive filters self learn. As the signal into the filter continues, the adaptive filter coefficients adjust themselves to achieve the desired result, such as identifying an unknown filter or cancelling noise in the input signal. In Figure 4-1, the shaded box represents the adaptive filter, comprising the adaptive filter and the adaptive RLS algorithm. For the general adaptive algorithm block diagram, look at Figure 4-2.

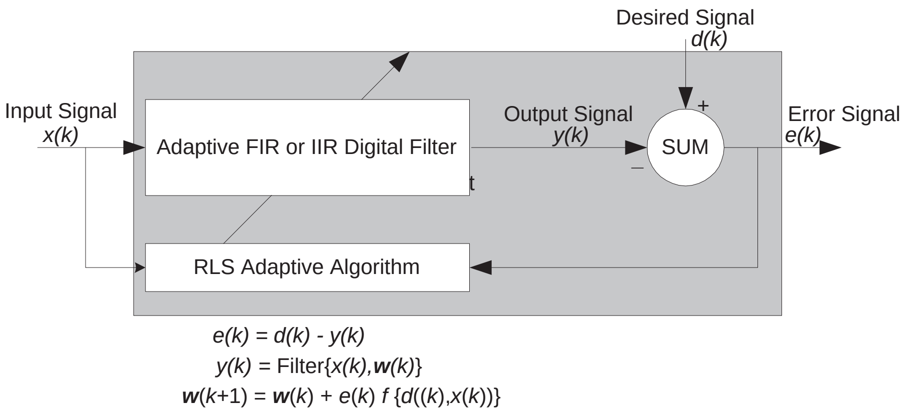


Figure 4-1: Block Diagram That Defines the Inputs and Output of a Generic RLS Adaptive Filter

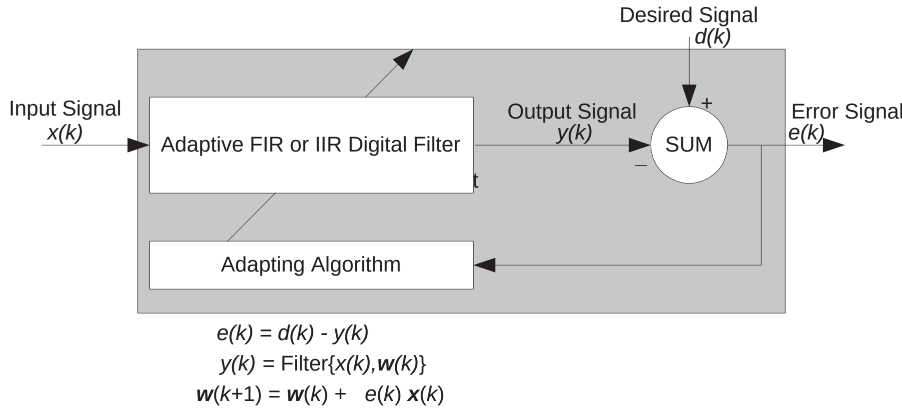


Figure 4-2: Block Diagram Defining General Adaptive Filter Algorithm Inputs and Outputs

An adaptive FIR or IIR filter designs itself based on the characteristics of the input signal to the filter and a signal which represent the desired behavior of the filter on its input. Designing the filter does not require any other frequency response information or specification. To define the self learning process the filter uses, you select the adaptive algorithm used to reduce the error between the output signal *y(k)* and the desired signal *d(k)*. When the LMS performance criteria for *e(k)* has achieved its minimum value through the iterations of the adapting algorithm, the adaptive filter is finished and its coefficients have converged to a solution. Now the output from the adaptive filter matches closely the desired signal *d(k)*. When you change the input data characteristics, sometimes called the filter environment, the filter adapts to the new environment by generating a new set of coefficients for the new data. Notice that when *e(k)* goes to zero and remains there you achieve perfect adaptation; the ideal result but not likely in the real world.

The adaptive filter functions in this toolbox implement the shaded portion of Figure 4-1, replacing the adaptive algorithm with an appropriate technique. Therefore, to use one of the functions you provide the input signal or signals and the initial values for the filter. A later section in this *User's Guide*, "Adaptive Filters in the Filter Design Toolbox" offers details about the algorithms available and the inputs required to use them in MATLAB.

Choosing an Adaptive Filter

With many adaptive filters to choose from, selecting the one that best meets your needs requires careful consideration. An exhaustive discussion of the criteria for selecting your approach is beyond the scope of this *User's Guide*. However, a few guidelines can help you make your choice.

Two main considerations frame the decision—the filter job to do and the filter algorithm to use.

When you begin to develop an adaptive filter for your needs, most likely the primary concern is whether using an adaptive filter is a cost-competitive approach to solving your filtering needs. Generally many areas determine the suitability of adaptive filters (these areas are common to most filtering and signal processing applications). Four such areas are:

- Filter consistency—does your filter performance degrade when the filter coefficients change slightly as a result of quantization, or you switch to fixed-point arithmetic? Will excessive noise in the signal hurt the performance of your filter?
- Filter performance—does your adaptive filter provide sufficient identification accuracy or fidelity, or does the filter provide sufficient signal discrimination or noise cancellation to meet your requirements?
- Tools—do tools exist that make your filter development process easier? Better tools can make it practical to use more complex adaptive algorithms.
- DSP requirements—can your filter perform its job within the constraints of your application. Does your processor have sufficient memory, throughput, and time to use your proposed adaptive filtering approach? Can you trade memory for throughput: use more memory to reduce the throughput requirements or use a faster signal processor?

Of the preceding considerations, characterizing filter consistency or robustness may be the most difficult.

Using the simulations in the Filter Design Toolbox offers a good first step in developing and studying these issues. Often, beginning your study using one of the least mean squares (LMS) algorithm filters provides both a relatively straightforward filter to implement and a sufficiently powerful tool for evaluating whether adaptive filtering can be useful for your problem.

Additionally, starting with an LMS approach can form a solid baseline against which you can study and compare the more complex adaptive filters available in the toolbox. Finally, your development process should, at some time, test your algorithm and adaptive filter with real data. For truly testing the value of your work there is no substitute for actual data.

With these considerations in mind, here are some applications that commonly use adaptive filters.

System Identification

One common application is to use adaptive filters to identify an unknown system, such as the response of an unknown communications channel or the frequency response of an auditorium, to pick fairly divergent applications. Other applications include echo cancellation and channel identification.

In the figure, the unknown system is placed in parallel with the adaptive filter.

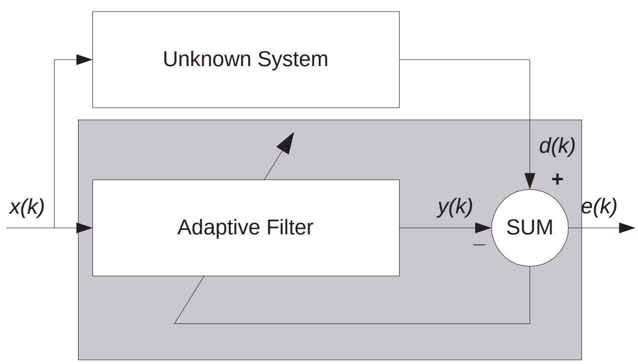


Figure 4-3: Using an Adaptive Filter to Identify an Unknown System

Clearly, when *e(k)* is very small, the adaptive filter response is close to the response of the unknown system. In this case the same input feeds both the adaptive filter and the unknown. When the unknown system is a modem, the input often represents white noise, and is the sound you hear from your modem when you log in to your Internet service provider.

Inverse System Identification

By placing the unknown system in series with your adaptive filter, your filter becomes the inverse of the unknown system when *e(k)* gets very small. As shown in the figure the process requires a delay inserted in the desired signal *d(k)* path to keep the data at the summation synchronized. Adding the delay keeps the system causal.

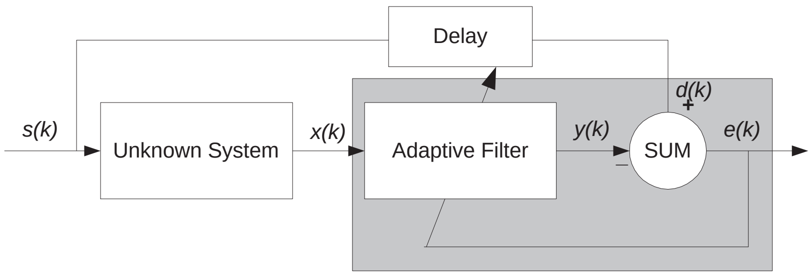


Figure 4-4: Determining an Inverse Response to an Unknown System

Without the delay element, the adaptive filter algorithm tries to match the output from the adaptive filter (*y(k)*) to input data (*x(k)*) that has not yet reached the adaptive elements because it is passing through the unknown system. In essence, the filter ends up trying to look ahead in time. As hard as it tries, the filter can never adapt: *e(k)* never reaches a very small value and your adaptive filter never compensates for the unknown system response. And it never provides a true inverse response to the unknown system. Including a delay equal to the delay caused by the unknown system prevents this condition.

Plain old telephone systems (POTS) commonly use inverse system identification to compensate for the copper transmission medium. When you send data or voice over telephone lines, the copper wires behave like a filter, having a response that rolls off at higher frequencies (or data rates) and possibly having other anomalies as well. Adding an adaptive filter which has a response that is the inverse of the wire response, adapting in real time, removes the rolloff and the anomalies, increasing the available frequency range and data rate for the telephone system.

Noise Cancellation (or Interference Cancellation)

In noise cancellation, adaptive filters let you remove noise from a signal in real time. Here, the desired signal, the one to clean up, combines noise and desired information. To remove the noise, feed a signal, *n'(k)* to the adaptive filter that represents noise that is correlated to the noise to remove from our desired signal.

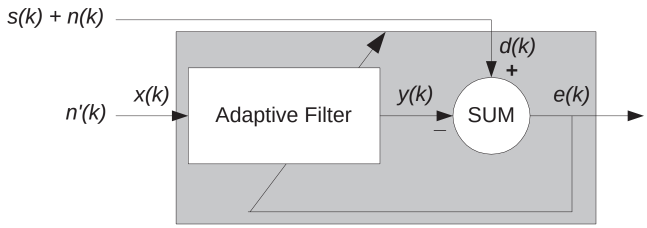


Figure 4-5: Using an Adaptive Filter to Remove Noise from an Unknown System

So long as the input noise to the filter remains correlated to the unwanted noise accompanying the desired signal, the adaptive filter adjusts its coefficients to reduce the value of the difference between *y(k)* and *d(k),* removing the noise and resulting in a clean signal in *e(k)*. Notice that in this application, the error signal actually converges to the input data signal, rather than converging to zero.

Prediction

Predicting signals may seem to be an impossible task, without some limiting assumptions. Assume that the signal is either steady or slowly varying over time, and periodic over time as well.

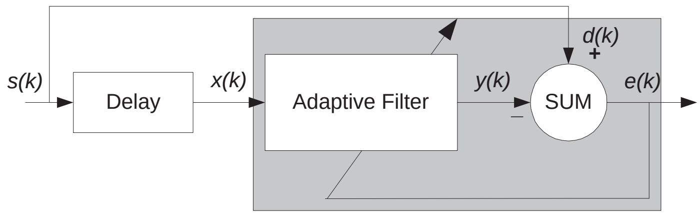


Figure 4-6: Predicting Future Values of a Periodic Signal

Accepting these assumptions, the adaptive filter must predict the future values of the desired signal based on past values. When *s(k)* is periodic and the filter is long enough to remember previous values, this structure with the delay in the input signal, can perform the prediction. You might use this structure to remove a periodic signal from stochastic noise signals.

Finally, notice that most systems of interest contain elements of more than one of the four adaptive filter structures. Carefully reviewing the real structure may be required to determine what the adaptive filter is adapting to. Also, for clarity in the figures, the analog-to-digital (A/D) and digital-to-analog (D/A) components do not appear. Since the adaptive filters are assumed to be digital in nature, and many of the problems produce analog data, converting the input signals to and from the analog domain is probably necessary.

Adaptive Filters in the Filter Design Toolbox

Filter Design Toolbox contains many objects for constructing and applying adaptive filters to data. As you see in the tables in the next section, the objects use various algorithms to determine the weights for the filter coefficients of the adapting filter. While the algorithms differ in their detail implementations, the LMS and RLS share a common operational approach—minimizing the error between the filter output and the desired signal.

Algorithms

For adaptive filter (adaptfilt) objects, the *`algorithm`* string determines which adaptive filter algorithm your `adaptfilt` object implements. Each available algorithm entry appears in one of the tables along with a brief description of the algorithm. Click on the algorithm in the first column to get more information about the associated adaptive filter technique.

- LMS based adaptive filters
- RLS based adaptive filters
- Affine projection adaptive filters
- Adaptive filters in the frequency domain
- Lattice based adaptive filters

Least Mean Squares (LMS) Based FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.adjlms`	Use the Adjoint LMS FIR adaptive filter algorithm
`adaptfilt.blms`	Use the Block LMS FIR adaptive filter algorithm
`adaptfilt.blmsfft`	Use the FFT-based Block LMS FIR adaptive filter algorithm
`adaptfilt.dlms`	Use the delayed LMS FIR adaptive filter algorithm
`adaptfilt.filtxlms`	Use the filtered-x LMS FIR adaptive filter algorithm
`adaptfilt.lms`	Use the LMS FIR adaptive filter algorithm
`adaptfilt.nlms`	Use the normalized LMS FIR adaptive filter algorithm
`adaptfilt.sd`	Use the sign-data LMS FIR adaptive filter algorithm
`adaptfilt.se`	Use the sign-error LMS FIR adaptive filter algorithm
`adaptfilt.ss`	Use the sign-sign LMS FIR adaptive filter algorithm

For further information about an adapting algorithm, refer to the reference page for the algorithm.

Recursive Least Squares (RLS) Based FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.ftf`	Use the fast transversal least-squares adaptation algorithm
`adaptfilt.qrdrls`	Use the QR-decomposition RLS adaptation algorithm
`adaptfilt.hrls`	Use the householder RLS adaptation algorithm
`adaptfilt.hswrls`	Use the householder SWRLS adaptation algorithm
`adaptfilt.rls`	Use the recursive-least squares (RLS) adaptation algorithm
`adaptfilt.swrls`	Use the sliding window (SW) RLS adaptation algorithm
`adaptfilt.swftf`	Use the sliding window ftf adaptation algorithm

For more complete information about an adapting algorithm, refer to the reference page for the algorithm.

Affine Projection (AP) FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.ap`	Use the affine projection algorithm that uses direct matrix inversion
`adaptfilt.apru`	Use the affine projection algorithm that uses recursive matrix updating
`adaptfilt.bap`	Use the block affine projection adaptation algorithm

To find more information about an adapting algorithm, refer to the reference page for the algorithm.

FIR Adaptive Filters in the Frequency Domain (FD)

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.fdaf`	Use the frequency domain adaptation algorithm
`adaptfilt.pbfdaf`	Use the partition block version of the fdaf algorithm
`adaptfilt.pbufdaf`	Use the partition block unconstrained version of the fdaf algorithm
`adaptfilt.tdafdct`	Use the transform domain adaptation algorithm using DCT
`adaptfilt.tdafdft`	Use the transform domain adaptation algorithm using DFT
`adaptfilt.ufdaf`	Use the unconstrained fdaf algorithm for adaptation

For more information about an adapting algorithm, refer to the reference page for the algorithm.

Lattice Based (L) FIR Adaptive Filters

adaptfilt.algorithm String	**Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation**
`adaptfilt.gal`	Use the gradient adaptive lattice filter adaptation algorithm
`adaptfilt.lsl`	Use the least squares lattice adaptation algorithm
`adaptfilt.qrdlsl`	Use the QR decomposition RLS adaptation algorithm

For more information about an adapting algorithm, refer to the reference page for the algorithm.

Presenting a detailed derivation of the Wiener-Hopf equation and determining solutions to it is beyond the scope of this *User's Guide*. Full descriptions of the theory appear in the adaptive filter references provided in the "Selected Bibliography" on page 4-48.

Using Adaptive Filter Objects

After you construct and adaptive filter object, how do you apply it to your data or system? Like quantizer objects, adaptive filter objects have a `filter` method that you use to apply the adaptfilt object to data. In the following sections, various examples of using LMS and RLS adaptive filters show you how `filter` works with the objects to apply them to data.

- "Examples of Adaptive Filters That Use LMS Algorithms" on page 4-17
- "Example of Adaptive Filter That Uses RLS Algorithm" on page 4-37

Examples of Adaptive Filters That Use LMS Algorithms

This section provides introductory examples using some of the least mean squares (LMS) adaptive filter functions in the toolbox.

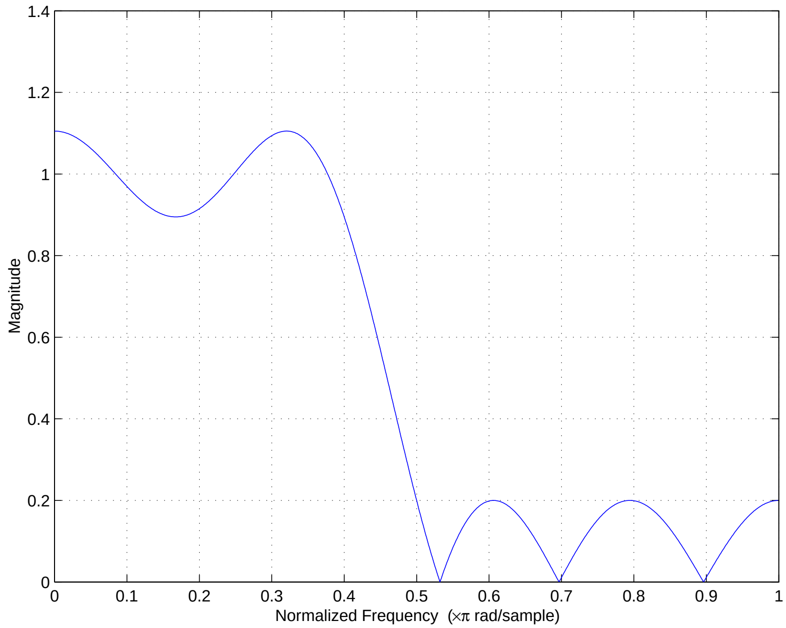
The Filter Design Toolbox provides many adaptive filter design functions that use the LMS algorithms to search for the optimal solution to the adaptive filter, including:

- `adaptfilt.lms`—implement the LMS algorithm to solve the Weiner-Hopf equation and find the filter coefficients for an adaptive filter.
- `adaptfilt.nlms`—implement the normalized variation of the LMS algorithm to solve the Weiner-Hopf equation and determine the filter coefficients of an adaptive filter.
- `adaptfilt.sd`—implement the sign-data variation of the LMS algorithm to solve the Weiner-Hopf equation and determine the filter coefficients of an adaptive filter. The correction to the filter weights at each iteration depends on the sign of the input $x(k)$.
- `adaptfilt.se`—implement the sign-error variation of the LMS algorithm to solve the Weiner-Hopf equation and determine the filter coefficients of an adaptive filter. The correction applied to the current filter weights for each successive iteration depends on the sign of the error, $e(k)$.
- `adaptfilt.ss`—implement the sign-sign variation of the LMS algorithm to solve the Weiner-Hopf equation and determine the filter coefficients of an adaptive filter. The correction applied to the current filter weights for each successive iteration depends on both the sign of $x(k)$ and the sign of $e(k)$.

To demonstrate the differences and similarities between the various LMS algorithms supplied in the toolbox, the LMS and NLMS adaptive filter examples use the same filter for the unknown system. In this case, the unknown filter is one of the filters used in the examples from "firgr Examples" on page 2-8—the constrained lowpass filter.

```
[b,err,res]=firgr(12,[0 0.4 0.5 1], [1 1 0 0], [1 0.2],...
{'w' 'c'});
```

From the figure you see that the filter is indeed lowpass and constrained to 0.2 ripple in the stopband. With this as the baseline, the adaptive LMS filter examples use the adaptive LMS algorithms and their initialization functions, to identify this filter in a system identification role. To review the general model for system ID mode, look at "System Identification" on page 4-7 for the layout.



For the sign variations of the LMS algorithm, the examples use noise cancellation as the demonstration application, as opposed to the system identification application used in the LMS examples.

adaptfilt.lms Example—System Identification

To use the adaptive filter functions in the toolbox you need to provide three things:

- An unknown system or process to adapt to. In this example, the filter designed by `firgr` is the unknown system.
- Appropriate input data to exercise the adaptation process. In terms of the generic LMS model, these are the desired signal *d(k)* and the input signal *x(k)*.

- The adaptive LMS function to use

Start by defining an input signal x.

```
x = 0.1*randn(1,250);
```

The input is broadband noise. For the unknown system filter, use firgr to create a twelfth-order lowpass filter:

```
[b,err,res] = firgr(12,[0 0.4 0.5 1], [1 1 0 0], [1 0.2],...
{'w' 'c'});
```

Although you do not need them here, include the err and res output arguments.

Now filter the signal through the unknown system to get the desired signal.

```
d = filter(b,1,x);
```

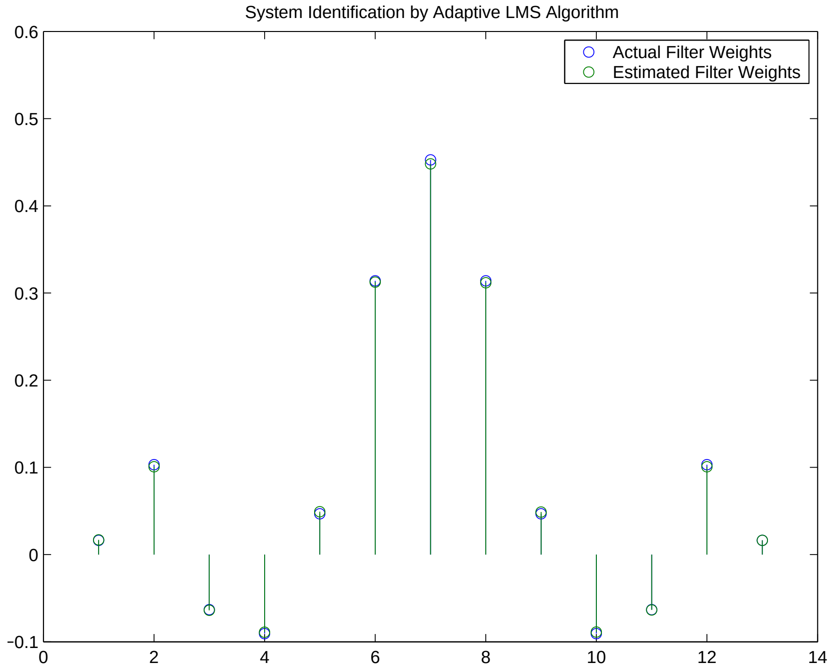
With the unknown filter designed and the desired signal in place you construct and apply the adaptive LMS filter object to identify the unknown.

Preparing the adaptive filter object requires that you provide starting values for estimates of the filter coefficients and the LMS step size. You could start with estimated coefficients of some set of nonzero values; this example uses zeros for the 12 initial filter weights. For the step size, 0.8 is a reasonable value—a good compromise between being large enough to converge well within the 250 iterations (250 input sample points) and small enough to create an accurate estimate of the unknown filter.

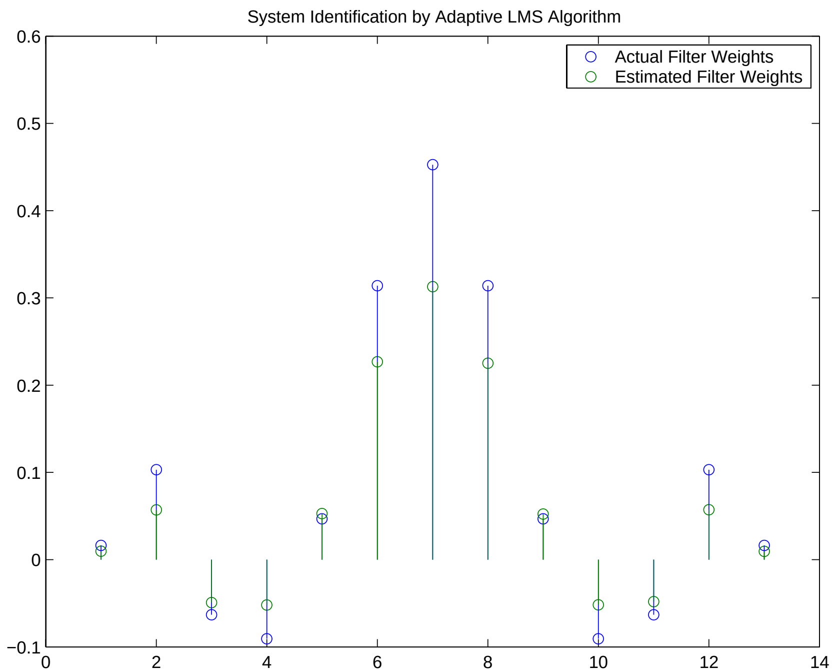
```
mu = 0.8;
ha = adaptfilt.lms(13,mu,w0)
```

Finally, using the adaptfilt object ha, desired signal, d, and the input to the filter, x, we run the adaptive filter to determine the unknown system and plot the results, comparing the actual coefficients from firgr to the coefficients found by adaptlms.

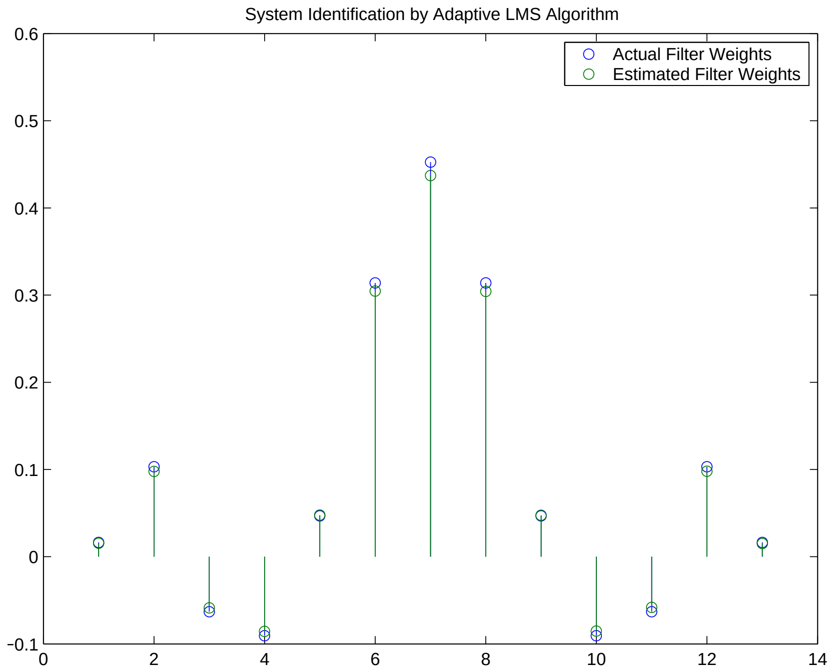
```
[y,e] = filter(ha,x,d);
stem([b.' ha.coefficients.'])
```



In the stem plot the actual and estimated filter weights are the same. As an experiment, try changing the step size to 0.2. Repeating the example with `mu = 0.2` results in the following stem plot. The estimated weights fail to approximate the actual weights closely.



Since this may be because we did not iterate over the LMS algorithm enough times, try using 1000 samples. With 1000 samples, the stem plot, shown in the next figure, looks much better, albeit at the expense of much more computation. Clearly you should take care to select the step size with both the computation required and the fidelity of the estimated filter in mind.



adaptfilt.nlms Example—System Identification

To improve the convergence performance of the LMS algorithm, the normalized variant uses an adaptive step size based on the signal power. As the input signal power changes, the algorithm calculates the input power and adjusts the step size to maintain an appropriate value. Thus the step size changes with time. As a result, the normalized algorithm converges more quickly with fewer samples in many cases. For input signals that change slowly over time, the normalized LMS can represent a more efficient LMS approach.

In the `adaptlms` example, we used `firgr` to create the filter that we would identify. So you can compare the results, we use the same filter, and replace `adaptlms` with `adaptnlms`, to use the normalized LMS algorithm variation. You should see better convergence with similar fidelity.

First, generate the input signal and the unknown filter.

```
x = 0.1*randn(1,500);
[b,err,res] = fircband(12,[0 0.4 0.5 1], [1 1 0 0], [1 0.2],...
{'w' 'c'});
d = filter(b,1,x);
```

Again `d` represents the desired signal *d(x)* as we defined it in Figure 4-1 and `b` contains the filter coefficients for our unknown filter.

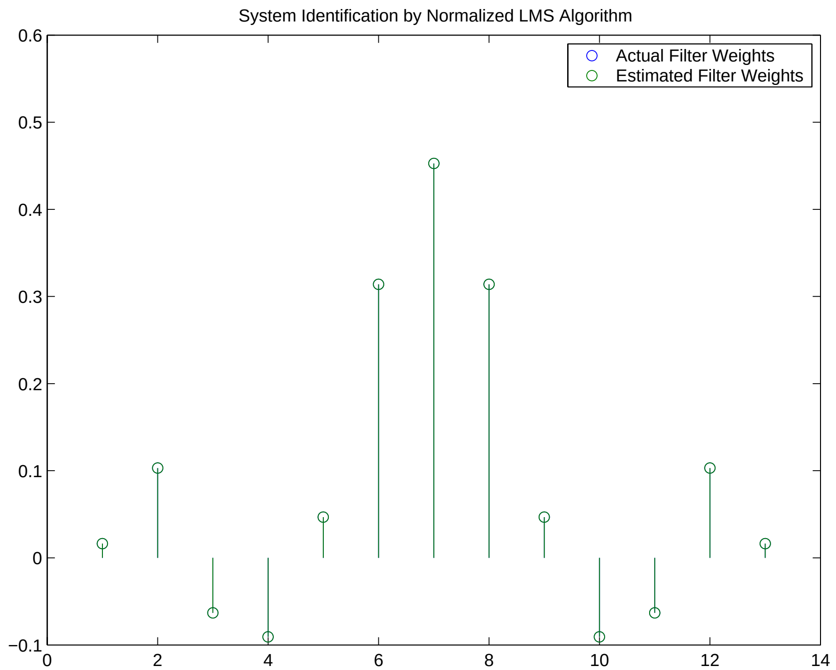
```
mu = 0.8;
ha = adaptfilt.nlms(13,mu);
```

We use the preceding code to initialize the normalized LMS algorithm. For more information about the optional input arguments, refer to `adaptfilt.nlms` in the reference section of this *User's Guide*.

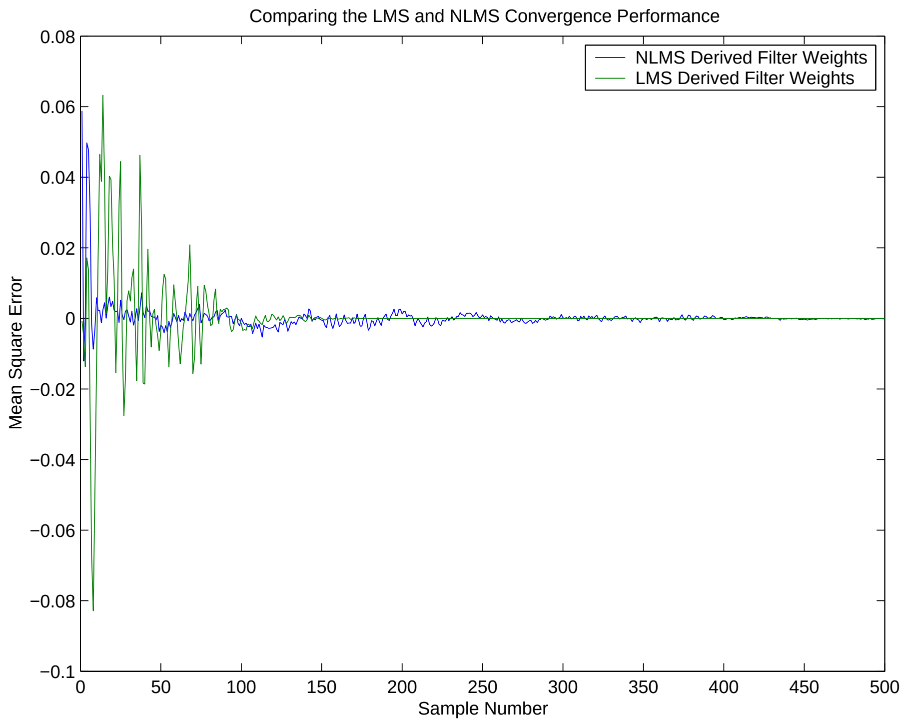
Running the system identification process is a matter of using `adaptfilt.nlms` with the desired signal, the input signal, and the initial filter coefficients and conditions specified in `s` as input arguments. Then plot the results to compare the adapted filter to the actual filter.

```
[y,e] = filter(ha,x,d);
stem([b.' ha.coefficients.'])
```

As shown in the following stem plot (a convenient way to compare the estimated and actual filter coefficients), the two are nearly identical.



If we compare the convergence performance of the regular LMS algorithm to the normalized LMS variant, you see the normalized version adapts in far fewer iterations to a result almost as good as the nonnormalized version.



adaptfilt.sd Example—Noise Cancellation

When the amount of computation required to derive an adaptive filter drives your development process, the sign-data variant of the LMS (SDLMS) algorithm may be a very good choice. Fortunately, the current state of digital signal processor (DSP) design has relaxed the need to minimize the operations count by making DSPs whose multiply and shift operations are as fast as add operations. Thus some of the impetus for the sign-data algorithm (and the sign-error and sign-sign variations) has been lost to DSP technology improvements.

In the standard and normalized variations of the LMS adaptive filter, coefficients for the adapting filter arise from the mean square error between the desired signal and the output signal from the unknown system. Using the sign-data algorithm changes the mean square error calculation by using the sign of the input data to change the filter coefficients. When the error is positive, the new coefficients are the previous coefficients plus the error multiplied by the step size μ. If the error is negative, the new coefficients are again the previous coefficients minus the error multiplied by μ—note the sign change. When the input is zero, the new coefficients are the same as the previous set.

In vector form, the sign-data LMS algorithm is

$$\mathbf{w}(k+1) = \mathbf{w}(k) + \mu e(k) sgn[\mathbf{x}(k)] \ , \ sgn[\mathbf{x}(k)] = \begin{cases} 1, & \mathbf{x}(k) > 0 \\ 0, & \mathbf{x}(k) = 0 \\ -1, & \mathbf{x}(k) < 0 \end{cases}$$

with vector **w** containing the weights applied to the filter coefficients and vector **x** containing the input data. *e(k)* (equal to desired signal - filtered signal) is the error at time *k* and is the quantity the SDLMS algorithm seeks to minimize. μ (`mu`) is the step size. As you specify `mu` smaller, the correction to the filter weights gets smaller for each sample and the SDLMS error falls more slowly. Larger `mu` changes the weights more for each step so the error falls more rapidly, but the resulting error does not approach the ideal solution as closely. To ensure good convergence rate and stability, select `mu` within the following practical bounds

$$0 < \mu < \frac{1}{N\{InputSignalPower\}}$$

where *N* is the number of samples in the signal. Also, define `mu` as a power of two for efficient computing.

Note How you set the initial conditions of the sign-data algorithm profoundly influences the effectiveness of the adaptation. Because the algorithm essentially quantizes the input signal, the algorithm can become unstable easily. A series of large input values, coupled with the quantization process may result in the error growing beyond all bounds. You restrain the tendency of the sign-data algorithm to get out of control by choosing a small step size (μ<< 1) and setting the initial conditions for the algorithm to nonzero positive and negative values.

In this noise cancellation example, `adaptfilt.sd` requires two input data sets:

- Data containing a signal corrupted by noise. In Figure 4-5, this is *d(k)*, the desired signal. The noise cancellation process removes the noise, leaving the signal.
- Data containing random noise (*x(k)* in Figure 4-5) that is correlated with the noise that corrupts the signal data. Without the correlation between the noise data, the adapting algorithm cannot remove the noise from the signal.

For the signal, use a sine wave. Note that `signal` is a column vector of 1000 elements.

```
signal = sin(2*pi*0.055*[0:1000-1]');
```

Now, add correlated white noise to `signal`. To ensure that the noise is correlated, pass the noise through a lowpass FIR filter, then add the filtered noise to the signal.

```
noise=randn(1,1000);
nfilt=fir1(11,0.4); % Eleventh order lowpass filter
fnoise=filter(nfilt,1,noise); % Correlated noise data
d=signal.'+fnoise;
```

`fnoise` is the correlated noise and `d` is now the desired input to the sign-data algorithm.

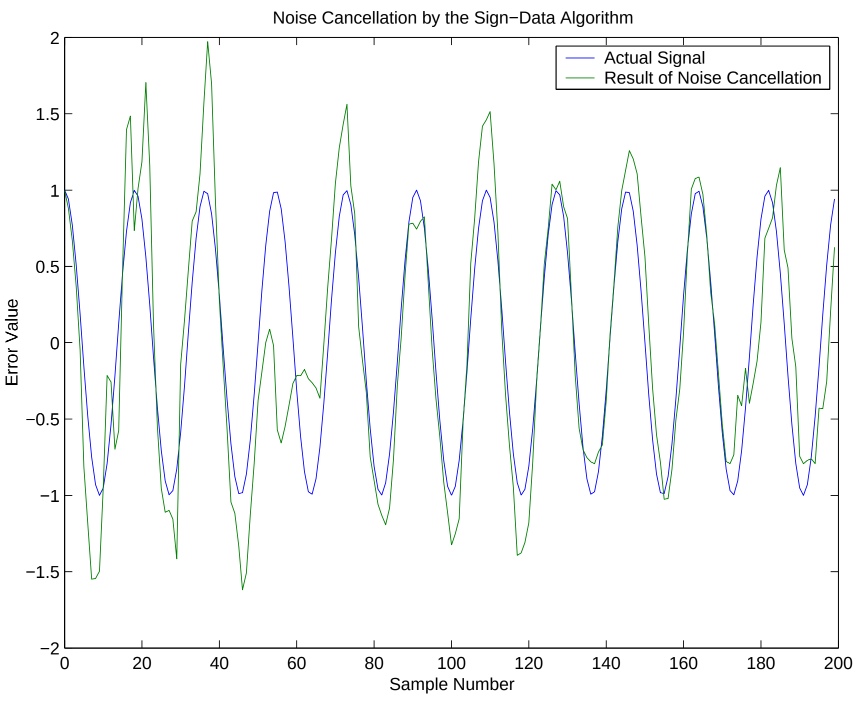
To prepare the adaptfilt object for processing, set the input conditions `coeffs` and `mu` for the object. As noted earlier in this section, the values you set for `coeffs` and `mu` determine whether the adaptive filter can remove the noise from the signal path. In "adaptfilt.lms Example—System Identification" on page 4-18, you constructed a default filter that sets the filter coefficients to zeros. Except in rare cases, that approach does not work for the sign-data algorithm. The closer you set your initial filter coefficients to the expected values, the more likely it is that the algorithm remains well behaved and converges to a filter solution that removes the noise effectively. For this example, we start with the coefficients in the filter we used to filter the noise (`nfilt`), and modify them slightly so the algorithm has to adapt.

```
coeffs = nfilt.' -0.01; % Set the filter initial conditions.
mu = 0.05;          % Set the step size for algorithm updating.
```

With the required input arguments for `adaptfilt.sd` prepared, construct the adaptfilt object, run the adaptation, and view the results.

```
ha = adaptfilt.sd(13,mu)
set(ha,'coefficients',coeffs);
[y,e] = filter(ha,noise,d);
plot(0:199,signal(1:200),0:199,e(1:200));
```

When `adaptfilt.sd` runs, it uses far fewer multiply operations than either of the LMS algorithms. Also, performing the sign-data adaptation requires only bit shifting multiplys when the step size is a power of two. Although the performance of the sign-data algorithm as shown in the next figure is quite good, the sign-data algorithm is much less stable than the standard LMS variations. In this noise cancellation example, the signal after processing is a very good match to the input signal, but the algorithm could very easily grow without bound rather than achieve good performance. Changing `coeffs`, `mu`, or even the lowpass filter you used to create the correlated noise can cause noise cancellation to fail and the algorithm to become useless.



adaptfilt.se Example—Noise Cancellation

In some cases, the sign-error variant of the LMS algorithm may be a very good choice for an adaptive filter application. In the standard and normalized variations of the LMS adaptive filter, the coefficients for the adapting filter arise from calculating the mean square error between the desired signal and the output signal from the unknown system, and applying the result to the current filter coefficients. Using the sign-error algorithm replaces the mean square error calculation by using the sign of the error to modify the filter coefficients. When the error is positive, the new coefficients are the previous coefficients plus the error multiplied by the step size μ. If the error is negative, the new coefficients are again the previous coefficients minus the error multiplied by μ—note the sign change. When the input is zero, the new coefficients are the same as the previous set.

In vector form, the sign-error LMS algorithm is

$$\mathbf{w}(k+1) = \mathbf{w}(k) + \mu \operatorname{sgn}[e(k)][\mathbf{x}(k)] \ , \ sgn[e(k)] = \begin{cases} 1, & e(k) > 0 \\ 0, & e(k) = 0 \\ -1, & e(k) < 0 \end{cases}$$

with vector **w** containing the weights applied to the filter coefficients and vector **x** containing the input data. *e(k)* (equal to desired signal - filtered signal) is the error at time *k* and is the quantity the SELMS algorithm seeks to minimize. μ (`mu`) is the step size. As you specify `mu` smaller, the correction to the filter weights gets smaller for each sample and the SELMS error falls more slowly. Larger `mu` changes the weights more for each step so the error falls more rapidly, but the resulting error does not approach the ideal solution as closely. To ensure good convergence rate and stability, select `mu` within the following practical bounds

$$0 < \mu < \frac{1}{N\{InputSignalPower\}}$$

where *N* is the number of samples in the signal. Also, define `mu` as a power of two for efficient computation.

Note How you set the initial conditions of the sign-data algorithm profoundly influences the effectiveness of the adaptation. Because the algorithm essentially quantizes the error signal, the algorithm can become unstable easily. A series of large error values, coupled with the quantization process may result in the error growing beyond all bounds. You restrain the tendency of the sign-error algorithm to get out of control by choosing a small step size (μ<< 1) and setting the initial conditions for the algorithm to nonzero positive and negative values.

In this noise cancellation example, `adaptfilt.se` requires two input data sets:

- Data containing a signal corrupted by noise. In Figure 4-5, this is *d(k)*, the desired signal. The noise cancellation process removes the noise, leaving the signal.
- Data containing random noise (*x(k)* in Figure 4-5) that is correlated with the noise that corrupts the signal data. Without the correlation between the noise data, the adapting algorithm cannot remove the noise from the signal.

For the signal, use a sine wave. Note that `signal` is a column vector of 1000 elements.

```
signal = sin(2*pi*0.055*[0:1000-1]');
```

Now, add correlated white noise to `signal`. To ensure that the noise is correlated, pass the noise through a lowpass FIR filter, then add the filtered noise to the signal.

```
noise=randn(1,1000);
nfilt=fir1(11,0.4); % Eleventh order lowpass filter.
fnoise=filter(nfilt,1,noise); % Correlated noise data.
d=signal.'+fnoise;
```

`fnoise` is the correlated noise and `d` is now the desired input to the sign-data algorithm.

To prepare the adaptfilt object for processing, set the input conditions `coeffs` and `mu` for the object. As noted earlier in this section, the values you set for `coeffs` and `mu` determine whether the adaptive filter can remove the noise from the signal path. In "adaptfilt.lms Example—System Identification" on page 4-18, you constructed a default filter that sets the filter coefficients to zeros. Except in rare cases, that approach does not work for the sign-error algorithm. The closer you set your initial filter coefficients to the expected values, the more likely it is that the algorithm remains well behaved and converges to a filter solution that removes the noise effectively. For this example, we start with the coefficients in the filter we used to filter the noise (`nfilt`), and modify them slightly so the algorithm has to adapt.

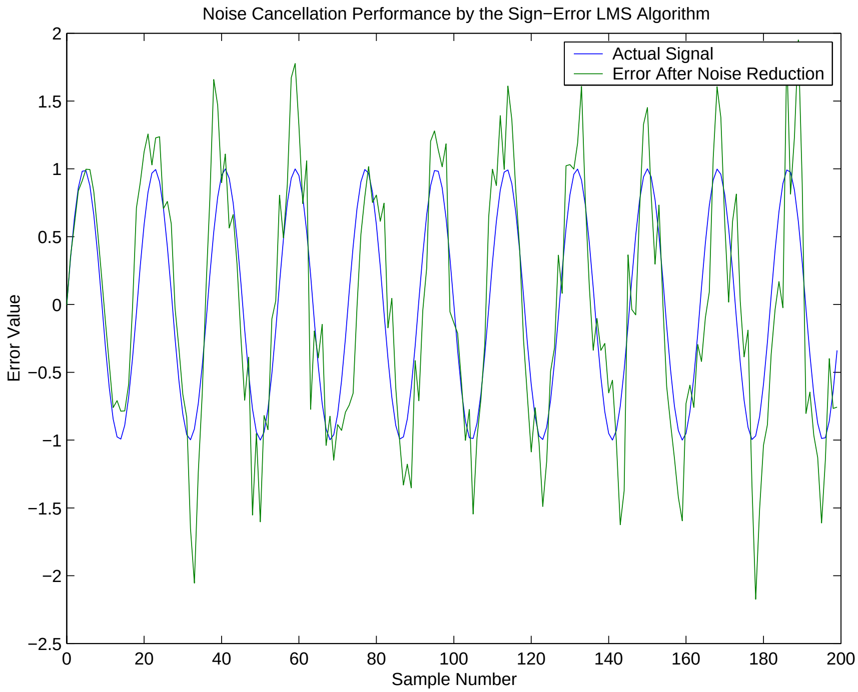
```
coeffs = nfilt.' -0.01; % Set the filter initial conditions.
mu = 0.05;              % Set the step size for algorithm update.
```

With the required input arguments for `adaptfilt.se` prepared, run the adaptation and view the results.

```
ha = adaptfilt.sd(12,mu)
set(ha,'coefficients',coeffs);
set(ha,'persistentmemory',true); % Prevent filter reset.
[y,e] = filter(ha,noise,d);
plot(0:199,signal(1:200),0:199,e(1:200));
```

Notice that you have to set the property `PersistentMemory` to `true` when you manually change the settings of object `ha`. If `PersistentMemory` is left to `false`, the default, when you try to apply `ha` with the method `filter`, the filtering process starts by resetting the object properties to their initial conditions at construction. To preserve the customized coefficients in this example, we set `PersistentMemory` to `true` so the coefficients do not get reset automatically back to zero.

When `adaptfilt.se` runs, it uses far fewer multiply operations than either of the LMS algorithms. Also, performing the sign-error adaptation requires only bit shifting multiplys when the step size is a power of two. Although the performance of the sign-data algorithm as shown in the next figure is quite good, the sign-data algorithm is much less stable than the standard LMS variations. In this noise cancellation example, the signal after processing is a very good match to the input signal, but the algorithm could very easily become unstable rather than achieve good performance. Changing `coeffs`, `mu`, or even the lowpass filter you used to create the correlated noise can cause noise cancellation to fail and the algorithm to become useless.



adaptfilt.ss Example—Noise Cancellation

One more example of a variation of the LMS algorithm in the toolbox is the sign-sign variant (SSLMS). The rationale for this version matches those for the sign-data and sign-error algorithms presented in preceding sections. For more details, refer to "adaptfilt.sd Example—Noise Cancellation" on page 4-25.

The sign-sign algorithm (SSLMS) replaces the mean square error calculation to using the sign of the input data to change the filter coefficients. When the error is positive, the new coefficients are the previous coefficients plus the error multiplied by the step size μ. If the error is negative, the new coefficients are again the previous coefficients minus the error multiplied by μ—note the sign change. When the input is zero, the new coefficients are the same as the previous set. In essence, the algorithm quantizes both the error and the input by applying the sign operator to them.

In vector form, the sign-sign LMS algorithm is

$$\mathbf{w}(k+1) = \mathbf{w}(k) + \mu \operatorname{sgn}[e(k)]\operatorname{sgn}[\mathbf{x}(k)] \quad ,\operatorname{sgn}[z(k)] = \begin{cases} 1, & z(k) > 0 \\ 0, & z(k) = 0 \\ -1, & z(k) < 0 \end{cases}$$

where

$$z(k) = [e(k)]\operatorname{sgn}[\mathbf{x}(k)]$$

Vector **w** contains the weights applied to the filter coefficients and vector **x** contains the input data. *e(k)* (= desired signal - filtered signal) is the error at time *k* and is the quantity the SSLMS algorithm seeks to minimize. μ (`mu`) is the step size. As you specify `mu` smaller, the correction to the filter weights gets smaller for each sample and the SSLMS error falls more slowly. Larger `mu` changes the weights more for each step so the error falls more rapidly, but the resulting error does not approach the ideal solution as closely. To ensure good convergence rate and stability, select `mu` within the following practical bounds

$$0 < \mu < \frac{1}{N\{InputSignalPower\}}$$

where *N* is the number of samples in the signal. Also, define `mu` as a power of two for efficient computation.

Note How you set the initial conditions of the sign-sign algorithm profoundly influences the effectiveness of the adaptation. Because the algorithm essentially quantizes the input signal and the error signal, the algorithm can become unstable easily. A series of large error values, coupled with the quantization process may result in the error growing beyond all bounds. You restrain the tendency of the sign-sign algorithm to get out of control by choosing a small step size (μ<< 1) and setting the initial conditions for the algorithm to nonzero positive and negative values.

In this noise cancellation example, `adaptfilt.ss` requires two input data sets:

- Data containing a signal corrupted by noise. In Figure 4-5, this is *d(k)*, the desired signal. The noise cancellation process removes the noise, leaving the cleaned signal as the content of the error signal.
- Data containing random noise (*x(k)* in Figure 4-5) that is correlated with the noise that corrupts the signal data, called. Without the correlation between the noise data, the adapting algorithm cannot remove the noise from the signal.

For the signal, use a sine wave. Note that `signal` is a column vector of 1000 elements.

```
signal = sin(2*pi*0.055*[0:1000-1]');
```

Now, add correlated white noise to `signal`. To ensure that the noise is correlated, pass the noise through a lowpass FIR filter, then add the filtered noise to the signal.

```
noise=randn(1,1000);
nfilt=fir1(11,0.4); % Eleventh order lowpass filter
fnoise=filter(nfilt,1,noise); % Correlated noise data
d=signal.'+fnoise;
```

`fnoise` is the correlated noise and `d` is now the desired input to the sign-data algorithm.

To prepare the adaptfilt object for processing, set the input conditions `coeffs` and `mu` for the object. As noted earlier in this section, the values you set for `coeffs` and `mu` determine whether the adaptive filter can remove the noise from the signal path. In "adaptfilt.lms Example—System Identification" on page 4-18, you constructed a default filter that sets the filter coefficients to zeros. Except in rare cases, that approach does not work for the sign-sign algorithm. The closer you set your initial filter coefficients to the expected values, the more likely it is that the algorithm remains well behaved and converges to a filter solution that removes the noise effectively. For this example, we start with the coefficients in the filter we used to filter the noise (`nfilt`), and modify them slightly so the algorithm has to adapt.

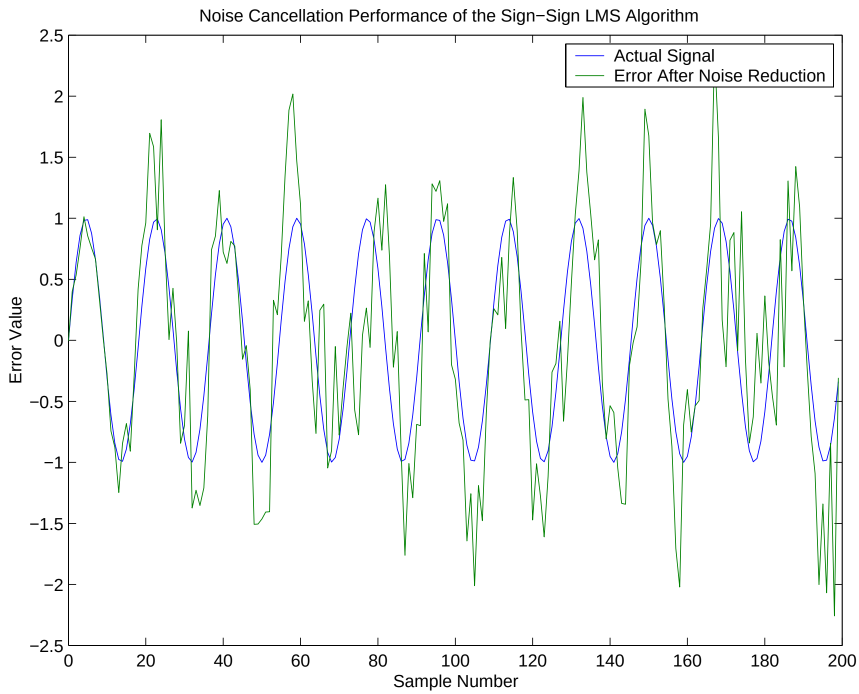
```
coeffs = nfilt.' -0.01; % Set the filter initial conditions.
mu = 0.05;          % Set the step size for algorithm updating.
```

With the required input arguments for `adaptfilt.ss` prepared, run the adaptation and view the results.

```
ha = adaptfilt.ss(12,mu)
set(ha,'coefficients',coeffs);
set(ha,'persistentmemory',true); % Prevent filter reset.
[y,e] = filter(ha,noise,d);
plot(0:199,signal(1:200),0:199,e(1:200));
```

Notice that you have to set the property `PersistentMemory` to `true` when you manually change the settings of object `ha`. If `PersistentMemory` is left to `false`, the default, when you try to apply `ha` with the method `filter`, the filtering process starts by resetting the object properties to their initial conditions at construction. To preserve the customized coefficients in this example, we set `PersistentMemory` to `true` so the coefficients do not get reset automatically back to zero.

When `adaptfilt.ss` runs, it uses far fewer multiply operations than either of the LMS algorithms. Also, performing the sign-sign adaptation requires only bit shifting multiplys when the step size is a power of two. Although the performance of the sign-sign algorithm as shown in the next figure is quite good, the sign-sign algorithm is much less stable than the standard LMS variations. In this noise cancellation example, the signal after processing is a very good match to the input signal, but the algorithm could very easily become unstable rather than achieve good performance. Changing `coeffs`, `mu`, or even the lowpass filter you used to create the correlated noise can cause noise cancellation to fail and the algorithm to become useless.



As an aside, the sign-sign LMS algorithm is part of the international CCITT standard for 32 Kb/s ADPCM telephony.

Example of Adaptive Filter That Uses RLS Algorithm

This section provides an introductory example that uses the RLS adaptive filter function `adaptfilt.rls`.

If LMS algorithms represent the simplest and most easily applied adaptive algorithms, the recursive least squares (RLS) algorithms represents increased complexity, computational cost, and fidelity. In performance, RLS approaches the Kalman filter in adaptive filtering applications, at somewhat reduced required throughput in the signal processor. Compared to the LMS algorithm, the RLS approach offers faster convergence and smaller error with respect to the unknown system, at the expense of requiring more computations.

In contrast to the least mean squares algorithm, from which it can be derived, the RLS adaptive algorithm minimizes the total squared error between the desired signal and the output from the unknown system. Referring to Figure 4-2, you see the signal flow graph (or model) for the RLS adaptive filter system. Note that the signal paths and identifications are the same whether the filter uses RLS or LMS. The difference lies in the adapting portion. Within limits, you can use any of the adaptive filter algorithms to solve an adaptive filter problem by replacing the adaptive portion of the application with a new algorithm. Examples of the sign variants of the LMS algorithms demonstrated this feature to demonstrate the differences between the sign-data, sign-error, and sign-sign variations of the LMS algorithm.

One interesting input option that applies to RLS algorithms is not present in the LMS processes—a forgetting factor, λ, that determines how the algorithm treats past data input to the algorithm. When the LMS algorithm looks at the error to minimize, it considers only the current error value. In the RLS method, the error considered is the total error from the beginning to the current data point. Said another way, the RLS algorithm has infinite memory—all error data is given the same consideration in the total error. In cases where the error value might come from a spurious input data point or points, the forgetting factor lets the RLS algorithm reduce the value of older error data by multiplying the old data by the forgetting factor. Since $0 \leq \lambda < 1$, applying the factor is equivalent to weighting the older error. When $\lambda = 1$, all previous error is considered of equal weight in the total error. As λ approaches zero, the past errors play a smaller role in the total. For example, when $\lambda = 0.9$, the RLS algorithm multiplies an error value from 50 samples in the past by an attenuation factor of $0.9^{50} = 5.15 \times 10^{-3}$, considerably deemphasizing the influence of the past error on the current total error.

adaptfilt.rls Example—Inverse System Identification

Rather than use a system identification application, or a noise cancellation model, this example use the inverse system identification model shown in Figure 4-4. Cascading the adaptive filter with the unknown filter causes the adaptive filter to converge to a solution that is the inverse of the unknown system. If the transfer function of the unknown is H(z) and the adaptive filter transfer function is G(z), the error measured between the desired signal and the signal from the cascaded system reaches its minimum when the product of H(z) and G(z) is 1, G(z)*H(z) = 1. For this relation to be true, G(z) must equal -H(z), the inverse of the transfer function of the unknown system.

To demonstrate that this is true, create a signal to input to the cascaded filter pair.

```
x = randn(1,3000);
```

In the cascaded filters case, like this one, the unknown filter results in a delay in the signal arriving at the summation point after both filters. To prevent the adaptive filter from trying to adapt to a signal it has not yet seen (equivalent to predicting the future), delay the desired signal by 32 samples, the order of the unknown system. Generally, you do not know the order of the system you are trying to identify. In that case, delay the desired signal by about the number of samples that is equal to half the order of the adaptive filter. Delaying the input requires prepending 12 zero-values samples to `x`.

```
delay = zeros(1,12);
d = [delay x(1:2988)]; % Concatenate the delay and the signal.
```

You have to keep the desired signal vector `d` the same length as `x`, hence adjust the signal element count to allow for the delay samples. Although not generally true, for this example you know the order of the unknown filter, so you add a delay equal to the order of the unknown filter.

For the unknown system, use a lowpass, 12th-order FIR filter.

```
ufilt = fir1(12,0.55,'low');
```

Filtering `x` provides the input data signal for the adaptive algorithm function.

```
xdata = filter(ufilt,1,x);
```

To set the input argument values for the adaptfilt.rls object, use the constructor `adaptfilt.rls`, providing the needed arguments `l`, `lambda`, and invcov. For more information about the input conditions to prepare the RLS algorithm object, refer to adaptfilt.rls in the reference section of this *User's Guide*.

```
p0 = 2*eye(13);
lambda = 0.99;
ha = adaptfilt.rls(13,lambda,p0);
```

Most of the process to this point is the same as the preceding examples. However, since this example is looking to develop an inverse solution, you need to be careful about which signal carries the data and which is the desired signal. Earlier examples of adaptive filters use the filtered noise as the desired signal. In this case, the filtered noise (xdata) carries the unknown system information. With Gaussian distribution and variance of 1, the unfiltered noise d is the desired signal. The code to run this adaptive filter example is

```
[y,e] = filter(ha,xdata,d);
```

where y returns the coefficients of the adapted filter and e contains the error signal as the filter adapts to find the inverse of the unknown system. You can review the returned elements of the adapted filter in the properties of ha.

Figure 4-7 presents the results of the adaptation. In the figure, we present the magnitude response curves for the unknown and adapted filters. As a reminder, the unknown filter was a lowpass filter with cutoff at 0.55, on the normalized frequency scale from 0 to 1.

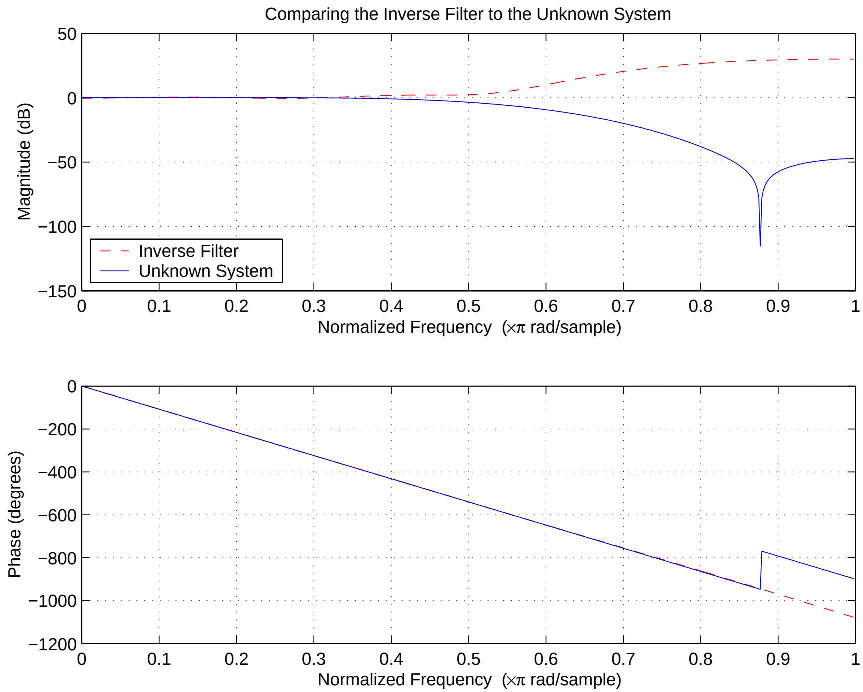


Figure 4-7: Comparing the Results of the RLS Inverse System Identification

Viewed alone in Figure 4-8, the inverse system looks like a fair compensator for the unknown lowpass filter—a high pass filter with linear phase.

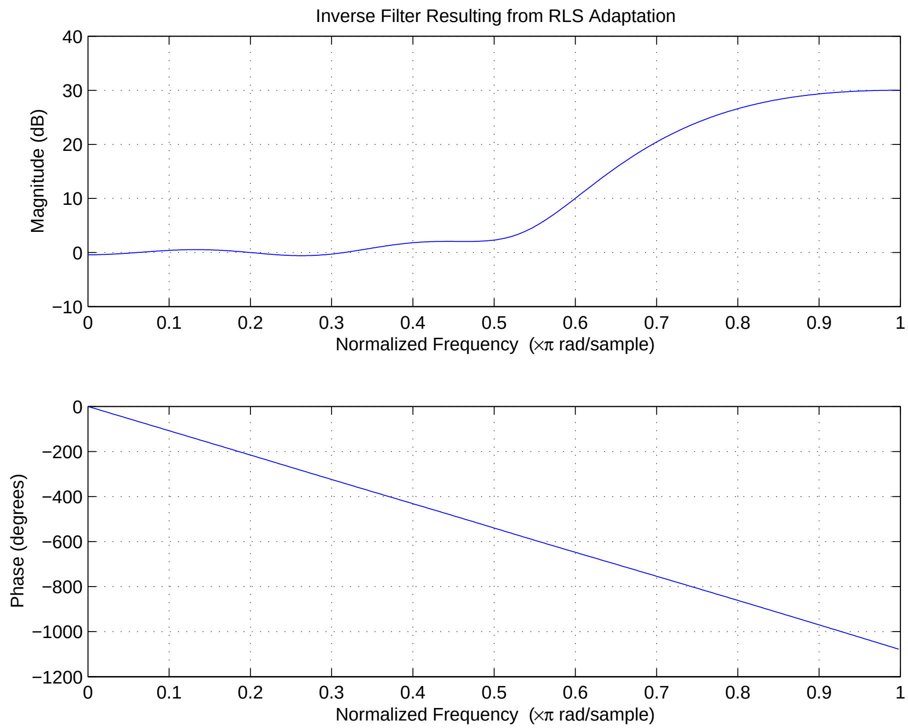


Figure 4-8: After Adapting, the RLS Algorithm Produces a Highpass Filter

Adaptive Filter Properties Reference

Like dfilts, adaptfilts are objects and have properties that govern their behavior and store some of the results of filtering operations. This section lists, in alphabetical order, the name of every property associated with adaptfilt objects. Note that not all adaptfilt objects have all of these properties. To view the properties of a particular adaptive filter, such as an adaptfilt.bap filter, use get with the object handle, like this:

```
ha = adaptfilt.bap(32,0.5,4,1.0);
get(ha)
     PersistentMemory: false
    NumSamplesProcessed: 0
              Algorithm: 'Block Affine Projection FIR Adaptive Filter'
           FilterLength: 32
           Coefficients: [1x32 double]
                 States: [35x1 double]
               StepSize: 0.5000
        ProjectionOrder: 4
              OffsetCov: [4x4 double]
```

get shows you the properties for ha and the values for the properties.

Adaptive Filter Properties

Algorithm

Reports the algorithm the object uses for adaptation. When you construct your adaptive filter object, this property is set automatically by the constructor, such as adaptfilt.nlms creating an adaptive filter that uses the normalized LMS algorithm as shown in the code example here. You cannot change the value—it is read only.

```
ha=adaptfilt.nlms(32,1,1,50)

ha =

              Algorithm: 'Direct-Form FIR Normalized LMS Adaptive Filter'
           FilterLength: 32
           Coefficients: [1x32 double]
                 States: [31x1 double]
               StepSize: 1
                Leakage: 1
                 Offset: 50
     PersistentMemory: false
```

```
    NumSamplesProcessed: 0
ha.algorithm='lms' % Try to change the algorithm.
??? Changing the 'Algorithm' property of adaptfilt.baseclass is
not allowed.
```

AvgFactor

Averaging factor used to compute the exponentially-windowed estimates of the powers in the transformed signal bins for the coefficient updates. `AvgFactor` should lie between zero and one. For default filter objects, `AvgFactor` equals (1 - `step`). `lambda` is the input argument that represent `AvgFactor`

BkwdPredErrorPower

BkwdPrediction

Blocklength

Block length for the coefficient updates. This must be a positive integer such that (`l/blocklen`) is also an integer. For faster execution, `blocklen` should be a power of two. `blocklen` defaults to two.

Coefficients

Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.

ConversionFactor

Conversion factor defaults to the matrix [1 -1] that specifies soft-constrained initialization. This is the `gamma` input argument for some of the fast transversal algorithms.

Delay

Update delay given in time samples. This scalar should be a positive integer—negative delays do not work. `delay` defaults to 1 for most algorithms.

DesiredSignalStates

Desired signal states of the adaptive filter. `dstates` defaults to a zero vector with length equal to (`blocklen` - 1) or (`swblocklen` - 1) depending on the algorithm.

EpsilonStates

Vector of the epsilon values of the adaptive filter. `EpsilonStates` defaults to a vector of zeros with (`projectord` - 1) elements.

ErrorStates

Vector of the adaptive filter error states. `ErrorStates` defaults to a zero vector with length equal to (`projectord` - 1).

FFTCoefficients

Stores the discrete Fourier transform of the filter coefficients in `coeffs`.

FFTStates

Stores the states of the FFT of the filter coefficients during adaptation.

FilteredInputStates

Vector of filtered input states with length equal to `l` - 1.

FilterLength

Contains the length of the filter. Note that this is not the filter order. Filter length is 1 greater than filter order. Thus a filter with length equal to 10 has filter order equal to 9.

ForgettingFactor

Determines how the RLS adaptive filter uses past data in each iteration. You use the forgetting factor to specify whether old data carries the same weight in the algorithm as more recent data.

This is a scalar and should lie in the range (0, 1]. It defaults to 1. Setting `forgetting factor = 1` denotes infinite memory while adapting to find the new filter. Note that this is the `lambda` input argument.

FwdPredErrorPower

Returns the minimum mean-squared prediction error in the forward direction. Refer to [12] in the bibliography for details about linear prediction.

FwdPrediction

Contains the predicted values for samples during adaptation. Compare these to the actual samples to get the error and power.

InitFactor

Soft-constrained initialization factor. This scalar should be positive and sufficiently large to prevent an excessive number of Kalman gain rescues. Called `delta` as an input argument, this defaults to one.

InvCov

Upper-triangular Cholesky (square root) factor of the input covariance matrix. Initialize this matrix with a positive definite upper triangular matrix. Dimensions are `l`-by-`l`, where `l` is the filter length.

KalmanGain

Empty when you construct the object, this gets populated after you run the filter.

KalmanGainStates

Contains the states of the Kalman gain updates during adaptation.

Leakage

Contains the setting for leakage in the adaptive filter algorithm. Using a leakage factor that is not 1 forces the weights to adapt even when they have found the minimum error solution. Forcing the adaptation can improve the numerical performance of the LMS algorithm.

NumSamplesProcessed

Reports the number of input samples processed by your adaptfilt object when you filter a data set. When you set the `PersistentMemory` property to `false`, `NumSamplesProcessed` reports the number of processed input samples for your most recent filtering operation with the object.

With `PersistentMemory` set to `true`, `NumSamplesProcessed` accumulates the total number of samples processed for all preceding filtering operations.

OffsetCov

Contains the offset covariance matrix.

Offset

Specifies an optional offset for the denominator of the step size normalization term. You must specify offset to be a scalar greater than or equal to zero. Nonzero offsets can help avoid a divide-by-near-zero condition that causes errors.

Use this to avoid dividing by zero or by very small numbers when input signal amplitude becomes very small, or dividing by very small numbers when any of the FFT input signal powers become very small. `offset` defaults to one.

Power

A vector of 2*`l` elements, each initialized with the value `delta` from the input arguments. As you filter data, `Power` gets updated by the filter process.

ProjectionOrder

Projection order of the affine projection algorithm. `projectord` defines the size of the input signal covariance matrix and defaults to two.

ReflectionCoeffs

Stores the reflection coefficients for adaptive filters, where applicable.

ReflectionCoeffsStep

Stores the step size used to determine the reflection coefficients during adaptation.

PersistentMemory

Determine whether the filter states and coefficients get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter.

PersistentMemory returns to zero any property value that the filter changes during processing. Property values that the filter does not change are not affected. Defaults to false.

SecondaryPathCoeffs

A vector that contains the coefficient values of your secondary path from the output actuator to the error sensor.

SecondaryPathEstimate

An estimate of the secondary path filter model.

SecondaryPathStates

The states of the secondary path filter, the unknown system.

SqrtCov

Upper-triangular Cholesky (square root) factor of the input covariance matrix. Initialize this matrix with a positive definite upper triangular matrix.

SqrtInvCov

Square root of the inverse of the sliding window input signal covariance matrix. This square matrix should be full-ranked.

States

Vector of the adaptive filter states. states defaults to a vector of zeros whose length depends on the chosen algorithm. Usually the length is a function of the filter length l and another input argument to the filter object, such as projectord.

StepSize

Reports the size of the step taken between iterations of the adaptive filter process. Each adaptfilt object has a default value that best meets the needs of the algorithm.

SwBlockLength

Block length of the sliding window. This integer must be at least as large as the filter length. swblocklen defaults to 16.

Selected Bibliography

[1] Hayes, Monson H., *Statistical Digital Signal Processing and Modeling*, John Wiley & Sons, 1996, 493–552.

[2] Haykin, Simon, *Adaptive Filter Theory*, Prentice-Hall, Inc., 1996

5

Designing Multirate Filters

Introducing Multirate Filters

Over the last few years, developments in multirate filter design and implementation have brought rapid growth in applying multirate filtering to signals in digital signal processing. Improved processors and development tools allow system designers to use multirate filters in a broad range of application areas, such as:

- POTS audio encryption—encrypts voice sent over plain old telephone systems (POTS).
- Digital audio—sound handled in digital rather than analog form. Enncompasses various signal compression schemes, analog-to-digital conversion techniques, and the opposite conversions, signal reproduction, and audio improvements.
- Subband speech and image coding—uses the techniques of separating a signal or image into subbands that each containing only a portion of the original signal. Then processing the subbands through filters before reconstructing the original signal from the processed subbands.

  Polyphase filters—filters that separate an input signal into constituent bands that are easier to process, and can then be either recombined or used after processing—represent one way to accomplish signal separation. Filter performance depends on the phase differences between the input signals.
- Transmultiplexer design—uses filters to convert time division multiplexing (TDM) signals to frequency division multiplexing (FDM) format, and the reverse. FDM combines numerous signals for transmission on a single communications line or channel. Each signal is assigned a different frequency (subchannel) within the main channel. TDM puts multiple data streams in a single signal by separating the signal into many segments, each having a very short duration. Based on the timing of the signals, each individual data stream is reassembled at the receiving end.

These represent a few of the growing number of areas in which systems designers use multirate filters.

Listed below are the examples in this chapter that introduce multirate filters. Each example includes a tutorial that uses toolbox features to demonstrate how you work with multirate filters:

- “Getting Started—Designing Multirate Filters” on page 5-4

- “Audio Example—Audio Sample Rate Conversion” on page 5-47
- “CIC Filter Example—Using CIC Decimation Filters” on page 5-24

- “Audio Example—Audio Sample Rate Conversion” on page 5-47

Getting Started—Designing Multirate Filters

This section demonstrates how to use the multirate filter (`mfilt`) objects available in the toolbox. By following these procedures you get introduced to multirate filter development. This tutorial covers the following tasks:

- "Creating Multirate Filters" on page 5-4
- "Getting and Setting Filter Coefficients" on page 5-6
- "Analyzing Multirate and Multistage Filters" on page 5-8
- "Specifying Initial Conditions to the Filter" on page 5-10
- "Streaming Data to the Filter" on page 5-12
- "Filtering Multichannel Signals" on page 5-12
- "Generating Simulink Blocks" on page 5-14
- "Getting Help About Multirate Filters" on page 5-15

Creating Multirate Filters

To develop a multirate filter (`mfilt`) object, you select the filter structure to be used by selecting the constructor function, such as `mfilt.firdecim` or `mfilt.firinterp`. Most multirate filter constructors take the coefficients of the filter as an optional final right-hand input argument. If you do not specify the coefficients, the toolbox functions design a default filter according to the interpolation or decimation factor(s) you provide as input for L or M in the calling syntax, or both in the case of fractional rate changer filters.

Here is an example that creates an interpolating filter with order of three interpolation and a decimating filter that decimates by two.

```
l = 3; % Interpolation factor
m = 2; % Decimation factor
hm1 = mfilt.firinterp(l);
hm2 = mfilt.firtdecim(m);
```

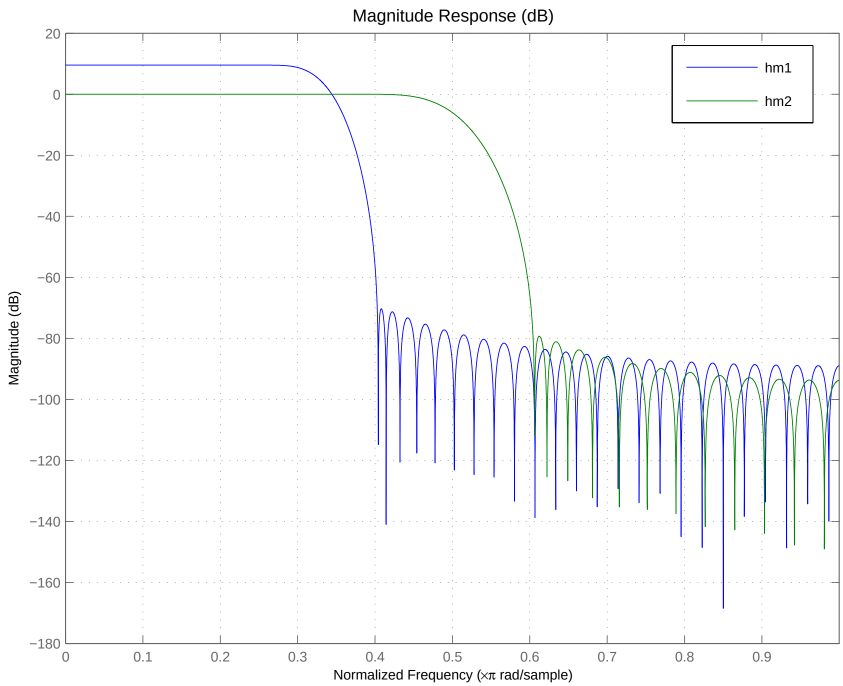
Both filter constructors return direct form FIR polyphase Nyquist filters by default.

```
hm1

hm1 =

          FilterStructure: 'Direct-Form FIR Polyphase Interpolator'
                Numerator: [1x72 double]
```

```
      InterpolationFactor: 3
       PersistentMemory: false
                 States: [23x1 double]
      NumSamplesProcessed: 0

hm2

hm2 =

          FilterStructure: 'Direct-Form Transposed FIR Polyphase Decimator'
                Numerator: [1x48 double]
         DecimationFactor: 2
       PersistentMemory: false
                 States: [23x1 double]
      NumSamplesProcessed: 0
```

Filter `hm1` is a direct-form FIR polyphase interpolator filter with the cutoff frequency of π/`l` and gain of `l`. `hm2` is a direct-form transposed FIR polyphase decimator with a cutoff frequency of π/`m` and a gain of 1.

For confirmation, here is the frequency response displayed by the Filter Visualization Tool (FVTool).



`hm1` and `hm2` are filters and `mfilt` objects. As objects, they work with a range of functions (methods) such as `filter`, `freqz`, and `tf`, or `display`.

Getting and Setting Filter Coefficients

To access and manipulate the coefficients of a filter as a regular MATLAB vector, you use the common object functions `set` and `get` or dot notation. You can always get the coefficients from the `mfilt` object (filter). To modify the coefficients of an existing `mfilt` object, you set new ones. Direct-form FIR structures like those of `hm1` and `hm2` have numerator coefficients only—also known as the filter weights.

Here are the filter coefficients for hm2.

```
b = get(hm2,'numerator') % Could use hm2.numerator as well. Assign the coefficients
                        % to vector b.
b =

  Columns 1 through 8

       0   -0.0001         0    0.0004         0   -0.0010         0    0.0022

  Columns 9 through 16

       0   -0.0043         0    0.0077         0   -0.0128         0    0.0207

  Columns 17 through 24

       0   -0.0331         0    0.0542         0   -0.1002         0    0.3163

  Columns 25 through 32

    0.5000    0.3163         0   -0.1002         0    0.0542         0   -0.0331

  Columns 33 through 40

       0    0.0207         0   -0.0128         0    0.0077         0   -0.0043

  Columns 41 through 48
```

After you get the coefficients, create a new Nyquist FIR filter bmod and set the coefficients of hm2 to the coefficients from bmod.

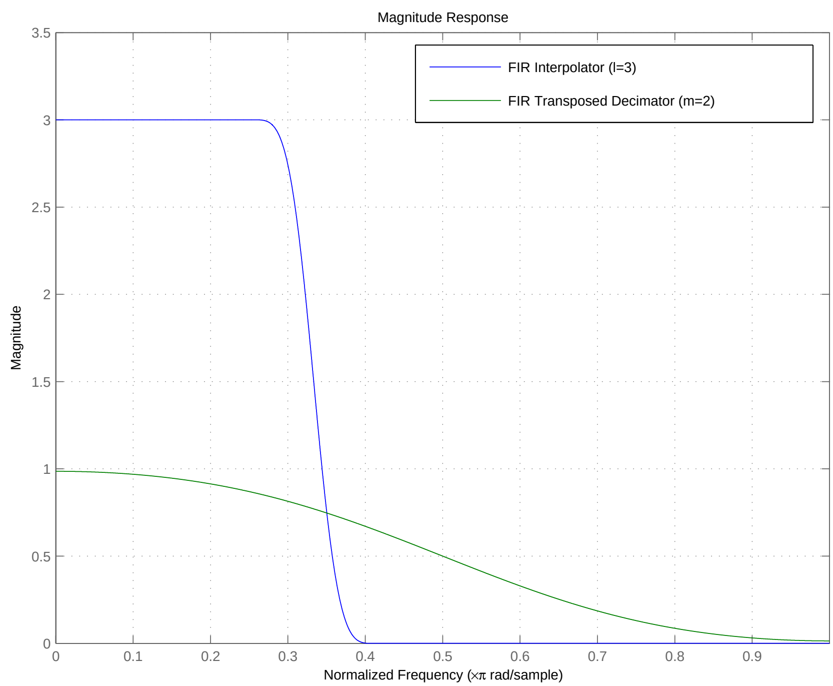
```
bmod = firnyquist(8,m,kaiser(9,0.1102*(80-8.71)));
set(hm2,'Numerator',bmod); % Set the modified coefficients.
hm2.numerator

ans =

         0   -0.0092         0    0.2522    0.5000    0.2522         0   -0.0092
```

You do not have to use a Nyquist filter to get new filter coefficients; other FIR filter design techniques in the toolbox work as well.

Analyzing Multirate and Multistage Filters

Analyzing multirate or multistage filter objects is similar to analyzing discrete-time filter (`dfilt`) objects. Many if not all of the analysis functions for `dfilt` objects apply to `mfilt` objects equally. In particular, the Filter Visualization Tool (FVTool) provides most of the filter analysis tools you need.

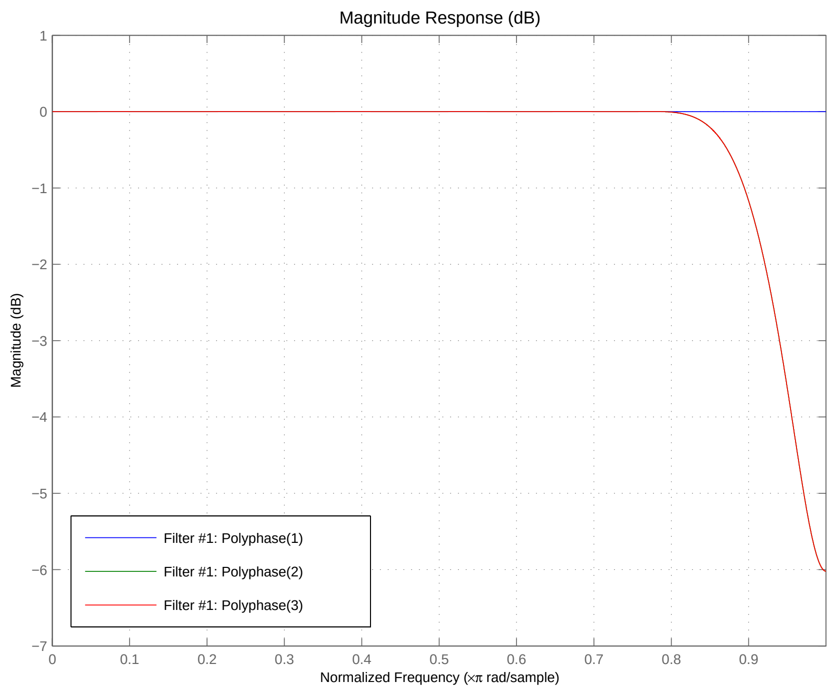
```
h = fvtool(hm1,hm2);
```



But one difference is very important. In analyzing multirate and multistage filters, the filter sample rates become important. The toolbox and tools let you specify sample rates for all of your analyses.

Additionally, `polyphase` for `mfilt` objects provides a tool for analyzing the polyphase components of `mfilt` objects. Calling the `polyphase` method without output arguments (as shown here using filter `hm1`) starts an FVTool session with the polyphase subfilters ready for you to analyze.

```
polyphase(hm1)
```



Filtering with Multirate Filters

By default, multirate filters begin with zero-valued filter states. Furthermore the `PersistentMemory` property is set to `false`, meaning that the filter object properties, such as the filter states, are reset before each filter run. This built-in reset process allows you to filter the same sequence input data sequence twice and produce the same output. For example:

```
x = 1:6;
y1 = filter(hm2,x) % First run
```

```
y1 =

         0   -0.0184    0.9676
```

At this point, you can verify that the filter `hm2` holds nonzero final conditions in the filter states.

```
zf1 = hm2.States

zf1 =

    3.0133
    3.4904
   -0.0369
```

Run the filter again using the same input data `x`.

```
y2 = filter(hm2,x) % Second run

y2 =

         0   -0.0184    0.9676

zf2 = hm2.States

zf2 =

    3.0133
    3.4904
   -0.0369
```

After the second run, the states of the filter are the same as they were after the first run. With `PersistentMemory` property set to `false`, the filter states were reinitialized to zeros before the second run.

Specifying Initial Conditions to the Filter

You make it possible to specify the initial conditions for your filter by seting both of the following:

- The `PersistentMemory` property to `true`
- The `States` property to your initial conditions (ICs)

Setting the `PersistentMemory` property to `true` is essential in the process of specifying initial conditions. If you set your filter ICs to specific values, but you do not enable the filter memory, when you use the filter with input data, the ICs get reset to zeros before the filter runs. As a result you lose your desired ICs and the results of filtering are not correct.

When you set the ICs, if you provide a scalar, that value is expanded to the correct number of states. If you specify a vector of values, its length must be equal to the number of states for the filter.

For example, using `hm2` as the filter, experiment with setting the filter states before filtering an input data set.

```
hm2.persistentmemory='true'


hm2.states=zf1

hm2 =

       FilterStructure: 'Direct-Form Transposed FIR Polyphase Decimator'
             Numerator: [0 -0.0092 0 0.2522 0.5000 0.2522 0 -0.0092]
      DecimationFactor: 2
      PersistentMemory: true
                States: [3x1 double]
    NumSamplesProcessed: 6

y3=filter(hm2,x)

y3 =

    2.9580    4.9853    2.4440


zf3=hm2.states

zf3 =

    2.9580
    3.4904
   -0.0369
```

As you might have anticipated, the filter output and the filter states are different now than they were after the first run.

Streaming Data to the Filter

Setting the filter property PersistentMemory to true is a valuable feature when you are filtering streaming data. Since breaking a signal into sections and filtering the sections in a loop is equivalent to filtering the entire signal at once, this example simulates filtering streaming data by using the filter hm2 in a loop.

```
reset(hm2); % Clear history of the filter by resetting all states.
xsec = reshape(x(:),2,3);   % Break the input signal into
                              % three sections.
yloop = zeros(1,3);           % Preallocate memory for storing
                                % intermediate results.
for i=1:3,
    yloop(i)=filter(hm2,xsec(:,i));
end
yloop

yloop =

         0   -0.0184    0.9676

y1

y1 =

         0   -0.0184    0.9676
```

You have verified that yloop (the signal filtered by three sections) is equal to y1 (entire signal filtered at once). Without changing the property value for PersistentMemory, this test does not work.

Filtering Multichannel Signals

Up to this point you have only done single channel filtering, entering a vector of data x for the filter. When the input signal x is a matrix, the filter interprets each column of x as an independent input channel. Thus an 11-by-4 matrix provides 4 channels of input data where each channel contains 11 samples.

Before you can continue this tutorial and experiment with multichannel filtering, you must either reset your filter to the initial states, or set the PersistentMemory property to false. The toolbox does not let you switch between single channel and multichannel filtering unless PersistentMemory is false or you reset the filter manually. If you forget to do this step, MATLAB returns an error message to tell you to reset your filter.

This example begins by resetting hm2 and defining some data to filter.

```
reset(hm2);
x = randn(10,3); % Three channel signal; each channel providing
                 % ten samples.
y = filter(hm2,x)

y =

         0         0         0
   -0.0094    0.0095   -0.0022
    0.0794    0.3678    0.5956
    0.0440   -0.2253    1.1980
    0.6913    0.3884    0.3812


zf = hm2.States

zf =

    0.9268   -0.0027    0.4663
   -0.5359   -0.6960    0.3092
    0.0066    0.0123   -0.0029
```

Notice that the filter object stores the final conditions for each channel separately. Each column of the States property corresponds to one input channel or column in the input matrix x.

Filtering Multichannel Data in Loops

When x is a matrix, the filter treats each matrix column as an independent channel. When you are filtering multichannel data, dim lets you specify which dimension of the input matrix to filter along—whether a row represents a channel or a column represents a channel. To filter multichannel data in a loop environment, you must use the dim input argument to set the processing dimension.

You specify the initial conditions for each channel individually, when needed, by setting `hm.states` to a matrix of `nstates(hm)` rows (each individual row containing the states for one channel of input data) and `size(x,2)` columns (one column containing the filter states for each channel).

Here is an example that uses the `dim` input argument to filter the multichannel input data matrix `x`.

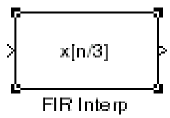
```
Fs = 44.1e3;           % Original sampling frequency 44.1kHz
 n = [0:10239].';         % 10240 samples, 0.232s signal.
x  = sin(2*pi*1e3/Fs*n); % Original signal, sinusoid at 1kHz.
 M = 2;                   % Decimation factor.
 Hm = mfilt.firdecim(M);  % We use the default filter.

 % No initial conditions
 y1 = filter(Hm,x);       % PersistentMemory is false.
 zf = Hm.States;          % Final conditions.

 % Non-zero initial conditions.
 Hm.PersistentMemory = true;
 Hm.States = 1;           % Uses scalar expansion.
 y2 = filter(Hm,x);
 stem([y1(1:60) y2(1:60)])% Different sequences at the
                          % beginning.
 % Streaming data
 reset(Hm);               % Clear filter history.
 y3 = filter(Hm,x);       % Filter the entire signal in one
                          % block.
 reset(Hm);               % Clear filter history.
 yloop = [];
 xblock = reshape(x,[2048 5]);
 % Filtering the signal section by section is equivalent to
 % filtering the entire signal at once.
 for i=1:5,
   yloop = [yloop; filter(Hm,xblock(:,i))];
 end
```

Generating Simulink Blocks

When the Signal Processing Blockset is installed, you can generate a Simulink® block of the `mfilt` object if the Signal Processing Blockset supports the filter structure. For example `hm1`, the direct-form FIR polyphase interpolator that you have been using throughout these examples, can be rendered as a Simulink block.

```
block(hm1,'destination','new','blockname','FIR Interp');
```

This figure shows the block as generated by the toolbox from the filter `hm1`.



Getting Help About Multirate Filters

Entering `helpwin mfilt` in the MATLAB Command Window returns a list of multirate structures that the toolbox supports, as well as functions that operate on `mfilt` objects. For further information about a particular structure or function, enter `helpwin mfilt/`*`functionname`*, which returns the help information about *`functionname`* in a formatted HTML view, or enter `help mfilt/`*`functionname`* that returns the help information as plain text. For example:

```
help mfilt/firinterp % Help on the FIRINTERP structure
```

returns the following text in the Command Window.

```
FIRINTERP Direct-Form FIR Polyphase Interpolator.

Hm = mfilt.FIRINTERP(L,NUM) returns a direct-form FIR polyphase
interpolator Hm.

L is the interpolation factor. It must be an integer. If not
specified, it defaults to 2.

NUM is a vector containing the coefficients of the FIR lowpass filter
used for interpolation. If omitted, a low-pass Nyquist filter of gain
L

EXAMPLE: Interpolation by a factor of 2 (used to convert from
22.05kHz to 44.1kHz)

L = 2;                     % Interpolation factor.

Hm = mfilt.firinterp(L);   % We use the default filter.

Fs = 22.05e3;              % Original sampling frequency: 22.05kHz.
```

```
n = 0:5119;                  % 5120 samples, 0.232 second long
signal.
x  = sin(2*pi*1e3/Fs*n);     % Original signal, sinusoid at 1kHz.
y = filter(Hm,x);            % 10240 samples, still 0.232 seconds.
stem(n(1:22)/Fs,x(1:22),'filled')   % Plot original sampled at
                                    % 22.05kHz.
hold on                      % Plot interpolated signal (44.1kHz) in
                             % red.
stem(n(1:44)/(Fs*L),y(25:68),'r')
xlabel('Time (sec)');
ylabel('Signal value')
See also mfilt/HOLDINTERP, mfilt/LINEARINTERP, mfilt/FFTFIRINTERP,
mfilt/FIRFRACINTERP, mfilt/CICINTERP, mfilt/CICINTERPZEROLAT,
FDESIGN/INTERP, FDESIGN/SRC.
```

You can also enter

`help mfilt`/polyphase % Get help on the POLYPHASE method

at the MATLAB prompt to return this information about `polyphase`.

```
POLYPHASE Polyphase decomposition of multirate filters.
P=POLYPHASE(Hm) returns the polyphase matrix of the multirate filter
Hm. The ith row of the matrix P represents the ith subfilter.
POLYPHASE(Hm) called with no outputs launches the Filter
Visualization Tool (FVTool) with all the polyphase subfilters to
allow analysis of each component individually.
```

To use the online help system, use the `doc` function instead of `help`.

```
doc mfilt
```

opens the Help browser and displays the general help text for multirate filter objects.

To obtain information about CIC decimation filter objects, enter either

```
helpwin mfilt/cicdecim
helpwin mfilt/cicinterp
```

at the command prompt, depending on which structure you need to know about.

For a complete list of the multirate filters that are available in the toolbox, enter `help mfilt`.

FIR Decimation—Filtering with FIR Decimators

This section demonstrates how you can decrease the sampling rate of a signal using FIR decimators from the toolbox. To show you how this works, this section takes you through the following tasks:

- "Creating FIR Decimators" on page 5-18
- "Understanding Input Sample Processing and the InputOffset Property" on page 5-19
- "Filtering with FIR Decimators" on page 5-21

Creating FIR Decimators

The Filter Design Toolbox supports different structures to perform decimation including different FIR-based structures and cascaded integrator-comb (CIC) structures. Entering `helpwin mfilt` at the prompt gives you a list of all supported structures.

Start by defining the filter decimation factor for your FIR decimator.

```
m = 3; % Specify the decimation factor as m.
```

Since the toolbox uses objects to implement multirate filters, you use the same methods to create most decimators. First you specify the decimation factor and then the FIR filter coefficients. If you do not include filter coefficients when you construct the filter, the toolbox filter constructor returns a lowpass filter with a cutoff frequency of (π/decimation factor) and a gain of 1. This example uses `mfilt.firdecim` to create a direct-form polyphase FIR decimator. After constructing the filter, you can change the filter coefficients that are stored in the `Numerator` property.

Begin by designing an FIR decimator with the decimation factor set to `3`.

```
hm1 = mfilt.firdecim(m); % Default decimator filter
```

`mfilt.firdecim` produces filters that decimate signals by integer factors. To change the sampling rate of a signal by a fractional factor, you might use a direct-form FIR polyphase sample rate converter. One way to create such a rate-changing filter is `mfilt.firsrc`. This structure uses L polyphase subfilters where L is the interpolation factor. Sample rate convertors use both a decimation factor and interpolation factor to perform fractional rate changing.

```
l = 2; % Set the interpolation factor.
hm2 = mfilt.firsrc(l,m); % Create the rate changing filter.
```

Here is the configuration information about hm2.

```
hm2 =

       FilterStructure: 'Direct-Form FIR Polyphase Sample-Rate Converter'
             Numerator: [1x72 double]
     RateChangeFactors: [2 3]
      PersistentMemory: false
                States: [35x1 double]
   NumSamplesProcessed: 0
```

Understanding Input Sample Processing and the InputOffset Property

When you decimate signals whose length is not a multiple of the decimation factor M, the last samples—(nM +1) to [(n+1)(M) -1], where n is an integer—are processed and used to track where the filter stopped processing input data and when to expect the next output sample. If you think of the filtering process as generating an output for a block of input data, where each block has M elements, every complete input data block yields one output sample. Incomplete blocks of data (one or more input samples up to one less than the decimation factor) increment the InputOffset property by one for each sample in the incomplete block.

Note InputOffset applies only when you set PersistentMemory to true. Otherwise, InputOffset is not available for you to use.

Two different cases can arise when you decimate a signal:

1 The input signal is a multiple of the filter decimation factor. In this case, the filter processes the input samples and generates output samples for all inputs as determined by the decimation factor. For example, processing 99 input samples with a filter that decimates by three returns 33 output samples.

2 The input signal is not a multiple of the decimation factor. When this occurs, the filter processes all of the input samples, generates output samples as

determined by the decimation factor, and has one or more input samples that were processed but did not generate an output sample.

For example, when you filter 100 input samples with a filter which has decimation factor of 3, you get 33 output samples, and 1 sample that did not generate an output. In this case, `InputOffset` stores the value 1 after the filter run.

`InputOffset` equal to 1 indicates that, if you divide your input signal into blocks of data with length equal to your filter decimation factor, the filter processed one sample from a new block of data. Subsequent inputs to the filter are concatenated with this single sample to form the next block of length m.

One way to define the value stored in `InputOffset` is

```
InputOffset = mod(length(nx),m)
```

where `nx` is the number of input samples in the data set and `m` is the decimation factor.

Storing `InputOffset` in the filter allows you to stop filtering a signal at any point and start over from there (provided that the `PersistentMemory` property is set to `true`). Being able to resume filtering after stopping a signal lets you break large data sets in to smalller pieces for filtering. With `PersistentMemory` set to `true` and the `InputOffset` property in the filter, breaking a signal into sections of arbitrary length and filtering the sections is equivalent to filtering the entire signal at once.

```
xtot=[x,x];
ytot=filter(hm1,xtot)
ytot =

       0   -0.0003    0.0005   -0.0014    0.0028   -0.0054    0.0092
reset(hm1);  % Clear history of the filter
hm1.PersistentMemory='true';
ysec=[filter(hm1,x) filter(hm1,x)]

ysec =

       0   -0.0003    0.0005   -0.0014    0.0028   -0.0054    0.0092
```

This test verifies that ysec (the signal filtered by sections) is equal to ytot (the entire signal filtered at once).

All of the preceding discussion applies to interpolation filters as well, with appropriate changes from decimation to interpolation.

Filtering with FIR Decimators

After creating your decimator, you are ready to filter data. Rather than use random data, as you did earlier, this example uses a more realistic data set.

For this example, define the input signal x as a 1 kHz sinusoid sampled at 44.1 kHz. Here is one way to create x[n].

```
N = 159;
fs = 44.1e3;
n = 0:N-1;
x = sin(2*pi*n*1e3/fs); % Signal as required. 159 data points.
```

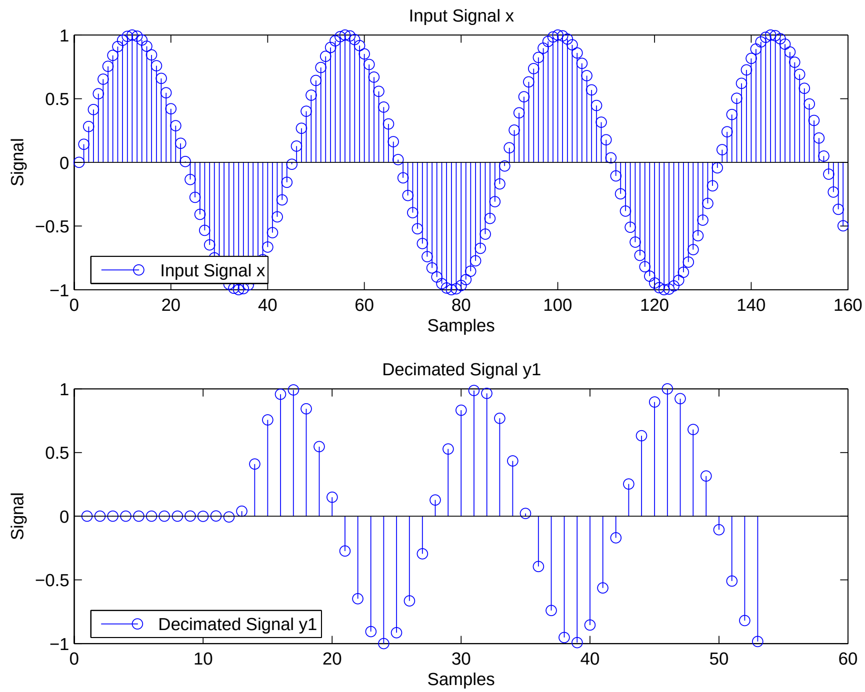
Now you can use filter hm1 you designed earlier to try decimating x.

Filtering with the Direct-Form FIR Polyphase Decimator hm1

You have data and a decimator in your workspace. Applying the filter to the data takes two steps—reset the filter and use filter to apply the decimator to x.

```
reset(hm1) % Reset the filter history and states to zeros.
y1 = filter(hm1,x);
```

Two stem plots give a sense of the decimation.



`y1` contains 53 samples—one-third of the number in `x`. Filter `hm1` decimated `x` by two-thirds. Since multirate filters support sample-by-sample processing, all input samples passed through the filter.

For further information about filtering options in general and specifying initial conditions for filters in particular, refer to "Getting Started—Designing Multirate Filters" on page 5-4.

The previous stem plot shows a feature of the filter—a delay of a number of samples before the filter starts to output the decimated input signal. Called the transient response, the length of the transient response of the decimator is equal to half the order of a polyphase subfilter. For `hm1`, the subfilter order is 24, so the transient response should be 12 samples. This is also the group delay of the filter.

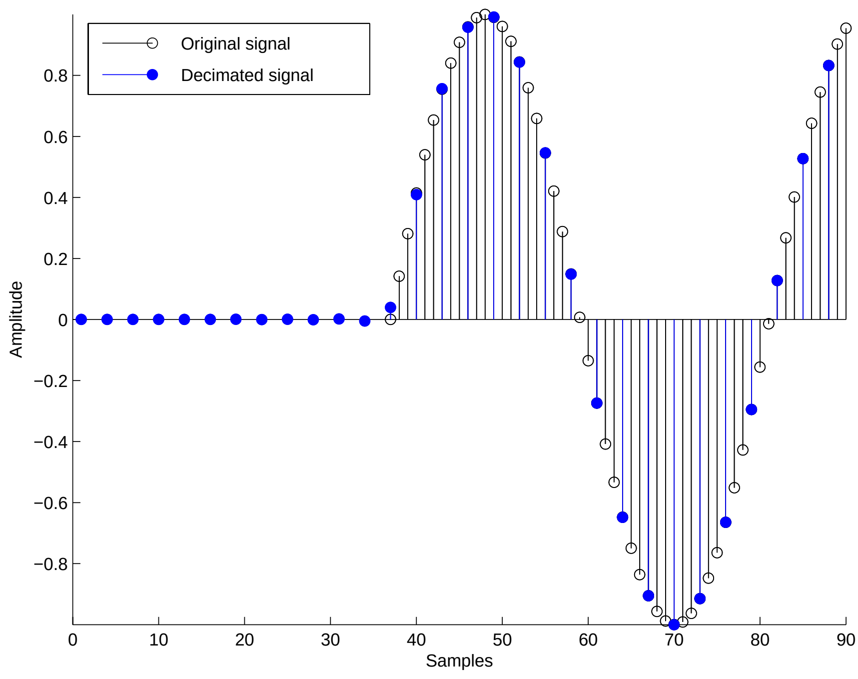
From the plot, it appears that the transient response is about 12 samples long. The next plot makes this more clear by plotting the decimated signal with a delayed version of the input x.

```
delay = mean(grpdelay(hm1)); % Constant group delay equal to its
                               % mean.
tx = delay+[1:length(x)];
ty = 1:m:m*length(y1);
```

Plot the output of the direct-form FIR polyphase decimator hm1 and overlay a shifted version of the original signal using tx and ty.

```
stem(tx,x,'k');hold on;stem(ty,y1,'filled');
```

Using the delayed signals makes the transient response clear.



CIC Filter Example—Using CIC Decimation Filters

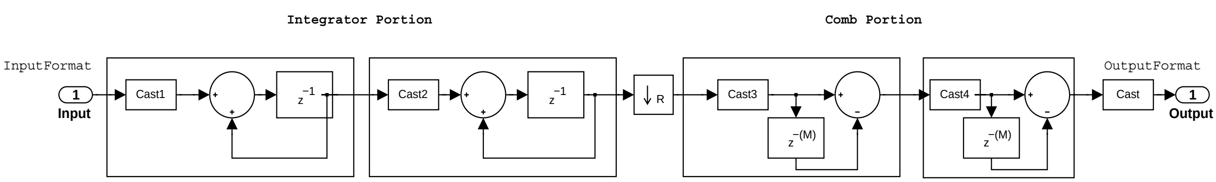
This demonstration shows how to use multirate cascaded integrator-comb (CIC) decimation filters in the Filter Design Toolbox. CIC filters are efficient, multiplierless structures that are often used in high-decimation ratio or high-interpolation ratio systems.

Digital down convertors and digital up convertors commonly use CIC filters. Refer to the demonstration program "Design of a Digital Down-Converter for GSM (Group Speciale Mobile)" in the Filter Design Toolbox demos for an example that uses a CIC decimator for digital down-conversion processing of a signal.

To help you understand what CIC filters do and how, this example includes the following sections:

Creating CIC Decimator filters

The Filter Design Toolbox provides a CIC decimating filter structure—the Cascaded Integrator-Comb Decimator. As you see in the figure below, the structure is optimized for pipelined implementations such as might be used on field-programmable gate arrays (FPGAs). The following Simulink model provides a signal-flow graph of the structure.



With the Fixed-Point Toolbox installed (required for you to use CIC filters), you create a default cascaded integrator-comb decimator object with this command

```
hm = mfilt.cicdecim
```

at the prompt. MATLAB returns the CIC filter with the specifications shown here.

```
hm =

        FilterStructure: 'Cascaded Integrator-Comb Decimator'
             Arithmetic: 'fixed'
      DifferentialDelay: 1
       NumberOfSections: 2
       DecimationFactor: 2
       PersistentMemory: false

        InputWordLength: 16
        InputFracLength: 15

  SectionWordLengthMode: 'MinWordLengths'

       OutputWordLength: 16
```

The CIC decimation filter comprises three portions—an integrator portion, a rate change factor, and a comb portion. Similarly, you can completely specify a CIC decimation filter with three parameters—a decimation factor r, the number of individual integrator or comb sections n, and the differential delay of the comb section m.

The display of the multirate filter object (`mfilt`) in the Command Window groups the filter properties together in a logical manner, making the filter specification more clear.

Only the writable properties appear in the display by default. Changing a filter property, such as resetting `PersistentMemory` from `false` to `true` reveals more properties as they become writable—in this case the `States` property appears when `PersistentMemory` is `true`.

Unlike other multirate filters and discrete-time objects, CIC filter objects allow only fixed-point arithmetic (the `Arithmetic` property is always set to `fixed`) since these filters are inherently fixed-point filters. Check the value of the `Arithmetic` property

```
set(hm,'arithmetic')

ans =

    'fixed'
```

to see that `fixed` is the only option. As with all filter objects, and all objects in general, the `get` function returns the complete set of properties (read-only and writable) for the filter and object.

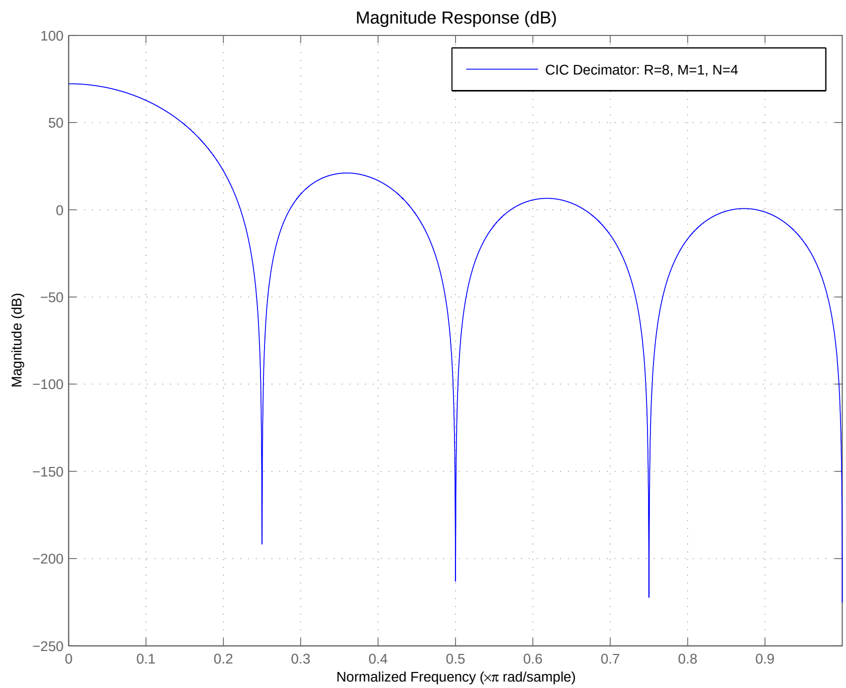
```
get(hm)
```

Analyzing CIC Decimation Filters

Analyzing CIC filters is the same as analyzing any multirate filter object in the Filter Design Toolbox. The Filter Visualization Tool (FVTool) provides graphical access to all analyses.

```
hm = mfilt.cicdecim(8,1,4);
hfvt=fvtool(hm);
hfvt.showreference='off';
```

FVTool returns the magnitude response for `hm`, shown here. As `hm` is a fixed-point filter, we suppress the reference filter in the display by setting the `ShowReference` property in FVTool to `off`.



After you have the filter displayed in FVTool, you can use any of the filter analysis capabilities provided to learn more about `hm`. To perform an analysis, select one of the analytical options, such as **Impulse Response** or **Round-off Noise Power Spectrum** from **Analysis** on the FVTool menu bar.

About the MSB at the Filter Output

A significant consideration in CIC filters is the size (number of bits) of data that can pass through the filter without loss. The most significant bit (MSB) of the filter represents the maximum number of bits that can be propagated through the filter while maintaining the integrity of the data.

Parameters R, M, N and the `InputWordLength` specify the MSB of the filter output. Since the output of the integrator sections of the filter can grow without bounds, the MSB at the filter output is also the MSB for all filter sections.

Called B_{max} in the reference, the maximum wordlength in the filter, or most significant bit (MSB), is both the maximum word length for all of the filter sections as well as the MSB at the filter output.

Hogenauer defines B_{max} , the MSB at the filter output, as follows:

$$B_{max} = [N\log_2 RM + B_{in} - 1]$$

with

- N is the number of filter sections
- M is the comb portion differential delay
- R is the decimation factor
- B_{in} is the input word length in bits

Working with Section Word Lengths

CIC filters include a property that defines how you specify the section word and fraction lengths for the filter. Called `SectionWordLengthMode`, this property specifies the specific data format (word length and fraction length) the filter uses when accumulating data in the integrator sections or subtracting data in the comb sections. `SectionWordLengthMode` can take one of two values:

- `MinWordLengths`—the filter calculates the optimal section word lengths given the filter parameters R (the rate change factor), M (the differential delay), N (the number of filter sections), and the input and output word lengths.
- `SpecifyWordLengths`—you specify the word lengths for the sections by entering a scalar or a vector of length 2*N. When you provide a scalar, the filter method expands the scalar into a vector with 2*N elements, applying the same word length to all sections. If you specify a vector, it must meet these requirements:
  - It must contain 2*N elements.
  - The values of the vector elements must be monotonically decreasing.

When you construct a new CIC decimating filter, `SectionWordLengthMode` is set to `MinWordLength` by default.

Using hm as an example, here is the `SectionWordLengthMode`.

```
set(hm,'SectionWordLengthMode')
```

```
ans =

    'MinWordLengths'
    'SpecifyWordLengths'
```

In the reference provided later in this section ([1] on page 5-35), Hogenauer shows that during filtering you can discard least significant bits (LSBs) from each section (refer to Equation 21 of the reference) of the filter so long as the error introduced by removing the LSBs is acceptable at the filter output. In this case, the section word lengths reported by the filter are computed by subtracting the LSBs from the maximum word lengths in the filter (refer to Equation 11 in the reference for details).

To help connect the CIC filter designs in the toolbox to the analysis by Hogenauer, the next example designs a CIC decimator that matches the design on page 159 of the Hogenauer paper.

```
m=1;   % Set the differential delay to one.
n=4;   % Specify the number of sections.
r=25;  % Set the rate change factor.
inwl=16;   % Set the word length at the filter input.
outwl=16;  % Set the filter output word length.

% With the specifications prepared, design the CIC decimator.
hm=mfilt.cicdecim(r,m,n,inwl,outwl);
```

`hm` reproduces the referenced filter exactly. To see the correspondence, check that the word lengths applied to each filter section match those developed in the reference example, where the MSB is 34 bits.

Filter Section	**Number of LSBs Discarded**	**Word Length Calculated in [1] on page 5-35 (MSB-Discarded LSBs)**
1	1	33 (34-1)
2	6	28 (34-6)
3	9	25 (34-9)
4	13	21 (34-13)

Filter Section	Number of LSBs Discarded	Word Length Calculated in [1] on page 5-35 (MSB-Discarded LSBs)
5	14	20 (34-14)
6	15	19 (34-15)
7	16	18 (34-16)
8	17	17 (34-17)

In the referenced paper by Hogenauer, the MSB is also called B_{max}. Use `get` to verify the match.

```
get(hm,'sectionwordlengths')

ans =

    33    28    25    21    20    19    18    17
```

For cases where you enter the word lengths explicitly when you construct the filter, rather than letting the `mfilt` constructor determine them, by setting `SectionWordLengthMode` to `SpecifyWordLengths`, you enter the word lengths to use as either a scalar or a vector of length 2*`n`. Recall from earlier that the input vector containing the section word lengths must meet two criteria—the number of elements must be twice the number of filter sections `n`, and the element values must be monotonically decreasing.

As you see in this example, when you enter the word length as a scalar, the filter constructor expands the scalar to apply it as the section word length for all of the filter sections.

```
set(hm,'sectionWordLengthMode','SpecifyWordLengths');
hm.sectionWordLengths=32;
get(hm,'sectionWordLengths')

ans =

    32    32    32    32    32    32    32    32
```

CIC Filter States

The `States` property of CIC decimation filters contains an object—`filtstates.cic`. This object represents or stores the initial conditions of the filter before filtering and the final conditions after filtering. `filtstates.cic` has two properties, `Integrator` and `Comb`, that correspond to their respective portions of the filter. When you construct a CIC filter, the states contain zeros. After you filter data with the filter, the states contain the values stored in the filter delay elements. To demonstrate the filter states, the following example creates a decimator, and then applies the filter to a set of fixed-point input data.

```
% Construct the input data set for filter filter some fixed-point
% ones.
x = fi(ones(1,10),true,16,0);
% Construct a decimator to use to filter x.
hm = mfilt.cicdecim(2,1,2,16,16,16);
```

Take a look at `x` and `hm` to see what you have.

```
x

x =

     1     1     1     1     1     1     1     1     1     1


          DataTypeMode: Fixed-point: binary point scaling
                Signed: true
            WordLength: 16
        FractionLength: 0

             RoundMode: round
          OverflowMode: saturate
           ProductMode: FullPrecision
  MaxProductWordLength: 128
               SumMode: FullPrecision
      MaxSumWordLength: 128
         CastBeforeSum: true
hm

hm =
```

```
         FilterStructure: 'Cascaded Integrator-Comb Decimator'
              Arithmetic: 'fixed'
       DifferentialDelay: 1
        NumberOfSections: 2
        DecimationFactor: 2
        PersistentMemory: false

         InputWordLength: 16
         InputFracLength: 15

   SectionWordLengthMode: 'SpecifyWordLengths'

      SectionWordLengths: [16 16 16 16]

        OutputWordLength: 16
get(hm,'states')

ans =

    Integrator: [2x1 States]
          Comb: [2x1 States]
```

At this point, the states for the filter are zeros. That changes after you filter a set of data.

```
hm.inputfraclength = 0; % Set the input to use integer data.
y = filter(hm,x);
```

You can extract the final states by using the int function and assigning the output to a variable.

```
sts = int(Hm.states)
sts =

     10     45
     28     13
```

As you see, the states now contain nonzero values related to the filtering operation.

This states matrix has dimensions M+1-by-N, where M is the differential delay of the comb section and N is the number of sections. Filter `hm` stores the integrator sections states (`hm.states.integrator`) in the first row of the states matrix and stores the states for the comb portion in the remaining rows in the matrix.

You might have noticed that the `States` property is not displayed by the default filter display. When `PersistentMemory` is set to `false`, you do not see the states property in the default listing in MATLAB.

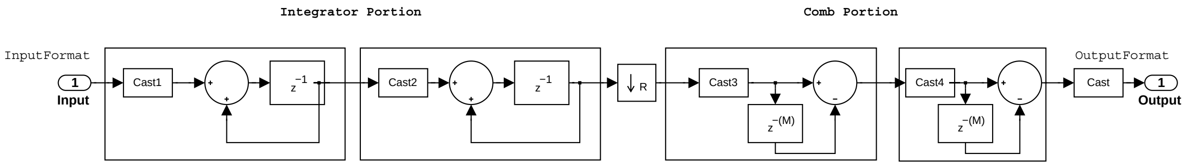
```
hm % Generate the default filter display.

hm =

       FilterStructure: 'Cascaded Integrator-Comb Decimator'
            Arithmetic: 'fixed'
     DifferentialDelay: 1
      NumberOfSections: 2
      DecimationFactor: 2
      PersistentMemory: false

       InputWordLength: 16
       InputFracLength: 15

SectionWordLengthMode: 'MinWordLengths'

      OutputWordLength: 16
```

Setting `PersistentMemory` to `true` reveals the `States` property in the filter display. However, when you use `get` to review the properties, you see the `States` property listed in all instances.

For more information about the `fi` object used in `x` above, refer to the Fixed-Point Toolbox documentation in the online Help system.

Filter Implementation—Signal Flow Graph

The toolbox implements a structure that differs slightly from the one in the referenced paper by [1] on page 5-35. The difference lies in the location of the delays in the integrator portion of the filter. We made this change to optimize the filter for pipelining on hardware such as field-programmable gate arrays (FPGAs). The following figure shows the flow graph as implemented by mfilt.cicdecim. After the table following the figure, is a short example that should help interpret the entries in the figure.



The word length and fraction length at each stage of the decimator are shown in the following table. Either you specify the word length for each filter stage in the SectionWordLengths property as a vector of integers, or you let the filter constructor set the word lengths by making MinWordLengths the value for SectionWordLengthMode. The calculation for each fraction length is shown below:

Decimator Word Lengths and Fraction Lengths

Position in the Signal Flow	Word Length	Fraction Length
Filter Input	InputWL	InputFL
1^{st} Section Output	SectionOneWL	InputFL
2^{nd} Section Output	SectionTwoWL	InputFL (SectionTwoWL - SectionOneWL)
3^{rd} Section Output	SectionThreeWL	SectionTwoFL + (SectionThreeWL - SectionTwoWL)
4^{th} Section Output	SectionFourWL	SectionThreeFL + (SectionFourWL - SectionThreeWL)
N^{th} Section Output	Section(N)WL	Section(N-1)FL + (Section(N)WL - Section(N-1)WL)
Filter Output	OutputWL	FinalSectionFL + (OutputWL - FinalSectionWL)

Reference

The following paper formed the basis for developing the CIC filters in the Filter Design Toolbox. Many more details of the CIC multirate filters are discussed in this reference.

[1] Hogenauer, E. B., “An Economical Class of Digital Filters for Decimation and Interpolation,” *IEEE Transactions on Acoustics, Speech, and Signal Processing*, Vol. ASSP-29, No. 2, April 1981, pp. 155-162.

Analyzing Multirate and Multistage Filters

Multirate filter analysis presents some differences from analyzing single-rate discrete-time filters. While most of the same analytical tools apply, the primary difference is the filter sample rate—what the sample rate is, how it is defined, and where. Filter sample rate, called Fs in the toolbox, changes depending on the type of multirate filter you are using. Or more precisely, how the sample rate is defined changes according to the multirate filter under discussion.

Generally, filter sample rate refers to the rate at which the filter is running:

- For decimators, the filter sample rate equals the sample rate at the filter input, prior to decimating the input.
- For interpolators, the filter sample rate is equal to the sample rate at the output of the filter, after interpolation.
- For sample rate change filters, Fs is the input rate multiplied by the interpolation factor. The decimation factor does not apply to define the sample rate.

When you provide a sampling frequency for the analysis, the analytical tool, such as FVTool, assume that the rate specified is the sampling frequency at which the filter is operating.

Another feature of analyzing multirate filters that have more than one stage is that the analysis process applies to a filter that is the overall equivalent of the multistage filter under consideration. Recognizing that the analytical tool you choose first computes an equivalent filter makes understanding the analytical process somewhat easier.

For example, a multistage filter that included

- Multiple interpolators
- Multiple decimators

might be reduced to an equivalent filter with

- One equivalent interpolation filter
- One equivalent decimation stage

For more about how the tools develop the equivalent filter they use to analyze your filter, refer to “Performing Multistage Filter Analysis” on page 5-40.

A pair of definitions will help as you read this section:

- Multirate filters consist of *sections*.
- Multistage filters are the result of using `dfilt.cascade` or `dfilt.parallel` to create filters by combining other filters. Each filter that composes the multistage filter is called a *stage*.

This tutorial demonstrates how to perform analysis on single-stage and multistage multirate filters by presenting the following topics:

Analyzing Single-Stage Multirate Filters

You analyze single-stage multirate filters at the rate the filter is operating. As mentioned in the introduction to this tutorial section, the sample rate you use depends on the filter your are analyzing.

The following plot overlays the magnitude response of a sample-rate converter, an interpolator, and a decimator. For the first filter, the input sampling frequency is 1000/5 and the output sampling frequency is 1000/3. For the interpolator, the input `fs` is 1000/4 and the output `fs` is 1000. Finally, for the decimator, the input `fs` is 1000 and the output `fs` is 1000/3.

Here are the commands to create the three filters to analyze.

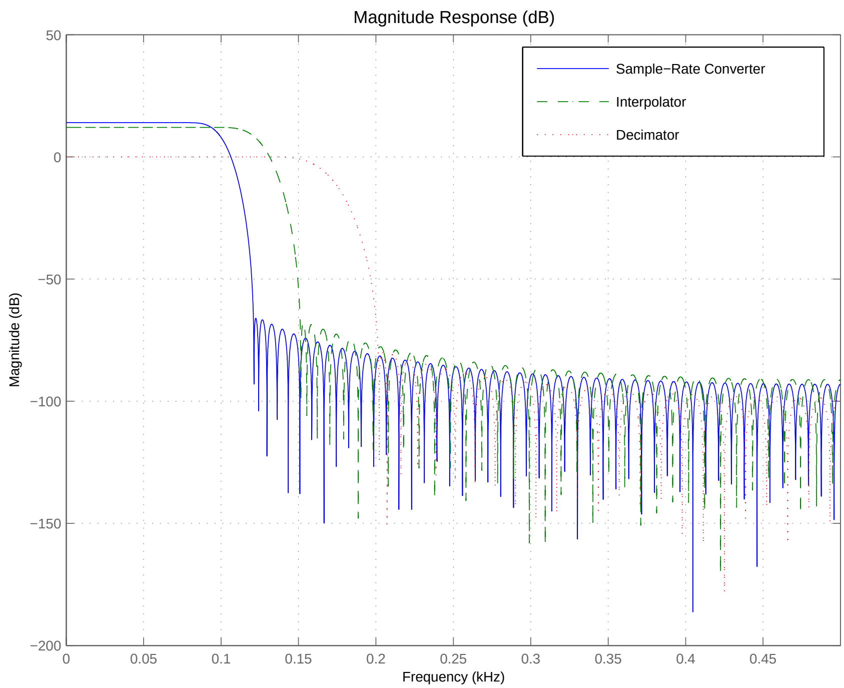
```
h1 = mfilt.firsrc(5,3);  % Use a default filter.
h2 = mfilt.firinterp(4); % Use a default filter.
h3 = mfilt.firdecim(3);  % Use a default filter.
```

Now you need to specify the sampling rate and the number of points in the FFT used.

```
fs = 1000; nfft = 8192;
```

With the filters in your workspace and the sampling frequency set, use FVTool to visualize the filters using a common sampling rate.

```
fvtool(h1,h2,h3,'fs',fs);
```



Comparing Interpolators

Interpolators and decimators exhibit a lowpass magnitude response. Simple interpolators, like the CIC interpolator and the hold or linear interpolators, have a poor lowpass response. However, they are easy to implement and they do not require the filter to perform any multiplications in real-time while filtering data. The following plot compares the lowpass response of four different interpolators:

- An FIR interpolator (`mfilt.firinterp`)
- A linear interpolator (`mfilt.linearinterp`)
- A hold interpolator (`mfilt.holdinterp`)

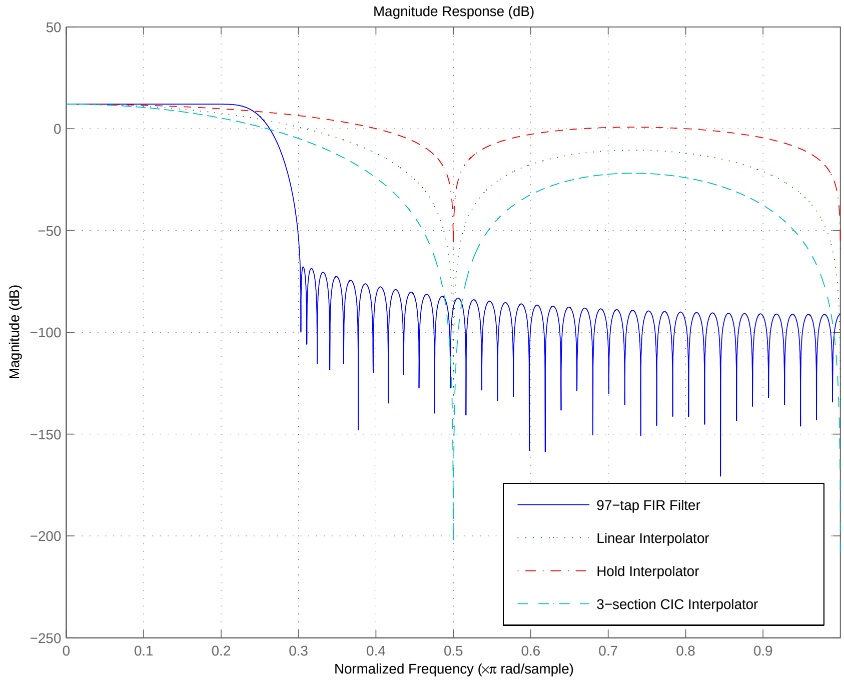
- A CIC interpolator (`mfilt.cicinterp`)

They each have an interpolation factor of 4. You can see that the quality of the lowpass filter, such as the sharpness of the lowpass cutoff, depends on which type of interpolator you use. By design, the CIC interpolator has more gain than the other interpolators. For the purposes of this analysis, we include a scalar in cascade with the CIC filter to normalize its gain. Normalizing the gain makes comparing the different filters easier.

```
h(1) = mfilt.firinterp(4); % Use the default filter.
h(2) = mfilt.linearinterp(4);
h(3) = mfilt.holdinterp(4);
hcic = mfilt.cicinterp(4,1,3); % 3-section CIC with
                                 % differential delay = 1.
hscalar = dfilt.scalar(1/gain(hcic));
h(4) = dfilt.cascade(hscalar,hcic); % Add a gain correction
filter...
                                % in cascade.
```

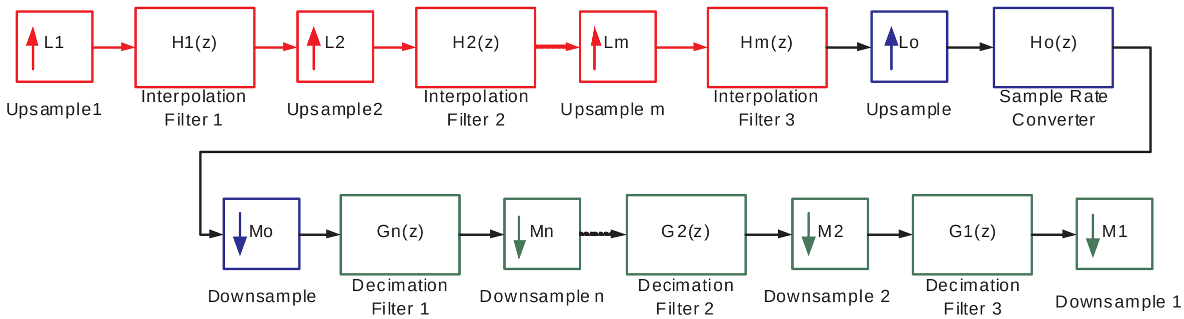
Use FVTool to see the results of the four filters. An interesting trick you might notice—naming the filters as indexes of the variable `h` lets you plot all four interpolators by passing `h` to FVTool.

```
fvtool(h);
```



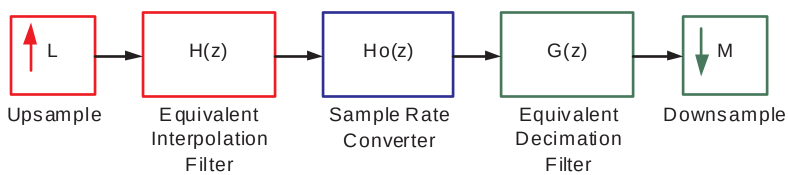
Performing Multistage Filter Analysis

Using the tools provided in the toolbox, either from the command line or in FVtool, you can analyze multirate filters of the following form.



In a multirate filter, any of the blue, red, or green sections is optional. Since that is true, you can perform analysis on multistage interpolators, multistage decimators, or multistage sample-rate converters.

When you choose to perform the analysis, the analysis tool computes an equivalent overall filter for the interpolation section and/or the decimation section as shown in the next figure, and performs the requested analysis on the equivalent filter.



In the equivalent filter shown in the figure, the following conversions apply.

- Upsample block L = Lo*L1*L2*...*Lm; (convolved interpolators)
- Downsample block M = Mo*M1*M2*...*Mn; (convolved decimators)
- Interpolator transfer function
  H(z) = H1(z^(Lo*L1*...Lm))*H2(z^(Lo*L2...Lm))...Hm(z^(Lo));
- Decimator transfer function
  G(z) = G1(z^(Mo*M1*...Mn))*G2(z^(Mo*M2...Mn))...Gn(z^(Mo))

Finally, filters H(z), G(z), and Ho(z) are all operating at the same rate and can be combined into a single filter on which to perform the analysis. If you specify a sampling frequency as an input to the analytical tool, the analysis assumes that the single overall filter (equivalent to the subfilters that have been combined) is operating at the rate you specified.

Analyzing Multistage Interpolators

Here is an example of how you might analyze a multistage interpolator. Refer to the demo "Design of a Digital Down-Converter for GSM" in the Filter Design Toolbox demos for an example in which the Global System for Mobile Communications (GSM) uses a multistage decimator.

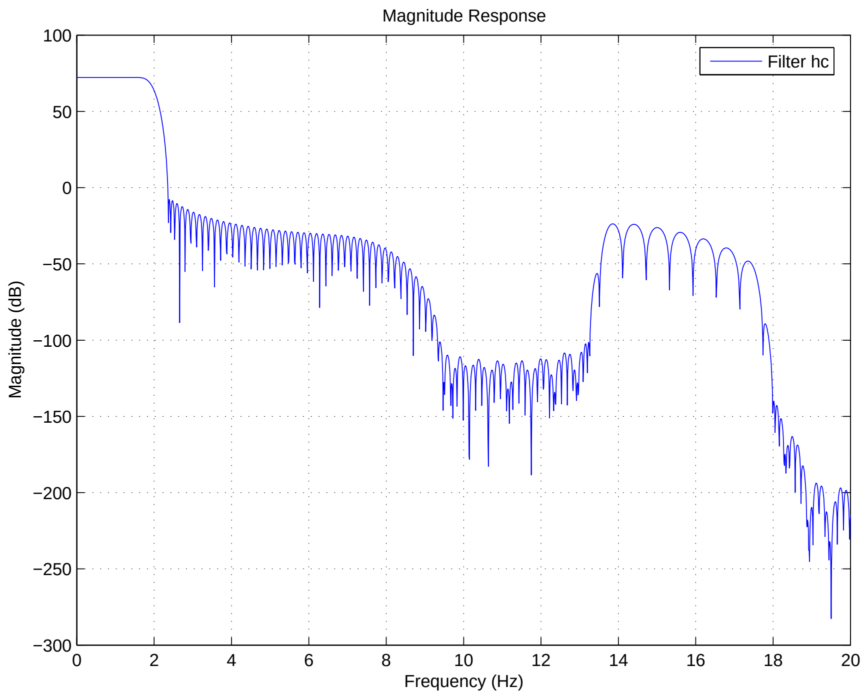
This section cascades four interpolators to form a four stage filter. The fourth interpolator is a CIC filter. In this case, the sampling frequency specified for the filter corresponds to the output of the four stage interpolator because this is the rate at which the equivalent filter operates.

```
h(1) = mfilt.firinterp(4);
h(2) = mfilt.firinterp(2);
h(3) = mfilt.firinterp(2);
h(4) = mfilt.cicinterp(16);
hc = dfilt.cascade(h);

hc

hc =

         FilterStructure: Cascade
                Stage(1): Direct-Form FIR Polyphase Interpolator
                Stage(2): Direct-Form FIR Polyphase Interpolator
                Stage(3): Direct-Form FIR Polyphase Interpolator
                Stage(4): Cascaded Integrator-Comb Interpolator
        PersistentMemory: false
```

To perform the analysis on `hc`, compute the frequency response between 0 and 200 Hz. set the sampling frequency Fs to 1000 Hz.

```
[hf,f] = freqz(hc,0:1e-2:20,1000);
plot(f,20*log10(abs(hf)))
```

`freqz` returns the transfer function for the cascaded filter at the sampling frequency you entered as an input argument.



Analyzing a Multistage Sample-Rate Converter

To demonstrate working with multistage sample rate convertors, add some decimation stages to filter hc to form a multistage sample-rate converter. Again, the sampling frequency fs you specify as input to freqz once again represents and is assumed to be the rate of the equivalent filter. And this is the rate at which the frequency response of hc2 is analyzed. This fs is the fastest rate in the entire system in this case.

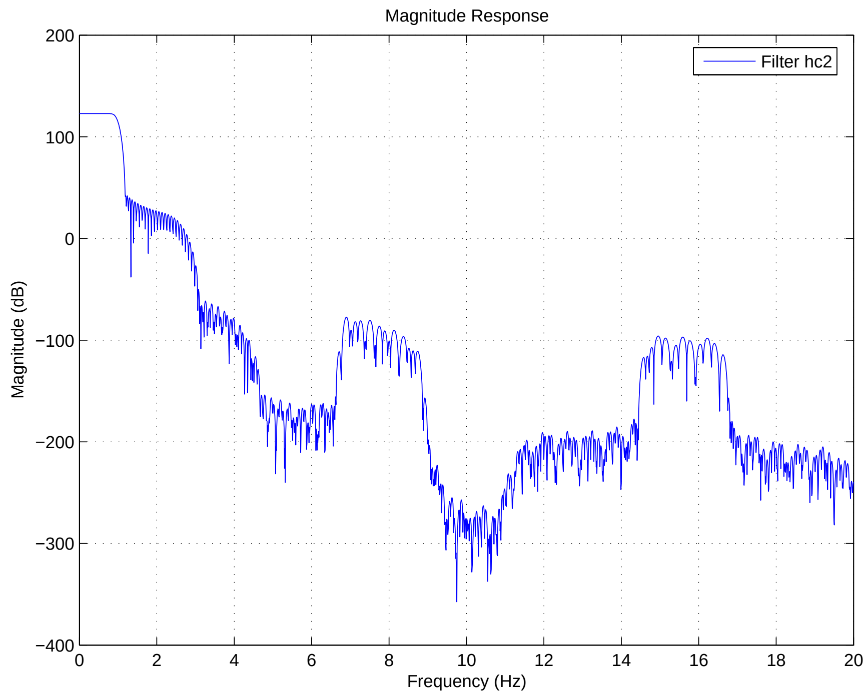
```
h(5) = mfilt.firsrc(2,3);
h(6) = mfilt.cicdecim(13);
h(7) = mfilt.firdecim(5);
hc2 = dfilt.cascade(h)
```

```
hc2 =

         FilterStructure: Cascade
                Stage(1): Direct-Form FIR Polyphase Interpolator
                Stage(2): Direct-Form FIR Polyphase Interpolator
                Stage(3): Direct-Form FIR Polyphase Interpolator
                Stage(4): Cascaded Integrator-Comb Interpolator
               Stage(5): Direct-Form FIR Polyphase Sample-Rate Converter
                Stage(6): Cascaded Integrator-Comb Decimator
                Stage(7): Direct-Form FIR Polyphase Decimator
        PersistentMemory: false
```

As you did in the preceding section, compute the frequency response between 0 and 200 Hz using Fs equal to 1000 Hz.

```
[hf,f] = freqz(hc2,0:1e-2:20,1000);
plot(f,20*log10(abs(hf)))
```

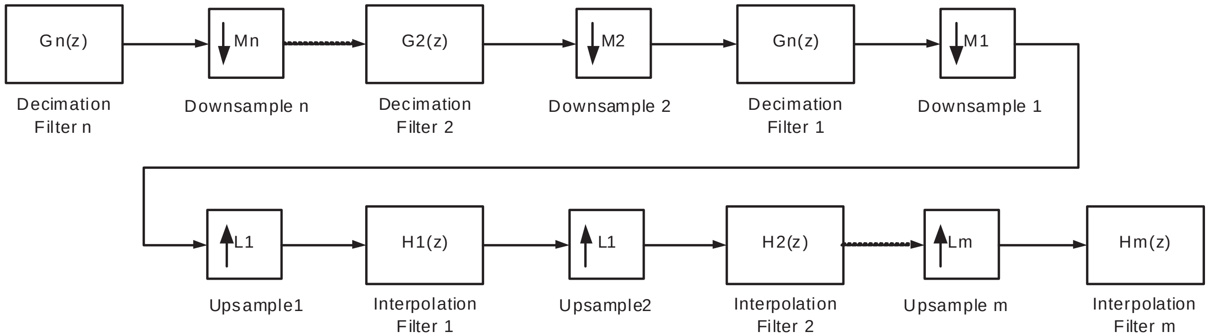
The figure show the frequency response of `hc2`, the result of adding decimators and a rate changing filter to `hc`.



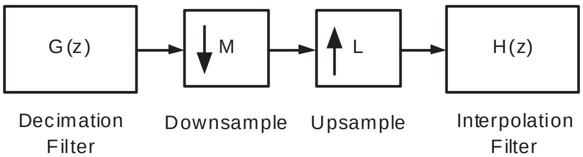
Analyzing Other Multistage Configurations

In addition to the multistage filters `hc` and `hc2` shown, the toolbox lets you analyze multistage filters where decimation occurs prior to interpolation, provided the overall filter interpolation and decimation factors are the same. Notice that this does not necessarily mean that there is an equal number of decimation and interpolation stages.

One multistage structure that you could analyze in the toolbox is this one:



In this case, the analysis tools develop two equivalent filters as shown in the next figure, where M = M1*M2*…Mn = L1*L2*…Lm = L.



Because the overall interpolation factor L is equal to the overall decimation factor M, the equivalent filters are operating at the same rate.

As before, when you provide a sampling frequency for the analysis, the tools assume that the supplied rate is the rate at which both filters are operating. For this case, this would also be equal to the input and output rate for these filters.

To see a demonstration about this type of analysis, where you are analyzing multistage, multirate filters, refer to “Multirate Multistage FIR Filter Design” in the Filter Design Toolbox demos.

Audio Example—Audio Sample Rate Conversion

For a more concrete application of multirate filters, this section illustrates multirate filters that you might use to perform sample rate conversion on different audio formats. During this section, you create each of the following:

- FIR sample rate conversion filter
- FIR fractional interpolator
- FIR fractional decimator

To do these tasks, this section contains the following topics:

Creating the Multirate Filters

All sample rate conversion filters are created in the same way. You first specify the interpolation factor L, and then the decimation factor M, and finally the FIR filter coefficients. Note that L and M must be relatively prime. Two integers a and b are relatively prime when they do not share any common factors. For example, 21 and 54 are not relatively prime—7 is a factor common to both. 14 and 25 are relatively prime. When L and M are not relatively prime, they are converted to relatively prime factors and you get a warning in MATLAB.

If you do not provide filter coefficients when you construct your filter, the filter design process returns a lowpass filter with a cutoff frequency of π/`max`(L,M) and a gain of L in the passband.

Begin by designing a default rate change filter `hm1`.

```
hm1 = mfilt.firsrc(4,3); % Default sample rate change filter.
hm2 = mfilt.firfracinterp(8,6);

Warning: L and M are not relatively prime. Converting ratio 8/6
to 4/3.
The cutoff frequency of the filter should be approximately pi/4.
```

MATLAB notifies you that the factors 8 and 6 do not meet the relatively prime specification and reduces each by the common factor 2. Then MATLAB designs the filter.

Decreasing the Sample Rate by a Fractional Factor

Suppose you are converting an audio signal recorded at 48kHz to 32kHz for broadcasting. Consider the following audio sample recorded at 48kHz (Copyright 2002 FingerBomb) by loading the sample into MATLAB and then playing the file.

```
load audio48;
```

To listen to the original 48 kHz signal, you can use an audio player object in MATLAB.

```
p48 = audioplayer(signal48kHz,Fs48); % Create audio player
                                     % object.
play(p48); % Play the track. Use stop(p48) to stop play.
```

In all, the track lasts about 9 seconds.

Constructing the Fractional Decimator

Reducing the 48kHz sample rate for the signal to 32 kHz requires decimating the signal by two-thirds (discard one sample out of every three). Decimation by two-thirds is an example of fractional decimation.

The interpolation factor for this case is 2 and the decimation factor is 3. You can use a fractional decimator to achieve this sample rate modification. To avoid making this example more complicated, use the default filter that `mfilt.firfracdecim` designs for now.

```
hfd = mfilt.firfracdecim(2,3); % Use default decimator filter.
hfd
```

```
hfd =

        FilterStructure: 'Direct-Form FIR Polyphase Fractional Decimator'
              Numerator: [1x72 double]
      RateChangeFactors: [2 3]
     PersistentMemory: false
                 States: [36x1 double]
    NumSamplesProcessed: 0
```

You could also use your own lowpass filter by specifying the coefficients as a third input argument

```
hfd = mfilt.firfracdecim(l,m,coeffs)
```

where `coeffs` contains the FIR filter coefficients to use.

Filtering to Change the Sample Rate

To use the fractional decimator `hfd` to convert the sample rate of the signal, you invoke the `filter` method with the signal `signal48kHz` and `hfd`.

```
s32 = filter(hfd,signal48kHz);
```

Once again, you can use an `audioplayer` object to listen to the down-converted signal.

```
p32 = audioplayer(s32,32e3); % Create a new audio player.
play(p32);
```

Comparing the Resampled Signals

You now have about 9 seconds of audio. Of course, you can find the exact length in seconds from

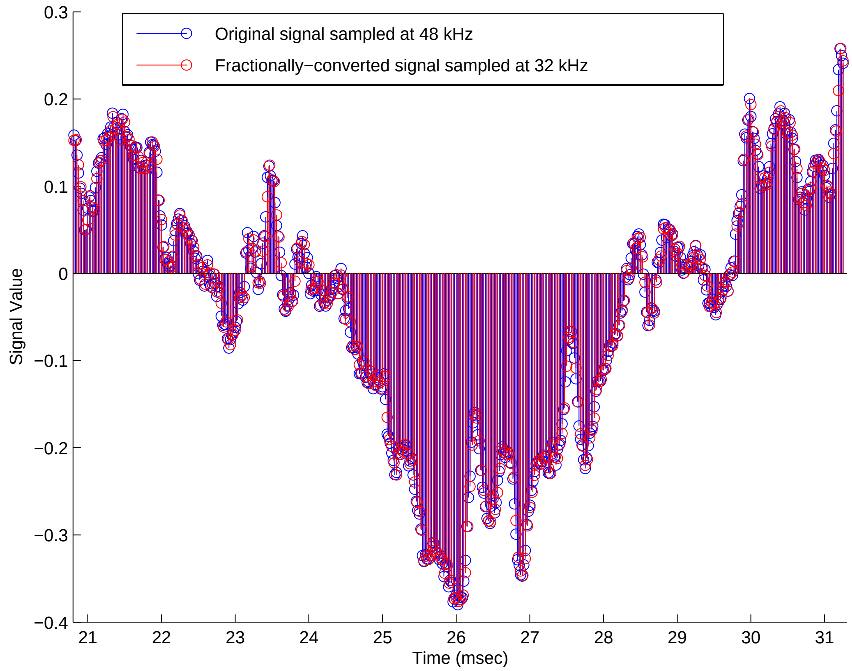
```
length(signal48kHz)/Fs48 % Or length(s32)/32e3.

ans =

     8.9634
```

For clarity, you should overlay the two signals on a plot to compare them. Because the audio track contains some 430,000 samples, you show only a small signal segment. You also have to account for the delay the filter introduces in the 32 kHz signal (the transient response mentioned earlier). Filter `hfd` has a group-delay of 36 samples. Since it is running three times faster than the 32 kHz signal, the delay is equivalent to 12 low speed samples.

Note that there are three samples of the 48 kHz signal for every two samples of the 32 kHz signal. Now to pick some audio data samples to display.

To make the overlay work, you need the same starting point for each signal. The following code finds common points for the 48 kHz and 32 kHz signals and displays them in a stem plot.

```
xindx = 999:1500; % 0.0105 seconds of audio at 48 kHz.
figure
stem(xindx/Fs48*1e3,signal48kHz(xindx));
hold on;
xindx2 = xindx(1)*32e3/48e3:xindx(end)*32e3/48e3; % Find the same
                                                   % start and
                                                   % stop times.

stem(xindx2/32,s32(xindx2+12),'r'); % Add 12 samples to account
                                    % for filter transient delay.
```



Increasing the Sample Rate by a Fractional Factor

You can convert the broadcast quality signal at 32 kHz back to 48 kHz with a fractional interpolator, perhaps to store it on a digital audio tape (DAT). Moving from 32 to 48 requires upsampling by 50 percent, achieved using an interpolation factor of 3 and decimation by 2. Again, you use the fractional FIR interpolator.

```
hfi = mfilt.firfracinterp(3,2);
s48 = filter(hfi,s32);
```

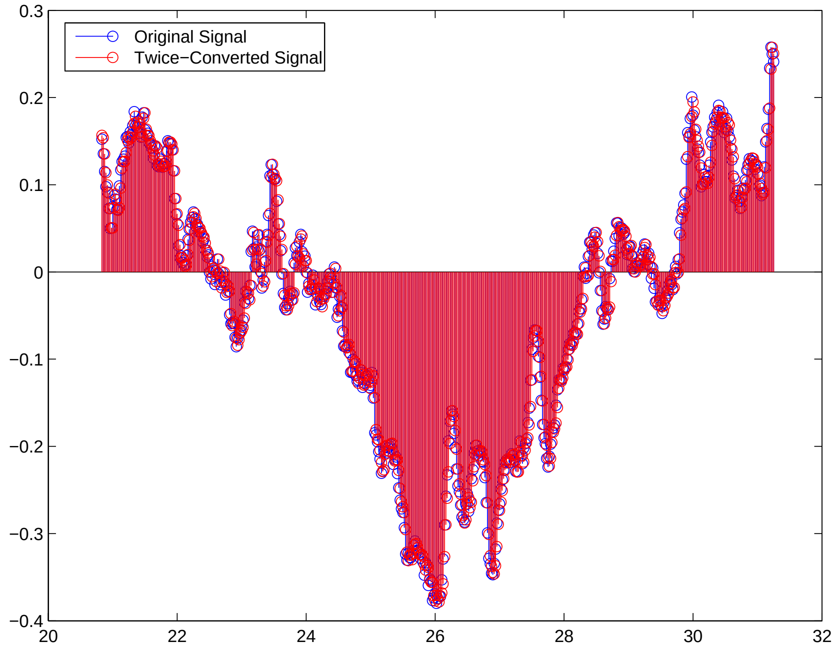
Listening to the up-converted audio might be interesting. Use an audio player again.

```
ps48 = audioplayer(s48,Fs48);
play(ps48);
```

Plotting the Original Signal and the Reconverted Signal

To compare both 48 kHz signals—the original and the twice-converted signal, you must account for the delay introduced by both the fractional decimation and the fractional interpolation processes when you converted the signal down to 32 kHz and back to 48 kHz. In the stem plot shown here, notice that most of the reconverted signal samples have moved slightly from where they were originally. This is distortion introduced by converting down to 32 kHz by decimation and then converting back up to 48 kHz by interpolation.

```
figure;
xindx = 1000:1500;
stem(xindx/Fs48*1e3,signal48kHz(xindx));
hold on;
stem(xindx/Fs48*1e3,s48(1037:1537),'r'); % Account for the
                                         % process-induced
                                         % delays.
```



Different filters achieve different results. You used the default filters which do not optimize the output.

Converting from 48 kHz to 44.1 kHz

To convert from studio quality audio at 48 kHz to CD quality audio, 44.1 kHz, you would use a multirate filter better suited for this ratio change (interpolation factor of 147, decimation factor of 160; decimation by 1.088). To avoid the startup delay (latency) introduced by the filter, preload half of the filter states with the beginning of the signal. Doing this step compensates for the delay caused by filtering and decimation. For this rate change, you use the FIR sample rate change multirate filter—`firsrc`.

```
hsrc = mfilt.firsrc(147,160) % Use default filter coefficients.
hsrc =
```

```
      FilterStructure: 'Direct-Form FIR Polyphase Sample-Rate Converter'
            Numerator: [1x3840 double]
    RateChangeFactors: [147 160]
     PersistentMemory: true
               States: [26x1 double]
  NumSamplesProcessed: 430080

hsrc.persistentmemory = 'true';    % Allows you to set the states
                                   % to eliminate delay.
hsrc.States(13:-1:1) = signal48kHz(1:13); % Preload the states.
s441 = filter(hsrc,signal48kHz(14:end)); % This takes a few
                                         % seconds.
```

Again, you can play the down-converted signal at 44.1 kHz with a MATLAB audio player.

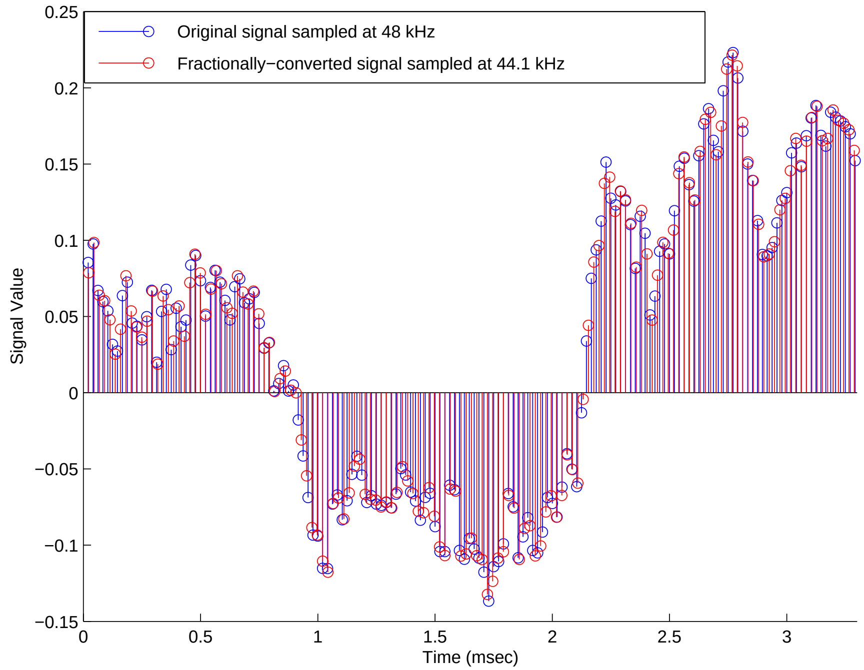
```
p441 = audioplayer(s441,44.1e3);
play(p441);
```

When you are doing sample-rate conversion with large values of L or M, as you are in this case where L=147 and M=160, using the `mfilt.firsrc` structure is the most effective approach. Other possible fractional rate change structures, such as `mfilt.firfracinterp` (where L > M) or `mfilt.firfracdecim` (where L < M) may have prohibitively large memory requirements for applications that require large rate changes.

Plotting the 48 kHz Signal and the 44.1 kHz Signal

Now compare segments of the two signals graphically. In this case you can verify visually in the stem plot shown that the filter does not introduce delay since you compensated for its group delay by preloading the states.

```
figure
xindx = 1:160;
stem(xindx/Fs48*1e3,signal48kHz(xindx));
hold on
xindx2 = 1:147;
stem(xindx2/44.1,s441(xindx2),'r');
```


Original signal sampled at 48 kHz
Fractionally−converted signal sampled at 44.1 kHz
Signal Value
Time (msec)
0.25
0.2
0.15
0.1
0.05
0
−0.05
−0.1
−0.15
0
0.5
1
1.5
2
2.5
3

6

Digital Frequency Transformations

Introduction

Converting existing FIR or IIR filter designs to a modified IIR form is often done using allpass frequency transformations. Although the resulting designs can be considerably more expensive in terms of dimensionality than the prototype (original) filter, their ease of use in fixed or variable applications is a big advantage.

The general idea of the frequency transformation is to take an existing prototype filter and produce another filter from it that retains some of the characteristics of the prototype, in the frequency domain. Transformation functions achieve this by replacing each delaying element of the prototype filter with an allpass filter carefully designed to have a prescribed phase characteristic for achieving the modifications requested by the designer.

This tutorial gives an overview and interpretation of the frequency transformations, and describes the range of transformations available to the toolbox user. To aid this purpose the tutorial has been arranged into three sections:

- "Definition of the Problem" on page 6-3 introduces the frequency transformation concept and provides its mathematical and intuitive interpretations.
- "Frequency Transformations for Real Filters" on page 6-11 describes the real frequency transformations available in the toolbox. Such transformations start from a real prototype filter and return a real target filter.
- "Frequency Transformations for Complex Filters" on page 6-26 describes complex frequency transformations available in the toolbox. Such transformations start from the any real or complex prototype filter and return a complex target filter.

Definition of the Problem

The basic form of mapping in common use is

$$H_T(z) = H_o[H_A(z)]$$

The $H_A(z)$ is an Nth-order allpass mapping filter given by

$$H_A(z) = S\frac{\sum_{i=0}^{N} \alpha_i z^{-i}}{\sum_{i=0}^{N} \alpha_i z^{-N+i}} = \frac{N_A(z)}{D_A(z)}$$

$$\alpha_0 = 1$$

where

$H_o(z)$— Transfer function of the prototype filter

$H_A(z)$— Transfer function of the allpass mapping filter

$H_T(z)$— Transfer function of the target filter

Let's look at a simple example of the transformation given by

$$H_T(z) = H_o(-z)$$

The target filter has its poles and zeroes flipped across the origin of the real and imaginary axes. For the real filter prototype, it gives a mirror effect against 0.5, which means that lowpass $H_o(z)$ gives rise to a real highpass $H_T(z)$. This is shown in the following figure for the prototype filter designed as a third-order halfband elliptic filter.

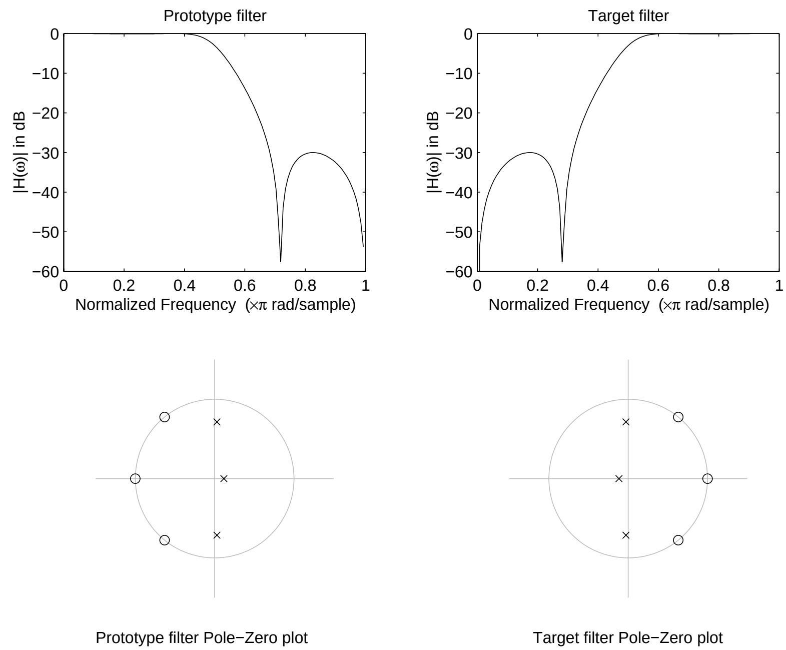


Figure 6-1: Example of a Simple Mirror Transformation

The choice of an allpass filter to provide the frequency mapping is necessary to provide the frequency translation of the prototype filter frequency response to the target filter by changing the frequency position of the features from the prototype filter without affecting the overall shape of the filter response.

The phase response of the mapping filter normalized to π can be interpreted as a translation function:

$$H(w_{new}) = \omega_{old}$$

The graphical interpretation of the frequency transformation is shown in the figure below. The complex multiband transformation takes a real lowpass filter and converts it into a number of passbands around the unit circle.

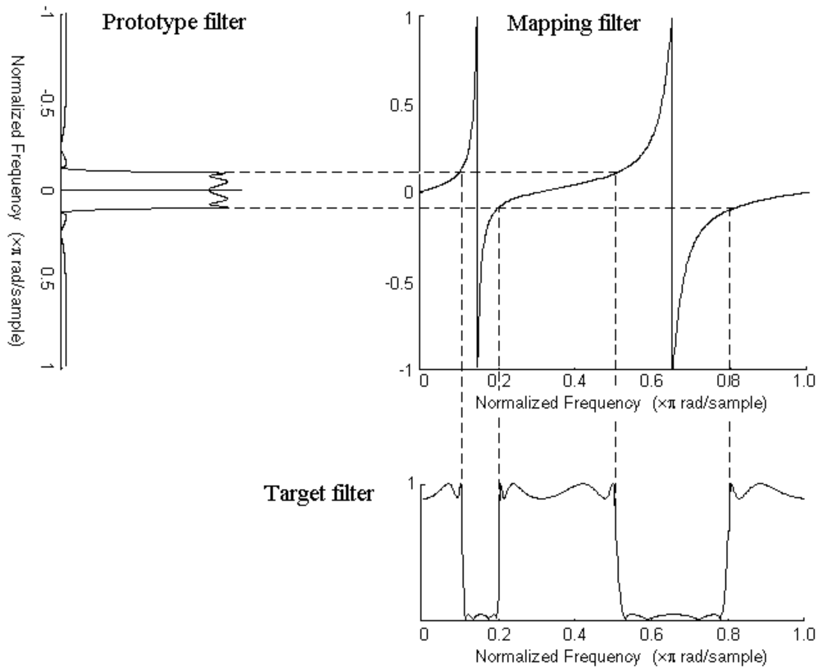


Figure 6-2: Graphical Interpretation of the Mapping Process

Most of the frequency transformations are based on the second-order allpass mapping filter:

$$H_A(z) = \pm\frac{1 + \alpha_1 z^{-1} + \alpha_2 z^{-2}}{\alpha_2 + \alpha_1 z^{-1} + z^{-2}}$$

The two degrees of freedom provided by α_1 and α_2 choices are not fully used by the usual restrictive set of “flat-top” classical mappings like lowpass to bandpass. Instead, any two transfer function features can be migrated to (almost) any two other frequency locations if α_1 and α_2 are chosen so as to keep the poles of $H_A(z)$ strictly outside the unit circle (since $H_A(z)$ is substituted for z in the prototype transfer function). Moreover, as first pointed out by Constantinides, the selection of the outside sign influences whether the original feature at zero can be moved (the minus sign, a condition known as

“DC mobility”) or whether the Nyquist frequency can be migrated (the “Nyquist mobility” case arising when the leading sign is positive).

All the transformations forming the package are explained in the next sections of the tutorial. They are separated into those operating on real filters and those generating or working with complex filters. The choice of transformation ranges from standard Constantinides first and second-order ones [16][17] up to the real multiband filter by Mullis and Franchitti [18], and the complex multiband filter and real/complex multipoint ones by Krukowski, Cain and Kale [19].

Selecting Features Subject to Transformation

Choosing the appropriate frequency transformation for achieving the required effect and the correct features of the prototype filter is very important and needs careful consideration. It is not advisable to use a first-order transformation for controlling more than one feature. The mapping filter will not give enough flexibility. It is also not good to use higher order transformation just to change the cutoff frequency of the lowpass filter. The increase of the filter order would be too big, without considering the additional replica of the prototype filter that may be created in undesired places.

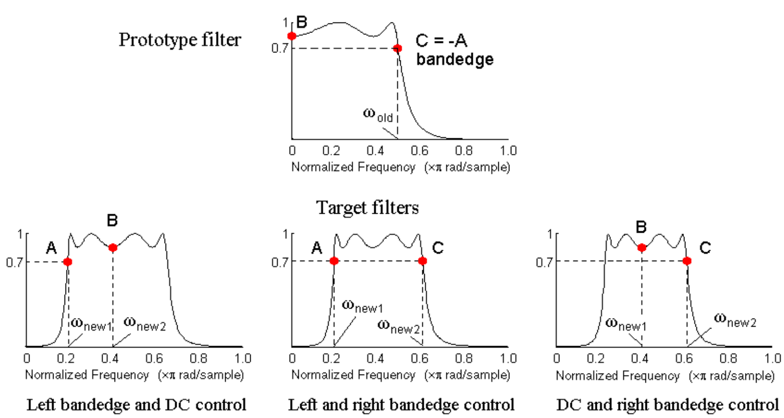


Figure 6-3: Feature Selection for Real Lowpass to Bandpass Transformation

To illustrate the idea, the second-order real multipoint transformation was applied three times to the same elliptic halfband filter in order to make it into a bandpass filter. In each of the three cases, two different features of the prototype filter were selected in order to obtain a bandpass filter with passband ranging from 0.25 to 0.75. The position of the DC feature was not important, but it would be advantageous if it were in the middle between the edges of the passband in the target filter. In the first case the selected features were the left and the right band edges of the lowpass filter passband, in the second case they were the left band edge and the DC, in the third case they were DC and the right band edge.

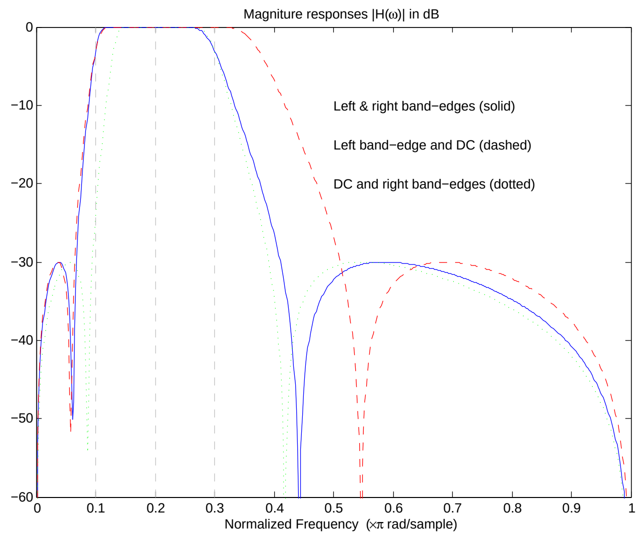


Figure 6-4: Result of choosing different features

The results of all three approaches are completely different. For each of them only the selected features were positioned precisely where they were required. In the first case the DC is moved toward the left passband edge just like all the other features close to the left edge being squeezed there. In the second case the right passband edge was pushed way out of the expected target as the precise position of DC was required. In the third case the left passband edge was pulled toward the DC in order to position it at the correct frequency. The conclusion is that if only the DC can be anywhere in the passband, the edges of the passband should have been selected for the transformation. For most of the cases requiring the positioning of passbands and stopbands, designers should always choose the position of the edges of the prototype filter in order to make sure that they get the edges of the target filter in the correct places. Frequency responses for the three cases considered are shown in the figure. The prototype filter was a third-order elliptic lowpass filter with cutoff frequency at 0.5.

The MATLAB code used to generate the figure is given here.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

In the example the requirements are set to create a real bandpass filter with passband edges at 0.1 and 0.3 out of the real lowpass filter having the cutoff frequency at 0.5. This is attempted in three different ways. In the first approach both edges of the passband are selected, in the second approach the left edge of the passband and the DC are chosen, while in the third approach the DC and the right edge of the passband are taken:

```
[num1,den1] = iirlp2xn(b, a, [-0.5, 0.5], [0.1, 0.3]);
[num2,den2] = iirlp2xn(b, a, [-0.5, 0.0], [0.1, 0.2]);
[num3,den3] = iirlp2xn(b, a, [ 0.0, 0.5], [0.2, 0.3]);
```

Mapping from Prototype Filter to Target Filter

In general the frequency mapping converts the prototype filter, $H_0(z)$, to the target filter, $H_T(z)$, using the N_Ath-order allpass filter, $H_A(z)$. The general form of the allpass mapping filter is given in Equation . The frequency mapping is a mathematical operation that replaces each delayer of the prototype filter with an allpass filter. There are two ways of performing such mapping. The choice of the approach is dependent on how prototype and target filters are represented.

When the Nth-order prototype filter is given with pole-zero form

$$H_o(z) = \frac{\sum_{i=1}^{N} (z - z_i)}{\sum_{i=1}^{N} (z - p_i)}$$

the mapping will replace each pole, p_i, and each zero, z_i, with a number of poles and zeros equal to the order of the allpass mapping filter:

$$H_o(z) = \frac{\sum_{i=1} \left(S \sum_{k=0} \alpha_k z^k - z_i \cdot \sum_{k=0} \alpha_k z^{N-k} \right)}{\sum^{N} \left(S \sum^{N-1} \alpha_k z^k - p_i \cdot \sum^{N-1} \alpha_k z^{N-k} \right)}$$

The root finding needs to be used on the bracketed expressions in order to find the poles and zeros of the target filter.

When the prototype filter is described in the numerator-denominator form:

$$H_T(z) = \left. \frac{\beta_0 z^N + \beta_1 z^{N-1} + \ldots + \beta_N}{\alpha_0 z^N + \alpha_1 z^{N-1} + \ldots + \alpha_N} \right|_{z = H_A(z)}$$

Then the mapping process will require a number of convolutions in order to calculate the numerator and denominator of the target filter:

$$H_T(z) = \frac{\beta_1 N_A(z)^N + \beta_2 N_A(z)^{N-1} D_A(z) + \ldots + \beta_N D_A(z)^N}{\beta_1 N_A(z)^N + \beta_2 N_A(z)^{N-1} D_A(z) + \ldots + \beta_N D_A(z)^N}$$

For each coefficient α_i and β_i of the prototype filter the N_Ath-order polynomials must be convolved N times. Such approach may cause rounding errors for large prototype filters and/or high order mapping filters. In such a case the user should consider the alternative of doing the mapping using via poles and zeros.

Summary of Frequency Transformations

Advantages

- Most frequency transformations are described by closed-form solutions or can be calculated from the set of linear equations.
- They give predictable and familiar results.
- Ripple heights from the prototype filter are preserved in the target filter.
- They are architecturally appealing for variable and adaptive filters.

Disadvantages

- There are cases when using optimization methods to design the required filter gives better results.
- High-order transformations increase the dimensionality of the target filter, which may give expensive final results.
- Starting from fresh designs helps avoid locked-in compromises.

Frequency Transformations for Real Filters

This section discusses real frequency transformations that take the real lowpass prototype filter and convert it into a different real target filter. The target filter has its frequency response modified in respect to the frequency response of the prototype filter according to the characteristic of the applied frequency transformation:

Real Frequency Shift

Real frequency shift transformation uses a second-order allpass mapping filter. It performs an exact mapping of one selected feature of the frequency response into its new location, additionally moving both the Nyquist and DC features. This effectively moves the whole response of the lowpass filter by the distance specified by the selection of the feature from the prototype filter and the target filter. As a real transformation, it works in a similar way for positive and negative frequencies.

$$H_A(z) = z^{-1} \cdot \frac{1 - \alpha z^{-1}}{\alpha - z^{-1}}$$

with α given by

$$\alpha = \begin{cases} \dfrac{\cos\frac{\pi}{2}(\omega_{old} - 2\omega_{new})}{\cos\frac{\pi}{2}\omega_{old}} & \text{for} \quad \left|\cos\frac{\pi}{2}(\omega_{old} - 2\omega_{new})\right| < 1 \\ \dfrac{\sin\frac{\pi}{2}(\omega_{old} - 2\omega_{new})}{\sin\frac{\pi}{2}\omega_{old}} & \text{otherwise} \end{cases}$$

where

ω_{old} — Frequency location of the selected feature in the prototype filter

ω_{new} — Position of the feature originally at ω_{old} in the target filter

The example below shows how this transformation can be used to move the response of the prototype lowpass filter in either direction. Please note that because the target filter must also be real, the response of the target filter will inherently be disturbed at frequencies close to Nyquist and close to DC. Here is the MATLAB code for generating the example in the figure.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example transformation moves the feature originally at 0.5 to 0.9:

```
[num,den] = iirshift(b, a, 0.5, 0.9);
```

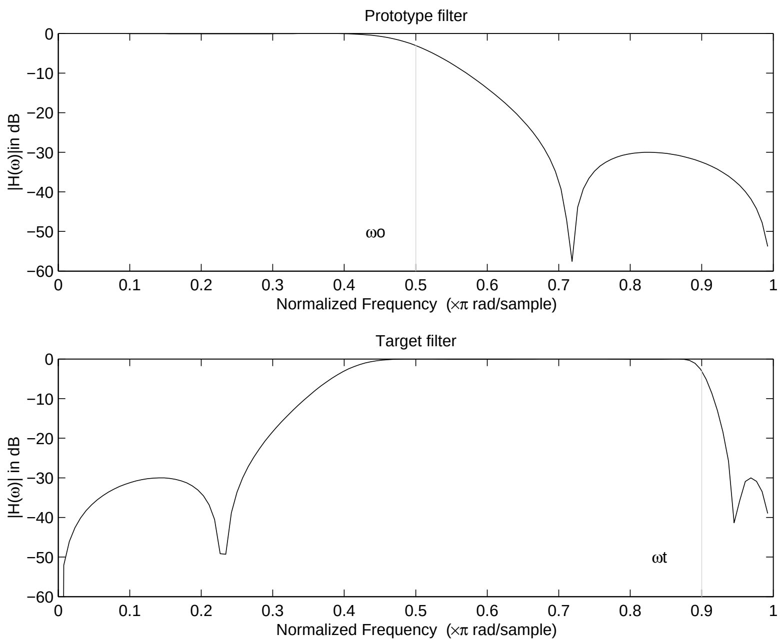


Figure 6-5: Example of Real Frequency Shift Mapping

Real Lowpass to Real Lowpass

Real lowpass filter to real lowpass filter transformation uses a first-order allpass mapping filter. It performs an exact mapping of one feature of the frequency response into the new location keeping DC and Nyquist features fixed. As a real transformation, it works in a similar way for positive and negative frequencies. It is important to mention that using first-order mapping ensures that the order of the filter after the transformation is the same as it was originally.

$$H_A(z) = -\left(\frac{1-\alpha z^{-1}}{\alpha - z^{-1}}\right)$$

with α given by

$$\alpha = \frac{\sin\frac{\pi}{2}(w_{old} - w_{new})}{\sin\frac{\pi}{2}(w_{old} + w_{new})}$$

where

ω_{old} — Frequency location of the selected feature in the prototype filter

ω_{new} — Frequency location of the same feature in the target filter

The example below shows how to modify the cutoff frequency of the prototype filter. The MATLAB code for this example is shown in the figure below.

The prototype filter is a halfband elliptic, real, third-order filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The cutoff frequency moves from 0.5 to 0.75:

```
[num,den] = iirlp2lp(b, a, 0.5, 0.75);
```

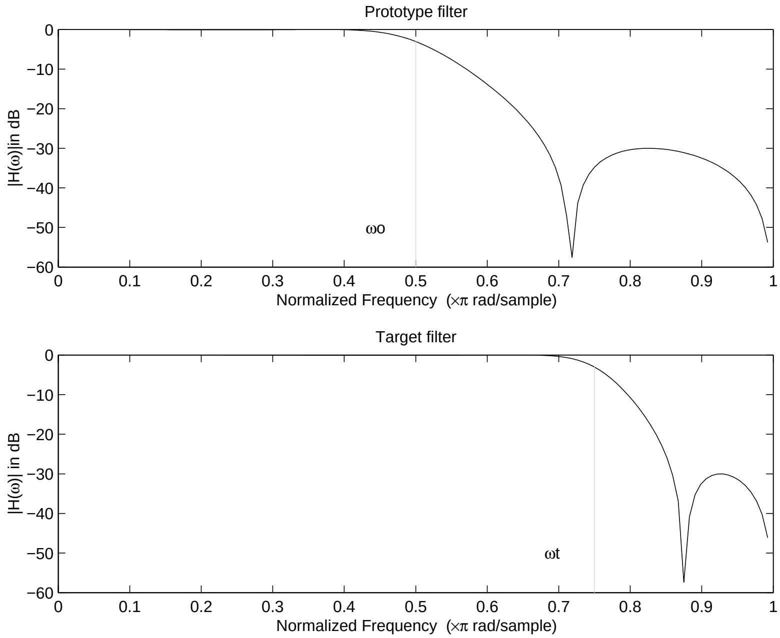


Figure 6-6: Example of Real Lowpass to Real Lowpass Mapping

Real Lowpass to Real Highpass

Real lowpass filter to real highpass filter transformation uses a first-order allpass mapping filter. It performs an exact mapping of one feature of the frequency response into the new location additionally swapping DC and Nyquist features. As a real transformation, it works in a similar way for positive and negative frequencies. Just like in the previous transformation because of using a first-order mapping, the order of the filter before and after the transformation is the same.

$$H_A(z) = -\left(\frac{1 + \alpha z^{-1}}{\alpha + z^{-1}}\right)$$

with α given by

$$\alpha = -\left(\frac{\cos\frac{\pi}{2}(w_{old} + w_{new})}{\cos\frac{\pi}{2}(w_{old} - w_{new})}\right)$$

where

ω_{old} — Frequency location of the selected feature in the prototype filter

ω_{new} — Frequency location of the same feature in the target filter

The example below shows how to convert the lowpass filter into a highpass filter with arbitrarily chosen cutoff frequency. In the MATLAB code below, the lowpass filter is converted into a highpass with cutoff frequency shifted from0.5 to 0.75. Results are shown in the figure.

The prototype filter is a halfband elliptic, real, third-order filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example moves the cutoff frequency from 0.5 to 0.75:

```
[num,den] = iirlp2lp(b, a, 0.5, 0.75);
```

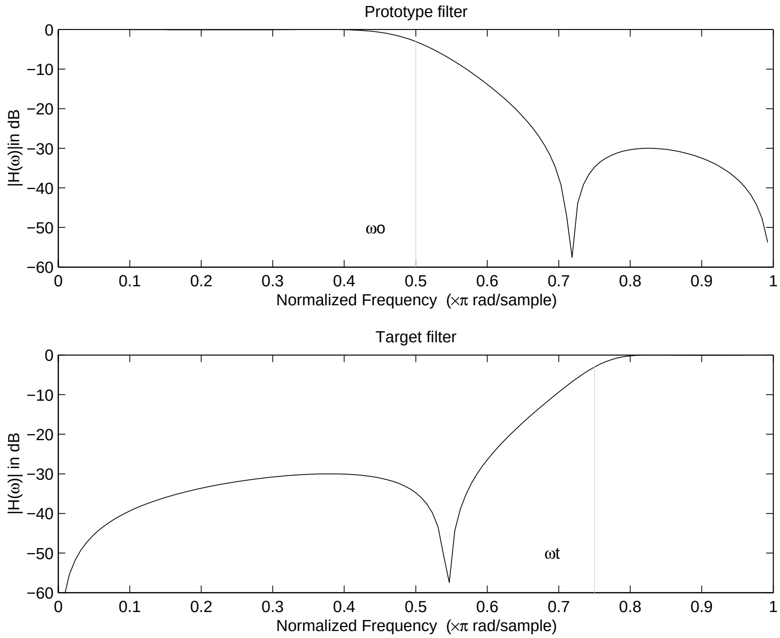


Figure 6-7: Example of Real Lowpass to Real Highpass Mapping

Real Lowpass to Real Bandpass

Real lowpass filter to real bandpass filter transformation uses a second-order allpass mapping filter. It performs an exact mapping of two features of the frequency response into their new location additionally moving a DC feature and keeping the Nyquist feature fixed. As a real transformation, it works in a similar way for positive and negative frequencies.

$$H_A(z) = -\left(\frac{1-\beta(1+\alpha)z^{-1}-\alpha z^{-2}}{\alpha-\beta(1+\alpha)z^{-1}+z^{-2}}\right)$$

with α and β given by

$$\alpha = \frac{\sin\frac{\pi}{4}(2w_{old} - w_{new,2} + w_{new,1})}{\sin\frac{\pi}{4}(2w_{old} + w_{new,2} - w_{new,1})}$$

$$\beta = \cos\frac{\pi}{2}(w_{new,1} + w_{new,2})$$

where

ω_{old} — Frequency location of the selected feature in the prototype filter

$\omega_{new,1}$ — Position of the feature originally at $(-\omega_{old})$ in the target filter

$\omega_{new,2}$ — Position of the feature originally at $(+\omega_{old})$ in the target filter

The example below shows how to move the response of the prototype lowpass filter in either direction. Please note that because the target filter must also be real, the response of the target filter will inherently be disturbed at frequencies close to Nyquist and close to DC. Here is the MATLAB code for generating the example in the figure.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example transformation creates the passband between 0.5 and 0.75:

```
[num,den] = iirlp2bp(b, a, 0.5, [0.5, 0.75]);
```

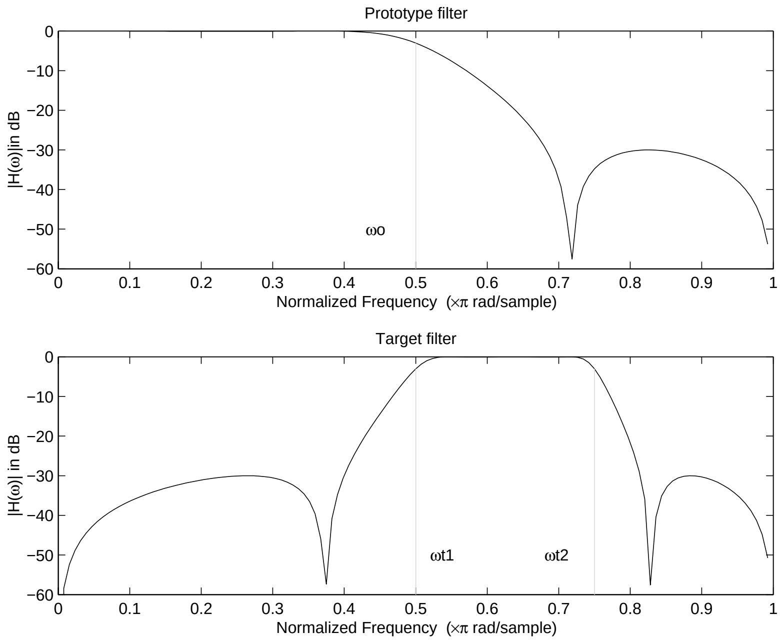


Figure 6-8: Example of Real Lowpass to Real Bandpass Mapping

Real Lowpass to Real Bandstop

Real lowpass filter to real bandstop filter transformation uses a second-order allpass mapping filter. It performs an exact mapping of two features of the frequency response into their new location additionally moving a Nyquist feature and keeping the DC feature fixed. This effectively creates a stopband between the selected frequency locations in the target filter. As a real transformation, it works in a similar way for positive and negative frequencies.

$$H_A(z) = \frac{1 - \beta(1+\alpha)z^{-1} + \alpha z^{-2}}{\alpha - \beta(1+\alpha)z^{-1} + z^{-2}}$$

with α and β given by

$$\alpha = \frac{\cos\frac{\pi}{4}(2w_{old} + w_{new,2} - w_{new,1})}{\cos\frac{\pi}{4}(2w_{old} - w_{new,2} + w_{new,1})}$$

$$\beta = \cos\frac{\pi}{2}(w_{new,1} + w_{new,2})$$

where

ω_{old} — Frequency location of the selected feature in the prototype filter

$\omega_{new,1}$ — Position of the feature originally at $(-\omega_{old})$ in the target filter

$\omega_{new,2}$ — Position of the feature originally at $(+\omega_{old})$ in the target filter

The example below shows how this transformation can be used to convert the prototype lowpass filter with cutoff frequency at 0.5 into a real bandstop filter with the same passband and stopband ripple structure and stopband positioned between 0.5 and 0.75. Here is the MATLAB code for generating the example in the figure.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example transformation creates a stopband from 0.5 to 0.75:

```
[num,den] = iirlp2bs(b, a, 0.5, [0.5, 0.75]);
```

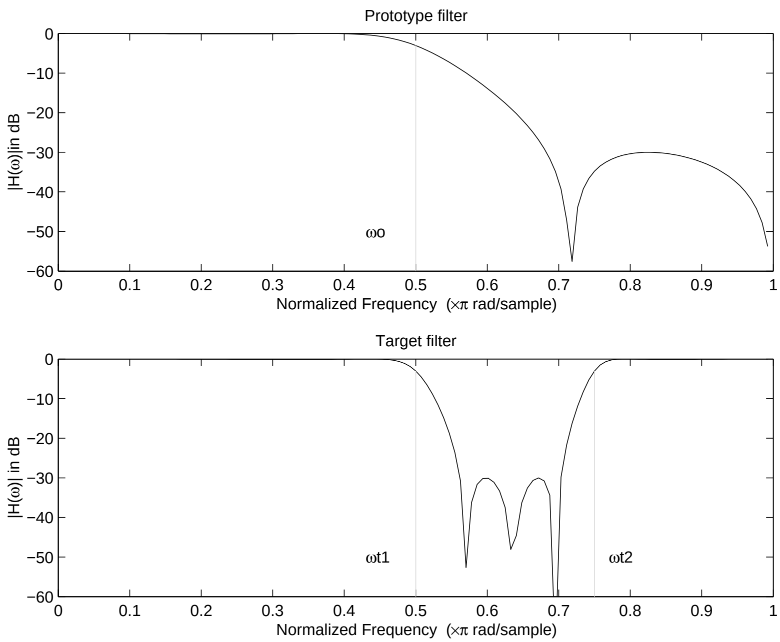


Figure 6-9: Example of Real Lowpass to Real Bandstop Mapping

Real Lowpass to Real Multiband

This high-order transformation performs an exact mapping of one selected feature of the prototype filter frequency response into a number of new locations in the target filter. Its most common use is to convert a real lowpass with predefined passband and stopband ripples into a real multiband filter with N arbitrary band edges, where N is the order of the allpass mapping filter.

$$H_A(z) = S\frac{\sum_{i=0}^{N} \alpha_i z^{-i}}{\sum_{i=0}^{N} \alpha_i z^{-N+i}}$$

$$\alpha_0 = 1$$

The coefficients α are given by

$$\begin{cases} \alpha_0 = 1 & k = 1, \ldots, N \\ \alpha_k = -S\dfrac{\sin\frac{\pi}{2}(N\omega_{new} + (-1)^k\omega_{old})}{\sin\frac{\pi}{2}((N-2k)\omega_{new} + (-1)^k\omega_{old})} \end{cases}$$

where

$\omega_{old,k}$ – Frequency location of the first feature in the prototype filter

$\omega_{new,k}$ – Position of the feature originally at $\omega_{old,k}$ in the target filter

The mobility factor, `S`, specifies the mobility or either DC or Nyquist feature:

$$S = \begin{cases} 1 & Nyquist \\ -1 & DC \end{cases}$$

The example below shows how this transformation can be used to convert the prototype lowpass filter with cutoff frequency at 0.5 into a filter having a number of bands positioned at arbitrary edge frequencies 1/5, 2/5, 3/5 and 4/5. Parameter `S` was such that there is a passband at DC. Here is the MATLAB code for generating the figure.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example transformation creates three passbands, from DC to 0.2, from 0.4 to 0.6 and from 0.8 to Nyquist:

```
[num,den] = iirlp2mb(b, a, 0.5, [0.2, 0.4, 0.6, 0.8], `pass');
```

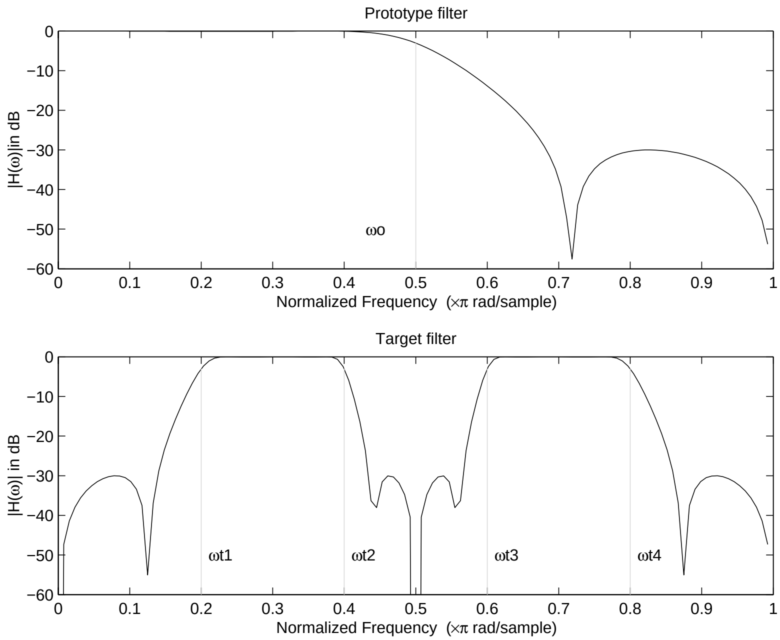


Figure 6-10: Example of Real Lowpass to Real Multiband Mapping

Real Lowpass to Real Multipoint

This high-order frequency transformation performs an exact mapping of a number of selected features of the prototype filter frequency response to their new locations in the target filter. The mapping filter is given by the general IIR polynomial form of the transfer function as given below.

$$H_A(z) = S\frac{\sum_{i=0}^{N} \alpha_i z^{-i}}{\sum_{i=0}^{N} \alpha_i z^{-N+i}}$$

$$\alpha_0 = 1$$

For the *N*th-order multipoint frequency transformation the coefficients α are

$$\begin{cases} \sum_{i=1}^{N} \alpha_{N-i} z_{old,k} \cdot z_{new,k}^{i} - S \cdot z_{new,k}^{N-i} = -z_{old,k} - S \cdot z_{new,k} \\ z_{old,k} = e^{j\pi\omega_{old,k}} \\ z_{new,k} = e^{j\pi\omega_{new,k}} \\ k = 1, \ldots, N \end{cases}$$

where

$\omega_{old,k}$ – Frequency location of the first feature in the prototype filter

$\omega_{new,k}$ – Position of the feature originally at $\omega_{old,k}$ in the target filter

The mobility factor, `S`, specifies the mobility of either DC or Nyquist feature:

$$S = \begin{cases} 1 & Nyquist \\ -1 & DC \end{cases}$$

The example below shows how this transformation can be used to move features of the prototype lowpass filter originally at -0.5 and 0.5 to their new locations at 0.5 and 0.75, effectively changing a position of the filter passband. Here is the MATLAB code for generating the figure.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example transformation creates a passband from 0.5 to 0.75:

```
[num,den] = iirlp2xn(b, a, [-0.5, 0.5], [0.5, 0.75], `pass');
```

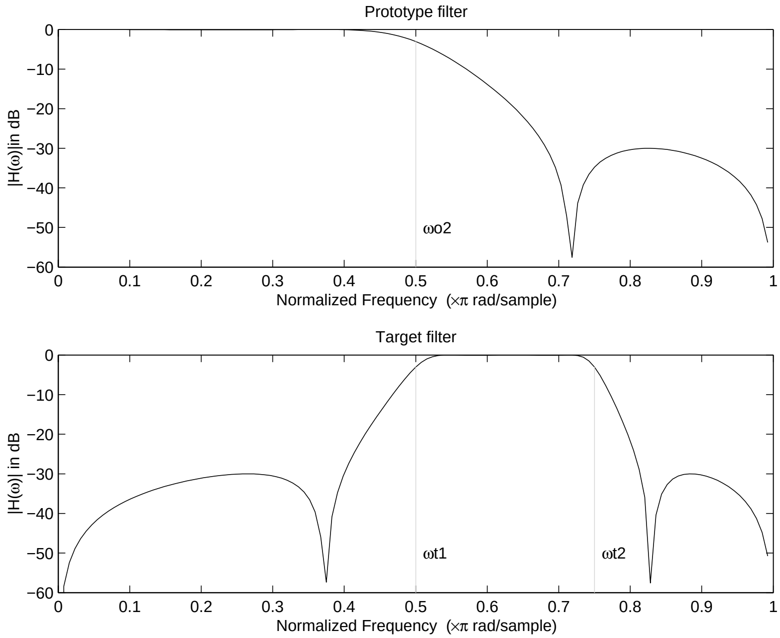


Figure 6-11: Example of Real Lowpass to Real Multipoint Mapping

Frequency Transformations for Complex Filters

This section discusses complex frequency transformation that take the complex prototype filter and convert it into a different complex target filter. The target filter has its frequency response modified in respect to the frequency response of the prototype filter according to the characteristic of the applied frequency transformation from:

- "Complex Frequency Shift" on page 6-26
- "Real Lowpass to Complex Bandpass" on page 6-28
- "Real Lowpass to Complex Bandstop" on page 6-29
- "Real Lowpass to Complex Multiband" on page 6-31
- "Real Lowpass to Complex Multipoint" on page 6-33
- "Complex Bandpass to Complex Bandpass" on page 6-36

Complex Frequency Shift

Complex frequency shift transformation is the simplest first-order transformation that performs an exact mapping of one selected feature of the frequency response into its new location. At the same time it rotates the whole response of the prototype lowpass filter by the distance specified by the selection of the feature from the prototype filter and the target filter.

$$H_A(z) = \alpha z^{-1}$$

with α given by

$$\alpha = e^{j2\pi(\nu_{new} - \nu_{old})}$$

where

ω_{old} — Frequency location of the selected feature in the prototype filter

ω_{new} — Position of the feature originally at ω_{old} in the target filter

A special case of the complex frequency shift is a, so called, Hilbert Transformer. It can be designed by setting the parameter to $|\alpha|=1$, that is

$$\alpha = \begin{cases} 1 & forward \\ -1 & inverse \end{cases}$$

The example below shows how to apply this transformation to rotate the response of the prototype lowpass filter in either direction. Please note that because the transformation can be achieved by a simple phase shift operator, all features of the prototype filter will be moved by the same amount. Here is the MATLAB code for generating the example in the figure.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example transformation moves the feature originally at 0.5 to 0.9:

```
[num,den] = iirshift(b, a, 0.5, 0.9);
```

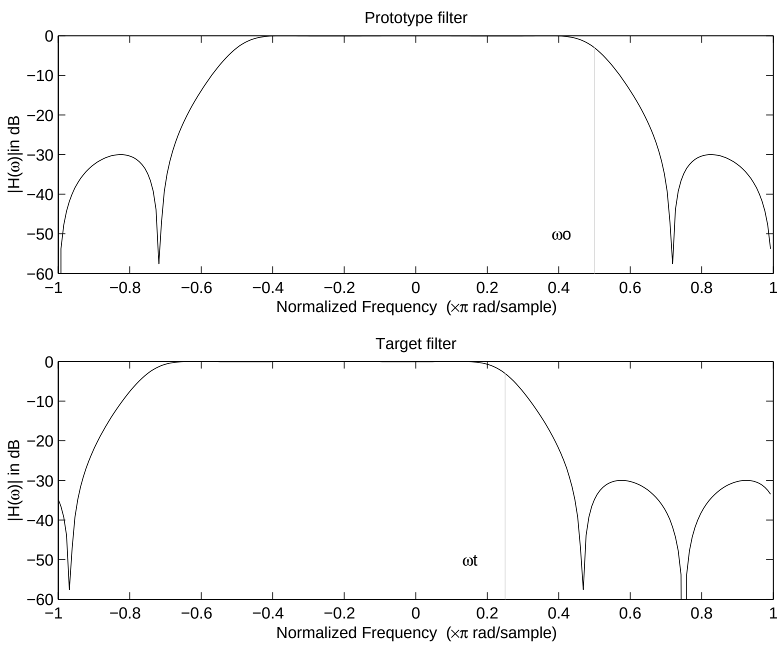


Figure 6-12: Example of Complex Frequency Shift Mapping

Real Lowpass to Complex Bandpass

This first-order transformation performs an exact mapping of one selected feature of the prototype filter frequency response into two new locations in the target filter creating a passband between them. Both Nyquist and DC features can be moved with the rest of the frequency response.

$$H_A(z) = \frac{\beta - \alpha z^{-1}}{z^{-1} - \alpha\beta}$$

with α and β are given by

$$\alpha = \frac{\sin\frac{\pi}{4}(2w_{old} - w_{new,2} + w_{new,1})}{\sin\pi(2w_{old} + w_{new,2} - w_{new,1})}$$

$$\beta = e^{-j\pi(w_{new} - w_{old})}$$

where

ω_{old} — Frequency location of the selected feature in the prototype filter

$\omega_{new,1}$ — Position of the feature originally at $(-\omega_{old})$ in the target filter

$\omega_{new,2}$ — Position of the feature originally at $(+\omega_{old})$ in the target filter

The example below shows the use of such a transformation for converting a real halfband lowpass filter into a complex bandpass filter with band edges at 0.5 and 0.75. Here is the MATLAB code for generating the example in the figure.

The prototype filter is a half band elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The transformation creates a passband from 0.5 to 0.75:

```
[num,den] = iirlp2bpc(b, a, 0.5, [0.5 0.75]);
```

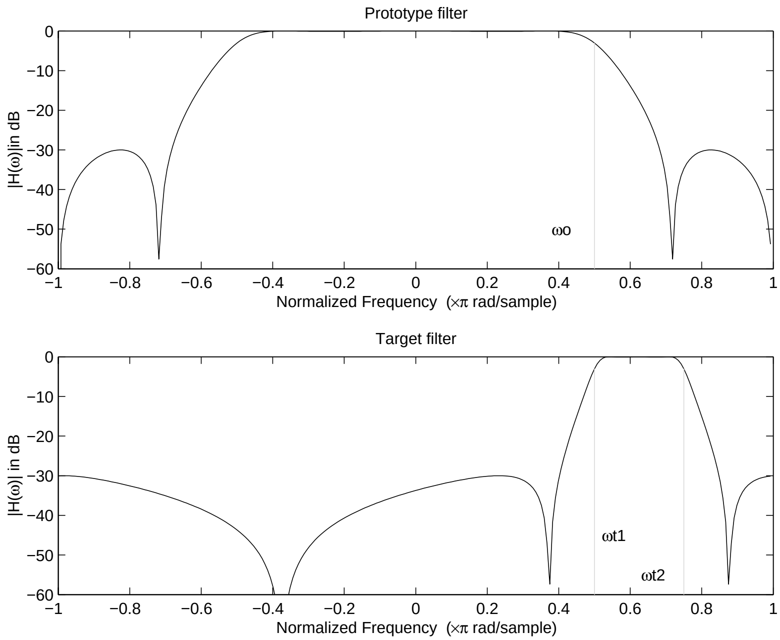


Figure 6-13: Example of Real Lowpass to Complex Bandpass Mapping

Real Lowpass to Complex Bandstop

This first-order transformation performs an exact mapping of one selected feature of the prototype filter frequency response into two new locations in the target filter creating a stopband between them. Both Nyquist and DC features can be moved with the rest of the frequency response.

$$H_A(z) = \frac{\beta - \alpha z^{-1}}{\alpha\beta - z^{-1}}$$

with α and β are given by

$$\alpha = \frac{\cos\pi(2w_{old} + \nu_{new,2} - \nu_{new,1})}{\cos\pi(2w_{old} - \nu_{new,2} + \nu_{new,1})}$$

$$\beta = e^{-j\pi(w_{new} - w_{old})}$$

where

ω_{old} — Frequency location of the selected feature in the prototype filter

$\omega_{new,1}$ — Position of the feature originally at $(-\omega_{old})$ in the target filter

$\omega_{new,2}$ — Position of the feature originally at $(+\omega_{old})$ in the target filter

The example below shows the use of such a transformation for converting a real halfband lowpass filter into a complex bandstop filter with band edges at 0.5 and 0.75. Here is the MATLAB code for generating the example in the figure.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The transformation creates a stopband from 0.5 to 0.75:

```
[num,den] = iirlp2bsc(b, a, 0.5, [0.5 0.75]);
```

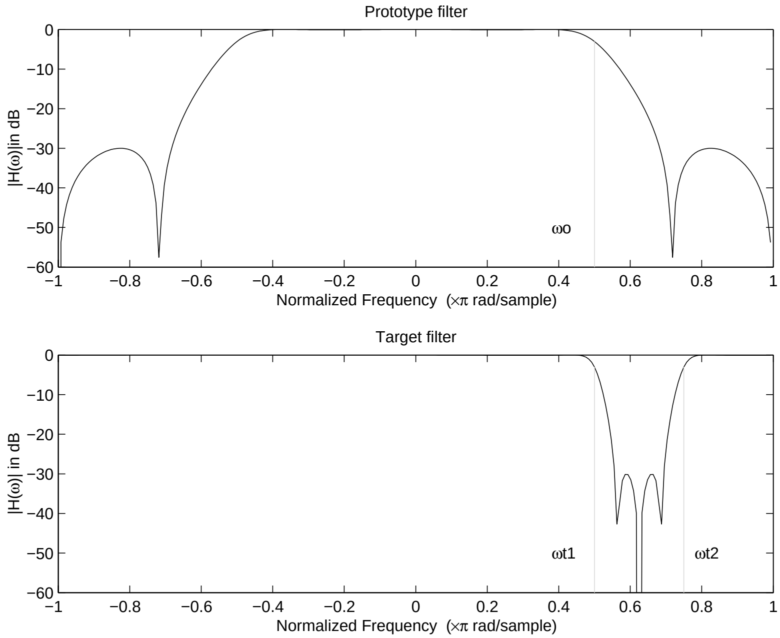


Figure 6-14: Example of Real Lowpass to Complex Bandstop Mapping

Real Lowpass to Complex Multiband

This high-order transformation performs an exact mapping of one selected feature of the prototype filter frequency response into a number of new locations in the target filter. Its most common use is to convert a real lowpass with predefined passband and stopband ripples into a multiband filter with arbitrary band edges. The order of the mapping filter must be even, which corresponds to an even number of band edges in the target filter. The Nth-order complex allpass mapping filter is given by the general transfer function form as shown below.

$$H_A(z) = S\frac{\sum_{i=0}^{N} \alpha_i z^{-i}}{\sum_{i=0}^{N} \alpha_i^* z^{-N+i}}$$

$$\alpha_0 = 1$$

The coefficients α are calculated from the system of linear equations:

$$\begin{cases} \sum_{i=1}^{N} \Re(\alpha_i) \cdot [\cos\beta_{1,k} - \cos\beta_{2,k}] + \Im(\alpha_i) \cdot [\sin\beta_{1,k} + \sin\beta_{2,k}] = \cos\beta_{3,k} \\ \sum_{i=1}^{N} \Re(\alpha_i) \cdot [\sin\beta_{1,k} - \sin\beta_{2,k}] - \Im(\alpha_i) \cdot [\cos\beta_{1,k} + \cos\beta_{2,k}] = \sin\beta_{3,k} \\ \beta_{1,k} = -\pi[\nu_{old} \cdot (-1)^k + \nu_{new,k}(N-k)] \\ \beta_{2,k} = -\pi[\Delta C + \nu_{new,k} k] \\ \beta_{3,k} = -\pi[\nu_{old} \cdot (-1)^k + \nu_{new,k} N] \\ k = 1 \ldots N \end{cases}$$

where

ω_{old} — Frequency location of the selected feature in the prototype filter

$\omega_{new,i}$ — Position of features originally at $\pm\omega_{old}$ in the target filter

Parameter `S` is the additional rotation factor by the frequency distance ΔC, giving the additional flexibility of achieving the required mapping:

$$S = e^{-j\pi\Delta C}$$

The example shows the use of such a transformation for converting a prototype real lowpass filter with the cutoff frequency at 0.5 into a multiband complex filter with band edges at 0.2, 0.4, 0.6 and 0.8, creating two passbands around the unit circle. Here is the MATLAB code for generating the figure.

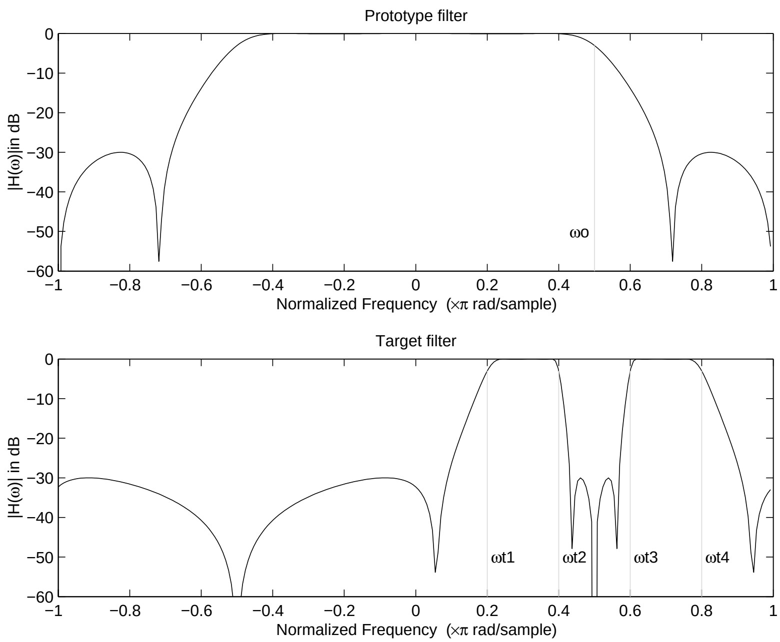


Figure 6-15: Example of Real Lowpass to Complex Multiband Mapping

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example transformation creates two complex passbands:

```
[num,den] = iirlp2mbc(b, a, 0.5, [0.2, 0.4, 0.6, 0.8]);
```

Real Lowpass to Complex Multipoint

This high-order transformation performs an exact mapping of a number of selected features of the prototype filter frequency response to their new locations in the target filter. The Nth-order complex allpass mapping filter is given by the general transfer function form as shown below.

$$H_A(z) = S\frac{\sum_{i=0}^{N} \alpha_i z^{-i}}{\sum_{i=0}^{N} \alpha_i^* z^{-N+i}}$$

$$\alpha_0 = 1$$

The coefficients α can be calculated from the system of linear equations:

$$\begin{cases} \sum_{i=1}^{N} \Re(\alpha_i) \cdot [\cos\beta_{1,k} - \cos\beta_{2,k}] + \Im(\alpha_i) \cdot [\sin\beta_{1,k} + \sin\beta_{2,k}] = \cos\beta_{3,k} \\ \sum_{i=1}^{N} \Re(\alpha_i) \cdot [\sin\beta_{1,k} - \sin\beta_{2,k}] - \Im(\alpha_i) \cdot [\cos\beta_{1,k} + \cos\beta_{2,k}] = \sin\beta_{3,k} \\ \beta_{1,k} = -\frac{\pi}{2}[w_{old,k} + w_{new,k}(N-k)] \\ \beta_{2,k} = -\frac{\pi}{2}[2\Delta C + w_{new,k}k] \\ \beta_{3,k} = -\frac{\pi}{2}[w_{old,k} + w_{new,k}N] \\ k = 1 \ldots N \end{cases}$$

where

$\omega_{old,k}$ — Frequency location of the first feature in the prototype filter

$\omega_{new,k}$ — Position of the feature originally at $\omega_{old,k}$ in the target filter

Parameter `S` is the additional rotation factor by the frequency distance ΔC, giving the additional flexibility of achieving the required mapping:

$$S = e^{-j\pi\Delta C}$$

The example below shows how this transformation can be used to move one selected feature of the prototype lowpass filter originally at -0.5 to two new frequencies -0.5 and 0.1, and the second feature of the prototype filter from 0.5 to new locations at -0.25 and 0.3. This creates two nonsymmetric passbands around the unit circle, creating a complex filter. Here is the MATLAB code for generating the figure.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example transformation creates two nonsymmetric passbands:

```
[num,den] = iirlp2xc(b,a,0.5*[-1,1,-1,1], [-0.5,-0.25,0.1,0.3]);
```

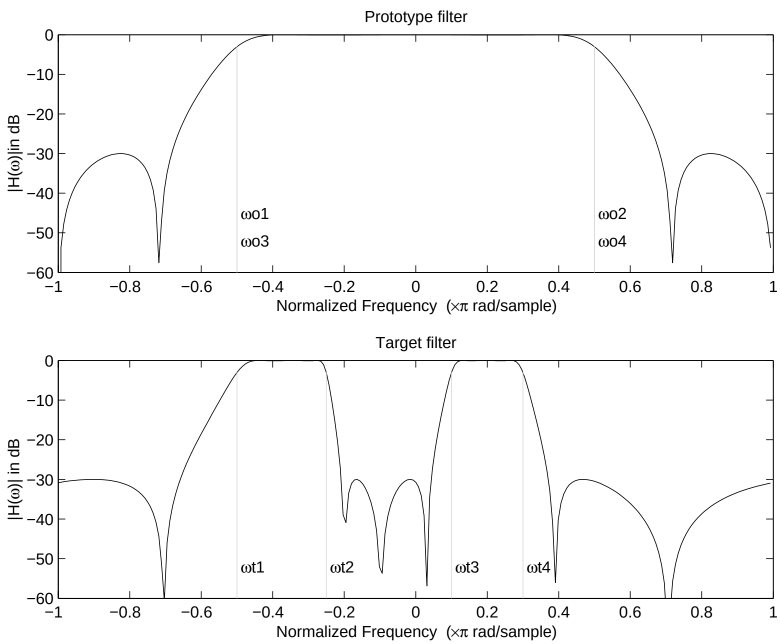


Figure 6-16: Example of Real Lowpass to Complex Multipoint Mapping

Complex Bandpass to Complex Bandpass

This first-order transformation performs an exact mapping of two selected features of the prototype filter frequency response into two new locations in the target filter. Its most common use is to adjust the edges of the complex bandpass filter.

$$H_A(z) = \frac{\alpha(\gamma - \beta z^{-1})}{z^{-1} - \beta\gamma}$$

with α and β are given by

$$\alpha = \frac{\sin\frac{\pi}{4}((w_{old,2} - w_{old,1}) - (w_{new,2} - w_{new,1}))}{\sin\frac{\pi}{4}((w_{old,2} - w_{old,1}) + (w_{new,2} - w_{new,1}))}$$

$$\alpha = e^{-j\pi(w_{old,2} - w_{old,1})}$$

$$\gamma = e^{-j\pi(w_{new,2} - w_{new,1})}$$

where

$\omega_{old,1}$ — Frequency location of the first feature in the prototype filter

$\omega_{old,2}$ — Frequency location of the second feature in the prototype filter

$\omega_{new,1}$ — Position of the feature originally at $\omega_{old,1}$ in the target filter

$\omega_{new,2}$ — Position of the feature originally at $\omega_{old,2}$ in the target filter

The example below shows how this transformation can be used to modify the position of the passband of the prototype filter, either real or complex. In the example below the prototype filter passband spanned from 0.5 to 0.75. It was converted to having a passband between -0.5 and 0.1. Here is the MATLAB code for generating the figure.

The prototype filter is a halfband elliptic, real, third-order lowpass filter:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

The example transformation creates a passband from 0.25 to 0.75:

```
[num,den] = iirbpc2bpc(b, a, [0.25, 0.75], [-0.5, 0.1]);
```

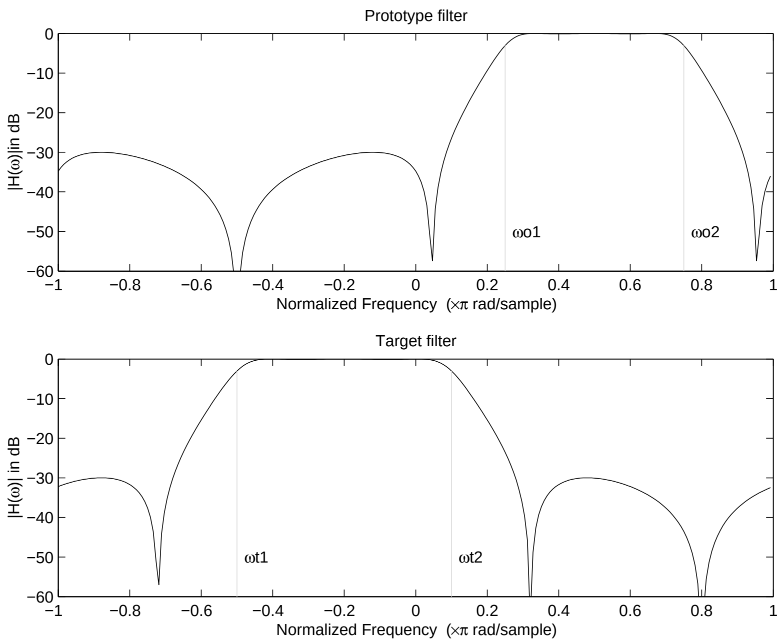


Figure 6-17: Example of Complex Bandpass to Complex Bandpass Mapping

7

Using FDATool with the Filter Design Toolbox

The Filter Design Toolbox adds new dialogs and operating modes, and new menu selections, to the Filter Design and Analysis Tool (FDATool) provided by the Signal Processing Toolbox. From the new dialogs, one titled **Set Quantization Parameters** and one titled **Frequency Transformations**, you can:

- Design advanced filters that Signal Processing Toolbox does not provide the design tools to develop.
- View Simulink models of the filter structures available in the toolbox.
- Quantize double-precision filters you design in this GUI using the design mode.
- Quantize double-precision filters you import into this GUI using the import mode.
- Analyze quantized filters.
- Scale second-order section filters.
- Select the quantization settings for the properties of the quantized filter displayed by the tool:
  - Coefficients—select the quantization options applied to the filter coefficients
  - Input/output—control how the filter processes input and output data
  - Filter Internals—specify how the arithmetic for the filter behaves
- Design multirate filters.
- Transform both FIR and IIR filters from one response to another.

After you import a filter in to FDATool, the options on the quantization dialog let you quantize the filter and investigate the effects of various quantization settings.

Options in the frequency transformations dialog let you change the frequency response of your filter, keeping various important features while changing the response shape.

This section presents the following information and procedures for using FDATool:

- “Designing Advanced Filters in FDATool” on page 7-5
- “Switching FDATool to Quantization Mode” on page 7-10
- “Quantizing Filters in the Filter Design and Analysis Tool” on page 7-14

Designing Advanced Filters in FDATool

Adding the Filter Design Toolbox to your tool suite adds a number of new filter design techniques to FDATool. Use the new filter responses to develop filters that meet more complex requirements than those you can design in the Signal Processing Toolbox. While the designs in FDATool are available as command line functions, the graphical user interface of FDATool makes the design process more clear and easier to accomplish.

In addition to the designs provided by FDATool and the Signal Processing Toolbox, you gain the following response types in FDATool when you have the Filter Design Toolbox:

Added Lowpass Responses

Response Type	Description
Halfband lowpass	Design a lowpass filter using the halfband algorithm
Inverse sinc lowpass	Design a lowpass filter using the inverse sinc technique
Nyquist	Design a lowpass Nyquist filter

Added Highpass Responses

Response Type	Description
Halfband highpass	Design a highpass filter using the halfband algorithm
Inverse sinc highpass	Design a highpass filter using the inverse sinc technique

Added Assorted Others

Response Type	Description
Arbitrary group delay	Design a filter whose group delay meets your specific needs
Notching	Design filter responses that you carefully tune to de-emphasize specific frequencies in your input data
Peaking	Design filter responses that you tune to emphasize specific frequencies in your input data

As you select a response type, the options in the panels to the right in FDATool change to let you set the values that define your filter. You also see that the analysis area includes a diagram that describes the options for the filter response you choose. By reviewing the diagram you can see how the options are defined and how to use them. While this is usually straightforward for lowpass or highpass filter responses, setting the options for the arbitrary response types or the peaking/notching filters is more complicated. Having the diagrams leads you to your result more easily.

Changing the filter design method changes the available response type options. Similarly, the response type you select may change the filter design methods you can choose.

Example—Design a Notch Filter

Notch filters aim to remove one or a few frequencies from a broader spectrum.You must specify the frequencies to remove by setting the filter design options in FDATool appropriately:

- Response Type
- Design Method
- Frequency Specifications
- Magnitude Specifications

Here is how you design a notch filter that removes concert A (440 Hz) from an input musical signal spectrum.

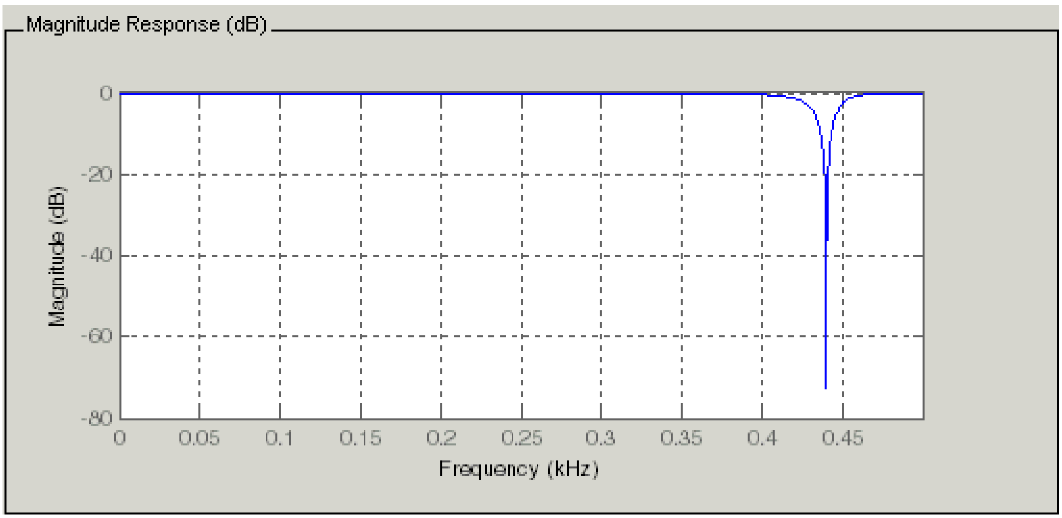
1. Select `Notching` from the **Differentiator** list in **Response Type**.
2. Select **IIR** in **Filter Design Method** and choose `Single Notch` from the list**.**
3. For the **Frequency Specifications**, set **Units** to `Hz` and **Fs**, the full scale frequency, to `10000`.
4. Set the location of the center of the notch, in either normalized frequency or Hz. For the notch center at 440 Hz, enter `440`.
5. To shape the notch, enter the **bandwidth**, `bw`, to be `40`.
6. Leave the **Magnitude Specification** in `dB` (the default) and leave **Apass** as `1`.

7 Click Design Filter.

FDATool computes the filter coefficients and plots the filter magnitude response in the analysis area for you to review.

When you design a single notch filter, you do not have the option of setting the filter order—the **Filter Order** options are disabled.

Your filter should look about like this:



For more information about a design method, refer to the online Help system. For instance, to get further information about the **Q** setting for the notch filter in FDATool, enter

```
doc iirnotch
```

at the prompt. This opens the Help browser and displays the reference page for function `iirnotch`.

Designing other filters follows a similar procedure, adjusting for different design specification options as each design requires.

Any one of the designs may be quantized in FDATool and analyzed with the available analyses on the **Analysis** menu. For more general information about FDATool, such as the user interface and areas, refer to the FDATool documentation in the Signal Processing Toolbox documentation. One way to do this is to enter

```
doc signal/fdatool
```

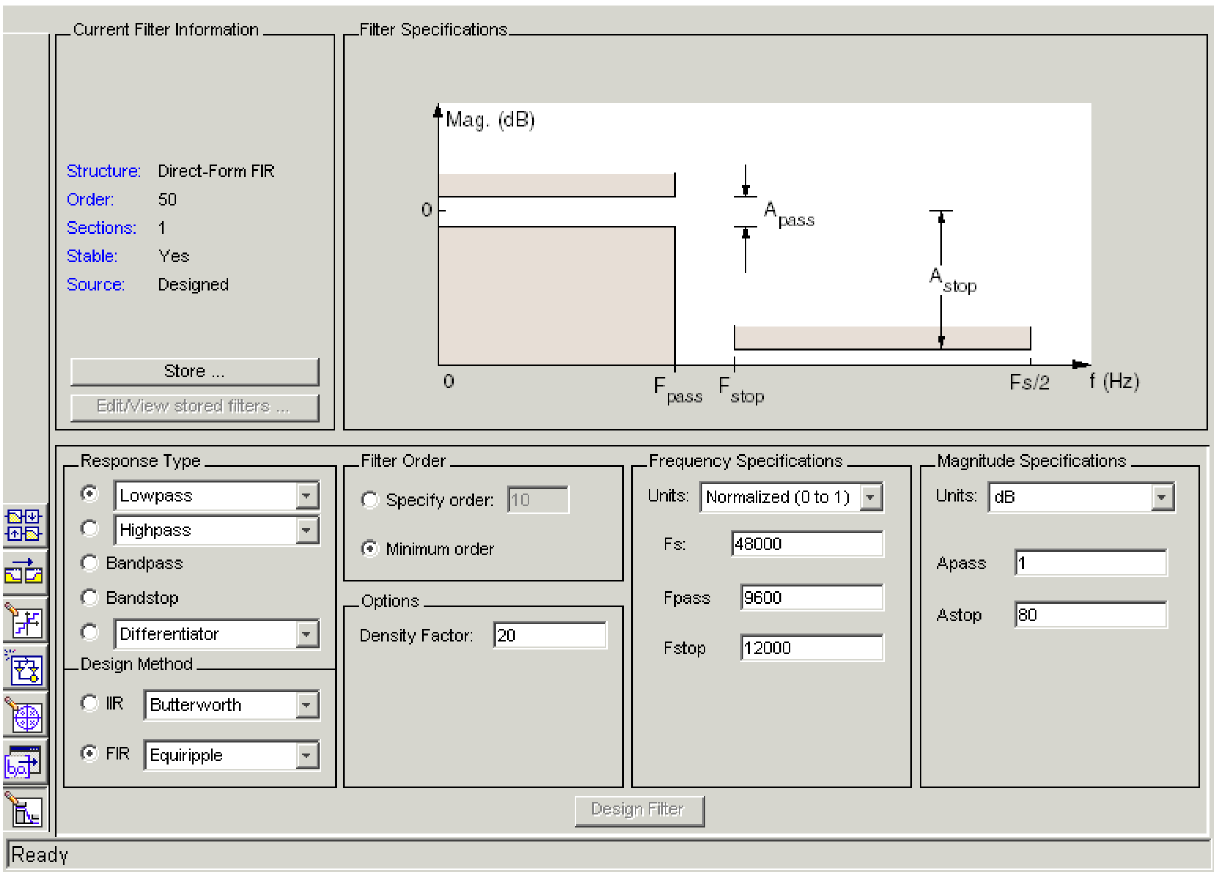
at the prompt. The `signal` qualifier is necessary to open the reference page in the Signal Processing Toolbox documentation, rather than the page in the Filter Design Toolbox documentation. You might also look at the general section on FDATool in the *Signal Processing Toolbox User's Guide*.

Switching FDATool to Quantization Mode

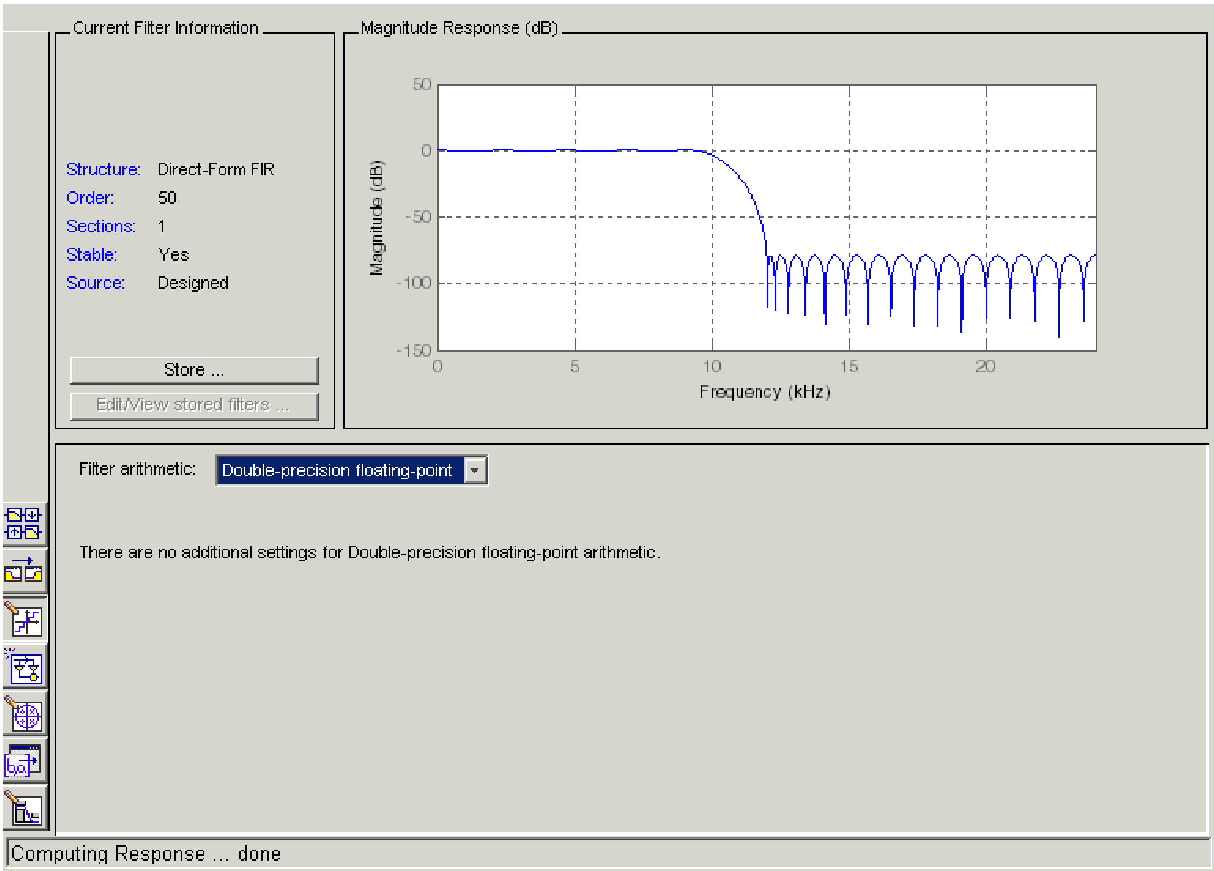
You use the quantization mode in FDATool to quantize filters. Quantization represents the fourth operating mode for FDATool, along with the filter design, filter transformation, and import modes. To switch to quantization mode, open FDATool from the MATLAB command prompt by entering

```
fdatool
```

You see FDATool in this configuration.

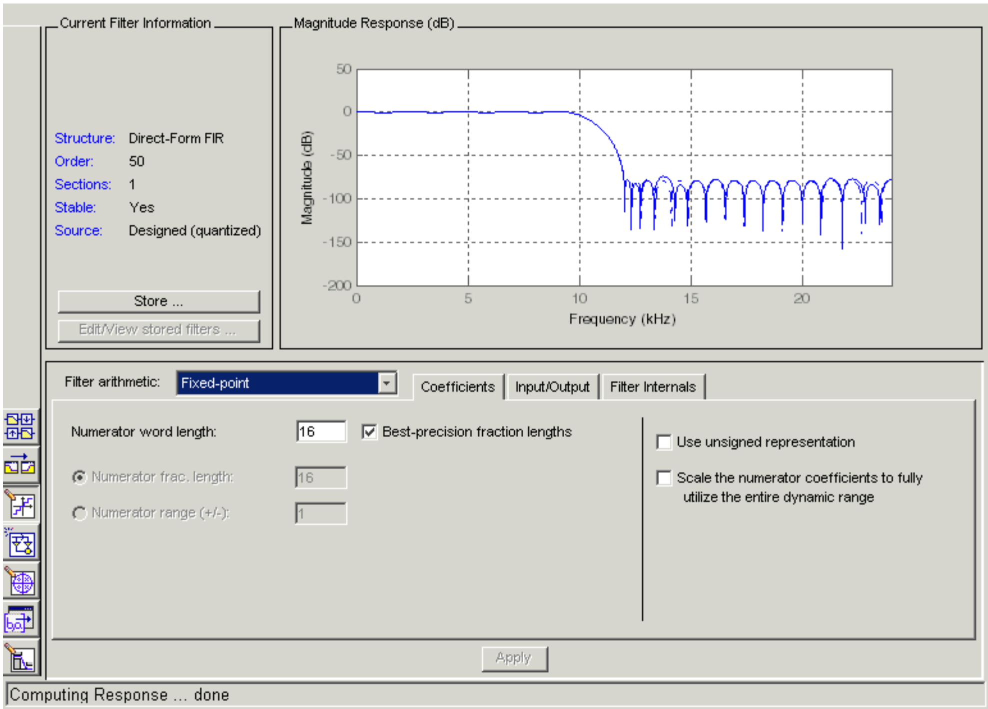


When FDATool opens, click the **Set Quantization Parameters** button on the side bar. FDATool switches to quantization mode and you see the following panel at the bottom of FDATool, with the default double-precision option shown for **Filter arithmetic**.



The **Filter arithmetic** option lets you quantize filters and investigate the effects of changing quantization settings. To enable the quantization settings in FDATool, select `Fixed-point` from the **Filter Arithmetic**.

The quantization options appear in the lower panel of FDATool. You see tabs that access various sets of options for quantizing your filter.



You use the following tabs in the dialog to perform tasks related to quantizing filters in FDATool:

- **Coefficients** provides access the settings for defining the coefficient quantization. This is the default active panel when you switch FDATool to quantization mode without a quantized filter in the tool. When you import a fixed-point filter into FDATool, this is the active pane when you switch to quantization mode.
- **Input/Output** switches FDATool to the options for quantizing the inputs and outputs for your filter.
- **Filter Internals** lets you set a variety of options for the arithmetic your filter performs, such as how the filter handles the results of multiplication operations or how the filter uses the accumulator.

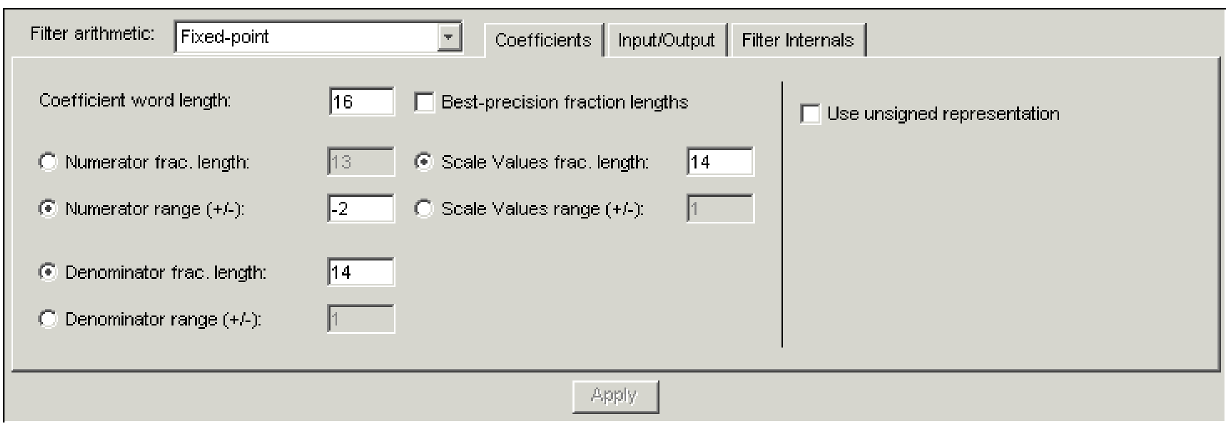
- **Apply**—applies changes you make to the quantization parameters for your filter.

Quantizing Filters in the Filter Design and Analysis Tool

Quantized filters have properties that define how they quantize data you filter. Use the **Set Quantization Parameters** dialog in FDATool to set the properties. Using options in the **Set Quantization Parameters** dialog, FDATool lets you perform a number of tasks:

- Create a quantized filter from a double-precision filter after either importing the filter from your workspace, or using FDATool to design the prototype filter.
- Create a quantized filter that has the default structure (Direct form II transposed) or any structure you choose, and other property values you select.
- Change the quantization property values for a quantized filter after you design the filter or import it from your workspace.

When you click **Set Quantization Parameters**, and then change **Filter Arithmetic** to `Fixed-point`, the quantized filter panel opens in FDATool, with the coefficient quantization options set to default values. In this image, you see the options for an SOS filter. Some of the options shown apply only to SOS filters. Other filter structures present a subset of the options you see here.



Coefficients Options

To let you set the properties for the filter coefficients that make up your quantized filter, FDATool lists options for numerator word length (and denominator word length if you have an IIR filter). Table 7-1 lists every coefficients option and a short description of what the option setting does in the filter.

Table 7-1: Filter Options

Option Name	When Used	Description
Numerator Word Length	FIR filters only	Sets the word length used to represent numerator coefficients in FIR filters.
Numerator Frac. Length	FIR/IIR	Sets the fraction length used to interpret numerator coefficients in FIR filters.
Numerator Range (+/-)	FIR/IIR	Lets you set the range the numerators represent. You use this instead of the **Numerator Frac. Length** option to set the precision. When you enter a value x, the resulting range is -x to x. Range must be a positive integer.
Coefficient Word Length	IIR filters only	Sets the word length used to represent both numerator and denominator coefficients in IIR filters. You cannot set different word lengths for the numerator and denominator coefficients.
Denominator Frac. Length	IIR filters	Sets the fraction length used to interpret denominator coefficients in IIR filters.

Table 7-1: Filter Options

Option Name	When Used	Description
Denominator Range (+/-)	IIR filters	Lets you set the range the denominator coefficients represent. You use this instead of the **Denominator Frac. Length** option to set the precision. When you enter a value x, the resulting range is -x to x. Range must be a positive integer.
Best-precision fraction lengths	All filters	Directs FDATool to select the fraction lengths for numerator (and denominator where available) values to maximize the filter performance. Selecting this option disables all of the fraction length options for the filter.
Scale Values frac. length	SOS IIR filters	Sets the fraction length used to interpret the scale values in SOS filters.
Scale Values range (+/-)	SOS IIR filters	Lets you set the range the SOS scale values represent. You use this with SOS filters to adjust the scaling used between filter sections. Setting this value disables the **Scale Values frac. length** option. When you enter a value x, the resulting range is -x to x. Range must be a positive integer.
Use unsigned representation	All filters	Tells FDATool to interpret the coefficients as unsigned values.
Scale the numerator coefficients to fully utilize the entire dynamic range	All filters	Directs FDATool to scale the numerator coefficients to effectively use the dynamic range defined by the numerator word length and fraction length format.

Input/Output Options

The options that specify how the quantized filter uses input and output values are listed in Table 7-2. In the following picture you see the options for an SOS filter.

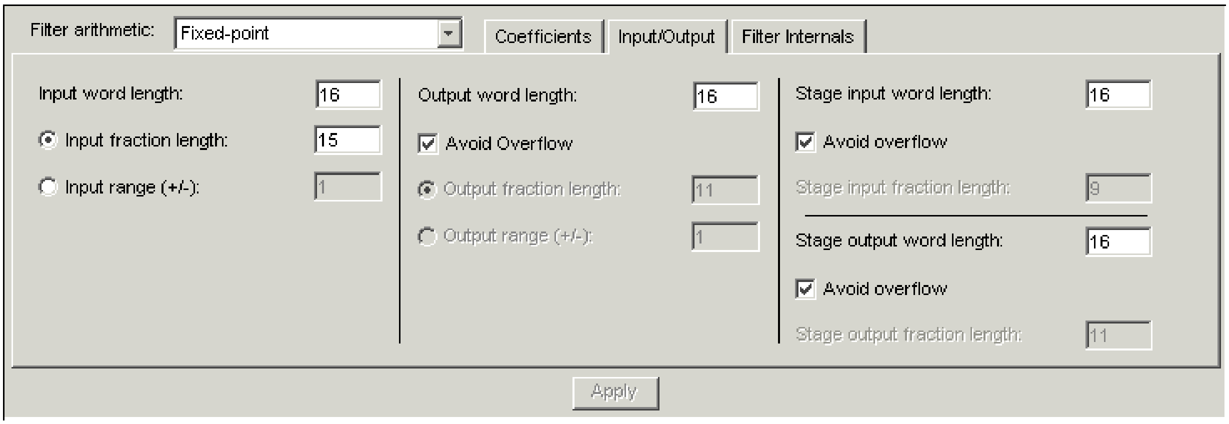


Table 7-2: Input/Output Options

Option Name	When Used	Description
Input Word Length	All filters	Sets the word length used to represent the input to a filter.
Input fraction length	All filters	Sets the fraction length used to interpret input values to filter.
Input range (+/-)	All filters	Lets you set the range the inputs represent. You use this instead of the **Input fraction length** option to set the precision. When you enter a value x, the resulting range is -x to x. Range must be a positive integer.
Output word length	All filters	Sets the word length used to represent the output from a filter.

Table 7-2: Input/Output Options (Continued)

Option Name	When Used	Description
Avoid overflow	All filters	Directs the filter to set the fraction length for the input to prevent the output values from exceeding the available range as defined by the word length. Clearing this option lets you set **Output fraction length**.
Output fraction length	All filters	Sets the fraction length used to represent output values from a filter.
Output range (+/-)	All filters	Lets you set the range the outputs represent. You use this instead of the **Output fraction length** option to set the precision. When you enter a value x, the resulting range is -x to x. Range must be a positive integer.
Stage input word length	SOS filters only	Sets the word length used to represent the input to an SOS filter section.
Avoid overflow	SOS filters only	Directs the filter to use a fraction length for stage inputs that prevents overflows in the values. When you clear this option, you can set **Stage input fraction length**.
Stage input fraction length	SOS filters only	Sets the fraction length used to represent input to a section of an SOS filter.
Stage output word length	SOS filters only	Sets the word length used to represent the output from an SOS filter section.

Table 7-2: Input/Output Options (Continued)

Option Name	When Used	Description
Avoid overflow	SOS filters only	Directs the filter to use a fraction length for stage outputs that prevents overflows in the values. When you clear this option, you can set **Stage output fraction length**.
Stage output fraction length	SOS filters only	Sets the fraction length used to represent the output from a section of an SOS filter.

Filter Internals Options

The options that specify how the quantized filter performs arithmetic operations are listed in Table 7-3. In the following picture you see the options for an SOS filter.

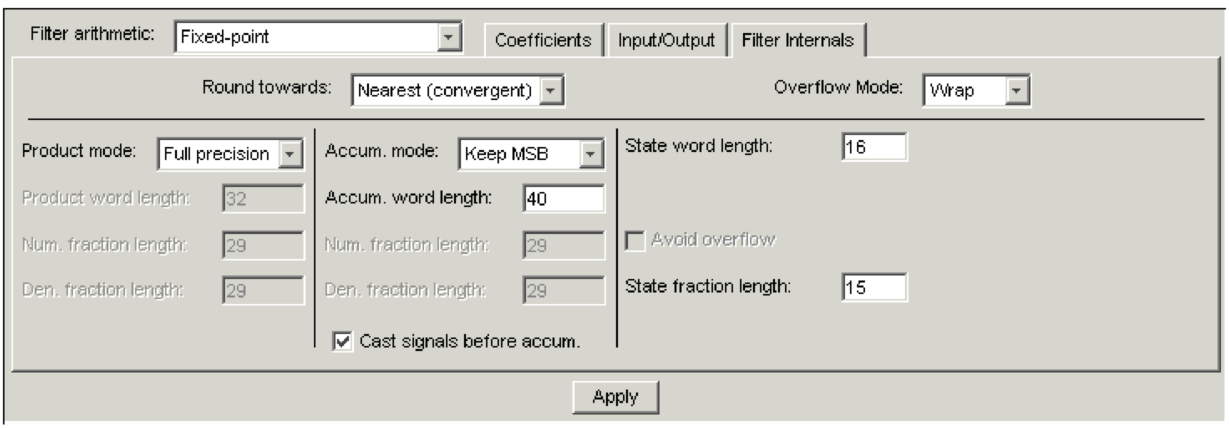


Table 7-3: Options for the Filter Internals

Option	Equivalent Filter Property (using wildcard *)	Description
Round towards	RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). Choose from one of: • convergent—round up to the next allowable quantized value • ceil—round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1 • fix—round negative numbers up and positive numbers down to the next allowable quantized value • floor—round down to the next allowable quantized value • round—round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up.

Table 7-3: Options for the Filter Internals (Continued)

Option	Equivalent Filter Property (using wildcard *)	Description
Overflow Mode	`OverflowMode`	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either `saturate` (limit the output to the largest positive or negative representable value) or `wrap` (set overflowing values to the nearest representable value using modular arithmetic.
Filter Product (Multiply) Options		
Product Mode	`ProductMode`	Determines how the filter handles the output of product operations. Choose from full precision (`FullPrecision`), or whether to keep the most significant bit (`KeepMSB`) or least significant bit (`KeepLSB`) in the result when you need to shorten the word length. `Specify all` lets you set the fraction length applied to the results of product operations.
Product word length	`*ProdWordLength`	Sets the word length applied to interpret the results of multiply operations.
Num. fraction length	`NumProdFracLength`	Sets the fraction length used to interpret the results of product operations that involve numerator coefficients.

Table 7-3: Options for the Filter Internals (Continued)

Option	Equivalent Filter Property (using wildcard *)	Description
Den. fraction length	`DenProdFracLength`	Sets the fraction length used to interpret the results of product operations that involve denominator coefficients.
Filter Sum Options		
Accum. mode	`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set this to `Specify all`.
Accum. word length	`*AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.
Num. fraction length	`NumAccumFracLength`	Sets the fraction length used to interpret the numerator coefficients.
Den. fraction length	`DenAccumFracLength`	Sets the fraction length the filter uses to interpret denominator coefficients.
Cast signals before sum	`CastBeforeSum`	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams for each filter structure) before performing sum operations.
Filter State Options		

Table 7-3: Options for the Filter Internals (Continued)

Option	Equivalent Filter Property (using wildcard *)	Description
State word length	`*StateWordLength`	Sets the word length used to represent the filter states. Applied to both numerator- and denominator-related states
Avoid overflow	`None`	Prevent overflows in arithmetic calculations by setting the fraction length appropriately.
State fraction length	`*StateFracLength`	Lets you set the fraction length applied to interpret the filter states. Applied to both numerator- and denominator-related states

Example—Quantize Double-Precision Filters

When you are quantizing a double-precision filter by switching to fixed-point or single-precision floating point arithmetic, follow these steps.

1 Click **Set Quantization Parameters** to display the **Set Quantization Parameters** pane in FDATool.

2 Select `Single-precision floating point` or `Fixed-point` from **Filter arithmetic**.

   When you select one of the optional arithmetic settings, FDATool quantizes the current filter according to the settings of the options in the Set Quantization Parameter panes, and changes the information displayed in the analysis area to show quantized filter data.

3 In the quantization panes, set the options for your filter. Set options for **Coefficients**, **Input/Output**, and **Filter Internals**.

4 Click **Apply**.

   FDATool quantizes your filter using your new settings.

5 Use the analysis features in FDATool to determine whether your new quantized filter meets your requirements.

Example—Change the Quantization Properties of Quantized Filters

When you are changing the settings for the quantization of a quantized filter, or after you import a quantized filter from your MATLAB workspace, follow these steps to set the property values for the filter:

1 Verify that the current filter is quantized.

2 Click **Set Quantization Parameters** to display the **Set Quantization Parameters** panel.

3 Review and select property settings for the filter quantization: **Coefficients**, **Input/Output**, and **Filter Internals**. Settings for options on these panes determine how your filter quantizes data during filtering operations.

4 Click **Apply** to update your current quantized filter to use the new quantization property settings from Step 3.

5 Use the analysis features in FDATool to determine whether your new quantized filter meets your requirements.

Analyzing Filters with a Noise-Based Method

One technique for estimating the frequency response for quantized filters is the magnitude response estimate. FDATool offers this noise-based method as a filter analysis tool accessible from the toolbar.

Using the Magnitude Response Estimate Method

After you design and quantize your filter, the **Magnitude Response Estimate** option on the **Analysis** menu lets you apply the noise loading method to your filter. When you select **Analysis -> Magnitude Response Estimate** from the menubar, FDATool immediately starts the Monte Carlo trials that form the basis for the method and runs the analysis, ending by displaying the results in the analysis area in FDATool.

With the noise-based method, you estimate the complex frequency response for your filter as determined by applying a noise- like signal to the filter input. **Magnitude Response Estimate** uses the Monte Carlo trials to generate a noise signal that contains complete frequency content across the range 0 to Fs. The first time you run the analysis, magnitude response estimate uses default settings for the various conditions that define the process, such as the number of test points and the number of trials.

Analysis Parameter	Default Setting	Description
Number of Points	`512`	Number of equally spaced points around the upper half of the unit circle.
Frequency Range	`0 to Fs/2`	Frequency range of the plot x-axis.
Frequency Units	`Hz`	Units for specifying the frequency range.
Sampling Frequency	`48000`	Inverse of the sampling period.

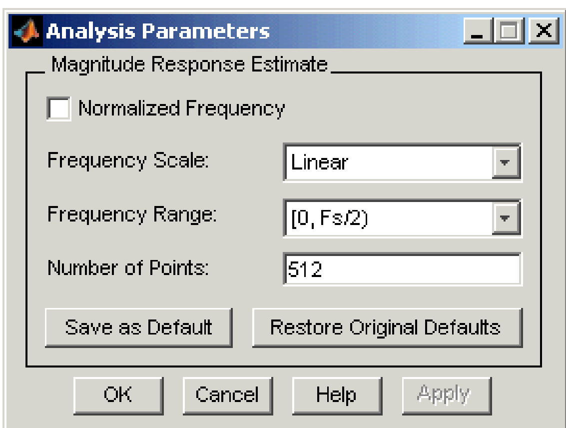
Analysis Parameter	Default Setting	Description
Frequency Scale	`dB`	Units used for the y-axis display of the output.
Normalized Frequency	`Off`	Use normalized frequency for the display.

After your first analysis run ends, open the **Analysis Parameters** dialog and adjust your settings appropriately, such as changing the number of trials or number of points.

To open the **Analysis Parameters** dialog, use either of the next procedures when you have a quantized filter in FDATool:

- Select **Analysis -> Analysis Parameters** from the menu bar
- Right-click in the filter analysis area and select **Analysis Parameters** from the context menu

Whichever option you choose opens the dialog as shown in the figure. Notice that the settings for the options reflect the defaults.



Example—Noise Method Applied to a Filter

To demonstrate the magnitude response estimate method, start by creating a quantized filter. For this example, use FDATool to design a sixth-order Butterworth IIR filter.

To Use Noise-Based Analysis in FDATool

1. Enter `fdatool` at the MATLAB prompt to launch FDATool.

2. Under **Response Type**, select **Highpass**.

3. Select `IIR` in **Design Method**. Then select `Butterworth`.

4. To set the filter order to 6, select **Specify order** under **Filter Order**. Enter `6` in the text box.

5. Click **Design Filter**.

   In FDATool, the analysis area changes to display the magnitude response for your filter.

6. To generate the quantized version of your filter, using default quantizer settings, click  on the side bar.

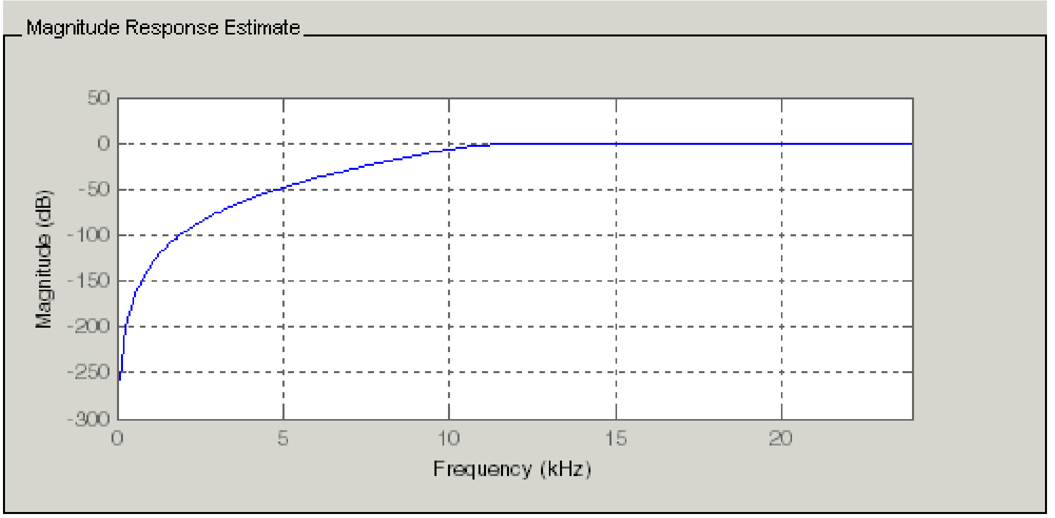
   FDATool switches to quantization mode and displays the quantization panel.

7. From **Filter arithmetic**, select `fixed-point`.

   Now the analysis areas shows the magnitude response for both filters—your original filter and the fixed-point arithmetic version.

8. Finally, to use noise-based estimation on your quantized filter, select **Analysis -> Magnitude Response Estimate** from the menubar.

   FDATool runs the trial, calculates the estimated magnitude response for the filter, and displays the result in the analysis area as shown in this figure.



In the figure you see the magnitude response as estimated by the analysis method.

To View the Noise Power Spectrum

When you use the noise method to estimate the magnitude response of a filter, FDATool simulates and applies a spectrum of noise values to test your filter response. While the simulated noise is essentially white, you might want to see the actual spectrum that FDATool used to test your filter.

From the **Analysis** menu bar option, select **Round-off Noise Power Spectrum**. In the analysis area in FDATool, you see the spectrum of the noise used to estimate the filter response. The details of the noise spectrum, such as the range and number of data points, appear in the **Analysis Parameters** dialog.

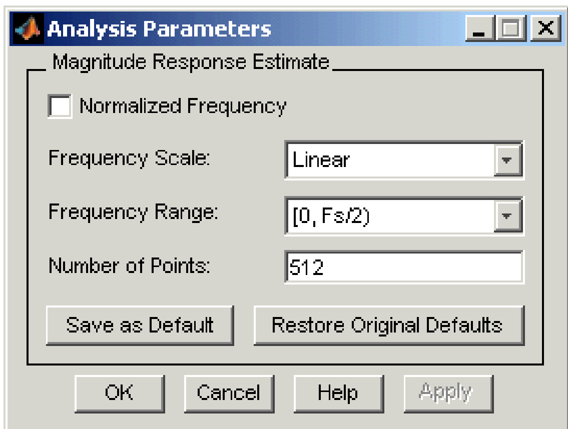
To Change Your Noise Analysis Parameters

In “Example—Noise Method Applied to a Filter”, you used synthetic white noise to estimate the magnitude response for a fixed-point highpass Butterworth filter. Since you ran the estimate only once in FDATool, your noise analysis used the default analysis parameters settings shown in “Using the Magnitude Response Estimate Method”.

To change the settings, follow these steps after the first time you use the noise estimate on your quantized filter.

1 With the results from running the noise estimating method displayed in the FDATool analysis area, select **Analysis->Analysis Parameters** from the menubar.

   To give you access to the analysis parameters, the **Analysis Parameters** dialog opens as shown here (with default settings).



2 To use more points in the spectrum to estimate the magnitude response, change **Number of Points** to 1024 and click **OK** to run the analysis.

   FDATool closes the **Analysis Parameters** dialog and reruns the noise estimate, returning the results in the analysis area.

   To rerun the test without closing the dialog, press **Enter** after you type your new value into a setting, then click **Apply**. Now FDATool runs the test without closing the dailog. When you want to try many different settings for the noise-based analysis, this is a useful shortcut.

Comparing the Estimated and Theoretical Magnitude Responses

An important measure of the effectiveness of the noise method for estimating the magnitude response of a quantized filter is to compare the estimated response to the theoretical response.

One way to do this comparison is to overlay the theoretical response on the estimated response. While you have the Magnitude Response Estimate displaying in FDATool, select **Analysis->Overlay Analysis** from the menu bar. Then select **Magnitude Response** to show both response curves plotted together in the analysis area.

Choosing Quantized Filter Structures

FDATool lets you change the structure of any quantized filter. Use the **Convert structure** option to change the structure of your filter to one that meets your needs.

To learn about changing the structure of a filter in FDATool, refer to "Converting to a New Structure" in your Signal Processing Toolbox documentation.

Converting the Structure of a Quantized Filter

You use the **Convert structure** option to change the structure of filter. When the **Source** is **Designed(Quantized)** or **Imported(Quantized)**, **Convert structure** lets you recast the filter to one of the following structures:

- "Direct Form II Transposed Filter Structure" on page 8-55
- "Direct Form I Transposed Filter Structure" on page 8-51
- "Direct Form II Filter Structure" on page 8-52
- "Direct Form I Filter Structure" on page 8-50
- "Direct Form Finite Impulse Response (FIR) Filter Structure" on page 8-60
- "Direct Form FIR Transposed Filter Structure" on page 8-61
- "Lattice Autoregressive Moving Average (ARMA) Filter Structure" on page 8-67
- "dfilt.calattice" on page 9-281
- "dfilt.calatticepc" on page 9-284

- “Direct Form Symmetric FIR Filter Structure (Any Order)” on page 8-69

Starting from any quantized filter, you can convert to one of the following representation:

- Direct form I
- Direct form II
- Direct form I transposed
- Direct form II transposed
- Lattice ARMA

Additionally, FDATool lets you do the following conversions:

- Minimum phase FIR filter to Lattice MA minimum phase
- Maximum phase FIR filter to Lattice MA maximum phase
- Allpass filters to Lattice allpass

Refer to “FilterStructure” on page 8-46 for details about each of these structures.

Converting Filters to Second-Order Sections Form

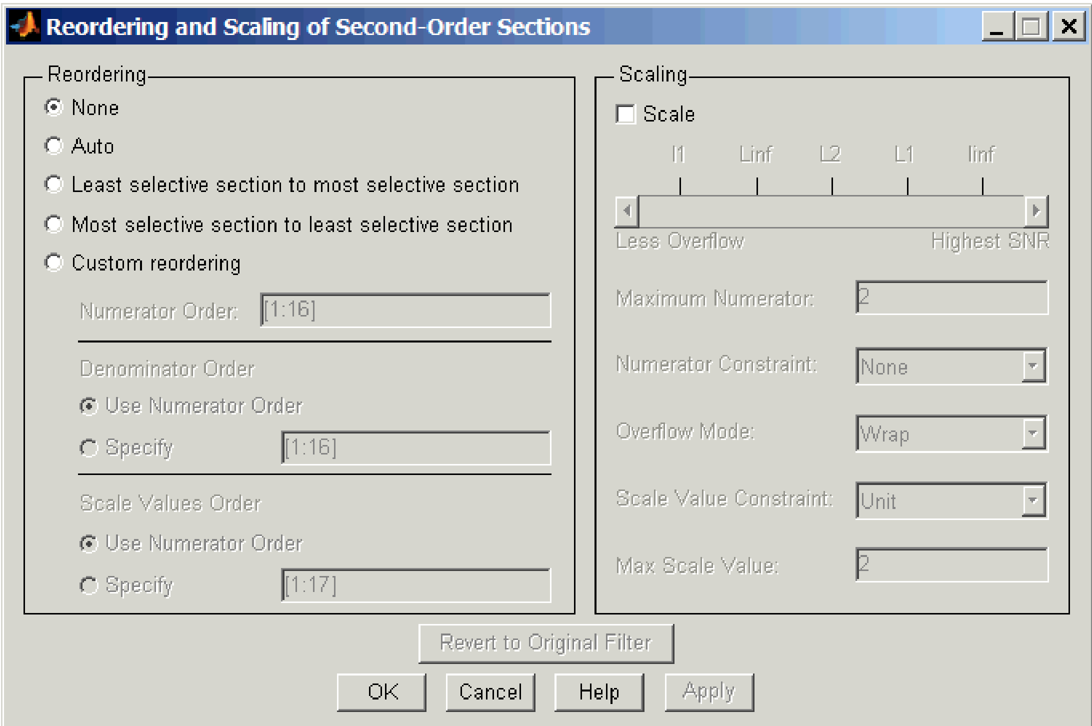
To learn about using FDATool to convert your quantized filter to use second-order sections, refer to “Converting to Second-Order Sections” in your Signal Processing Toolbox documentation. You might notice that filters you design in FDATool, rather than filters you imported, are implemented in SOS form.

To View Filter Structures in FDATool

To open the demonstration, click **Help -> Show filter structures**. After the Help browser opens, you see the reference page for the current filter. You find the filter structure signal flow diagram on this reference page, or you can navigate to reference pages for other filter.

Scaling Second-Order Section Filters

FDATool provides the ability to scale SOS filters after you create them. Using options on the Reordering and Scaling Second-Order Sections dialog, FDATool scales either or both the filter numerators and filter scale values according to your choices for the scaling options.



Parameter	Description and Valid Value
Scale	Apply any scaling options to the filter. Select this when you are reordering your SOS filter and you want to scale it at the same time. Or when you are scaling your filter, with or without reordering. Scaling is disabled by default.
No Overflow—High SNR slider	Lets you set whether scaling favors reducing arithmetic overflow in the filter or maximizing the signal-to-noise ratio (SNR)) at the filter output. Moving the slider to the right increases the emphasis on SNR at the expense of possible overflows. The markings indicate the P-norm applied to achieve the desired result in SNR or overflow protection. For more information about the P-norm settings, refer to `norm` for details.
Maximum Numerator	Maximum allowed value for numerator coefficients after scaling.
Numerator Constraint	Specifies whether and how to constrain numerator coefficient values. Options are `none`, `normalize`, `power of 2`, and `unit`. Choosing `none` lets the scaling use any scale value for the numerators by removing any constraints on the numerators. `Normalize`. The `power of 2` option forces scaling to use numerator values that are powers of 2, such as 2 or 0.5.

Parameter	Description and Valid Value
Overflow Mode	Sets the way the filter handles arithmetic overflow situations during scaling. Choose from either `saturate` (limit the output to the largest positive or negative representable value) or `wrap` (set overflowing values to the nearest representable value using modular arithmetic.
Scale Value Constraint	Specify whether to constrain the filter scale values, and how to constrain them. Valid options are `none`, `power of 2`, and `unit`. Choosing `unit` for the constraint disables the **Max. Scale Value** setting and limits scale values to one. `Power of 2` constrains the scale values to be powers of 2, such as 2 or 0.5, while `none` removes any constraint on the scale values.
Max. Scale Value	Sets the maximum allowed scale values. SOS filter scaling applies the **Max. Scale Value** limit only when you set **Scale Value Constraint** to a value other than `unit` (the default setting). Note that setting a maximum scale value removes any other limits on the scale values.
Revert to Original Filter	Returns your filter to the original scaling. Being able to revert to your original filter makes it easier to assess the results of scaling your filter.

Various combinations of settings let you scale filter numerators without changing the scale values, or adjust the filter scale values without changing the numerators. There is no scaling control for denominators.

Example—Scale An SOS Filter

Start the process by designing a lowpass elliptical filter in FDATool.

1 Launch FDATool.

2 In **Response Type**, select **Lowpass**.

3 In Design Method, select **IIR** and `Elliptic` from the IIR design methods list.

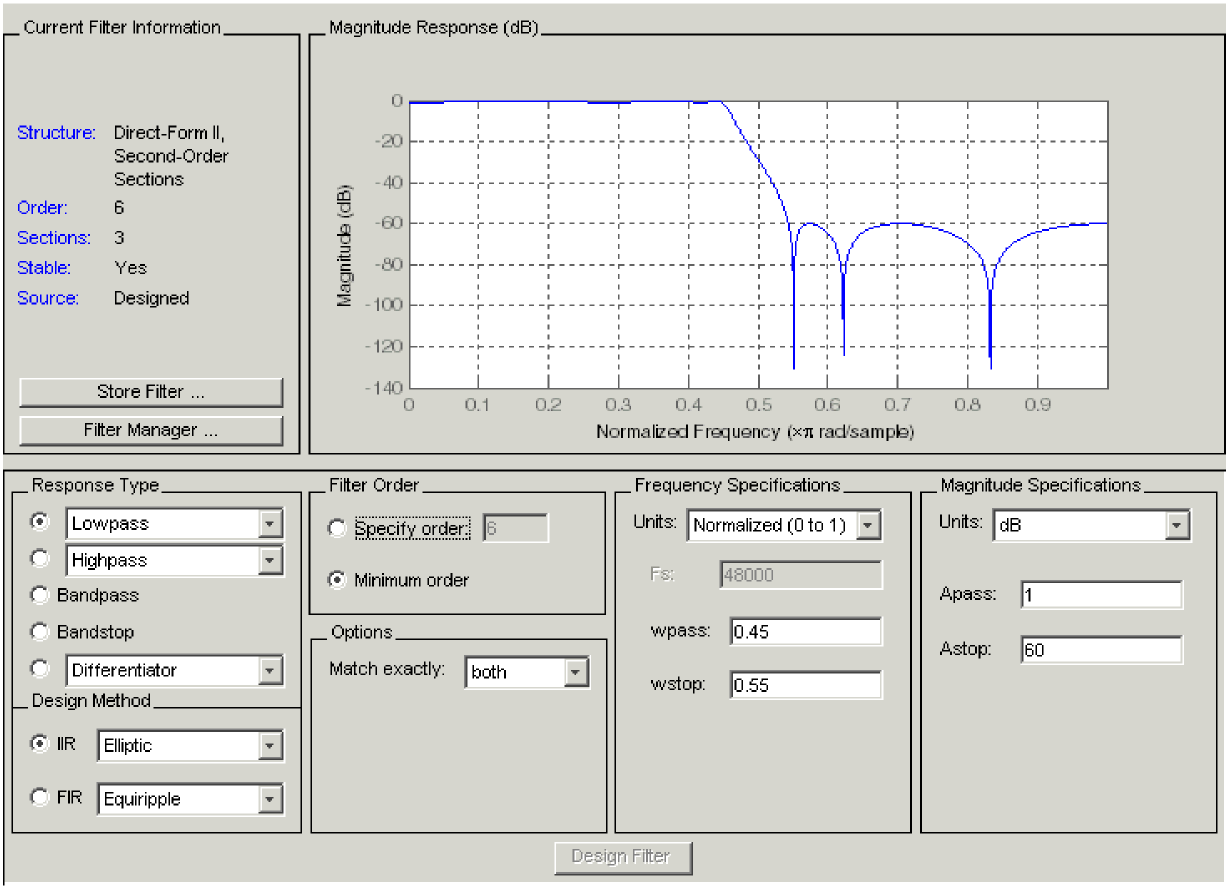
4 Select **Minimum Order** for the filter.

5 Switch the frequency units by choosing `Normalized(0 to 1)` from the **Units** list.

6 To set the passband specifications, enter `0.45` for **wpass** and `0.55` for **wstop**. Finally, in **Magnitude Specifications**, set **Astop** to `60`.

7 Click **Design Filter** to design the filter.

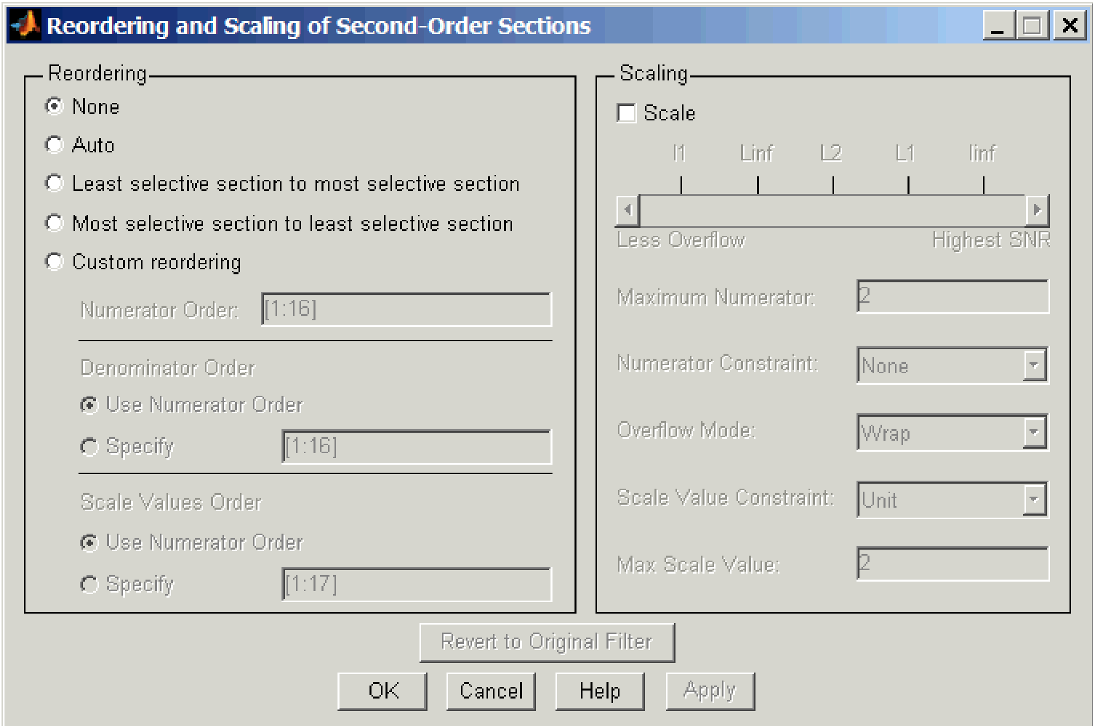
   After FDATool finishes designing the filter, you see the following plot and settings in the tool.



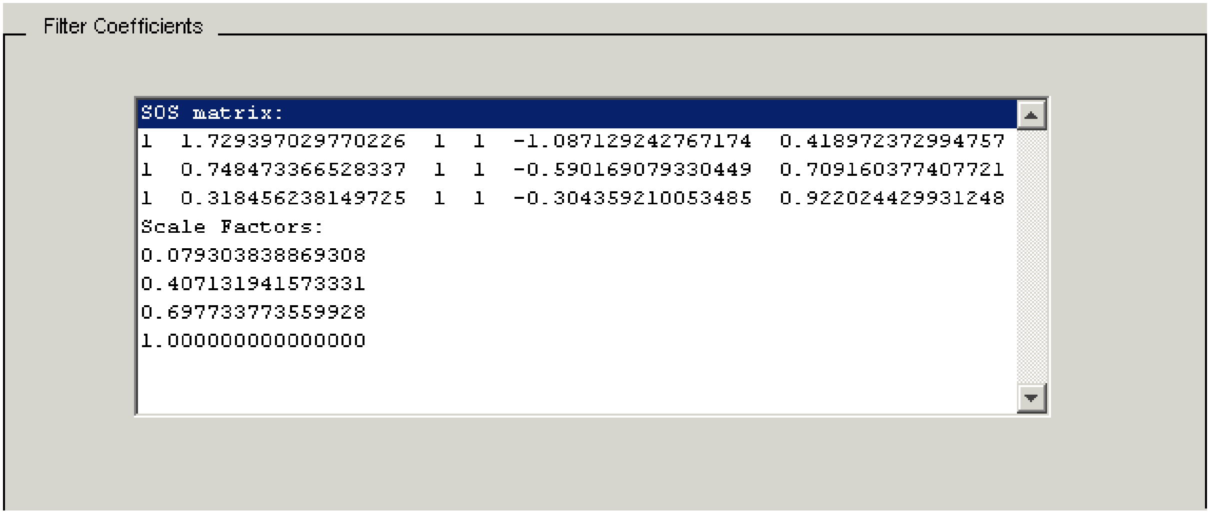
You kept the **Options** setting for **Match exactly** as `both`, meaning the filter design matches the specification for the passband and the stopband.

8 To switch to scaling the filter, select **Edit—>Reorder and Scale Second-Order Sections** from the menu bar.

Your selection opens the **Reordering and Scaling Second-Order Sections** dialog shown here.



9 To see the filter coefficients, return to FDATool and select **Filter Coefficients** from the **Analysis** menu. FDATool displays the coefficients and scale values in FDATool.



With the coefficients displayed you can see the effects of scaling your filter directly in the scale values and filter coefficients.

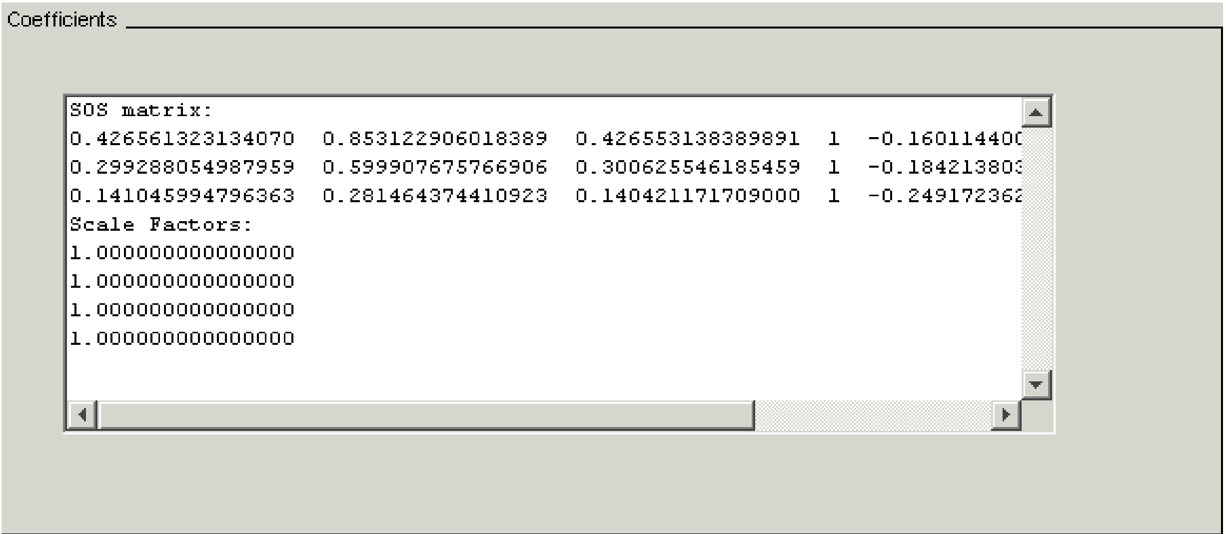
Now try scaling the filter in a few different ways. First scale the filter to maximize the SNR.

1 Return to the **Reordering and Scaling Second-Order Sections** dialog and select **None** for **Reordering** in the left pane. This prevents FDATool from reordering the filter sections when you rescale the filter.

2 Move the **No Overflow—High SNR** slider from **No Overflow** to **High SNR**.

3 Click **Apply** to scale the filter and leave the dialog open.

After a few moments, FDATool updates the coefficients displayed so you see the new scaling, as shown here.



All of the scale factors are now 1, and the SOS matrix of coefficients shows that none of the numerator coefficients are 1 and the first denominator coefficient of each section is 1.

4 Click **Revert to Original Filter** to restore the filter to the original settings for scaling and coefficients.

Reordering the Sections of Second-Order Section Filters

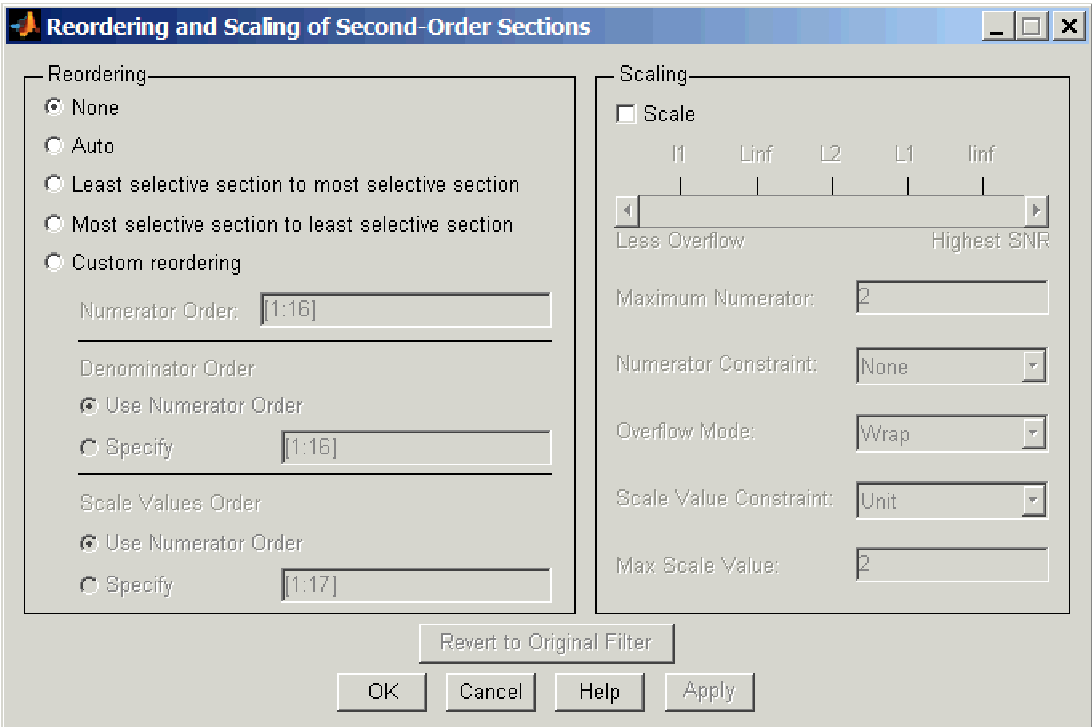
FDATool design most discrete-time filters in second-order sections. Generally, SOS filters resist the effects of quantization changes when you create fixed-point filters. After you have a second-order section filter in FDATool, either one you designed in the tool, or one you imported, FDATool provides the capability to change the order of the sections that compose the filter.

Any SOS filter in FDATool allows reordering of the sections.

Switching FDATool to Reorder Filters

To reorder the sections of a filter, you access the Reorder and Scaling of Second-Order Sections dialog in FDATool.

With your SOS filter in FDATool, select **Edit—>Reorder and Scale Second-Order Sections** from the menu bar. FDATool returns the reordering dialog shown here with the default settings.



Controls on the Reordering and Scaling of Second-Order Sections Dialog

In this dialog, the left-hand side contains options for reordering SOS filters. On the right you see the scaling options. These are independent—reordering your filter does not require scaling (note the **Scale** option) and scaling does not require that you reorder your filter (note the **None** option under **Reordering**). For more about scaling SOS filters, refer to "Scaling Second-Order Section Filters" on page 7-32 and to `scale` in the reference section.

Reordering SOS filters involves using the options in the **Reordering and Scaling of Second-Order Sections** dialog. The following table lists each reorder option and provides a description of what the option does.

Control Option	Description
Auto	Reorders the filter sections to minimize the output noise power of the filter. Note that different ordering applies to each specification type, such as lowpass or highpass. Automatic ordering adapts to the specification type of your filter.
None	Does no reordering on your filter. Selecting **None** lets you scale your filter without applying reordering at the same time. When you access this dialog with a current filter, this is the default setting—no reordering is applied.
Least selective section to most selective section	Rearranges the filter sections so the least restrictive (lowest Q) section is the first section and the most restrictive (highest Q) section is the last section.
Most selective section to least selective section	Rearranges the filter sections so the most restrictive (highest Q) section is the first section and the least restrictive (lowest Q) section is the last section.
Custom reordering	Lets you specify the section ordering to use by enabling the Numerator Order and Denominator Order options
Numerator Order	Specify new ordering for the sections of your SOS filter. Enter a vector of the indices of the sections in the order in which to rearrange them. For example, a filter with five sections has indices 1, 2, 3, 4, and 5. To switch the second and fourth sections, the vector would be [1,4,3,2,5].

Control Option	Description
Use Numerator Order	Rearranges the denominators in the order assigned to the numerators.
Specify	Lets you specify the order of the denominators, rather than using the numerator order. Enter a vector of the indices of the sections to specify the order of the denominators to use. For example, a filter with five sections has indices 1, 2, 3, 4, and 5. To switch the second and fourth sections, the vector would be [1,4,3,2,5].
Use Numerator Order	Reorders the scale values according to the order of the numerators.
Specify	Lets you specify the order of the scale values, rather than using the numerator order. Enter a vector of the indices of the sections to specify the order of the denominators to use. For example, a filter with five sections has indices 1, 2, 3, 4, and 5. To switch the second and fourth sections, the vector would be [1,4,3,2,5].
Revert to Original Filter	Returns your filter to the original section ordering. Being able to revert to your original filter makes comparing the results of changing the order of the sections easier to assess.

Example—Reorder an SOS Filter

With FDATool open and a second-order filter as the current filter, you use the following process to access the reordering capability and reorder you filter. Start by launching FDATool from the command prompt.

1. Enter `fdatool` at the command prompt to launch FDATool.

2. Design a lowpass Butterworth filter with order 10 and the default frequency specifications by entering the following settings:

   - Under **Response Type** select `Lowpass`.

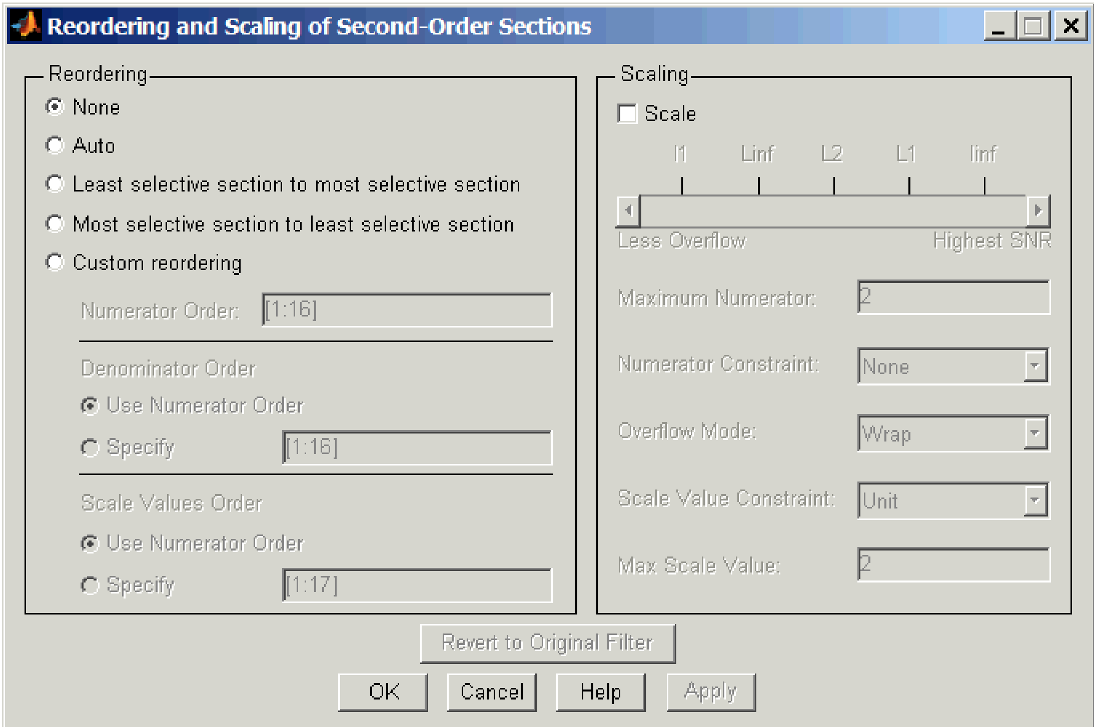
- Under **Design Method**, select **IIR** and `Butterworth` from the list.
- Specify the order equal to 10 in **Specify order** under **Filter Order**.
- Keep the default **Fs** and **Fc** values in **Frequency Specifications**.

3 Click **Design Filter**.

FDATool design the Butterworth filter and returns your filter as a Direct-Form II filter implemented with second-order sections. You see the specifications in the **Current Filter Information** area.

With the second-order filter in FDATool, reordering the filter uses the **Reordering and Scaling of Second-Order Sections** feature in FDATool (also available in Filter Visualization Tool, `fvtool`).

4 To reorder your filter, select **Edit—>Reorder and Scale Second-Order Sections** from the FDATool menus. FDATool opens the following dialog that controls reordering of the sections of your filter.



Now you are ready to reorder the sections of your filter. Note that FDATool performs the reordering on the current filter in the session.

Use Least Selective to Most Selective Section Reordering

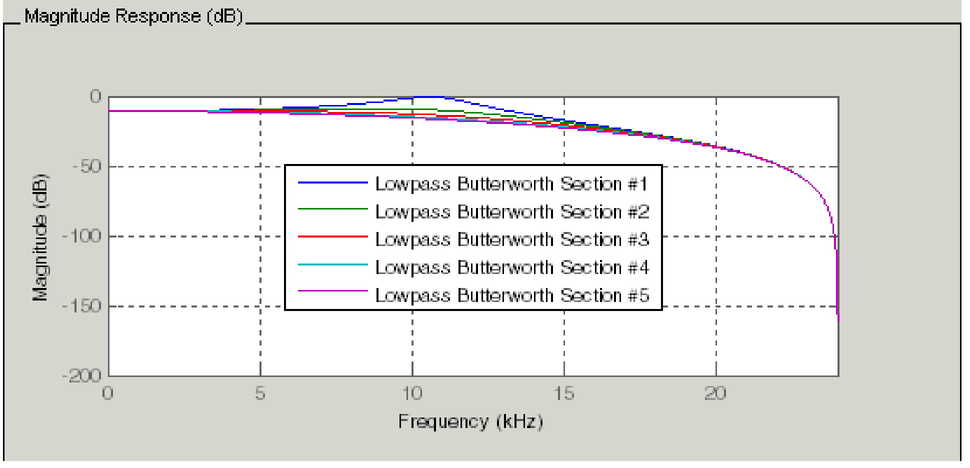
To let FDATool reorder your filter so the least selective section is first and the most selective section is last, perform the following steps in the **Reordering and Scaling of Second-Order Sections** dialog.

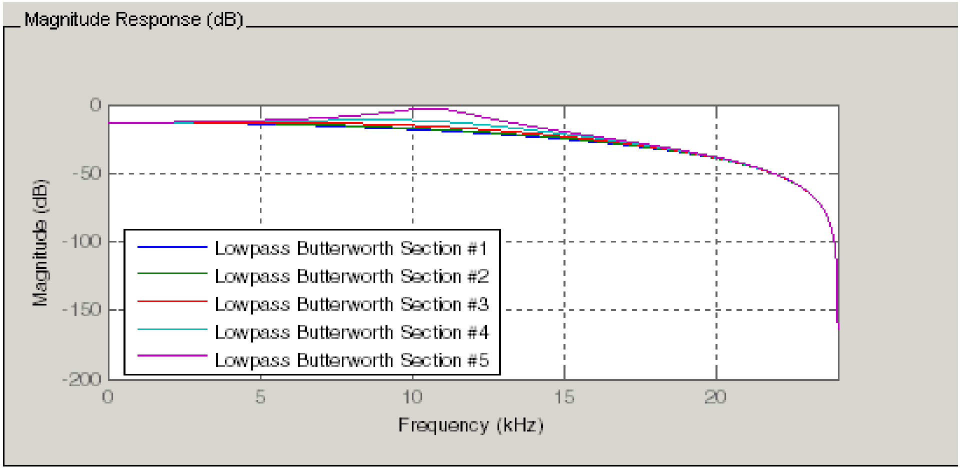
1. In **Reordering**, select **Least selective section to most selective section**.
2. To prevent filter scaling at the same time, clear **Scale** in **Scaling**.
3. In FDATool, select **View—>SOS View** from the menu bar so you see the sections of your filter displayed in FDATool.

4 In the **SOS View** dialog, select **Individual sections**. Making this choice configures FDATool to show the magnitude response curves for each section of your filter in the analysis area.

5 Back in the **Reordering and Scaling of Second-Order Sections** dialog, click **Apply** to reorder your filter according to the Qs of the filter sections, and keep the dialog open. In response, FDATool presents the responses for each filter section (there should be five sections) in the analysis area.

In the next two figures you can compare the ordering of the sections of your filter. In the first figure, your original filter sections appear. In the second figure, the sections have been rearranged from least selective to most selective.





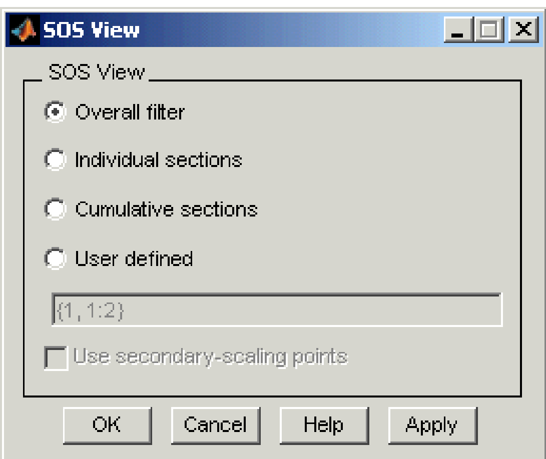
You see what reordering does, although the result is a bit subtle. Now try custom reordering the sections of your filter or using the most selective to least selective reordering option.

Viewing SOS Filter Sections

Since you can design and reorder the sections of SOS filters, FDATool provides the ability to view the filter sections in the analysis area—SOS View. Once you have a second-order section filter as your current filter in FDATool, you turn on the SOS View option to see the filter sections individually, or cumulatively, or even only some of the sections. Enabling SOS View puts FDATool in a mode where all second-order section filters display sections until you disable the SOS View option. SOS View mode applies to any analysis you display in the analysis area. For example, if you configure FDATool to show the phase responses for filters, enabling SOS View means FDATool displays the phase response for each section of SOS filters.

Controls on the SOS View Dialog

SOS View uses a few options to control how FDATool displays the sections, or which sections to display. When you select **View—>SOS View** from the FDATool menu bar, you see this dialog containing options to configure SOS View operation.



By default, SOS View shows the overall response of SOS filters. Options in the SOS View dialog let you change the display. This table lists all the options and describes the effects of each.

Option	Description
Overall Filter	This is the familiar display in FDATool. For a second-order section filter you see only the overall response rather than the responses for the individual sections. This is the default configuration.
Individual sections	When you select this option, FDATool displays the response for each section as a curve. If your filter has five sections you see five response curves, one for each section, and they are independent. Compare to **Cumulative sections**.
Cumulative sections	When you select this option, FDATool displays the response for each section as the accumulated response of all prior sections in the filter. If your filter has five sections you see five response curves: • The first curve plots the response for the first filter section. • The second curve plots the response for the combined first and second sections. • The third curve plots the response for the first, second, and third sections combined. And so on until all filter sections appear in the display. The final curve represents the overall filter response. Compare to **Cumulative sections** and **Overall Filter**.

Option	Description
User defined	Here you define which sections to display, and in which order. Selecting this option enables the text box where you enter a cell array of the indices of the filter sections. Each index represents one section. Entering one index plots one response. Entering something like {1:2} plots the combined response of sections 1 and 2. If you have a filter with four sections, the entry {1:4} plots the combined response for all four sections, whereas {1,2,3,4} plots the response for each section. Note that after you enter the cell array, you need to click **OK** or **Apply** to update the FDATool analysis area to the new SOS View configuration.
Use secondary-scaling points	This directs FDATool You use this with the **Cumulative sections** option only.

Example—View the Sections of SOS Filters

After you design or import an SOS filter in to FDATool, the SOS view option lets you see the per section performance of your filter. Enabling SOS View from the View menu in FDATool configures the tool to display the sections of SOS filters whenever the current filter is an SOS filter.

These next steps demonstrate using SOS View to see your filter sections displayed in FDATool.

1 Launch FDATool.

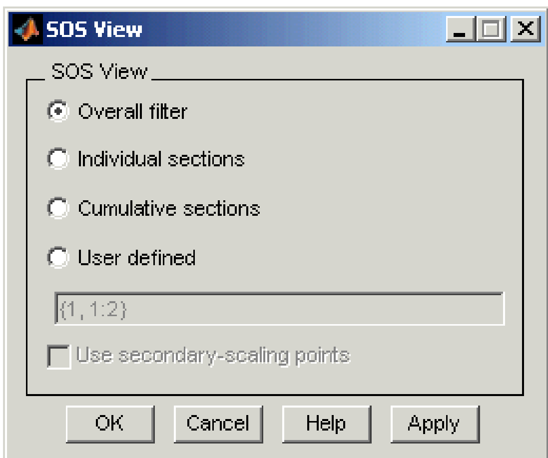
2 Create a lowpass SOS filter using the Butterworth design method. Specify the filter order to be 6. Using a low order filter makes seeing the sections more clear.

3 Design your new filter by clicking **Design Filter**.

FDATool design your filter and show you the magnitude response in the analysis area. In Current Filter Information you see the specifications for your filter. You should have a sixth-order Direct-Form II, Second-Order Sections filter with three sections.

4 To enable SOS View, select **View—>SOS View** from the menu bar.

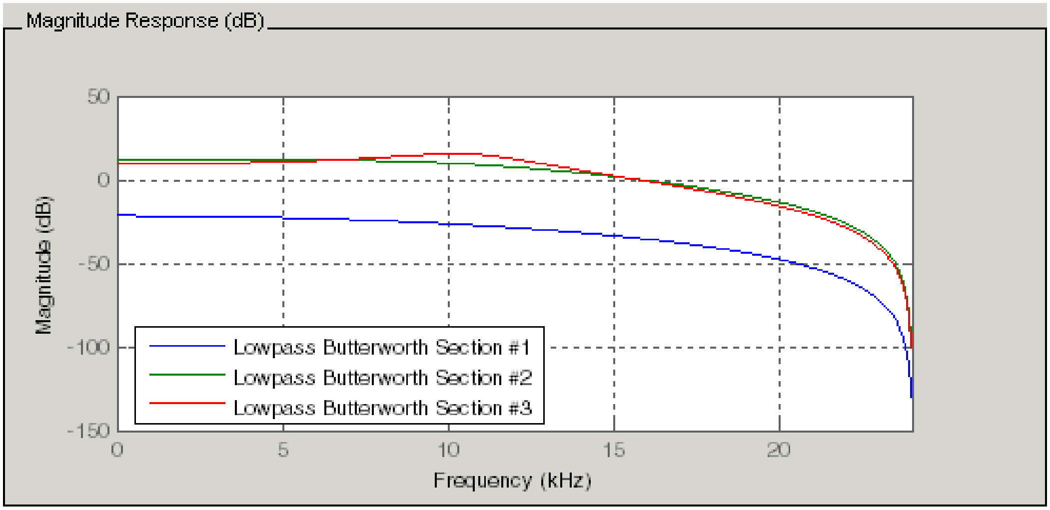
Now you see the **SOS View** dialog in FDATool. Options here let you specify how to display the filter sections.



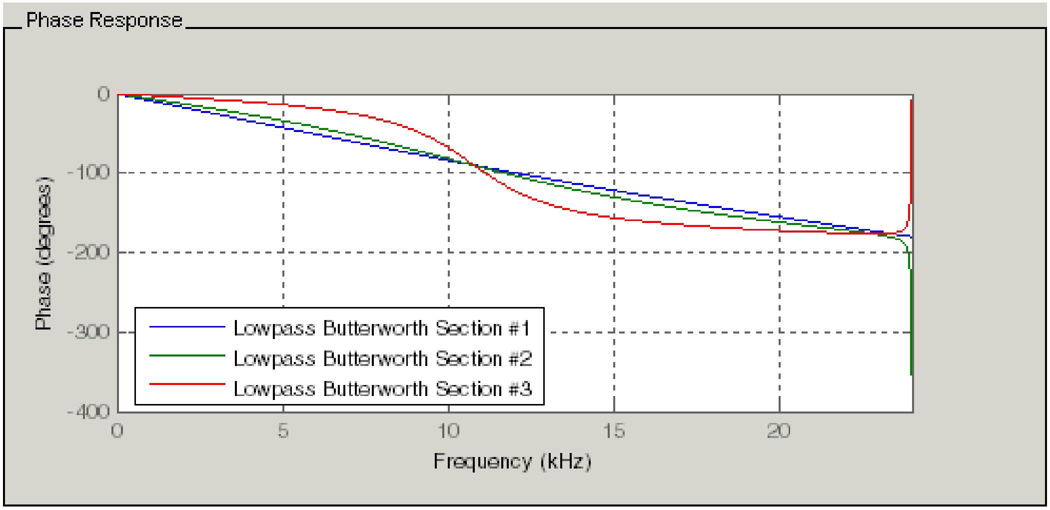
By default the analysis area in FDATool shows the overall filter response, not the individual filter section responses. This dialog lets you change the display configuration to see the sections.

5 To see the magnitude responses for each filter section, select **Individual sections**.

6 Click **Apply** to update FDATool to display the responses for each filter section. The analysis area changes to show you something like the following figure.



If you switch FDATool to display filter phase responses, you see the phase response for each filter section in the analysis area.

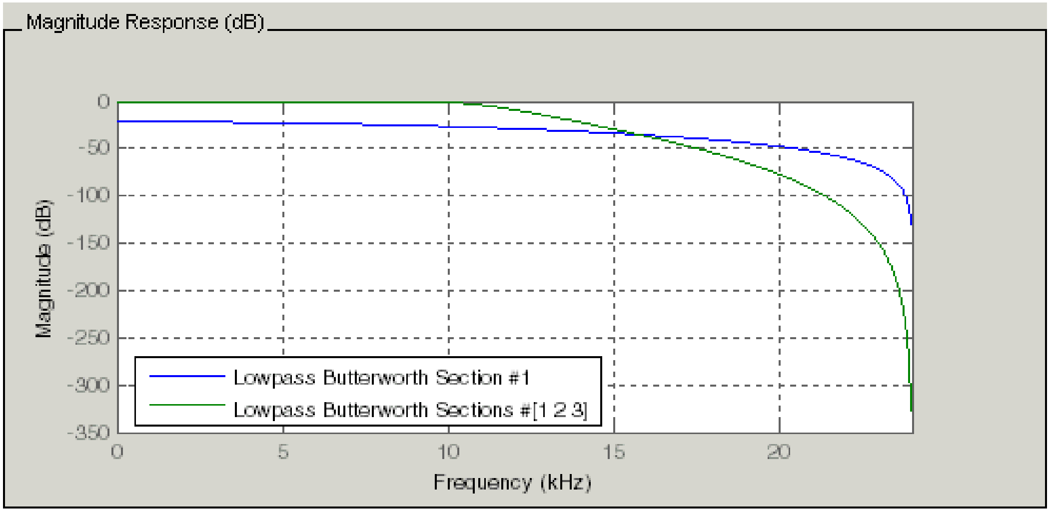


7 To define your own display of the sections, you use the **User defined** option and enter a vector of section indices to display. Now we display the first section response, and the cumulative first, second. and third sections response:

- Select **User defined** to enable the text entry box in the dialog.
- Enter the cell array `{1,1:3}` to specify that FDATool should display the response of the first section and the cumulative response of the first three sections of the filter.

8 To apply your new SOS View selection, click **Apply** or **OK** (which closes the **SOS View** dialog).

In the FDATool analysis area you see two curves—one for the response of the first filter section and one for the combined response of sections 1, 2, and 3.


Magnitude Response (dB)
Magnitude (dB)
0
-50
-100
-150
-200
-250
-300
-350
0
5
10
15
20
Frequency (kHz)
Lowpass Butterworth Section #1
Lowpass Butterworth Sections #[1 2 3]

Importing and Exporting Quantized Filters

When you import a quantized filter into FDATool, or export a quantized filter from FDATool to your workspace, the import and export functions use objects and you specify the filter as a variable. This contrasts with importing and exporting nonquantized filters, where you select the filter structure and enter the filter numerator and denominator for the filter transfer function.

You have the option of exporting quantized filters to your MATLAB workspace, exporting them to text files, or exporting them to MAT-files.

This section includes:

- "Example—Import Quantized Filters"
- "To Export Quantized Filters"

For general information about importing and exporting filters in FDATool, refer to "Filter Design and Analysis Tool" section in your *Signal Processing Toolbox User's Guide*.

FDATool imports quantized filters having the following structures:

- Direct form I
- Direct form II
- Direct form I transposed
- Direct form II transposed
- Direct form symmetric FIR
- Direct form antisymmetric FIR
- Lattice allpass
- Lattice AR
- Lattice MA minimum phase
- Lattice MA maximum phase
- Lattice ARMA
- Lattice coupled-allpass
- Lattice coupled-allpass power complementary

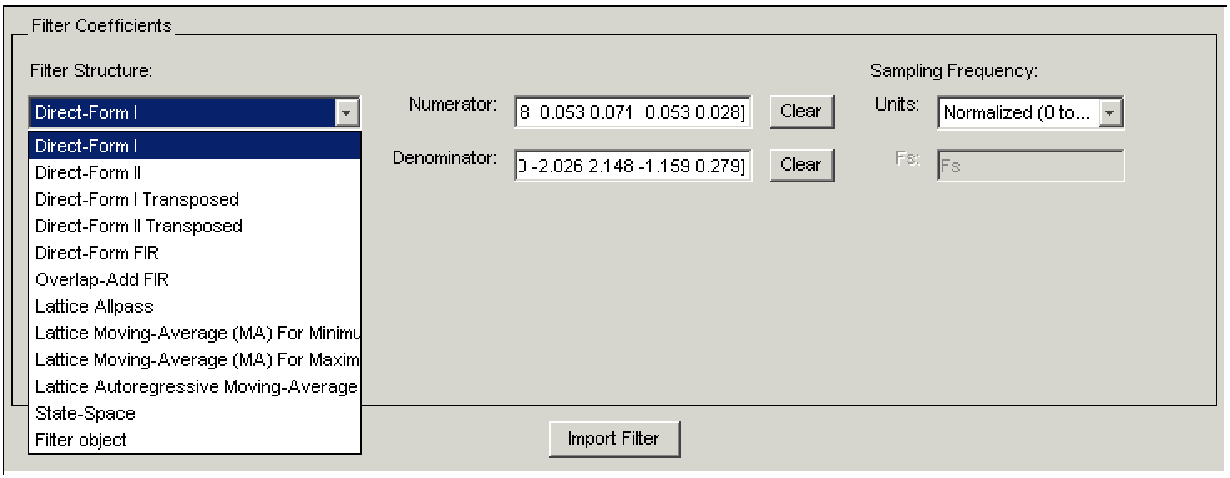
Example—Import Quantized Filters

After you design or open a quantized filter in your MATLAB workspace, FDATool lets you import the filter for analysis. Follow these steps to import your filter in to FDATool:

1 Open FDATool.

2 Select **Filter->Import Filter** from the menu bar.

In the lower region of FDATool, the **Design Filter** pane becomes **Import Filter**, and options appear for importing quantized filters, as shown.



3 From the **Filter Structure** list, select `Filter object`.

The options for importing filters change to include:

- **Discrete filter**—Enter the variable name for the discrete-time, fixed-point filter in your workspace.
- **Frequency units**—Select the frequency units from the **Units** list under **Sampling Frequency**, and specify the sampling frequency value in **Fs** if needed. Your sampling frequency must correspond to the units you select. For example, when you select `Normalized (0 to 1)`, **Fs** defaults to one. But if you choose one of the frequency options, enter the sampling

frequency in your selected units. If you have the sampling frequency defined in your workspace as a variable, enter the variable name for the sampling frequency.

4 Click **Import** to import the filter.

FDATool checks your workspace for the specified filter. It imports the filter if it finds it, displaying the magnitude response for the filter in the analysis area. If it cannot find the filter it returns an **FDATool Error** dialog.

Note If, during any FDATool session, you switch to quantization mode and create a fixed-point filter, FDATool remains in quantization mode. If you import a double-precision filter, FDATool automatically quantizes your imported filter applying the most recent quantization parameters.

When you check the current filter information for your imported filter, it will indicate that the filter is **Source:** `imported (quantized)` even though you did not import a quantized filter.

To Export Quantized Filters

To save your filter design, FDATool lets you export the quantized filter to your MATLAB workspace (or you can save the current session in FDATool). When you choose to save the quantized filter by exporting it, you select one of these options:

- Export to your MATLAB workspace
- Export to a text file
- Export to a MAT-file

Example—Export Coefficients or Objects to the Workspace

You can save the filter as filter coefficients variables or as a `dfilt` filter object variable. To save the filter to the MATLAB workspace:

1 Select **Export** from the **File** menu. The **Export** dialog appears.

2 Select `Workspace` from the **Export To** list.

3 Select `Coefficients` from the **Export As** list to save the filter coefficients or select `Objects` to save the filter in a filter object.

4 For coefficients, assign variable names using the **Numerator** and **Denominator** options under **Variable Names**. For objects, assign the variable name in the **Discrete** or **Quantized filter** option. If you have variables with the same names in your workspace and you want to overwrite them, select the **Overwrite Variables** box.

5 Click the **OK** button

   If you try to export the filter to a variable name that exists in your workspace, and you did not select **Overwrite existing variables**, FDATool stops the export operation and returns a warning that the variable you specified as the quantized filter name already exists in the workspace. To continue to export the filter to the existing variable, click **OK** to dismiss the warning dialog, select the **Overwrite existing variables** check box and click **OK** or **Apply**.

Getting Filter Coefficients after Exporting

To extract the filter coefficients from your quantized filter after you export the filter to MATLAB, use the `celldisp` function in MATLAB. For example, create a quantized filter in FDATool and export the filter as `Hq`. To extract the filter coefficients for `Hq`, use

```
celldisp(Hq.referencecoefficients)
```

which returns the cell array containing the filter reference coefficients, or

```
celldisp(Hq.quantizedcoefficients)
```

to return the quantized coefficients.

Example—Exporting as a Text File

To save your quantized filter as a text file, follow these steps:

1 Select **Export** from the **File** menu.

2 Select `Text-file` under **Export to**.

3 Click **OK** to export the filter and close the dialog. Click **Apply** to export the filter without closing the **Export** dialog. Clicking **Apply** lets you export your quantized filter to more than one name without leaving the **Export** dialog.

   The **Export Filter Coefficients to Text-file** dialog appears. This is the standard Microsoft Windows save file dialog.

4 Choose or enter a directory and filename for the text file and click **OK**.

   FDATool exports your quantized filter as a text file with the name you provided, and the MATLAB editor opens, displaying the file for editing.

Example—Exporting as a MAT-File

To save your quantized filter as a MAT-file, follow these steps:

1. Select **Export** from the **File** menu.
2. Select `MAT-file` under **Export to**.
3. Assign a variable name for the filter.
4. Click **OK** to export the filter and close the dialog. Click **Apply** to export the filter without closing the **Export** dialog. Clicking **Apply** lets you export your quantized filter to more than one name without leaving the **Export** dialog.

   The **Export Filter Coefficients to MAT-file** dialog appears. This is the standard Microsoft Windows save file dialog.

5. Choose or enter a directory and filename for the text file and click **OK**.

   FDATool exports your quantized filter as a MAT-file with the specified name.

Importing XILINX Coefficient (.COE) Files

You can import XILINX coefficients (.coe) files into FDATool to create quantized filters directly using the imported filter coefficients.

Example—Import XILINX .COE Files

To use the new import file feature:

1 Select **File->Import Filter From XILINX Coefficient (.COE) File** in FDATool.

2 In the **Import Filter From XILINX Coefficient (.COE) File** dialog, find and select the .coe file to import.

3 Click **Open** to dismiss the dialog and start the import process.

   FDATool imports the coefficient file and creates a quantized, single-section, direct-form FIR filter.

Transforming Filters

The toolbox provides functions for transforming filters between various forms. When you use FDATool with the Toolbox installed, a side bar button and a menu bar option enable you to use the **Transform Filter** panel to transform filters as well as using the command line functions.

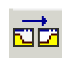
From the selection on the FDATool menu bar—**Transformations**—you can transform lowpass FIR and IIR filters to a variety of passband shapes.

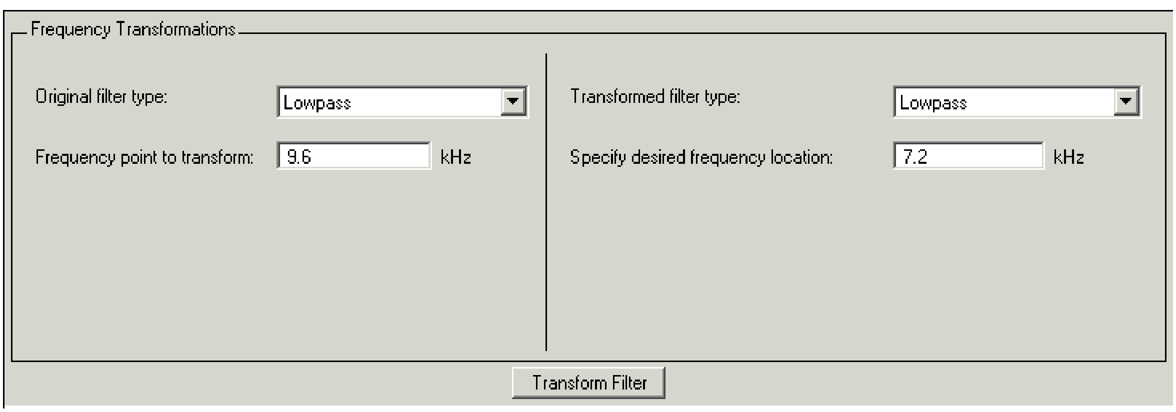
You can convert your FIR filters from:

- Lowpass to lowpass.
- Lowpass to highpass.

For IIR filters, you can convert from:

- Lowpass to lowpass.
- Lowpass to highpass.
- Lowpass to bandpass.
- Lowpass to bandstop.

When you click the **Transform Filter** button, , on the side bar, the **Transform Filter** panel opens in FDATool, as shown here.



Your options for **Original filter type** refer to the type of your current filter to transform. If you select lowpass, you can transform your lowpass filter to another lowpass filter or to a highpass filter, or to numerous other filter formats, real and complex.

Note When your original filter is an FIR filter, both the FIR and IIR transformed filter type options appear on the **Transformed filter type** list. Both options remain active because you can apply the IIR transforms to an FIR filter. If your source is as IIR filter, only the IIR transformed filter options show on the list.

Original Filter Type

Select the magnitude response of the filter you are transforming from the list. Your selection changes the types of filters you can transform to. For example:

- When you select **Lowpass** with an IIR filter, your transformed filter type can be
  - **Lowpass**
  - **Highpass**
  - **Bandpass**
  - **Bandstop**
  - **Multiband**
  - **Bandpass (complex)**
  - **Bandstop (complex)**
  - **Multiband (complex)**
- When you select **Lowpass** with an FIR filter, your transformed filter type can be
  - **Lowpass**
  - **Lowpass (FIR)**
  - **Highpass**
  - **Highpass (FIR) narrowband**
  - **Highpass (FIR) wideband**
  - **Bandpass**
  - **Bandstop**

- **Multiband**
- **Bandpass (complex)**
- **Bandstop (complex)**
- **Multiband (complex)**

In the following table you see each available original filter type and all the types of filter to which you can transform your original.

Original Filter	Available Transformed Filter Types
Lowpass FIR	• **Lowpass** • **Lowpass (FIR)** • **Highpass** • **Highpass (FIR) narrowband** • **Highpass (FIR) wideband** • **Bandpass** • **Bandstop** • **Multiband** • **Bandpass (complex)** • **Bandstop (complex)** • **Multiband (complex)**
Lowpass IIR	• **Lowpass** • **Highpass** • **Bandpass** • **Bandstop** • **Multiband** • **Bandpass (complex)** • **Bandstop (complex)** • **Multiband (complex)**

Original Filter	Available Transformed Filter Types
Highpass FIR	• **Lowpass** • **Lowpass (FIR) narrowband** • **Lowpass (FIR) wideband** • **Highpass (FIR)** • **Highpass** • **Bandpass** • **Bandstop** • **Multiband** • **Bandpass (complex)** • **Bandstop (complex)** • **Multiband (complex)**
Highpass IIR	• **Lowpass** • **Highpass** • **Bandpass** • **Bandstop** • **Multiband** • **Bandpass (complex)** • **Bandstop (complex)** • **Multiband (complex)**
Bandpass FIR	• **Bandpass** • **Bandpass (FIR)**
Bandpass IIR	**Bandpass**
Bandstop FIR	• **Bandstop** • **Bandstop (FIR)**
Bandstop IIR	**Bandstop**

Note also that the transform options change depending on whether your original filter is FIR or IIR. Starting from an IIR filter, you can transform to IIR or FIR forms. With an IIR original filter, you are limited to IIR target filters.

After selecting your response type, use **Frequency point to transform** to specify the magnitude response point in your original filter to transfer to your target filter. Your target filter inherits the performance features of your original filter, such as passband ripple, while changing to the new response form.

For more information about transforming filters, refer to “Frequency Transformations for Real Filters” on page 6-11 and “Frequency Transformations for Complex Filters” on page 6-26.

Frequency Point To Transform

The frequency point you enter in this field identifies a magnitude response value (in dB) on the magnitude response curve.

When you enter frequency values in the **Specify desired frequency location** option, the frequency transformation tries to set the magnitude response of the transformed filter to the value identified by the frequency point you enter in this field.

While you can enter any location, generally you should specify a filter passband or stopband edge, or a value in the passband or stopband.

The **Frequency point to transform** sets the magnitude response at the values you enter in **Specify desired frequency location**. Specify a value that lies at either the edge of the stopband or the edge of the passband. If, for example, you are creating a bandpass filter from a highpass filter, the transformation algorithm sets the magnitude response of the transformed filter at the **Specify desired frequency location** to be the same as the response at the **Frequency point to transform** value. Thus you get a bandpass filter whose response at the low and high frequency locations is the same. Notice that the passband between them is undefined. In the next two figures you see the original highpass filter and the transformed bandpass filter.

For more information about transforming filters, refer to “Digital Frequency Transformations” on page 6-1.

Transformed Filter Type

Select the magnitude response for the target filter from the list. The complete list of transformed filter types is:

- **Lowpass**
- **Lowpass (FIR)**
- **Highpass**
- **Highpass (FIR) narrowband**
- **Highpass (FIR) wideband**
- **Bandpass**
- **Bandstop**
- **Multiband**
- **Bandpass (complex)**
- **Bandstop (complex)**
- **Multiband (complex)**

Not all types of transformed filters are available for all filter types on the **Original filter types** list. You can transform bandpass filters only to bandpass filters. Or bandstop filters to bandstop filters. Or IIR filters to IIR filters.

For more information about transforming filters, refer to "Frequency Transformations for Real Filters" on page 6-11 and "Frequency Transformations for Complex Filters" on page 6-26.

Specify Desired Frequency Location

The frequency point you enter in **Frequency point to transform** matched a magnitude response value. At each frequency you enter here, the transformation tries to make the magnitude response the same as the response identified by your **Frequency point to transform** value.

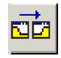
While you can enter any location, generally you should specify a filter passband or stopband edge, or a value in the passband or stopband.

For more information about transforming filters, refer to "Digital Frequency Transformations" on page 6-1.

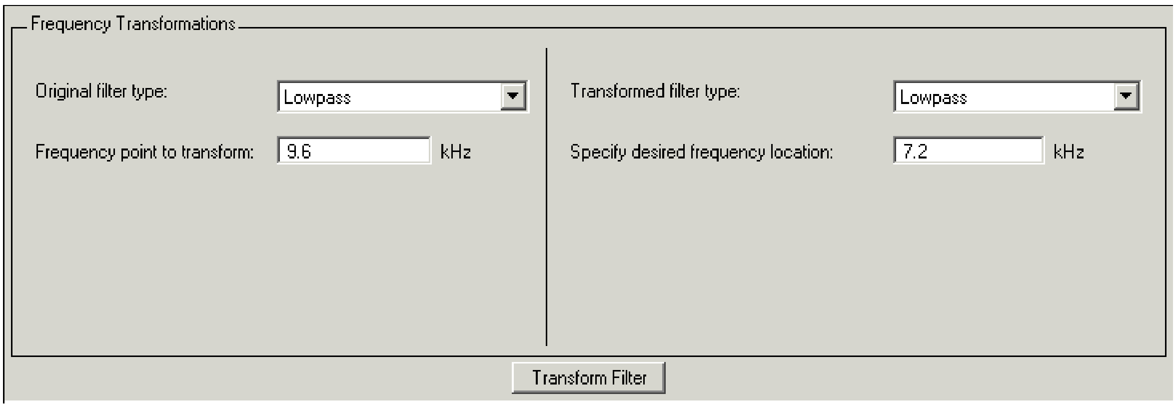
Example—Transform Filters

To transform the magnitude response of your filter, use the **Transform Filter** option on the side bar.

1 Design or import your filter into FDATool.

2 Click **Transform Filter**,  **,** on the side bar.

FDATool opens the **Transform Filter** panel in FDATool**.**



3 From the **Original filter type** list, select the response form of the filter you are transforming.

When you select the type, whether is **lowpass**, **highpass**, **bandpass**, or **bandstop**, FDATool recognizes whether your filter form is FIR or IIR. Using both your filter type selection and the filter form, FDATool adjusts the entries on the **Transformed filter type** list to show only those that apply to your original filter.

4 Enter the frequency point to transform value in **Frequency point to transform**. Notice that the value you enter must be in KHz; for example, enter 0.1 for 100 Hz or 1.5 for 1500 Hz.

5 From the **Transformed filter type** list, select the type of filter you want to transform to.

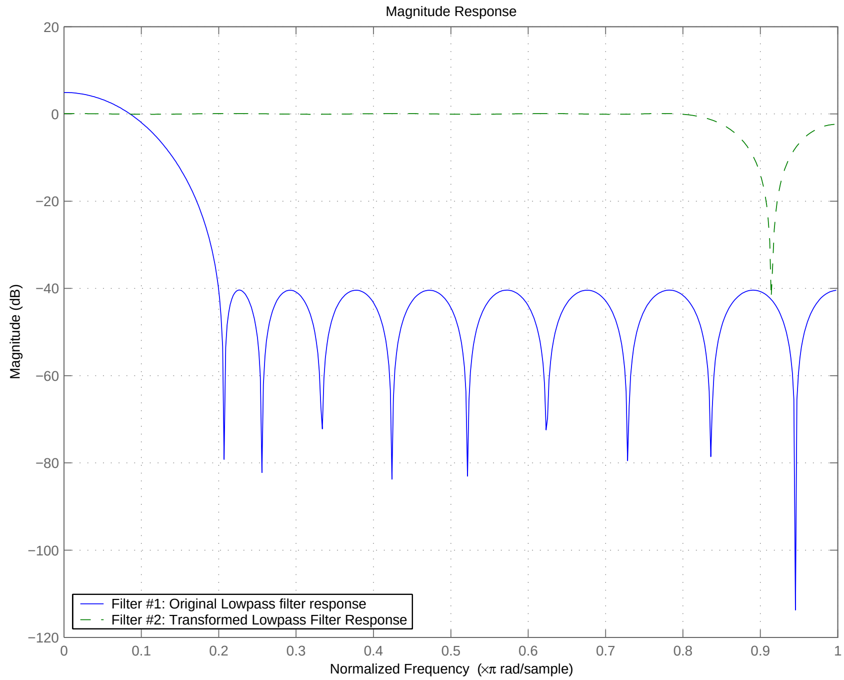
Your filter type selection changes the options here.

- When you pick a lowpass or highpass filter type, you enter one value in **Specify desired frequency location**.

- When you pick a bandpass or bandstop filter type, you enter two values— one in **Specify desired low frequency location** and one in **Specify desired high frequency location**. Your values define the edges of the passband or stopband.
- When you pick a multiband filter type, you enter values as elements in a vector in Specify a vector or desired frequency locations— one element for each desired location. Your values define the edges of the passbands and stopbands.

After you click **Transform Filter**, FDATool transforms your filter, displays the magnitude response of your new filter, and updates the **Current Filter Information** to show you that your filter has been transformed. In the filter information, the **Source** is **Transformed**.

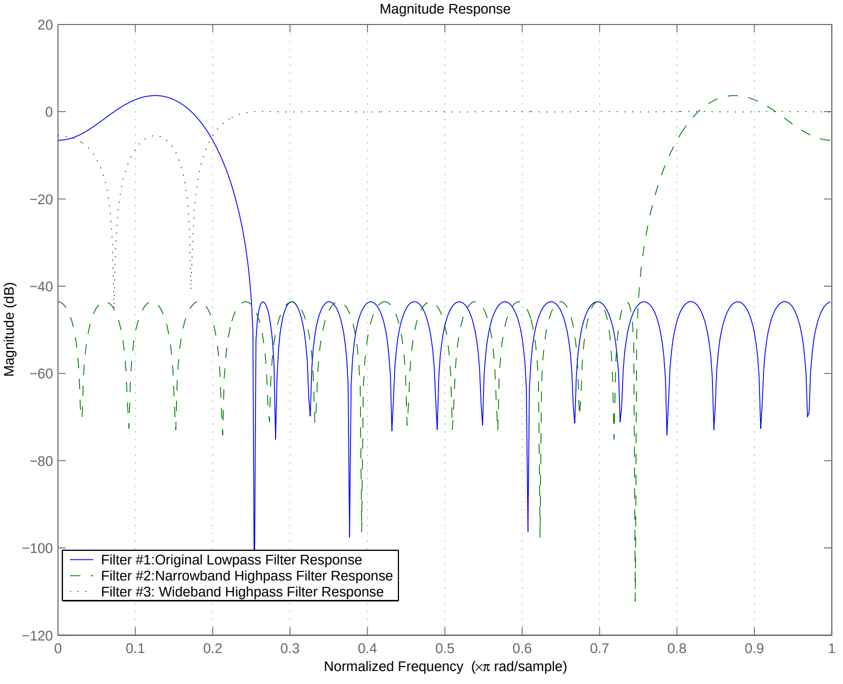
For example, the figure shown here includes the magnitude response curves for two filter. The original filter is a lowpass filter with rolloff between 0.2 and 0.25. The transformed filter is a lowpass filter with rolloff region between 0.8 and 0.85.



- To transform your lowpass filter to a highpass filter, select **Lowpass to Highpass**.

When you select **Lowpass to Highpass**, FDATool returns the dialog shown here. More information about the **Select Transform...** dialog follows the figure.

To demonstrate the effects of selecting **Narrowband Highpass** or **Wideband Highpass**, the next figure presents the magnitude response curves for a source lowpass filter after it is transformed to both narrow- and wideband highpass filters. For comparison, the response of the original filter appears as well.



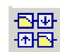
For the narrowband case, the transformation algorithm essentially reverses the magnitude response, like reflecting the curve around the *y*-axis, then translating the curve to the right until the origin lies at 1 on the *x*-axis. After reflecting and translating, the passband at high frequencies is the reverse of the passband of the original filter at low frequencies with the same rolloff and ripple characteristics.

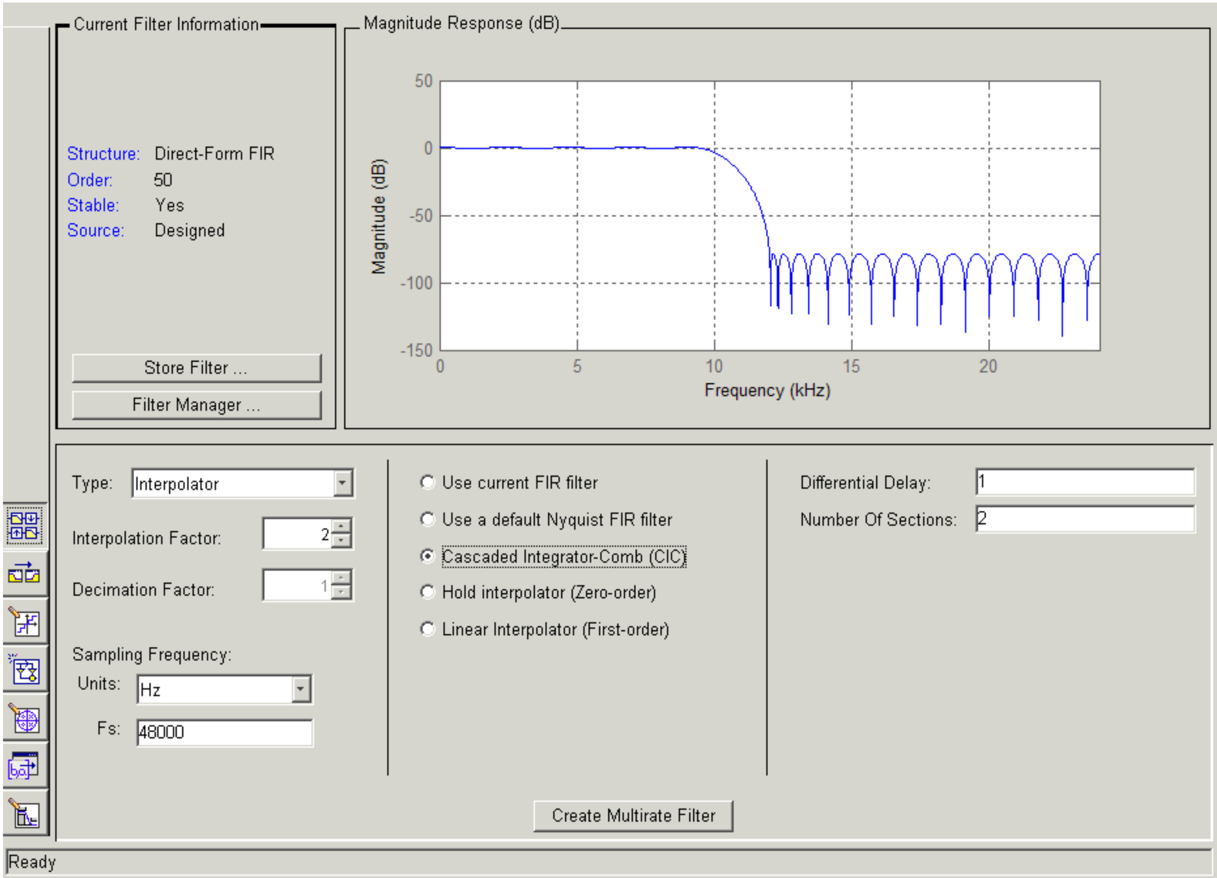
Designing Multirate Filters in FDATool

Not only can you design multirate filters from the MATLAB command prompt, FDATool provides the same design capability in a graphical user interface tool. By starting FDATool and switching to the multirate filter design mode you have access to all of the multirate design capabilities in the toolbox—decimators, interpolators, and fractional rate changing filters, among others.

Switching FDATool to Multirate Filter Design Mode

The multirate filter design mode in FDATool lets you specify and design a wide range of multirate filters, including decimators and interpolators.

With FDATool open, click **Create a Multirate Filter**,  , on the side bar. You see FDATool switch to the design mode showing the multirate filter design options. Shown in the figure below is the default multirate design configuration that designs an interpolating filter with an interpolation factor of 2. The design uses the current FIR filter in FDATool.



When the current filter in FDATool is not an FIR filter, the multirate filter design panel removes the **Use current FIR filter** option and selects the **Use default Nyquist FIR filter** option instead as the default setting.

Controls on the Multirate Design Panel

You see the options that allow you to design a variety of multirate filters. The Type option is your starting point. From this list you select the multirate filter to design. Based on your selection, other options change to provide the controls you need to specify your filter.

Notice the separate sections of the design panel. On the left is the filter type area where you choose the type of multirate filter to design and set the filter performance specifications.

In the center section FDATool provides choices that let you pick the filter design method to use.

The rightmost section offers options that control filter configuration when you select **Cascaded-Integrator Comb (CIC)** as the design method in the center section. Both the `Decimator` type and `Interpolator` type filters let you use the **Cascaded-Integrator Comb (CIC)** option to design multirate filters.

Here are all the options available when you switch to multirate filter design mode. Each option listed includes a brief description of what the option does when you use it.

Options for Selecting and Configuring Your Filter

Option	Description
Type	Specifies the type of multirate filter to design. Choose from `Decimator`, `Interpolator`, or `Fractional-rate convertor`. • When you choose `Decimator`, set **Decimation Factor** to specify the decimation to apply. • When you choose `Interpolator`, set **Interpolation Factor** to specify the interpolation amount applied. • When you choose `Fractional-rate convertor`, set both **Interpolation Factor** and **Decimation Factor**. FDATool uses both to determine the fractional rate change by dividing **Interpolation Factor** by **Decimation Factor** to determine the fractional rate change in the signal. You should select values for interpolation and decimation that are relatively prime. When your interpolation factor and decimation factor are not relatively prime, FDATool reduces the interpolation/decimation fractional rate to the lowest common denominator and issues a message in the status bar in FDATool. For example, if the interpolation factor is 6 and the decimation factor is 3, FDATool reduces 6/3 to 2/1 when you design the rate changer. But if the interpolation factor is 8 and the decimation factor is 3, FDATool designs the filter without change.
Interpolation Factor	Use the up-down control arrows to specify the amount of interpolation to apply to the signal. Factors range upwards from 2.
Decimation Factor	Use the up-down control arrows to specify the amount of decimation to apply to the signal. Factors range upwards from 2.
Sampling Frequency	No settings here. Just **Units** and **Fs** below.

Option	Description
Units	Specify whether Fs is specified in `Hz`, `kHz`, `MHz`, `GHz`, or `Normalized (0 to 1)` units.
Fs	Set the full scale sampling frequency in the frequency units you specified in **Units**. When you select `Normalized` for **Units**, you do not enter a value for **Fs**.

Options for Designing Your Filter

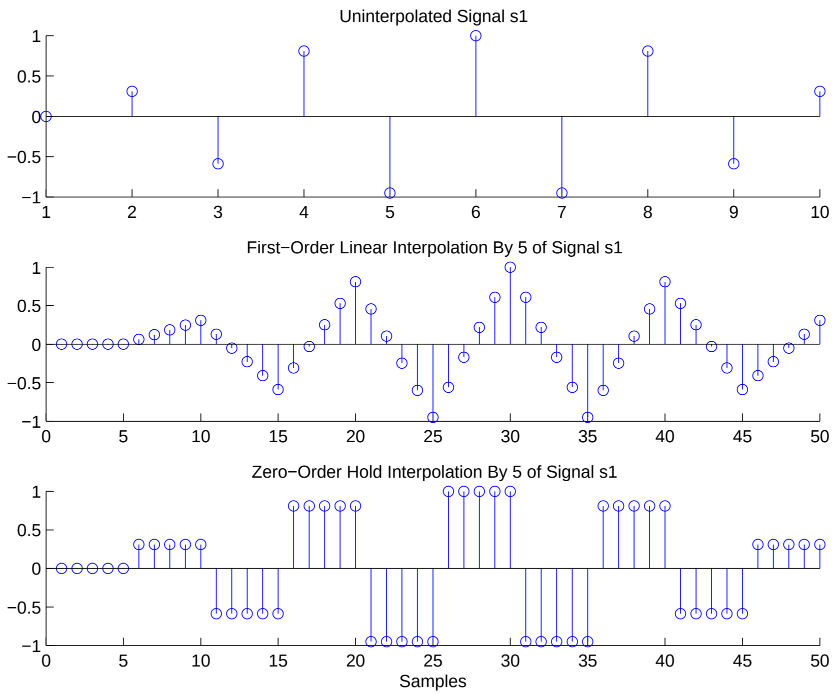
Option	Description
Use current FIR filter	Directs FDATool to use the current FIR filter to design the multirate filter. If the current filter is an IIR form, you cannot select this option. For an IIR filter, the multirate design process uses either a Nyquist filter, one of the CIC filter options, or designs an interpolator.
Use a default Nyquist Filter	Tells FDATool to use the default Nyquist design method when the current filter in FDATool is not an FIR filter.
Cascaded Integrator-Comb (CIC)	Design CIC filters using the options provided in the right-hand area of the multirate design panel.

Option	Description
Hold Interpolator (Zero-order)	When you design an interpolator, you can specify how the filter sets interpolated values between signal values. When you select this option, the interpolator applies the most recent signal value for each interpolated value until it processes the next signal value. This is similar to sample-and-hold techniques. Compare to the **Linear Interpolator** option.
Linear Interpolator (First-order)	When you design an interpolator, you can specify how the filter sets interpolated values between signal values. When you select this option, the interpolator applies linear interpolation between signal value to set the interpolated value until it processes the next signal value. Compare to the **Linear Interpolator** option.

To see the difference between hold interpolation and linear interpolation, the following figure presents a sine wave signal s1 in three forms:

- The top subplot in the figure presents s1 without interpolation.
- The middle subplot shows signal s1 interpolated by a linear interpolator with an interpolation factor of 5.
- The bottom subplot shows signal s1 interpolated by a hold interpolator with an interpolation factor of 5.

You see in the bottom figure the sample and hold nature of hold interpolation, and the first-order linear interpolation applied by the linear interpolator.



We used FDATool to create interpolators similar to the following code for the figure:

- Linear interpolator—`hm=mfilt.linearinterp(5)`
- Hold interpolator—`hm=mfilt.holdinterp(5)`

Options for Designing CIC Filters

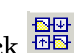
Option	Description
Differential Delay	Sets the differential delay for the CIC filter. Usually a value of one or two is appropriate.
Number of Sections	Specifies the number of sections in a CIC decimator. The default number of sections is 2 and the range is any positive integer.
Input Word Length	Specifies the word length in bits of the input samples to the CIC filter. Usually 8, 16, or 32 bits. 16 bits is the default value.
Output Word Length	Specifies the word length in bits of the output samples after filtering by the CIC filter. Ranging from 1 to 32 bits, the default is 16 bits.
Bits Per Section	Defines the number of bits per stage used while accumulating the data in the integrator stages or while subtracting the data during the comb stages (using wrap arithmetic) of a CIC decimator. Enter a positive integer number of bits. The default is 16 for each stage in the decimator. Applies only when you design CIC decimators (the **Filter Type** is `Decimator`).

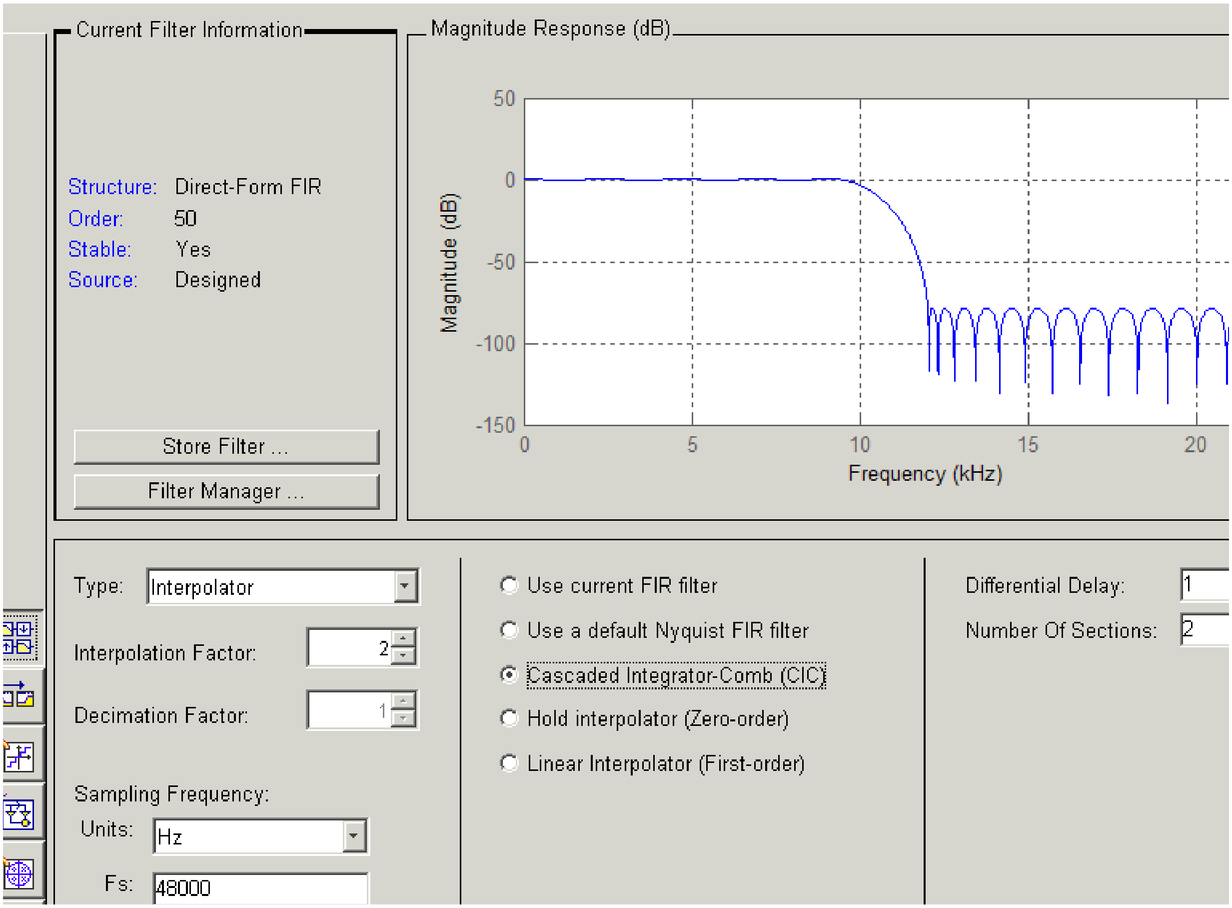
Example—Design a Fractional Rate Convertor

To introduce the process you use to design a multirate filter in FDATool, this example uses the options to design a fractional rate convertor which uses 7/3 as the fractional rate. Begin the design by creating a default lowpass FIR filter in FDATool. You do not have to begin with this FIR filter, but the default filter works fine.

1 Launch FDATool.

2 Select the settings for a minimum-order lowpass FIR filter, using the `Equiripple` design method.

3 When FDATool displays the magnitude response for the filter, click  in the side bar. FDATool switches to multirate filter design mode, showing the multirate design panel, shown here.



4 To design a fractional rate filter, select `Fractional-rate convertor` from the **Type** list. The **Interpolation Factor** and **Decimation Factor** options become available.

5 In **Interpolation Factor**, use the up arrow to set the interpolation factor to 7.

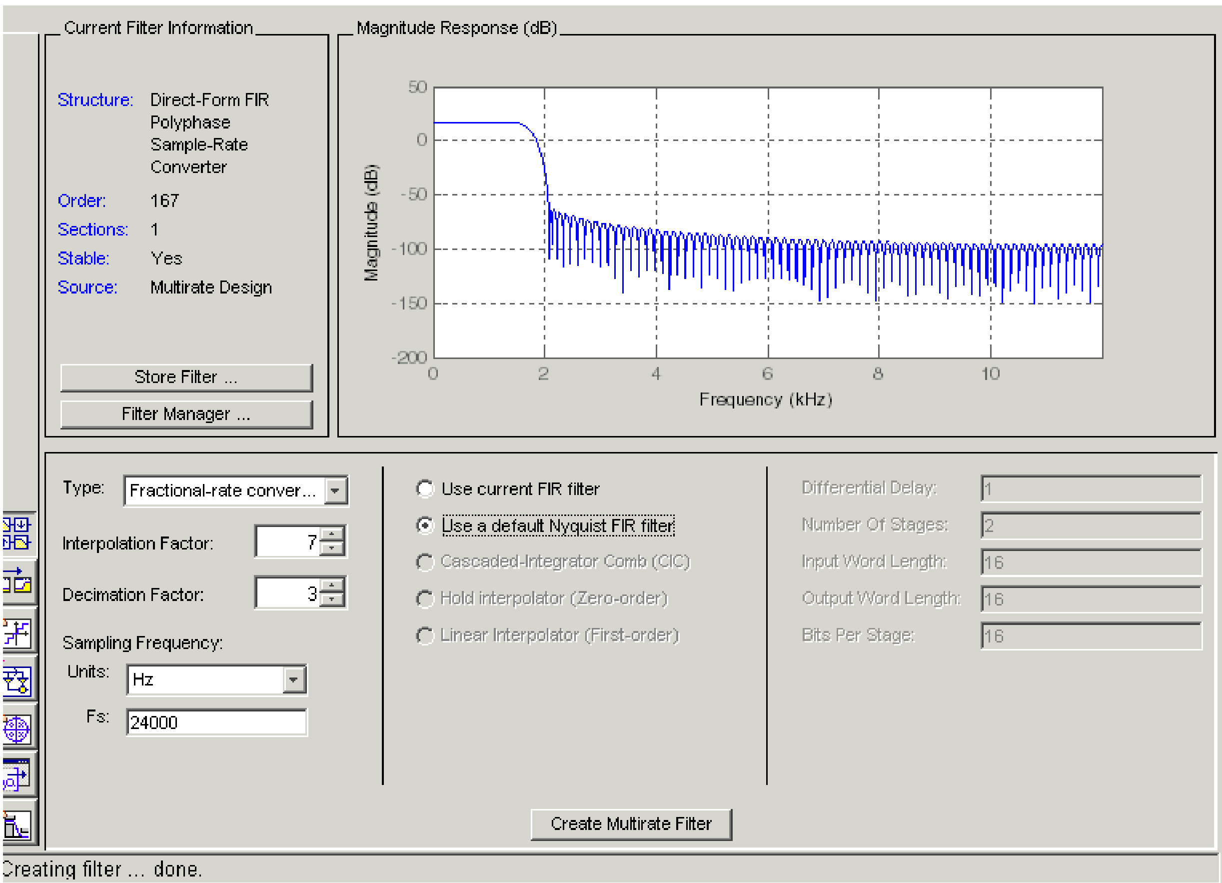
6 Using the up arrow in **Decimation Factor**, set 3 as the decimation factor.

7 Select `Use a default Nyquist FIR filter`. You could design the rate convertor with the current FIR filter as well.

8 Enter 24000 to set **Fs**.

9 Click **Design Multirate Filter**.

After designing the filter, FDATool returns with the specifications for your new filter displayed in **Current Filter Information**, and shows the magnitude response of the filter.



You can test the filter by exporting it to your workspace and using it to filter a signal. For information about exporting filters, refer to “Importing and Exporting Quantized Filters” on page 7-55.

Example—Design a CIC Decimator for 8 Bit Input/Output Data

Another kind of filter you can design in FDATool is Cascaded-Integrator Comb (CIC) filters. FDATool provides the options needed to configure your CIC to meet your needs.

1 Launch FDATool and design the default FIR lowpass filter. Designing a filter at this time is an optional step.

2 Switch FDATool to multirate design mode by clicking  on the side bar.

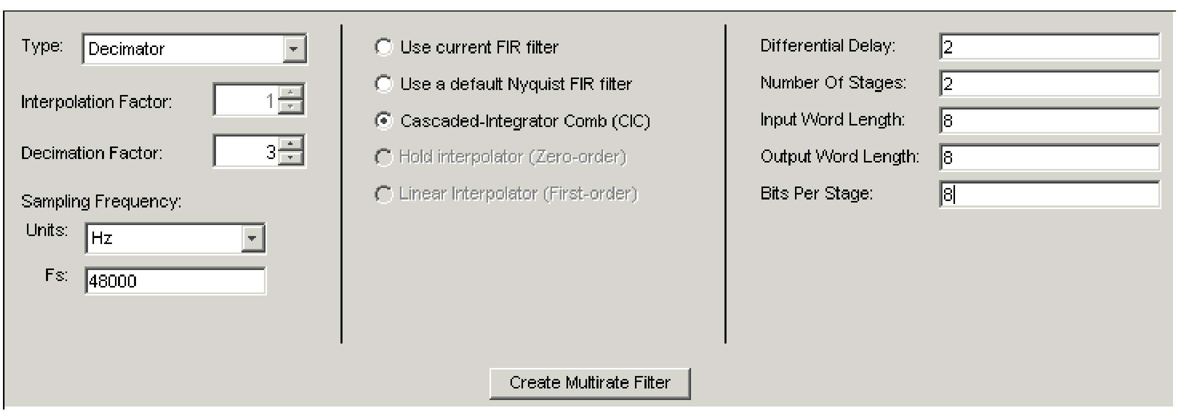
3 For **Type**, select `Decimator`, and set **Decimation Factor** to `3`.

4 To design the decimator using a CIC implementation, select **Cascaded-Integrator Comb (CIC)**. This enables the CIC-related options on the right of the panel.

5 Set Differential Delay to 2. Generally, 1 or 2 are good values to use.

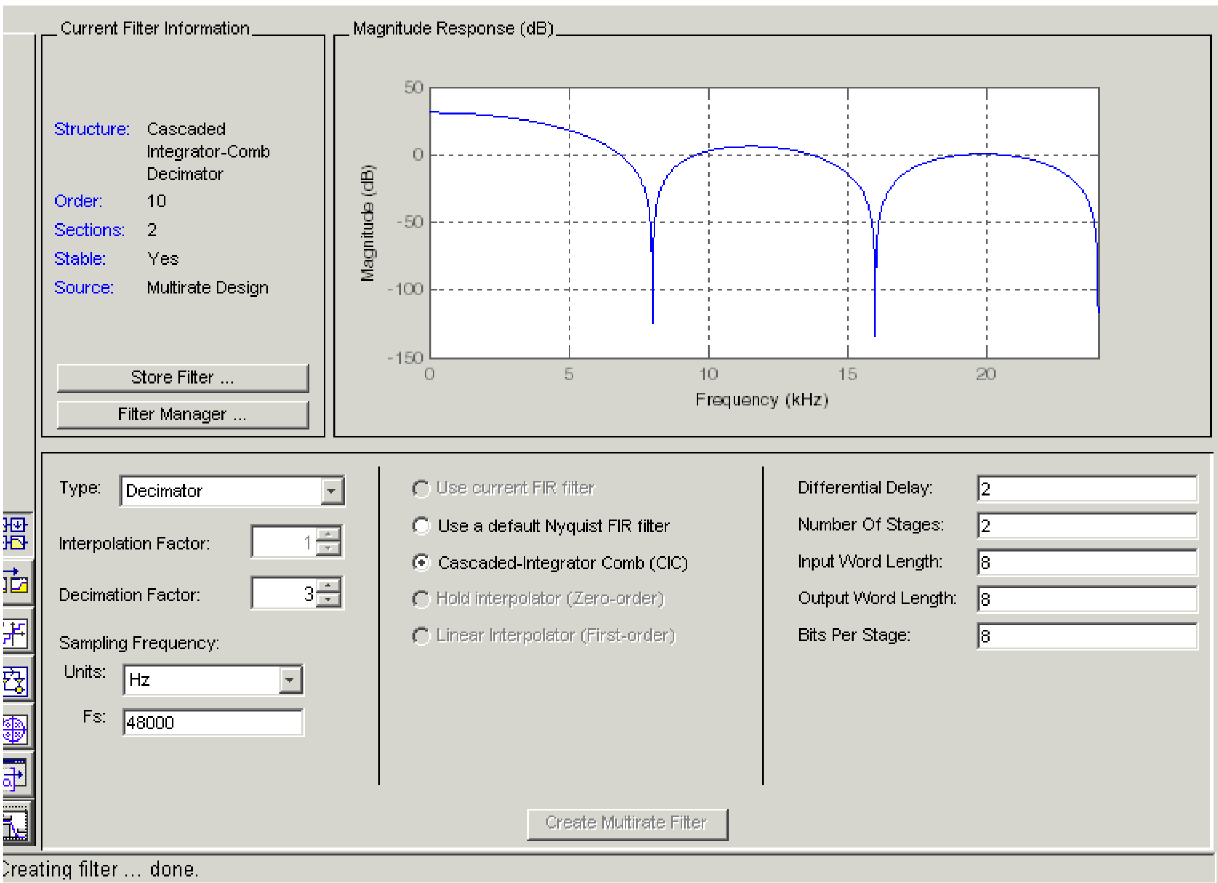
6 Enter `8` for the **Input Word Length** and **Output Word Length**.

7 For **Bits Per Section**, enter `8`. Settings in the multirate design panel should look like this.



8 Click **Design Multirate Filter**.

FDATool designs the filter, shows the magnitude response in the analysis area, and updates the current filter information to show that you designed a tenth-order cascaded-integrator comb decimator with two sections. Notice the source is Multirate Design, indicating you used the multirate design mode in FDATool to make the filter. FDATool should look like this now.



Designing other multirate filters follows the same pattern.

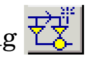
To design a multirate filter from an IIR prototype rather than from an FIR filter, do one of the following depending on the filter to design:

- To design an interpolator, select one of these options.
  - **Use a default Nyquist FIR filter**
  - **Cascaded-Integrator Comb (CIC)**
  - **Hold Interpolator (Zero-order)**
  - **Linear Interpolator (First-order)**
- To design a decimator, select from these options.
  - **Use a default Nyquist FIR filter**
  - **Cascaded-Integrator Comb (CIC)**
- To design a fractional-rate convertor, select **Use a default Nyquist FIR filter**. This is the only options for designing a rate convertor from an IIR prototype.

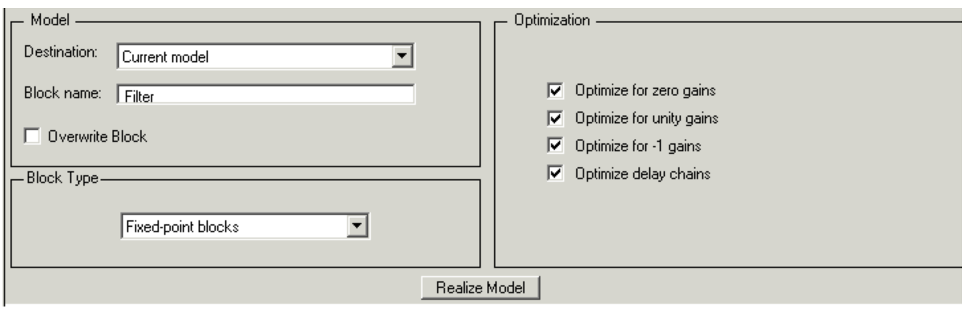
Realizing Filters as Simulink Subsystem Blocks

After you design or import a filter in FDATool, the realize model feature lets you create a Simulink subsystem block that implements your filter. The generated filter subsystem block uses the delay, gain, and sum blocks in fixed-point mode from Simulink. If you do not own Simulink Fixed Point, FDATool still realizes your model using blocks in fixed-point mode from Simulink, but you cannot run any model that includes your filter subsystem block in Simulink.

About the Realize Model Panel in FDATool

Switching FDATool to realize model mode, by clicking  on the sidebar, gives you access to the Realize Model panel and the options for realizing your quantized filter as a Simulink subsystem block.

On the panel, as shown here, are the options provided for configuring how FDATool realizes your model.



Model Options

Under **Model**, you set options that direct FDATool where to put your new subsystem block and what to name the block.

Destination. Tells FDATool whether to put the new block in your current Simulink model or open a new Simulink model and add the block to that window. Select `Current model` to add the block to your current model, or select `New model` to create a new model for the block.

Block name. Provides FDATool with a name to assign to your block. When you realize your filter as a subsystem, the resulting block shows the name you enter here as the block name, positioned just below the block.

Overwrite block. Directs FDATool whether to overwrite an existing block with this block in the destination model. The result is that the new filter realization subsystem block replaces the existing filter subsystem block. Selecting this option replaces your existing filter realization subsystem block with the one you create when you click **Realize Model**. Clearing **Overwrite block** causes FDATool to create a new block in the destination model, rather than replacing the existing block.

Block Type Option

To realize your quantized filter as a subsystem block, the most appropriate choice is to select `Fixed-point blocks` from the list. When you are licensed to use the fixed-point blocks in DSP Blockset, you have the option of realizing your model as either fixed- or floating-point blocks. Since your filter is designed to use quantized coefficients, the fixed-point blocks option usually matches your needs most closely.

You can elect to realize your filter using floating-point blocks, with the understanding that while the coefficients and gains of your filter retain their fixed-point values (the filter uses the fixed-point values for both gain and coefficients, in floating-point format), the math performed during filtering uses floating-point arithmetic and does not truly match the output of your filter running in fixed-point mode. Although realizing your quantized filter with floating-point blocks is not recommended, selecting `Floating-point blocks` from the list creates your filter from blocks in Simulink and the DSP Blockset.

If you do not own a license for the fixed-point blockset, realizing your quantized filter as a subsystem generates a subsystem block that uses fixed-point blocks, but you cannot run or edit the block. If you use the filter subsystem in a Simulink model, you cannot run the model.

Optimization Options

Four options enable you to tailor the way the realized model optimizes various filter features such as delays and gains. When you open the Realize Model panel, these options are selected by default.

Optimize for zero gains. Specify whether to remove zero-gain blocks from the realized filter.

Optimize for unity gains. Specify whether to replace unity-gain blocks with direct connections in the filter subsystem.

Optimize for -1 gains. Specify whether to replace negative unity-gain blocks with a sign change at the nearest sum block in the filter.

Optimize delay chains. Specify whether to replace cascaded chains of delay blocks with a single integer delay block to provide an equivalent delay.

Each of these options can optimize the way your filter performs in simulation and in code you might generate from your model.

Example—Realize a Filter Using FDATool

After your quantized filter in FDATool is performing the way you want, with your desired phase and magnitude response, and with the right coefficients and form, follow these steps to realize your filter as a subsystem that you can use in a Simulink model.

1 Click **Realize Model** on the sidebar to change FDATool to realize model mode.

2 From the **Destination** list under **Model**, select either:
   - `Current model`—to add the realized filter subsystem to your current model
   - `New model`—to open a new Simulink model window and add your filter subsystem to the new window

3 Provide a name for your new filter subsystem in the **Name** field.

4 Decide whether to overwrite an existing block with this new one, and select or clear **Overwrite block** to direct FDATool which way to go—overwrite or not.

5 Select `Fixed-point blocks` from the list in **Block Type**.

6 Select or clear the optimizations to apply.
   - **Optimize for zero gains**—removes zero gain blocks from the model realization

- **Optimize for unity gains**—replaces unity gain blocks with direct connections to adjacent blocks
- **Optimize for -1 gains**—replaces negative gain blocks by a change of sign at the nearest sum block
- **Optimize delay chains—**replaces cascaded delay blocks with a single delay block that produces the equivalent gain

7 Click **Realize Model** to realize your quantized filter as a subsystem block according to the settings you selected.

If you double-click the filter block subsystem created by FDATool, you see the filter implementation in Simulink model form. Depending on the options you chose when you realized your filter, and the filter you started with, you might see one or more sections, or different architectures based on the form of your quantized filter. From this point on, the subsystem filter block acts like any other block that you use in Simulink models.

Getting Help for FDATool

To find out more about the buttons or options in the FDATool dialogs, use the **What's This?** button to access context-sensitive help.

Context-Sensitive Help—The What's This? Option

To find information on a particular option or region of the dialog:

1 Click the **What's This?** button .

Your cursor changes to .

2 Click on the region or option of interest.

For example, click **Turn quantization on** to find out what this option does.

You can also select **What's this?** from the **Help** menu to launch context-sensitive help.

Additional Help for FDATool

For help about importing filters into FDATool, or for details about using FDATool to create and analyze double-precision filters, refer to the "Filter Design and Analysis Tool Overview" in your Signal Processing Toolbox documentation.

8

Reference for the Properties of Filter Objects

Overview

This chapter presents all of the properties for adaptive filters (`adaptfilt` objects), discrete-time filters (both floating-point and fixed-point `dfilt` objects), and multirate filters (`mfilt` objects).

- “Fixed-Point Filter Properties” on page 8-3
- “Adaptive Filter Properties” on page 8-108
- “Multirate Filter Properties” on page 8-122

Fixed-Point Filter Properties

There is a distinction between fixed-point filters and quantized filters—quantized filters represent a superset that includes fixed-point filters.

When `dfilt` objects have their `Arithmetic` property set to `single` or `fixed`, they are quantized filters. However, after you set the `Arithmetic` property to `fixed`, the resulting filter is both quantized and fixed-point. Fixed-point filters perform arithmetic operations without allowing the binary point to move in response to the calculation—hence the name fixed-point. You can find out more about fixed-point arithmetic in your Fixed-Point Toolbox documentation or from the Help system.

With the `Arithmetic` property set to `single`, meaning the filter uses single-precision floating-point arithmetic, the filter allows the binary point to move during mathematical operations, such as sums or products. Therefore these filters cannot be considered fixed-point filters. But they are quantized filters.

This section presents the properties for fixed-point filters, which includes all the properties for double-precision and single-precision floating-point filters as well.

Fixed-Point Objects and Filters

Fixed-point filters depend in part on fixed-point objects from the Fixed-Point Toolbox. You can see this when you display a fixed-point filter at the command prompt.

```
hd=dfilt.df2t

hd =

         FilterStructure: 'Direct-Form II Transposed'
              Arithmetic: 'double'
               Numerator: 1
             Denominator: 1
        PersistentMemory: false
                  States: [0x1 double]
     NumSamplesProcessed: 0

set(hd,'arithmetic','fixed')
```

```
hd

hd =

         FilterStructure: 'Direct-Form II Transposed'
              Arithmetic: 'fixed'
               Numerator: 1
             Denominator: 1
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
        OutputFracLength: 15

         StateWordLength: 16
          StateAutoScale: true

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'
```

Look at the States property, shown here

```
States: [1x1 embedded.fi]
```

The notation `embedded.fi` indicates that the states are being represented by fixed-point objects, usually called `fi` objects. If you take a closer look at the property `States`, you see how the properties of the fi object represent the values for the filter states.

```
hd.states

ans =

[]

              DataType: Fixed
               Scaling: BinaryPoint
                Signed: true
            WordLength: 16
        FractionLength: 15

             RoundMode: round
          OverflowMode: saturate
           ProductMode: FullPrecision
  MaxProductWordLength: 128
               SumMode: FullPrecision
      MaxSumWordLength: 128
         CastBeforeSum: true
```

To learn more about `fi` objects (fixed-point objects) in general, refer to your Fixed-Point Toolbox documentation. Commands like the following can help you get the information you are looking for:

```
docsearch(fixed-point object)
```

or

```
docsearch(fi)
```

Either command opens the Help system and searches for information about fixed-point objects in the Fixed Point Toolbox.

As inputs (data to be filtered), fixed-point filters accept both regular double-precision values and `fi` objects. Which you use depends on your needs. How your filter responds to the input data is determined by the settings of the filter properties, discussed in the next few sections.

Summary—Fixed-Point Filter Properties

Discrete-time filters in this toolbox use objects that perform the filtering and configuration of the filter. As objects, they include properties and methods (that we often call functions—not strictly the same as MATLAB functions but mostly so) to provide filtering capability. In discrete-time filters, or `dfilt` objects, many of the properties are dynamic, meaning they become available depending on the settings of other properties in the `dfilt` object or filter.

Dynamic Properties

When you use a `dfilt.`*`structure`* function to create a filter, MATLAB displays the filter properties in the command window in return (unless you end the command with a semicolon which suppresses the output display). Generally you see six or seven properties, ranging from the property `FilterStructure` to `PersistentMemory`. These first properties are always present in the filter. One of the most important properties is `Arithmetic`. The `Arithmetic` property controls all of the dynamic properties for a filter.

Dynamic properties become available when you change another property in the filter. For example, when you change the `Arithmetic` property value to `fixed`, the display now shows many more properties for the filter, all of them considered dynamic. Here is an example that uses a direct form II filter. First create the default filter:

```
hd=dfilt.df2

hd =

          FilterStructure: 'Direct-Form II'
               Arithmetic: 'double'
                Numerator: 1
              Denominator: 1
         PersistentMemory: false
                   States: [0x1 double]
      NumSamplesProcessed: 0
```

With the filter `hd` in the workspace, convert the arithmetic to fixed-point. Do this by setting the property `Arithmetic` to `fixed`. Notice the display. Instead of a few properties, the filter now has many more, each one related to a particular part of the filter and its operation. Each of the now-visible properties is dynamic.

```
hd.arithmetic='fixed'

hd =
```

```
      FilterStructure: 'Direct-Form II'
           Arithmetic: 'fixed'
            Numerator: 1
          Denominator: 1
     PersistentMemory: false
               States: [1x1 embedded.fi]
  NumSamplesProcessed: 0

      CoeffWordLength: 16
       CoeffAutoScale: true
               Signed: true

      InputWordLength: 16
      InputFracLength: 15

     OutputWordLength: 16
           OutputMode: 'AvoidOverflow'

      StateWordLength: 16
      StateFracLength: 15

          ProductMode: 'FullPrecision'

            AccumMode: 'KeepMSB'
      AccumWordLength: 40
        CastBeforeSum: true

            RoundMode: 'convergent'
         OverflowMode: 'wrap'
```

Even this list of properties is not yet complete. Changing the value of other properties such as the `ProductMode` or `CoeffAutoScale` properties may reveal even more properties that control how the filter works. Remember this feature about `dfilt` objects and dynamic properties as you review the rest of this section about properties of fixed-point filters.

An important distinction is you cannot change the value of a property unless you see the property listed in the default display for the filter. Entering the filter name at the MATLAB prompt generates the default property display for the named filter. Using `get(filtername)` does not generate the default display—it lists all of the filter properties, both those that you can change and those that are not available yet.

The following table summarizes the properties, static and dynamic, of fixed-point filters and provides a brief description of each. Full descriptions of each property, in alphabetical order, follow the table.

Property Name	Valid Values [default value]	Brief Description
`AccumFracLength`	Any positive or negative integer number of bits [29]	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters and lattice filters. IIR filters have two similar properties—`DenAccumFracLength` and `NumAccumFracLength`—that let you set the precision for numerator and denominator operations separately.
`AccumMode`	FullPrecision, KeepLSB, [KeepMSB], SpecifyPrecision	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Any positive integer number of bits [40]	Sets the word length used to store data in the accumulator/buffer.
`Arithmetic`	[Double], single, fixed	Defines the arithmetic the filter uses. Gives you the options `double`, `single`, and `fixed`. In short, this property defines the operating mode for your filter.

Property Name (Continued)	Valid Values [default value]	Brief Description
CastBeforeSum	[True] or false	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.
CoeffAutoScale	[True] or false	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to `false` enables you to change the `NumFracLength` and `DenFracLength` properties to specify the precision used.
CoeffFracLength	Any positive or negative integer number of bits [14]	Set the fraction length the filter uses to interpret coefficients. `CoeffFracLength` is not available until you set `CoeffAutoScale` to `false`. Scalar filters include this property.
CoeffWordLength	Any positive integer number of bits [16]	Specifies the word length to apply to filter coefficients.
DenAccumFracLength	Any positive or negative integer number of bits [29]	Specifies how the filter algorithm interprets the results of addition operations involving denominator coefficients. You can change the value for this property after you set `AccumMode` to `SpecifyPrecision`.
DenFracLength	Any positive or negative integer number of bits [14]	Sets the fraction length the filter uses to interpret denominator coefficients. `DenFracLength` is always available, but it is read-only until you set `CoeffAutoScale` to `false`.

Property Name (Continued)	Valid Values [default value]	Brief Description
`Denominator`	Any filter coefficient value [1]	Holds the denominator coefficients for IIR filters.
`DenProdFracLength`	Any positive or negative integer number of bits [29]	Specifies how the filter algorithm interprets the results of product operations involving denominator coefficients. You can change this property value after you set `ProductMode` to `SpecifyPrecision`.
`DenStateFracLength`	Any positive or negative integer number of bits [15]	Specifies the fraction length used to interpret the states associated with denominator coefficients in the filter.
`DenStateWordLength`	Any positive integer number of bits [16]	Specifies the word length used to represent the states associated with denominator coefficients in the filter.
`FilterInternals`	`[FullPrecision]`, `SpecifyPrecision`	Controls whether the filter sets the output word and fraction lengths, and the accumulator word and fraction lengths automatically to maintain the best precision results during filtering. The default value, `FullPrecision`, sets automatic word and fraction length determination by the filter. `SpecifyPrecision` exposes the output and accumulator related properties so you can set your own word and fraction lengths for them.
`FilterStructure`	Not applicable.	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.

Property Name (Continued)	Valid Values [default value]	Brief Description
`InputFracLength`	Any positive or negative integer number of bits [15]	Specifies the fraction length the filter uses to interpret data to be processed by the filter.
`InputWordLength`	Any positive integer number of bits [16]	Specifies the word length applied to represent input data.
`Ladder`	Any ladder coefficients in double-precision data type [1]	`latticearma` filters include this property to store the ladder coefficients.
`LadderAccumFracLength`	Any positive or negative integer number of bits [29]	`latticearma` filters use this to define the fraction length applied to values output by the accumulator that stores the results of ladder computations.
`LadderFracLength`	Any positive or negative integer number of bits [14]	`latticearma` filters use ladder coefficients in the signal flow. This property determines the fraction length used to interpret the coefficients.
`Lattice`	Any lattice structure coefficients. No default value.	Stores the lattice coefficients for lattice-based filters.
`LatticeAccumFracLength`	Any positive or negative integer number of bits [29]	Specifies how the accumulator outputs the results of operations on the lattice coefficients.
`LatticeFracLength`	Any positive or negative integer number of bits [15]	Specifies the fraction length applied to the lattice coefficients.

Property Name (Continued)	Valid Values [default value]	Brief Description
MultiplicandFracLength	Any positive or negative integer number of bits [15]	Sets the fraction length for values used in product operations in the filter. Direct-form I transposed (df1t) filter structures include this property.
MultiplicandWordLength	Any positive integer number of bits [16]	Sets the word length applied to the values input to a multiply operation (the multiplicands). The filter structure df1t includes this property.
NumAccumFracLength	Any positive or negative integer number of bits [29]	Specifies how the filter algorithm interprets the results of addition operations involving numerator coefficients. You can change the value of this property after you set AccumMode to SpecifyPrecision.
Numerator	Any double-precision filter coefficients [1]	Holds the numerator coefficient values for the filter.
NumFracLength	Any positive or negative integer number of bits [14]	Sets the fraction length used to interpret the numerator coefficients.
NumProdFracLength	Any positive or negative integer number of bits [29]	Specifies how the filter algorithm interprets the results of product operations involving numerator coefficients. You can change the property value after you set ProductMode to SpecifyPrecision.

Property Name (Continued)	Valid Values [default value]	Brief Description
NumSamplesProcessed	Starts at 0 samples and grows to any number of samples processed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is true, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to false causes this property to report the total number of samples processed for all data sets, not each one.
NumStateFracLength	Any positive or negative integer number of bits [15]	For IIR filters, this defines the fraction length applied to the numerator states of the filter. Specifies the fraction length used to interpret the states associated with numerator coefficients in the filter.
NumStateWordLength	Any positive integer number of bits [16]	For IIR filters, this defines the word length applied to the numerator states of the filter. Specifies the word length used to interpret the states associated with numerator coefficients in the filter.
OutputFracLength	Any positive or negative integer number of bits—[15] or [12] bits depending on the filter structure	Determines how the filter interprets the filtered data. You can change the value of OutputFracLength after you set OutputMode to SpecifyPrecision.

Property Name (Continued)	Valid Values [default value]	Brief Description
`OutputMode`	[AvoidOverflow], BestPrecision, SpecifyPrecision	Sets the mode the filter uses to scale the filtered input data. You have the following choices: - `AvoidOverflow`—directs the filter to set the output data fraction length to avoid causing the data to overflow. - `BestPrecision`—directs the filter to set the output data fraction length to maximize the precision in the output data. - `SpecifyPrecision`—lets you set the fraction length used by the filtered data.
`OutputWordLength`	Any positive integer number of bits [16]	Determines the word length used for the filtered data.
`OverflowMode`	Saturate or [wrap]	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either `saturate` (limit the output to the largest positive or negative representable value) or `wrap` (set overflowing values to the nearest representable value using modular arithmetic. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name (Continued)	Valid Values [default value]	Brief Description
ProductFracLength	Any positive or negative integer number of bits [29]	For the output from a product operation, this sets the fraction length used to interpret the numeric data. This property becomes writable (you can change the value) after you set ProductMode to SpecifyPrecision.
ProductMode	[FullPrecision], KeepLSB, KeepMSB, SpecifyPrecision	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Any positive number of bits. Default is 16 or 32 depending on the filter structure	Specifies the word length to use for the results of multiplication operations. This property becomes writable (you can change the value) after you set ProductMode to SpecifyPrecision.
PersistentMemory	True or [false]	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. True is the default setting.

Property Name (Continued)	Valid Values [default value]	Brief Description
`RoundMode`	[Convergent], ceil, fix, floor, round	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • `convergent`—Round up to the next allowable quantized value. • `ceil`—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to `1`. • `fix`—Round negative numbers up and positive numbers down to the next allowable quantized value. • `floor`—Round down to the next allowable quantized value. • `round`—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Property Name (Continued)	Valid Values [default value]	Brief Description
ScaleValueFracLength	Any positive or negative integer number of bits [29]	Scale values work with SOS filters. Setting this property controls how your filter interprets the scale values by setting the fraction length. Available only when you disable CoeffAutoScale by setting it to false.
ScaleValues	[2 x 1 double] array with values of 1	Stores the scaling values for sections in SOS filters.
Signed	[True] or false	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
sosMatrix	[1 0 0 1 0 0]	Holds the filter coefficients as property values. Displays the matrix in the format [sections x coefficients/section datatype]. A [15x6 double] SOS matrix represents a filter with 6 coefficients per section and 15 sections, using data type double to represent the coefficients.
SectionInputAutoScale	[True] or false	Specifies whether the filter automatically chooses the proper fraction length to prevent overflow by data entering a section of an SOS filter. Setting this property to false enables you to change the SectionInputFracLength property to specify the precision used. Available only for SOS filters.

Property Name (Continued)	Valid Values [default value]	Brief Description
`SectionInputFracLength`	Any positive or negative integer number of bits [29]	Section values work with SOS filters. Setting this property controls how your filter interprets the section values between sections of the filter by setting the fraction length. This applies to data entering a section. Compare to `SectionOutputFracLength`. Available only when you disable `SectionInputAutoScale` by setting it to `false`.
`SectionInputWordLength`	Any positive or negative integer number of bits [29]	Sets the word length used to represent the data moving into a section of an SOS filter.
`SectionOutputAutoScale`	[True] or false	Specifies whether the filter automatically chooses the proper fraction length to prevent overflow by data leaving a section of an SOS filter. Setting this property to `false` enables you to change the `SectionOutputFracLength` property to specify the precision used.
`SectionOutputFracLength`	Any positive or negative integer number of bits [29]	Section values work with SOS filters. Setting this property controls how your filter interprets the section values between sections of the filter by setting the fraction length. This applies to data leaving a section. Compare to `SectionInputFracLength`. Available after you disable `SectionOutputAutoScale` by setting it to `false`.
`SectionOutputWordLength`	Any positive or negative integer number of bits [32]	Sets the word length used to represent the data moving out of one section of an SOS filter.

Property Name (Continued)	Valid Values [default value]	Brief Description
`StateFracLength`	Any positive or negative integer number of bits [15]	Lets you set the fraction length applied to interpret the filter states.
`States`	[1x1 embedded `fi`]	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use `fi` objects, with the associated properties from those objects. For details, refer to `filtstates` in your Signal Processing Toolbox documentation or in the Help system.
`StateWordLength`	Any positive integer number of bits [16]	Sets the word length used to represent the filter states.
`TapSumFracLength`	Any positive or negative integer number of bits [15]	Sets the fraction length used to represent the filter tap values in addition operations. This is available after you set `TapSumMode` to `false`. Symmetric and antisymmetric FIR filters include this property.

Property Name (Continued)	Valid Values [default value]	Brief Description
`TapSumMode`	FullPrecision, KeepLSB, [KeepMSB], SpecifyPrecision	Determines how the accumulator outputs stored that involve filter tap weights. Choose from full precision (`FullPrecision`) to prevent overflows, or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when outputting results from the accumulator. To let you set the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`. Symmetric and antisymmetric FIR filters include this property.
`TapSumWordLength`	Any positive number of bits [17]	Sets the word length used to represent the filter tap weights during addition. Symmetric and antisymmetric FIR filters include this property.

Property Details for Fixed-Point Filters

When you create a fixed-point filter, you are creating a filter object (a `dfilt` object). In this manual, we use filter, `dfilt` object, and filter object interchangeably. To filter data, you apply the filter object to your data set. The output of the operation is the data filtered by the filter and the filter property values.

Filter objects have properties to which you assign property values. You use these property values to assign various characteristics to the filters you create, including

- The type of arithmetic to use in filtering operations
- The structure of the filter used to implement the filter (not a property you can set or change—you select it by the `dfilt.`*`structure`* function you choose)
- The locations of quantizations and cast operations in the filter
- The data formats used in quantizing, casting, and filtering operations

Details of the properties associated with fixed-point filters are described in alphabetical order on the following pages.

AccumFracLength

Except for state-space filters, all `dfilt` objects that use fixed arithmetic have this property that defines the fraction length applied to data in the accumulator. Combined with `AccumWordLength` and `AccumMode`, `AccumFracLength` helps fully specify how the accumulator outputs data after processing addition operations. As with all fraction length properties, `AccumFracLength` can be any integer, including integers larger than `AccumWordLength`, and positive or negative integers.

AccumMode

This property, available after your filter is in fixed-point mode, specifies how the filter outputs the results of summation operations. All `dfilt` objects include this property when they use fixed-point arithmetic.

When you switch to fixed-point filtering from floating-point, you are most likely going to throw away some data bits after sum operations, perhaps because you have limited resources or the accumulator allows more bits (often 40 bits) than the filter itself, perhaps 32 bits or 16 bits.

When you have to change the word length of results from the accumulator, you might choose to discard the least significant bits (LSB) from the result because the quantization error caused by discarding the LSBs would be small since the LSBs carry less weight.

Or you might choose to discard the most significant bits (MSB) from the result because the results have MSBs that are mostly zero—your result values are small relative to the range of the format in which they are represented. The options for `AccumMode` let you choose how to maintain the information you need from the accumulator, or which bits to discard.

You select from one of the following values for `AccumMode`:

- `FullPrecision`—means the filter automatically chooses the word length to represent the results of sum operations so they retain all of the accuracy provided by the inputs (addends). This setting prevents overflows.

- KeepMSB—means you specify the word length for representing summation results. The filter sets the fraction length to discard the LSBs, keep the higher order bits in the data, and maintain the precision.
- KeepLSB—means you specify the word length for representing the sum operation results. The filter sets the fraction length to discard the MSBs, keep the lower order bits, and maintain the precision. Compare to the KeepMSB option.
- SpecifyPrecision—means you specify the word length and the fraction length to apply to data output from the accumulator.

For more information about data formats, word length, and fraction length in fixed-point arithmetic, refer to "Notes About Fraction Length, Word Length, and Precision" on page 8-32.

AccumWordLength

You use AccumWordLength to define the data word length used in the accumulator. Set this property to a value that matches your intended hardware. For example, many digital signal processors use 40-bit accumulators, so set AccumWordLength to 40 in your fixed-point filter:

```
set(hq,'arithmetic','fixed');
set(hq,'AccumWordLength',40);
```

Note that AccumWordLength only applies to filters whose Arithmetic property value is fixed.

Arithmetic

Perhaps the most important property when you are working with dfilt objects, Arithmetic determines the type of arithmetic the filter uses, and the properties or quantizers that compose the fixed-point or quantized filter. You use strings to set the Arithmetic property value.

The next table shows the valid strings for the Arithmetic property. Following the table, each property string appears with more detailed information about what happens when you select the string as the value for `Arithmetic` in your `dfilt`.

Arithmetic Property String	Brief Description of Effect on the Filter
`double`	All filtering operations and coefficients use double-precision floating-point representations and math. When you use `dfilt.`*`structure`* to create a filter object, `double` is the default value for the `Arithmetic` property.
`single`	All filtering operations and coefficients use single-precision floating-point representations and math.
`fixed`	This string applies selected default values for the properties in the fixed-point filter object, including such properties as coefficient word lengths, fraction lengths, and various operating modes. Generally, the default values match those you use on many digital signal processors. Allows signed fixed data types only. Fixed-point arithmetic filters are available only when you install the Fixed-Point Toolbox with this toolbox.

double

When you use one of the `dfilt.`*`structure`* methods to create a filter, the `Arithmetic` property value is `double` by default. Your filter is identical to the same filter without the `Arithmetic` property, as you would create if you used the Signal Processing Toolbox.

`Double` means that the filter uses double-precision floating-point arithmetic in all operations while filtering:

- All input to the filter must be double data type. Any other data type returns an error.
- The states and output are doubles as well.
- All internal calculations are done in double math.

When you use `double` data type filter coefficients, the reference and quantized (fixed-point) filter coefficients are identical. The filter stores the reference coefficients as double data type.

single

When your filter should use single-precision floating-point arithmetic, set the `Arithmetic` property to `single` so all arithmetic in the filter processing gets restricted to single-precision data type.

- Input data must be single data type. Other data types return errors.
- The filter states and filter output use single data type.

When you choose `single`, you can provide the filter coefficients in either of two ways:

- Double data type coefficients. With `Arithmetic` set to `single`, the filter casts the double data type coefficients to single data type representation.
- Single data type. These remain unchanged by the filter.

Depending on whether you specified single or double data type coefficients, the reference coefficients for the filter are stored in the data type you provided. If you provide coefficients in double data type, the reference coefficients are double as well. Providing single data type coefficients generates single data type reference coefficients. Note that the arithmetic used by the reference filter is always double.

When you use `reffilter` to create a reference filter from the reference coefficients, the resulting filter uses double-precision versions of the reference filter coefficients.

To set the `Arithmetic` property value, create your filter, then use `set` to change the `Arithmetic` setting, as shown in this example using a direct form FIR filter.

```
b=fir1(7,0.45);

hd=dfilt.dffir(b)
```

```
hd =

          FilterStructure: 'Direct-Form FIR'
               Arithmetic: 'double'
                Numerator: [1x8 double]
         PersistentMemory: false
                   States: [7x1 double]
      NumSamplesProcessed: 0

set(hd,'arithmetic','single')
hd

hd =

          FilterStructure: 'Direct-Form FIR'
               Arithmetic: 'single'
                Numerator: [1x8 double]
         PersistentMemory: false
                   States: [7x1 single]
      NumSamplesProcessed: 0
```

fixed

Converting your `dfilt` object to use fixed arithmetic results in a filter structure that uses properties and property values to match how the filter would behave on digital signal processing hardware.

Note The `fixed` option for the property `Arithmetic` is available only when you install the Fixed-Point Toolbox as well as the Filter Design Toolbox.

After you set `Arithmetic` to `fixed`, you are free to change any property value from the default value to a value that more closely matches your needs. You cannot, however, mix floating-point and fixed-point arithmetic in your filter when you select `fixed` as the `Arithmetic` property value. Choosing `fixed` restricts you to using either fixed-point or floating point throughout the filter (the data type must be homogenous). Also, all data types must be signed. `fixed` does not support unsigned data types except for unsigned coefficients when you set the property `Signed` to `false`. Mixing word and fraction lengths within the fixed object is acceptable. In short, using fixed arithmetic assumes

- fixed word length.
- fixed size and dedicated accumulator and product registers.
- the ability to do either saturation or wrap arithmetic.
- that multiple rounding modes are available.

Making these assumptions simplifies your job of creating fixed-point filters by reducing repetition in the filter construction process, such as only requiring you to enter the accumulator word size once, rather than for each step that uses the accumulator.

Default property values are a starting point in tailoring your filter to common hardware, such as choosing 40-bit word length for the accumulator, or 16-bit words for data and coefficients.

In this `dfilt` object example, `get` returns the default values for `dfilt.df1t` structures.

```
[b,a]=butter(6,0.45);
hd=dfilt.df1(b,a)

hd =

         FilterStructure: 'Direct-Form I'
              Arithmetic: 'double'
               Numerator: [1x7 double]
             Denominator: [1x7 double]
        PersistentMemory: false
                  States: Numerator:  [6x1 double]
                          Denominator:[6x1 double]
     NumSamplesProcessed: 0


set(hd,'arithmetic','fixed')
get(hd)
        PersistentMemory: false
     NumSamplesProcessed: 0
         FilterStructure: 'Direct-Form I'
                  States: [1x1 filtstates.dfiir]
```

```
           Numerator: [1x7 double]
         Denominator: [1x7 double]
          Arithmetic: 'fixed'
      CoeffWordLength: 16
       CoeffAutoScale: 1
               Signed: 1
            RoundMode: 'convergent'
         OverflowMode: 'wrap'
      InputWordLength: 16
      InputFracLength: 15
          ProductMode: 'FullPrecision'
            AccumMode: 'KeepMSB'
     OutputWordLength: 16
     OutputFracLength: 15
        NumFracLength: 16
        DenFracLength: 14
    ProductWordLength: 32
    NumProdFracLength: 31
    DenProdFracLength: 29
      AccumWordLength: 40
   NumAccumFracLength: 31
   DenAccumFracLength: 29
        CastBeforeSum: 1
```

Here is the default display for hd.

```
hd

hd =

         FilterStructure: 'Direct-Form I'
              Arithmetic: 'fixed'
               Numerator: [1x7 double]
             Denominator: [1x7 double]
        PersistentMemory: false
                  States: Numerator:  [6x1 fi]
                          Denominator:[6x1 fi]
     NumSamplesProcessed: 0


         CoeffWordLength: 16
```

```
           CoeffAutoScale: true
                   Signed: true

          InputWordLength: 16
          InputFracLength: 15

         OutputWordLength: 16
         OutputFracLength: 15

              ProductMode: 'FullPrecision'

                AccumMode: 'KeepMSB'
          AccumWordLength: 40
            CastBeforeSum: true

                RoundMode: 'convergent'
             OverflowMode: 'wrap'
```

This second example shows the default property values for `dfilt.latticemamax` filter objects, using the coefficients from an `fir1` filter.

```
b=fir1(7,0.45)


hdlat=dfilt.latticemamax(b)

hdlat =

         FilterStructure: [1x45 char]
              Arithmetic: 'double'
                 Lattice: [1x8 double]
        PersistentMemory: false
                  States: [8x1 double]
     NumSamplesProcessed: 0

hdlat.arithmetic='fixed'

hdlat =

         FilterStructure: [1x45 char]
              Arithmetic: 'fixed'
```

```
                Lattice: [1x8 double]
       PersistentMemory: false
                 States: [1x1 embedded.fi]
    NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

         StateWordLength: 16
         StateFracLength: 15

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'
```

Unlike the `single` or `double` options for `Arithmetic`, `fixed` uses properties to define the word and fraction lengths for each portion of your filter. By changing the property value of any of the properties, you control your filter performance. Every word length and fraction length property is independent—set the one you need and the others remain unchanged, such as setting the input word length with `InputWordLength`, while leaving the fraction length the same.

```
d=fdesign.lowpass('n,fc',6,0.45)

d =

           ResponseType: 'Lowpass with cutoff'
      SpecificationType: 'N,Fc'
            Description: {2x1 cell}
```

```
    NormalizedFrequency: true
                     Fs: 'Normalized'
            FilterOrder: 6
                Fcutoff: 0.4500


designmethods(d)


Design Methods for class fdesign.lowpass:


butter

hd=butter(d)

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [3x6 double]
             ScaleValues: [4x1 double]
        PersistentMemory: false
                  States: [2x3 double]
     NumSamplesProcessed: 0

hd.arithmetic='fixed'

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'fixed'
               sosMatrix: [3x6 double]
             ScaleValues: [4x1 double]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

          CoeffWordLength: 16
           CoeffAutoScale: true
```

```
                    Signed: true

           InputWordLength: 16
           InputFracLength: 15

    SectionInputWordLength: 16
     SectionInputAutoScale: true

   SectionOutputWordLength: 16
   Section OutputAutoScale: true

          OutputWordLength: 16
                OutputMode: 'AvoidOverflow'

           StateWordLength: 16
           StateFracLength: 15

               ProductMode: 'FullPrecision'

                 AccumMode: 'KeepMSB'
           AccumWordLength: 40
             CastBeforeSum: true

                 RoundMode: 'convergent'
              OverflowMode: 'wrap'

hd.inputWordLength=12

hd =

          FilterStructure: 'Direct-Form II, Second-Order Sections'
                Arithmetic: 'fixed'
                 sosMatrix: [3x6 double]
               ScaleValues: [4x1 double]
          PersistentMemory: false
                    States: [1x1 embedded.fi]
       NumSamplesProcessed: 0

           CoeffWordLength: 16
            CoeffAutoScale: true
```

```
                  Signed: true

         InputWordLength: 12
         InputFracLength: 15

  SectionInputWordLength: 16
   SectionInputAutoScale: true

 SectionOutputWordLength: 16
  SectionOutputAutoScale: true

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

         StateWordLength: 16
         StateFracLength: 15

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'
```

Notice that the properties for the lattice filter `hdlat` and direct-form II filter `hd` are different, as befits their differing filter structures. Also, some properties are common to both objects, such as `RoundMode` and `PersistentMemory` and behave the same way in both objects.

Notes About Fraction Length, Word Length, and Precision

Word length and fraction length combine to make the format for a fixed-point number, where word length is the number of bits used to represent the value and fraction length specifies, in bits, the location of the binary point in the fixed-point representation. Therein lies a problem—fraction length, which you specify in bits, can be larger than the word length, or a negative number of bits. This section explains how that idea works and how you might use it.

Unfortunately fraction length is somewhat misnamed (although it continues to be used in this User's Guide and elsewhere for historical reasons).

Fraction length defined as the number of fractional bits (bits to the right of the binary point) is true only when the fraction length is positive and less than or equal to the word length. In MATLAB format notation we use [word length fraction length]. For example, for the format [16 16], the second 16 (the fraction length) is the number of fractional bits or bits to the right of the binary point. In this example, all 16 bits are to the right of the binary point.

But it is also possible to have fixed-point formats of [16 18] or [16 -45]. In these cases the fraction length can no longer be the number of bits to the right of the binary point since the format says the word length is 16—there cannot be 18 fraction length bits on the right. And how can there be a negative number of bits for the fraction length, such as [16 -45]?

A better way to think about fixed-point format [word length fraction length] and what it means is that the representation of a fixed-point number is a weighted sum of powers of two driven by the fraction length, or the two's complement representation of the fixed-point number.

Consider the format [B L], where the fraction length L can be positive, negative, 0, greater than B (the word length) or less than B. (B and L are always integers and B is always positive.)

Given a binary string b(1) b(2) b(3) ... b(B), to determine the two's-complement value of the string in the format described by [B L], use the value of the individual bits in the binary string in the following formula, where b(1) is the first binary bit (and most significant bit, MSB), b(2) is the second, and on up to b(B).

The decimal numeric value that those bits represent is given by

```
value =-b(1)*2^(B-L-1)+b(2)*2^(B-L-2)+b(3)*2^(B-L-3)+...+ b(B)*2^(-L)
```

L, the fraction length, represents the negative of the weight of the last, or least significant bit (LSB). L is also the step size or the precision provided by a given fraction length.

Precision

Here is how precision works.

When all of the bits of a binary string are zero except for the LSB (which is therefore equal to one), the value represented by the bit string is given by $2^{(-L)}$.

If L is negative, for example L=-16, the value is 2^{16}. The smallest step between numbers that can be represented in a format where L=-16 is given by 1 x 2^{16} (the rightmost term in the formula above), which is 65536. Note the precision does not depend on the word length.

Take a look at another example. When the word length set to 8 bits, the decimal value 12 is represented in binary by 00001100. That 12 is the decimal equivalent of 00001100 tells us we are using [8 0] data format representation—the word length is 8 bits and fraction length 0 bits, and the step size or precision (the smallest difference between two adjacent values in the format [8,0], is 2^0=1.

Suppose you plan to keep only the upper 5 bits and discard the other three. The resulting precision after removing the right-most three bits comes from the weight of the lowest remaining bit, the fifth bit from the left, which is 2^3=8, so the format would be [5,-3].

Note that in this format the step size is 8, I cannot represent numbers that are between multiples of 8.

In MATLAB, with the Fixed-Point Toolbox installed:

```
x=8;
q=quantizer([8,0]); % Word length = 8, fraction length = 0
xq=quantize(q,x);
binxq=num2bin(q,xq);
q1=quantizer([5 -3]); % Word length = 5, fraction length = -3
xq1 = quantize(q1,xq);
binxq1=num2bin(q1,xq1);
binxq

binxq =

00001000

binxq1

binxq1 =

00001
```

But notice that in [5,-3] format, 00001 is the two's complement representation for 8, not for 1; `q = quantizer([8 0])` and `q1 = quantizer([5 -3])` are not the same. They cover the about the same range—`range(q)>range(q1)`—but their quantization step is different—`eps(q)`= 8, and `eps(q1)=1`.

Look at one more example. When you construct a quantizer `q`

```
q = quantizer([a,b])
```

the first element in `[a,b]` is `a`, the word length used for quantization. The second element in the expression, `b`, is related to the quantization step—the numerical difference between the two closest values that the quantizer can represent. This is also related to the weight given to the LSB. Note that `2^(-b) = eps(q)`.

Now construct two quantizers, `q1` and `q2`. Let `q1` use the format [32,0] and let `q2` use the format [16, -16].

```
q1 = quantizer([32,0])
q2 = quantizer([16,-16])
```

Quantizers `q1` and `q2` cover the same range, but `q2` has less precision. It covers the range in steps of 2^{16}, while `q` covers the range in steps of 1.

This lost precision is due to (or can be used to model) throwing out 16 least-significant bits.

An important point to understand is that in `dfilt` objects and filtering you control which bits are carried from the sum and product operations in the filter to the filter output by setting the format for the output from the sum or product operation.

For instance, if you use [16 0] as the output format for a 32-bit result from a sum operation when the original format is [32 0], you take the lower 16 bits from the result. If you use [16 -16], you take the higher 16 bits of the original 32 bits. You could even take 16 bits somewhere in between the 32 bits by choosing something like [16 -8], but you probably do not want to do that.

Filter scaling is directly implicated in the format and precision for a filter. When you know the filter input and output formats, as well as the filter internal formats, you can scale the inputs or outputs to stay within the format ranges. For more information about scaling filters, refer to "Working with Fixed-Point Direct-Form FIR Filters" on page 3-33.

Notice that overflows or saturation might occur at the filter input, filter output, or within the filter itself, such as during add or multiply or accumulate operations. Improper scaling at any point in the filter can result in numerical errors that dramatically change the performance of your fixed-point filter implementation.

CastBeforeSum

Setting the `CastBeforeSum` property determines how the filter handles the input values to sum operations in the filter. After you set your filter `Arithmetic` property value to `fixed`, you have the option of using `CastBeforeSum` to control the data type of some inputs (addends) to summations in your filter. To determine which addends reflect the `CastBeforeSum` property setting, refer to the reference page for the signal flow diagram for the filter structure.

`CastBeforeSum` specifies whether to cast selected addends to summations in the filter to the output format from the addition operation before performing the addition. When you specify `true` for the property value, the results of the affected sum operations match most closely the results found on most digital signal processors. Performing the cast operation before the summation adds one or two additional quantization operations that can add error sources to your filter results.

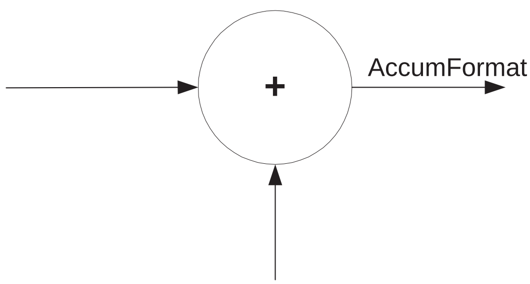
Specifying `CastBeforeSum` to be `false` prevents the addends from being cast to the output format before the addition operation. Choose this setting to get the most accurate results from summations without considering the hardware your filter might use.

Notice that the output format for every sum operation reflects the value of the output property specified in the filter structure diagram. Which input property is referenced by CastBeforeSum depends on the structure.

Property Value	Description
false	Configures filter summation operations to retain the addends in the format carried from the previous operation.
true	Configures filter summation operations to convert the input format of the addends to match the summation output format before performing the summation operation. Usually this generates results from the summation that more closely match those found from digital signal processors

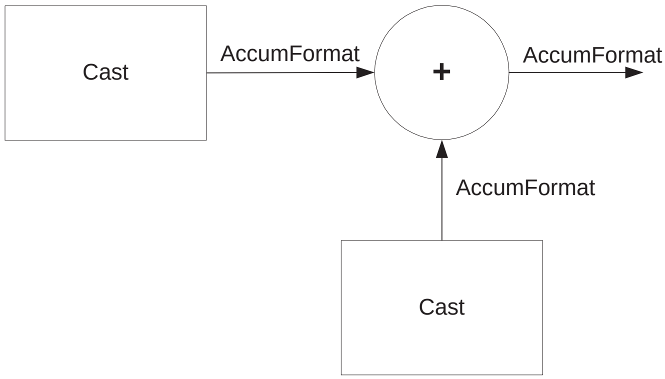
Diagrams of CastBeforeSum Settings

When CastBeforeSum is false, sum elements in filter signal flow diagrams look like this:



showing that the input data to the sum operations (the addends) retain their format word length and fraction length from previous operations. The addition process uses the existing input formats and then casts the output to the format defined by AccumFormat. Thus the output data has the word length and fraction length defined by AccumWordLength and AccumFracLength or AccumMode and AccumWordLength.

When CastBeforeSum is true, sum elements in filter signal flow diagrams look like this:



showing that the input data gets recast to the accumulator format word length and fraction length (AccumFormat) before the sum operation occurs. The data output by the addition operation has the word length and fraction length defined by AccumWordLength and AccumFracLength and AccumMode.

CoeffAutoScale

How the filter represents the filter coefficients depends on the property value of CoeffAutoScale. When you create a dfilt object, you use coefficients in double-precision format. Converting the dfilt object to fixed-point arithmetic forces the coefficients into a fixed-point representation. The representation the filter uses depends on whether the value of CoeffAutoScale is true or false.

- CoeffAutoScale = true means the filter chooses the fraction length to maintain the value of the coefficients as close to the double-precision values as possible. When you change the word length applied to the coefficients, the filter object changes the fraction length to try to accommodate the change. true is the default setting.
- CoeffAutoScale = false removes the automatic scaling of the fraction length for the coefficients and exposes the property that controls the coefficient fraction length so you can change it. For example, if the filter is a direct form FIR filter, setting CoeffAutoScale = false exposes the NumFracLength property that specifies the fraction length applied to numerator coefficients. If the filter is an IIR filter, setting

CoeffAutoScale = false exposes both the NumFracLength and DenFracLength properties.

Here is an example of using CoeffAutoScale with a direct form filter.

```
hd2=dfilt.dffir([0.3 0.6 0.3])

hd2 =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'double'
               Numerator: [0.3000 0.6000 0.3000]
        PersistentMemory: false
                  States: [2x1 double]
     NumSamplesProcessed: 0

hd2.arithmetic='fixed'

hd2 =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'fixed'
               Numerator: [0.3000 0.6000 0.3000]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
```

```
          CastBeforeSum: true

              RoundMode: 'convergent'
           OverflowMode: 'wrap'
```

To this point, the filter coefficients retain the original values from when you created the filter as shown in the `Numerator` property. Now change the `CoeffAutoScale` property value from `true` to `false`.

```
hd2.coeffautoScale=false

hd2 =

        FilterStructure: 'Direct-Form FIR'
             Arithmetic: 'fixed'
              Numerator: [0.3000 0.6000 0.3000]
       PersistentMemory: false
                 States: [1x1 embedded.fi]
    NumSamplesProcessed: 0

        CoeffWordLength: 16
         CoeffAutoScale: false
          NumFracLength: 15
                 Signed: true

        InputWordLength: 16
        InputFracLength: 15

       OutputWordLength: 16
             OutputMode: 'AvoidOverflow'

            ProductMode: 'FullPrecision'

              AccumMode: 'KeepMSB'
        AccumWordLength: 40
          CastBeforeSum: true

              RoundMode: 'convergent'
           OverflowMode: 'wrap'
```

With the `NumFracLength` property now available, change the word length to 5 bits.

Notice the coefficient values. Setting `CoeffAutoScale` to `false` removes the automatic fraction length adjustment and the filter coefficients cannot be represented by the current format of [5 15]—a word length of 5 bits, fraction length of 15 bits.

```
hd2.coeffwordlength=5

hd2 =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'fixed'
               Numerator: [4.5776e-004 4.5776e-004 4.5776e-004]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 5
          CoeffAutoScale: false
           NumFracLength: 15
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'
```

Restoring `CoeffAutoScale` to `true` goes some way to fixing the coefficient values. Automatically scaling the coefficient fraction length results in setting the fraction length to 4 bits. You can check this with `get(hd2)` as shown below.

```
hd2.coeffautoScale=true

hd2 =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'fixed'
               Numerator: [0.3125 0.6250 0.3125]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 5
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'

get(hd2)
        PersistentMemory: false
     NumSamplesProcessed: 0
         FilterStructure: 'Direct-Form FIR'
                  States: [1x1 embedded.fi]
               Numerator: [0.3125 0.6250 0.3125]
```

```
          Arithmetic: 'fixed'
     CoeffWordLength: 5
      CoeffAutoScale: 1
              Signed: 1
           RoundMode: 'convergent'
        OverflowMode: 'wrap'
     InputWordLength: 16
     InputFracLength: 15
    OutputWordLength: 16
          OutputMode: 'AvoidOverflow'
         ProductMode: 'FullPrecision'
           AccumMode: 'KeepMSB'
       NumFracLength: 4
    OutputFracLength: 12
   ProductWordLength: 21
   ProductFracLength: 19
     AccumWordLength: 40
     AccumFracLength: 19
       CastBeforeSum: 1
```

Clearly five bits is not enough to represent the coefficients accurately.

CoeffFracLength

Fixed-point scalar filters that you create using `dfilt.scalar` use this property to define the fraction length applied to the scalar filter coefficients. Like the coefficient-fraction-length-related properties for the FIR, lattice, and IIR filters, `CoeffFracLength` is not displayed for scalar filters until you set `CoeffAutoScale` to `false`. Once you change the automatic scaling you can set the fraction length for the coefficients to any value you require.

As with all fraction length properties, the value you enter here can be any negative or positive integer, or zero. Fraction length can be larger than the associated word length, as well. By default, the value is 14 bits, with the `CoeffWordlength` of 16 bits.

CoeffWordLength

One primary consideration in developing filters for hardware is the length of a data word. `CoeffWordLength` defines the word length for these data storage and arithmetic locations:

- Numerator and denominator filter coefficients
- Tap sum in dfilt.dfsymfir and dfilt.dfasymfir filter objects
- Section input, multiplicand, and state values in direct-form SOS filter objects such as dfilt.df1t and dfilt.df2
- Scale values in second-order filters
- Lattice and ladder coefficients in lattice filter objects, such as dfilt.latticearma and dfilt.latticemamax
- Gain in dfilt.scalar

Setting this property value controls the word length for the data listed. In most cases, the data words in this list have separate fraction length properties to define the associated fraction lengths.

Any positive, integer word length works here, limited by the machine you use to develop your filter and the hardware you use to deploy your filter.

DenAccumFracLength

Filter structures df1, df1t, df2, and df2t that use fixed arithmetic have this property that defines the fraction length applied to denominator coefficients in the accumulator. In combination with AccumWordLength, the properties fully specify how the accumulator outputs data stored there.

As with all fraction length properties, DenAccumFracLength can be any integer, including integers larger than AccumWordLength, and positive or negative integers. To be able to change the property value for this property, you set AccumMode to SpecifyPrecision.

DenFracLength

Property DenFracLength contains the value that specifies the fraction length for the denominator coefficients for your filter. DenFracLength specifies the fraction length used to interpret the data stored in C. Used in combination with CoeffWordLength, these two properties define the interpretation of the coefficients stored in the vector that contains the denominator coefficients.

As with all fraction length properties, the value you enter here can be any negative or positive integer, or zero. Fraction length can be larger than the associated word length, as well. By default, the value is 15 bits, with the CoeffWordLength of 16 bits.

Denominator

The denominator coefficients for your IIR filter, taken from the prototype you start with, are stored in this property. Generally this is a 1-by-N array of data in double format, where N is the length of the filter.

All IIR filter objects include `Denominator`, except the lattice-based filters which store their coefficients in the `Lattice` property, and second-order section filters, such as `dfilt.df1tsos`, which use the `SosMatrix` property to hold the coefficients for the sections.

DenProdFracLength

A property of all of the direct form IIR `dfilt` objects, except the ones that implement second-order sections, `DenProdFracLength` specifies the fraction length applied to data output from product operations that the filter performs on denominator coefficients.

Looking at the signal flow diagram for the `dfilt.df1t` filter, for example, you see that denominators and numerators are handled separately. When you set `ProductMode` to `SpecifyPrecision`, you can change the `DenProdFracLength` setting manually. Otherwise, for multiplication operations that use the denominator coefficients, the filter sets the fraction length as defined by the `ProductMode` setting.

DenStateFracLength

When you look at the flow diagram for the `dfilt.df1sos` filter object, the states associated with denominator coefficient operations take the fraction length from this property. In combination with the `DenStateWordLength` property, these properties fully specify how the filter interprets the states.

As with all fraction length properties, the value you enter here can be any negative or positive integer, or zero. Fraction length can be larger than the associated word length, as well. By default, the value is 15 bits, with the `DenStateWordLength` of 16 bits.

DenStateWordLength

When you look at the flow diagram for the `dfilt.df1sos` filter object, the states associated with the denominator coefficient operations take the data format from this property and the DenStateFracLength property. In combination, these properties fully specify how the filter interprets the state it uses.

By default, the value is 16 bits, with the DenStateFracLength of 15 bits.

FilterInternals

Similar to the FilterInternals pane in FDATool, this property controls whether the filter sets the output word and fraction lengths automatically, and the accumulator word and fraction lengths automatically as well, to maintain the best precision results during filtering. The default value, FullPrecision, sets automatic word and fraction length determination by the filter. Setting FilterInternals to SpecifyPrecision exposes the output and accumulator related properties so you can set your own word and fraction lengths for them. Note that

FilterStructure

Every dfilt object has a FilterStructure property. This is a read-only property containing a string that declares the structure of the filter object you created.

When you construct filter objects, the FilterStructure property value is returned containing one of the strings shown in the following table. Property FilterStructure indicates the filter architecture and comes from the constructor you use to create the filter.

After you create a filter object, you cannot change the FilterStructure property value. To make filters that use different structures, you construct new filters using the appropriate methods, or use convert to switch to a new structure.

Default value: Since this depends on the constructor you use and the constructor includes the filter structure definition, there is no default value.

When you try to create a filter without specifying a structure, MATLAB returns an error.

Filter Constructor Name	FilterStructure Property String and Filter Type
'dfilt.df1'	Direct form I
'dfilt.df1sos'	Direct form I filter implemented using second-order sections
'dfilt.df1t'	Direct form I transposed
'dfilt.df2'	Direct form II
'dfilt.df2sos'	Direct form II filter implemented using second order sections
'dfilt.df2t'	Direct form II transposed
'dfilt.dfasymfir'	Antisymmetric finite impulse response (FIR). Even and odd forms.
'dfilt.dffir'	Direct form FIR
'dfilt.dffirt'	Direct form FIR transposed
'dfilt.latticeallpass'	Lattice allpass
'dfilt.latticear'	Lattice autoregressive (AR)
'dfilt.latticemamin'	Lattice moving average (MA) minimum phase
'dfilt.latticemamax'	Lattice moving average (MA) maximum phase
'dfilt.latticearma'	Lattice ARMA
'dfilt.dfsymfir'	Symmetric FIR. Even and odd forms
'dfilt.scalar'	Scalar

Filter Structures with Quantizations Shown in Place

To help you understand how and where the quantizations occur in filter structures in this toolbox, Figure 8-1 presents the structure for a Direct Form 2 filter, including the quantizations (fixed-point formats) that compose part of the fixed-point filter. You see that one or more quantization processes, specified by the *format label, accompany each filter element, such as a delay, product, or summation element. The input to or output from each element reflects the result of applying the associated quantization as defined by the word length and fraction length format. Wherever a particular filter element appears in a filter structure, recall the quantization process that accompanies the element as it appears in this figure. Each filter reference page, such as the `dfilt.df2` reference page, includes the signal flow diagram showing the formatting elements that define the quantizations that occur throughout the filter flow.

For example, a product quantization, either numerator or denominator, follows every product (gain) element and a sum quantization, also either numerator or denominator, follows each sum element. In this figure, we set the `Arithmetic` property value to `fixed`.

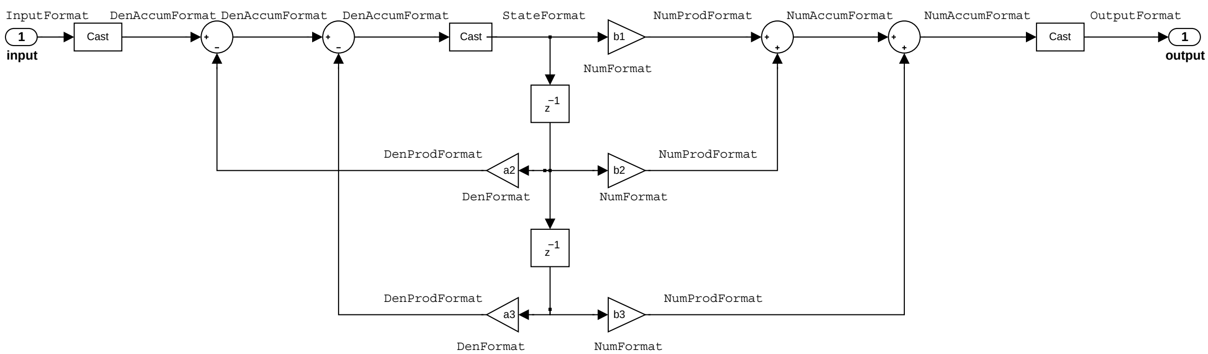


Figure 8-1: df2 IIR Filter Structure Including The Formatting Objects, With Arithmetic Property Value fixed

When your `df2` filter uses the `Arithmetic` property set to `fixed`, the filter structure contains the formatting features shown in the diagram. The formats included in the structure are fixed-point objects that include properties to set various word and fraction length formats. For example, the `NumFormat` or `DenFormat` in the fixed-point arithmetic filter set the properties for quantizing numerator or denominator coefficients according to word and fraction length settings.

When the leading denominator coefficient a(1) in your filter is not 1, choose it to be a power of two so that a shift replaces the multiply that would otherwise be used.

Fixed-Point Arithmetic Filter Structures

You choose among several filter structures when you create fixed-point filters. You can also specify filters with single or multiple cascaded sections of the same type. Because quantization is a nonlinear process, different fixed-point filter structures produce different results.

To specify the filter structure, you select the appropriate `dfilt.`*`structure`* method to construct your filter. Refer to the function reference information for `dfilt` and `set` for details on setting property values for quantized filters.

The figures in the following subsections of this section serve as aids to help you determine how to enter your filter coefficients for each filter structure. Each subsection contains an example for constructing a filter of the given structure.

Scale factors for the input and output for the filters do not appear in the block diagrams. The default filter structures do not include, nor assume, the scale factors. For filter scaling information, refer to `scale` in the Help system.

About the Filter Structure Diagrams

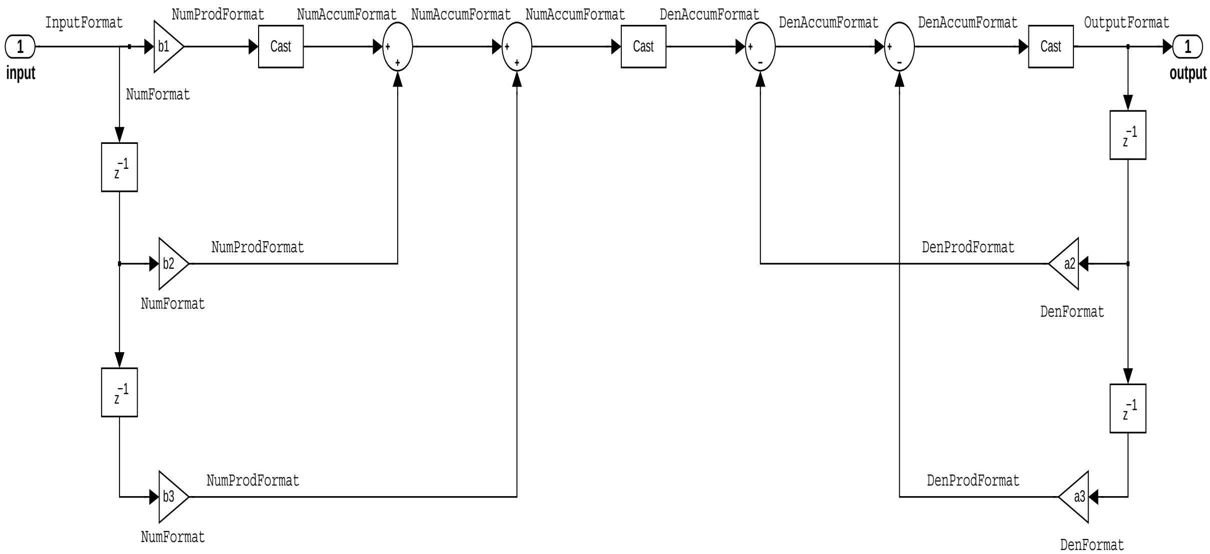
In the diagrams that accompany the following filter structure descriptions, you see the active operators that define the filter, such as sums and gains, and the formatting features that control the processing in the filter. Notice also that the coefficients are labeled in the figure. This tells you the order in which the filter processes the coefficients.

While the meaning of the block elements is straightforward, the labels for the formats that form part of the filter are less clear. Each figure includes text in the form *`labelFormat`* that represents the existence of a formatting feature at that point in the structure. The *`Format`* stands for formatting object and the *`label`* specifies the data that the formatting object affects.

For example, in the `dfilt.df2` filter shown on page 8-48, the entries `InputFormat` and `OutputFormat` are the formats applied, that is the word length and fraction length, to the filter input and output data. For example, filter properties like `OutputWordLength` and `InputWordLength` specify values that control filter operations at the input and output points in the structure and are represented by the formatting objects `InputFormat` and `OutputFormat` shown in the filter structure diagrams.

Direct Form I Filter Structure

The following figure depicts the *direct form I* filter structure that directly realizes a transfer function with a second-order numerator and denominator. The numerator coefficients are numbered $b(i)$, i=1, 2, 3; the denominator coefficients are numbered $a(i)$, i = 1, 2, 3; and the states (used for initial and final state values in filtering) are labeled $z(i)$. In the figure, the `Arithmetic` property is set to `fixed`.



Example—Specifying a Direct Form I Filter. You can specify a second-order direct form I structure for a quantized filter `hq` with the following code.

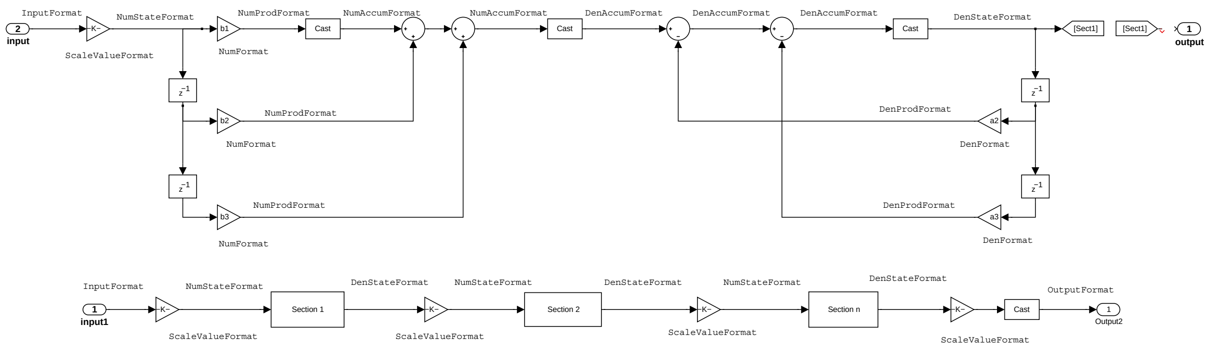
```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hq = dfilt.df1(b,a);
```

To create the fixed-point filter, set the `Arithmetic` property to `fixed` as shown here.

```
set(hq,'arithmetic','fixed');
```

Direct Form I Filter Structure With Second-Order Sections

The following figure depicts a *direct form I* filter structure that directly realizes a transfer function with a second-order numerator and denominator and second-order sections. The numerator coefficients are numbered $b(i)$, i=1, 2, 3; the denominator coefficients are numbered $a(i)$, i = 1, 2, 3; and the states (used for initial and final state values in filtering) are labeled $z(i)$. In the figure, the `Arithmetic` property is set to `fixed` to place the filter in fixed-point mode.



Example—Specifying a Direct Form I Filter with Second-Order Sections. You can specify an eighth-order direct form I structure for a quantized filter `hq` with the following code.

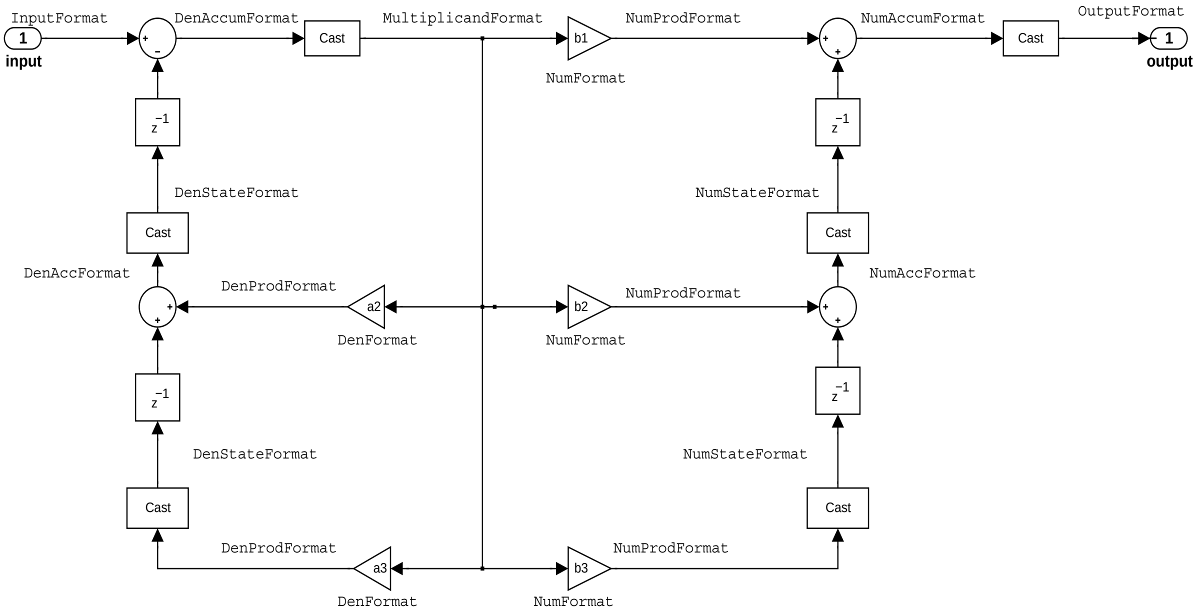
```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hq = dfilt.df1sos(b,a);
```

To create the fixed-point filter, set the `Arithmetic` property to `fixed`, as shown here.

```
set(hq,'arithmetic','fixed');
```

Direct Form I Transposed Filter Structure

The next signal flow diagram depicts a *direct form I transposed* filter structure that directly realizes a transfer function with a second-order numerator and denominator. The numerator coefficients are $b(i)$, i = 1, 2, 3; the denominator coefficients are $a(i)$, i = 1, 2, 3; and the states (used for initial and final state values in filtering) are labeled $z(i)$. With the `Arithmetic` property value set to `fixed`, the figure shows the filter with the properties indicated.

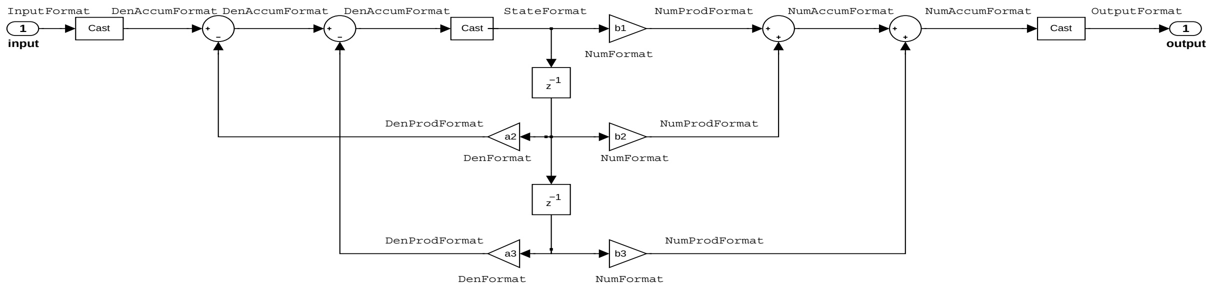


Example—Specifying a Direct Form I Transposed Filter. You can specify a second-order direct form I transposed filter structure for a quantized filter `hq` with the following code.

```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hq = dfilt.df1t(b,a);
set(hq,'arithmetic','fixed');
```

Direct Form II Filter Structure

The following graphic depicts a *direct form II* filter structure that directly realizes a transfer function with a second-order numerator and denominator. In the figure, the `Arithmetic` property value is `fixed`. Numerator coefficients are named $b(i)$; denominator coefficients are named $a(i)$, $i = 1, 2, 3$; and the states (used for initial and final state values in filtering) are named $z(i)$.



Use the method `dfilt.df2` to construct a quantized filter whose `FilterStructure` property is `Direct-Form II`.

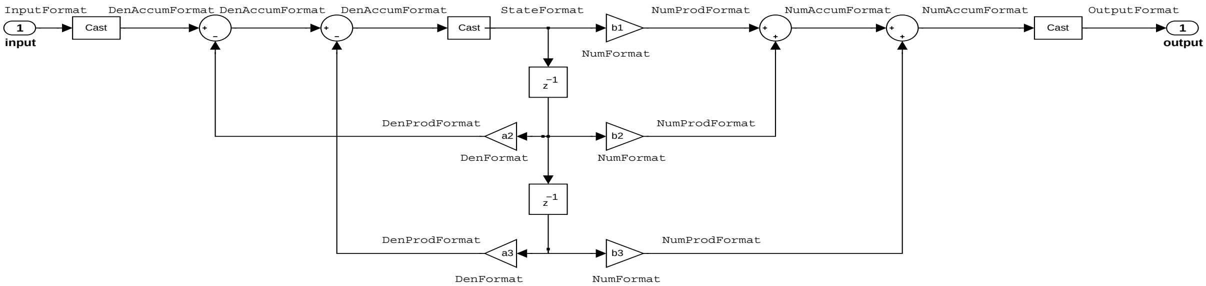
Example—Specifying a Direct Form II Filter. You can specify a second-order direct form II filter structure for a quantized filter `hq` with the following code.

```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hq = dfilt.df2({b,a});
hq.arithmetic = 'fixed'
```

To convert your initial double-precision filter `hq` to a quantized or fixed-point filter, set the `Arithmetic` property to `fixed`, as shown.

Direct Form II Filter Structure With Second-Order Sections

The following figure depicts *direct form II* filter structure using second-order sections that directly realizes a transfer function with a second-order numerator and denominator sections. In the figure, the `Arithmetic` property value is `fixed`. Numerator coefficients are labeled $b(i)$; denominator coefficients are labeled $a(i)$, $i = 1, 2, 3$; and the states (used for initial and final state values in filtering) are labeled $z(i)$.



Use the method `dfilt.df2sos` to construct a quantized filter whose `FilterStructure` property is `Direct-Form II`.

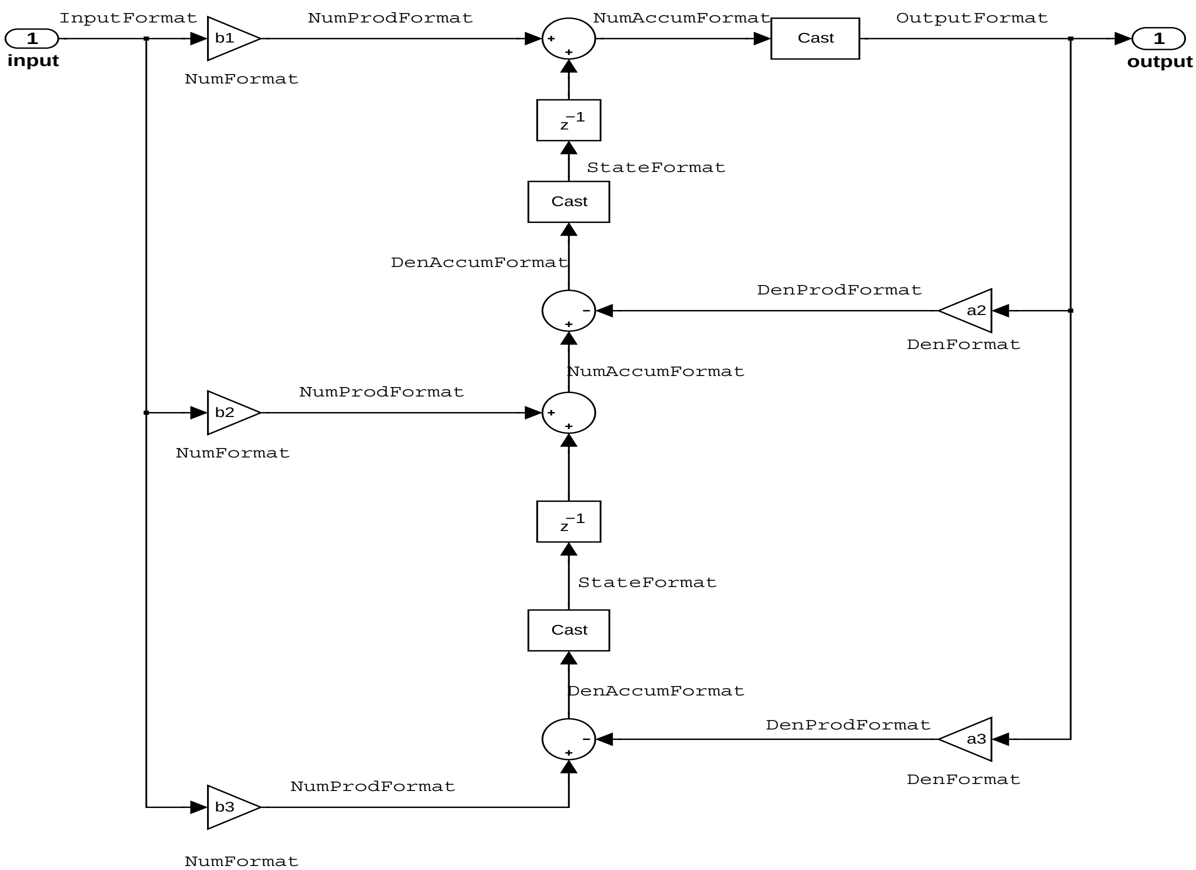
Example—Specifying a Direct Form II Filter with Second-Order Sections. You can specify a tenth-order direct form II filter structure that uses second-order sections for a quantized filter `hq` with the following code.

```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hq = dfilt.df2sos({b,a});
hq.arithmetic = 'fixed'
```

To convert your prototype double-precision filter `hq` to a fixed-point filter, set the `Arithmetic` property to `fixed`, as shown.

Direct Form II Transposed Filter Structure

The following figure depicts the *direct form II transposed* filter structure that directly realizes transfer functions with a second-order numerator and denominator. The numerator coefficients are labeled $b(i)$, the denominator coefficients are labeled $a(i)$, $i = 1, 2, 3$, and the states (used for initial and final state values in filtering) are labeled $z(i)$. In the first figure, the `Arithmetic` property value is `fixed`.



Use the constructor `dfilt.df2t` to specify the value of the `FilterStructure` property for a filter with this structure that you can convert to fixed-point filtering.

Example—Specifying a Direct Form II Transposed Filter. Specifying or constructing a second-order direct form II transposed filter for a fixed-point filter `hq` starts with the following code to define the coefficients and construct the filter.

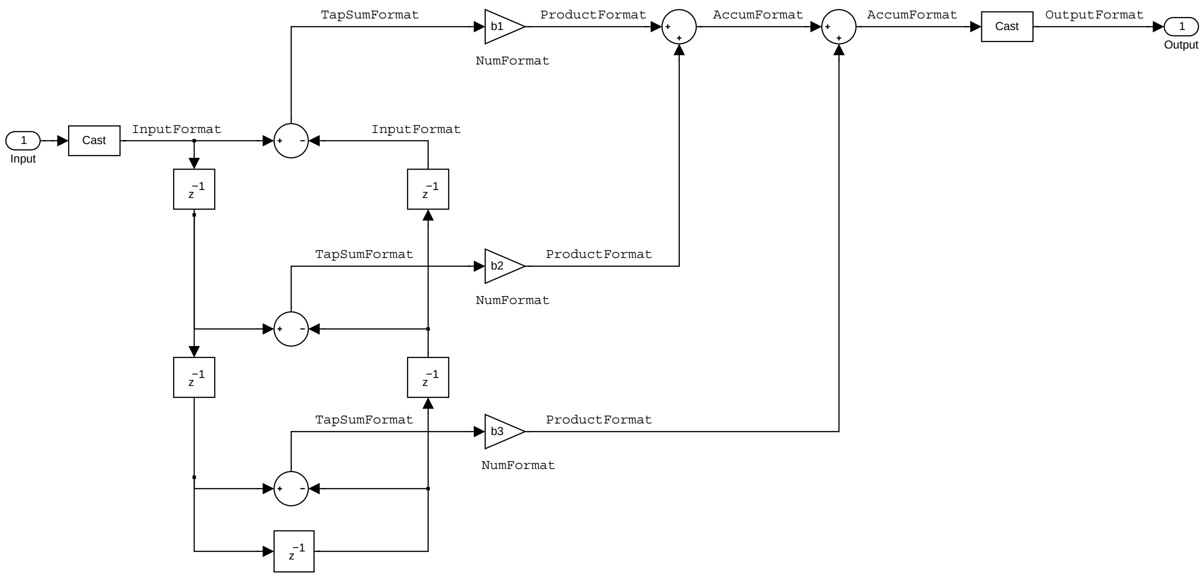
```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hd = dfilt.df2t({b,a});
```

Now create the fixed-point filtering version of the filter from `hd`, which is floating point.

```
hq = set(hd,'arithmetic','fixed');
```

Direct Form Antisymmetric FIR Filter Structure (Any Order)

The following figure depicts a *direct form antisymmetric FIR* filter structure that directly realizes a second-order antisymmetric FIR filter. The filter coefficients are labeled *b(i)*, and the initial and final state values in filtering are labeled $z(i)$. This structure reflects the `Arithmetic` property set to `fixed`.



Use the method `dfilt.dfasymfir` to construct the filter, and then set the `Arithmetic` property to `fixed` to convert to a fixed-point filter with this structure.

Example—Specifying an Odd-Order Direct Form Antisymmetric FIR Filter. Specify a fifth-order direct form antisymmetric FIR filter structure for a fixed-point filter `hq` with the following code.

```
b = [-0.008 0.06 -0.44 0.44 -0.06 0.008];
hq = dfilt.dfasymfir(b);
set(hq,'arithmetic','fixed')
```

```
hq

hq =

          FilterStructure: 'Direct-Form Antisymmetric FIR'
               Arithmetic: 'fixed'
                Numerator: [-0.0080 0.0600 -0.4400 0.4400 -0.0600 0.0080]
         PersistentMemory: false
                   States: [1x1 fi object]
      NumSamplesProcessed: 0

           CoeffWordLength: 16
            CoeffAutoScale: true
                    Signed: true

           InputWordLength: 16
           InputFracLength: 15

          OutputWordLength: 16
                OutputMode: 'AvoidOverflow'

                TapSumMode: 'KeepMSB'
          TapSumWordLength: 17

               ProductMode: 'FullPrecision'

                 AccumMode: 'KeepMSB'
           AccumWordLength: 40

             CastBeforeSum: true
                 RoundMode: 'convergent'
              OverflowMode: 'wrap'

           InheritSettings: false
```

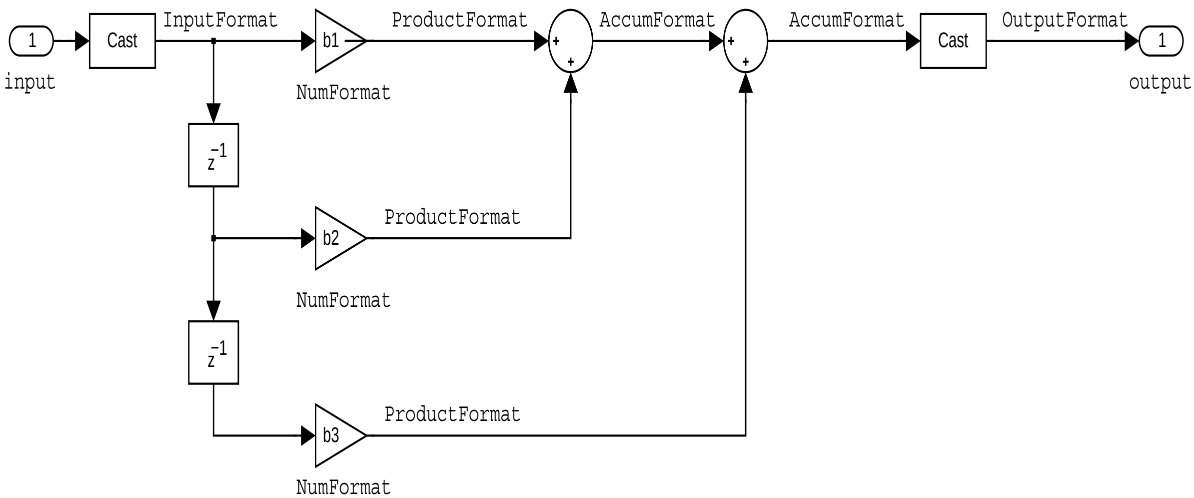
Example—Specifying an Even-Order Direct Form Antisymmetric FIR Filter. You can specify a fourth-order direct form antisymmetric FIR filter structure for a fixed-point filter hq with the following code.

```
b = [-0.01 0.1 0.0 -0.1 0.01];
```

```
hq = dfilt.dfasymfir(b);
hq.arithmetic='fixed'

hq =

         FilterStructure: 'Direct-Form Antisymmetric FIR'
              Arithmetic: 'fixed'
               Numerator: [-0.0100 0.1000 0 -0.1000 0.0100]
        PersistentMemory: false
                  States: [1x1 fi object]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

              TapSumMode: 'KeepMSB'
        TapSumWordLength: 17

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40

           CastBeforeSum: true
               RoundMode: 'convergent'
            OverflowMode: 'wrap'

         InheritSettings: false
```

Direct Form Finite Impulse Response (FIR) Filter Structure

In the next figure, you see the signal flow graph for a *direct form finite impulse response (FIR)* filter structure that directly realizes a second-order FIR filter. The filter coefficients are $b(i)$, $i = 1, 2, 3$, and the states (used for initial and final state values in filtering) are $z(i)$. To generate the figure, set the `Arithmetic` property to `fixed` after you create your prototype filter in double-precision arithmetic.



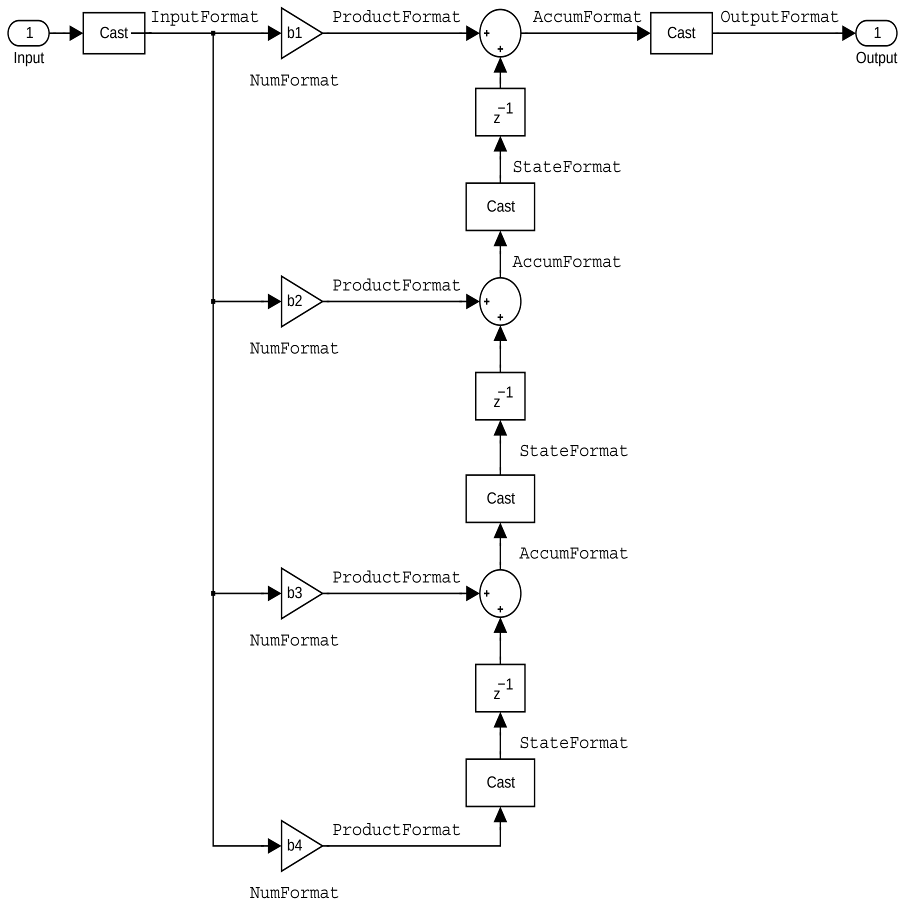
Use the `dfilt.dffir` method to generate a filter that uses this structure.

Example—Specifying a Direct Form FIR Filter. You can specify a second-order direct form FIR filter structure for a fixed-point filter `hq` with the following code.

```
b = [0.05 0.9 0.05];
hd = dfilt.dffir({b});
hq = set(hd,'arithmetic','fixed');
```

Direct Form FIR Transposed Filter Structure

This figure uses the filter coefficients labeled $b(i)$, i = 1, 2, 3, and states (used for initial and final state values in filtering) are labeled $z(i)$. These depict a *direct form finite impulse response (FIR) transposed* filter structure that directly realizes a second-order FIR filter.



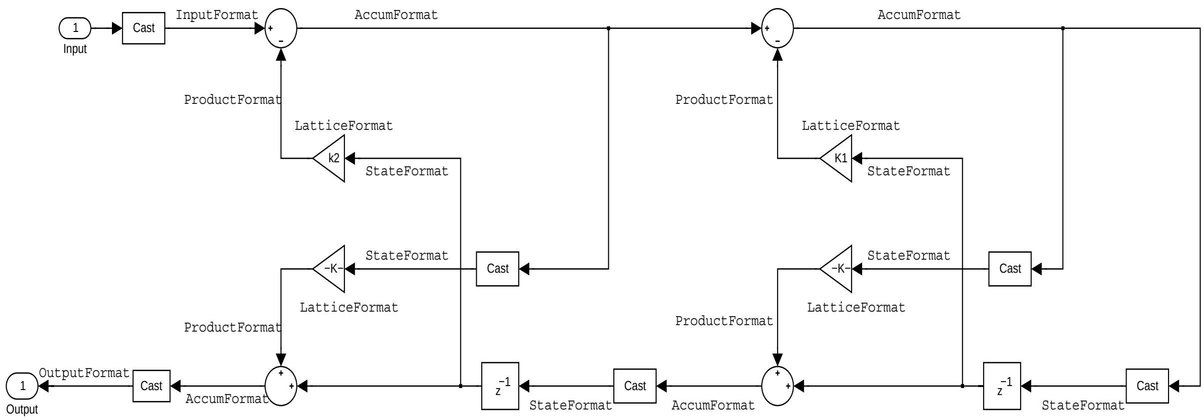
With the `Arithmetic` property set to `fixed`, your filter matches the figure. Using the method `dfilt.dffirt` returns a double-precision filter that you convert to a fixed-point filter.

Example—Specifying a Direct Form FIR Transposed Filter. You can specify a second-order direct form FIR transposed filter structure for a fixed-point filter hq with the following code.

```
b = [0.05 0.9 0.05];
hd=dfilt.dffirt({b});
hq = copy(hd);
hq.arithmetic = 'fixed';
```

Lattice Allpass Filter Structure

The following figure depicts the *lattice allpass* filter structure. The pictured structure directly realizes third-order lattice allpass filters using fixed-point arithmetic. The filter reflection coefficients are labeled $k1(i)$, $i = 1, 2, 3$. The states (used for initial and final state values in filtering) are labeled $z(i)$.



To create a quantized filter that uses the lattice allpass structure shown in the figure, use the `dfilt.latticeallpass` method and set the `Arithmetic` property to `fixed`.

Example—Specifying a Lattice Allpass Filter. You can create a third-order lattice allpass filter structure for a quantized filter `hq` with the following code.

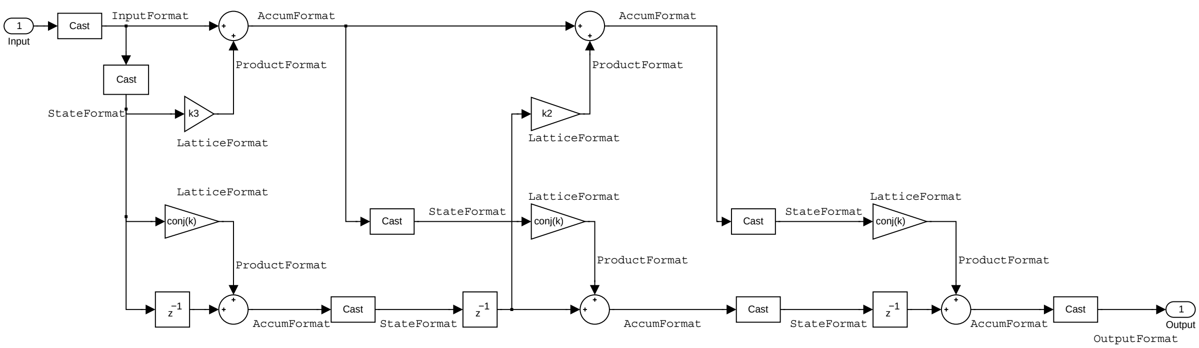
```
k = [.66 .7 .44];
hd=dfilt.latticeallpass({k});
hq = copy(hd)
set(hq,'arithmetic','fixed');
```

Lattice Moving Average Maximum Phase Filter Structure

In the next figure you see a *lattice moving average maximum phase* filter structure. This signal flow diagram directly realizes a third-order lattice moving average (MA) filter with the following phase form depending on the initial transfer function:

- When you start with a minimum phase transfer function, the upper branch of the resulting lattice structure returns a minimum phase filter. The lower branch returns a maximum phase filter.
- When your transfer function is neither minimum phase nor maximum phase, the lattice moving average maximum phase structure will not be maximum phase.
- When you start with a maximum phase filter, the resulting lattice filter is maximum phase also.

The filter reflection coefficients are labeled $k(i)$, $i = 1, 2, 3$. The states (used for initial and final state values in filtering) are labeled $z(i)$. In the figure, we set the `Arithmetic` property to `fixed` to reveal the fixed-point arithmetic format features that control such options as word length and fraction length.

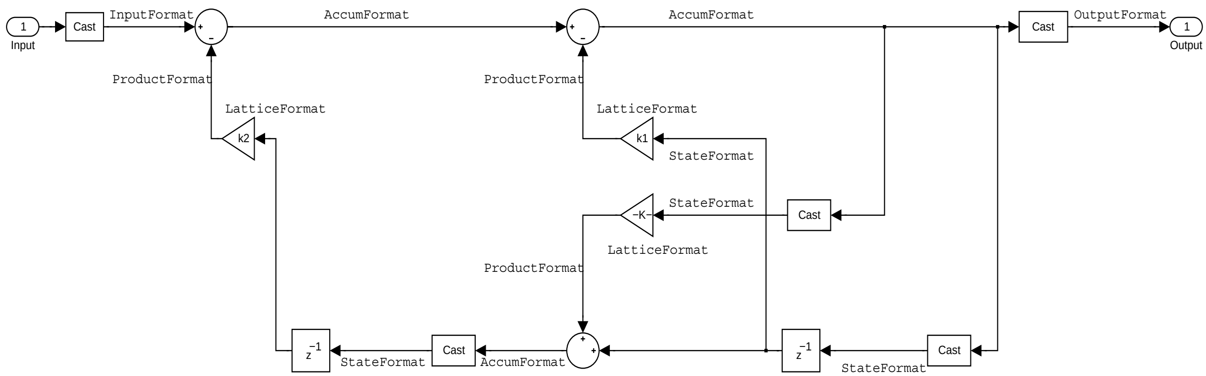


Example—Constructing a Lattice Moving Average Maximum Phase Filter. Constructing a fourth-order lattice MA maximum phase filter structure for a quantized filter `hq` begins with the following code.

```
k = [.66 .7 .44 .33];
hd=dfilt.latticemamax({k});
```

Lattice Autoregressive (AR) Filter Structure

The method `dfilt.latticear` directly realizes lattice autoregressive filters in the toolbox. The following figure depicts the third-order *lattice autoregressive (AR)* filter structure—with the `Arithmetic` property equal to `fixed`. The filter reflection coefficients are labeled $k(i)$, $i = 1, 2, 3$, and the states (used for initial and final state values in filtering) are labeled $z(i)$.



Example—Specifying a Lattice AR Filter. You can specify a third-order lattice AR filter structure for a quantized filter `hq` with the following code.

```
k = [.66 .7 .44];
hd=dfilt.latticear({k});
hq = copy(hd);
hq.arithmetic = 'custom';
```

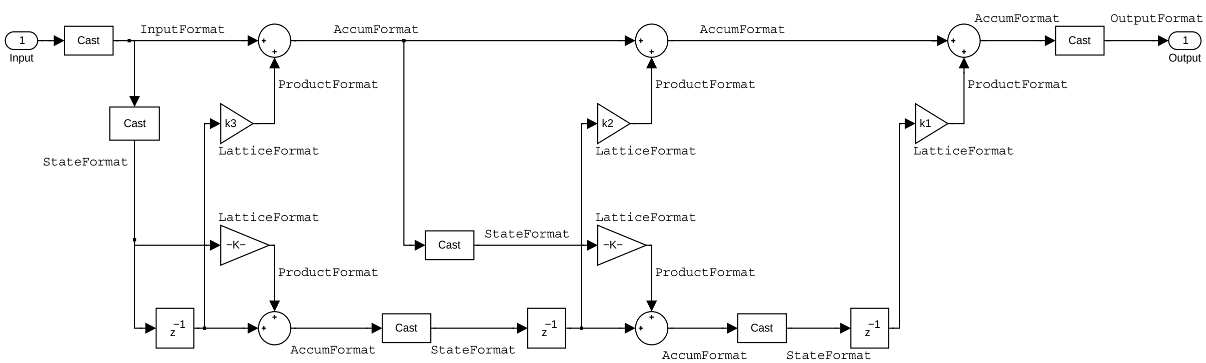
Lattice Moving Average (MA) Filter Structure for Minimum Phase

The following figures depict *lattice moving average (MA)* filter structures that directly realize third-order lattice MA filters for minimum phase. The filter reflection coefficients are labeled $k(i)$, $i = 1, 2, 3$, and the states (used for initial and final state values in filtering) are labeled $z(i)$. Setting the `Arithmetic` property of the filter to `fixed` results in a fixed-point filter that matches the figure.

This signal flow diagram directly realizes a third-order lattice moving average (MA) filter with the following phase form depending on the initial transfer function:

- When you start with a minimum phase transfer function, the upper branch of the resulting lattice structure returns a minimum phase filter. The lower branch returns a minimum phase filter.
- When your transfer function is neither minimum phase nor maximum phase, the lattice moving average minimum phase structure will not be minimum phase.
- When you start with a minimum phase filter, the resulting lattice filter is minimum phase also.

The filter reflection coefficients are labeled $k(i)$, $i = 1, 2, 3$. The states (used for initial and final state values in filtering) are labeled $z(i)$. In the figure, we set the `Arithmetic` property to `fixed` to reveal the fixed-point arithmetic format features that control such options as word length and fraction length.

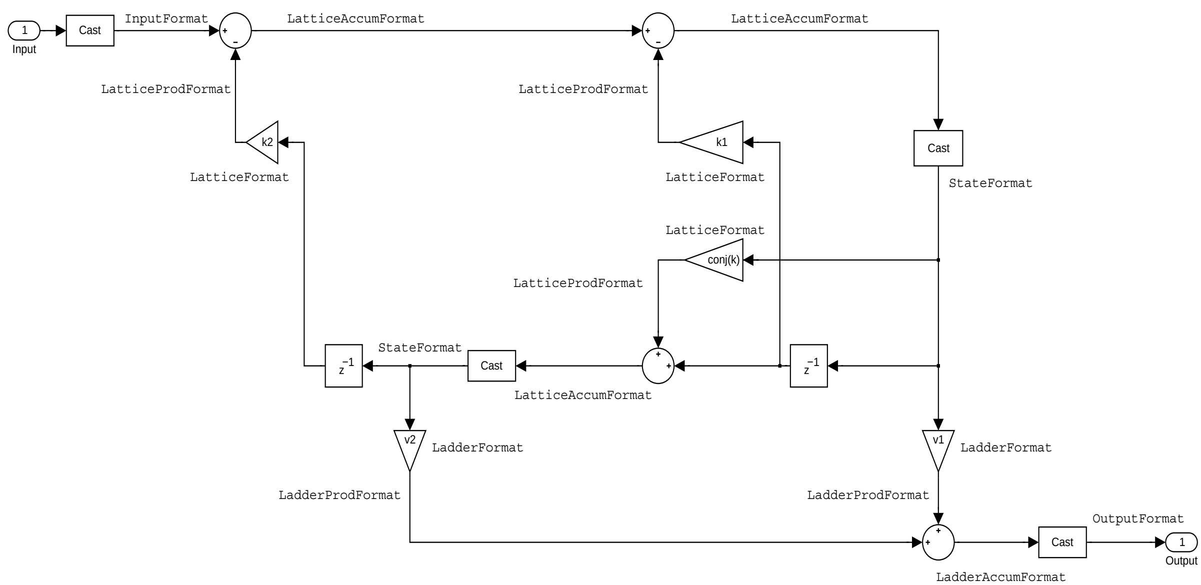


Example—Specifying a Minimum Phase Lattice MA Filter. You can specify a third-order lattice MA filter structure for minimum phase applications using variations of the following code.

```
k = [.66 .7 .44];
hd=dfilt.latticemamin({k});
hq = copy(hd);
set(hq,'arithmetic','fixed');
```

Lattice Autoregressive Moving Average (ARMA) Filter Structure

The figure below depicts a *lattice autoregressive moving average (ARMA)* filter structure that directly realizes a fourth-order lattice ARMA filter. The filter reflection coefficients are labeled $k(i)$, i = 1, ..., 4; the ladder coefficients are labeled $v(i)$, i = 1, 2, 3; and the states (used for initial and final state values in filtering) are labeled $z(i)$.

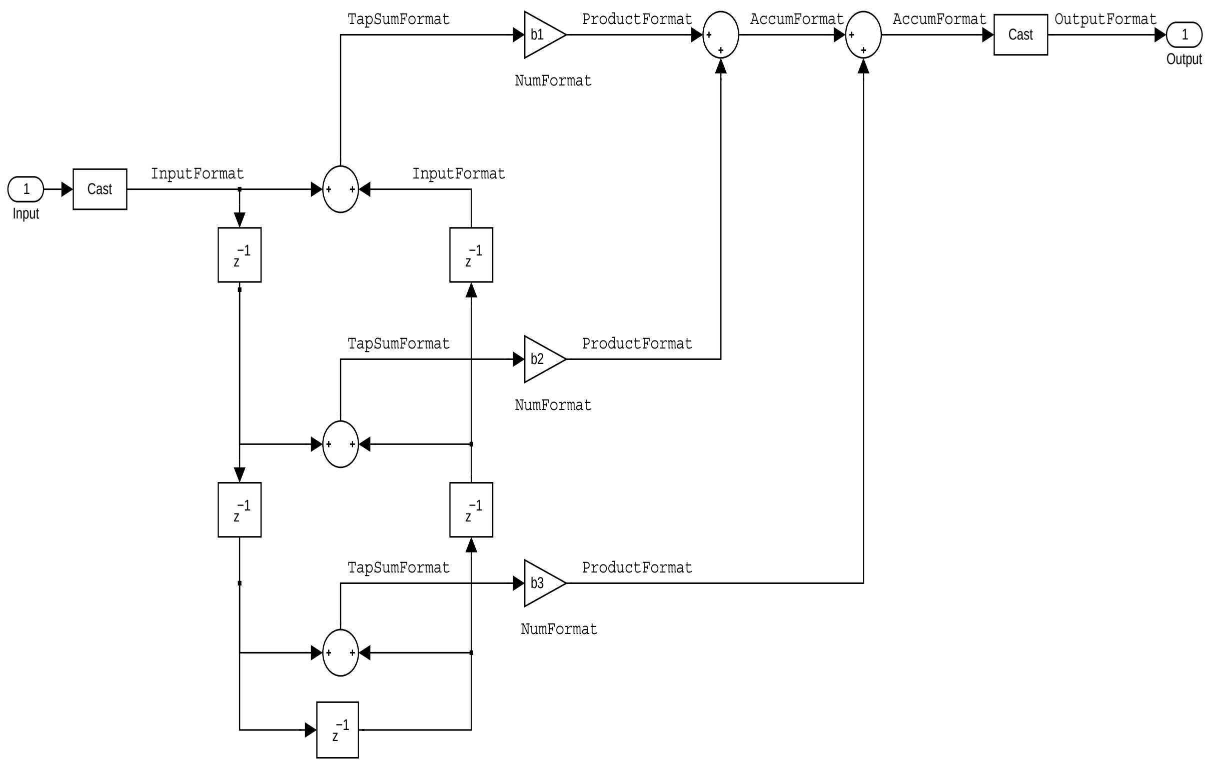


Example—Specifying an Lattice ARMA Filter. The following code specifies a fourth-order lattice ARMA filter structure for a quantized filter `hq`, starting from `hd`, a floating-point version of the filter.

```
k = [.66 .7 .44 .66];
v = [1 0 0];
hd=dfilt.latticearma({k,v});
hq = copy(hd);
hq.arithmetic = 'fixed';
```

Direct Form Symmetric FIR Filter Structure (Any Order)

Shown in the next figure, you see signal flow that depicts a *direct form symmetric FIR* filter structure that directly realizes a fifth-order direct form symmetric FIR filter. Filter coefficients are labeled $b(i)$, $i = 1, ..., n$, and states (used for initial and final state values in filtering) are labeled $z(i)$. Showing the filter structure used when you select `fixed` for the `Arithmetic` property value, the first figure details the properties in the filter object.



Example—Specifying an Odd-Order Direct Form Symmetric FIR Filter. By using the following code in MATLAB, you can specify a fifth-order direct form symmetric FIR filter for a fixed-point filter `hq`:

```
b = [-0.008 0.06 0.44 0.44 0.06 -0.008];
hd=dfilt.dfsymfir({b});
```

```
hq = copy(hd);
set(hq,'arithmetic','fixed');
```

Assigning Filter Coefficients

The syntax you use to assign filter coefficients for your floating-point or fixed-point filter depends on the structure you select for your filter.

Converting Filters Between Representations

Filter conversion functions in this toolbox and in the Signal Processing Toolbox let you convert filter transfer functions to other filter forms, and from other filter forms to transfer function form. Relevant conversion functions include the following functions.

Conversion Function	Description
`ca2tf`	Converts from a coupled allpass filter to a transfer function.
`cl2tf`	Converts from a lattice coupled allpass filter to a transfer function.
`convert`	Convert a discrete-time filter from one filter structure to another.
`sos`	Converts quantized filters to create second-order sections. We recommend this method for converting quantized filters to second-order sections.
`tf2ca`	Converts from a transfer function to a coupled allpass filter.
`tf2cl`	Converts from a transfer function to a lattice coupled allpass filter.
`tf2latc`	Converts from a transfer function to a lattice filter.
`tf2sos`	Converts from a transfer function to a second-order section form.

Conversion Function	Description
tf2ss	Converts from a transfer function to state-space form.
tf2zp	Converts from a rational transfer function to its factored (single section) form (zero-pole-gain form).
zp2sos	Converts a zero-pole-gain form to a second-order section form.
zp2ss	Conversion of zero-pole-gain form to a state-space form.
zp2tf	Conversion of zero-pole-gain form to transfer functions of multiple order sections.

Note that these conversion routines do not apply to dfilt objects.

Function convert is a special case—when you use convert to change the filter structure of a fixed-point filter, you lose all of the filter states and settings. Your new filter has default values for all properties, and it in not fixed-point.

To demonstrate the changes that occur, convert a fixed-point direct form I transposed filter to direct form II structure.

```
hd=dfilt.df1t

hd =

          FilterStructure: 'Direct-Form I Transposed'
               Arithmetic: 'double'
                Numerator: 1
              Denominator: 1
         PersistentMemory: false
                   States: Numerator:  [0x0 double]
                           Denominator:[0x0 double]
      NumSamplesProcessed: 0


hd.arithmetic='fixed'
```

```
hd =

         FilterStructure: 'Direct-Form I Transposed'
              Arithmetic: 'fixed'
               Numerator: 1
             Denominator: 1
        PersistentMemory: false
                  States: Numerator:  [0x0 fi]
                          Denominator:[0x0 fi]
     NumSamplesProcessed: 0

convert(hd,'df2')

Warning: Using reference filter for structure conversion.
Fixed-point attributes will not be converted.

ans =

         FilterStructure: 'Direct-Form II'
              Arithmetic: 'double'
               Numerator: 1
             Denominator: 1
        PersistentMemory: false
                  States: [0x1 double]
     NumSamplesProcessed: 0
```

You can specify a filter with L sections of arbitrary order by

1. Factoring your entire transfer function with `tf2zp`. This converts your transfer function to zero-pole-gain form.

2. Using `zp2tf` to compose the transfer function for each section from the selected first-order factors obtained in step 1.

Note You are not required to normalize the leading coefficients of each section's denominator polynomial when you specify second-order sections, though `tf2sos` does.

Gain

`dfilt.scalar` filters have a gain value stored in the `gain` property. By default the gain value is one—the filter acts as a wire.

InputFracLength

`InputFracLength` defines the fraction length assigned to the input data for your filter. Used in tandem with `InputWordLength`, the pair defines the data format for input data you provide for filtering.

As with all fraction length properties in `dfilt` objects, the value you enter here can be any negative or positive integer, or zero. Fraction length can be larger than the associated word length, in this case `InputWordLength`, as well.

InputWordLength

Specifies the number of bits your filter uses to represent your input data. Your word length option is limited by the arithmetic you choose—up to 32 bits for `double`, `float`, and `fixed`. Setting `Arithmetic` to `single` (single-precision floating-point) limits word length to 16 bits. The default value is 16 bits.

Ladder

Included as a property in `dfilt.latticearma` filter objects, `Ladder` contains the denominator coefficients that form an IIR lattice filter object. For instance, the following code creates a high pass filter object that uses the lattice ARMA structure.

```
[b,a]=cheby1(5,.5,.5,'high')

b =

    0.0282   -0.1409    0.2817   -0.2817    0.1409   -0.0282
```

```
a =

    1.0000    0.9437    1.4400    0.9629    0.5301    0.1620

hd=dfilt.latticearma(b,a)

hd =

         FilterStructure: [1x44 char]
              Arithmetic: 'double'
                 Lattice: [1x6 double]
                  Ladder: [1 0.9437 1.4400 0.9629 0.5301 0.1620]
     PersistentMemory: false
                  States: [6x1 double]
     NumSamplesProcessed: 0

hd.arithmetic='fixed'

hd =

         FilterStructure: [1x44 char]
              Arithmetic: 'fixed'
                 Lattice: [1x6 double]
                  Ladder: [1 0.9437 1.4400 0.9629 0.5301 0.1620]
     PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

         StateWordLength: 16
         StateFracLength: 15

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
```

```
OverflowMode: 'wrap'
```

LadderAccumFracLength

Autoregressive, moving average lattice filter objects (`lattticearma`) use ladder coefficients to define the filter. In combination with `LadderFracLength` and `CoeffWordLength`, these three properties specify or reflect how the accumulator outputs data stored there. As with all fraction length properties, `LadderAccumFracLength` can be any integer, including integers larger than `AccumWordLength`, and positive or negative integers. The default value is 29 bits.

LadderFracLength

To let you control the way your `latticearma` filter interprets the denominator coefficients, `LadderFracLength` sets the fraction length applied to the ladder coefficients for your filter. The default value is 14 bits.

As with all fraction length properties, `LadderFracLength` can be any integer, including integers larger than `AccumWordLength`, and positive or negative integers.

Lattice

When you create a lattice-based IIR filter, your numerator coefficients (from your IIR prototype filter or the default `dfilt` lattice filter function) get stored in the `Lattice` property of the `dfilt` object. The properties `CoeffWordLength` and `LatticeFracLength` define the data format the object uses to represent the lattice coefficients. By default, lattice coefficients are in double-precision format.

LatticeAccumFracLength

Lattice filter objects (`latticeallpass`, `latticearma`, `latticemamax`, and `latticemamin`) use lattice coefficients to define the filter. In combination with `LatticeFracLength` and `CoeffWordLength`, these three properties specify how the accumulator outputs lattice coefficient-related data stored there. As with all fraction length properties, `LatticeAccumFracLength` can be any integer, including integers larger than `AccumWordLength`, and positive or negative integers. By default, the property is set to 31 bits.

LatticeFracLength

To let you control the way your filter interprets the denominator coefficients, `LatticeFracLength` sets the fraction length applied to the lattice coefficients for your lattice filter. When you create the default lattice filter, `LatticeFracLength` is 16 bits.

As with all fraction length properties, `LatticeFracLength` can be any integer, including integers larger than `CoeffWordLength`, and positive or negative integers.

MultiplicandFracLength

Each input data element for a multiply operation has both word length and fraction length to define its representation. `MultiplicandFracLength` sets the fraction length to use when the filter object performs any multiply operation during filtering. For default filters, this is set to 15 bits.

As with all word and fraction length properties, `MultiplicandFracLength` can be any integer, including integers larger than `CoeffWordLength`, and positive or negative integers.

MultiplicandWordLength

Each input data element for a multiply operation has both word length and fraction length to define its representation. `MultiplicandWordLength` sets the word length to use when the filter performs any multiply operation during filtering. For default filters, this is set to 16 bits. Only the `df1t` and `df1tsos` filter objects include the `MultiplicandFracLength` property.

Only the `df1t` and `df1tsos` filter objects include the `MultiplicandWordLength` property.

NumAccumFracLength

Filter structures `df1`, `df1t`, `df2`, and `df2t` that use `fixed` arithmetic have this property that defines the fraction length applied to numerator coefficients in output from the accumulator. In combination with `AccumWordLength`, the NumAccumFracLength property fully specifies how the accumulator outputs numerator-related data stored there.

As with all fraction length properties, `NumAccumFracLength` can be any integer, including integers larger than `AccumWordLength`, and positive or negative integers. 30 bits is the default value when you create the filter object. To be able to change the value for this property, set `AccumMode` for the filter to `SpecifyPrecision`.

Numerator

The numerator coefficients for your filter, taken from the prototype you start with or from the default filter, are stored in this property. Generally this is a 1-by-N array of data in double format, where N is the length of the filter.

All of the filter objects include `Numerator`, except the lattice-based and second-order section filters, such as `dfilt.latticema` and `dfilt.df1tsos`.

NumFracLength

Property `NumFracLength` contains the value that specifies the fraction length for the numerator coefficients for your filter. `NumFracLength` specifies the fraction length used to interpret the numerator coefficients. Used in combination with `CoeffWordLength`, these two properties define the interpretation of the coefficients stored in the vector that contains the numerator coefficients.

As with all fraction length properties, the value you enter here can be any negative or positive integer, or zero. Fraction length can be larger than the associated word length, as well. By default, the value is 15 bits, with the `CoeffWordLength` of 16 bits.

NumProdFracLength

A property of all of the direct form IIR `dfilt` objects, except the ones that implement second-order sections, `NumProdFracLength` specifies the fraction length applied to data output from product operations the filter performs on numerator coefficients.

Looking at the signal flow diagram for the `dfilt.df1t` filter, for example, you see that denominators and numerators are handled separately. When you set `ProductMode` to `SpecifyPrecision`, you can change the `NumProdFracLength` setting manually. Otherwise, for multiplication operations that use the numerator coefficients, the filter sets the word length as defined by the `ProductMode` setting.

NumSamplesProcessed

Contains the count of the samples processed by the filter, for each iteration or the total for all iterations. How the filter counts the number of processed samples depends on the property value you set for `PersistentMemory`. When you set `PersistentMemory` to `false`, the number of samples processed resets to zero for each filtering process. For example, in a loop, `NumSamplesProcessed` reports the number of samples for each loop iteration, resetting to zero each time the loop start again. Starting at zero for each pass through the loop, this value lets you know how many samples the filter actually processed during each run.

When you set `ResetBeforFiltering` to `off`, `NumSamplesProcessed` does not reset to zero for each loop iteration in your test or program, thus it contains the total count of the samples processed during all filtering.

NumStateFracLength

All the variants of the direct form I structure include the property `NumStateFracLength` to store the fraction length applied to the numerator states for your filter object. By default, this property has the value 15 bits, with the `CoeffWordLength` of 16 bits, which you can change after you create the filter object.

As with all fraction length properties, the value you enter here can be any negative or positive integer, or zero. Fraction length can be larger than the associated word length, as well.

NumStateWordLength

When you look at the flow diagram for the `df1sos` filter object, the states associated with the numerator coefficient operations take the data format from this property and the `NumStateFracLength` property. In combination, these properties fully specify how the filter interprets the state it uses.

As with all fraction length properties, the value you enter here can be any negative or positive integer, or zero. Fraction length can be larger than the associated word length, as well. By default, the value is 16 bits, with the `NumStateFracLength` of 11 bits.

OutputFracLength

To define the output from your filter object, you need both the word and fraction lengths. `OutputFracLength` determines the fraction length applied to interpret the output data. Combining this with `OutputWordLength` fully specifies the format of the output.

Your fraction length can be any negative or positive integer, or zero. In addition, the fraction length you specify can be larger than the associated word length. Generally, the default value is 11 bits.

OutputMode

Sets the mode the filter uses to scale the filtered (output) data. You have the following choices:

- `AvoidOverflow`—directs the filter to set the property that controls the output data fraction length to avoid causing the data to overflow. In a `df2` filter, this would be the `OutputFracLength` property.
- `BestPrecision`—directs the filter to set the property that controls the output data fraction length to maximize the precision in the output data. For `df1t` filters, this is the `OutputFracLength` property. When you change the word length (`OutputWordLength`), the filter adjusts the fraction length to maintain the best precision for the new word size.
- `SpecifyPrecision`—lets you set the fraction length used by the filtered data. When you select this choice, you can set the output fraction length using the `OutputFracLength` property to define the output precision.

All filters include this property except the direct form I filter which takes the output format from the filter states.

Here is an example that changes the mode setting to `bestprecision`, and then adjusts the word length for the output.

```
hd=dfilt.df2

hd =

         FilterStructure: 'Direct-Form II'
              Arithmetic: 'double'
               Numerator: 1
             Denominator: 1
```

```
        PersistentMemory: false
                  States: [0x1 double]
      NumSamplesProcessed: 0

hd.arithmetic='fixed'

hd =

          FilterStructure: 'Direct-Form II'
               Arithmetic: 'fixed'
                Numerator: 1
              Denominator: 1
         PersistentMemory: false
                   States: [1x1 embedded.fi]
      NumSamplesProcessed: 0

           CoeffWordLength: 16
            CoeffAutoScale: true
                    Signed: true

           InputWordLength: 16
           InputFracLength: 15

          OutputWordLength: 16
                OutputMode: 'AvoidOverflow'

           StateWordLength: 16
           StateFracLength: 15

               ProductMode: 'FullPrecision'

                 AccumMode: 'KeepMSB'
           AccumWordLength: 40
             CastBeforeSum: true

                 RoundMode: 'convergent'
              OverflowMode: 'wrap'

get(hd)
          PersistentMemory: false
```

```
     NumSamplesProcessed: 0
         FilterStructure: 'Direct-Form II'
                  States: [1x1 embedded.fi]
               Numerator: 1
             Denominator: 1
              Arithmetic: 'fixed'
         CoeffWordLength: 16
          CoeffAutoScale: 1
                  Signed: 1
               RoundMode: 'convergent'
            OverflowMode: 'wrap'
         InputWordLength: 16
         InputFracLength: 15
        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'
             ProductMode: 'FullPrecision'
               AccumMode: 'KeepMSB'
         StateWordLength: 16
         StateFracLength: 15
           NumFracLength: 14
           DenFracLength: 14
        OutputFracLength: 13
       ProductWordLength: 32
       NumProdFracLength: 29
       DenProdFracLength: 29
         AccumWordLength: 40
      NumAccumFracLength: 29
      DenAccumFracLength: 29
           CastBeforeSum: 1

hd.outputMode='bestprecision'

hd =

         FilterStructure: 'Direct-Form II'
              Arithmetic: 'fixed'
               Numerator: 1
             Denominator: 1
        PersistentMemory: false
                  States: [1x1 embedded.fi]
```

```
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'BestPrecision'

         StateWordLength: 16
         StateFracLength: 15

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'

hd.outputWordLength=8;

get(hd)
        PersistentMemory: false
     NumSamplesProcessed: 0
         FilterStructure: 'Direct-Form II'
                  States: [1x1 embedded.fi]
               Numerator: 1
             Denominator: 1
              Arithmetic: 'fixed'
         CoeffWordLength: 16
          CoeffAutoScale: 1
                  Signed: 1
               RoundMode: 'convergent'
            OverflowMode: 'wrap'
         InputWordLength: 16
```

```
      InputFracLength: 15
     OutputWordLength: 8
           OutputMode: 'BestPrecision'
          ProductMode: 'FullPrecision'
            AccumMode: 'KeepMSB'
      StateWordLength: 16
      StateFracLength: 15
        NumFracLength: 14
        DenFracLength: 14
     OutputFracLength: 5
    ProductWordLength: 32
    NumProdFracLength: 29
    DenProdFracLength: 29
      AccumWordLength: 40
   NumAccumFracLength: 29
   DenAccumFracLength: 29
        CastBeforeSum: 1
```

Changing the OutputWordLength to 8 bits caused the filter to change the OutputFracLength to 5 bits to keep the best precision for the output data.

OutputWordLength

Use the property OutputWordLength to set the word length used by the output from your filter. Set this property to a value that matches your intended hardware. For example, some digital signal processors use 32-bit output so you would set OutputWordLength to 32.

```
[b,a] = butter(6,.5);
hd=dfilt.df1t(b,a);

set(hd,'arithmetic','fixed')

hd

hd =

         FilterStructure: 'Direct-Form I Transposed'
              Arithmetic: 'fixed'
               Numerator: [1x7 double]
             Denominator: [1 0 0.7777 0 0.1142 0 0.0018]
        PersistentMemory: false
                  States: Numerator:  [6x1 fi]
```

```
                     Denominator:[6x1 fi]
     NumSamplesProcessed: 0


         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

  MultiplicandWordLength: 16
  MultiplicandFracLength: 15

         StateWordLength: 16
          StateAutoScale: true

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'
```

```
hd.outputwordLength=32

hd =

          FilterStructure: 'Direct-Form I Transposed'
               Arithmetic: 'fixed'
                Numerator: [1x7 double]
              Denominator: [1 0 0.7777 0 0.1142 0 0.0018]
         PersistentMemory: false
                   States: Numerator:  [6x1 fi]
                           Denominator:[6x1 fi]
      NumSamplesProcessed: 0


          CoeffWordLength: 16
```

```
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 32
              OutputMode: 'AvoidOverflow'

  MultiplicandWordLength: 16
  MultiplicandFracLength: 15

         StateWordLength: 16
          StateAutoScale: true

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'
```

When you create a filter object, this property starts with the value 16.

OverflowMode

The `OverflowMode` property is specified as one of the following two strings indicating how to respond to overflows in fixed-point arithmetic:

- `'saturate'`—saturate overflows.

  When the values of data to be quantized lie outside of the range of the largest and smallest representable numbers (as specified by the applicable word length and fraction length properties), these values are quantized to the value of either the largest or smallest representable value, depending on which is closest.

- `'wrap'`—wrap all overflows to the range of representable values.

  When the values of data to be quantized lie outside of the range of the largest and smallest representable numbers (as specified by the data format properties), these values are wrapped back into that range using modular arithmetic relative to the smallest representable number. You can learn more about modular arithmetic in the Fixed-Point Toolbox.

These rules apply to the `OverflowMode` property.

- Applies to the accumulator and output data only.
- Does not apply to coefficients or input data. These always saturate the results.
- Does not apply to products. Products maintain full precision at all times. Your filters do not lose precision in the products.

Default value: `'saturate'`

Note Numbers in floating-point filters that extend beyond the dynamic range overflow to `±inf`.

ProductFracLength

After you set `ProductMode` for a fixed-point filter to `SpecifyPrecision`, this property becomes available for you to change. `ProductFracLength` sets the fraction length the filter uses for the results of multiplication operations. Only the FIR filters such as asymmetric FIRs or lattice autoregressive filters include this dynamic property.

Your fraction length can be any negative or positive integer, or zero. In addition, the fraction length you specify can be larger than the associated word length. Generally, the default value is 11 bits.

ProductMode

This property, available when your filter is in fixed-point arithmetic mode, specifies how the filter outputs the results of multiplication operations. All `dfilt` objects include this property when they use fixed-point arithmetic.

When available, you select from one of the following values for `ProductMode`:

- `FullPrecision`—means the filter automatically chooses the word length and fraction length it uses to represent the results of multiplication operations.

The setting allow the product to retain the precision provided by the inputs (multiplicands) to the operation.

- `KeepMSB`—means you specify the word length for representing product operation results. The filter sets the fraction length to discard the LSBs, keep the higher order bits in the data, and maintain the precision.
- `KeepLSB`—means you specify the word length for representing the product operation results. The filter sets the fraction length to discard the MSBs, keep the lower order bits, and maintain the precision. Compare to the `KeepMSB` option.
- `SpecifyPrecision`—means you specify the word length and the fraction length to apply to data output from product operations.

When you switch to fixed-point filtering from floating-point, you are most likely going to throw away some data bits after product operations in your filter, perhaps because you have limited resources. When you have to discard some bits, you might choose to discard the least significant bits (LSB) from a result since the resulting quantization error would be small as the LSBs carry less weight. Or you might choose to keep the LSBs because the results have MSBs that are mostly zero, such as when your values are small relative to the range of the format in which they are represented. So the options for `ProductMode` let you choose how to maintain the information you need from the accumulator.

For more information about data formats, word length, and fraction length in fixed-point arithmetic, refer to “Notes About Fraction Length, Word Length, and Precision” on page 8-32.

ProductWordLength

You use `ProductWordLength` to define the data word length used by the output from multiplication operations. Set this property to a value that matches your intended application. For example, the default value is 32 bits, but you can set any word length.

```
set(hq,'arithmetic','fixed');
set(hq,'ProductWordLength',64);
```

Note that `ProductWordLength` applies only to filters whose `Arithmetic` property value is `fixed`.

PersistentMemory

Note that PersistentMemory affects the reported number of samples processed by the filter and stored in the property NumSamplesProcessed.

Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter object. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false—the filter does not retain memory about filtering operations from one to the next. Maintaining memory (setting PersistentMemory to true) lets you filter large data sets as collections of smaller subsets and get the same result.

In this example, filter hd first filters data xtot in one pass. Then we use hd to filter x as two separate data sets. The results ytot and ysec are the same in both cases.

```
xtot=[x,x];
ytot=filter(hd,xtot)
ytot =

         0   -0.0003    0.0005   -0.0014    0.0028   -0.0054    0.0092
reset(hm1);  % Clear history of the filter
hm1.PersistentMemory='true';
ysec=[filter(hd,x) filter(hd,x)]

ysec =

         0   -0.0003    0.0005   -0.0014    0.0028   -0.0054    0.0092
```

This test verifies that ysec (the signal filtered by sections) is equal to ytot (the entire signal filtered at once).

RoundMode

The RoundMode property value specifies the rounding method used for quantizing numerical values. Specify the RoundMode property values as one of the following five strings.

RoundMode String	Description of Rounding Algorithm
'ceil'	Round up to the next representable quantized value.
'convergent'	Round to the nearest representable quantized value. Numbers that are exactly halfway between the two nearest representable quantized values are rounded up when the least significant bit would be set to 1 after rounding. Otherwise, the number is rounded down. Filter objects use convergent rounding by default.
'fix'	Round negative numbers up and positive numbers down to the next representable quantized value.
'floor'	Round down to the next representable quantized value.
'round'	Round to the nearest representable quantized value. Numbers that are halfway between the two nearest representable quantized values are rounded up.

Default value: 'convergent'

The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

ScaleValueFracLength

Filter structures df1sos, df1tsos, df2sos, and df2tsos that use fixed arithmetic have this property that defines the fraction length applied to the scale values the filter uses between sections. In combination with CoeffWordLength, these two properties fully specify how the filter interprets and uses the scale values stored in the property ScaleValues. As with fraction length properties, ScaleValueFracLength can be any integer, including integers larger than CoeffWordLength, and positive or negative integers. 15 bits is the default value when you create the filter.

ScaleValues

The ScaleValues property values are specified as a scalar (or vector) that introduces scaling for inputs (and the outputs from cascaded sections in the vector case) during filtering:

- When you only have a single section in your filter:
  - Specify the ScaleValues property value as a scalar if you only want to scale the input to your filter.
  - Specify the ScaleValues property as a vector of length 2 if you want to specify scaling to the input (scaled with the first entry in the vector) and the output (scaled with the last entry in the vector).
- When you have L cascaded sections in your filter:
  - Specify the ScaleValues property value as a scalar if you only want to scale the input to your filter.
  - Specify the value for the ScaleValues property as a vector of length L+1 if you want to scale the inputs to every section in your filter, along with the output:
    - The first entry of your vector specifies the input scaling
  - Each successive entry specifies the scaling at the output of the next section
  - The final entry specifies the scaling for the filter output.

The interpretation of this property is described below with diagrams in “Interpreting the ScaleValues Property”.

Default value: 0

Remarks: The value of the ScaleValues property is not quantized. Data affected by the presence of a scaling factor in the filter is quantized according to the appropriate data format.

When you apply `normalize` to a fixed-point filter, the value for the `ScaleValues` property is changed accordingly.

It is good practice to choose values for this property that are either positive or negative powers of two.

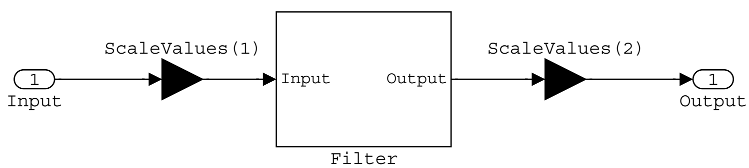
Interpreting the ScaleValues Property

When you specify the values of the `ScaleValues` property of a quantized filter, the values are entered as a vector, the length of which is determined by the number of cascaded sections in your filter:

- When you have only one section, the value of the `Scalevalues` property can be a a scalar or a two-element vector.
- When you have L cascaded sections in your filter, the value of the `Scalevalues` property can be a scalar or an L+1-element vector.

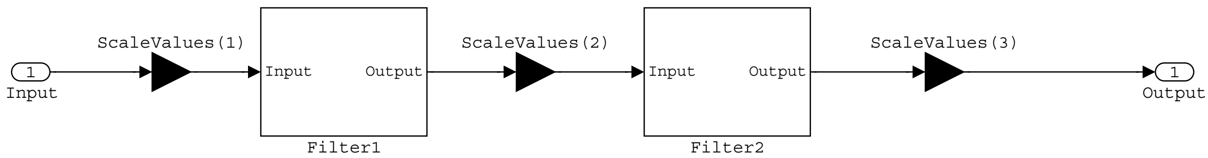
The following diagram shows how the `ScaleValues` property values are applied to a quantized filter with only one section.

Application of ScaleValues to a Single Section



The following diagram shows how the `ScaleValues` property values are applied to a quantized filter with two sections.

Application of ScaleValues to Multiple Sections



Signed

When you create a `dfilt` object for fixed-point filtering (you set the property `Arithmetic` to `fixed`, the property `Signed` specifies whether the filter interprets coefficients as signed or unsigned. This setting applies only to the coefficients. While the default setting is `true`, meaning that all coefficients are assumed to be signed, you can change the setting to `false` after you create the fixed-point filter.

For example, create a fixed-point direct-form II transposed filter with both negative and positive coefficients, and then change the property value for `Signed` from `true` to `false` to see what happens to the negative coefficient values.

```
hd=dfilt.df2t(-5:5)

hd =

         FilterStructure: 'Direct-Form II Transposed'
              Arithmetic: 'double'
               Numerator: [-5 -4 -3 -2 -1 0 1 2 3 4 5]
             Denominator: 1
        PersistentMemory: false
                  States: [10x1 double]
     NumSamplesProcessed: 0

set(hd,'arithmetic','fixed')
hd.numerator
```

```
ans =

    -5    -4    -3    -2    -1     0     1     2     3     4     5

set(hd,'signed',false)
hd.numerator

ans =

     0     0     0     0     0     0     1     2     3     4     5
```

Using unsigned coefficients limits you to using only positive coefficients in your filter. `Signed` is a dynamic property—you cannot set or change it until you switch the setting for the `Arithmetic` property to `fixed`.

SosMatrix

When you convert a `dfilt` object to second-order section form, or create a second-order section filter, `sosMatrix` holds the filter coefficients as property values. Using the `double` data type by default, the matrix is in [sections coefficients per section] form, displayed as `[15-x-6]` for filters with 6 coefficients per section and 15 sections, [15 6].

To demonstrate, the following code creates an order 30 filter using second-order sections in the direct-form II transposed configuration. Notice the `sosMatrix` property contains the coefficients for all the sections.

```
d = fdesign.lowpass('n,fc',30,0.5);
hd = butter(d);

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [15x6 double]
             ScaleValues: [16x1 double]
        PersistentMemory: false
                  States: [2x15 double]
     NumSamplesProcessed: 0
```

```
hd.arithmetic='fixed'

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'fixed'
               sosMatrix: [15x6 double]
             ScaleValues: [16x1 double]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

  SectionInputWordLength: 16
   SectionInputAutoScale: true

 SectionOutputWordLength: 16
  SectionOutputAutoScale: true

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

         StateWordLength: 16
         StateFracLength: 15

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'
```

```
hd.sosMatrix

ans =

    1.0000    2.0000    1.0000    1.0000         0    0.9005
    1.0000    2.0000    1.0000    1.0000         0    0.7294
    1.0000    2.0000    1.0000    1.0000         0    0.5888
    1.0000    2.0000    1.0000    1.0000         0    0.4724
    1.0000    2.0000    1.0000    1.0000         0    0.3755
    1.0000    2.0000    1.0000    1.0000         0    0.2948
    1.0000    2.0000    1.0000    1.0000         0    0.2275
    1.0000    2.0000    1.0000    1.0000         0    0.1716
    1.0000    2.0000    1.0000    1.0000         0    0.1254
    1.0000    2.0000    1.0000    1.0000         0    0.0878
    1.0000    2.0000    1.0000    1.0000         0    0.0576
    1.0000    2.0000    1.0000    1.0000         0    0.0344
    1.0000    2.0000    1.0000    1.0000         0    0.0173
    1.0000    2.0000    1.0000    1.0000         0    0.0062
    1.0000    2.0000    1.0000    1.0000         0    0.0007
```

The SOS matrix is an M-by-6 matrix, where M is the number of sections in the second-order section filter. Filter `hd` has M equal to 15 as shown above (15 rows). Each row of the SOS matrix contains the numerator and denominator coefficients (b's and a's) and the scale factors of the corresponding section in the filter.

SectionInputAutoScale

Second-order section filters include this property that determines who the filter handles data in the transitions from one section to the next in the filter.

How the filter represents the data passing from one section to the next depends on the property value of `SectionInputAutoScale`. The representation the filter uses between the filter sections depends on whether the value of `SectionInputAutoScale` is `true` or `false`.

- `SectionInputAutoScale = true` means the filter chooses the fraction length to maintain the value of the data between sections as close to the output values from the previous section as possible. `true` is the default setting.
- `SectionInputAutoScale = false` removes the automatic scaling of the fraction length for the intersection data and exposes the property that

controls the coefficient fraction length (SectionInputFracLength) so you can change it. For example, if the filter is a second-order, direct form FIR filter, setting SectionInputAutoScale = false exposes the SectionInputFracLength property that specifies the fraction length applied to data between the sections.

SectionInputFracLength

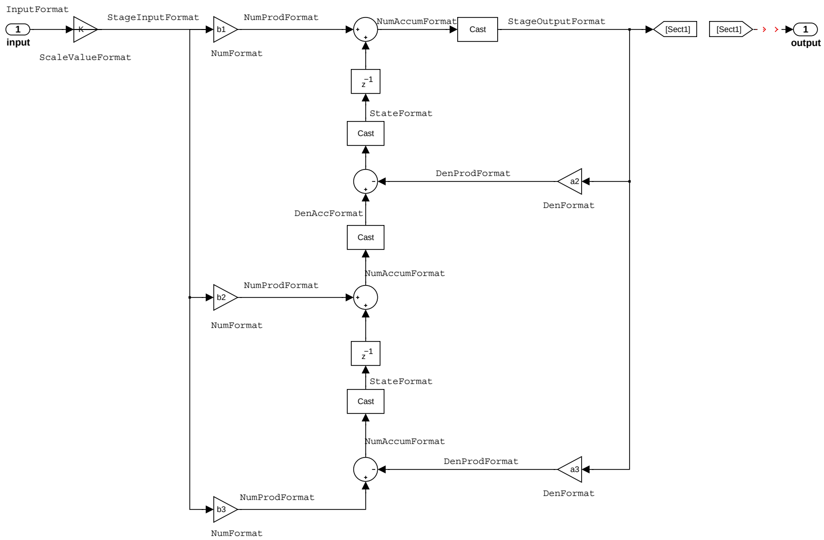
Second-order section filters use quantizers at the input to each section of the filter. The quantizers apply to the input data entering each filter section. Note that the quantizers for each section are the same. To set the fraction length for interpreting the input values, use the property value in SectionInputFracLength.

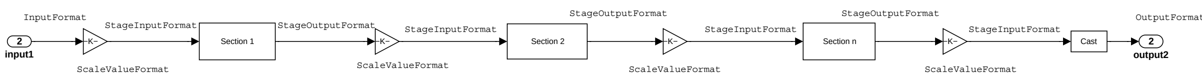
In combination with CoeffWordLength, SectionInputFracLength fully determines how the filter interprets and uses the state values stored in the property States. As with all word and fraction length properties, SectionInputFracLength can be any integer, including integers larger than CoeffWordLength, and positive or negative integers. 15 bits is the default value when you create the filter object.

SectionInputWordLength

SOS filters are composed of sections, each one a second-order filter. Filtering data input to the filter involves passing the data through each filter section. SectionInputWordLength specifies the word length applied to data as it enters one filter section from the previous section. Only second-order implementations of direct-form I transposed and direct-form II transposed filters include this property.

By looking at one of the SOS transposed filter structures, such as this one for the transposed direct-form I filter implemented using second-order sections, you see the filter sections at the bottom of the figure.





`SectionInputWordLength` defaults to 16 bits.

SectionOutputAutoScale

Second-order section filters include this property that determines who the filter handles data in the transitions from one section to the next in the filter.

How the filter represents the data passing from one section to the next depends on the property value of `SectionOutputAutoScale`. The representation the filter uses between the filter sections depends on whether the value of `SectionOutputAutoScale` is `true` or `false`.

- `SectionOutputAutoScale` = `true` means the filter chooses the fraction length to maintain the value of the data between sections as close to the output values from the previous section as possible. `true` is the default setting.

- SectionOutputAutoScale = false removes the automatic scaling of the fraction length for the intersection data and exposes the property that controls the coefficient fraction length (SectionOutputFracLength) so you can change it. For example, if the filter is a second-order, direct form FIR filter, setting SectionOutputAutoScale = false exposes the SectionOutputFracLength property that specifies the fraction length applied to data between the sections.

SectionOutputFracLength

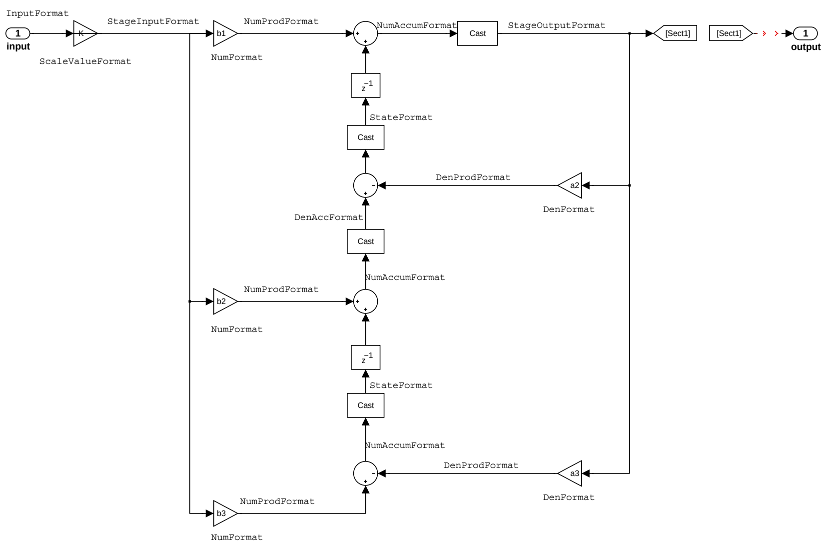
Second-order section filters use quantizers at the output from each section of the filter. The quantizers apply to the output data leaving each filter section. Note that the quantizers for each section are the same. To set the fraction length for interpreting the output values, use the property value in SectionOutputFracLength.

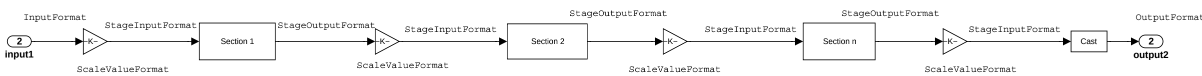
In combination with CoeffWordLength, SectionOutputFracLength determines how the filter interprets and uses the state values stored in the property States. As with all fraction length properties, SectionOutputFracLength can be any integer, including integers larger than CoeffWordLength, and positive or negative integers. 15 bits is the default value when you create the filter object.

SectionOutputWordLength

SOS filters are composed of sections, each one a second-order filter. Filtering data input to the filter involves passing the data through each filter section. SectionOutputWordLength specifies the word length applied to data as it leaves one filter section to go to the next. Only second-order implementations direct-form I transposed and direct-form II transposed filters include this property.

By looking at one of the SOS transposed filter structures, such as this one for the transposed direct-form I filter implemented using second-order sections, you see the filter sections at the bottom of the figure.





`SectionOutputWordLength` defaults to 16 bits.

StateAutoScale

Although all filters use states, some do not allow you to choose whether the filter automatically scales the state values to prevent overruns or bad arithmetic errors. You select either of the following settings:

- `StateAutoScale` = `true` means the filter chooses the fraction length to maintain the value of the states as close to the double-precision values as possible. When you change the word length applied to the states (where allowed by the filter structure), the filter object changes the fraction length to try to accommodate the change. `true` is the default setting.
- `StateAutoScale` = `false` removes the automatic scaling of the fraction length for the states and exposes the property that controls the coefficient fraction length so you can change it. For example, in a direct form I

transposed SOS FIR filter, setting StateAutoScale = false exposes the NumStateFracLength and DenStateFracLength properties that specify the fraction length applied to states.

Each of the following filter structures provides the StateAutoScale property:

- df1t
- df1tsos
- df2t
- df2tsos
- dffirt

Other filter structures do not include this property.

StateFracLength

Filter states stored in the property States have both word length and fraction length. To set the fraction length for interpreting the stored filter object state values, use the property value in StateFracLength.

In combination with CoeffWordLength, StateFracLength fully determines how the filter interprets and uses the state values stored in the property States.

As with all fraction length properties, StateFracLength can be any integer, including integers larger than CoeffWordLength, and positive or negative integers. 15 bits is the default value when you create the filter object.

States

Digital filters are dynamic systems. The behavior of dynamic systems (their response) depends on the input (stimulus) to the system and the current or previous *state* of the system. You can say the system has memory or inertia. All fixed- or floating-point digital filters (as well as analog filters) have states.

Filters use the states to compute the filter output for each input sample, as well using them while filtering in loops to maintain the filter state between loop iterations. In the toolbox we assume zero-valued initial conditions (the dynamic system is at rest) by default when we filter the first input sample. Assuming the states are zero initially does not mean the states are not used; they are, but arithmetically they do not have any effect.

Filter objects store the state values in the property `States`. The number of stored states depends on the filter implementation, since the states represent the delays in the filter implementation.

When you review the display for a filter object with fixed arithmetic, notice that the states return an embedded `fi` object, as you see here.

```
b = ellip(6,3,50,300/500);
hd=dfilt.dffir(b)

hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'double'
               Numerator: [0.0773 0.2938 0.5858 0.7239 0.5858 0.2938 0.0773]
        PersistentMemory: false
                  States: [6x1 double]
     NumSamplesProcessed: 0


hd.arithmetic='fixed'

hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'fixed'
               Numerator: [0.0773 0.2938 0.5858 0.7239 0.5858 0.2938 0.0773]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: 'on'
                  Signed: 'on'

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: 'on'

               RoundMode: 'convergent'
            OverflowMode: 'wrap'

         InheritSettings: 'off'
```

`fi` objects provide fixed-point support for the filters. To learn more about the details about `fi` objects, refer to your Fixed-Point Toolbox documentation.

The property `States` lets you use a `fi` object to define how the filter interprets the filter states. For example, you can create a `fi` object in MATLAB, then assign the object to States, as follows:

```
statefi=fi([],16,12)

statefi =

[]
                DataTypeMode = Fixed-point: binary point scaling
                      Signed = true
                  Wordlength = 16
              Fractionlength = 12
```

This `fi` object does not have a value associated (notice the `[]` input argument to `fi` for the value), and it has word length of 16 bits and fraction length of 12 bit. Now you can apply `statefi` to the `States` property of the filter `hd`.

```
set(hd,'States',statefi);
Warning: The 'States' property will be reset to the value
specified at construction before filtering.
Set the 'PersistentMemory' flag to 'True' to avoid changing this
property value.
hd

hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'fixed'
               Numerator: [0.0773 0.2938 0.5858 0.7239 0.5858
                           0.2938 0.0773]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: 'on'
                  Signed: 'on'

         InputWordLength: 16
         InputFracLength: 15
```

```
 OutputWordLength: 16
       OutputMode: 'AvoidOverflow'

      ProductMode: 'FullPrecision'

        AccumMode: 'KeepMSB'
  AccumWordLength: 40
    CastBeforeSum: 'on'

        RoundMode: 'convergent'
     OverflowMode: 'wrap'
```

StateWordLength

While all filters use states, some do not allow you to directly change the state representation—the word length and fraction lengths—independently. For the others, `StateWordLength` specifies the word length, in bits, the filter uses to represent the states. Filters that do not provide direct state word length control include:

- `df1`
- `dfasymfir`
- `dffir`
- `dfsymfir`

For these structures, the filter derives the state format from the input format you choose for the filter—except for the `df1` IIR filter. In this case, the numerator state format comes from the input format and the denominator state format comes from the output format. All other filter structures provide control of the state format directly.

TapSumFracLength

Direct-form FIR filter objects, both symmetric and antisymmetric, use this property. To set the fraction length for output from the sum operations that involve the filter tap weights, use the property value in `TapSumFracLength`. To enable this property, set the `TapSumMode` to `SpecifyPrecision` in your filter.

As you can see in this code example that creates a fixed-point asymmetric FIR filter, the `TapSumFracLength` property becomes available after you change the `TapSumMode` property value.

```
hd=dfilt.dfasymfir

hd =

         FilterStructure: 'Direct-Form Antisymmetric FIR'
              Arithmetic: 'double'
               Numerator: 1
        PersistentMemory: false
                  States: [0x1 double]
     NumSamplesProcessed: 0

set(hd,'arithmetic','fixed');
hd

hd =

         FilterStructure: 'Direct-Form Antisymmetric FIR'
              Arithmetic: 'fixed'
               Numerator: 1
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

              TapSumMode: 'KeepMSB'
        TapSumWordLength: 17
```

```
           ProductMode: 'FullPrecision'

             AccumMode: 'KeepMSB'
       AccumWordLength: 40

         CastBeforeSum: true
             RoundMode: 'convergent'
          OverflowMode: 'wrap'
```

With the filter now in fixed-point mode, you can change the `TapSumMode` property value to `SpecifyPrecision`, which gives you access to the `TapSumFracLength` property.

```
set(hd,'TapSumMode','SpecifyPrecision');
hd

hd =

       FilterStructure: 'Direct-Form Antisymmetric FIR'
            Arithmetic: 'fixed'
             Numerator: 1
      PersistentMemory: false
                States: [1x1 embedded.fi]
   NumSamplesProcessed: 0

       CoeffWordLength: 16
        CoeffAutoScale: true
                Signed: true

       InputWordLength: 16
       InputFracLength: 15

      OutputWordLength: 16
            OutputMode: 'AvoidOverflow'

            TapSumMode: 'SpecifyPrecision'
      TapSumWordLength: 17
      TapSumFracLength: 15

           ProductMode: 'FullPrecision'
```

```
        AccumMode: 'KeepMSB'
  AccumWordLength: 40

    CastBeforeSum: true
        RoundMode: 'convergent'
     OverflowMode: 'wrap'
```

In combination with TapSumWordLength, TapSumFracLength fully determines how the filter interprets and uses the state values stored in the property States.

As with all fraction length properties, TapSumFracLength can be any integer, including integers larger than TapSumWordLength, and positive or negative integers. 15 bits is the default value when you create the filter object.

TapSumMode

This property, available only after your filter is in fixed-point mode, specifies how the filter outputs the results of summation operations that involve the filter tap weights. Only symmetric (dfilt.dfsymfir) and antisymmetric (dfilt.dfasymfir) FIR filters use this property.

When available, you select from one of the following values:

- FullPrecision—means the filter automatically chooses the word length and fraction length to represent the results of the sum operation so they retain all of the precision provided by the inputs (addends).
- KeepMSB—means you specify the word length for representing tap sum summation results to keep the higher order bits in the data. The filter sets the fraction length to discard the LSBs from the sum operation. This is the default property value.
- KeepLSB—means you specify the word length for representing tap sum summation results to keep the lower order bits in the data. The filter sets the fraction length to discard the MSBs from the sum operation. Compare to the KeepMSB option.
- SpecifyPrecision—means you specify the word and fraction lengths to apply to data output from the tap sum operations.

TapSumWordLength

Specifies the word length the filter uses to represent the output from tap sum operations. The default value is 17 bits. Only `dfasymfir` and `dfsymfir` filters include this property.

Adaptive Filter Properties

The following table summarizes the adaptive filter properties and provides a brief description of each. Full descriptions of each property, in alphabetical order, follow the table.

Property	Description
`Algorithm`	Reports the algorithm the object uses for adaptation. When you construct your adaptive filter object, this property is set automatically by the constructor, such as `adaptfilt.nlms` creating an adaptive filter that uses the normalized LMS algorithm. You cannot change the value—it is read only.
`AvgFactor`	Averaging factor used to compute the exponentially-windowed estimates of the powers in the transformed signal bins for the coefficient updates. `AvgFactor` should lie between zero and one. For default filter objects, `AvgFactor` equals (1 - `step`). `lambda` is the input argument that represents `AvgFactor`
`BkwdPredErrorPower`	Returns the minimum mean-squared prediction error. Refer to [12] in the bibliography for details about linear prediction.
`BkwdPrediction`	Returns the predicted samples generated during adaptation.Refer to [12] in the bibliography for details about linear prediction.

Property (Continued)	Description
`Blocklength`	Block length for the coefficient updates. This must be a positive integer such that (`l/blocklength`) is also an integer. For faster execution, `blocklength` should be a power of two. `blocklength` defaults to two.
`Coefficients`	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.
`ConversionFactor`	Conversion factor defaults to the matrix [1 -1] that specifies soft-constrained initialization. This is the `gamma` input argument for some of the fast transversal algorithms.
`Delay`	Update delay given in time samples. This scalar should be a positive integer—negative delays do not work. `delay` defaults to 1 for most algorithms.
`DesiredSignalStates`	Desired signal states of the adaptive filter. `dstates` defaults to a zero vector with length equal to (`blocklen` - 1) or (`swblocklen` - 1) depending on the algorithm.
`EpsilonStates`	Vector of the epsilon values of the adaptive filter. `EpsilonStates` defaults to a vector of zeros with (`projectord` - 1) elements.

Property (Continued)	Description
ErrorStates	Vector of the adaptive filter error states. ErrorStates defaults to a zero vector with length equal to (projectord - 1).
FFTCoefficients	Stores the discrete Fourier transform of the filter coefficients in coeffs.
FFTStates	Stores the states of the FFT of the filter coefficients during adaptation.
FilteredInputStates	Vector of filtered input states with length equal to l - 1.
FilterLength	Contains the length of the filter. Note that this is not the filter order. Filter length is 1 greater than filter order. Thus a filter with length equal to 10 has filter order equal to 9.
ForgettingFactor	Determines how the RLS adaptive filter uses past data in each iteration. You use the forgetting factor to specify whether old data carries the same weight in the algorithm as more recent data.
FwdPredErrorPower	Returns the minimum mean-squared prediction error in the forward direction. Refer to [12] in the bibliography for details about linear prediction.
FwdPrediction	Contains the predicted values for samples during adaptation. Compare these to the actual samples to get the error and power.

Property (Continued)	Description
`InitFactor`	Soft-constrained initialization factor. This scalar should be positive and sufficiently large to prevent an excessive number of Kalman gain rescues. Called `delta` as an input argument, this defaults to one.
`InvCov`	Upper-triangular Cholesky (square root) factor of the input covariance matrix. Initialize this matrix with a positive definite upper triangular matrix. Dimensions are `l`-by-`l`, where `l` is the filter length.
`KalmanGain`	Empty when you construct the object, this gets populated after you run the filter.
`KalmanGainStates`	Contains the states of the Kalman gain updates during adaptation.
`Leakage`	Contains the setting for leakage in the adaptive filter algorithm. Using a leakage factor that is not 1 forces the weights to adapt even when they have found the minimum error solution. Forcing the adaptation can improve the numerical performance of the LMS algorithm.
`NumSamplesProcessed`	Reports the number of samples actually processed by the filter.
`OffsetCov`	Contains the offset covariance matrix.

Property (Continued)	Description
`Offset`	Specifies an optional offset for the denominator of the step size normalization term. You must specify offset to be a scalar greater than or equal to zero. Nonzero offsets can help avoid a divide-by-near-zero condition that causes errors.
`Power`	A vector of 2*`l` elements, each initialized with the value `delta` from the input arguments. As you filter data, `Power` gets updated by the filter process.
`ProjectionOrder`	Projection order of the affine projection algorithm. `projectord` defines the size of the input signal covariance matrix and defaults to two.
`ReflectionCoeffs`	Coefficients determined for the reflection portion of the filter during adaptation.
`ReflectionCoeffsStep`	Size of the steps used to determine the reflection coefficients.
`PersistentMemory`	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states and coefficients from previous filtering runs.

Property (Continued)	Description
SecondaryPathCoeffs	A vector that contains the coefficient values of your secondary path from the output actuator to the error sensor.
SecondaryPathEstimate	An estimate of the secondary path filter model.
SecondaryPathStates	The states of the secondary path filter, the unknown system.
SqrtCov	Upper-triangular Cholesky (square root) factor of the input covariance matrix. Initialize this matrix with a positive definite upper triangular matrix.
SqrtlnvCov	Square root of the inverse of the sliding window input signal covariance matrix. This square matrix should be full-ranked.
States	Vector of the adaptive filter states. states defaults to a vector of zeros whose length depends on the chosen algorithm. Usually the length is a function of the filter length l and another input argument to the filter object, such as projectord.

Property (Continued)	Description
StepSize	Reports the size of the step taken between iterations of the adaptive filter process. Each adaptfilt object has a default value that best meets the needs of the algorithm.
SwBlockLength	Block length of the sliding window. This integer must be at least as large as the filter length. swblocklen defaults to 16.

Like dfilt objects, adaptfilt objects have properties that govern their behavior and store some of the results of filtering operations. The following pages list, in alphabetical order, the name of every property associated with adaptfilt objects. Note that not all adaptfilt objects have all of these properties. To view the properties of a particular adaptive filter, such as an adaptfilt.bap filter, use get with the object handle, like this:

```
ha = adaptfilt.bap(32,0.5,4,1.0);
get(ha)
        PersistentMemory: false
     NumSamplesProcessed: 0
               Algorithm: 'Block Affine Projection FIR Adaptive Filter'
            FilterLength: 32
            Coefficients: [1x32 double]
                  States: [35x1 double]
                StepSize: 0.5000
         ProjectionOrder: 4
               OffsetCov: [4x4 double]
```

get shows you the properties for ha and the values for the properties. Entering the object handle returns the same values and properties without the formatting of the list and the more familiar property names.

Algorithm

Reports the algorithm the object uses for adaptation. When you construct you adaptive filter object, this property is set automatically. You cannot change the value—it is read only.

AvgFactor

Averaging factor used to compute the exponentially-windowed estimates of the powers in the transformed signal bins for the coefficient updates. AvgFactor should lie between zero and one. For default filter objects, AvgFactor equals (1 - step). lambda is the input argument that represent AvgFactor

BkwdPredErrorPower

BkwdPrediction

When you use an adaptive filter that does backward prediction, such as adaptfilt.ftf, one property of the filter contains the backward prediction coefficients for the adapted filter. With these coefficient, the forward coefficients, and the system under test, you have the full set of knowledge of how the adaptation occurred. Two values stored in properties compose the `BkwdPrediction` property:

- Coefficients, which contains the coefficients of the system under test, as determined using backward predictions process.
- Error, which is the difference between the filter coefficients determined by backward prediction and the actual coefficients of the sample filter. In this example, `adaptfilt.ftf` identifies the coefficients of an unknown FIR system.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
N  = 31;               % Adaptive filter order
lam = 0.99;            % RLS forgetting factor
del = 0.1;             % Soft-constrained initialization factor
ha = adaptfilt.ftf(32,lam,del);
[y,e] = filter(ha,x,d);

ha

ha =

               Algorithm: 'Fast Transversal Least-Squares Adaptive Filter'
            FilterLength: 32
            Coefficients: [1x32 double]
                  States: [31x1 double]
        ForgettingFactor: 0.9900
              InitFactor: 0.1000
           FwdPrediction: [1x1 struct]
          BkwdPrediction: [1x1 struct]
              KalmanGain: [32x1 double]
        ConversionFactor: 0.7338
        KalmanGainStates: [32x1 double]
        PersistentMemory: false
     NumSamplesProcessed: 500
ha.coefficients

ans =

  Columns 1 through 8

   -0.0055    0.0048    0.0045    0.0146   -0.0009    0.0002   -0.0019    0.0008

  Columns 9 through 16

   -0.0142   -0.0226    0.0234    0.0421   -0.0571   -0.0807    0.1434    0.4620

  Columns 17 through 24

    0.4564    0.1532   -0.0879   -0.0501    0.0331    0.0361   -0.0266   -0.0220

  Columns 25 through 32
```

```
    0.0231    0.0026   -0.0063   -0.0079    0.0032    0.0082    0.0033    0.0065

ha.bkwdprediction

ans =

    Coeffs: [1x32 double]
     Error: 82.3394

>> ha.bkwdprediction.coeffs

ans =

  Columns 1 through 8

    0.0067    0.0186    0.1114   -0.0150   -0.0239   -0.0610   -0.1120   -0.1026

  Columns 9 through 16

    0.0093   -0.0399   -0.0045    0.0622    0.0997    0.0778    0.0646   -0.0564

  Columns 17 through 24

    0.0775    0.0814    0.0057    0.0078    0.1271   -0.0576    0.0037   -0.0200

  Columns 25 through 32

   -0.0246    0.0180   -0.0033    0.1222    0.0302   -0.0197   -0.1162    0.0285
```

Blocklength

Block length for the coefficient updates. This must be a positive integer such that (`l/blocklen`) is also an integer. For faster execution, `blocklen` should be a power of two. `blocklen` defaults to two.

Coefficients

Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.

ConversionFactor

Conversion factor defaults to the matrix [1 -1] that specifies soft-constrained initialization. This is the `gamma` input argument for some of the fast transversal algorithms.

Delay

Update delay given in time samples. This scalar should be a positive integer—negative delays do not work. `delay` defaults to 1 for most algorithms.

DesiredSignalStates

Desired signal states of the adaptive filter. `dstates` defaults to a zero vector with length equal to (`blocklen` - 1) or (`swblocklen` - 1) depending on the algorithm.

EpsilonStates

Vector of the epsilon values of the adaptive filter. `EpsilonStates` defaults to a vector of zeros with (`projectord` - 1) elements.

ErrorStates

Vector of the adaptive filter error states. `ErrorStates` defaults to a zero vector with length equal to (`projectord` - 1).

FFTCoefficients

Stores the discrete Fourier transform of the filter coefficients in `coeffs`.

FFTStates

Stores the states of the FFT of the filter coefficients during adaptation.

FilteredInputStates

Vector of filtered input states with length equal to `l` - 1.

FilterLength

Contains the length of the filter. Note that this is not the filter order. Filter length is 1 greater than filter order. Thus a filter with length equal to 10 has filter order equal to 9.

ForgettingFactor

Determines how the RLS adaptive filter uses past data in each iteration. You use the forgetting factor to specify whether old data carries the same weight in the algorithm as more recent data.

This is a scalar and should lie in the range (0, 1]. It defaults to 1. Setting `forgetting factor = 1` denotes infinite memory while adapting to find the new filter. Note that this is the `lambda` input argument.

FwdPredErrorPower

Returns the minimum mean-squared prediction error in the forward direction. Refer to [12] in the bibliography for details about linear prediction.

FwdPrediction

Contains the predicted values for samples during adaptation. Compare these to the actual samples to get the error and power.

InitFactor

Returns the soft-constrained initialization factor. This scalar should be positive and sufficiently large to prevent an excessive number of Kalman gain rescues. `delta` defaults to one.

InvCov

Upper-triangular Cholesky (square root) factor of the input covariance matrix. Initialize this matrix with a positive definite upper triangular matrix. Dimensions are `l`-by-`l`, where `l` is the filter length.

KalmanGain

Empty when you construct the object, this gets populated after you run the filter.

KalmanGainStates

Contains the states of the Kalman gain updates during adaptation.

Leakage

Contains the setting for leakage in the adaptive filter algorithm. Using a leakage factor that is not 1 forces the weights to adapt even when they have found the minimum error solution. Forcing the adaptation can improve the numerical performance of the LMS algorithm.

NumSamplesProcessed

Reports the number of input samples processed by your `adaptfilt` object when you filter a data set. When you set the `PersistentMemory` property to `false`, `NumSamplesProcessed` reports the number of processed input samples for your most recent filtering operation with the object.

With PersistentMemory set to true, NumSamplesProcessed accumulates the total number of samples processed for all preceding filtering operations.

OffsetCov

Contains the offset covariance matrix.

Offset

Specifies an optional offset for the denominator of the step size normalization term. You must specify offset to be a scalar greater than or equal to zero. Nonzero offsets can help avoid a divide-by-near-zero condition that causes errors.

Use this to avoid dividing by zero or by very small numbers when input signal amplitude becomes very small, or dividing by very small numbers when any of the FFT input signal powers become very small. offset defaults to one.

Power

A vector of 2*l elements, each initialized with the value delta from the input arguments. As you filter data, Power gets updated by the filter process.

ProjectionOrder

Projection order of the affine projection algorithm. projectord defines the size of the input signal covariance matrix and defaults to two.

ReflectionCoeffs

For adaptive filters that use reflection coefficients, this property stores them.

ReflectionCoeffsStep

As the adaptive filter changes coefficient values during adaptation, the step size used between runs is stored here.

PersistentMemory

Determines whether the filter states and coefficients get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter.

`PersistentMemory` returns to zero any property value that the filter changes during processing. Property values that the filter does not change are not affected. Defaults to `false`.

SecondaryPathCoeffs

A vector that contains the coefficient values of your secondary path from the output actuator to the error sensor.

SecondaryPathEstimate

An estimate of the secondary path filter model.

SecondaryPathStates

The states of the secondary path filter, the unknown system.

SqrtCov

Upper-triangular Cholesky (square root) factor of the input covariance matrix. Initialize this matrix with a positive definite upper triangular matrix.

SqrtInvCov

Square root of the inverse of the sliding window input signal covariance matrix. This square matrix should be full-ranked.

States

Vector of the adaptive filter states. `states` defaults to a vector of zeros whose length depends on the chosen algorithm. Usually the length is a function of the filter length `l` and another input argument to the filter object, such as projectord.

StepSize

Reports the size of the step taken between iterations of the adaptive filter process. Each `adaptfilt` object has a default value that best meets the needs of the algorithm.

SwBlockLength

Block length of the sliding window. This integer must be at least as large as the filter length. `swblocklength` defaults to 16.

Multirate Filter Properties

The following table summarizes the multirate filter properties and provides a brief description of each. Full descriptions of each property follow in the next section.

Name	Values	Default	Description
`BlockLength`	Positive integers	100	Length of each block of data input to the FFT used in the filtering. `fftfirinterp` multirate filters include this property.
`DecimationFactor`	Any positive integer	2	Amount to reduce the input sampling rate.
`DifferentialDelay`	Any integer	1	Sets the differential delay for the filter. Usually a value of one or two is appropriate.
`FilterInternals`	`FullPrecision`, `SpecifyPrecision`	`FullPrecision`	Controls whether the filter sets the output word and fraction lengths, and the accumulator word and fraction lengths automatically to maintain the best precision results during filtering. The default value, `FullPrecision`, sets automatic word and fraction length determination by the filter. `SpecifyPrecision` exposes the output and accumulator related properties so you can set your own word and fraction lengths for them.

Name	Values	Default	Description
FilterStructure	mfilt structure string	None	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output. You cannot set this property—it is always read only and results from your choice of mfilt object.
InputBitWidth	8, 16, 32	16	Word length of the input samples.
InputOffset	Integers	0	Contains the number of input data samples processed without generating an output sample.
InterpolationFactor	Positive integers	2	Interpolation factor for the filter. l specifies the amount to increase the input sampling rate.
NumberOfSections	Any positive integer	2	Number of sections used in the decimator, or in the comb and integrator portions of CIC filters.
Numerator	Array of double values	No default values	Vector containing the coefficients of the FIR lowpass filter used for interpolation.
OutputBitWidth	1 to 32	16	Word length of the output samples.

Name	Values	Default	Description
OverflowMode	saturate, [wrap]	wrap	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
PolyphaseAccum	Values depend on filter type. Either double,single, or fixed-point object	0	Stores the value remaining in the accumulator after the filter processes the last input sample. The stored value for PolyphaseAccum affects the next output when PersistentMemory is true and InputOffset is not equal to 0. Always provides full precision values. Compare the AccumWordLength and AccumFracLength.

Name	Values	Default	Description
PersistentMemory	false or true	false	Determines whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected.
RateChangeFactors	[l,m]	[2,3] or [3,2]	Reports the decimation (m) and interpolation (l) factors for the filter object. Combining these factors results in the final rate change for the signal. The default changes depending on whether the filter decimates or interpolates.
States	Any m+1-by-n matrix of double values	2-by-2 matrix, int32	Stored conditions for the filter, including values for the integrator and comb sections. n is the number of filter sections and m is the differential delay. Stored in a filtstates object.

Name	Values	Default	Description
SectionWordLengthMode	MinWordLengths or SpecifyWordLengths	MinWordLength	Determines whether the filter object sets the section word lengths or you provide the word lengths explicitly. By default, the filter uses the input and output word lengths in the command to determine the proper word lengths for each section, according to the information in [1]. When you choose SpecifyWordLengths, you provide the word length for each section. In addition, choosing SpecifyWordLengths exposes the SectionWordLengths property for you to modify as needed.
SpecifyWordLengths	Vector of integers	[16 16 16 16] bits	
WordLengthPerSection	Any integer or a vector of length 2*n	16	Defines the word length used in each section while accumulating the data in the integrator sections or while subtracting the data during the comb sections (using 'wrap' arithmetic). Enter WordLengthPerSection as a scalar or vector of length 2*n, where n is the number of sections. When WordLengthPerSection is a scalar, the scalar value is applied to each filter section. The default is 16 for each section in the decimator.

The following sections provide details about the properties that govern the way multirate filter work. Creating any multirate filter object puts in place a number of these properties. On the next pages, we list the `mfilt` object properties in alphabetical order.

BitsPerSection

Any integer or a vector of length 2*`n`.

Defines the bits per section used while accumulating the data in the integrator sections or while subtracting the data during the comb sections (using `wrap` arithmetic). Enter `bps` as a scalar or vector of length 2*`n`, where `n` is the number of sections. When `bps` is a scalar, the scalar value is applied to each filter section. The default is 16 for each section in the decimator.

BlockLength

Length of each block of input data used in the filtering.

`mfilt.fftfirinterp` objects process data in blocks whose length is determined by the value you set for the BlockLength property. By default the property value is 100. When you set the `BlockLength` value, try choosing a value so that [`BlockLength` + `length`(filter order)] is a power of two.

Larger block lengths generally reduce the computation time.

DecimationFactor

Decimation factor for the filter. `m` specifies the amount to reduce the sampling rate of the input signal. It must be an integer. You can enter any integer value. The default value is 2.

DifferentialDelay

Sets the differential delay for the filter. Usually a value of one or two is appropriate. While you can set any value, the default is one and the maximum is usually two.

FilterInternals

Similar to the FilterInternals pane in FDATool, this property controls whether the filter sets the output word and fraction lengths automatically, and the accumulator word and fraction lengths automatically as well, to maintain the best precision results during filtering. The default value, `FullPrecision`, sets automatic word and fraction length determination by the filter. Setting `FilterInternals` to `SpecifyPrecision` exposes the output and accumulator related properties so you can set your own word and fraction lengths for them.

FilterStructure

Reports the type of filter object, such as a decimator or fractional integrator. You cannot set this property—it is always read only and results from your choice of `mfilt` object. Because of the length of the names of multirate filters, `FilterStructure` often returns a vector specification for the string. For example, when you use `mfilt.firfracinterp` to design a filter, `FilterStructure` returns as `[1x49 char]`.

```
hm=mfilt.firfracinterp

hm =

         FilterStructure: [1x49 char]
               Numerator: [1x72 double]
       RateChangeFactors: [3 2]
        PersistentMemory: false
                  States: [24x1 double]
     NumSamplesProcessed: 0
```

InputBitWidth

Word length of the input samples. The default is 16 bits and the accepted values are 8, 16, and 32 bits. CIC filters use this property.

InputOffset

When you decimate signals whose length is not a multiple of the decimation factor M, the last samples—(nM +1) to [(n+1)(M) -1], where n is an integer—are processed and used to track where the filter stopped processing input data and when to expect the next output sample. If you think of the filtering process as generating an output for a block of input data, `InputOffset` contains a count of the number of samples in the last incomplete block of input data.

Note `InputOffset` applies only when you set `PersistentMemory` to `true`. Otherwise, `InputOffset` is not available for you to use.

Two different cases can arise when you decimate a signal:

1. The input signal is a multiple of the filter decimation factor. In this case, the filter processes the input samples and generates output samples for all inputs as determined by the decimation factor. For example, processing 99 input samples with a filter that decimates by three returns 33 output samples.

2. The input signal is not a multiple of the decimation factor. When this occurs, the filter processes all of the input samples, generates output samples as determined by the decimation factor, and has one or more input samples that were processed but did not generate an output sample.

   For example, when you filter 100 input samples with a filter which has decimation factor of 3, you get 33 output samples, and 1 sample that did not generate an output. In this case, `InputOffset` stores the value 1 after the filter run.

   `InputOffset` equal to 1 indicates that, if you divide your input signal into blocks of data with length equal to your filter decimation factor, the filter processed one sample from a new (incomplete) block of data. Subsequent inputs to the filter are concatenated with this single sample to form the next block of length m.

One way to define the value stored in `InputOffset` is

```
InputOffset = mod(length(nx),m)
```

where `nx` is the number of input samples in the data set and `m` is the decimation factor.

Storing `InputOffset` in the filter allows you to stop filtering a signal at any point and start over from there, provided that the `PersistentMemory` property is set to `true`. Being able to resume filtering after stopping a signal lets you break large data sets in to smalller pieces for filtering. With `PersistentMemory` set to `true` and the `InputOffset` property in the filter, breaking a signal into sections of arbitrary length and filtering the sections is equivalent to filtering the entire signal at once.

```
xtot=[x,x];
ytot=filter(hm1,xtot)
ytot =
```

```
        0   -0.0003    0.0005   -0.0014    0.0028   -0.0054    0.0092
reset(hm1);  % Clear history of the filter
hm1.PersistentMemory='true';
ysec=[filter(hm1,x) filter(hm1,x)]

ysec =

        0   -0.0003    0.0005   -0.0014    0.0028   -0.0054    0.0092
```

This test verifies that ysec (the signal filtered by sections) is equal to ytot (the entire signal filtered at once).

InterpolationFactor

Amount to increase the sampling rate. Interpolation factor for the filter. It specifies the amount to increase the input sampling rate. It must be an integer. Two is the default value. You may use any positive value.

NumberOfSections

Number of sections used in the multirate filter. By default multirate filters use two sections, but any positive integer works.

OutputBitWidth

Word length of the output samples. While 16 bits is the default value, you can enter any integer from 1 to 32 bits. Compare to InputBitWidth.

OverflowMode

The OverflowMode property is specified as one of the following two strings indicating how to respond to overflows in fixed-point arithmetic:

- 'saturate'—saturate overflows.

  When the values of data to be quantized lie outside of the range of the largest and smallest representable numbers (as specified by the applicable word length and fraction length properties), these values are quantized to the value of either the largest or smallest representable value, depending on which is closest.

- 'wrap'—wrap all overflows to the range of representable values.

  When the values of data to be quantized lie outside of the range of the largest and smallest representable numbers (as specified by the data format properties), these values are wrapped back into that range using modular arithmetic relative to the smallest representable number. You can learn more about modular arithmetic in the Fixed-Point Toolbox.

These rules apply to the OverflowMode property.

- Applies to the accumulator and output data only.
- Does not apply to coefficients or input data. These always saturate the results.
- Does not apply to products. Products maintain full precision at all times. Your filters do not lose precision in the products.

Default value: 'saturate'

Note Numbers in floating-point filters that extend beyond the dynamic range overflow to ±inf.

PolyphaseAccum

The idea behind PolyphaseAccum and AccumWordLength/AccumFracLength is to distinguish between the adders that always work in full precision (PolyphaseAccum) from the others [the adders that are controlled by the user (through AccumWordLength and AccumFracLength) and that may introduce quantization effects when you set property FilterInternals to SpecifyPrecision].

Given a product format determined by the input word and fraction lengths, and the coefficients word and fraction lengths, doing full precision accumulation means allowing enough guard bits to avoid overflows and underflows.

Property PolyphaseAccum stores the value that was in the accumulator the last time your filter ran out of input samples to process. The default value for PolyphaseAccum affects the next output only if PersistentMemory is true and InputOffset is not equal to 0.

PolyphaseAccum stores data in the format for the filter arithmetic. Double-precision filters store doubles in PolyphaseAccum. Single-precision filter store singles in PolyphaseAccum. Fixed-point filters store fi objects in PolyphaseAccum.

PersistentMemory

Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected.

Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter object. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to true—the filter retains memory about filtering operations from one to the next. Maintaining memory lets you filter large data sets as collections of smaller subsets and get the same result.

```
xtot=[x,x];
ytot=filter(hm1,xtot)
ytot =

       0   -0.0003    0.0005   -0.0014    0.0028   -0.0054    0.0092
reset(hm1);  % Clear history of the filter
hm1.PersistentMemory='true';
ysec=[filter(hm1,x) filter(hm1,x)]

ysec =

       0   -0.0003    0.0005   -0.0014    0.0028   -0.0054    0.0092
```

This test verifies that ysec (the signal filtered by sections) is equal to ytot (the entire signal filtered at once).

RateChangeFactors

Reports the decimation (m) and interpolation (l) factors for the filter object when you create fractional integrators and decimators, although m and l are used as arguments to both decimators and integrators, applying the same meaning. Combining these factors as input arguments to the fractional decimator or integrator results in the final rate change for the signal.

For decimating filters, the default is [2,3]. For integrators, [3,2].

States

Stored conditions for the filter, including values for the integrator and comb sections. `m` is the differential delay and `n` is the number of sections in the filter.

About the States of Multirate Filters

In the `states` property you find the states for both the integrator and comb portions of the filter, stored in a `filtstates` object. `states` is a matrix of dimensions `m+1`-by-`n`, with the states in CIC filters apportioned as follows:

- States for the integrator portion of the filter are stored in the first row of the state matrix.
- States for the comb portion fill the remaining rows in the state matrix.

In the state matrix, state values are specified and stored in `double` format.

States stores conditions for the delays between each interpolator phase, the filter states, and the states at the output of each phase in the filter, including values for the interpolator and comb states.

The number of states is (`lh`-1)*`m`+(l-1)*(`lo`+`mo`) where `lh` is the length of each subfilter, and `l` and `m` are the interpolation and decimation factors. `lo` and `mo`, the input and output delays between each interpolation phase, are integers from Euclid's theorem such that `lo`*l-`mo`*`m` = -1 (refer to the reference for more details). Use `euclidfactors` to get `lo` and `mo` for an `mfilt.firfracdecim` object.

`States` defaults to a vector of zeros that has length equal to `nstates(hm)`

9

Function Reference

Functions—Categorical List

With the Filter Design (FD) Toolbox, you can create, apply, and analyze quantized filters, adaptive filters, and multirate filters.

This chapter contains brief descriptions of all FD Toolbox functions grouped by subject area, and continues with the detailed reference entries listed alphabetically.

In addition to the constructors for discrete time filters, you find listed here the constructors for adaptive filter and multirate filter preferences objects, which look like functions but return objects when you use them.

The following tables list the functions and constructors in the FD Toolbox, separated by application—adaptive filter, discrete-time filter, multirate filter, and filter analysis. In many instances, you can apply a function to more than one application; those functions are called *overloaded* functions and they appear in more than one table.

Adaptive Filter Constructors

For adaptive filter (`adaptfilt`) objects, the *`algorithm`* string determines which adaptive filter algorithm your `adaptfilt` object implements. Each available algorithm entry appears in one of the following tables along with a brief description of the algorithm. Click on the algorithm in the first column to get more information about the associated adaptive filter technique.

- "Least Mean Squares (LMS) Based FIR Adaptive Filters" on page 9-3
- "Recursive Least Squares (RLS) Based FIR Adaptive Filters" on page 9-4
- "Affine Projection (AP) FIR Adaptive Filters" on page 9-5
- "FIR Adaptive Filters in the Frequency Domain (FD)" on page 9-6
- "Lattice Based (L) FIR Adaptive Filters" on page 9-7

Least Mean Squares (LMS) Based FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.adjlms`	Use the adjoint LMS FIR adaptive filter algorithm
`adaptfilt.blms`	Use the block LMS FIR adaptive filter algorithm
`adaptfilt.blmsfft`	Use the FFT-based block LMS FIR adaptive filter algorithm
`adaptfilt.dlms`	Use the delayed LMS FIR adaptive filter algorithm
`adaptfilt.filtxlms`	Use the filtered-x LMS FIR adaptive filter algorithm
`adaptfilt.lms`	Use the LMS FIR adaptive filter algorithm
`adaptfilt.nlms`	Use the normalized LMS FIR adaptive filter algorithm
`adaptfilt.sd`	Use the sign-data LMS FIR adaptive filter algorithm
`adaptfilt.se`	Use the sign-error LMS FIR adaptive filter algorithm
`adaptfilt.ss`	Use the sign-sign LMS FIR adaptive filter algorithm

For further information about an adapting algorithm, refer to the reference page for the algorithm.

Recursive Least Squares (RLS) Based FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.ftf`	Use the fast transversal least squares adaptation algorithm
`adaptfilt.qrdrls`	Use the QR-decomposition RLS adaptation algorithm
`adaptfilt.hrls`	Use the householder RLS adaptation algorithm
`adaptfilt.hswrls`	Use the householder sliding window SWRLS adaptation algorithm
`adaptfilt.rls`	Use the recursive least squares (RLS) adaptation algorithm
`adaptfilt.swrls`	Use the sliding window (SW) RLS adaptation algorithm
`adaptfilt.swftf`	Use the sliding window FTF adaptation algorithm

For more complete information about an adapting algorithm, refer to the reference page for the algorithm.

Affine Projection (AP) FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.ap`	Use the affine projection (AP) algorithm that uses direct matrix inversion
`adaptfilt.apru`	Use the affine projection (AP) algorithm that uses recursive matrix updating
`adaptfilt.bap`	Use the block affine projection (AP) adaptation algorithm

To find more information about an adapting algorithm, refer to the reference page for the algorithm.

FIR Adaptive Filters in the Frequency Domain (FD)

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
adaptfilt.fdaf	Use the frequency domain (FD) adaptation algorithm
adaptfilt.pbfdaf	Use the partition block version of the FDAF algorithm
adaptfilt.pbufdaf	Use the partition block unconstrained version of the FDAF algorithm
adaptfilt.tdafdct	Use the transform domain adaptation algorithm using DCT
adaptfilt.tdafdft	Use the transform domain adaptation algorithm using DFT
adaptfilt.ufdaf	Use the unconstrained FDAF algorithm for adaptation

For more information about an adapting algorithm, refer to the reference page for the algorithm.

Lattice Based (L) FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.gal`	Use the gradient adaptive lattice filter adaptation algorithm
`adaptfilt.lsl`	Use the least squares lattice adaptation algorithm
`adaptfilt.qrdlsl`	Use the QR decomposition LSL adaptation algorithm

For more information about an adapting algorithm, refer to the reference page for the algorithm.

Discrete-Time Filter Constructors

For discrete-time filter (`dfilt`) objects, the *`structure`* string determines which filter structure your `dfilt` object implements. Each available structure entry appears in the following table along with a brief description of the structure. Click on the structure in the first column to get more information about the associated filter implementation.

Filter Structure	Description
`dfilt.cascade`	Construct and cascade filter object
`dfilt.df1`	Construct a direct-form I IIR filter object
`dfilt.df1sos`	Construct a direct-form I IIR filter object that uses second-order sections
`dfilt.df1t`	Construct a transposed direct-form I IIR filter object
`dfilt.df1tsos`	Construct a transposed direct-form I IIR filter object that uses second-order sections
`dfilt.df2`	Construct a direct-form II IIR filter object

Filter Structure	Description
`dfilt.df2sos`	Construct a direct-form II IIR filter object that uses second-order sections
`dfilt.df2t`	Construct a transposed direct-form II IIR filter object
`dfilt.df2tsos`	Construct a transposed direct-form II IIR filter object that uses second-order sections
`dfilt.dfasymfir`	Construct an antisymmetric FIR filter object
`dfilt.dffir`	Construct a direct-form FIR filter object
`dfilt.dffirt`	Construct a transposed direct-form FIR filter object
`dfilt.dfsymfir`	Construct a symmetric FIR filter object
`dfilt.latticeallpass`	Construct an allpass lattice filter object
`dfilt.latticear`	Construct an autoregressive lattice filter object
`dfilt.latticearma`	Construct a moving average, autoregressive lattice filter object
`dfilt.latticemamax`	Construct a moving average, maximum phase lattice filter object
`dfilt.latticemamin`	Construct a moving average, minimum phase lattice filter object
`dfilt.parallel`	Construct a parallel filter object
`dfilt.scalar`	Construct a scalar FIR filter object

Discrete-Time Filter Design Objects—Response Types

Rather than choosing the filter structure when you create a discrete-time filter, you have the option of selecting the filter specifications and magnitude response to use to describe your filter, and then selecting the design method to implement your filter. The next two table list the kind of filter responses you can choose from and the methods the toolbox provides to implement the filters.

Magnitude Response Shape	Description of the Design Object
`fdesign.bandpass`	Design a bandpass filter
`fdesign.bandstop`	Design a bandstop filter
`fdesign.decim`	Design a decimating filter
`fdesign.halfband`	Design a halfband filter
`fdesign.highpass`	Design a highpass filter
`fdesign.interp`	Design an interpolating filter
`fdesign.lowpass`	Design a lowpass filter
`fdesign.nyquist`	Design a Nyquist filter
`fdesign.src`	Design a sample rate change filter

Filter Design Objects—Methods

Design Method	Description
`butter`	Use the Butterworth design method to construct a filter from a design object
`cheby1`	Use the Chebyshev I design method to construct a filter from a design object
`cheby2`	Use the Chebyshev II design method to construct a filter from a design object
`ellip`	Use the elliptic design method to construct a filter from a design object
`equiripple`	Use the equiripple design method to construct a filter from a design object

Design Method	Description
`firls`	Use a least-squares technique to construct a filter from a design object
`kaiserwin`	Use the Kaiser window design method to construct a filter from a design object
`window`	Use the window design method to construct a filter from a design object

Multirate Filter Constructors

Filter Design Toolbox includes functions that create multirate filtering objects, as shown in this table.

Multirate Filter Object	Filter Object Description
`mfilt.cicdecim`	Construct a fixed-point cascaded integrator-comb decimation filter object
`mfilt.cicinterp`	Construct a fixed-point cascaded integrator-comb interpolation filter object
`mfilt.fftfirinterp`	Construct an overlap-add FIR polyphase interpolation filter object
`mfilt.firdecim`	Construct a direct-form FIR polyphase decimation filter object
`mfilt.firfracdecim`	Construct a direct-form FIR polyphase fractional decimation filter object
`mfilt.firfracinterp`	Construct a direct-form FIR polyphase fractional interpolation filter object
`mfilt.firinterp`	Construct a direct-form FIR polyphase interpolation filter object
`mfilt.firsrc`	Construct a direct-form FIR polyphase sample rate conversion filter object
`mfilt.firtdecim`	Construct a direct-form, transposed FIR polyphase decimation filter object
`mfilt.holdinterp`	Construct an FIR interpolation filter object that uses “hold” interpolation between input samples
`mfilt.linearinterp`	Construct an FIR linear interpolation filter object that applies linear interpolation between input samples

For more information about a multirate filter object, refer to the reference page for the object.

Methods For Analyzing Filters

The following methods support the filter objects—either adaptive filters (`adaptfilt` objects), discrete-time filters (`dfilt` objects), or multirate filters (`mfilt` objects). Many of the functions support more than one type of object. In addition, existing filter analysis methods such as `freqz` work with the objects just as they do for most filter objects.

Method Name	Supported Objects	Description
`block`	Multirate (some)	Generate a Signal Processing Blockset block that duplicates the filter object. Works only when Signal Processing Blockset is installed. Some multirate objects cannot be modelled as blocks since the blockset does not provide blocks for all the multirate filters in the toolbox.
`coefficients`	Multirate	Return the multirate filter objects coefficients.
`cumsec`	Discrete-time filters	Return a vector of the cumulative second-order sections for an SOS filter
`denormalize`	Discrete-time filters	Restore a filter after a `normalize` operation.
`euclidfactors`	Multirate	Use Euclid's theorem to determine the integer factors for an `mfilt` object.
`filter`	All filters	Apply a filter to a signal.
`filtmsb`	Multirate filters	Return the most significant bit of a CIC filter.

Method Name	Supported Objects	Description
`firtype`	Multirate filters	Return the type (1 to 4) of a linear phase FIR filter.
`freqz`	All filters	Plot the frequency response of a filter.
`grpdelay`	All filters	Plot the group delay for a filter.
`impz`	All filters	Plot the impulse response for a filter.
`isfir`	All filters	Test whether a filter is an FIR filter.
`islinphase`	All filters	Test whether a filter is linear phase.
`ismaxphase`	All filters	Test whether a filter is maximum phase.
`isminphase`	All filters	Test whether a filter is minimum phase.
`isreal`	All filters	Test whether a filter is real.
`isstable`	All filters	Test whether a filter is stable.
`maxstep`	Adaptive filter	Return the maximum step size for an `adaptfilt` object.
`msepred`	Adaptive filter	Return the predicted mean-square error for the `adaptfilt` object.
`msesim`	Adaptive filter	Return the measured mean-square error for the `adaptfilt` object via simulation.

Method Name	Supported Objects	Description
`normalize`	Discrete-time filters	Normalize the filter coefficients.
`nstates`	Multirate filter	Return the number of states in an `mfilt` object.
`phasez`	All filters	Return the filter phase response.
`polyphase`	Multirate filter	Return the polyphase matrix for an `mfilt` object.
`reffilter`	Discrete-time filters	Return the reference filter that corresponds to fixed-point filter.
`reorder`	SOS discrete-time filters	Change the order of the sections in an SOS filter.
`reset`	Adaptive and Multirate filters	Reset the filter to the original conditions.
`scale`	SOS discrete-time filters	Scale the sections of an SOS filter.
`scalecheck`	SOS discrete-time filters	Check the scaling of an SOS filter.
`specifyall`	Discrete-time filters	Set the properties of a fixed-point filter so you can specify them manually.
`stepz`	Adaptive and Multirate filters	Return the step response for a filter
`zerophase`	All filters	Return the zerophase response for a filter.
`zplane`	All filters	Return the pole-zero plot for a filter.

To see the full listing of analysis methods that apply to the `adaptfilt`, `dfilt`, and `mfilt` objects, type `help adaptfilt`, `help dfilt`, and `help mfilt` at the MATLAB prompt.

The Filter Visualization Tool (FVTool) supports these objects so you can use the full power of FVTool to analyze the objects you create.

Fixed-Point Filter Construction and Property Functions

Function	Description
`get`	Get properties of a quantized filter
`isreal`	Test if filter coefficients are real
`num2bin`	Convert a number to two's-complement binary string
`num2hex`	Convert a number to hexadecimal string
`reset`	Reset the properties of a quantized filter to their initial values
`scale`	Scale the sections of second-order section filters
`scalecheck`	Check the scaling of a second-order sections filter
`scaleopts`	Create an object that contains scaling options for second-order section scaling
`set`	Set properties of a quantized filter
`cell2sos`	Convert a cell array to a second-order sections matrix
`sos`	Convert a quantized filter to second-order sections form, order, and scale
`sos2cell`	Convert a second-order sections matrix to a cell array

Quantized Filter Analysis Functions

Function	Description
`freqz`	Compute the frequency response for a quantized filter
`impz`	Compute the impulse response for a quantized filter
`isallpass`	Test quantized filters to determine if they are allpass structures
`isfir`	Test quantized filters to see if they are FIR filters
`islinphase`	Test quantized filters to see if they are linear phase
`ismaxphase`	Test quantized filters to see if they are maximum phase filters
`isminphase`	Test quantized filters to see if they are minimum phase filters
`isreal`	Test quantized filters for purely real coefficients
`issos`	Test whether quantized filters are composed of second-order sections
`isstable`	Test for stability of quantized filters
`noisepsd`	Compute the power spectral density (PSD) of filter output caused by round-off noise during the quantization process
`noisepsdopts`	Create an object that contains options for running the output noise PSD computation `noisepsd` on a filter
`zplane`	Compute a pole-zero plot for a quantized filter

Table 9-1: Quantized Filtering Functions

Function	Description
`filter`	Filter data with a quantized filter
`normalize`	Normalize quantized filter coefficients

Second-Order Sections Conversion Functions

Function	Description
cell2sos	Convert a cell array to a second-order sections matrix
sos	Convert a quantized filter to second-order sections form, order, and scale
sos2cell	Convert a second-order sections matrix to a cell array

Filter Design Functions

Function	Description
fircband	Design constrained-band Remez FIR filters
firceqrip	Design constrained, equiripple, finite impulse response (FIR) filters
firgr	Design optimal equiripple FIR (finite impulse response) digital filters based on the Parks-McClellan algorithm
firhalfband	Design half-band FIR filters
firlpnorm	Design least-pth norm optimal FIR filters
firminphase	Compute the minimum phase FIR spectral factor of linear phase FIR filters
firnyquist	Design lowpass Nyquist (L-th band) FIR filters
ifir	Design interpolated FIR filters
iircomb	Design comb IIR filters with periodic frequency response
iirgrpdelay	Design least-pth norm IIR filters with given group delay
iirlpnorm	Design least-pth norm IIR filters
iirlpnormc	Design constrained least-pth norm IIR filters

Filter Design Functions

Function	Description
iirnotch	Design notch IIR filters to attenuate a fixed frequency
iirpeak	Design peaking IIR filters for boosting or cutting specific frequencies

Filter Conversion Functions

Function	Description
ca2tf	Convert coupled allpass filters to transfer function form
cl2tf	Convert lattice coupled allpass filters to transfer function form
convert	Convert dfilt objects from one structure to another
firlp2lp	Transform lowpass FIR filters to lowpass filters with different passband specifications
firlp2hp	Transform lowpass FIR filters to highpass FIR filters
iirlp2bp	Transform lowpass IIR filters to bandpass filters
iirlp2bs	Transform lowpass IIR filters to bandstop filters
iirlp2hp	Transform lowpass IIR filters to highpass filters
iirlp2lp	Transform lowpass IIR filters to lowpass filters
iirpowcomp	Compute the power complementary IIR filter
tf2ca	Convert transfer function form to coupled allpass form
tf2cl	Convert transfer function form to lattice coupled allpass form

Functions Operating on Discrete-Time Filters

The following table lists functions that operate directly on discrete-time filters. Some are overloaded and operate on other quantized objects such as multirate filters. Overloaded functions are marked in the table.

Functions	Functions That Operate Directly on Quantized Filters	Overloaded Functions
convert	÷	
cumsec	÷	÷
disp	÷	÷
denormalize	÷	
double	÷	
filter	÷	÷
freqz	÷	÷
get	÷	÷
grpdelay	÷	÷
impz	÷	÷
isallpass	÷	
isfir	÷	
islinphase	÷	
ismaxphase	÷	
isminphase	÷	
isreal	÷	÷
issos	÷	
isstable	÷	

Functions	Functions That Operate Directly on Quantized Filters	Overloaded Functions
normalize	÷	
num2bin	÷	
num2hex	÷	
order	÷	
phase	÷	÷
phasez	÷	÷
range	÷	÷
reffilter	÷	
reorder	÷	
reset	÷	÷
scale	÷	
scalecheck	÷	
scaleopts	÷	
set	÷	÷
specifyall	÷	
sos	÷	
zerophase	÷	÷
zplane	÷	÷

To get command line help on an overloaded function `functionname` for quantized filters, type either one of the following syntax options:

```
help dfilt/functionname
help dfilt.functionname
```

To get online help from the Help system, type these syntax options instead:

```
doc dfilt.functionname
doc functionname
```

For most functions and methods, use

```
doc functionname
```

Functions for Designing Discrete-Time Filters

The following functions design digital FIR filters:

- `fircband`
- `firceqrip`
- `firgr`
- `firlpnorm`
- `firhalfband`
- `firminphase`
- `firnyquist`
- `firgr`
- `ifir`

The following functions design digital IIR filters:

- `iircomb`
- `iirgrpdelay`
- `iirlpnorm`
- `iirlpnormc`
- `iirnotch`
- `iirpeak`

The following functions return adaptive (`adaptfilt`) filter objects:

- `adaptfilt.adjlms`
- `adaptfilt.ap`
- `adaptfilt.apru`
- `adaptfilt.bap`
- `adaptfilt.blms`
- `adaptfilt.blmsfft`
- `adaptfilt.dlms`
- `adaptfilt.fdaf`
- `adaptfilt.filtxlms`
- `adaptfilt.ftf`
- `adaptfilt.gal`
- `adaptfilt.hrls`
- `adaptfilt.hswrls`
- `adaptfilt.lms`
- `adaptfilt.lsl`

- `adaptfilt.nlms`
- `adaptfilt.pbfdaf`
- `adaptfilt.pbufdaf`
- `adaptfilt.qrdrls`
- `adaptfilt.rls`
- `adaptfilt.sd`
- `adaptfilt.se`
- `adaptfilt.ss`
- `adaptfilt.swrls`
- `adaptfilt.swftf`
- `adaptfilt.tdafdct`
- `adaptfilt.tdafdft`
- `adaptfilt.ufdaf`

The following functions return discrete-time filter objects (`dfilt` objects):

- `dfilt.calattice`
- `dfilt.calatticepc`
- `dfilt.cascade`
- `dfilt.dfasymfir`
- `dfilt.df1`
- `dfilt.df1sos`
- `dfilt.df1t`
- `dfilt.df2`
- `dfilt.df2sos`
- `dfilt.df2t`
- `dfilt.dffir`
- `dfilt.dffirt`
- `dfilt.latticeallpass`
- `dfilt.latticear`
- `dfilt.latticemamin`
- `dfilt.latticemamax`
- `dfilt.latticearma`
- `dfilt.dfsymfir`

The following functions return multirate filter objects:

- `mfilt.cicdecim`
- `mfilt.cicinterp`

- `mfilt.fftfirinterp`
- `mfilt.firdecim`
- `mfilt.firfracdecim`
- `mfilt.firfracinterp`
- `mfilt.firinterp`
- `mfilt.firsrc`
- `mfilt.firtdecim`
- `mfilt.holdinterp`
- `mfilt.linearinterp`

The following functions transform the frequency response of digital filters from one type to another, such as lowpass to highpass:

IIR transforms

- `firlp2lp`
- `firlp2hp`
- `iirlp2bp`
- `iirlp2bs`
- `iirlp2hp`
- `iirlp2lp`
- `iirlp2mb`
- `iirlp2xn`
- `iirlp2bpc`
- `iirlp2bsc`
- `iirshiftc`
- `iirlp2mbc`
- `iirlp2xc`
- `iirbpc2bpc`
- `iirrateup`
- `iirftransf`

ZPK transforms

- `zpklp2lp`
- `zpklp2hp`
- `zpklp2bp`
- `zpklp2bs`
- `zpkshift`

- `zpklp2mb`
- `zpklp2xn`
- `zpklp2bpc`
- `zpklp2bsc`
- `zpkshiftc`
- `zpklp2mbc`
- `zpklp2xc`
- `zpkbpc2bpc`
- `zpkrateup`
- `zpkftransf`

The following functions convert the structures of digital filters:

- `ca2tf`
- `cl2tf`
- `convert`
- `iirpowcomp`
- `tf2ca`
- `tf2cl`

To get command line help on a design or conversion function such as `firgr` or `fdesign`, type either

- `doc` *`functionname`* (for example `doc firgr` or `doc fdesign`)
- `doc` *`objecttype.function`* where `objecttype` is one of the following strings that specify the version of help to see:
  - `adaptfilt`
  - `dfilt`
  - `fdesign`
  - `mfilt`

  For example, for information about the method you use to construct a direct form 2 filter, use `doc dfilt.df2`. Or for information about creating lowpass filter design objects, use `doc fdesign.lowpass`.

Functions — Alphabetical List

This following pages provide the reference information for every function in the toolbox, in alphabetical order by the name of the function.

adaptfilt

Purpose

Construct an adaptive filter object

Syntax

```
ha = adaptfilt.algorithm(input1,input2, )
```

Description

`ha = adaptfilt.`*`algorithm`*`('input1',input2, )` returns the adaptive filter object `ha` that uses the adaptive filtering technique specified by *`algorithm`*. When you construct an adaptive filter object, include an *`algorithm`* specifier to implement a specific adaptive filter. Note that you do not enclose the algorithm option in single quotation marks as you do for most strings. To construct an adaptive filter object you must supply an *`algorithm`* string—there is no default algorithm, although every constructor creates a default adaptive filter when you do not provide input arguments such as `input1` or `input2` in the calling syntax.

Algorithms

For adaptive filter (`adaptfilt`) objects, the *`algorithm`* string determines which adaptive filter algorithm your `adaptfilt` object implements. Each available algorithm entry appears in one of the tables along with a brief description of the algorithm. Click on the algorithm in the first column to get more information about the associated adaptive filter technique.

- LMS based adaptive filters
- RLS based adaptive filters
- Affine projection adaptive filters
- Adaptive filters in the frequency domain
- Lattice based adaptive filters

Least Mean Squares (LMS) Based FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.adjlms`	Use the Adjoint LMS FIR adaptive filter algorithm
`adaptfilt.blms`	Use the Block LMS FIR adaptive filter algorithm
`adaptfilt.blmsfft`	Use the FFT-based Block LMS FIR adaptive filter algorithm
`adaptfilt.dlms`	Use the delayed LMS FIR adaptive filter algorithm
`adaptfilt.filtxlms`	Use the filtered-x LMS FIR adaptive filter algorithm
`adaptfilt.lms`	Use the LMS FIR adaptive filter algorithm
`adaptfilt.nlms`	Use the normalized LMS FIR adaptive filter algorithm
`adaptfilt.sd`	Use the sign-data LMS FIR adaptive filter algorithm
`adaptfilt.se`	Use the sign-error LMS FIR adaptive filter algorithm
`adaptfilt.ss`	Use the sign-sign LMS FIR adaptive filter algorithm

For further information about an adapting algorithm, refer to the reference page for the algorithm.

Recursive Least Squares (RLS) Based FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.ftf`	Use the fast transversal least squares adaptation algorithm
`adaptfilt.qrdrls`	Use the QR-decomposition RLS adaptation algorithm
`adaptfilt.hrls`	Use the householder RLS adaptation algorithm
`adaptfilt.hswrls`	Use the householder SWRLS adaptation algorithm
`adaptfilt.rls`	Use the recursive-least squares (RLS) adaptation algorithm
`adaptfilt.swrls`	Use the sliding window (SW) RLS adaptation algorithm
`adaptfilt.swftf`	Use the sliding window FTF adaptation algorithm

For more complete information about an adapting algorithm, refer to the reference page for the algorithm.

Affine Projection (AP) FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.ap`	Use the affine projection algorithm that uses direct matrix inversion
`adaptfilt.apru`	Use the affine projection algorithm that uses recursive matrix updating
`adaptfilt.bap`	Use the block affine projection adaptation algorithm

To find more information about an adapting algorithm, refer to the reference page for the algorithm.

FIR Adaptive Filters in the Frequency Domain (FD)

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.fdaf`	Use the frequency domain adaptation algorithm
`adaptfilt.pbfdaf`	Use the partition block version of the FDAF algorithm
`adaptfilt.pbufdaf`	Use the partition block unconstrained version of the FDAF algorithm
`adaptfilt.tdafdct`	Use the transform domain adaptation algorithm using DCT
`adaptfilt.tdafdft`	Use the transform domain adaptation algorithm using DFT
`adaptfilt.ufdaf`	Use the unconstrained FDAF algorithm for adaptation

For more information about an adapting algorithm, refer to the reference page for the algorithm.

Lattice Based (L) FIR Adaptive Filters

adaptfilt.algorithm String	Description of the Adapting Algorithm Used to Generate Filter Coefficients During Adaptation
`adaptfilt.gal`	Use the gradient adaptive lattice filter adaptation algorithm
`adaptfilt.lsl`	Use the least squares lattice adaptation algorithm
`adaptfilt.qrdlsl`	Use the QR decomposition least squares lattice adaptation algorithm

For more information about an adapting algorithm, refer to the reference page for the algorithm.

Properties for all Adaptive Filter Objects

Each reference page for an algorithm and `adaptfilt.algorithm` object specifies which properties apply to the adapting algorithm and how to use them.

Methods for Adaptive Filter Objects

As is true with all objects, methods enable you to perform various operations on `adaptfilt` objects. To use the methods, you apply them to the object handle that you assigned when you constructed the `adaptfilt` object.

Most of the analysis methods that apply to `dfilt` objects also work with `adaptfilt` objects. Methods like `freqz` rely on the filter coefficients in the `adaptfilt` object. Since the coefficients change each time the filter adapts to data, you should view the results of using a method as an analysis of the filter at a moment in time for the object. Use caution when you apply an analysis method to your adaptive filter objects—always check that your result approached your expectation.

In particular, the Filter Visualization Tool (FVTool) supports all of the `adaptfilt` objects. Analyzing and viewing your `adaptfilt` objects is straightforward—use the `fvtool` method with the name of your object

```
fvtool(objectname)
```

to launch FVTool and work with your object.

Some methods share their names with functions in the Signal Processing Toolbox, or even functions in this toolbox. Functions that share names with methods behave in a similar way. Using the same name for more than one function or method is called *overloading* and is common is many toolboxes.

Method	Description
`adaptfilt/coefficients`	Return the instantaneous adaptive filter coefficients
`adaptfilt/filter`	Apply an `adaptfilt` object to your signal
`adaptfilt/freqz`	Plot the instantaneous adaptive filter frequency response
`adaptfilt/grpdelay`	Plot the instantaneous adaptive filter group delay
`adaptfilt/impz`	Plot the instantaneous adaptive filter impulse response.
`adaptfilt/info`	Return the adaptive filter information.
`adaptfilt/isfir`	Test whether an adaptive filter is an finite impulse response (FIR) filters.
`adaptfilt/islinphase`	Test whether an adaptive filter is linear phase
`adaptfilt/ismaxphase`	Test whether an adaptive filter is maximum phase
`adaptfilt/isminphase`	Test whether an adaptive filter is minimum phase
`adaptfilt/isreal`	True whether an adaptive filter has real coefficients
`adaptfilt/isstable`	Test whether an adaptive filter is stable

Method	Description
adaptfilt/maxstep	Return the maximum step size for an adaptive filter
adaptfilt/msepred	Return the predicted mean square error
adaptfilt/msesim	Return the measured mean square error via simulation.
adaptfilt/phasez	Plot the instantaneous adaptive filter phase response
adaptfilt/reset	Reset an adaptive filter to initial conditions
adaptfilt/stepz	Plot the instantaneous adaptive filter step response
adaptfilt/tf	Return the instantaneous adaptive filter transfer function
adaptfilt/zerophase	Plot the instantaneous adaptive filter zerophase response
adaptfilt/zpk	Return a matrix containing the instantaneous adaptive filter zero, pole, and gain values
adaptfilt/zplane	Plot the instantaneous adaptive filter in the Z-plane

Working with Adaptive Filter Objects

The next sections cover viewing and changing the properties of `adaptfilt` objects. Generally, modifying the properties is the same for `adaptfilt`, `dfilt`, and `mfilt` objects and most of the same methods apply to all.

Viewing Object Properties

As with any object, you can use `get` to view a `adaptfilt` object's properties. To see a specific property, use

```
get(ha,'property')
```

To see all properties for an object, use

```
get(ha)
```

Changing Object Properties

To set specific properties, use

```
set(ha,'property1',value1,'property2',value2,...)
```

You must use single quotation marks around the property name so MATLAB treats them as strings.

Copying an Object

To create a copy of an object, use `copy`.

```
ha2 = copy(ha)
```

Note Using the syntax `ha2 = ha` copies only the object handle and does not create a new object—`ha` and `ha2` are not independent. When you change the characteristics of `ha2`, those of `ha` change as well.

Using Filter States

Two properties control your adaptive filter states.

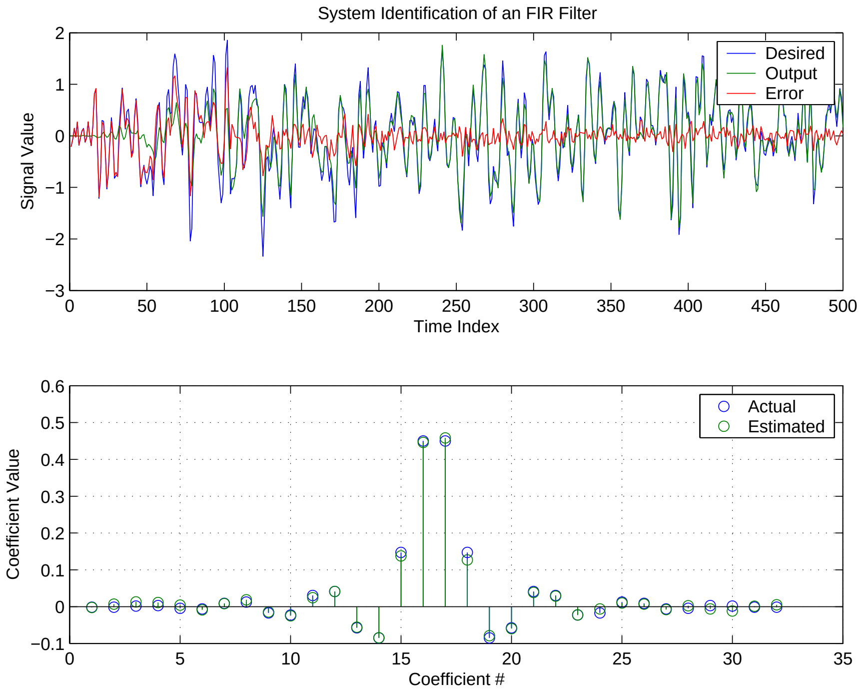
- `States`—stores the current states of the filter. Before the filter is applied, the states correspond to the initial conditions and after the filter is applied, the states correspond to the final conditions.
- `PersistentMemory`—resets the filter before filtering. The default value is `false` which causes the properties that are modified by the filter, such as `coefficients` and `states`, to be reset to the value you specified when you constructed the object, before you use the object to filter data. Setting `PersistentMemory` to `true` allows the object to retain its current properties between filtering operations, rather than resetting the filter to its property values at construction.

Examples

Construct an LMS adaptive filter object and use it to identify an unknown system. For this example, use 500 iteration of the adapting process to determine the unknown filter coefficients. Using the LMS algorithm represents one of the most straightforward technique for adaptive filters.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
mu = 0.008;            % LMS step size.
ha = adaptfilt.lms(32,mu);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.coefficients.']);
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```

Glancing at the figure shows you the coefficients after adapting closely match the desired unknown FIR filter.



See Also

`dfilt`, `filter`, `mfilt`

Purpose

Adjoint LMS FIR adaptive filter that adapts using an adjoint LMS algorithm

Syntax

```
ha = adaptfilt.adjlms(l,step,leakage,pathcoeffs,pathest,
   errstates,pstates,coeffs,states)
```

Description

`ha = adaptfilt.adjlms(l,step,leakage,pathcoeffs,pathest, errstates,pstates,coeffs,states)` constructs object `ha`, an FIR adjoint LMS adaptive filter. `l` is the adaptive filter length (the number of coefficients or taps) and must be a positive integer. `l` defaults to 10 when you omit the argument. `step` is the adjoint LMS step size. It must be a nonnegative scalar. When you omit the `step` argument, `step` defaults to 0.1.

`leakage` is the adjoint LMS leakage factor. It must be a scalar between 0 and 1. When `leakage` is less than one, you implement a leaky version of the `adjlms` algorithm to determine the filter coefficients. `leakage` defaults to 1 specifying no leakage in the algorithm.

`pathcoeffs` is the secondary path filter model. This vector should contain the coefficient values of the secondary path from the output actuator to the error sensor.

`pathest` is the estimate of the secondary path filter model. `pathest` defaults to the values in `pathcoeffs`.

`errstates` is a vector of error states of the adaptive filter. It must have a length equal to the filter order of the secondary path model estimate. `errstates` defaults to a vector of zeros of appropriate length. `pstates` contains the secondary path FIR filter states. It must be a vector of length equal to the filter order of the secondary path model. `pstates` defaults to a vector of zeros of appropriate length. The initial filter coefficients for the secondary path filter compose vector `coeffs`. It must be a length `l` vector. `coeffs` defaults to a length `l` vector of zeros. `states` is a vector containing the initial filter states. It must be a vector of length `l+ne-1`, where `ne` is the length of `errstates`. When you omit `states`, it defaults to an appropriate length vector of zeros.

Properties

In the syntax for creating the `adaptfilt` object, the input options are properties of the object created. This table list all the properties for the adjoint LMS object, their default values, and a brief description of the property.

Property	Default Value	Description
`Algorithm`		Specifies the adaptive filter algorithm the object uses during adaptation
`Coefficients`	Length l vector with zeros for all elements	Adjoint LMS FIR filter coefficients. Should be initialized with the initial coefficients for the FIR filter prior to adapting. You need `l` entries in `coefficients`. Updated filter coefficients are returned in `coefficients` when you use `s` as an output argument.
`NumSamplesProcessed`	Any positive integer	Specifies the number performed during the adaptation process
`ErrorStates`	[0,...,0]	A vector of the error states for your adaptive filter, with length equal to the order of your secondary path filter
`FilterLength`	10	The number of coefficients in your adaptive filter

Property	Default Value	Description
`Leakage`	1	Specifies the leakage parameter. Allows you to implement a leaky algorithm. Including a leakage factor can improve the results of the algorithm by forcing the algorithm to continue to adapt even after it reaches a minimum value. Ranges between 0 and 1.
`SecondaryPathCoeffs`	No default	A vector that contains the coefficient values of your secondary path from the output actuator to the error sensor
`SecondaryPathEstimate`	`pathcoeffs` values	An estimate of the secondary path filter model
`SecondaryPathStates`	Length of the secondary path filter. All elements are zeros.	The states of the secondary path filter, the unknown system

Property	Default Value	Description
States	l+ne+1, where ne is length(errstates)	Contains the initial conditions for your adaptive filter and returns the states of the FIR filter after adaptation. If omitted, it defaults to a zero vector of length equal to l+ne+1. When you use adaptfilt.adjlms in a loop structure, use this element to specify the initial filter states for the adapting FIR filter.

Property	Default Value	Description
`Stepsize`	0.1	Sets the adjoint LMS algorithm step size used for each iteration of the adapting algorithm. Determines both how quickly and how closely the adaptive filter converges to the filter solution.
`PersistentMemory`	`false` or `true`	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. `PersistentMemory` returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to `false`.

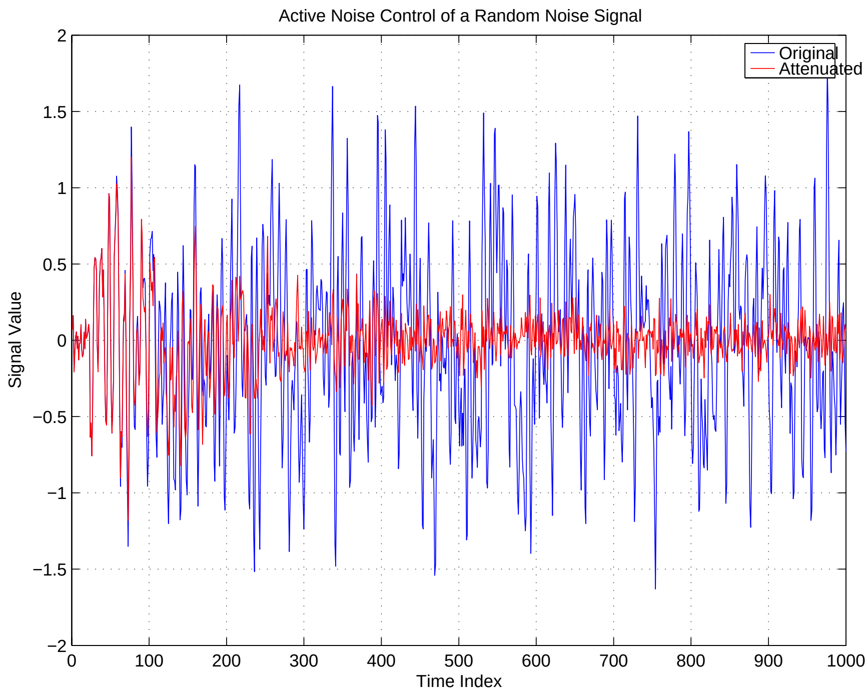
Example

Demonstrate active noise control of a random noise signal that runs for 1000 samples.

```
x  = randn(1,1000);      % Noise source
g  = fir1(47,0.4);       % FIR primary path system model
n  = 0.1*randn(1,1000);  % Observation noise signal
d  = filter(g,1,x)+n;    % Signal to be canceled (desired)
b  = fir1(31,0.5);       % FIR secondary path system model
mu = 0.008;              % Adjoint LMS step size
```

```
ha = adaptfilt.adjlms(32,mu,1,b);
[y,e] = filter(ha,x,d);
plot(1:1000,d,'b',1:1000,e,'r');
title('Active Noise Control of a Random Noise Signal');
legend('Original','Attenuated');
xlabel('Time Index'); ylabel('Signal Value');  grid on;
```

Reviewing the figure shows that the adaptive filter attenuates the original noise signal as you expect.



See Also

`adaptfilt.dlms`, `adaptfilt.filtxlms`

References

Wan, Eric., "Adjoint LMS: An Alternative to Filtered-X LMS and Multiple Error LMS," Proceedings of the International Conference on Acoustics, Speech, and Signal Processing (ICASSP), pp. 1841-1845, 1997

Purpose

Construct an affine projection FIR adaptive filter object that uses direct matrix inversion

Syntax

```
ha = adaptfilt.ap(l,step,projectord,offset,coeffs,states,
   errstates,epsstates)
```

Description

`ha = adaptfilt.ap(l,step,projectord,offset,coeffs,states, errstates,epsstates)` constructs an affine projection FIR adaptive filter `ha` using direct matrix inversion.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.ap`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	Affine projection step size. This is a scalar that should be a value between zero and one. Setting `step` equal to one provides the fastest convergence during adaptation. `step` defaults to 1.
`projectord`	Projection order of the affine projection algorithm. `projectord` defines the size of the input signal covariance matrix and defaults to two.
`offset`	Offset for the input signal covariance matrix. You should initialize the covariance matrix to a diagonal matrix whose diagonal entries are equal to the offset you specify. `offset` should be positive. `offset` defaults to one.

Input Argument	Description
coeffs	Vector containing the initial filter coefficients. It must be a length l vector, the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
states	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).
errstates	Vector of the adaptive filter error states. errstates defaults to a zero vector with length equal to (projectord - 1).
epsstates	Vector of the epsilon values of the adaptive filter. epsstates defaults to a vector of zeros with (projectord - 1) elements.

Properties

Since your adaptfilt.ap filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.ap objects. To show you the properties that apply, this table lists and describes each property for the affine projection filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps
NumSamplesProcessed	Any positive integer	Specifies the number performed during the adaptation process

Name	Range	Description
ProjectionOrder		Projection order of the affine projection algorithm. projectord defines the size of the input signal covariance matrix and defaults to two.
OffsetCov		Contains the offset covariance matrix
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector, the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
States	Vector of elements, data type double	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).
ErrorStates	Vector of elements	Vector of the adaptive filter error states. errstates defaults to a zero vector with length equal to (projectord - 1).
EpsilonStates	Vector of elements	Vector of the epsilon values of the adaptive filter. epsstates defaults to a vector of zeros with (projectord - 1) elements.

Name	Range	Description
`StepSize`	Any scalar from zero to one, inclusive	Specifies the step size taken between filter coefficient updates
`PersistentMemory`	`false` or `true`	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. `PersistentMemory` returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to `true`.

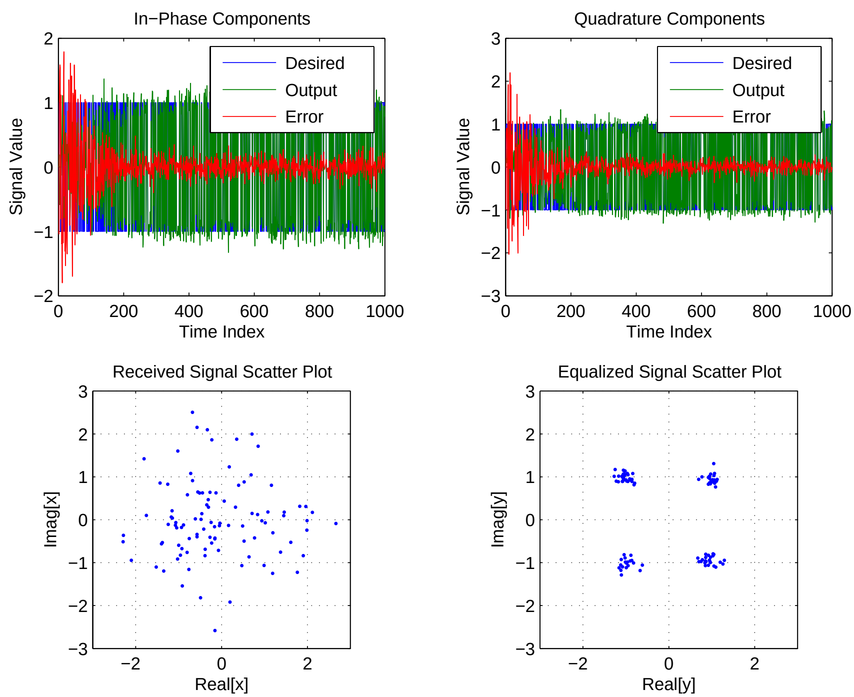
Example

Quadrature phase shift keying (QPSK) adaptive equalization using a 32-coefficient FIR filter. Run the adaptation for 1000 iterations.

```
D  = 16;                      % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];    % Numerator coefficients of channel
a  = [1 -0.7];              % Denominator coefficients of channel
ntr= 1000;                    % Number of iterations
s  = sign(randn(1,ntr+D)) + j*sign(randn(1,ntr+D));% Baseband
                                                   % QPSK signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));     % Noise signal
r  = filter(b,a,s)+n;         % Received signal
x  = r(1+D:ntr+D);    % Input signal (received signal)
d  = s(1:ntr);        % Desired signal (delayed QPSK signal)
mu = 0.1;             % Step size
po = 4;               % Projection order
offset = 0.05;        % Offset for covariance matrix
ha = adaptfilt.ap(32,mu,po,offset);
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
title('In-Phase Components');
```

```
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```

The four plots shown reveal the QPSK process at work.



See also `msesim`

References

[1] K. Ozeki and Umeda, T., "An Adaptive Filtering Algorithm Using an Orthogonal Projection to an Affine Subspace and Its Properties," Electronics and Communications in Japan, vol.67-A, no. 5, pp. 19-27, May 1984

[2] Y. Maruyama, "A Fast Method of Projection Algorithm," Proc. 1990 IEICE Spring Conf., B-744

Purpose

Return an affine projection FIR adaptive filter object that uses recursive matrix updating

Syntax

```
ha = adaptfilt.apru(l,step,projectord,offset,coeffs,states,
   errstates,epsstates)
```

Description

`ha = adaptfilt.apru(l,step,projectord,offset,coeffs,states, errstates,epsstates)` constructs an affine projection FIR adaptive filter `ha` using recursive matrix updating.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.apru`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps). It must be a positive integer. `l` defaults to 10.
`step`	Affine projection step size. This is a scalar that should be a value between zero and one. Setting `step` equal to one provides the fastest convergence during adaptation. `step` defaults to 1.
`projectord`	Projection order of the affine projection algorithm. `projectord` defines the size of the input signal covariance matrix and defaults to two.
`offset`	Offset for the input signal covariance matrix. You should initialize the covariance matrix to a diagonal matrix whose diagonal entries are equal to the offset you specify. `offset` should be positive. `offset` defaults to one.

Input Argument	Description
coeffs	Vector containing the initial filter coefficients. It must be a length l vector, the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
states	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).
errstates	Vector of the adaptive filter error states. errstates defaults to a zero vector with length equal to (projectord - 1).
epsstates	Vector of the epsilon values of the adaptive filter. epsstates defaults to a vector of zeros with (projectord - 1) elements.

Properties

Since your adaptfilt.apru filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.apru objects. To show you the properties that apply, this table lists and describes each property for the affine projection filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps
NumSamplesProcessed	Any positive integer	Specifies the number performed during the adaptation process

Name	Range	Description
ProjectionOrder		Projection order of the affine projection algorithm. projectord defines the size of the input signal covariance matrix and defaults to two.
OffsetCov		Contains the offset covariance matrix
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector, the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
States	Vector of elements, data type double	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).
ErrorStates	Vector of elements	Vector of the adaptive filter error states. errstates defaults to a zero vector with length equal to (projectord - 1).
EpsilonStates	Vector of elements	Vector of the epsilon values of the adaptive filter. epsstates defaults to a vector of zeros with (projectord - 1) elements.

Name	Range	Description
StepSize	Any scalar from zero to one, inclusive	Specifies the step size taken between filter coefficient updates
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to true.

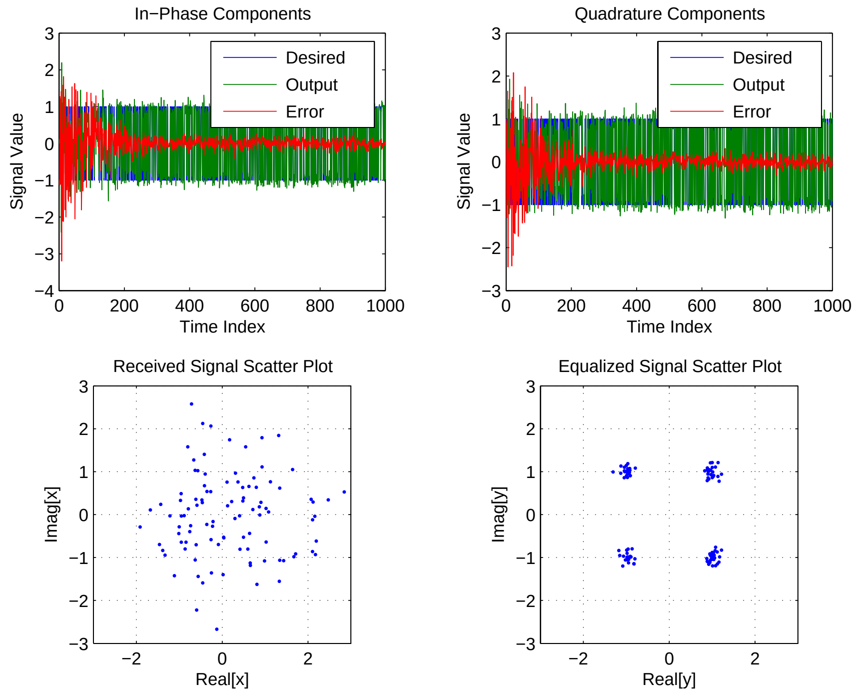
Example

Demonstrate quadrature phase shift keying (QPSK) adaptive equalization using a 32-coefficient FIR filter. In this example we run the adaptation for 1000 iterations.

```
D  = 16;                        % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];    % Numerator coefficients of channel
a  = [1 -0.7];               % Denominator coefficients of channel
ntr= 1000;                      % Number of iterations
s  = sign(randn(1,ntr+D)) + j*sign(randn(1,ntr+D)); % Baseband
                                                    % QPSK sig
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));  % Noise signal
r  = filter(b,a,s)+n;                          % Received signal
x  = r(1+D:ntr+D);             % Input signal (received signal)
d  = s(1:ntr);             % Desired signal (delayed QPSK signal)
mu = 0.1;                      % Step size
po = 4;                        % Projection order
del = 0.05;                    % Offset
ha = adaptfilt.apru(32,mu,po,offset);
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
```

```
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```

In the component and scatter plots below, you see the results of QPSK equalization.



See Also

`adaptfilt`, `adaptfilt.ap`, `adaptfilt.bap`

References

[1] K. Ozeki, Omeda, T, "An Adaptive Filtering Algorithm Using an Orthogonal Projection to an Affine Subspace and Its Properties,", Electronics and Communications in Japan, vol. 67-A, no. 5, pp. 19-27, May 1984

[2] Y. Maruyama, "A Fast Method of Projection Algorithm," Proceedings 1990 IEICE Spring Conference, B-744

Purpose Return a block affine projection FIR adaptive filter object

Syntax

```
ha = adaptfilt.bap(l,step,projectord,offset,coeffs,states)
```

Description

`ha = adaptfilt.bap(l,step,projectord,offset,coeffs,states)` constructs a block affine projection FIR adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.bap`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	Affine projection step size. This is a scalar that should be a value between zero and one. Setting `step` equal to one provides the fastest convergence during adaptation. `step` defaults to 1.
`projectord`	Projection order of the affine projection algorithm. `projectord` defines the size of the input signal covariance matrix and defaults to two.
`offset`	Offset for the input signal covariance matrix. You should initialize the covariance matrix to a diagonal matrix whose diagonal entries are equal to the offset you specify. `offset` should be positive. `offset` defaults to one.

Input Argument	Description
`coeffs`	Vector containing the initial filter coefficients. It must be a length `l` vector, the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.
`states`	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (`l` + `projectord` - 2).

Properties

Since your `adaptfilt.bap` filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating `adaptfilt.bap` objects. To show you the properties that apply, this table lists and describes each property for the affine projection filter object.

Name	Range	Description
`Algorithm`	None	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps
`NumSamplesProcessed`	Any positive integer	Specifies the number performed during the adaptation process
`ProjectionOrder`		Projection order of the affine projection algorithm. `projectord` defines the size of the input signal covariance matrix and defaults to two.

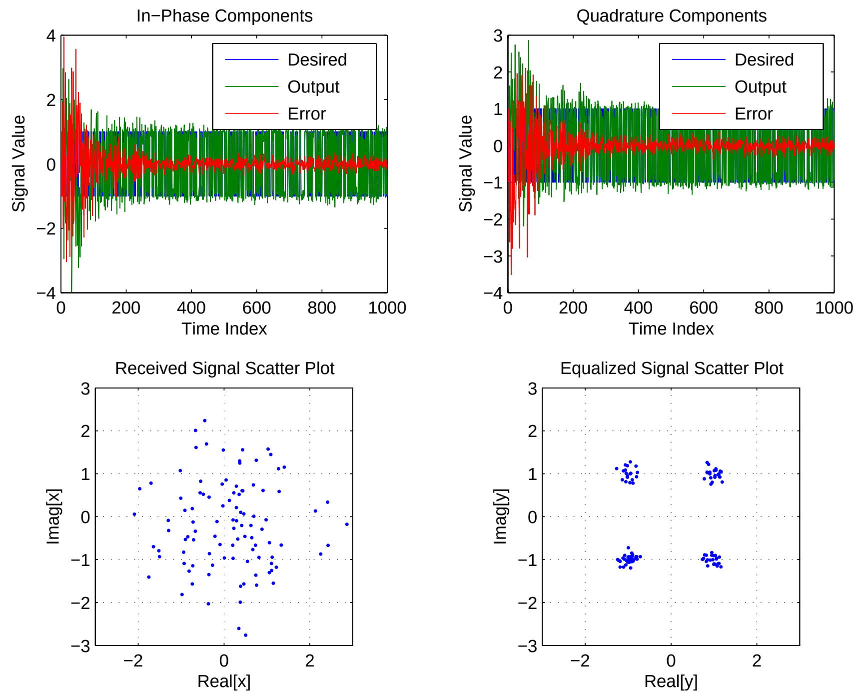
Name	Range	Description
OffsetCov		Contains the offset covariance matrix
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector, the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
States	Vector of elements, data type double	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).
StepSize	Any scalar from zero to one, inclusive	Specifies the step size taken between filter coefficient updates
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to true.

Example

Show an example of quadrature phase shift keying (QPSK) adaptive equalization using a 32-coefficient FIR filter.

```
D  = 16;                           % Number of samples of delay
```

```
b  = exp(j*pi/4)*[-0.7 1];          % Numerator coefficients of
                                     % channel
a  = [1 -0.7];                       % Denominator coefficients
                                     % of channel
ntr= 1000;                           % Number of iterations
s  = sign(randn(1,ntr+D)) + j*sign(randn(1,ntr+D));  % Baseband
                                     % QPSK signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));      % Noise signal
r  = filter(b,a,s)+n;                % Received signal
x  = r(1+D:ntr+D);               % Input signal (received signal)
d  = s(1:ntr);                       % Desired signal (delayed
                                     % QPSK signal)
mu = 0.5;                            % Step size
po = 4;                              % Projection order
offset = 1.0;                     % Offset for covariance matrix
ha = adaptfilt.bap(32,mu,po,offset);
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```



Using the block affine projection object in QPSK results in the plots shown here.

See Also

`adaptfilt`, `adaptfilt.ap`, `adaptfilt.apru`

References

[1] K. Ozeki, Omeda, T, "An Adaptive Filtering Algorithm Using an Orthogonal Projection to an Affine Subspace and Its Properties," Electronics and Communications in Japan, vol. 67-A, no. 5, pp. 19-27, May 1984

[2] M. Montazeri, M, Duhamel, P, "A Set of Algorithms Linking NLMS and Block RLS Algorithms," IEEE Transactions Signal Processing, vol. 43, no. 2, pp, 444-453, February 1995

adaptfilt.blms

Purpose Construct a Block LMS (BLMS) FIR adaptive filter

Syntax

```
ha = adaptfilt.blms(l,step,leakage,blocklen,coeffs,states)
```

Description

ha = adaptfilt.blms(l,step,leakage,blocklen,coeffs,states) constructs an FIR block LMS adaptive filter ha, where l is the adaptive filter length (the number of coefficients or taps) and must be a positive integer. l defaults to 10.

step is the block LMS step size. You must set step to a nonnegative scalar. You can use function maxstep to determine a reasonable range of step size values for the signals being processed. When unspecified, step defaults to 0.

leakage is the block LMS leakage factor. It must be a scalar between 0 and 1. If you set leakage to be less than one, you implement the leaky block LMS algorithm. leakage defaults to 1 specifying no leakage in the adapting algorithm.

blocklen is the block length used. It must be a positive integer and the signal vectors d and x should be divisible by blocklen. Larger block lengths result in faster per-sample execution times but with poor adaptation characteristics. When you choose blocklen such that blocklen + length(coeffs) is a power of 2, use adaptfilt.blmsfft. blocklen defaults to l.

coeffs is a vector of initial filter coefficients. it must be a length l vector. coeffs defaults to length l vector of zeros.

states contains a vector of your initial filter states. It must be a length l vector and defaults to a length l vector of zeros when you do not include it in your calling function.

Properties

In the syntax for creating the adaptfilt object, the input options are properties of the object created. This table list all the properties for the adjoint LMS object, their default values, and a brief description of the property.

Property	Default Value	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps
NumSamplesProcessed	Any positive integer	Specifies the number performed during the adaptation process
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector where l is the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
States	Vector of elements	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to l

Property	Default Value	Description
Leakage		Specifies the leakage parameter. Allows you to implement a leaky algorithm. Including a leakage factor can improve the results of the algorithm by forcing the algorithm to continue to adapt even after it reaches a minimum value. Ranges between 0 and 1.
BlockLength	Vector of length 1	Size of the blocks of data processed in each iteration

Property	Default Value	Description
StepSize	0.1	Sets the block LMS algorithm step size used for each iteration of the adapting algorithm. Determines both how quickly and how closely the adaptive filter converges to the filter solution. Use maxstep to determine the maximum usable step size.
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.

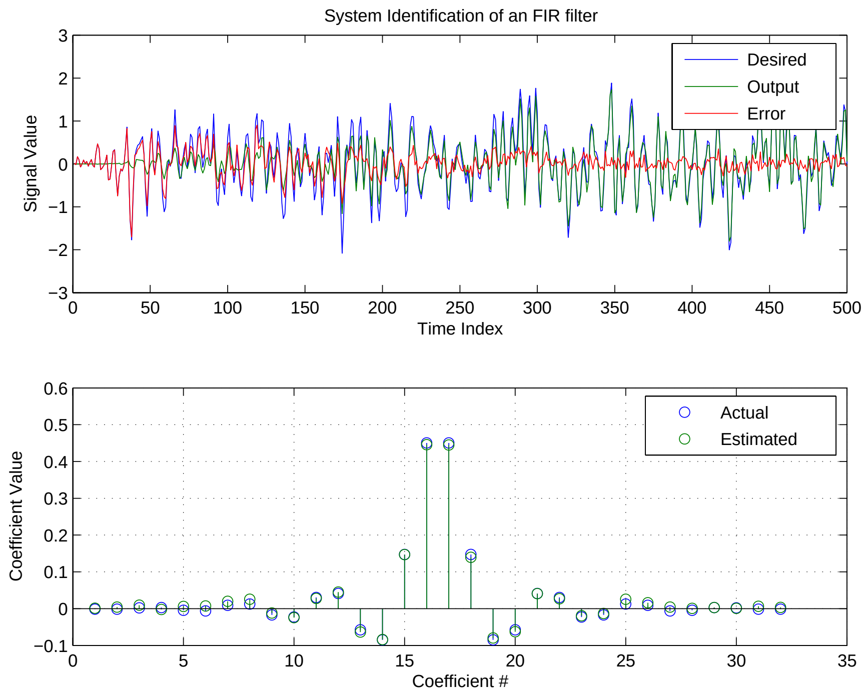
Example

Use an adaptive filter to identify an unknown 32nd-order FIR filter. In this example we input 500 samples to result in 500 iterations of the adaptation process. You see in the plot that follows the example code that the adaptive filter has determined the coefficients of the unknown system under test.

```
x  = randn(1,500);             % Input to the filter
```

```
b  = fir1(31,0.5);              % FIR system to be identified
no  = 0.1*randn(1,500);         % Observation noise signal
d  = filter(b,1,x)+no;          % Desired signal
mu = 0.008;                     % Block LMS step size
n  = 5;                         % Block length
ha = adaptfilt.blms(32,mu,1,n);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.coefficients.']);
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```

Based on looking at the figures here, the adaptive filter correctly identified the unknown system after 500 iterations, or fewer. In the lower plot, you see the comparison between the actual filter coefficients and those determined by the adaptation process.



See Also

`adaptfilt.blmsfft`, `adaptfilt.fdaf`, `adaptfilt.lms`

Reference

J.J. Shynk, “Frequency-Domain and Multirate Adaptive Filtering,” IEEE Signal Processing Magazine, vol. 9, no. 1, pp. 14-37, Jan. 1992.

Purpose

Construct an FFT-based block LMS FIR adaptive filter

Syntax

```
ha = adaptfilt.blmsfft(l,step,leakage,blocklen,coeffs,states)
```

Description

`ha = adaptfilt.blmsfft(l,step,leakage,blocklen,coeffs,states)` constructs an FIR block LMS adaptive filter object `ha` where `l` is the adaptive filter length (the number of coefficients or taps) and must be a positive integer. `l` defaults to 10. `step` is the block LMS step size. It must be a nonnegative scalar. The function maxstep may be helpful to determine a reasonable range of step size values for the signals you are processing. `step` defaults to 0.

`leakage` is the block LMS leakage factor. It must also be a scalar between 0 and 1. When `leakage` is less than one, the `adaptfilt.blmsfft` implements a leaky block LMS algorithm. `leakage` defaults to 1 (no leakage). `blocklen` is the block length used. It must be a positive integer such that

```
blocklen + length(coeffs)
```

is a power of two; otherwise, an `adaptfilt.blms` algorithm is used for adapting. Larger block lengths result in faster execution times, with poor adaptation characteristics as the cost of the speed gained. `blocklen` defaults to `l`. Enter your initial filter coefficients in `coeffs`, a vector of length `l`. When omitted, `coeffs` defaults to a length `l` vector of all zeros. `states` contains a vector of initial filter states; it must be a length `l` vector. `states` defaults to a length `l` vector of all zeros when you omit the `states` argument in the calling syntax.

Properties

In the syntax for creating the adaptfilt object, the input options are properties of the object you create. This table list all the properties for the block LMS object, their default values, and a brief description of the property.

Property	Default Value	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps
NumSamplesProcessed	Any positive integer	Specifies the number performed during the adaptation process
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector where l is the number of filter coefficients. coefficients defaults to length l vector of zeros when you do not provide the argument for input.
States	Vector of elements of length l	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to l

Property	Default Value	Description
Leakage	1	Specifies the leakage parameter. Allows you to implement a leaky algorithm. Including a leakage factor can improve the results of the algorithm by forcing the algorithm to continue to adapt even after it reaches a minimum value. Ranges between 0 and 1.
BlockLength	Vector of length 1	Size of the blocks of data processed in each iteration

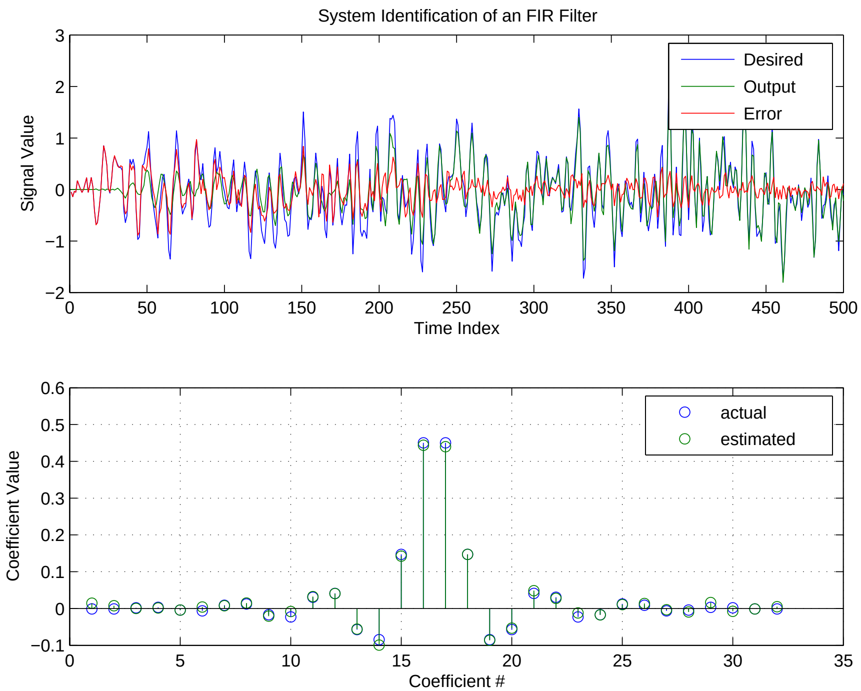
Property	Default Value	Description
`StepSize`	0.1	Sets the block LMS algorithm step size used for each iteration of the adapting algorithm. Determines both how quickly and how closely the adaptive filter converges to the filter solution. Use `maxstep` to determine the maximum usable step size.
`PersistentMemory`	`false` or `true`	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. `PersistentMemory` returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to `false`.

Example

Identify an unknown FIR filter with 32 coefficients using 512 iterations of the adapting algorithm.

```
x  = randn(1,512);       % Input to the filter
b  = fir1(31,0.5);       % FIR system to be identified
no  = 0.1*randn(1,512);  % Observation noise signal
```

```
d  = filter(b,1,x)+no;    % Desired signal
mu = 0.008;               % Step size
n  = 16;                  % Block length
ha = adaptfilt.blmsfft(32,mu,1,n);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d(1:500);y(1:500);e(1:500)]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.coefficients.']);
legend('actual','estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```



As a result of running the adaptation process, filter object `ha` now matches the unknown system FIR filter `b`, based on comparing the filter coefficients derived during adaptation.

See Also

`adaptfilt.blms`, `adaptfilt.fdaf`, `adaptfilt.lms`, `filter`

Reference

J.J. Shynk, "Frequency-Domain and Multirate Adaptive Filtering," IEEE Signal Processing Magazine, vol. 9, no. 1, pp. 14-37, Jan. 1992.

adaptfilt.dlms

Purpose

Create a delayed LMS FIR adaptive filter object

Syntax

```
ha = adaptfilt.dlms(l,step,leakage,delay,errstates,coeffs,
   states)
```

Description

`ha = adaptfilt.dlms(l,step,leakage,delay,errstates,coeffs, states)` constructs an FIR delayed LMS adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.dlms`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	LMS step size. It must be a nonnegative scalar. You can use `maxstep` to determine a reasonable range of step size values for the signals being processed. `step` defaults to 0.
`leakage`	Your LMS leakage factor. It must be a scalar between 0 and 1. When `leakage` is less than one, `adaptfilt.lms` implements a leaky LMS algorithm. When you omit the `leakage` property in the calling syntax, it defaults to 1 providing no leakage in the adapting algorithm.
`delay`	Update delay given in time samples. This scalar should be a positive integer—negative delays do not work. `delay` defaults to 1.

Input Argument	Description
errstates	Vector of the error states of your adaptive filter. It must have a length equal to the update delay (delay) in samples. errstates defaults to an appropriate length vector of zeros.
coeffs	Vector of initial filter coefficients. it must be a length l vector. coeffs defaults to length l vector with elements equal to zero.
states	Vector of initial filter states for the adaptive filter. It must be a length l-1 vector. states defaults to a length l-1 vector of zeros.

Properties

In the syntax for creating the adaptfilt object, the input options are properties of the object you create. This table list all the properties for the block LMS object, their default values, and a brief description of the property.

Property	Default Value	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps

Property	Default Value	Description
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector where l is the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input. LMS FIR filter coefficients. Should be initialized with the initial coefficients for the FIR filter prior to adapting. You need l entries in coeffs.
Delay	1	Specifies the update delay for the adaptive algorithm
ErrorStates	Vector of zeros with the number of elements equal to delay	A vector comprising the error states for the adaptive filter
States	Vector of elements, data type double	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).

Property	Default Value	Description
StepSize	0.1	Sets the LMS algorithm step size used for each iteration of the adapting algorithm. Determines both how quickly and how closely the adaptive filter converges to the filter solution.
Leakage	1	Specifies the leakage parameter. Allows you to implement a leaky algorithm. Including a leakage factor can improve the results of the algorithm by forcing the algorithm to continue to adapt even after it reaches a minimum value. Ranges between 0 and 1.

Property	Default Value	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

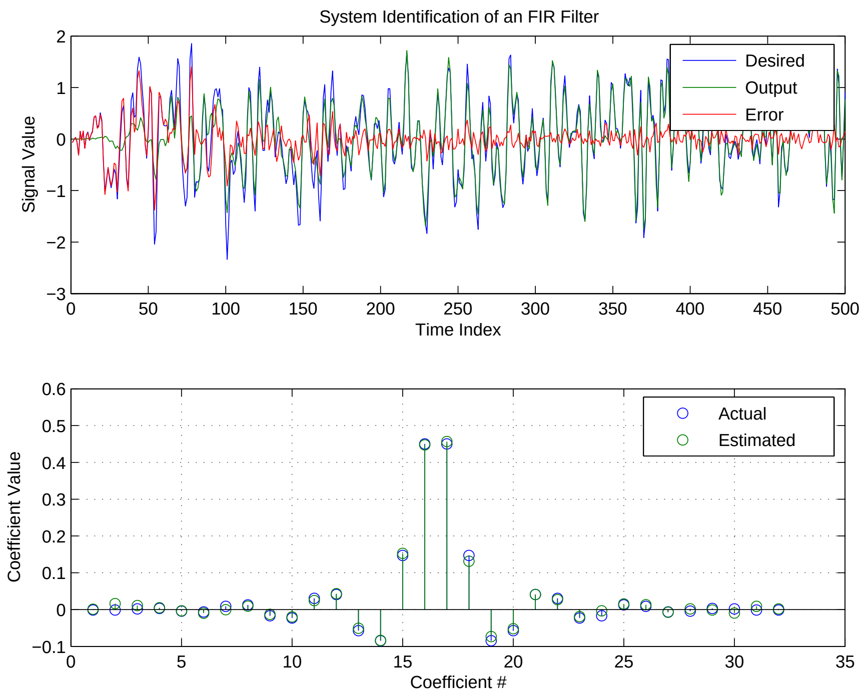
Example

System identification of a 32-coefficient FIR filter. Refer to the figure that follows to see the results of the adapting filter process.

```
x  = randn(1,500);      % Input to the filter
b  = fir1(31,0.5);      % FIR system to be identified
n  = 0.1*randn(1,500);  % Observation noise signal
d  = filter(b,1,x)+n;   % Desired signal
mu = 0.008;             % LMS step size.
delay = 1;              % Update delay
ha = adaptfilt.dlms(32,mu,1,delay);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
```

```
subplot(2,1,2); stem([b.',ha.coefficients.']);
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```

Using a delayed LMS adaptive filter in the process to identify an unknown filter appears to work as planned, as shown in this figure.



See Also

adaptfilt.adjlms, adaptfilt.filtxlms, adaptfilt.lms

Reference

J.J. Shynk, "Frequency-Domain and Multirate Adaptive Filtering," IEEE Signal Processing Magazine, vol. 9, no. 1, pp. 14-37, Jan. 1992.

adaptfilt.fdaf

Purpose

Construct a frequency-domain FIR adaptive filter with bin step size normalization

Syntax

```
ha = adaptfilt.fdaf(l,step,leakage,delta,lambda,blocklen,
   offset,coeffs,states)
```

Description

`ha = adaptfilt.fdaf(l,step,leakage,delta,lambda,blocklen,offset,coeffs,states)` constructs a frequency-domain FIR adaptive filter `ha` with bin step size normalization. If you omit all the input arguments you create a default object with l = 10 and `step` = 1.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.fdaf`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps). l must be a positive integer; it defaults to 10 when you omit the argument.
`step`	Step size of the adaptive filter. This is a scalar and should lie in the range (0,1]. `step` defaults to 1.
`leakage`	Leakage parameter of the adaptive filter. If this parameter is set to a value between zero and one, you implement a leaky FDAF algorithm. `leakage` defaults to 1—no leakage provided in the algorithm.
`delta`	Initial common value of all of the FFT input signal powers. Its initial value should be positive. `delta` defaults to 1.
`lambda`	Specifies the averaging factor used to compute the exponentially-windowed FFT input signal powers for the coefficient updates. `lambda` should lie in the range (0,1]. `lambda` defaults to 0.9.

Input Argument	Description
blocklen	Block length for the coefficient updates. This must be a positive integer. For faster execution, (blocklen + l) should be a power of two. blocklen defaults to l.
offset	Offset for the normalization terms in the coefficient updates. Use this to avoid divide by zeros or by very small numbers when any of the FFT input signal powers become very small. offset defaults to zero.
coeffs	Initial time-domain coefficients of the adaptive filter. coeff should be a length l vector. The adaptive filter object uses these coefficients to compute the initial frequency-domain filter coefficients via an FFT computed after zero-padding the time-domain vector by the blocklen.
states	The adaptive filter states. states defaults to a zero vector that has length equal to l.

Properties

Since your adaptfilt.fdaf filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.fdaf objects. To show you the properties that apply, this table lists and describes each property for the adaptfilt.fdaf filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps

Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.
Leakage		Leakage parameter of the adaptive filter. if this parameter is set to a value between zero and one, you implement a leaky FDAF algorithm. leakage defaults to 1—no leakage provided in the algorithm.
Power		A vector of 2*l elements, each initialized with the value delta from the input arguments. As you filter data, Power gets updated by the filter process.

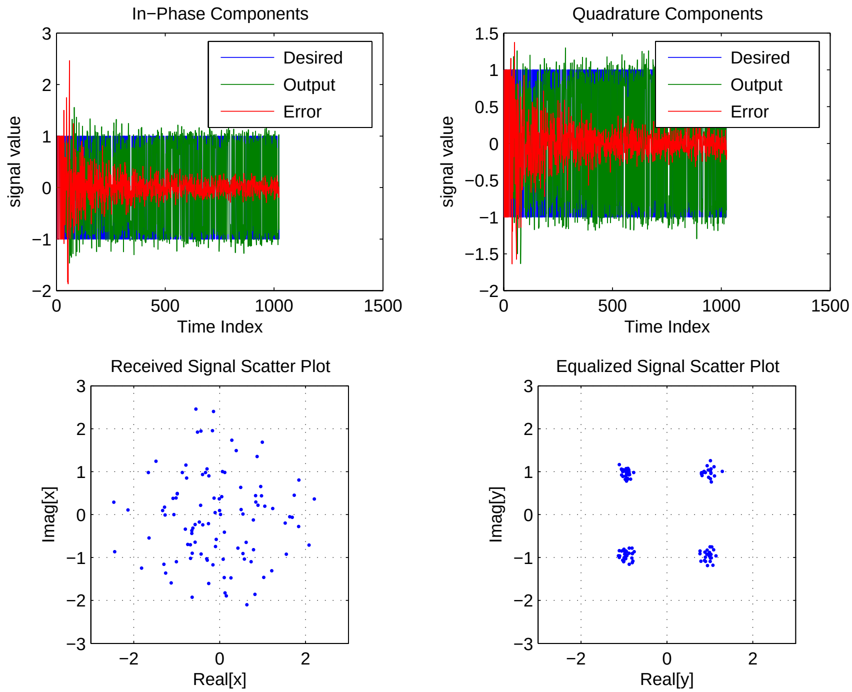
Name	Range	Description
AvgFactor	(0, 1]	Specifies the averaging factor used to compute the exponentially-windowed FFT input signal powers for the coefficient updates. Same as the input argument lambda.
BlockLength	Any integer	Block length for the coefficient updates. This must be a positive integer. For faster execution, (blocklen + 1) should be a power of two. blocklen defaults to 1.
Offset	Any positive real value	Offset for the normalization terms in the coefficient updates. Use this to avoid dividing by zero or by very small numbers when any of the FFT input signal powers become very small. offset defaults to zero.
FFTCoefficients		Stores the discrete Fourier transform of the filter coefficients in coeffs.
FFTStates		States for the FFT operation.
StepSize	Any scalar from zero to one, inclusive	Specifies the step size taken between filter coefficient updates

Examples

Quadrature Phase Shift Keying (QPSK) adaptive equalization using 1024 iterations of a 32-coefficient FIR filter. After this example code, a figure demonstrates the equalization results.

```
D = 16;                          % Number of samples of delay
```

```
b  = exp(j*pi/4)*[-0.7 1];     % Numerator coefficients of channel
a  = [1 -0.7];               % Denominator coefficients of channel
ntr= 1024;                         % Number of iterations
s  = sign(randn(1,ntr+D)) + j*sign(randn(1,ntr+D));  % Baseband
                                   % QPSK signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));      % Noise signal
r  = filter(b,a,s)+n;               % Received signal
x  = r(1+D:ntr+D);                 % Input signal (received signal)
d  = s(1:ntr);                      % Desired signal (delayed QPSK
                                    % signal)
del = 1;                            % Initial FFT input powers
mu  = 0.1;                          % Step size
lam = 0.9;                          % Averaging factor
ha = adaptfilt.fdaf(32,mu,1,del,lam);
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('signal value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('signal value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```



See Also

`adaptfilt.ufdaf`, `adaptfilt.pbfdaf`, `adaptfilt.blms`, `adaptfilt.blmsfft`

Reference

J.J. Shynk, “Frequency-Domain and Multirate Adaptive Filtering,” IEEE Signal Processing Magazine, vol. 9, no. 1, pp. 14-37, Jan. 1992

adaptfilt.filtxlms

Purpose

Create an filtered-x LMS FIR adaptive filter

Syntax

```
ha = adaptfilt.filtxlms(l,step,leakage,pathcoeffs,pathest,
   errstates,pstates,coeffs,states)
```

Description

`ha = adaptfilt.filtxlms(l,step,leakage,pathcoeffs,pathest, errstates,pstates,coeffs,states)` constructs an filtered-x LMS adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.filtxlms`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	Filtered LMS step size. it must be a nonnegative scalar. `step` defaults to 0.1.
`leakage`	is the filtered-x LMS leakage factor. it must be a scalar between 0 and 1. If it is less than one, a leaky version of `adaptfilt.filtxlms` is implemented. `leakage` defaults to 1 (no leakage).
`pathcoeffs`	is the secondary path filter model. this vector should contain the coefficient values of the secondary path from the output actuator to the error sensor.
`pathest`	is the estimate of the secondary path filter model. `pathest` defaults to the values in `pathcoeffs`.
`fstates`	is a vector of filtered input states of the adaptive filter. `fstates` defaults to a zero vector of length equal to (`l` - 1).

Input Argument	Description
pstates	are the secondary path FIR filter states. it must be a vector of length equal to the (length(pathcoeffs) - 1). pstates defaults to a vector of zeros of appropriate length.
coeffs	is a vector of initial filter coefficients. it must be a length l vector. coeffs defaults to length l vector of zeros.
states	Vector of initial filter states. states defaults to a zero vector of length equal to the larger of (length(pathcoeffs) - 1) and (length(pathest) - 1).

Properties

In the syntax for creating the adaptfilt object, the input options are properties of the object created. This table list all the properties for the adjoint LMS object, their default values, and a brief description of the property.

Property	Default Value	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps

Property	Default Value	Description
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector where l is the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
States	Vector of elements	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2)
SecondaryPathCoeffs	No default	A vector that contains the coefficient values of your secondary path from the output actuator to the error sensor
SecondaryPathEstimate	pathcoeffs values	An estimate of the secondary path filter model
SecondaryPathStates	Vector of size (length(pathcoeffs)-1) with all elements equal to zero.	The states of the secondary path FIR filter—the unknown system

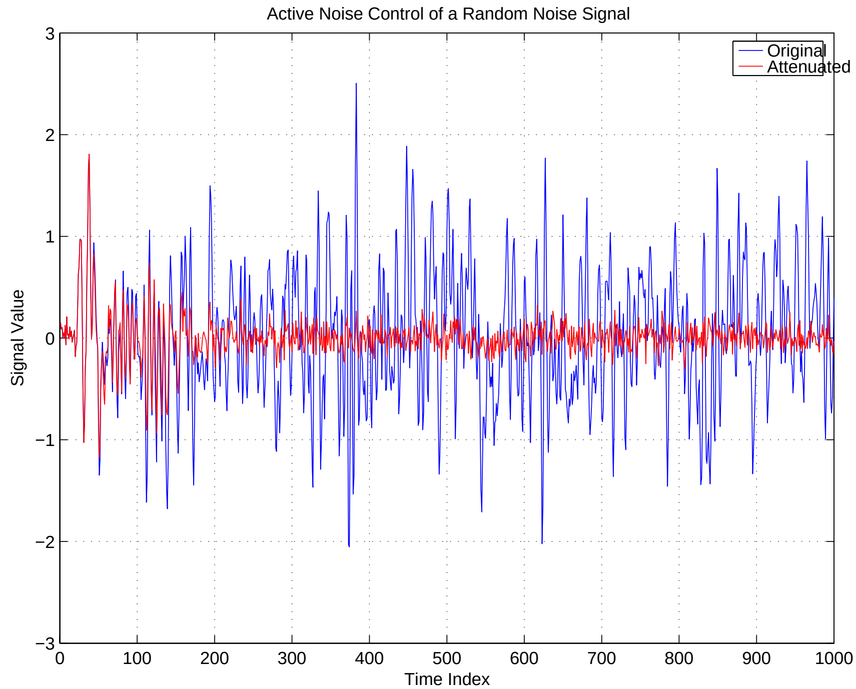
Property	Default Value	Description
FilteredInputStates	l-1	Vector of filtered input states with length equal to l - 1.
StepSize	0.1	Sets the filtered-x algorithm step size used for each iteration of the adapting algorithm. Determines both how quickly and how closely the adaptive filter converges to the filter solution.

Example

Demonstrate active noise control of a random noise signal over 1000 iterations.

As the figure that follows this code demonstrates, the filtered-x LMS filter successfully controls random noise in this context.

```
x  = randn(1,1000);      % Noise source
g  = fir1(47,0.4);       % FIR primary path system model
n  = 0.1*randn(1,1000);  % Observation noise signal
d  = filter(g,1,x)+n;    % Signal to be cancelled (desired)
b  = fir1(31,0.5);       % FIR secondary path system model
mu = 0.008;              % Filtered-X LMS step size
ha = adaptfilt.filtxlms(32,mu,1,b);
[y,e] = filter(ha,x,d);
plot(1:1000,d,'b',1:1000,e,'r');
title('Active Noise Control of a Random Noise Signal');
legend('Original','Attenuated');
xlabel('Time Index'); ylabel('Signal Value');  grid on;
```



See also

`adaptfilt.dlms`, `adaptfilt.lms`

Reference

J.J. Shynk, “Frequency-Domain and Multirate Adaptive Filtering,” IEEE Signal Processing Magazine, vol. 9, no. 1, pp. 14-37, Jan. 1992.

Purpose Construct a fast transversal least squares adaptive filter object

Syntax

```
ha = adaptfilt.ftf(l,lambda,delta,gamma,gstates,coeffs,states)
```

Description

`ha = adaptfilt.ftf(l,lambda,delta,gamma,gstates,coeffs,states)` constructs a fast transversal least squares adaptive filter object `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.ftf`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`lambda`	RLS forgetting factor. This is a scalar that should lie in the range (1-0.5/`l`, 1]. `lambda` defaults to 1.
`delta`	Soft-constrained initialization factor. This scalar should be positive and sufficiently large to prevent an excessive number of Kalman gain rescues. `delta` defaults to one.
`gamma`	Conversion factor. `gamma` defaults to one specifying soft-constrained initialization.
`gstates`	States of the Kalman gain updates. `gstates` defaults to a zero vector of length `l`.
`coeffs`	Length `l` vector of initial filter coefficients. coeffs defaults to a length `l` vector of zeros.
`states`	Vector of initial filter States. `states` defaults to a zero vector of length (`l`-1).

Properties

Since your `adaptfilt.ftf` filter is an object, it has properties that define its operating behavior. Note that many of the properties are also input arguments for creating `adaptfilt.ftf` objects. To show you the properties that apply, this table lists and describes each property for the fast transversal least squares filter object.

Name	Range	Description
`Algorithm`	None	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	Vector of elements	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.
`States`	Vector of elements, data type double	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to (`l` + `projectord` - 2).
`ForgettingFactor`		RLS forgetting factor. This is a scalar that should lie in the range (1-0.5/`l`, 1]. `lambda` defaults to 1.
`InitFactor`		Soft-constrained initialization factor. This scalar should be positive and sufficiently large to prevent an excessive number of Kalman gain rescues. `delta` defaults to one.

Name	Range	Description
FwdPrediction		Contains the predicted values for samples during adaptation. Compare these to the actual samples to get the error and power.
BkwdPrediction		Returns the predicted samples generated during adaptation.Refer to [12] in the bibliography for details about linear prediction.
KalmanGain		Empty when you construct the object, this gets populated after you run the filter.
ConversionFactor		Conversion factor. Called gamma when it is an input argument, it defaults to the matrix [1 -1] that specifies soft-constrained initialization.

Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

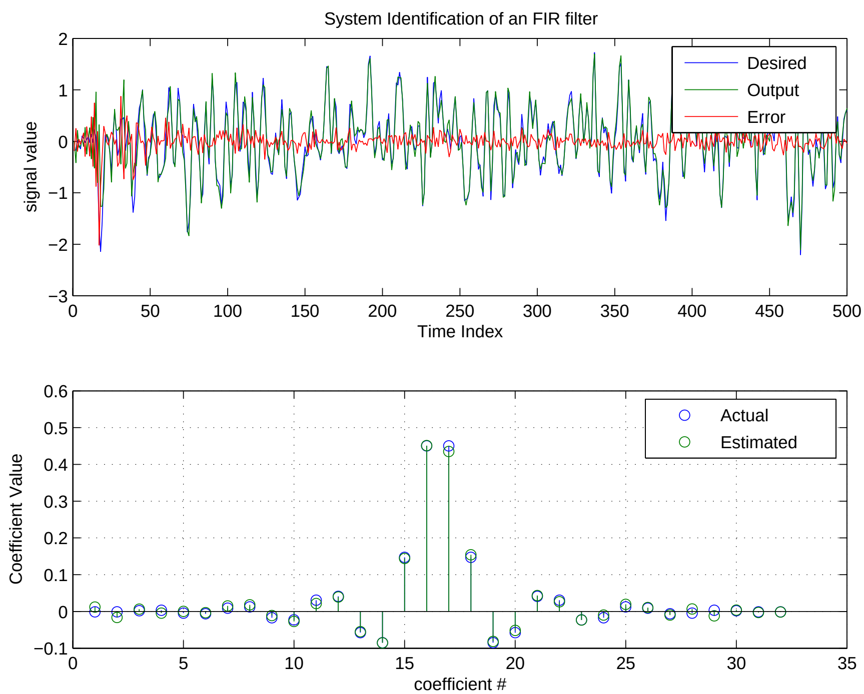
Examples

System Identification of a 32-coefficient FIR filter by running the identification process for 500 iterations.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
N  = 31;               % Adaptive filter order
lam = 0.99;            % RLS forgetting factor
del = 0.1;             % Soft-constrained initialization factor
ha = adaptfilt.ftf(32,lam,del);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('signal value');
subplot(2,1,2); stem([b.',ha.Coefficients.']);
legend('Actual','Estimated');
```

```
xlabel('coefficient #'); ylabel('Coefficient Value'); grid on;
```

For this example of identifying an unknown system, the figure shows that the adaptation process identifies the filter coefficients for the unknown FIR filter within the first 150 iterations.



See Also

`adaptfilt.swftf`, `adaptfilt.rls`, `adaptfilt.lsl`

Reference

D.T.M. Slock and Kailath, T., "Numerically Stable Fast Transversal Filters for Recursive Least Squares Adaptive Filtering," IEEE Trans. Signal Processing, vol. 38, no. 1, pp. 92-114.

adaptfilt.gal

Purpose

Construct a gradient adaptive lattice FIR filter

Syntax

```
ha = adaptfilt.gal(l,step,leakage,offset,rstep,delta,lambda,
   rcoeffs,coeffs,states)
```

Description

`ha = adaptfilt.gal(l,step,leakage,offset,rstep,delta,lambda, rcoeffs,coeffs,states)` constructs a gradient adaptive lattice FIR filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.gal`.

Input Argument	Description
`l`	Length of the joint process filter coefficients. It must be a positive integer and must be equal to the length of the reflection coefficients plus one. `l` defaults to 10.
`step`	Joint process step size of the adaptive filter. This scalar should be a value between zero and one. `step` defaults to 0.
`leakage`	Leakage factor of the adaptive filter. It must be a scalar between 0 and 1. Setting leakage less than one implements a leaky algorithm to estimate both the reflection and the joint process coefficients. `leakage` defaults to 1 (no leakage).
`offset`	Specifies an optional offset for the denominator of the step size normalization term. It must be a scalar greater or equal to zero. A non-zero `offset` is useful to avoid divide-by-near-zero conditions when the input signal amplitude becomes very small. `offset` defaults to 1.

Input Argument	Description
rstep	Reflection process step size of the adaptive filter. This scalar should be a value between zero and one. rstep defaults to step.
delta	Initial common value of the forward and backward prediction error powers. It should be a positive value. 0.1 is the default value for delta.
lambda	Specifies the averaging factor used to compute the exponentially windowed forward and backward prediction error powers for the coefficient updates. lambda should lie in the range (0, 1]. lambda defaults to the value (1 - step).
rcoeffs	Vector of initial reflection coefficients. It should be a length (l-1) vector. rcoeffs defaults to a zero vector of length (l-1).
coeffs	Vector of initial joint process filter coefficients. It must be a length l vector. coeffs defaults to a length l vector of zeros.
states	Vector of the backward prediction error states of the adaptive filter. states defaults to a zero vector of length (l-1).

Properties

Since your adaptfilt.gal filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.gal objects. To show you the properties that apply, this table lists and describes each property for the affine projection filter object.

Name	Range	Description
`Algorithm`	None	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	Vector of elements	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.
`States`	Vector of elements	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to (`l` + `projectord` - 2).
`StepSize`	0 to 1	Specifies the step size taken between filter coefficient updates
`Leakage`	0 to 1	Leakage parameter of the adaptive filter. If this parameter is set to a value between zero and one, you implement a leaky GAL algorithm. `leakage` defaults to 1—no leakage provided in the algorithm.

Name	Range	Description
Offset		Offset for the normalization terms in the coefficient updates. Use this to avoid dividing by zero or by very small numbers when input signal amplitude becomes very small. offset defaults to one.
ReflectionCoeffs		
FwdPredErrorPower		Returns the minimum mean-squared prediction error in the forward direction. Refer to [12] in the bibliography for details about linear prediction.
BkwdPredErrorPower		Returns the minimum mean-squared prediction error. Refer to [12] in the bibliography for details about linear prediction
ReflectionCoeffsStep		
AvgFactor		Specifies the averaging factor used to compute the exponentially-windowed forward and backward prediction error powers for the coefficient updates. Same as the input argument lambda.

Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

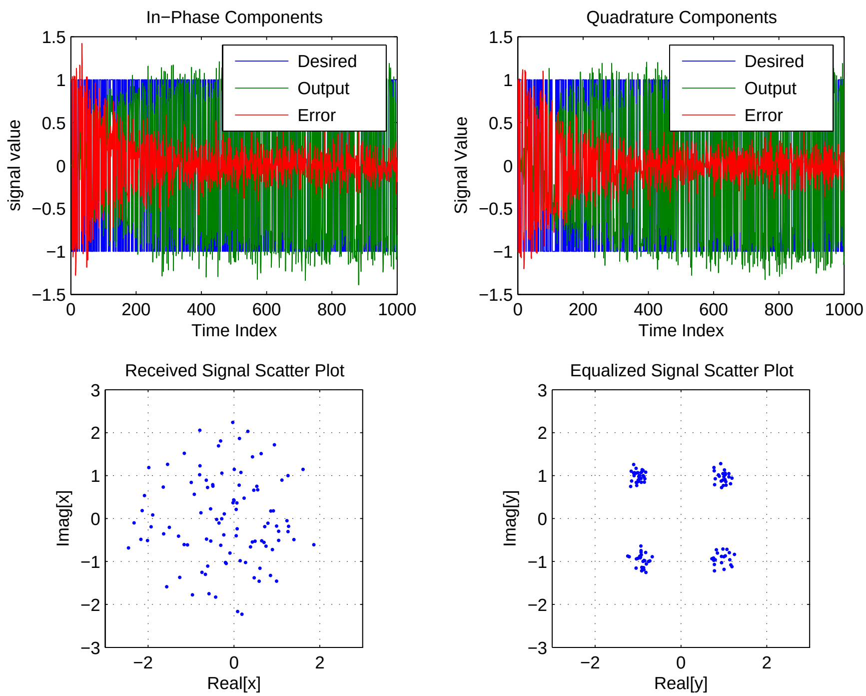
Examples

Perform a Quadrature Phase Shift Keying (QPSK) adaptive equalization using a 32-coefficient adaptive filter over 1000 iterations.

```
D  = 16;                          % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];     % Numerator coefficients of channel
a  = [1 -0.7];               % Denominator coefficients of channel
ntr= 1000;                        % Number of iterations
s  = sign(randn(1,ntr+D)) + j*sign(randn(1,ntr+D)); % Baseband
                                  % QPSK signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));      % Noise signal
r  = filter(b,a,s)+n;             % Received signal
x  = r(1+D:ntr+D);              % Input signal (received signal)
d  = s(1:ntr);               % Desired signal (delayed QPSK signal)
L = 32;                         % filter length
mu = 0.007;                     % Step size
ha = adaptfilt.gal(L,mu);
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
```

```
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('signal value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```

To see the results, look at this figure.



See Also

`adaptfilt.qrdlsl`, `adaptfilt.lsl`, `adaptfilt.tdafdft`

References

L.J. Griffiths, "A Continuously Adaptive Filter Implemented as a Lattice Structure," Proc. IEEE Int. Conf. on Acoustics, Speech, and Signal Processing, Hartford, CT, pp. 683-686, 1977

S. Haykin, *Adaptive Filter Theory*, 3rd Ed., Upper Saddle River, NJ, Prentice Hall, 1996

Purpose

Construct a householder recursive least squares (RLS) FIR adaptive filter object

Syntax

```
ha = adaptfilt.hrls(l,lambda,sqrtinvcov,coeffs,states)
```

Description

`ha = adaptfilt.hrls(l,lambda,sqrtinvcov,coeffs,states)` constructs an FIR householder RLS adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.hrls`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`lambda`	RLS forgetting factor. This is a scalar and should lie in the range (0, 1]. `lambda` defaults to 1 meaning the adaptation process retains infinite memory.
`sqrtinvcov`	Square-root of the inverse of the sliding window input signal covariance matrix. This square matrix should be full-ranked.
`coeffs`	Vector of initial filter coefficients. It must be a length `l` vector. `coeffs` defaults to being a length `l` vector of zeros.
`states`	Vector of initial filter states. It must be a length `l-1` vector. `states` defaults to a length `l-1` vector of zeros.

Properties

Since your `adaptfilt.hrls` filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating `adaptfilt.hrls` objects. To show you the properties that apply, this table lists and describes each property for the affine projection filter object.

Name	Range	Description
`Algorithm`	None	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	Vector of elements	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.
`States`	Vector of elements, data type double	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to (`l` - 1).
`ForgettingFactor`	Scalar	RLS forgetting factor. This is a scalar and should lie in the range (0, 1]. Same as input argument `lambda`. It defaults to 1 meaning the adaptation process retains infinite memory.
`KalmanGain`	Vector of size (`l`,1)	Empty when you construct the object, this gets populated after you run the filter.

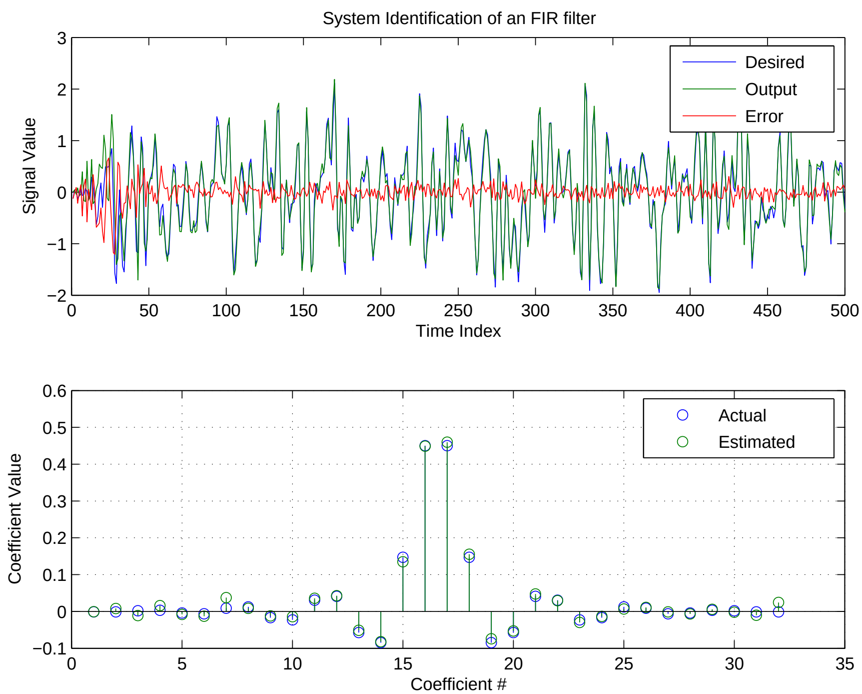
Name	Range	Description
SqrtInvCov	Matrix of doubles	Square root of the inverse of the sliding window input signal covariance matrix. This square matrix should be full-ranked.
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

Examples

Use 500 iterations of an adaptive filter object to identify a 32-coefficient FIR filter system. Both the example code and the resulting figure show the successful filter identification through adaptive filter processing.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
G0 = sqrt(10)*eye(32); % Initial sqrt correlation matrix inverse
lam = 0.99;            % RLS forgetting factor
ha = adaptfilt.hrls(32,lam,G0);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
```

```
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.Coefficients.']);
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```



See Also

`adaptfilt.rls`, `adaptfilt.qrdrls`, `adaptfilt.hswrls`

Purpose

Construct a householder sliding window RLS FIR adaptive filter

Syntax

```
ha = adaptfilt.hswrls(l, lambda, sqrtinvcov, swblocklen,
   dstates, coeffs, states)
```

Description

`ha = adaptfilt.hswrls(l, lambda, sqrtinvcov, swblocklen, dstates, coeffs, states)` constructs an FIR householder sliding window recursive-least-square adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.hswrls`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`lambda`	Recursive least square (RLS) forgetting factor. This is a scalar and should lie in the range (0, 1]. `lambda` defaults to 1 meaning the adaptation process retains infinite memory.
`sqrtinvcov`	Square-root of the inverse of the sliding window input signal covariance matrix. This square matrix should be full-ranked.
`swblocklen`	Block length of the sliding window. This integer must be at least as large as the filter length. `swblocklen` defaults to 16.
`dstates`	Desired signal states of the adaptive filter. `dstates` defaults to a zero vector with length equal to (`swblocklen` - 1).

Input Argument	Description
coeffs	Vector of initial filter coefficients. It must be a length `l` vector. `coeffs` defaults to being a length `l` vector of zeros.
states	Vector of initial filter states. It must be a length `(l + swblocklen -2)` vector. `states` defaults to a length `(l + swblocklen -2)` vector of zeros.

Properties

Since your `adaptfilt.hswrls` filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating `adaptfilt.hswrls` objects. To show you the properties that apply, this table lists and describes each property for the affine projection filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.

Name	Range	Description
States	Vector of elements, data type double	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (1 + projectord - 2).
ForgettingFactor	Scalar	Root-least-square (RLS) forgetting factor. This is a scalar and should lie in the range (0, 1]. Same as input argument lambda. It defaults to 1 meaning the adaptation process retains infinite memory.
KalmanGain	(l,1) vector	Empty when you construct the object, this gets populated after you run the filter.
SqrtInvCov	l-by-l Matrix	Square-root of the inverse of the sliding window input signal covariance matrix. This square matrix should be full-ranked.
SwBlockLength	Integer	Block length of the sliding window. This integer must be at least as large as the filter length. swblocklen defaults to 16.
DesiredSignalStates	Vector	Desired signal states of the adaptive filter. dstates defaults to a zero vector with length equal to (swblocklen - 1).

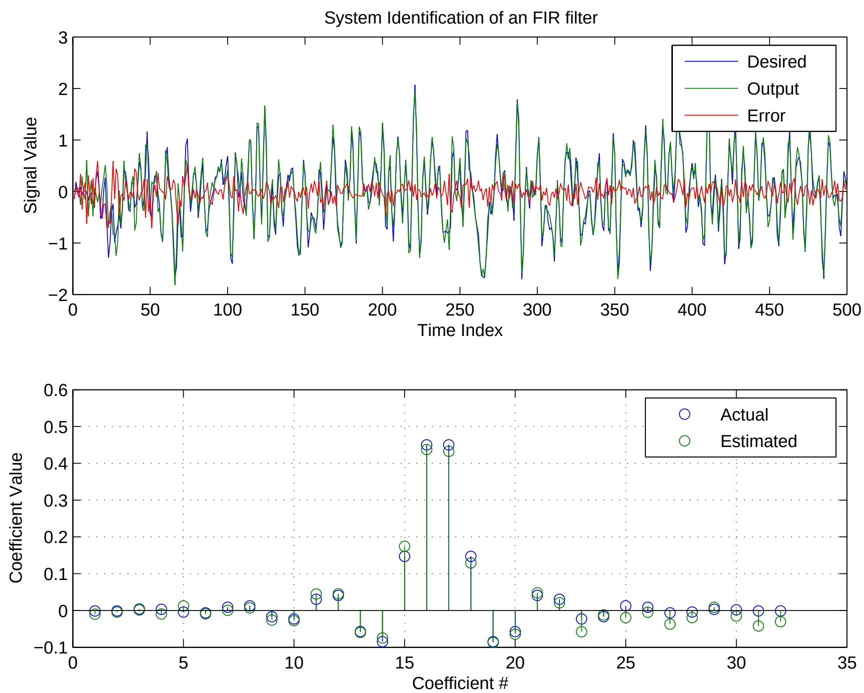
Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

Examples

System Identification of a 32-coefficient FIR filter.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
G0 = sqrt(10)*eye(32); % Initial sqrt correlation matrix inverse
lam = 0.99;            % RLS forgetting factor
N  = 64;               % block length
ha = adaptfilt.hswrls(32,lam,G0,N);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.Coefficients.']);
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```

In the pair of plots shown in the figure you see the comparison of the desired and actual output for the adapting filter and the coefficients of both filters, the unknown and the adapted.



See Also

`adaptfilt.rls`, `adaptfilt.qrdrls`, `adaptfilt.hrls`

Purpose

Construct a least-mean-square (LMS) FIR adaptive filter object

Syntax

```
ha = adaptfilt.lms(l,step,leakage,coeffs,states)
```

Description

`ha = adaptfilt.lms(l,step,leakage,coeffs,states)` constructs an FIR LMS adaptive filter object `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.lms`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	LMS step size. It must be a nonnegative scalar. You can use `maxstep` to determine a reasonable range of step size values for the signals being processed. `step` defaults to 0.1.
`leakage`	Your LMS leakage factor. It must be a scalar between 0 and 1. When `leakage` is less than one, `adaptfilt.lms` implements a leaky LMS algorithm. When you omit the `leakage` property in the calling syntax, it defaults to 1 providing no leakage in the adapting algorithm.
`coeffs`	Vector of initial filter coefficients. it must be a length `l` vector. `coeffs` defaults to length `l` vector with elements equal to zero.
`states`	Vector of initial filter states for the adaptive filter. It must be a length `l-1` vector. `states` defaults to a length l-1 vector of zeros.

Properties

In the syntax for creating the `adaptfilt` object, the input options are properties of the object created. This table list all the properties for the adaptfilt.lms object, their default values, and a brief description of the property.

Property	Range	Property Description
Algorithm	None	Reports the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector where l is the number of filter coefficients. coeffs defaults to a length l vector of zeros when you do not provide the vector as an input argument.
States	Vector of elements, data type double	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l - 1).
StepSize	0 to 1	LMS step size. It must be a scalar between zero and one. Setting this step size value to one provides the fastest convergence. step defaults to 0.1.
Leakage	0 to 1	LMS leakage factor. It must be a scalar between zero and one. When it is less than one, a leaky NLMS algorithm results. leakage defaults to 1 (no leakage).

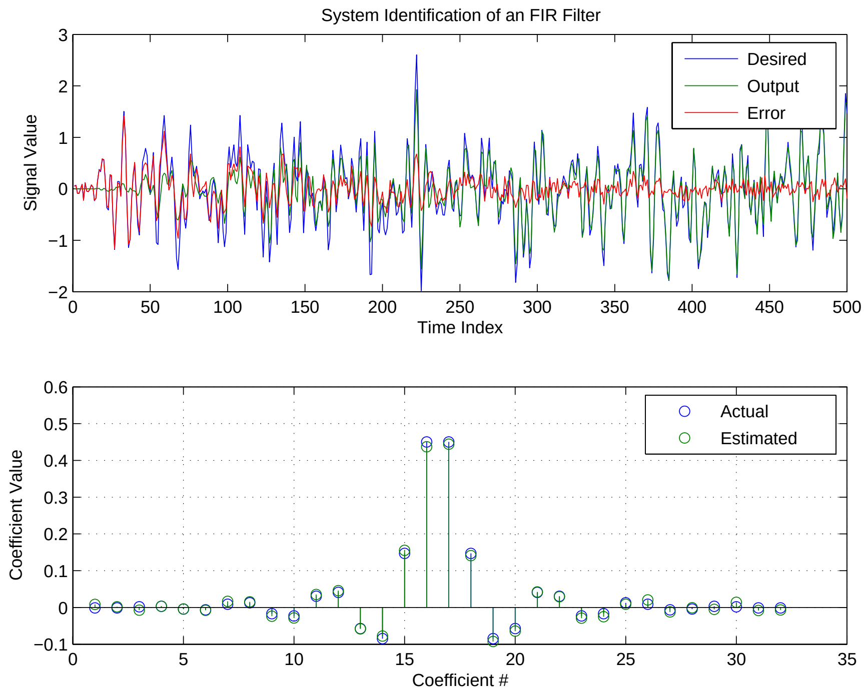
Property	Range	Property Description
PersistentMemory	false or true	Determine whether the filter states and coefficients get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any property value that the filter changes during processing. Property values that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

Example

Use 500 iterations of an adapting filter system to identify and unknown 32nd-order FIR filter.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
mu = 0.008;            % LMS step size.
ha = adaptfilt.lms(32,mu);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.coefficients.']);
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```

Using LMS filters in an adaptive filter architecture is a time honored means for identifying an unknown filter. By running the example code provided you can demonstrate one process to identify an unknown FIR filter.



See Also

`adaptfilt.blms`, `adaptfilt.blmsfft`, `adaptfilt.dlms`, `adaptfilt.nlms`, `adaptfilt.tdafdft`, `adaptfilt.sd`, `adaptfilt.se`, `adaptfilt.ss`

Reference

J.J. Shynk, “Frequency-Domain and Multirate Adaptive Filtering,” IEEE Signal Processing Magazine, vol. 9, no. 1, pp. 14-37, Jan. 1992.

adaptfilt.lsl

Purpose

Construct a least squares lattice (LSL) adaptive filter

Syntax

```
ha = adaptfilt.lsl(l,lambda,delta,coeffs,states)
```

Description

`ha = adaptfilt.lsl(l,lambda,delta,coeffs,states)` constructs a least squares lattice adaptive filter ha.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.lsl`.

Input Argument	Description
`l`	Length of the joint process filter coefficients. It must be a positive integer and must be equal to the length of the prediction coefficients plus one. L defaults to 10.
`lambda`	Forgetting factor of the adaptive filter. This is a scalar and should lie in the range (0, 1]. `lambda` defaults to 1. `lambda` = 1 denotes infinite memory while adapting to find the new filter.
`delta`	Soft-constrained initialization factor in the least squares lattice algorithm. It should be positive. `delta` defaults to 1.
`coeffs`	Vector of initial joint process filter coefficients. It must be a length `l` vector. `coeffs` defaults to a length `l` vector of all zeros.
`states`	Vector of the backward prediction error states of the adaptive filter. `states` defaults to a length `l` vector of all zeros, specifying soft-constrained initialization for the algorithm.

Properties

Since your `adaptfilt.lsl` filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating `adaptfilt.lsl` objects. To show you the properties that apply, this table lists and describes each property for the filter object.

Name	Range	Description
`Algorithm`	None	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	Vector of elements	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.
`States`	Vector of elements, data type double	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to `l`
`ForgettingFactor`		Forgetting factor of the adaptive filter. This is a scalar and should lie in the range (0, 1]. It defaults to 1. Setting `forgetting factor = 1` denotes infinite memory while adapting to find the new filter. Note that this is the `lambda` input argument.

Name	Range	Description
InitFactor		Soft-constrained initialization factor. This scalar should be positive and sufficiently large to prevent an excessive number of Kalman gain rescues. delta defaults to one.
FwdPrediction		
BkwdPrediction		Returns the predicted samples generated during adaptation.Refer to [12] in the bibliography for details about linear prediction.

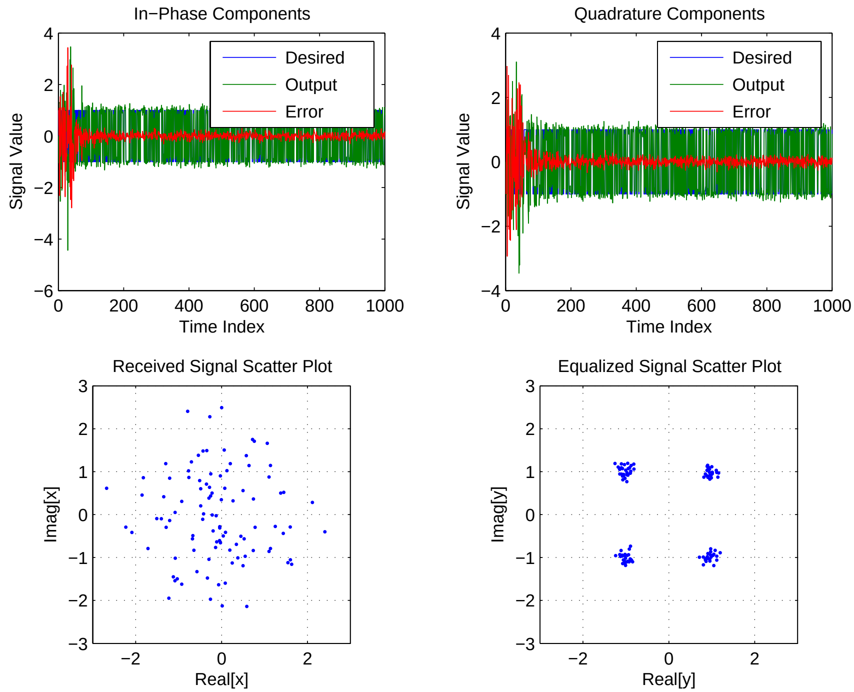
Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. As a check, the number of samples reported processed plus the number of nonprocessed samples should be the total number of input samples. Defaults to zero.

Examples

Demonstrate Quadrature Phase Shift Keying (QPSK) adaptive equalization using a 32-coefficient adaptive filter running for 1000 iterations. After you review the example code, the figure shows the results of running the example to use QPSK adaptive equalization with a 32nd-order FIR filter. Notice that the error between the in-phase and quadrature components, as shown by the errors plotted in the upper plots, falls to near zero. Also, the equalized signal shows the clear quadrature nature.

```
D  = 16;                          % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];      % Numerator coefficients of channel
a  = [1 -0.7];                 % Denominator coefficients of channel
ntr= 1000;                        % Number of iterations
```

```
s  = sign(randn(1,ntr+D)) + j*sign(randn(1,ntr+D));% Baseband
                                                   % QPSK signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));     % Noise signal
r  = filter(b,a,s)+n;        % Received signal
x  = r(1+D:ntr+D);           % Input signal (received signal)
d  = s(1:ntr);                 % Desired signal (delayed QPSK
                                % signal)
lam = 0.995;                    % Forgetting factor
del = 1;                     % Soft-constrained initialization
factor
ha = adaptfilt.lsl(32,lam,del);
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```



See Also

`adaptfilt.qrdlsl`, `adaptfilt.gal`, `adaptfilt.ftf`, `adaptfilt.rls`

References

S. Haykin, *Adaptive Filter Theory*, 2nd Edition, Prentice Hall, N.J., 1991

adaptfilt.nlms

Purpose Construct a normalized LMS FIR adaptive filter object

Syntax `ha = adaptfilt.nlms(l,step,leakage,offset,coeffs,states)`

Description `ha = adaptfilt.nlms(l,step,leakage,offset,coeffs,states)` constructs a normalized least-mean squares (NLMS) FIR adaptive filter object named `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.nlms`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	NLMS step size. It must be a scalar between 0 and 2. Setting this step size value to one provides the fastest convergence. `step` defaults to 1.
`leakage`	NLMS leakage factor. It must be a scalar between zero and one. When it is less than one, a leaky NLMS algorithm results. `leakage` defaults to 1 (no leakage).
`offset`	Specifies an optional offset for the denominator of the step size normalization term. You must specify offset to be a scalar greater than or equal to zero. Nonzero offsets can help avoid a divide-by-near-zero condition that causes errors. Use this to avoid dividing by zero (or by very small numbers) when the square of the input data norm becomes very small (when the input signal amplitude becomes very small). When you omit it, `offset` defaults to zero.

Input Argument	Description
`coeffs`	Vector composed of your initial filter coefficients. Enter a length `l` vector. `coeffs` defaults to a vector of zeros with length equal to the filter order.
`states`	Your initial adaptive filter states appear in the `states` vector. It must be a vector of length `l`-1. `states` defaults to a length `l`-1 vector with zeros for all of the elements.

Properties

In the syntax for creating the `adaptfilt` object, the input options are properties of the object you create. This table list all the properties for normalized LMS objects, their default values, and a brief description of the property.

Property	Range	Property Description
`Algorithm`	None	Reports the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	Vector of elements	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.

Property	Range	Property Description
`States`	Vector of elements, data type double	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to (`l` - 1).
`StepSize`	0 to 1	NLMS step size. It must be a scalar between zero and one. Setting this step size value to one provides the fastest convergence. `step` defaults to one.
`Leakage`	0 to 1	NLMS leakage factor. It must be a scalar between zero and one. When it is less than one, a leaky NLMS algorithm results. `leakage` defaults to 1 (no leakage).
`Offset`	0 or greater	Specifies an optional offset for the denominator of the step size normalization term. You must specify offset to be a scalar greater than or equal to zero. Nonzero offsets can help avoid a divide-by-near-zero condition that causes errors. Use this to avoid dividing by zero (or by very small numbers) when the square of the input data norm becomes very small (when the input signal amplitude becomes very small). When you omit it, `offset` defaults to zero.

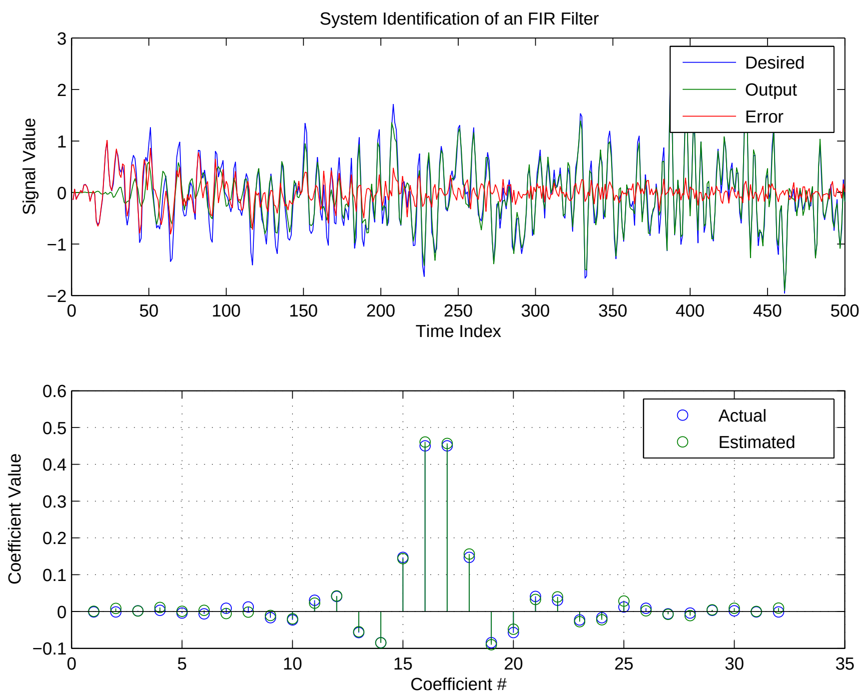
Property	Range	Property Description
PersistentMemory	false or true	Determine whether the filter states and coefficients get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any property value that the filter changes during processing. Property values that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

Example

To help you compare this algorithm's performance to other LMS-based algorithms, such as BLMS or LMS, this example demonstrates the NLMS adaptive filter in use to identify the coefficients of an unknown FIR filter of order equal to 32—an example used in other adaptive filter examples.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
mu = 1;                % NLMS step size
offset = 50;           % NLMS offset
ha = adaptfilt.nlms(32,mu,1,offset);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.coefficients.']);
```

```
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```



As you see from the figure, the nlms variant again closely matches the actual filter coefficients in the unknown FIR filter.

See Also

`adaptfilt.ap`, `adaptfilt.apru`, `adaptfilt.lms`, `adaptfilt.rls`, `adaptfilt.swrls`

Purpose

Construct a partitioned block frequency-domain (PBFDAF) FIR adaptive filter that includes binned step size normalization

Syntax

```
ha = adaptfilt.pbfdaf(l,step,leakage,delta,lambda,blocklen,offset,
   coeffs,states)
```

Description

`ha = adaptfilt.pbfdaf(l,step,leakage,delta,lambda,blocklen,offset, coeffs,states)` constructs a partitioned block frequency-domain FIR adaptive filter `ha` that uses bin step size normalization during adaptation.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.pbfdaf`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. L defaults to 10.
`step`	Step size of the adaptive filter. This is a scalar and should lie in the range (0,1]. `step` defaults to 1.
`leakage`	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, a leaky version of the PBFDAF algorithm is implemented. `leakage` defaults to 1— no leakage.
`delta`	Initial common value of all of the FFT input signal powers. Its initial value should be positive. delta defaults to 1.
`lambda`	Averaging factor used to compute the exponentially windowed FFT input signal powers for the coefficient updates. `lambda` should lie in the range (0,1]. `lambda` defaults to 0.9.

Input Argument	Description
blocklen	Block length for the coefficient updates. This must be a positive integer such that (l/blocklen) is also an integer. For faster execution, blocklen should be a power of two. blocklen defaults to two.
offset	Offset for the normalization terms in the coefficient updates. This can be useful to avoid divide by zeros conditions, or dividing by very small numbers, if any of the FFT input signal powers become very small. offset defaults to zero.
coeffs	Initial time-domain coefficients of the adaptive filter. It should be a vector of length l. The PBFDAF algorithm uses these coefficients to compute the initial frequency-domain filter coefficient matrix via FFTs.
states	Specifies the filter initial conditions. states defaults to a zero vector of length l.

Properties

Since your adaptfilt.pbfdaf filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.pbfdaf objects. To show you the properties that apply, this table lists and describes each property for the filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps

Name	Range	Description
StepSize	0 to 1	Step size of the adaptive filter. This is a scalar and should lie in the range (0,1]. step defaults to 1.
Leakage	0 to 1	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, a leaky version of the PBFDAF algorithm is implemented. leakage defaults to 1— no leakage.
Power		A vector of 2*l elements, each initialized with the value delta from the input arguments. As you filter data, Power gets updated by the filter process.
AvgFactor		Averaging factor used to compute the exponentially windowed FFT input signal powers for the coefficient updates. AvgFactor should lie in the range (0,1]. AvgFactor defaults to 0.9. Called lambda as an input argument.

Name	Range	Description
BlockLength		Block length for the coefficient updates. This must be a positive integer such that (l/blocklen) is also an integer. For faster execution, blocklen should be a power of two. blocklen defaults to two.
Offset		Offset for the normalization terms in the coefficient updates. This can be useful to avoid divide by zeros conditions, or dividing by very small numbers, if any of the FFT input signal powers become very small. offset defaults to zero.
FFTCoefficients		Stores the discrete Fourier transform of the filter coefficients in coeffs.
FFTStates		States for the FFT operation.

Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

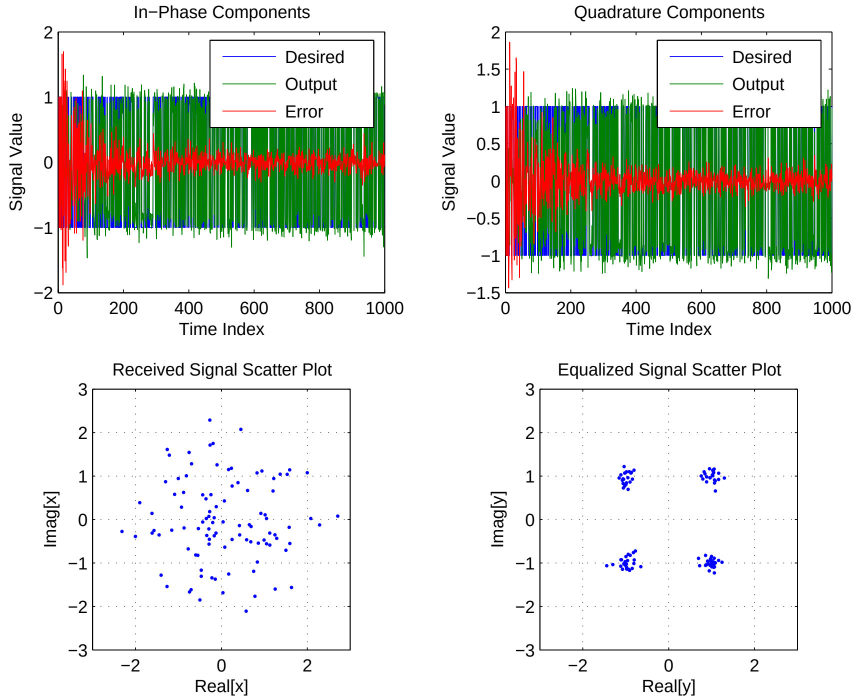
Examples

An example of Quadrature Phase Shift Keying (QPSK) adaptive equalization using a 32-coefficient FIR filter.

```
D  = 16;                          % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];    % Numerator coefficients of channel
a  = [1 -0.7];              % Denominator coefficients of channel
ntr = 1000;                       % Number of iterations
s  = sign(randn(1,ntr+D))+j*sign(randn(1,ntr+D)); % Baseband
                                                  % QPSK signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));     % Noise signal
      r  = filter(b,a,s)+n;    % Received signal
x  = r(1+D:ntr+D);                % Input signal (received signal)
d  = s(1:ntr);              % Desired signal (delayed QPSK signal)
del = 1;                           % Initial FFT input powers
mu  = 0.1;                         % Step size
lam = 0.9;                         % Averaging factor
N   = 8;                           % Block size
ha = adaptfilt.pbfdaf(32,mu,1,del,lam,N);
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
```

```
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```

In the figure shown, the four subplots provide the details of the results of the QPSK process used in the equalization for this example.



See Also

`adaptfilt.fdaf`, `adaptfilt.pbufdaf`, `adaptfilt.blmsfft`

References

J.S. So and K.K. Pang, "Multidelay Block Frequency Domain Adaptive Filter," IEEE Trans. Acoustics, Speech, and Signal Processing, vol. 38, no. 2, pp. 373-376, February 1990

J.M. Paez Borrallo and M.G. Otero, "On The Implementation of a Partitioned Block Frequency Domain Adaptive Filter (PBFDAF) For Long Acoustic Echo Cancellation," Signal Processing, vol. 27, no. 3, pp. 301-315, June 1992

adaptfilt.pbufdaf

Purpose

Construct a partitioned block unconstrained frequency-domain (PBUFDAF) FIR adaptive filter with binned step size normalization

Syntax

```
ha = adaptfilt.pbufdaf(l,step,leakage,delta,lambda,blocklen,
   offset,coeffs,states)(
```

Description

`ha = adaptfilt.pbufdaf(l,step,leakage,delta,lambda,blocklen, offset,coeffs,states)` constructs a partitioned block unconstrained frequency-domain FIR adaptive filter `ha` with bin step size normalization.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.pbufdaf`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. L defaults to 10.
`step`	Step size of the adaptive filter. This is a scalar and should lie in the range (0,1]. `step` defaults to 1.
`leakage`	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, a leaky version of the PBFDAF algorithm is implemented. `leakage` defaults to 1— no leakage.
`delta`	Initial common value of all of the FFT input signal powers. Its initial value should be positive. delta defaults to 1.
`lambda`	Averaging factor used to compute the exponentially windowed FFT input signal powers for the coefficient updates. `lambda` should lie in the range (0,1]. `lambda` defaults to 0.9.

Input Argument	Description
blocklen	Block length for the coefficient updates. This must be a positive integer such that (l/blocklen) is also an integer. For faster execution, blocklen should be a power of two. blocklen defaults to two.
offset	Offset for the normalization terms in the coefficient updates. This can be useful to avoid divide by zeros conditions, or dividing by very small numbers, if any of the FFT input signal powers become very small. offset defaults to zero.
coeffs	Initial time-domain coefficients of the adaptive filter. It should be a vector of length l. The PBFDAF algorithm uses these coefficients to compute the initial frequency-domain filter coefficient matrix via FFTs.
states	Specifies the filter initial conditions. states defaults to a zero vector of length l.

Properties

Since your adaptfilt.pbufdaf filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.pbufdaf objects. To show you the properties that apply, this table lists and describes each property for the filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps
StepSize	0 to 1	Step size of the adaptive filter. This is a scalar and should lie in the range (0,1]. step defaults to 1.
Leakage	0 to 1	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, a leaky version of the PBFDAF algorithm is implemented. leakage defaults to 1— no leakage.
Power	2*l element vector	A vector of 2*l elements, each initialized with the value delta from the input arguments. As you filter data, Power gets updated by the filter process.

Name	Range	Description
AvgFactor		Averaging factor used to compute the exponentially windowed FFT input signal powers for the coefficient updates. AvgFactor should lie in the range (0,1]. AvgFactor defaults to 0.9. Called lambda as an input argument.
BlockLength		Block length for the coefficient updates. This must be a positive integer such that (l/blocklen) is also an integer. For faster execution, blocklen should be a power of two. blocklen defaults to two.
Offset		Offset for the normalization terms in the coefficient updates. This can be useful to avoid divide by zeros conditions, or dividing by very small numbers, if any of the FFT input signal powers become very small. offset defaults to zero.
FFTCoefficients		Stores the discrete Fourier transform of the filter coefficients in coeffs.
FFTStates		States for the FFT operation.

Name	Range	Description
`PersistentMemory`	`false` or `true`	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. `PersistentMemory` returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to `false`.
`NumSamplesProcessed`	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

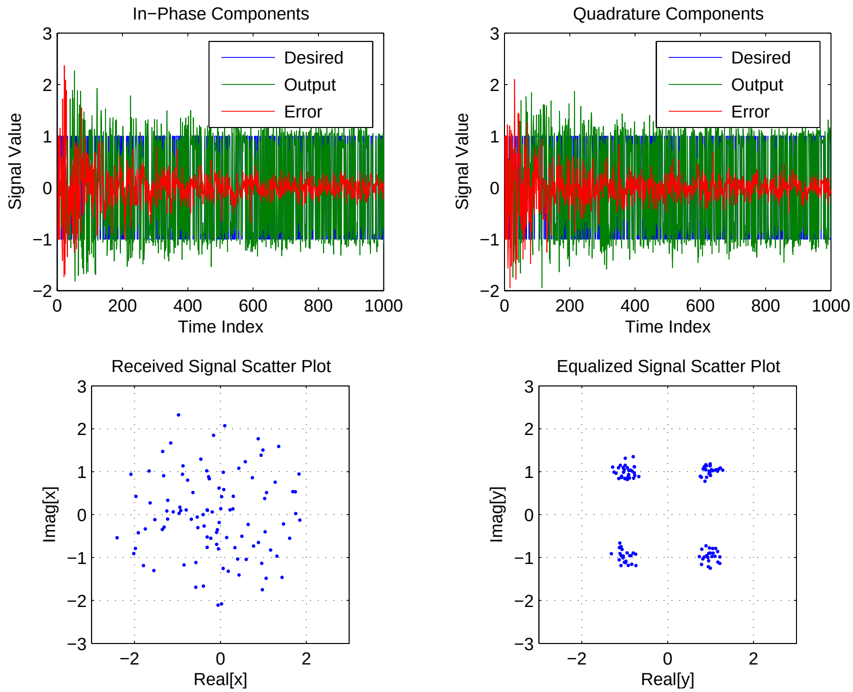
Examples

Demonstrating Quadrature Phase Shift Keying (QPSK) adaptive equalization using a 32-coefficient FIR filter. To perform the equalization, this example runs for 1000 iterations.

```
D  = 16;                      % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];    % Numerator coefficients of channel
a  = [1 -0.7];              % Denominator coefficients of channel
ntr= 1000;                    % Number of iterations
s  = sign(randn(1,ntr+D))+j*sign(randn(1,ntr+D)); % Baseband QPSK
                                                   % signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D)); % Noise signal
r  = filter(b,a,s)+n;         % Received signal
x  = r(1+D:ntr+D);            % Input signal (received signal)
d  = s(1:ntr);              % Desired signal (delayed QPSK signal)
del = 1;                      % Initial FFT input powers
mu  = 0.1;                    % Step size
lam = 0.9;                    % Averaging factor
N   = 8;                      % Block size
ha = adaptfilt.pbufdaf(32,mu,1,del,lam,N);
[y,e] = filter(ha,x,d);
```

```
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```

To allow you to compare this algorithm to another, such as the `pbfdaf` version, we use the same example of QPSK adaptation. The figure shows the results.



See Also

`adaptfilt.ufdaf`, `adaptfilt.pbfdaf`, `adaptfilt.blmsfft`

References

J.S. So and K.K. Pang, "Multidelay Block Frequency Domain Adaptive Filter," IEEE Trans. Acoustics, Speech, and Signal Processing, vol. 38, no. 2, pp. 373-376, February 1990

J.M. Paez Borrallo and M.G. Otero, "On The Implementation of a Partitioned Block Frequency Domain Adaptive Filter (PBFDAF) for Long Acoustic Echo Cancellation," Signal Processing, vol. 27, no. 3, pp. 301-315, June 1992

Purpose Return a QR-decomposition-based least squares lattice adaptive filter object

Syntax

```
ha = adaptfilt.qrdlsl(l,lambda,delta,coeffs,states)
```

Description

`ha = adaptfilt.qrdlsl(l,lambda,delta,coeffs,states)` returns a QR-decomposition-based least squares lattice adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.qrdlsl`.

Input Argument	Description
`l`	Length of the joint process filter coefficients. It must be a positive integer and must be equal to the length of the prediction coefficients plus one. L defaults to 10.
`lambda`	Forgetting factor of the adaptive filter. This is a scalar and should lie in the range (0, 1]. `lambda` defaults to 1. `lambda` = 1 denotes infinite memory while adapting to find the new filter.
`delta`	Soft-constrained initialization factor in the least squares lattice algorithm. It should be positive. `delta` defaults to 1.
`coeffs`	Vector of initial joint process filter coefficients. It must be a length `l` vector. `coeffs` defaults to a length `l` vector of all zeros.
`states`	Vector of the angle normalized backward prediction error states of the adaptive filter

Properties

Since your `adaptfilt.qrdlsl` filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating `adaptfilt.qrdlsl` objects. To show you the properties that apply, this table lists and describes each property for the filter object.

Name	Range	Description
`Algorithm`	None	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	Vector of elements	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.
`States`	Vector of elements, data type double	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to `l -1`
`ForgettingFactor`		Forgetting factor of the adaptive filter. This is a scalar and should lie in the range (0, 1]. It defaults to 1. Setting `forgetting factor = 1` denotes infinite memory while adapting to find the new filter. Note that this is the `lambda` input argument.

Name	Range	Description
InitFactor		Soft-constrained initialization factor. This scalar should be positive and sufficiently large to prevent an excessive number of Kalman gain rescues. delta defaults to one.
FwdPrediction		Returns the predicted samples generated during adaptation in the forward direction.Refer to [12] in the bibliography for details about linear prediction.
BkwdPrediction		Returns the predicted samples generated during adaptation.Refer to [12] in the bibliography for details about linear prediction.

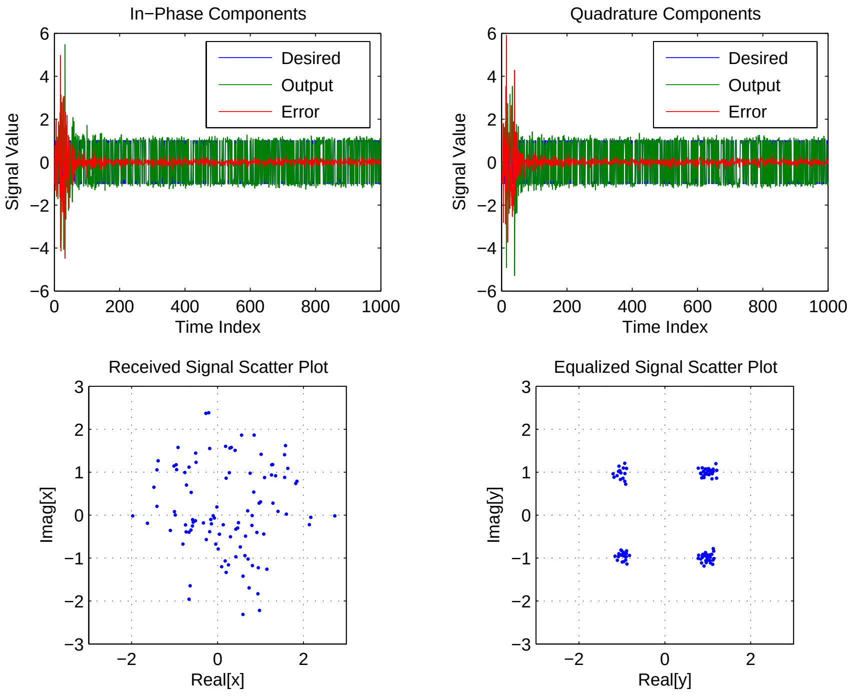
Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

Examples

Implement Quadrature Phase Shift Keying (QPSK) adaptive equalization using a 32-coefficient adaptive filter. To see the results of the equalization process in this example, look at the figure that follows the example code.

```
D  = 16;                        % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];    % Numerator coefficients of channel
a  = [1 -0.7];              % Denominator coefficients of channel
ntr= 1000;                      % Number of iterations
s  = sign(randn(1,ntr+D))+j*sign(randn(1,ntr+D)); % Baseband
QPSK                                 % signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));   % Noise signal
r  = filter(b,a,s)+n;           % Received signal
x  = r(1+D:ntr+D);              % Input signal (received signal)
d  = s(1:ntr);              % Desired signal (delayed QPSK signal)
lam = 0.995;                   % Forgetting factor
del = 1;                       % Soft-constrained initialization
factor
ha = adaptfilt.qrdlsl(32,lam,del);
```

```
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```



See Also `adaptfilt.qrdrls`, `adaptfilt.gal`, `adaptfilt.ftf`, `adaptfilt.lsl`

References

S. Haykin, *Adaptive Filter Theory*, 2nd Edition, Prentice Hall, N.J., 1991

Purpose

Create a QR-decomposition-based RLS FIR adaptive filter object

Syntax

```
ha = adaptfilt.qrdrls(l,lambda,sqrtcov,coeffs,states)
```

Description

`ha = adaptfilt.qrdrls(l,lambda,sqrtcov,coeffs,states)` constructs an FIR QR-decomposition-based recursive-least squares (RLS) adaptive filter object `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.qrdrls`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer.`l` defaults to 10.
`lambda`	RLS forgetting factor. This is a scalar and should lie within the range (0, 1]. `lambda` defaults to 1.
`sqrtcov`	Upper-triangular Cholesky (square root) factor of the input covariance matrix. Initialize this matrix with a positive definite upper triangular matrix.
`coeffs`	Vector of initial filter coefficients. It must be a length `l` vector. `coeffs` defaults to length `l` vector whose elements are zeros.
`states`	Vector of initial filter states. It must be a length `l`-1 vector. `states` defaults to a length `l`-1 vector of zeros.

Properties

Since your `adaptfilt.qrdrls` filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating `adaptfilt.qrdrls` objects. To show you the properties that apply, this table lists and describes each property for the filter object.

Name	Range	Description
`Algorithm`	None	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	Vector of length `l`	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.
`States`	Vector of elements	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to (`l` + `projectord` - 2).
`ForgettingFactor`	Scalar	Forgetting factor of the adaptive filter. This is a scalar and should lie in the range (0, 1]. It defaults to 1. Setting `forgetting factor = 1` denotes infinite memory while adapting to find the new filter. Note that this is the `lambda` input argument.

Name	Range	Description
SqrtCov	Square matrix with each dimension equal to the filter length l	Upper-triangular Cholesky (square root) factor of the input covariance matrix. Initialize this matrix with a positive definite upper triangular matrix.
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

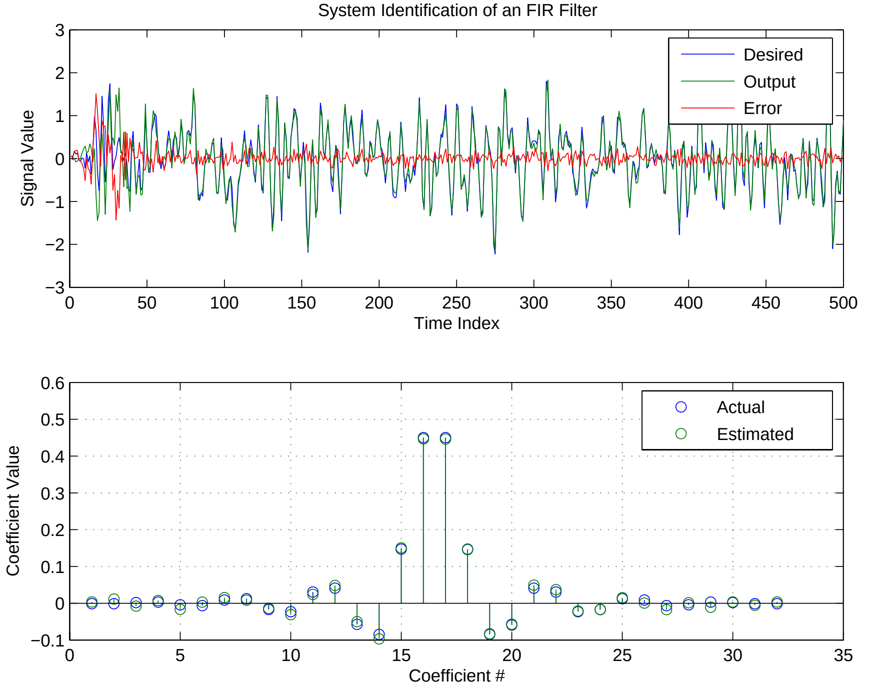
Examples

System Identification of a 32-coefficient FIR filter (500 iterations).

```
x  = randn(1,500);       % Input to the filter
b  = fir1(31,0.5);       % FIR system to be identified
n  = 0.1*randn(1,500);   % Observation noise signal
d  = filter(b,1,x)+n;    % Desired signal
G0 = sqrt(.1)*eye(32);   % Initial sqrt correlation matrix
lam = 0.99;              % RLS forgetting factor
ha = adaptfilt.qrdrls(32,lam,G0);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
```

```
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.Coefficients.']);
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```

Using this variant of the RLS algorithm successfully identifies the unknown FIR filter, as shown here.



See Also

`adaptfilt.rls`, `adaptfilt.hrls`, `adaptfilt.hswrls`, `adaptfilt.swrls`

Purpose

Construct a direct form recursive least squares (RLS) FIR adaptive filter object

Syntax

```
ha = adaptfilt.rls(l,lambda,invcov,coeffs,states)
```

Description

`ha = adaptfilt.rls(l,lambda,invcov,coeffs,states)` constructs an FIR direct form RLS adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.rls`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`lambda`	RLS forgetting factor. This is a scalar and should lie in the range (0, 1]. `lambda` defaults to 1.
`invcov`	Inverse of the input signal covariance matrix. For best performance, you should initialize this matrix to be a positive definite matrix.
`coeffs`	Vector of initial filter coefficients. it must be a length `l` vector. `coeffs` defaults to length `l` vector with elements equal to zero.
`states`	Vector of initial filter states for the adaptive filter. It must be a length `l`-1 vector. `states` defaults to a length l-1 vector of zeros.

Properties

Since your adaptfilt.rls filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.rls objects. To show you the properties that apply, this table lists and describes each property for the filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation.
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps. Remember that filter length is filter order + 1.
Coefficients	Vector containing l elements	Vector containing the initial filter coefficients. It must be a length l vector where l is the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
States	Double array	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).

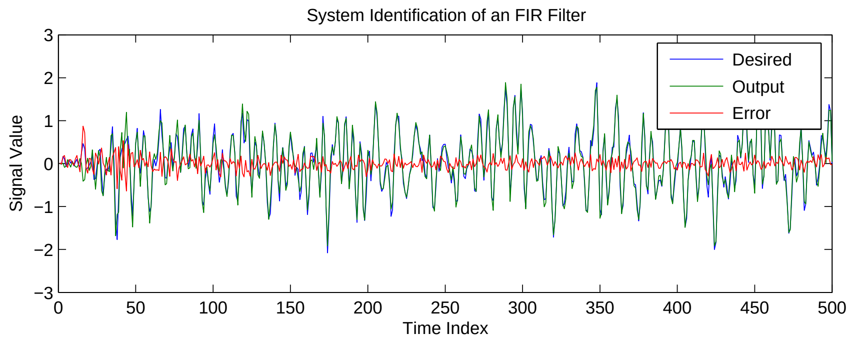
Name	Range	Description
ForgettingFactor	Scalar	Forgetting factor of the adaptive filter. This is a scalar and should lie in the range (0, 1]. It defaults to 1. Setting forgetting factor = 1 denotes infinite memory while adapting to find the new filter. Note that this is the lambda input argument.
KalmanGain	Vector of size (l,1)	Empty when you construct the object, this gets populated after you run the filter.
InvCov	Matrix of size l-by-l	Upper-triangular Cholesky (square root) factor of the input covariance matrix. Initialize this matrix with a positive definite upper triangular matrix.
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

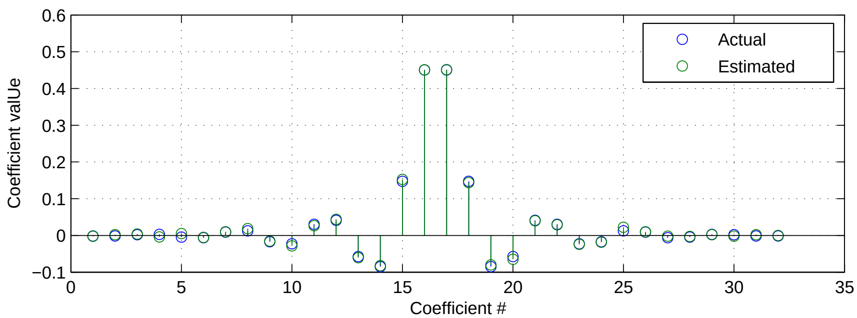
Examples

System Identification of a 32-coefficient FIR filter over 500 adaptation iterations.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
P0 = 10*eye(32); % Initial sqrt correlation matrix inverse
lam = 0.99;            % RLS forgetting factor
ha = adaptfilt.rls(32,lam,P0);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.Coefficients.']);
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient valUe'); grid on;
```

In this example of adaptive filtering using the RLS algorithm to update the filter coefficients for each iteration, the figure shown reveals the fidelity of the derived filter after adaptation.





See Also

`adaptfilt.hrls`,`adaptfilt.hswrls`, `adaptfilt.qrdrls`

adaptfilt.sd

Purpose Construct an FIR adaptive filter object that uses the sign-data algorithm

Syntax

```
ha = adaptfilt.sd(l,step,leakage,coeffs,states)
```

Description

`ha = adaptfilt.sd(l,step,leakage,coeffs,states)` constructs an FIR sign-data adaptive filter object `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.sd`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	SD step size. It must be a nonnegative scalar. `step` defaults to 0.1
`leakage`	Your SD leakage factor. It must be a scalar between 0 and 1. When `leakage` is less than one, `adaptfilt.sd` implements a leaky SD algorithm. When you omit the `leakage` property in the calling syntax, it defaults to 1 providing no leakage in the adapting algorithm.
`coeffs`	Vector of initial filter coefficients. it must be a length `l` vector. `coeffs` defaults to length `l` vector with elements equal to zero.
`states`	Vector of initial filter states for the adaptive filter. It must be a length `l`-1 vector. `states` defaults to a length l-1 vector of zeros.

Properties

In the syntax for creating the `adaptfilt` object, the input options are properties of the object you create. This table list all the properties for sign-data objects, their default values, and a brief description of the property.

Property	Default Value	Description
`Algorithm`	Sign-data	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	10	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	`zeros(1,l)`	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input. Should be initialized with the initial coefficients for the FIR filter prior to adapting. You need `l` entries in `coefficients`.
`States`	`zeros(l-1,1)`	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to (`l` - 1).

Property	Default Value	Description
StepSize	0.1	Sets the SD algorithm step size used for each iteration of the adapting algorithm. Determines both how quickly and how closely the adaptive filter converges to the filter solution.
Leakage	0	Specifies the leakage parameter. Allows you to implement a leaky algorithm. Including a leakage factor can improve the results of the algorithm by forcing the algorithm to continue to adapt even after it reaches a minimum value. Ranges between 0 and 1. DEFaults to 0

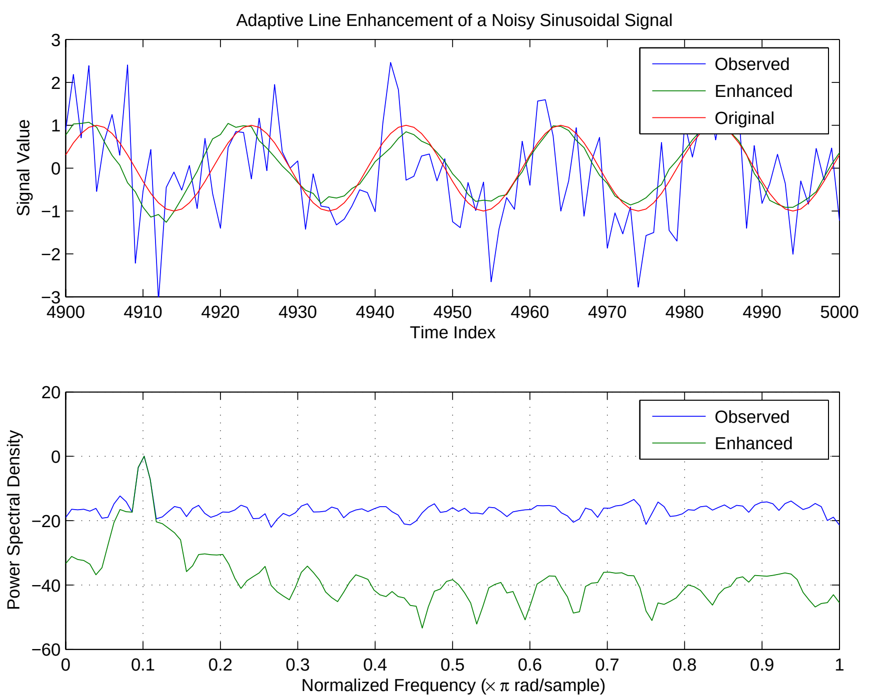
Property	Default Value	Description
PersistentMemory	false or true	Determine whether the filter states and coefficients get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any property value that the filter changes during processing. Property values that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	0	Returns the number of samples processed during filtering. Defaults to zero.

Example

Adaptive line enhancement using a 32-coefficient FIR filter to perform the enhancement. This example runs for 5000 iterations, as you see in property iter.

```
d  = 1;                                % Number of samples of delay
ntr= 5000;                              % Number of iterations
v  = sin(2*pi*0.05*[1:ntr+d]);          % Sinusoidal signal
n  = randn(1,ntr+d);                    % Noise signal
x  = v(1:ntr)+n(1:ntr);              % Input signal (delayed desired
                                        % signal)
d  = v(1+d:ntr+d)+n(1+d:ntr+d);         % Desired signal
mu = 0.0001;                            % Sign-data step size.
ha = adaptfilt.sd(32,mu);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:ntr,[d;y;v(1+d:ntr+d)]);
```

```
axis([ntr-100 ntr -3 3]);
title('Adaptive Line Enhancement of a Noisy Sinusoidal Signal');
legend('Observed','Enhanced','Original');
xlabel('Time Index'); ylabel('Signal Value');
[pxx,om] = pwelch(x(ntr-1000:ntr));
pyy = pwelch(y(ntr-1000:ntr));
subplot(2,1,2);
plot(om/pi,10*log10([pxx/max(pxx),pyy/max(pyy)]));
axis([0 1 -60 20]);
legend('Observed','Enhanced');
xlabel('Normalized Frequency (\times \pi rad/sample)');
ylabel('Power Spectral Density'); grid on;
```



Each of the variants—sign-data, sign-error, and sign-sign—uses the same example. You can compare the results by viewing the figure shown for each adaptive filter method—`adaptfilt.sd`, `adaptfilt.se`, and `adaptfilt.ss`.

See Also

`adaptfilt.lms`, `adaptfilt.se`, `adaptfilt.ss`

References

Moschner, J.L., “Adaptive Filter with Clipped Input Data,” Ph.D. thesis, Stanford Univ., Stanford, CA, June 1970.

Hayes, M., *Statistical Digital Signal Processing and Modeling,* New York Wiley, 1996.

adaptfilt.se

Purpose

Construct a sign-error algorithm FIR adaptive filter object

Syntax

```
ha = adaptfilt.se(l,step,leakage,coeffs,states)
```

Description

`ha = adaptfilt.se(l,step,leakage,coeffs,states)` constructs an FIR sign-error adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.se`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	SE step size. It must be a nonnegative scalar. You can use `maxstep` to determine a reasonable range of step size values for the signals being processed. `step` defaults to 0.1
`leakage`	Your SE leakage factor. It must be a scalar between 0 and 1. When `leakage` is less than one, `adaptfilt.se` implements a leaky SE algorithm. When you omit the `leakage` property in the calling syntax, it defaults to 1 providing no leakage in the adapting algorithm.
`coeffs`	Vector of initial filter coefficients. it must be a length `l` vector. `coeffs` defaults to length `l` vector with elements equal to zero.
`states`	Vector of initial filter states for the adaptive filter. It must be a length `l-1` vector. `states` defaults to a length l-1 vector of zeros.

Properties

In the syntax for creating the `adaptfilt` object, the input options are properties of the object you create. This table list all the properties for the sign-error SD object, their default values, and a brief description of the property.

Property	Default Value	Description
`Algorithm`	Sign-error	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	10	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	`zeros`(1,`l`)	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input. Should be initialized with the initial coefficients for the FIR filter prior to adapting.
`States`	`zeros`(`l`-1,1)	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to (`l` -1).
`StepSize`	0.1	Sets the SE algorithm step size used for each iteration of the adapting algorithm. Determines both how quickly and how closely the adaptive filter converges to the filter solution.

Property	Default Value	Description
Leakage	1	Specifies the leakage parameter. Allows you to implement a leaky algorithm. Including a leakage factor can improve the results of the algorithm by forcing the algorithm to continue to adapt even after it reaches a minimum value. Ranges between 0 and 1. Defaults to one if omitted.
PersistentMemory	false or true	Determine whether the filter states and coefficients get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any property value that the filter changes during processing. Property values that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	0	Returns the number of samples processed during filtering. Defaults to zero.

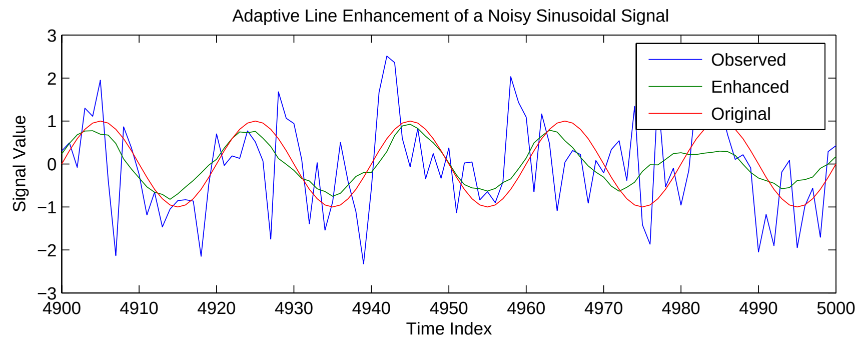
Use inspect(ha) to view or change the object properties graphically using the MATLAB Property Inspector.

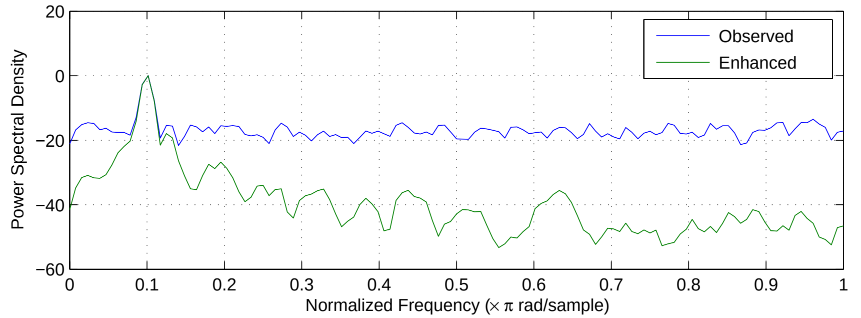
Examples

Adaptive line enhancement using a 32-coefficient FIR filter running over 5000 iterations.

```
d = 1;                              % Number of samples of delay
ntr= 5000;                          % Number of iterations
v = sin(2*pi*0.05*[1:ntr+d]);      % Sinusoidal signal
n = randn(1,ntr+d);                 % Noise signal
x = v(1:ntr)+n(1:ntr);          % Input signal (delayed desired
                                    % signal)
d = v(1+d:ntr+d)+n(1+d:ntr+d);     % Desired signal
mu = 0.0001;                        % Sign-error step size
ha = adaptfilt.se(32,mu);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:ntr,[d;y;v(1+d:ntr+d)]);
axis([ntr-100 ntr -3 3]);
title('Adaptive Line Enhancement of a Noisy Sinusoidal Signal');
legend('Observed','Enhanced','Original');
xlabel('Time Index'); ylabel('Signal Value');
[pxx,om] = pwelch(x(ntr-1000:ntr));
pyy = pwelch(y(ntr-1000:ntr));
subplot(2,1,2);
plot(om/pi,10*log10([pxx/max(pxx),pyy/max(pyy)]));
axis([0 1 -60 20]);
legend('Observed','Enhanced');
xlabel('Normalized Frequency (\times \pi rad/sample)');
ylabel('Power Spectral Density'); grid on;
```

Compare the figure shown here to the ones for `adaptfilt.sd` and `adaptfilt.ss` to see how the variants perform on the same example.





See Also

`adaptfilt.sd`, `adaptfilt.ss`, `adaptfilt.lms`

References

Gersho, A, "Adaptive Filtering With Binary Reinforcement," IEEE Trans. Information Theory, vol. IT-30, pp. 191-199, March 1984.

Hayes, M, *Statistical Digital Signal Processing and Modeling*, New York, Wiley, 1996.

Purpose Construct an adaptive FIR filter object that uses the sign-sign algorithm

Syntax

```
ha = adaptfilt.ss(l,step,leakage,coeffs,states)
```

Description

`ha = adaptfilt.se(l,step,leakage,coeffs,states)` constructs an FIR sign-error adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.ss`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	SS step size. It must be a nonnegative scalar. `step` defaults to 0.1.
`leakage`	Your SS leakage factor. It must be a scalar between 0 and 1. When `leakage` is less than one, `adaptfilt.lms` implements a leaky SS algorithm. When you omit the `leakage` property in the calling syntax, it defaults to 1 providing no leakage in the adapting algorithm.
The `coeffs`	Vector of initial filter coefficients. it must be a length `l` vector. `coeffs` defaults to length `l` vector with elements equal to zero.
`states`	Vector of initial filter states for the adaptive filter. It must be a length `l`-1 vector. `states` defaults to a length l-1 vector of zeros.

`adaptfilt.ss` can be called for a block of data, when `x` and `d` are vectors, or in "sample by sample mode" using a For-loop with the method filter:

```
for n = 1:length(x)
ha = adaptfilt.ss(25,0.9);
```

```
[y(n),e(n)] = filter(ha,(x(n),d(n),s));
% The property values of ha may be modified here.
end
```

Properties

In the syntax for creating the `adaptfilt` object, most of the input options are properties of the object you create. This table list all the properties for sign-sign objects, their default values, and a brief description of the property.

Property	Default Value	Description
`Algorithm`	Sign-sign	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	10	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	`zeros`(1,`l`)	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input. Should be initialized with the initial coefficients for the FIR filter prior to adapting.
`States`	`zeros`(`l`-1,1)	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to (`l` -1).

Property	Default Value	Description
StepSize	0.1	Sets the SE algorithm step size used for each iteration of the adapting algorithm. Determines both how quickly and how closely the adaptive filter converges to the filter solution.
Leakage	1	Specifies the leakage parameter. Allows you to implement a leaky algorithm. Including a leakage factor can improve the results of the algorithm by forcing the algorithm to continue to adapt even after it reaches a minimum value. Ranges between 0 and 1. 1 is the default value.

Property	Default Value	Description
PersistentMemory	false or true	Determine whether the filter states and coefficients get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any property value that the filter changes during processing. Property values that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	0	Returns the number of samples processed during filtering. Defaults to zero.

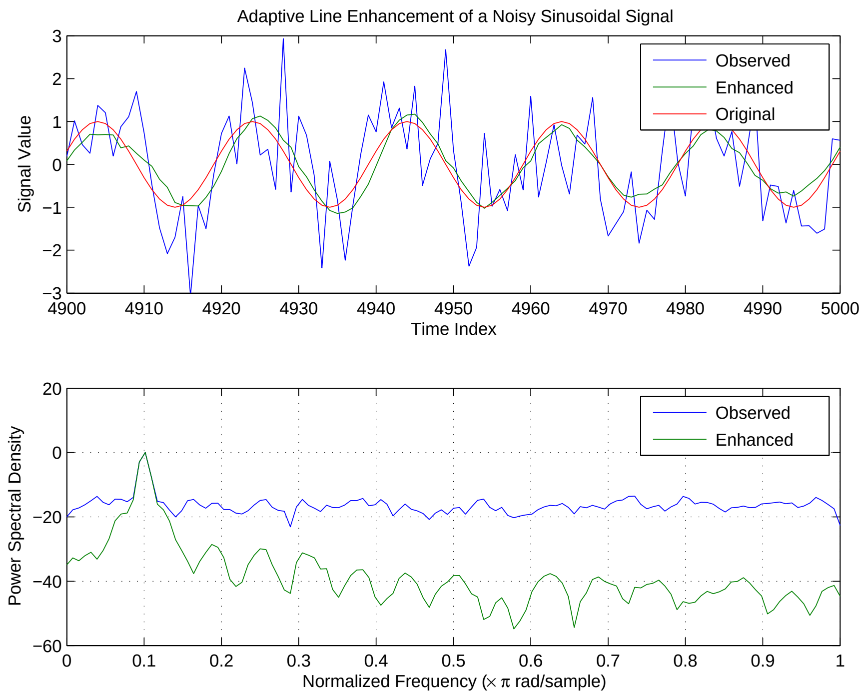
Examples

Demonstrating adaptive line enhancement using a 32-coefficient FIR filter provides a good introduction to the sign-sign algorithm.

```
d = 1;                              % number of samples of delay
ntr= 5000;                          % number of iterations
v = sin(2*pi*0.05*[1:ntr+d]);       % sinusoidal signal
n = randn(1,ntr+d);                 % noise signal
x = v(1:ntr)+n(1:ntr);              % Delayed input signal
d = v(1+d:ntr+d)+n(1+d:ntr+d);      % desired signal
mu = 0.0001;                        % sign-sign step size
ha = adaptfilt.ss(32,mu);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:ntr,[d;y;v(1+d:ntr+d)]);
axis([ntr-100 ntr -3 3]);
title('Adaptive Line Enhancement of a Noisy Sinusoidal Signal');
legend('Observed','Enhanced','Original');
xlabel('Time Index'); ylabel('Signal Value');
[pxx,om] = pwelch(x(ntr-1000:ntr));
```

```
pyy = pwelch(y(ntr-1000:ntr));
subplot(2,1,2);
plot(om/pi,10*log10([pxx/max(pxx),pyy/max(pyy)]));
axis([0 1 -60 20]);
legend('Observed','Enhanced');
xlabel('Normalized Frequency (\times \pi rad/sample)');
ylabel('Power Spectral Density'); grid on;
```

This example is the same as the ones used for the sign-data and sign-error examples. Comparing the figures shown for each of the others lets you assess the performance of each for the same task.



See Also

adaptfilt.se, adaptfilt.sd, adaptfilt.lms

References

Lucky, R.W, “Techniques For Adaptive Equalization of Digital Communication Systems,” Bell Systems Technical Journal, vol. 45, pp. 255-286, Feb. 1966

Hayes, M., *Statistical Digital Signal Processing and Modeling*, New York, Wiley, 1996.

Purpose Construct a sliding window fast transversal least squares adaptive filter object

Syntax

```
ha = adaptfilt.swftf(l,delta,blocklen,gamma,gstates,dstates,coeffs
   ,  states)
```

Description

`ha = adaptfilt.swftf(l,delta,blocklen,gamma,gstates,dstates,coeffs,states)` constructs a sliding window fast transversal least squares adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.swftf`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`delta`	Soft-constrained initialization factor. This scalar should be positive and sufficiently large to maintain stability. `delta` defaults to 1.
`blocklen`	Block length of the sliding window. This must be an integer at least as large as the filter length `l`, which is the default value.
`gamma`	Conversion factor. `gamma` defaults to the matrix [1 -1] that specifies soft-constrained initialization.
`gstates`	States of the kalman gain updates. `gstates` defaults to a zero vector of length (`l` + `blocklen` - 1).
`dstates`	Desired signal states of the adaptive filter. `dstates` defaults to a zero vector of length equal to (`blocklen` - 1). For a default object, dstates is (`l`-1).

Input Argument	Description
coeffs	Vector of initial filter coefficients. It must be a length l vector. coeffs defaults to length l vector of all zeros.
states	Vector of initial filter states. states defaults to a zero vector of length equal to (l + blocklen - 2).

Properties

Since your adaptfilt.swftf filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.swftf objects. To show you the properties that apply, this table lists and describes each property for the filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector where l is the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.

Name	Range	Description
States	Vector of elements, data type double	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).
BlockLength		Block length of the sliding window. This must be an integer at least as large as the filter length l, which is the default value.
InitFactor		Soft-constrained initialization factor. This scalar should be positive and sufficiently large to prevent an excessive number of Kalman gain rescues. delta defaults to one.
KalmanGain		Empty when you construct the object, this gets populated after you run the filter.
ConversionFactor		Conversion factor. Called gamma when it is an input argument, it defaults to the matrix [1 -1] that specifies soft-constrained initialization.
FwdPrediction		

Name	Range	Description
BkwdPredictions		Returns the predicted samples generated during adaptation.Refer to [12] in the bibliography for details about linear prediction.
DesiredSignalStates		Desired signal states of the adaptive filter. dstates defaults to a zero vector with length equal to (blocklen - 1).
KalmanGainStates		Contains the states of the Kalman gains for the adaptive algorithm. Initialized to a vector of double data type entries.

Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

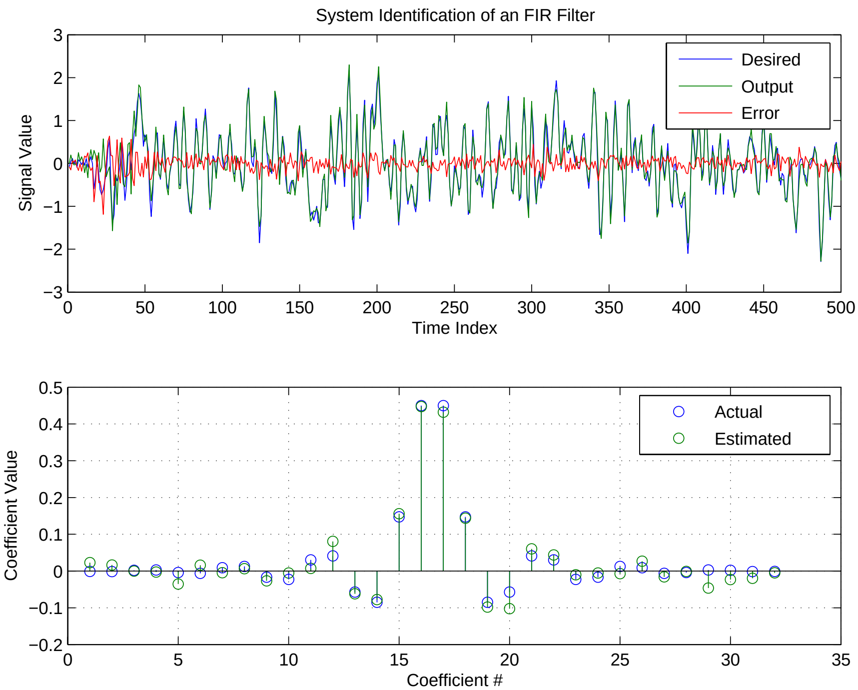
Examples

Over 500 iterations, perform a system identification of a 32-coefficient FIR filter.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
L  = 32;               % Adaptive filter length
del = 0.1;             % Soft-constrained initialization factor
N  = 64;               % block length
ha = adaptfilt.swftf(L,del,N);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
```

```
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.Coefficients.']);
legend('Actual','Estimated');
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```

Review the figure for the results of the example. When you evaluate the example you should get the same results, within the differences in the random noise signal you use.



See Also

adaptfilt.ftf, adaptfilt.swrls, adaptfilt.ap, adaptfilt.apru

References

D.T.M. Slock and Kailath, T., “A Modular Prewindowing Framework for Covariance FTF RLS Algorithms,” Signal Processing, vol. 28, pp. 47-61, 1992

D.T.M. Slock and Kailath, T., “A Modular Multichannel Multi-Experiment Fast Transversal Filter RLS Algorithm,” Signal Processing, vol. 28, pp. 25-45, 1992

adaptfilt.swrls

Purpose

Construct a sliding window recursive least squares FIR adaptive filter

Syntax

```
ha = adaptfilt.swrls(l,lambda,invcov,swblocklen,dstates,
   coeffs,states)
```

Description

`ha = adaptfilt.swrls(l,lambda,invcov,swblocklen,dstates, coeffs,states)` constructs an FIR sliding window RLS adaptive filter `ha`.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.swrls`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps). It must be a positive integer. `l` defaults to 10.
`lambda`	RLS forgetting factor. This is a scalar and should lie within the range (0, 1]. `lambda` defaults to 1.
`invcov`	Inverse of the input signal covariance matrix. You should initialize invcov to a positive definite matrix.
`swblocklen`	Block length of the sliding window. This integer must be at least as large as the filter length. `swblocklen` defaults to 16.
`dstates`	Desired signal states of the adaptive filter. `dstates` defaults to a zero vector with length equal to (`swblocklen` - 1).

Input Argument	Description
`coeffs`	Vector of initial filter coefficients. It must be a length `l` vector. `coeffs` defaults to length `l` vector of all zeros.
`states`	Vector of initial filter states. `states` defaults to a zero vector of length equal to (l + `swblocklen` - 2).

Properties

Since your `adaptfilt.swrls` filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating `adaptfilt.swrls` objects. To show you the properties that apply, this table lists and describes each property for the filter object.

Name	Range	Description
`Algorithm`	None	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps
`Coefficients`	Any vector of `l` elements	Vector containing the initial filter coefficients. It must be a length `l` vector where `l` is the number of filter coefficients. `coeffs` defaults to length `l` vector of zeros when you do not provide the argument for input.
`States`	Vector of elements, data type double	Vector of the adaptive filter states. `states` defaults to a vector of zeros which has length equal to (`l` + `swblocklen` - 2).

Name	Range	Description
`ForgettingFactor`	Scalar	Forgetting factor of the adaptive filter. This is a scalar and should lie in the range (0, 1]. It defaults to 1. Setting `forgetting factor = 1` denotes infinite memory while adapting to find the new filter. Note that this is the `lambda` input argument.
`KalmanGain`	Vector with dimensions (`l`,1)	Empty when you construct the object, this gets populated after you run the filter.
`InvCov`	Matrix	Square matrix with each dimension equal to the filter length `l`.
`SwBlockLength`	Integer	Block length of the sliding window. This integer must be at least as large as the filter length. `swblocklen` defaults to 16.
`DesiredSignalStates`	Vector	Desired signal states of the adaptive filter. `dstates` defaults to a zero vector with length equal to (`swblocklen` - 1).

Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering.

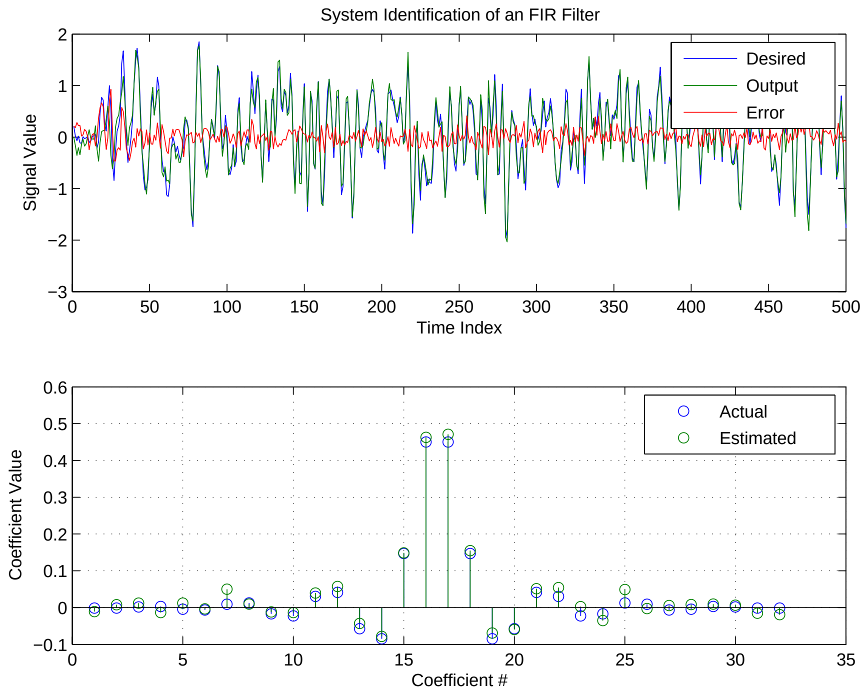
Examples

System Identification of a 32-coefficient FIR filter. Use 500 iterations to adapt to the unknown filter. After the example code, you see a figure that plots the results of the running the code.

```
x  = randn(1,500);     % Input to the filter
b  = fir1(31,0.5);     % FIR system to be identified
n  = 0.1*randn(1,500); % Observation noise signal
d  = filter(b,1,x)+n;  % Desired signal
P0 = 10*eye(32);       % Initial correlation matrix inverse
lam = 0.99;            % RLS forgetting factor
N  = 64;               % Block length
ha = adaptfilt.swrls(32,lam,P0,N);
[y,e] = filter(ha,x,d);
subplot(2,1,1); plot(1:500,[d;y;e]);
title('System Identification of an FIR Filter');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,1,2); stem([b.',ha.Coefficients.']);
legend('Actual','Estimated');
```

```
xlabel('Coefficient #'); ylabel('Coefficient Value'); grid on;
```

In the figure you see clearly that the adaptive filter process successfully identified the coefficients of the unknown FIR filter. But then you knew it had to or many things we take for granted, such as modems on computers, would not work.



See Also

`adaptfilt.rls`, `adaptfilt.qrdrls`, `adaptfilt.hswrls`

Purpose

Create a transform-domain (TDAFDFT) adaptive filter object that uses the discrete Fourier transform

Syntax

```
ha = adaptfilt.tdafdft(l,step,leakage,offset,delta,lambda,
   coeffs,states)
```

Description

`ha = adaptfilt.tdafdft(l,step,leakage,offset,delta,lambda, coeffs,states)` constructs a transform-domain adaptive filter object `ha` using a discrete Fourier transform.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.tdafdft`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	Adaptive filter step size. It must be a nonnegative scalar. You can use `maxstep` to determine a reasonable range of step size values for the signals being processed. `step` defaults to 0.
`leakage`	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, you are implementing a leaky version of the TDAFDFT algorithm. `leakage` defaults to 1—no leakage.
`offset`	Offset for the normalization terms in the coefficient updates. YOu can use this argument to avoid dividing by zeros or by very small numbers when any of the FFT input signal powers become very small. `offset` defaults to zero.

Input Argument	Description
`delta`	Initial common value of all of the transform domain powers. Its initial value should be positive. `delta` defaults to 5.
`lambda`	Averaging factor used to compute the exponentially-windowed estimates of the powers in the transformed signal bins for the coefficient updates. `lambda` should lie between zero and one. For default filter objects, LAMBDA equals (1 - `step`).
`coeffs`	Initial time domain coefficients of the adaptive filter. Set it to be a length `l` vector. `coeffs` defaults to a zero vector of length `l`.
`states`	Initial conditions of the adaptive filter. `states` defaults to a zero vector with length equal to (`l` - 1).

Properties

Since your `adaptfilt.tdafdft` filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating `adaptfilt.tdafdft` objects. To show you the properties that apply, this table lists and describes each property for the transform domain filter object.

Name	Range	Description
`Algorithm`	None	Defines the adaptive filter algorithm the object uses during adaptation
`FilterLength`	Any positive integer	Reports the length of the filter, the number of coefficients or taps

Name	Range	Description
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector where l is the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
States	Vector of elements, data type double	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).
StepSize	0 to 1	Step size. It must be a nonnegative scalar, greater than zero and less than or equal to 1. step defaults to 0.
Leakage	0 to 1	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, you are implementing a leaky version of the TDAFDFT algorithm. leakage defaults to 1—no leakage.

Name	Range	Description
Offset		Offset for the normalization terms in the coefficient updates. You can use this argument to avoid dividing by zeros or by very small numbers when any of the FFT input signal powers become very small. offset defaults to zero.
Power	2*l element vector	A vector of 2*l elements, each initialized with the value delta from the input arguments. As you filter data, Power gets updated by the filter process.
AvgFactor		Averaging factor used to compute the exponentially-windowed estimates of the powers in the transformed signal bins for the coefficient updates. AvgFactor should lie between zero and one. For default filter objects, AvgFactor equals (1 - step). lambda is the input argument that represent AvgFactor.

Name	Range	Description
PersistentMemory	false or true	Determines whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

Examples

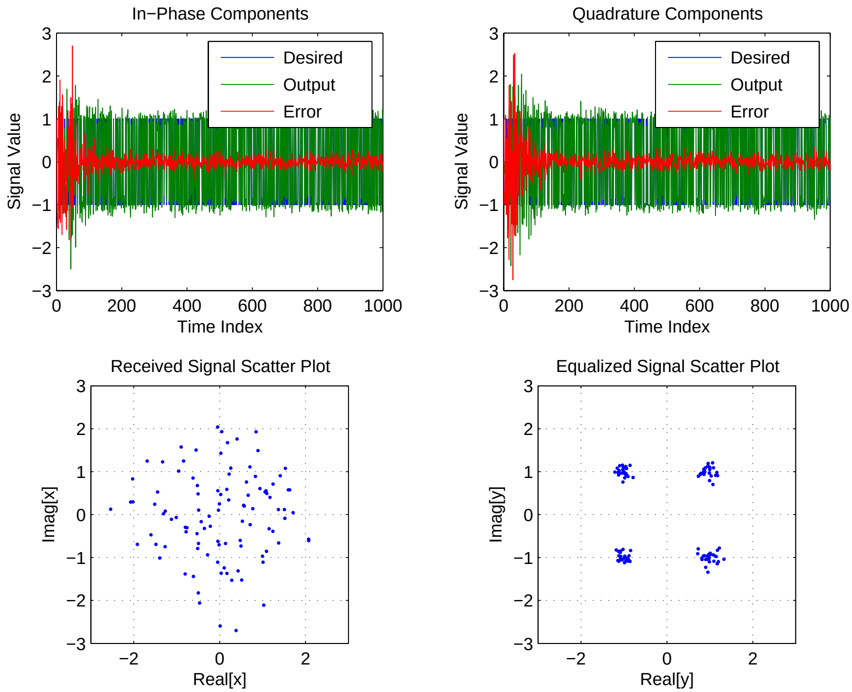
Quadrature Phase Shift Keying (QPSK) adaptive equalization using a 32-coefficient FIR filter (1000 iterations).

```
D  = 16;                        % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];    % Numerator coefficients of channel
a  = [1 -0.7];              % Denominator coefficients of channel
ntr= 1000;                      % Number of iterations
s  = sign(randn(1,ntr+D)) + j*sign(randn(1,ntr+D));% Baseband
                                                    % QPSK signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));     % Noise signal
r  = filter(b,a,s)+n;           % Received signal
x  = r(1+D:ntr+D);              % Input signal (received signal)
d  = s(1:ntr);             % Desired signal (delayed QPSK signal)
L  = 32;                        % filter length
mu = 0.01;                      % Step size
ha = adaptfilt.tdafdft(L,mu);
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
title('In-Phase Components');
legend('Desired','Output','Error');
```

```
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```

All of the time domain adaptive filter reference pages use this QPSK example. By comparing the results for each variation you get an idea of the differences in the way each one performs.

This figure demonstrates the results of running the example code shown.



See Also

`adaptfilt.tdafdct`, `adaptfilt.fdaf`, `adaptfilt.blms`

References

S. Haykin, *Adaptive Filter Theory*, 3rd Edition, Prentice Hall, N.J., 1996

Purpose

Construct a transform-domain (TDAFDCT) adaptive filter object that uses the discrete cosine transform

Syntax

```
ha = adaptfilt.tdafdct(l,step,leakage,offset,delta,lambda,coeffs,
states)
```

Description

`ha = adaptfilt.tdafdct(l,step,leakage,offset,delta,lambda,coeffs, states)` constructs a transform-domain adaptive filter ha object that uses the discrete cosine transform to perform filter adaptation.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.tdafdct`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	Adaptive filter step size. It must be a nonnegative scalar. You can use `maxstep` to determine a reasonable range of step size values for the signals being processed. `step` defaults to 0.
`leakage`	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, you are implementing a leaky version of the TDAFDCT algorithm. `leakage` defaults to 1—no leakage.
`offset`	Offset for the normalization terms in the coefficient updates. You can use this argument to avoid dividing by zero or by very small numbers when any of the FFT input signal powers become very small. `offset` defaults to zero.

Input Argument	Description
delta	Initial common value of all of the transform domain powers. Its initial value should be positive. delta defaults to 5.
lambda	Averaging factor used to compute the exponentially-windowed estimates of the powers in the transformed signal bins for the coefficient updates. lambda should lie between zero and one. For default filter objects, lambda equals (1 - step).
coeffs	Initial time domain coefficients of the adaptive filter. Set it to be a length l vector. coeffs defaults to a zero vector of length l.
states	Initial conditions of the adaptive filter. states defaults to a zero vector with length equal to (l - 1).

Properties

Since your adaptfilt.tdafdct filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.tdafdct objects. To show you the properties that apply, this table lists and describes each property for the transform domain filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps

Name	Range	Description
Coefficients	Vector of elements	Vector containing the initial filter coefficients. It must be a length l vector where l is the number of filter coefficients. coeffs defaults to length l vector of zeros when you do not provide the argument for input.
States	Vector of elements, data type double	Vector of the adaptive filter states. states defaults to a vector of zeros which has length equal to (l + projectord - 2).
StepSize	0 to 1	Step size. It must be a nonnegative scalar, greater than zero and less than or equal to 1. You can use maxstep to determine a reasonable range of step size values for the signals being processed. step defaults to 0.
Leakage	0 to 1	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, you are implementing a leaky version of the TDAFDFT algorithm. leakage defaults to 1—no leakage.

Name	Range	Description
Offset		Offset for the normalization terms in the coefficient updates. You can use this argument to avoid dividing by zeros or by very small numbers when any of the FFT input signal powers become very small. offset defaults to zero.
Power	2*l element vector	A vector of 2*l elements, each initialized with the value delta from the input arguments. As you filter data, Power gets updated by the filter process.
AvgFactor		Averaging factor used to compute the exponentially-windowed estimates of the powers in the transformed signal bins for the coefficient updates. AvgFactor should lie between zero and one. For default filter objects, AvgFactor equals (1 - step). lambda is the input argument that represent AvgFactor.

Name	Range	Description
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

For checking the values of properties for an adaptive filter object, use get(ha) or enter the object name, without a trailing semicolon, at the MATLAB prompt.

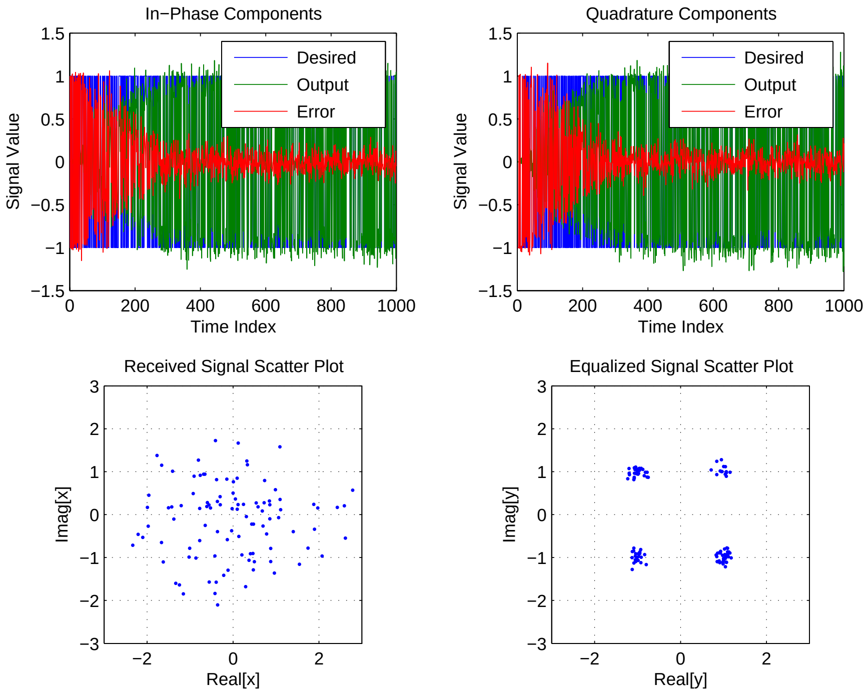
Examples

Using 1000 iterations, perform a Quadrature Phase Shift Keying (QPSK) adaptive equalization using a 32-coefficient FIR filter.

```
D  = 16;                        % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];     % Numerator coefficients of channel
a  = [1 -0.7];               % Denominator coefficients of channel
ntr= 1000;                      % Number of iterations
s  = sign(randn(1,ntr+D)) + j*sign(randn(1,ntr+D));% Baseband
                                                    % QPSK signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D)); % Noise signal
r  = filter(b,a,s)+n;           % Received signal
x  = r(1+D:ntr+D);              % Input signal (received signal)
d  = s(1:ntr);              % Desired signal (delayed QPSK signal)
L  = 32;                        % filter length
mu = 0.01;                      % Step size
```

```
ha = adaptfilt.tdafdct(L,mu);
[y,e] = filter(ha,x,d);
subplot(2,2,1); plot(1:ntr,real([d;y;e]));
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```

Compare the plots shown in this figure to those in the other time domain filter variations. The comparison should help you select and understand how the variants differ.



See Also

`adaptfilt.tdafdft`, `adaptfilt.fdaf`, `adaptfilt.blms`

References

S. Haykin, *Adaptive Filter Theory*, 3rd Edition, Prentice Hall, N.J., 1996.

Purpose

Construct a Unconstrained frequency-domain (UFDAF) FIR adaptive filter with binned step size normalization

Syntax

```
ha = adaptfilt.ufdaf(l,step,leakage,delta,lambda,blocklen,offset,c
oeffs,states)
```

Description

`ha = adaptfilt.ufdaf(l,step,leakage,delta,lambda,blocklen,offset,c oeffs,states)` constructs an unconstrained frequency-domain FIR adaptive filter `ha` with bin step size normalization.

Input Arguments

Entries in the following table describe the input arguments for `adaptfilt.ufdaf`.

Input Argument	Description
`l`	Adaptive filter length (the number of coefficients or taps) and it must be a positive integer. `l` defaults to 10.
`step`	Adaptive filter step size. It must be a nonnegative scalar. `step` defaults to 0.
`leakage`	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, you are implementing a leaky version of the UFDAF algorithm. `leakage` defaults to 1—no leakage.
`delta`	Initial common value of all of the FFT input signal powers. the initial value of delta should should be positive, and it defaults to 1.
`lambda`	Specifies the averaging factor used to compute the exponentially-windowed FFT input signal powers for the coefficient updates. `lambda` should lie in the range (0,1]. For default UFDAF filter objects, `lambda` defaults to 0.9.

Input Argument	Description
blocklen	Block length for the coefficient updates. This must be a positive integer. For faster execution, (blocklen + 1) should be a power of two. blocklen defaults to 1.
offset	Offset for the normalization terms in the coefficient updates. This can help you avoid divide by zero conditions, or divide by very small numbers conditions, when any of the FFT input signal powers become very small. Default value is zero.
coeffs	Initial time-domain coefficients of the adaptive filter. It should be a length l vector. The filter object uses these coefficients to compute the initial frequency-domain filter coefficients via an FFT computed after zero-padding the time-domain vector by blocklen.
states	Adaptive filter states. states defaults to a zero vector with length equal to l.

Properties

Since your adaptfilt.ufdaf filter is an object, it has properties that define its behavior in operation. Note that many of the properties are also input arguments for creating adaptfilt.ufdaf objects. To show you the properties that apply, this table lists and describes each property for the filter object.

Name	Range	Description
Algorithm	None	Defines the adaptive filter algorithm the object uses during adaptation
FilterLength	Any positive integer	Reports the length of the filter, the number of coefficients or taps

Name	Range	Description
StepSize	0 to 1	Adaptive filter step size. It must be a nonnegative scalar. You can use maxstep to determine a reasonable range of step size values for the signals being processed. step defaults to 0.
Leakage	0 to 1	Leakage parameter of the adaptive filter. When you set this argument to a value between zero and one, you are implementing a leaky version of the UFDAF algorithm. leakage defaults to 1—no leakage.
Power	2*l element vector	A vector of 2*l elements, each initialized with the value delta from the input arguments. As you filter data, Power gets updated by the filter process.

Name	Range	Description
AvgFactor		Specifies the averaging factor used to compute the exponentially-windowed FFT input signal powers for the coefficient updates. AvgFactor should lie in the range (0,1]. For default UFDAF filter objects, AvgFactor defaults to 0.9. Note that AvgFactor and lambda are the same thing—lambda is an input argument and AvgFactor a property of the object.
BlockLength		Block length for the coefficient updates. This must be a positive integer. For faster execution, (blocklen + 1) should be a power of two. blocklen defaults to 1.
Offset		Offset for the normalization terms in the coefficient updates. This can help you avoid divide by zero conditions, or divide by very small numbers conditions, when any of the FFT input signal powers become very small. Default value is zero.
FFTCoefficients		Stores the discrete Fourier transform of the filter coefficients in coeffs.

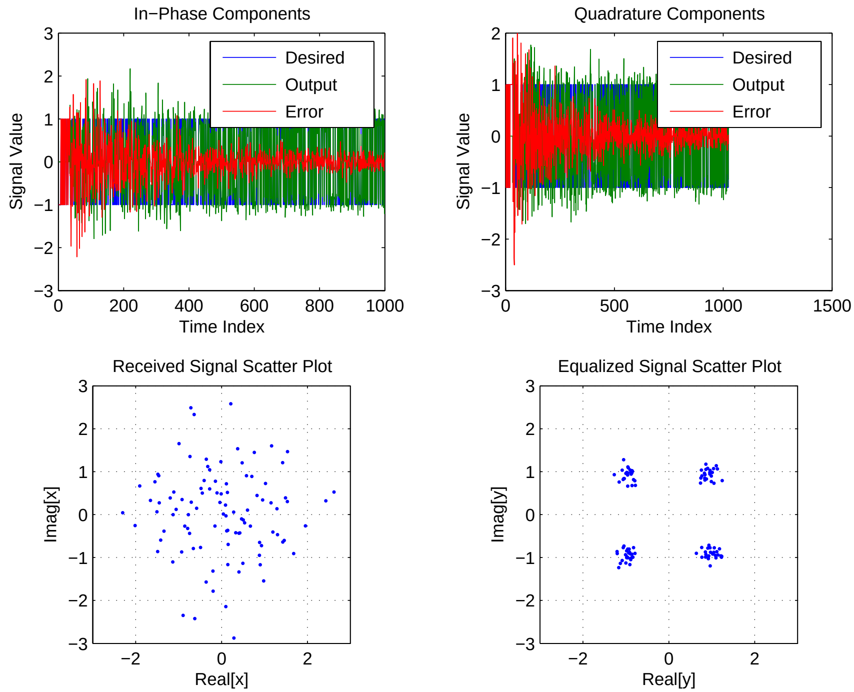
Name	Range	Description
FFTStates		States for the FFT operation.
PersistentMemory	false or true	Determine whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. Defaults to false.
NumSamplesProcessed	Any integer	Returns the number of samples processed during filtering. Defaults to zero.

Examples

Show an example of Quadrature Phase Shift Keying (QPSK) adaptive equalization using a 32-coefficient adaptive filter. For fidelity, use 1024 iterations. The figure that follows the code provides the information you need to assess the performance of the equalization process.

```
D  = 16;                        % Number of samples of delay
b  = exp(j*pi/4)*[-0.7 1];    % Numerator coefficients of channel
a  = [1 -0.7];               % Denominator coefficients of channel
ntr= 1024;                      % Number of iterations
s = sign(randn(1,ntr+D)) + j*sign(randn(1,ntr+D)); % Baseband
                                                    % QPSK signal
n  = 0.1*(randn(1,ntr+D) + j*randn(1,ntr+D));  % Noise signal
r  = filter(b,a,s)+n;           % Received signal
x  = r(1+D:ntr+D);              % Input signal (received signal)
d  = s(1:ntr);               % Desired signal (delayed QPSK signal)
del = 1;                        % Initial FFT input powers
mu  = 0.1;                      % Step size
lam = 0.9;                      % Averaging factor
```

```
ha = adaptfilt.ufdaf(32,mu,1,del,lam);
[y,e] = filter(ha,x,d);
subplot(2,2,1);
plot(1:1000,real([d(1:1000);y(1:1000);e(1:1000)]));
title('In-Phase Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,2); plot(1:ntr,imag([d;y;e]));
title('Quadrature Components');
legend('Desired','Output','Error');
xlabel('Time Index'); ylabel('Signal Value');
subplot(2,2,3); plot(x(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Received Signal Scatter Plot'); axis('square');
xlabel('Real[x]'); ylabel('Imag[x]'); grid on;
subplot(2,2,4); plot(y(ntr-100:ntr),'.'); axis([-3 3 -3 3]);
title('Equalized Signal Scatter Plot'); axis('square');
xlabel('Real[y]'); ylabel('Imag[y]'); grid on;
```



See Also

`adaptfilt.fdaf`, `adaptfilt.pbufdaf`, `adaptfilt.blms`, `adaptfilt.blmsfft`

References

J.J. Shynk, “Frequency-domain and Multirate Adaptive Filtering,” IEEE Signal Processing Magazine, vol. 9, no. 1, pp. 14-37, Jan. 1992

allpassbpc2bpc

Purpose

Return an allpass filter for complex bandpass transformation

Syntax

```
[AllpassNum,AllpassDen] = allpassbpc2bpc(Wo,Wt)
```

Description

`[AllpassNum,AllpassDen] = allpassbpc2bpc(Wo,Wt)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the first-order allpass mapping filter for performing a complex bandpass to complex bandpass frequency transformation. This transformation effectively places two features of an original filter, located at frequencies W_{o1} and W_{o2}, at the required target frequency locations W_{t1} and W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. In most of the cases the features selected for the transformation are the band edges of the filter passbands. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

This transformation can also be used for transforming other types of filters; e.g., complex notch filters or resonators can be repositioned at two distinct desired frequencies at any place around the unit circle. This is very attractive for adaptive systems.

Examples

Design the allpass filter changing the complex bandpass filter with the band edges originally at W_{o1}=0.2 and W_{o2}=0.4 to the new band edges of W_{t1}=0.3 and W_{t2}=0.6 precisely defined:

```
Wo = [0.2, 0.4];
Wt = [0.3, 0.6];
[AllpassNum, AllpassDen] = allpassbpc2bpc(Wo, Wt);
```

Calculate the frequency response of the mapping filter in the full range:

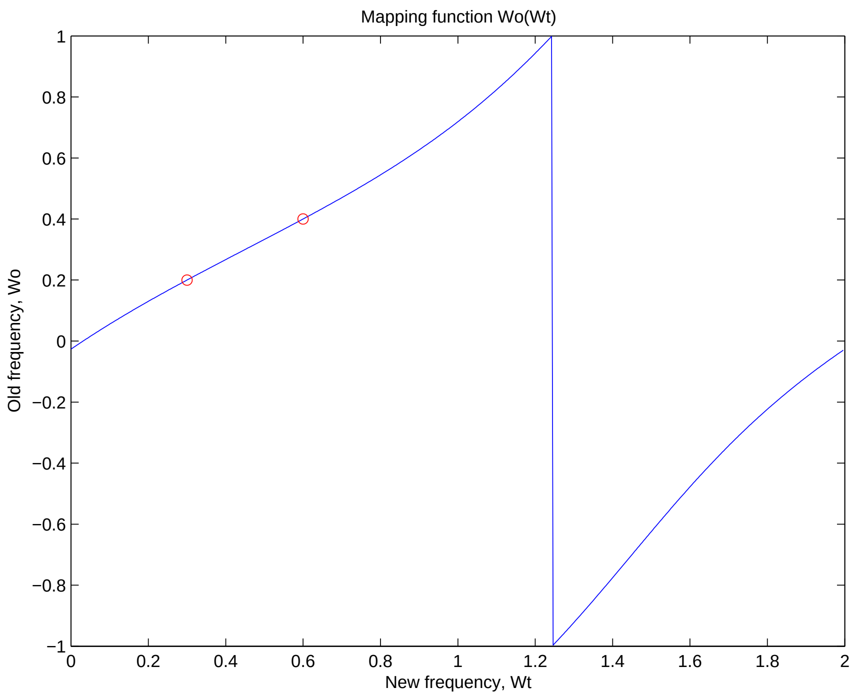
```
[ha, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$:

```
plot(f/pi, angle(ha)/pi, Wt, Wo, 'ro');
```

```
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

To demonstrate, the following figure shows the mapping function between old and new frequencies.



Arguments

`Wo`
Frequency values to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

See Also

`iirbpc2bpc`, `zpkbpc2bpc`

Purpose

Return an allpass filter for lowpass to bandpass transformation

Syntax

```
[AllpassNum,AllpassDen] = allpasslp2bp(Wo,Wt)
```

Description

`[AllpassNum,AllpassDen] = allpasslp2bp(Wo,Wt)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the second-order allpass mapping filter for performing a real lowpass to real bandpass frequency transformation. This transformation effectively places one feature of an original filter, located at frequency $-W_o$, at the required target frequency location, W_{t1}, and the second feature, originally at $+W_o$, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. This transformation implements the "DC mobility," which means that the Nyquist feature stays at Nyquist, but the DC feature moves to a location dependent on the selection of W_t.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandpass transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Lowpass to bandpass transformation can also be used for transforming other types of filters; e.g., real notch filters or resonators can be doubled and repositioned at two distinct desired frequencies.

Examples

Design the allpass filter changing the lowpass filter with cutoff frequency at W_o=0.5 to the real bandpass filter with cutoff frequencies at W_{t1}=0.25 and W_{t2}=0.375:

```
Wo = 0.5;
Wt = [0.25, 0.375];
[AllpassNum, AllpassDen] = allpasslp2bp(Wo, Wt);
```

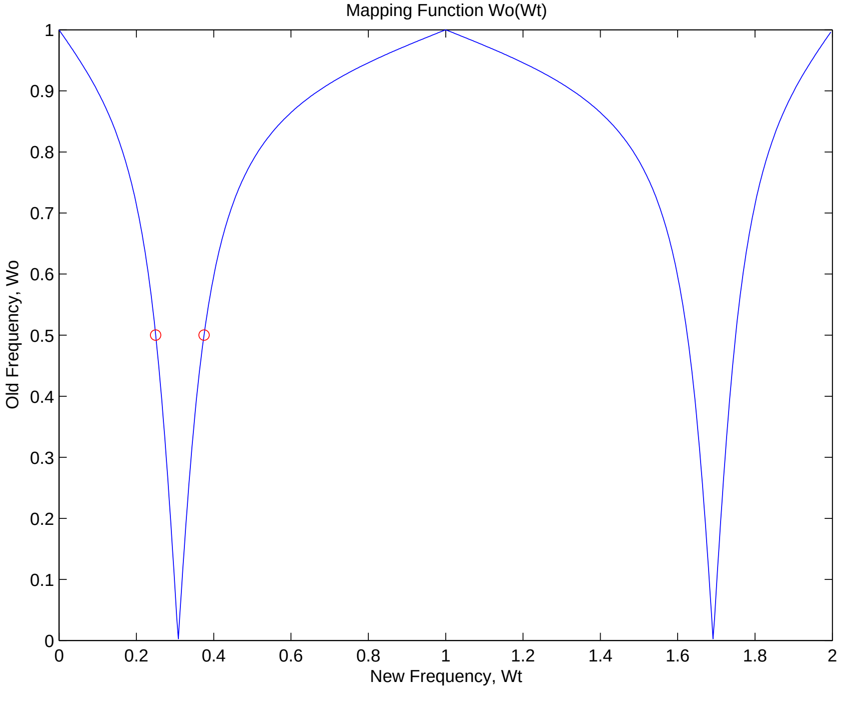
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$. Please note that the transformation works in the same way for both positive and negative frequencies:

```
plot(f/pi, abs(angle(h))/pi, Wt, Wo, 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

Shown in the figure, with the x-axis as the new frequency, you see the mapping filter for the example.



Arguments

Wo

Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirlp2bp`, `zpklp2bp`

References

[1] Constantinides, A.G., "Spectral transformations for digital filters," *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, "Prototype reference transfer function parameters in the discrete-time frequency transformations," *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, "Closed-form solutions for discrete-time elliptic transfer functions," *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

[4] Constantinides, A.G., "Design of bandpass digital filters," *IEEE Proceedings*, vol. 1, pp. 1129-1231, June 1969.

allpasslp2bpc

Purpose

Return an allpass filter for lowpass to complex bandpass transformation

Syntax

`[AllpassNum,AllpassDen] = allpasslp2bpc(Wo,Wt)`

Description

`[AllpassNum,AllpassDen] = allpasslp2bpc(Wo,Wt)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the first-order allpass mapping filter for performing a real lowpass to complex bandpass frequency transformation. This transformation effectively places one feature of an original filter, located at frequency $-W_o$, at the required target frequency location, W_{t1}, and the second feature, originally at $+W_o$, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandpass transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Lowpass to bandpass transformation can also be used for transforming other types of filters; e.g., real notch filters or resonators can be doubled and positioned at two distinct desired frequencies at any place around the unit circle forming a pair of complex notches/resonators. This transformation can be used for designing bandpass filters for radio receivers from the high-quality prototype lowpass filter.

Examples

Design the allpass filter changing the real lowpass filter with the cutoff frequency of W_o=0.5 into a complex bandpass filter with band edges of W_{t1}=0.2 and W_{t2}=0.4 precisely defined:

```
Wo = 0.5;
Wt = [0.2,0.4];
[AllpassNum, AllpassDen] = allpasslp2bpc(Wo, Wt);
```

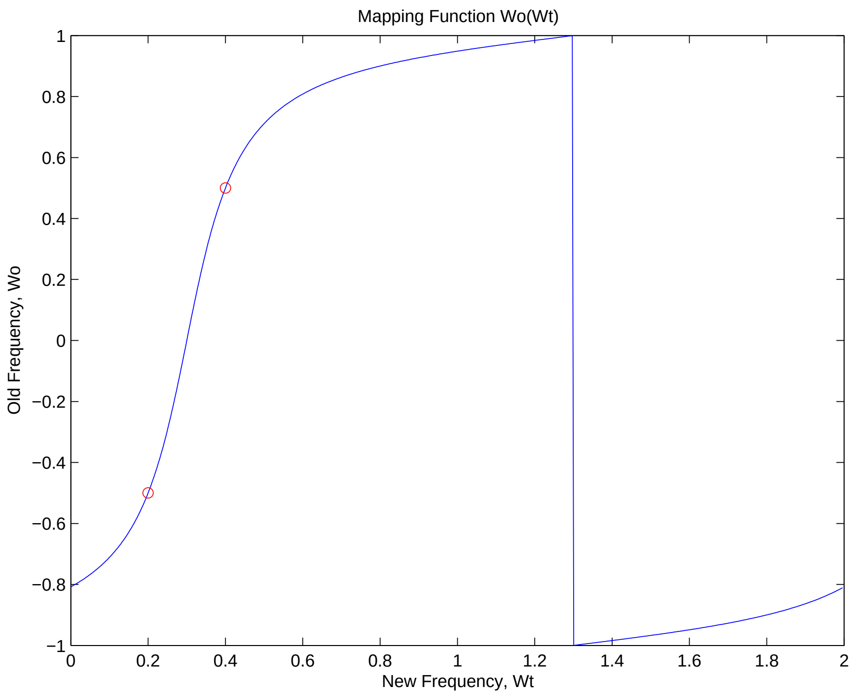
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$:

```
plot(f/pi, angle(h)/pi, Wt, Wo.*[-1,1], 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

The figure shown here details the mapping filter provided by the function.



Arguments

`Wo`

Frequency value to be transformed from the prototype filter. It should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wt`
Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also

`iirlp2bpc`, `zpklp2bpc`

Purpose Return an allpass filter for lowpass to bandstop transformation

Syntax

```
[AllpassNum,AllpassDen] = allpasslp2bs(Wo,Wt)
```

Description `[AllpassNum,AllpassDen] = allpasslp2bs(Wo,Wt)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the second-order allpass mapping filter for performing a real lowpass to real bandstop frequency transformation. This transformation effectively places one feature of an original filter, located at frequency $-W_o$, at the required target frequency location, W_{t1}, and the second feature, originally at $+W_o$, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. This transformation implements the "Nyquist Mobility," which means that the DC feature stays at DC, but the Nyquist feature moves to a location dependent on the selection of W_o and W_t.

Relative positions of other features of an original filter change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. After the transformation feature F_2 will precede F_1 in the target filter. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandstop transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Examples Design the allpass filter changing the lowpass filter with cutoff frequency at W_o=0.5 to the real bandstop filter with cutoff frequencies at W_{t1}=0.25 and W_{t2}=0.375:

```
Wo = 0.5;
Wt = [0.25, 0.375];
[AllpassNum, AllpassDen] = allpasslp2bs(Wo, Wt);
```

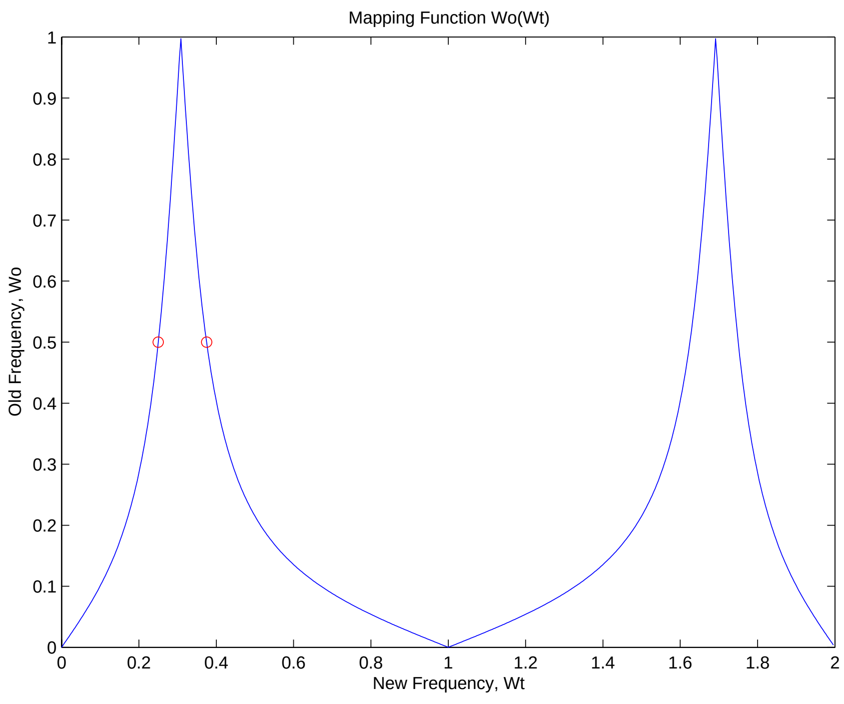
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$. Please note that the transformation works in the same way for both positive and negative frequencies:

```
plot(f/pi, abs(angle(h))/pi, Wt, Wo, 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

In the figure, you find the mapping filter function as determined by the example. Note the response is normalized to π, as mentioned earlier.



Arguments

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirlp2bs`, `zpklp2bs`

References

[1] Constantinides, A.G., “Spectral transformations for digital filters,” *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, “Prototype reference transfer function parameters in the discrete-time frequency transformations,” *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, “Closed-form solutions for discrete-time elliptic transfer functions,” *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

[4] Constantinides, A.G., “Design of bandpass digital filters,” *IEEE Proceedings*, vol. 1, pp. 1129-1231, June 1969.

Purpose Return an allpass filter for lowpass to complex bandstop transformation

Syntax

```
[AllpassNum,AllpassDen] = allpasslp2bsc(Wo,Wt)
```

Description `[AllpassNum,AllpassDen] = allpasslp2bsc(Wo,Wt)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the first-order allpass mapping filter for performing a real lowpass to complex bandstop frequency transformation. This transformation effectively places one feature of an original filter, located at frequency -W_o, at the required target frequency location, W_{t1}, and the second feature, originally at +W_o, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. Additionally the transformation swaps passbands with stopbands in the target filter.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandstop transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Lowpass to bandpass transformation can also be used for transforming other types of filters; e.g., real notch filters or resonators can be doubled and positioned at two distinct desired frequencies at any place around the unit circle forming a pair of complex notches/resonators. This transformation can be used for designing bandstop filters for band attenuation or frequency equalizers, from the high-quality prototype lowpass filter.

Examples Design the allpass filter changing the real lowpass filter with the cutoff frequency of W_o=0.5 into a complex bandstop filter with band edges of W_{t1}=0.2 and W_{t2}=0.4 precisely defined:

```
Wo = 0.5;
Wt = [0.2,0.4];
[AllpassNum, AllpassDen] = allpasslp2bsc(Wo, Wt);
```

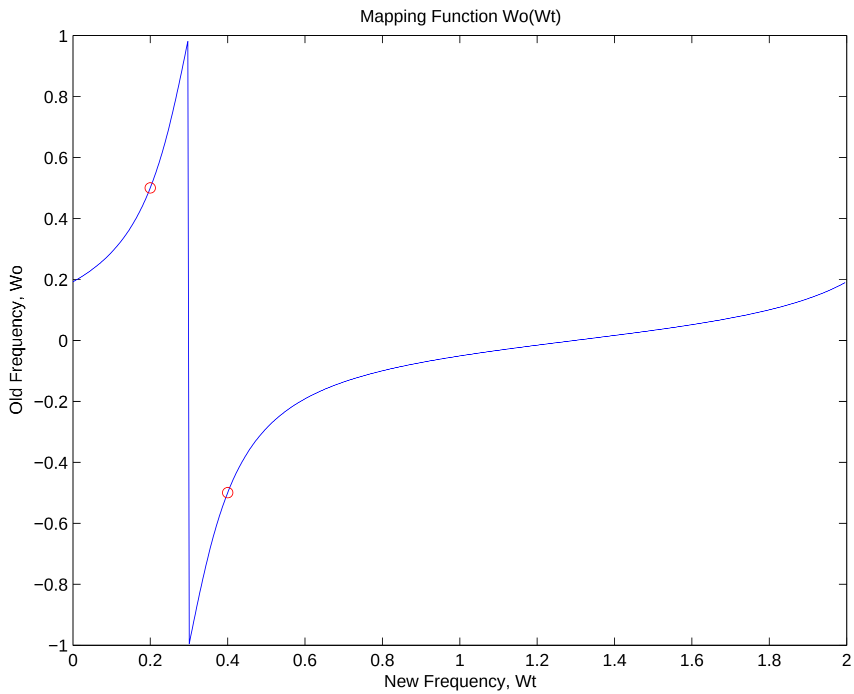
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$:

```
plot(f/pi, angle(h)/pi, Wt, Wo.*[1,-1], 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

We plot the resulting allpass mapping function response in this figure.



Arguments

Wo

Frequency value to be transformed from the prototype filter. It should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wt`
Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also `iirlp2bsc`, `zpklp2bsc`

Purpose Return an allpass filter for lowpass to highpass transformation

Syntax

```
[AllpassNum,AllpassDen] = allpasslp2hp(Wo,Wt)
```

Description

[AllpassNum,AllpassDen] = allpasslp2hp(Wo,Wt) returns the numerator, AllpassNum, and the denominator, AllpassDen, of the first-order allpass mapping filter for performing a real lowpass to real highpass frequency transformation. This transformation effectively places one feature of an original filter, located originally at frequency, W_o, at the required target frequency location, W_t, at the same time rotating the whole frequency response by half of the sampling frequency. Result is that the DC and Nyquist features swap places.

Relative positions of other features of an original filter change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. After the transformation feature F_2 will precede F_1 in the target filter. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to highpass transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband.

Lowpass to highpass transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can change their position in a simple way by using the lowpass to highpass transformation.

Examples

Design the allpass filter changing the lowpass filter to the highpass filter with its cutoff frequency moved from W_o=0.5 to W_t=0.25:

```
Wo = 0.5;
Wt = 0.25;
[AllpassNum, AllpassDen] = allpasslp2hp(Wo, Wt);
```

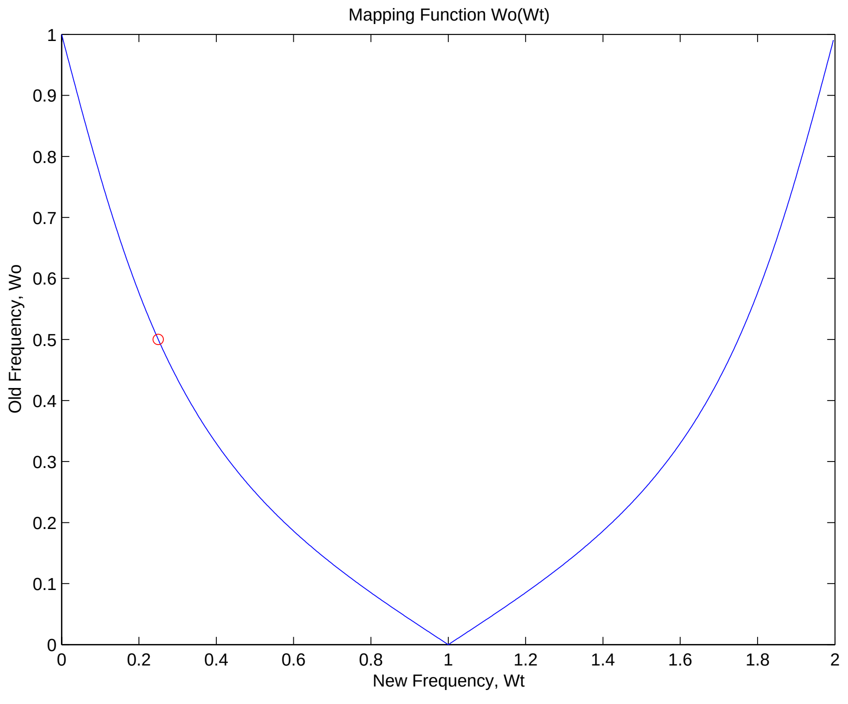
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$. Please note that the transformation works in the same way for both positive and negative frequencies:

```
plot(f/pi, abs(angle(h))/pi, Wt, Wo, 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

For transforming your lowpass filter to an highpass variation, the mapping function shown in this figure does the job.



Arguments

Wo

Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirlp2hp`, `zpklp2hp`

References

[1] Constantinides, A.G., "Spectral transformations for digital filters," *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, "Prototype reference transfer function parameters in the discrete-time frequency transformations," *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, "Closed-form solutions for discrete-time elliptic transfer functions," *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

[4] Constantinides, A.G., "Frequency transformations for digital filters," *Electronics Letters*, vol. 3, no. 11, pp. 487-489, November 1967.

Purpose Return an allpass filter for lowpass to lowpass transformation

Syntax

```
[AllpassNum,AllpassDen] = allpasslp2lp(Wo,Wt)
```

Description

`[AllpassNum,AllpassDen] = allpasslp2lp(Wo,Wt)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the first-order allpass mapping filter for performing a real lowpass to real lowpass frequency transformation. This transformation effectively places one feature of an original filter, located originally at frequency W_o, at the required target frequency location, W_t.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to lowpass transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband and so on.

Lowpass to lowpass transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can change their position in a simple way by applying the lowpass to lowpass transformation.

Examples

Design the allpass filter changing the lowpass filter cutoff frequency originally at W_o=0.5 to W_t=0.25:

```
Wo = 0.5;
Wt = 0.25;
[AllpassNum, AllpassDen] = allpasslp2lp(Wo, Wt);
```

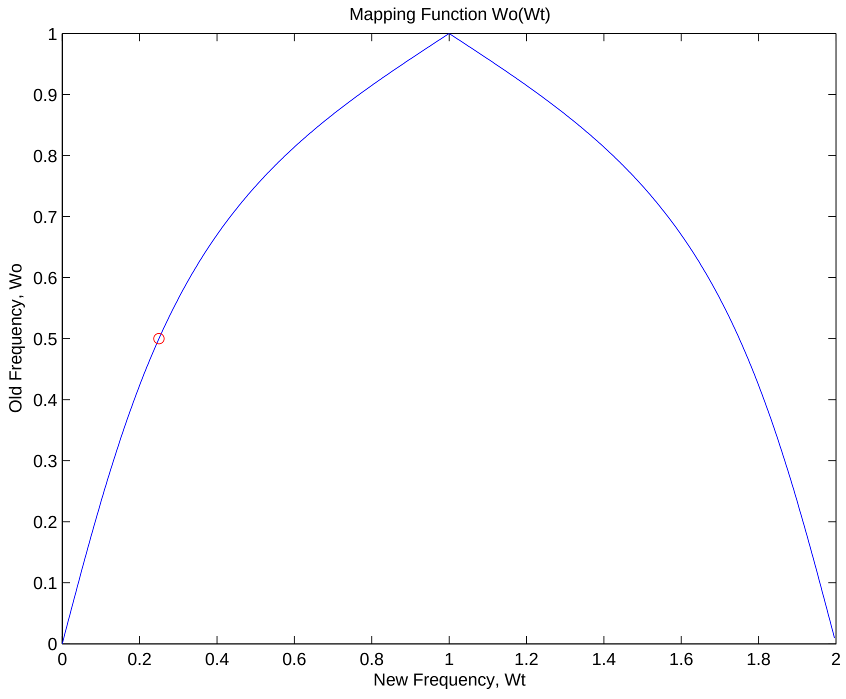
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$. Please note that the transformation works in the same way for both positive and negative frequencies:

```
plot(f/pi, abs(angle(h))/pi, Wt, Wo, 'ro');
```

```
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```



As shown in the figure, `allpasslp2lp` generates a mapping function that converts your prototype lowpass filter to a target lowpass filter with different passband specifications.

Arguments

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirlp2lp`, `zpklp2lp`

References

[1] Constantinides, A.G., "Spectral transformations for digital filters," *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, "Prototype reference transfer function parameters in the discrete-time frequency transformations," *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, "Closed-form solutions for discrete-time elliptic transfer functions," *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

[4] Constantinides, A.G., "Frequency transformations for digital filters," *Electronics Letters*, vol. 3, no. 11, pp. 487-489, November 1967.

Purpose Return an allpass filter for lowpass to M-band transformation

Syntax

```
[AllpassNum,AllpassDen] = allpasslp2mb(Wo,Wt)
[AllpassNum,AllpassDen] = allpasslp2mb(Wo,Wt,Pass)
```

Description `[AllpassNum,AllpassDen] = allpasslp2mb(Wo,Wt)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the `M`th-order allpass mapping filter for performing a real lowpass to real multipassband frequency transformation. Parameter `M` is the number of times an original feature is replicated in the target filter. This transformation effectively places one feature of an original filter, located at frequency W_o, at the required target frequency locations, W_{t1},...,W_{tM}. By default the DC feature is kept at its original location.

`[AllpassNum,AllpassDen]=allpasslp2mb(Wo,Wt,Pass)` allows you to specify an additional parameter, `Pass`, which chooses between using the "DC Mobility" and the "Nyquist Mobility". In the first case the Nyquist feature stays at its original location and the DC feature is free to move. In the second case the DC feature is kept at an original frequency and the Nyquist feature is movable.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations without redesigning them. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

Examples Design the allpass filter changing the real lowpass filter with the cutoff frequency of W_o`=0.5` into a real multiband filter with band edges of W_t`=[1:2:9]/10` precisely defined:

```
Wo = 0.5;
Wt = [1:2:9]/10;
[AllpassNum, AllpassDen] = allpasslp2mb(Wo, Wt);
```

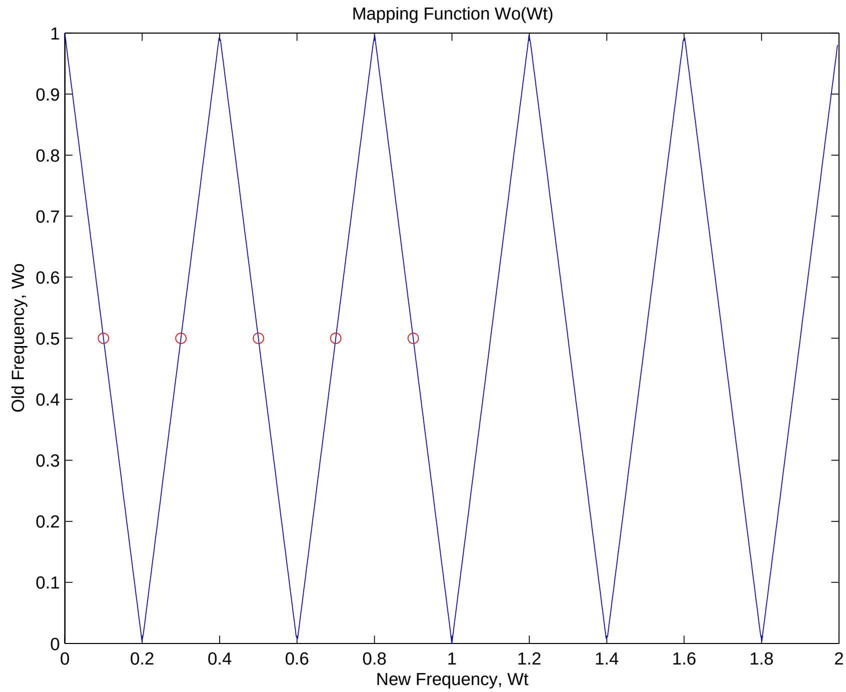
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$. Please note that the transformation works in the same way for both positive and negative frequencies:

```
plot(f/pi, abs(angle(h))/pi, Wt, Wo, 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

As the figure shows, the mapping function, or mapping filter, creates more than one band from your prototype.



Arguments

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`Pass`
Choice (`'pass'`/`'stop'`) of passband/stopband at DC, `` `pass' `` being the default

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirlp2mb`, `zpklp2mb`

References

[1] Franchitti, J.C., “All-pass filter interpolation and frequency transformation problems,” MSc Thesis, Dept. of Electrical and Computer Engineering, University of Colorado, 1985.

[2] Feyh, G., J.C. Franchitti and C.T. Mullis, “All-pass filter interpolation and frequency transformation problem,” *Proceedings 20th Asilomar Conference on Signals, Systems and Computers*, Pacific Grove, California, pp. 164-168, November 1986.

[3] Mullis, C.T. and R.A. Roberts, *Digital Signal Processing*, section 6.7, Reading, Massachusetts, Addison-Wesley, 1987.

[4] Feyh, G., W.B. Jones and C.T. Mullis, *An extension of the Schur Algorithm for frequency transformations, Linear Circuits, Systems and Signal Processing: Theory and Application*, C. J. Byrnes et al Eds, Amsterdam: Elsevier, 1988.

Purpose Return an allpass filter for lowpass to complex M-band transformation

Syntax

```
[AllpassNum,AllpassDen] = allpasslp2mbc(Wo,Wt)
```

Description

[AllpassNum,AllpassDen] = allpasslp2mbc(Wo,Wt) returns the numerator, AllpassNum, and the denominator, AllpassDen, of the Mth-order allpass mapping filter for performing a real lowpass to complex multipassband frequency transformation. Parameter M is the number of times an original feature is replicated in the target filter. This transformation effectively places one feature of an original filter, located at frequency W_o, at the required target frequency locations, W_{t1},...,W_{tM}.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations without the need to design them again. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

Examples

Design the allpass filter changing the real lowpass filter with the cutoff frequency of W_o=0.5 into a complex multiband filter with band edges of W_t=[-3+1:2:9]/10 precisely defined:

```
Wo = 0.5;
Wt = [-3+1:2:9]/10;
[AllpassNum, AllpassDen] = allpasslp2mbc(Wo, Wt);
```

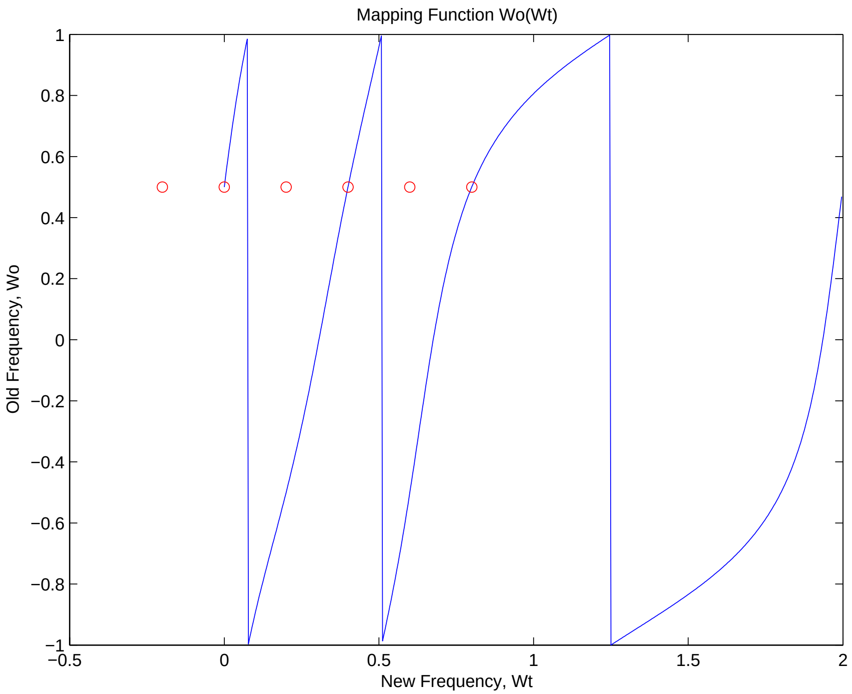
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$. Please note that the transformation works in the same way for both positive and negative frequencies:

```
plot(f/pi, angle(h)/pi, Wt, Wo, 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

In this example, the resulting mapping function converts real filters to multiband complex filters.



Arguments

Wo

Frequency value to be transformed from the prototype filter. It should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wt`
Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also

`iirlp2mbc`, `zpklp2mbc`

allpasslp2xc

Purpose

Return an allpass filter for lowpass to complex N-point transformation

Syntax

```
[AllpassNum,AllpassDen] = allpasslp2xc(Wo,Wt)
```

Description

[AllpassNum,AllpassDen] = allpasslp2xc(Wo,Wt) returns the numerator, AllpassNum, and the denominator, AllpassDen, of the Nth-order allpass mapping filter, where N is the allpass filter order, for performing a real lowpass to complex multipoint frequency transformation. Parameter N also specifies the number of replicas of the prototype filter created around the unit circle after the transformation. This transformation effectively places N features of the, original filter located at frequencies $W_{o1},...,W_{oN}$, at the required target frequency locations, $W_{t1},...,W_{tM}$.

Relative positions of other features of an original filter are the same in the target filter for the Nyquist mobility and are reversed for the DC mobility. For the Nyquist mobility this means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation. For DC mobility feature F_2 will precede F_1 after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones. The only condition is that the features must be selected in such a way that when creating N bands around the unit circle, there will be no band overlap.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

Examples

Design the allpass filter moving four features of an original complex filter given in W_o to the new independent frequency locations W_t. Please note that the transformation creates N replicas of an original filter around the unit circle, where N is the order of the allpass mapping filter:

```
Wo = [-0.2, 0.3, -0.7, 0.4];
Wt = [0.3, 0.5, 0.7, 0.9];
```

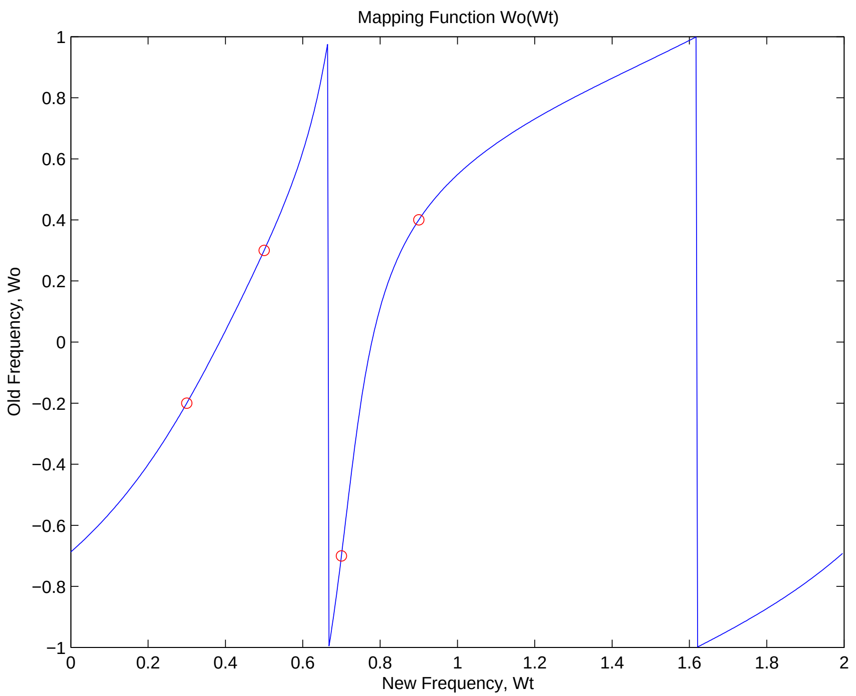
```
[AllpassNum, AllpassDen] = allpasslp2xc(Wo, Wt);
```

Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$:

```
plot(f/pi, angle(h)/pi, Wt, Wo, 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```



As shown, the mapping function copies four features of interest in your prototype to multiple, independent locations in your target filter.

Arguments

`Wo`
Frequency values to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

See Also

`iirlp2xc`, `zpklp2xc`

Purpose Return an allpass filter for lowpass to N-point transformation

Syntax

```
[AllpassNum,AllpassDen] = allpasslp2xn(Wo,Wt)
[AllpassNum,AllpassDen] = allpasslp2xn(Wo,Wt,Pass)
```

Description

`[AllpassNum,AllpassDen] = allpasslp2xn(Wo,Wt)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the `N`th-order allpass mapping filter, where `N` is the allpass filter order, for performing a real lowpass to real multipoint frequency transformation. Parameter `N` also specifies the number of replicas of the prototype filter created around the unit circle after the transformation. This transformation effectively places `N` features of an original filter, located at frequencies $W_{o1},...,W_{oN}$, at the required target frequency locations, $W_{t1},...,W_{tM}$. By default the DC feature is kept at its original location.

`[AllpassNum,AllpassDen]=allpasslp2xn(Wo,Wt,Pass)` allows you to specify an additional parameter, `Pass`, which chooses between using the “DC Mobility” and the “Nyquist Mobility”. In the first case the Nyquist feature stays at its original location and the DC feature is free to move. In the second case the DC feature is kept at an original frequency and the Nyquist feature is movable.

Relative positions of other features of an original filter are the same in the target filter for the Nyquist mobility and are reversed for the DC mobility. For the Nyquist mobility this means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation. For DC mobility feature F_2 will precede F_1 after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones. The only condition is that the features must be selected in such a way that when creating `N` bands around the unit circle, there will be no band overlap.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations without the need of designing them again. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

Arguments

`Wo`
Frequency values to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`Pass`
Choice (`'pass'`/`'stop'`) of passband/stopband at DC, `` `pass' `` being the default

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirlp2xn`, `zpklp2xn`

References

[1] Cain, G.D., A. Krukowski and I. Kale, “High Order Transformations for Flexible IIR Filter Design,” *VII European Signal Processing Conference (EUSIPCO'94)*, vol. 3, pp. 1582-1585, Edinburgh, United Kingdom, September 1994.

[2] Krukowski, A., G.D. Cain and I. Kale, “Custom designed high-order frequency transformations for IIR filters,” *38th Midwest Symposium on Circuits and Systems (MWSCAS'95)*, Rio de Janeiro, Brazil, August 1995.

Purpose

Return an allpass filter for integer upsample transformation

Syntax

```
[AllpassNum,AllpassDen] = allpassrateup(N)
```

Description

`[AllpassNum,AllpassDen] = allpassrateup(N)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the `N`th-order allpass mapping filter for performing the rateup frequency transformation, which creates `N` equal replicas of the prototype filter frequency response.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Examples

Design the allpass filter creating the effect of upsampling the digital filter four times:

```
N = 4;
```

Choose any feature from an original filter, say at W_o=0.2:

```
Wo = 0.2;
Wt = Wo/N + 2*[0:N-1]/N;
[AllpassNum, AllpassDen] = allpassrateup(N);
```

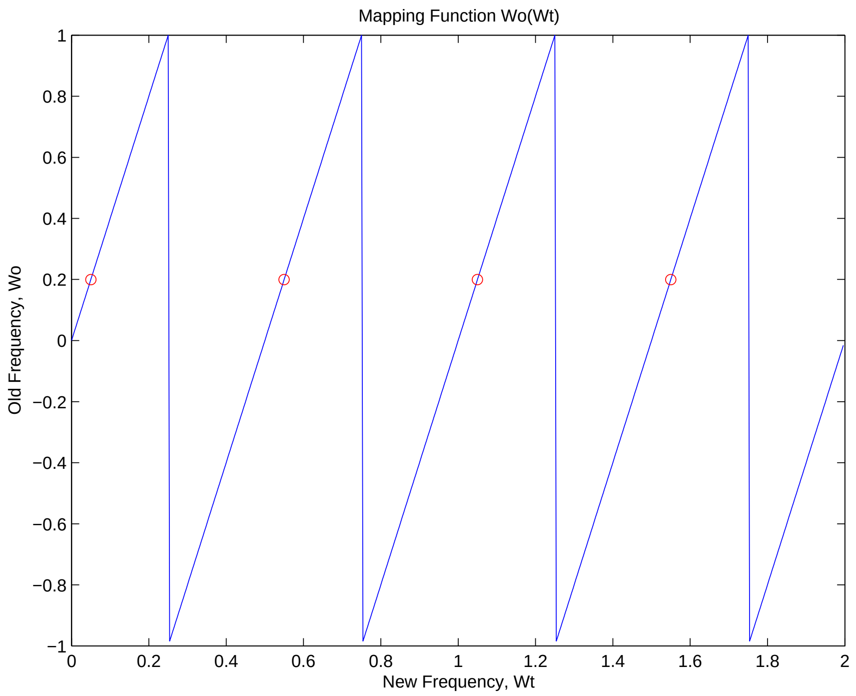
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$:

```
plot(f/pi, angle(h)/pi, Wt, Wo, 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

While this creates the effect of upsampling your prototype filter, compare the results to cicinterp for another approach to upsampling.



Arguments

`N`
Frequency replication ratio (upsampling ratio)

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also

`iirrateup`, `zpkrateup`

Purpose

Return an allpass filter for real shift transformation

Syntax

```
[AllpassNum,AllpassDen] = allpassshift(Wo,Wt)
```

Description

`[AllpassNum,AllpassDen] = allpassshift(Wo,Wt)` returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the second-order allpass mapping filter for performing a real frequency shift transformation. This transformation places one selected feature of an original filter, located at frequency W_o, at the required target frequency location, W_t. This transformation implements the "DC mobility," which means that the Nyquist feature stays at Nyquist, but the DC feature moves to a location dependent on the selection of W_o and W_t.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the real shift transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be moved to a different frequency by applying a shift transformation. In such a way you can avoid designing the filter from the beginning.

Examples

Design the allpass filter precisely shifting one feature of the lowpass filter originally at W_o=0.5 to the new frequencies of W_t=0.25:

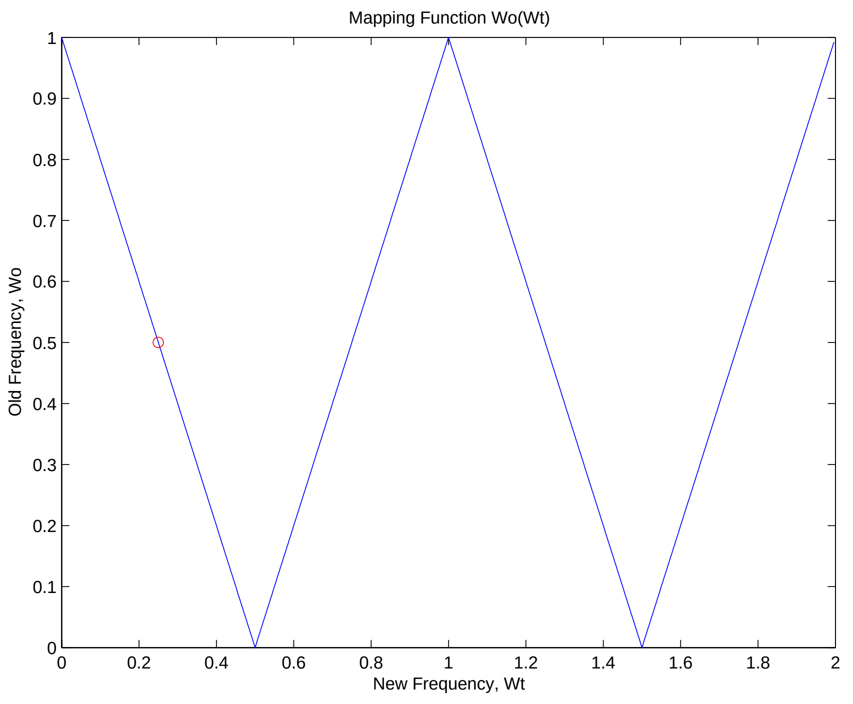
```
Wo = 0.5;
Wt = 0.25;
[AllpassNum, AllpassDen] = allpassshift(Wo, Wt);
```

Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$. Please note that the transformation works in the same way for both positive and negative frequencies:

```
plot(f/pi, abs(angle(h))/pi, Wt, Wo, 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```



Arguments

`Wo`

Frequency value to be transformed from the prototype filter

`Wt`

Desired frequency location in the transformed target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirshift`, `zpkshift`

allpassshiftc

Purpose

Return an allpass filter for complex shift transformation

Syntax

```
[AllpassNum,AllpassDen] = allpassshiftc(Wo,Wt)
```

Description

`[AllpassNum,AllpassDen] = allpassshiftc(Wo,Wt)` returns the numerator, `AllpassNum`, and denominator, `AllpassDen`, vectors of the allpass mapping filter for performing a complex frequency shift of the frequency response of the digital filter by an arbitrary amount.

`[AllpassNum,AllpassDen]=allpassshiftc(0,0.5)` calculates the allpass filter for doing the Hilbert transformation, i.e. a 90 degree counterclockwise rotation of an original filter in the frequency domain.

`[AllpassNum,AllpassDen]=allpassshiftc(0,-0.5)` calculates the allpass filter for doing an inverse Hilbert transformation, i.e. a 90 degree clockwise rotation of an original filter in the frequency domain.

Examples

Design the allpass filter precisely rotating the whole filter by the amount defined by the location of the selected feature from an original filter, W_o=0.5, and its required position in the target filter, W_t=0.25:

```
Wo = 0.5;
Wt = 0.25;
[AllpassNum, AllpassDen] = allpassshiftc(Wo, Wt);
```

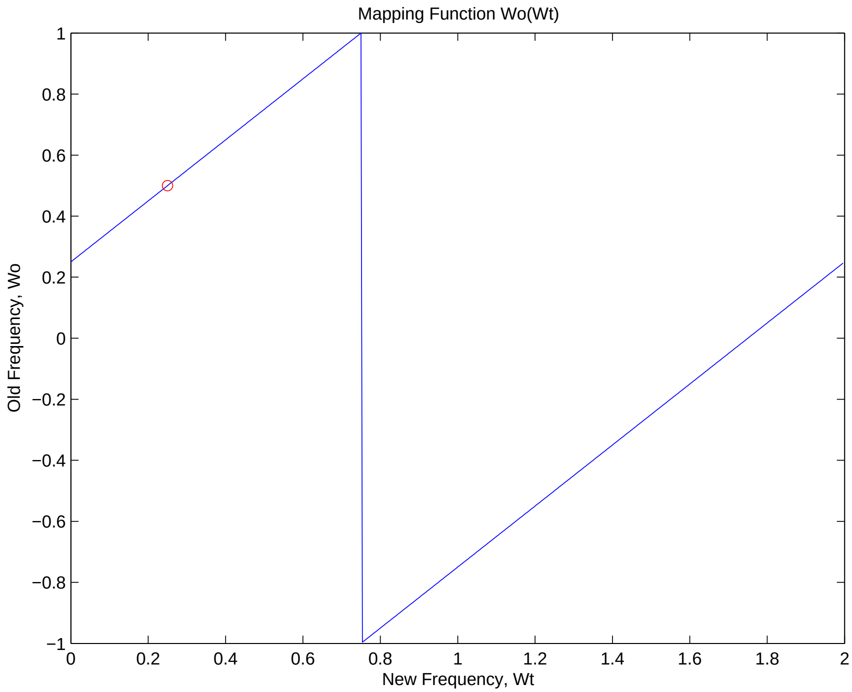
Calculate the frequency response of the mapping filter in the full range:

```
[h, f] = freqz(AllpassNum, AllpassDen, 'whole');
```

Plot the phase response normalized to π, which is in effect the mapping function $W_o(W_t)$:

```
plot(f/pi, angle(h)/pi, Wt, Wo, 'ro');
title('Mapping Function Wo(Wt)');
xlabel('New Frequency, Wt');
ylabel('Old Frequency, Wo');
```

The figure shows you that the transformation by the mapping filter does exactly what you intend.



Arguments

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

See Also

`iirshiftc`, `zpkshiftc`

References

[1] Oppenheim, A.V., R.W. Schafer and J.R. Buck, *Discrete-Time Signal Processing*, Prentice-Hall International Inc., 1989.

[2] Dutta-Roy, S.C. and B. Kumar, "On Digital Differentiators, Hilbert Transformers, and Half-band Low-pass Filters," *IEEE Transactions on Education*, vol. 32, pp. 314-318, August 1989.

Purpose

Generate a Signal Processing Blockset block from a multirate (`mfilt`) filter object

Syntax

```
block(hm)
block(hm,'propertyname1',propertyvalue1,'propertyname2',...
   propertyvalue2,...)
```

Description

`block(hm)` generates a Signal Processing Blockset block equivalent to `hm`.

`block(hm,'propertyname1',propertyvalue1,'propertyname2',... propertyvalue2,...)` generates a Signal Processing Blockset block using the options specified in the propertyname/propertyvalue pairs. The valid properties and their values are

Property Name	Description and Values
`Destination`	Determines which Simulink model gets the block. Choose either `current` or `new`. Specifying `new` opens a new Simulink model and adds the block. `Current` adds the block to your current Simulink model. `Current` is the default setting.
`Blockname`	Specifies the name of the generated block. The name appears below the block in the model. When you do not specify a block name, the default is `filter`.

Property Name	Description and Values
OverwriteBlock	Tells block whether to overwrite an existing block of the same name, or create a new block. Off is the default setting—block does not overwrite existing blocks with matching names. Switching from off to on directs block to overwrite existing blocks.
MapStates	Specifies whether to apply the current filter states to the new block. This lets you save states from a filter object you may have used or configured in a specific way. The default setting of off means the states are not transferred to the block. Choosing on preserves the current filter states in the block.

Examples

Two examples of using block demonstrate the syntax capabilities. Both examples start from an mfilt object with interpolation factor of 3. In the first example, use block with the default syntax, letting the function determine the block name and configuration.

```
l = 3; % Interpolation factor
hm = mfilt.firinterp(l);
```

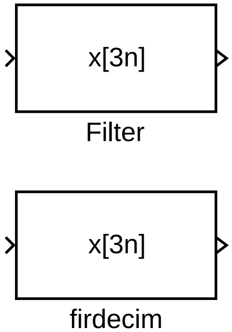
Now use the default syntax to create a block.

```
block(hm);
```

In this second example, define the block name to meet your needs by using property name/property value pairs.

```
block(hm, 'blockname', 'firinterp');
```

The figure below shows the blocks in a Simulink model. When you try these examples, you see that the second block writes over the first block location. You can avoid this by moving the first block before you generate the second, or setting the Destination property to new which puts the filter block in a new Simulink model.



See Also Refer to the Realize Model option in FDATool, and `realizemdl`

Purpose

Design Butterworth IIR digital filters using the specifications in a filter design object

Syntax

`hd = butter(d)`
`hd = butter(d,'`**`matchexactly`**`',`*`match`*`)`

Description

`hd = butter(d)` designs a Butterworth IIR digital filter using the specifications supplied in the object `d`.

`hd = butter(d,'`**`matchexactly`**`',`*`match`*`)` returns a Butterworth IIR filter where the filter response matches the specified response exactly for one filter band. `match`, which specifies which filter band to match, is either

- `passband`—match the passband specification exactly in the final filter.
- `stopband`—match the specified stopband performance exactly in the final filter. This is the default setting.

Use the **`matchexactly`** input option only when your filter object designs minimum order Butterworth filters. This condition applies when you do not specify the filter order in your filter design object. Lowpass, highpass, bandpass, and bandstop filter design objects support a `SpecificationType` string that does not include the filter order as an input specification.

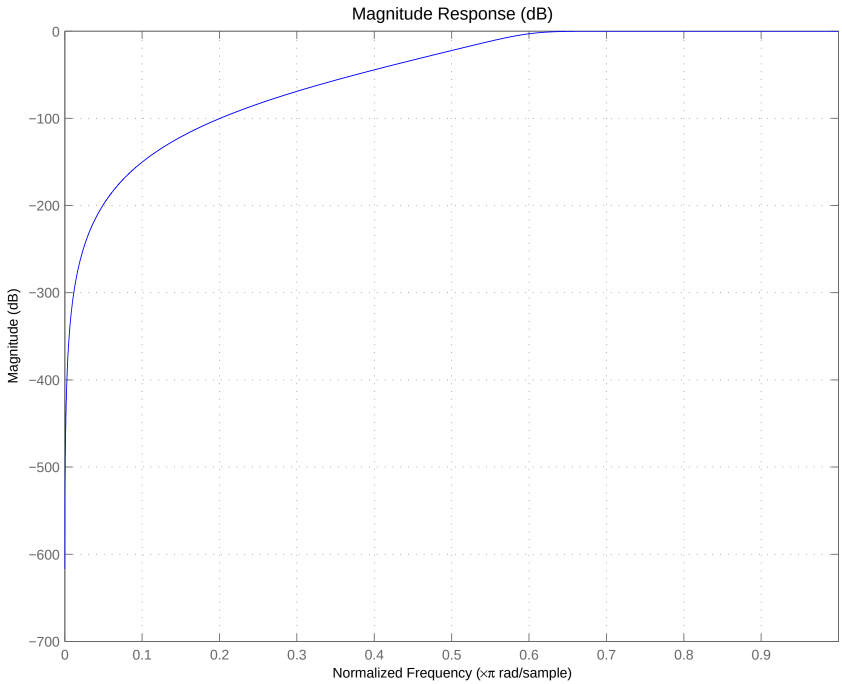
Examples

The first example constructs the default lowpass filter design object and uses it to design a Butterworth filter.

```
d = fdesign.lowpass;
hd = butter(d,'matchexactly','passband');
```

Example 2 constructs a highpass filter design object with order (`n`) and cutoff frequency (`fc`) specifications, and then designs a Butterworth filter from the object.

```
d = fdesign.highpass('n,fc',8,.6);
butter(d);
```



See Also

`cheby1`, `cheby2`, `ellip`

ca2tf

Purpose Convert coupled allpass filter form to transfer function forms

Syntax

```
[b,a] = ca2tf(d1,d2)
[b,a] = ca2tf(d1,d2,beta)
[b,a,bp] = ca2tf(d1,d2)
[b,a,bp] = ca2tf(d1,d2,beta)
```

Description

`[b,a]=ca2tf(d1,d2)` returns the vector of coefficients `b` and the vector of coefficients `a` corresponding to the numerator and the denominator of the transfer function

$$H(z) = B(z)/A(z) = \frac{1}{2}[H1(z) + H2(z)]$$

`d1` and `d2` are real vectors corresponding to the denominators of the allpass filters `H1(z)` and `H2(z)`.

`[b,a]=ca2tf(d1,d2,beta)` where `d1`, `d2` and `beta` are complex, returns the vector of coefficients `b` and the vector of coefficients `a` corresponding to the numerator and the denominator of the transfer function

$$H(z) = B(z)/A(z) = \frac{1}{2}[-(\bar{\beta}) \bullet H1(z) + \beta \bullet H2(z)]$$

`[b,a,bp]=ca2tf(d1,d2)`, where `d1` and `d2` are real, returns the vector `bp` of real coefficients corresponding to the numerator of the power complementary filter G(z)

$$G(z) = Bp(z)/A(z) = \frac{1}{2}[H1(z) - H2(z)]$$

`[b,a,bp]=ca2tf(d1,d2,beta)`, where `d1`, `d2` and `beta` are complex, returns the vector of coefficients `bp` of real or complex coefficients that correspond to the numerator of the power complementary filter G(z)

$$G(z) = Bp(z)/A(z) = \frac{1}{2j}[-(\bar{\beta}) \bullet H1(z) + \beta \bullet H2(z)]$$

Examples

Create a filter, convert the filter to coupled allpass form, and convert the result back to the original structure (create the power complementary filter as well).

```
[b,a]=cheby1(10,.5,.4);
[d1,d2,beta]=tf2ca(b,a);           % tf2ca returns the
                                   % denominators of the
                                   % allpasses.
[num,den,numpc]=ca2tf(d1,d2,beta); % Reconstruct the original
                                   % filter plus the power
                                   % complementary one.
[h,w,s]=freqz(num,den);
hpc = freqz(numpc,den);
s.plot = 'mag';
s.yunits = 'sq';
freqzplot([h hpc],w,s);            % Plot the mag response of the
                                   % original filter and the
                                   % power complementary one.
```

See Also

`cl2tf`, `iirpowcomp`, `tf2ca`, `tf2cl`

Purpose

Design Chebyshev Type I digital filter using the specifications in a filter design object

Syntax

```
hd = cheby1(d)
hd = cheby1(d,'matchexactly',match)
```

Description

`hd = cheby1(d)` designs a Chebyshev I IIR digital filter using the specifications supplied in the object `d`.

`hd = cheby1(d,'`**`matchexactly`**`',`*`match`*`)` returns a Chebyshev I IIR filter where the filter response matches the specified response exactly for one filter band. `match`, which specifies which filter band to match, is either

- `passband`—match the passband specification exactly in the final filter.
- `stopband`—match the specified stopband performance exactly in the final filter. This is the default setting.

Use the **`matchexactly`** input option only when your filter object designs minimum order Chebyshev filters. This condition applies when you do not specify the filter order in your filter design object. Lowpass, highpass, bandpass, and bandstop filter design objects support a `SpecificationType` string that does not include the filter order as an input argument.

Examples

These examples use filter design objects to construct Chebyshev type I filters. In the first example, you use the `matchexactly` option to ensure the performance of the filter in the passband.

```
d = fdesign.lowpass
hd = cheby1(d,'matchexactly','passband')

d =

           ResponseType: 'Minimum-order lowpass'
      SpecificationType: 'Fp,Fst,Ap,Ast'
            Description: {4x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
                  Fpass: 0.4500
                  Fstop: 0.5500
                  Apass: 1
```

```
                  Astop: 60


hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
               Arithmetic: 'double'
                sosMatrix: [5x6 double]
              ScaleValues: [6x1 double]
         PersistentMemory: false
                   States: [2x5 double]
      NumSamplesProcessed: 0
```

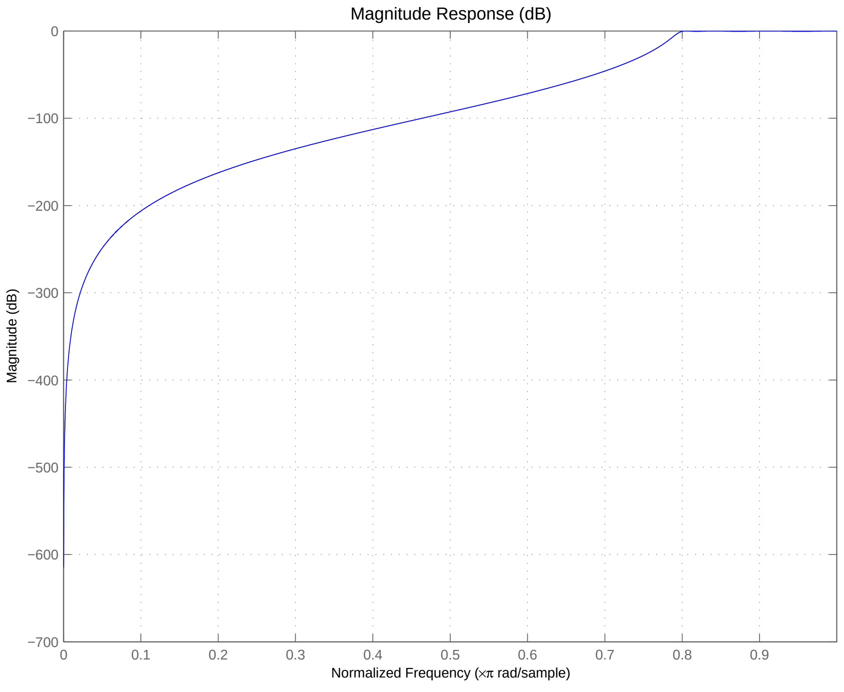
cheby1 also design highpass filters, among others. Specify the filter order, passband edge frequency. and the passband ripple to get the filter exactly as required.

```
d = fdesign.highpass('n,fp,ap',7,20,.4,50)
cheby1(d)

d =

            ResponseType: 'Highpass with passband-edge
specifications'
       SpecificationType: 'N,Fp,Ap'
             Description: {3x1 cell}
     NormalizedFrequency: false
                      Fs: 50
             FilterOrder: 7
                   Fpass: 20
                   Apass: 0.4000


hd=cheby1(d)

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
               Arithmetic: 'double'
```

```
               sosMatrix: [4x6 double]
             ScaleValues: [5x1 double]
        PersistentMemory: false
                  States: [2x4 double]
     NumSamplesProcessed: 0
```

Use `fvtool` to view the resulting filter.

```
fvtool(hd)
```



By design, `cheby1` returns filters that use second-order sections. For many applications, and for most fixed-point applications, SOS filters are particularly well-suited for use.

See Also

`butter`, `cheby2`, `ellip`

Purpose

Design Chebyshev Type II digital filter using the specifications in a filter design object

Syntax

```
hd = cheby2(d)
hd = cheby2(d,'matchexactly',match)
```

Description

`hd = cheby2(d)` designs a Chebyshev II IIR digital filter using the specifications supplied in the object `d`.

`hd = cheby2(d,'matchexactly',match)` returns a Chebyshev II IIR filter where the filter response matches the specified response exactly for one filter band. `match`, which specifies which filter band to match, is either

- `passband`—match the passband specification exactly in the final filter.
- `stopband`—match the specified stopband performance exactly in the final filter. This is the default setting.

Use the **`matchexactly`** input option only when your filter object designs minimum order Chebyshev filters. This condition applies when you do not specify the filter order in your filter design object. Lowpass, highpass, bandpass, and bandstop filter design objects support a `SpecificationType` string that does not include the filter order as an input argument.

Examples

These examples use filter design objects to construct Chebyshev type I filters. In the first example, you use the `matchexactly` option to ensure the performance of the filter in the passband.

```
d = fdesign.lowpass
hd = cheby2(d,'matchexactly','passband')

d =

           ResponseType: 'Minimum-order lowpass'
      SpecificationType: 'Fp,Fst,Ap,Ast'
            Description: {4x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
                  Fpass: 0.4500
                  Fstop: 0.5500
                  Apass: 1
```

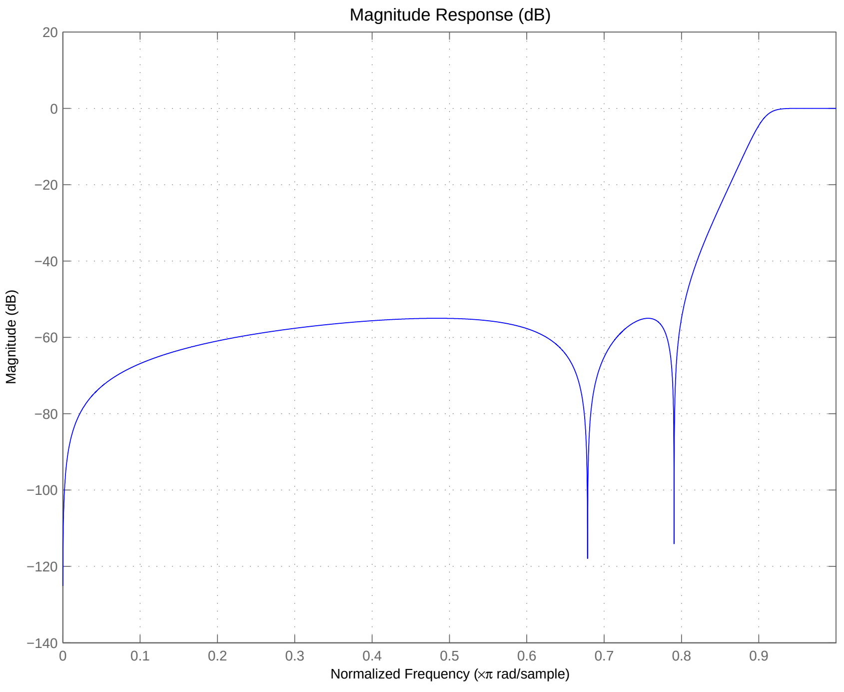
```
                 Astop: 60


hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
               Arithmetic: 'double'
                sosMatrix: [5x6 double]
              ScaleValues: [6x1 double]
         PersistentMemory: false
                   States: [2x5 double]
     NumSamplesProcessed: 0
```

cheby2 also design highpass, bandpass, and bandstop filters. Here is a highpass filter where you specify the filter order, the stopband edge frequency. and the stopband attenuation to get the filter exactly as required.

```
d = fdesign.highpass('n,fst,ast',5,20,55,50)

d =

           ResponseType: 'Highpass with stopband-edge specifications'
      SpecificationType: 'N,Fst,Ast'
            Description: {3x1 cell}
    NormalizedFrequency: false
                     Fs: 50
            FilterOrder: 5
                  Fstop: 20
                  Astop: 55


hd=cheby2(d)

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [3x6 double]
             ScaleValues: [4x1 double]
        PersistentMemory: false
                  States: [2x3 double]
     NumSamplesProcessed: 0
```

The Filter Visualization Tool shows the highpass filter meets the specifications.

```
fvtool(hd)
```



By design, `cheby2` returns filters that use second-order sections. For many applications, and for most fixed-point applications, SOS filters are particularly well-suited for use.

See Also

`butter`, `cheby1`, `ellip`

Purpose Convert coupled allpass lattice to transfer function form

Syntax

```
[b,a] = cl2tf(k1,k2)
[b,a] = cl2tf(k1,k2,beta)
[b,a,bp] = cl2tf(k1,k2)
[b,a,bp] = cl2tf(k1,k2,beta)
```

Description

`[b,a] = cl2tf(k1,k2)` returns the numerator and denominator vectors of coefficients `b` and `a` corresponding to the transfer function

$$H(z) = B(z)/A(z) = \frac{1}{2}[H1(z) + H2(z)]$$

where *H1(z)* and *H2(z)* are the transfer functions of the allpass filters determined by `k1` and `k2`, and `k1` and `k2` are real vectors of reflection coefficients corresponding to allpass lattice structures.

`[b,a] = cl2tf(k1,k2,beta)` where `k1`, `k2` and `beta` are complex, returns the numerator and denominator vectors of coefficients `b` and `a` corresponding to the transfer function

$$H(z) = B(z)/A(z) = \frac{1}{2}[-(\bar{\beta}) \bullet H1(z) + \beta \bullet H2(z)]$$

`[b,a,bp] = cl2tf(k1,k2)` where `k1` and `k2` are real, returns the vector `bp` of real coefficients corresponding to the numerator of the power complementary filter *G(z)*

$$G(z) = Bp(z)/A(z) = \frac{1}{2}[H1(z) - H2(z)]$$

`[b,a,bp] = cl2tf(k1,k2,beta)` where `k1`, `k2` and `beta` are complex, returns the vector of coefficients `bp` of possibly complex coefficients corresponding to the numerator of the power complementary filter *G(z)*

$$G(z) = Bp(z)/A(z) = \frac{1}{2j}[-(\bar{\beta}) \bullet H1(z) + \beta \bullet H2(z)]$$

Examples

```
[b,a]=cheby1(10,.5,.4);
[k1,k2,beta]=tf2cl(b,a); %TF2CL returns the reflection coeffs
```

```
% Reconstruct the original filter
% plus the power complementary one.
[num,den,numpc]=cl2tf(k1,k2,beta);
[h,w,s1]=freqz(num,den);
hpc = freqz(numpc,den);
s.plot = 'mag';
s.yunits = 'sq';
% Plot the mag response of the original filter and the power
% complementary one.
freqzplot([h hpc],w,s1);
```

See Also

`tf2cl`, `tf2ca`, `ca2tf`, `tf2latc`, `latc2tf`, `iirpowcomp`

Purpose

Return the filter coefficients for adaptive filters, discrete-time filters, and multirate filters

Syntax

```
c = coefficients(ha)
coefficients(ha)
c = coefficients(hd)
coefficients(hd)
c = coefficients(hm)
coefficients(hm)
```

Description

The next sections describe common `coefficients` operation with adaptive, discrete-time, and multirate filters.

Adaptive Filters

`c = coefficients(ha)` returns a cell array `c` containing the coefficients of adaptive filter `ha`. These are the instantaneous filter coefficients available at the time you use the function.

`coefficients(ha)` without an output argument opens FVTool in the coefficients analysis mode displaying the filter coefficients.

Discrete-Time Filters

`c = coefficients(hd)` returns a cell array `c` that contains the coefficients of discrete-time filter `hd`.

`coefficients(hd)` without an output argument opens FVTool in the coefficients analysis mode displaying the filter coefficients.

Multirate Filters

`c = coefficients(hm)` returns `c`, a cell array containing the coefficients of discrete-time filter `hm`. CIC-based filters do not have coefficients and this function does not work with constructors like `mfilt.cicdecim`.

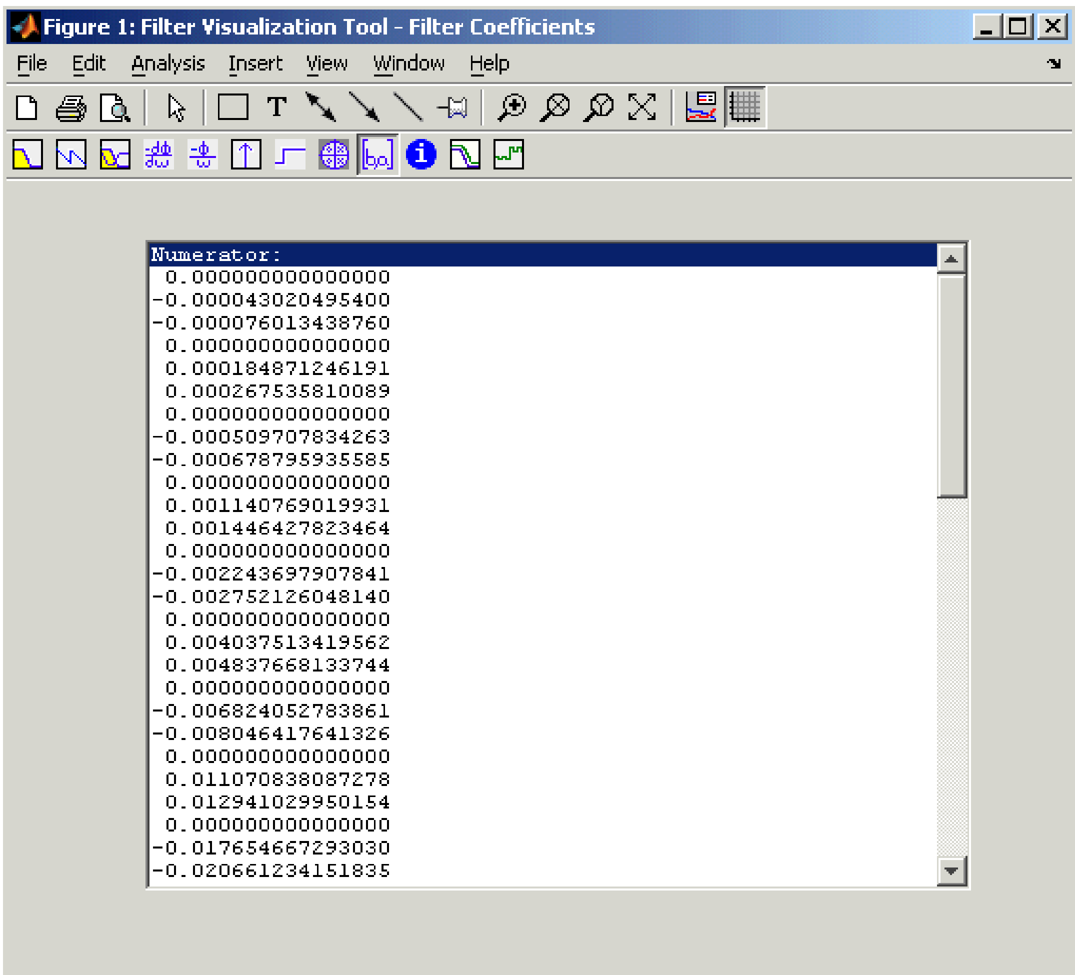
`coefficients(hm)` with no output argument opens FVTool in the coefficients analysis mode displaying the filter coefficients.

Examples

`coefficients` works the same way for all filters. This example uses a multirate filter `hm` to demonstrate the function.

```
hm=mfilt.firdecim(3)

hm =

         FilterStructure: 'Direct-Form FIR Polyphase Decimator'
               Numerator: [1x72 double]
        DecimationFactor: 3
        PersistentMemory: false
                  States: [69x1 double]
     NumSamplesProcessed: 0

c=coefficients(hm)

c =

    [1x72 double]

c{1}

ans =

  Columns 1 through 8

        0   -0.0000   -0.0001         0    0.0002    0.0003         0   -0.0005

  Columns 9 through 16

  -0.0007         0    0.0011    0.0014         0   -0.0022   -0.0028         0

  Columns 17 through 24

   0.0040    0.0048         0   -0.0068   -0.0080         0    0.0111    0.0129

  Columns 25 through 32

        0   -0.0177   -0.0207         0    0.0287    0.0342         0   -0.0513

  Columns 33 through 40

  -0.0659         0    0.1363    0.2749    0.3333    0.2749    0.1363         0

  Columns 41 through 48

  -0.0659   -0.0513         0    0.0342    0.0287         0   -0.0207   -0.0177

  Columns 49 through 56

        0    0.0129    0.0111         0   -0.0080   -0.0068         0    0.0048

  Columns 57 through 64
```

```
   0.0040         0   -0.0028   -0.0022         0    0.0014    0.0011         0

  Columns 65 through 72

  -0.0007   -0.0005         0    0.0003    0.0002         0   -0.0001   -0.0000

coefficients(hm)
```



See Also

`adaptfilt`, `freqz`, `grpdelay`, `impz`, `info`, `phasez`, `stepz`, `zerophase`, `zplane`

Purpose Read a XILINX CORE Generator™ coefficient (.COE) file

Syntax

```
hd = coeread('filename')
```

Description `hd = coeread(filename)` extracts the Distributed Arithmetic FIR filter coefficients defined in the XILINX CORE Generator .COE file specified by `filename`. It returns a `dfilt` object, the fixed-point filter `hd`. If you do not provide the file type extension `.coe` with the `filename`, the function assumes the `.coe` extension.

See Also `coewrite`, `dfilt`, `dfilt.dffir`

Purpose

Write a Xilinx CORE Generator™ coefficient (.COE) file

Syntax

```
coewrite(hd)
coewrite(hd,radix)
coewrite(...,filename)
```

Description

`coewrite(hd)` writes a XILINX Distributed Arithmetic FIR filter coefficient .COE file which can be loaded into the XILINX CORE Generator. The coefficients are extracted from the fixed-point `dfilt` object `hd`. Your fixed-point filter must be a direct form FIR structure `dfilt` object with one section and whose `Arithmetic` property is set to `fixed`. You cannot export single-precision, double-precision, or floating-point filters as `.coe` files, nor multiple-section filters. To enable you to provide a name for the file, `coewrite` displays a dialog where you fill in the file name. If you do not specify the name of the output file, the default file name is `untitled.coe`.

`coewrite(hd,radix)` indicates the radix (number base) used to specify the FIR filter coefficients. Valid `radix` values are `2` for binary, `10` for decimal, and `16` for hexadecimal (default).

`coewrite(...,filename)` writes a XILINX.COE file to `filename`. If you omit the file extension, `coewrite` adds the `.coe` extension to the name of the file.

Examples

`coewrite` generates an ASCII text file that contains the filter coefficients in a format the XILINX CORE Generator can read and load. In this example, you create a 30th-order fixed-point filter and generate the `.coe` file that include the filter coefficients as well as associated information about the filter.

```
b = firceqrip(30,0.4,[0.05 0.03]);
hq = dfilt.dffir(b);
set(hq,'arithmetic','fixed');
coewrite(hq,10,'mycoefile');
```

When you look at `mycoefile.coe`, you see the following:

```
;
; XILINX CORE Generator(tm) Distributed Arithmetic FIR filter
coefficient (.COE) File
; Generated by MATLAB(tm) and the Filter Design Toolbox.
;
```

```
; Generated on: 4-Dec-2003 13:47:15
;
Radix = 10;
Coefficient_Width = 16;
CoefData =   -41,
 -851,
 -366,
  308,
  651,
   22,
 -873,
 -658,
  749,
 1504,
   21,
-2367,
-2012,
 3014,
 9900,
....
```

`coewrite` puts the filter coefficients in column-major order and reports the radix, the coefficient width, and the coefficients. These represent the minimum set of data needed in a `.coe` file.

See Also

`coeread`, `dfilt`, `dfilt.dffir`

Purpose Convert filter structures of discrete-time and multirate filters

Syntax

```
hq = convert(hq,newstruct)
hm = convert(hm,newstruct)
```

Description

Discrete-Time Filters

`hq = convert(hq,newstruct)` returns a quantized filter whose structure has been transformed to the filter structure specified by string `newstruct`. You can enter any one of the following quantized filter structures:

- `'antisymmetricfir'`: Antisymmetric finite impulse response (FIR).
- `'df1'`: Direct form I.
- `'df1t'`: Direct form I transposed.
- `'df2'`: Direct form II.
- `'df2t'`: Direct form II transposed. Default filter structure.
- `'fir'`: FIR.
- `'firt'`: Direct form FIR transposed.
- `'latcallpass'`: Lattice allpass.
- `'latticeca'`: Lattice coupled-allpass.
- `'latticecapc'`: Lattice coupled-allpass power-complementary.
- `'latticear'`: Lattice autoregressive (AR).
- `'latticema'`: Lattice moving average (MA) minimum phase.
- `'latcmax'`: Lattice moving average (MA) maximum phase.
- `'latticearma'`: Lattice ARMA.
- `'statespace'`: Single-input/single-output state-space.
- `'symmetricfir'`: Symmetric FIR. Even and odd forms.

All filters can be converted to the following structures:

- `df1`
- `df1t`
- `df2`
- `df2t`
- `statespace`
- `latticearma`

For the following filter classes, you can specify other conversions as well:

- Minimum phase FIR filters can be converted to `latticema`
- Maximum phase FIR filters can be converted to `latcmax`
- Allpass filters can be converted to `latcallpass`

`convert` generates an error when you specify a conversion that is not possible.

Multirate Filters

`hm = convert(hm,newstruct)` returns a multirate filter whose structure has been transformed to the filter structure specified by string `newstruct`. You can enter any one of the following multirate filter structures, defined by the strings shown, for `newstruct`:

Cascaded Integrator-Comb Structures

- `cicdecim`—CIC-based decimator
- `cicdecimzerolat`—CIC-based decimator that exhibits no latency
- `cicinterp`—CIC-based interpolator
- `cicinterpzerolat`—CIC-based interpolater that does not induce latency

FIR Structures

- `firdecim`—FIR decimator
- `firtdecim`—transposed FIR decimator
- `firfracdecim`—FIR fractional decimator
- `firinterp`—FIR interpolator
- `firfracinterp`—FIR fractional interpolator
- `firsrc`—FIR sample rate change filter
- `firholdinterp`—FIR interpolator that uses hold interpolation between input samples
- `firlinearinterp`—FIR interpolator that uses linear interpolation between input samples
- `fftfirinterp`—FFT-based FIR interpolator

You cannot convert between the FIR and CIC structures.

Examples

```
[b,a]=ellip(5,3,40,.7);
hq = dfilt.df2t(b,a);
hq2 = convert(hq,'df1')
hq2 =

         FilterStructure: 'Direct-Form I'
              Arithmetic: 'double'
               Numerator: [0.1980 0.7886 1.4236 1.4236 0.7886 0.1980]
             Denominator: [1 1.4339 1.8021 0.6139 0.2047 -0.2342]
        PersistentMemory: false
                  States: Numerator:  [5x1 double]
                          Denominator:[5x1 double]
     NumSamplesProcessed: 0
```

For an example of changing the structure of a multirate filter, try the following conversion from a CIC interpolator to a CIC interpolator with zero latency.

```
hm = mfilt.cicinterp(2,2,3,8,8)

hm =

         FilterStructure: 'Cascaded Integrator-Comb Interpolator'
              Arithmetic: 'int'
       DifferentialDelay: 2
        NumberOfSections: 3
     InterpolationFactor: 2
               RoundMode: 'floor'
        PersistentMemory: false
                  States: Integrator: [3x1 States]
                                Comb: [3x1 States]
     NumSamplesProcessed: 0

         InputWordLength: 8

   SectionWordLengthMode: 'MinWordLengths'

        OutputWordLength: 8

hm2=convert(hm,'cicinterpzerolat')

hm2 =

         FilterStructure: 'Zero-Latency Cascaded Integrator-Comb Interpolator'
              Arithmetic: 'int'
       DifferentialDelay: 2
        NumberOfSections: 3
     InterpolationFactor: 2
               RoundMode: 'floor'
        PersistentMemory: false
```

```
                 States: Integrator: [3x1 States]
                               Comb: [3x1 States]
    NumSamplesProcessed: 0

        InputWordLength: 8

  SectionWordLengthMode: 'MinWordLengths'

       OutputWordLength: 8
```

See Also

`mfilt`

`dfilt` in the Signal Processing Toolbox documentation

Purpose

Return a vector of filters for the cumulative sections

Syntax

```
h = cumsec(hd)
h = cumsec(hd,indices)
h = cumsec(hd,indices,secondary)
cumsec(hd)
```

Description

`h = cumsec(hd)` returns a vector `h` of SOS filter objects with the cumulative sections. Each element in `h` is a filter with the structure of the original filter. The first element is the first filter section of `hd`. The second element of `h` is a filter that represents the combination of the first and second sections of `hd`. The third element of `h` is a filter which combines sections 1, 2 and 3 of `hd`. this pattern continues until the final element of `h` contains all the sections of `hd` and should be identical to `hd`.

`h = cumsec(hd,indices)` returns a vector `h` of SOS filter objects whose indices into the original filter are in the vector `indices`. Now you can specify the filter sections `cumsec` uses to compute the cumulative responses.

`h = cumsec(hd,indices,secondary)` when `secondary` is true, `cumsec` uses the secondary scaling points in the sections to determine where the sections should be split. This option applies only when `hd` is a `df2sos` and `df1tsos` filter. For these second-order section structures, the secondary scaling points refer to the scaling locations between the recursive and the nonrecursive parts of the section (the "middle" of the section). Argument `secondary` accepts either `true` or `false`. By default, `secondary` is `false`.

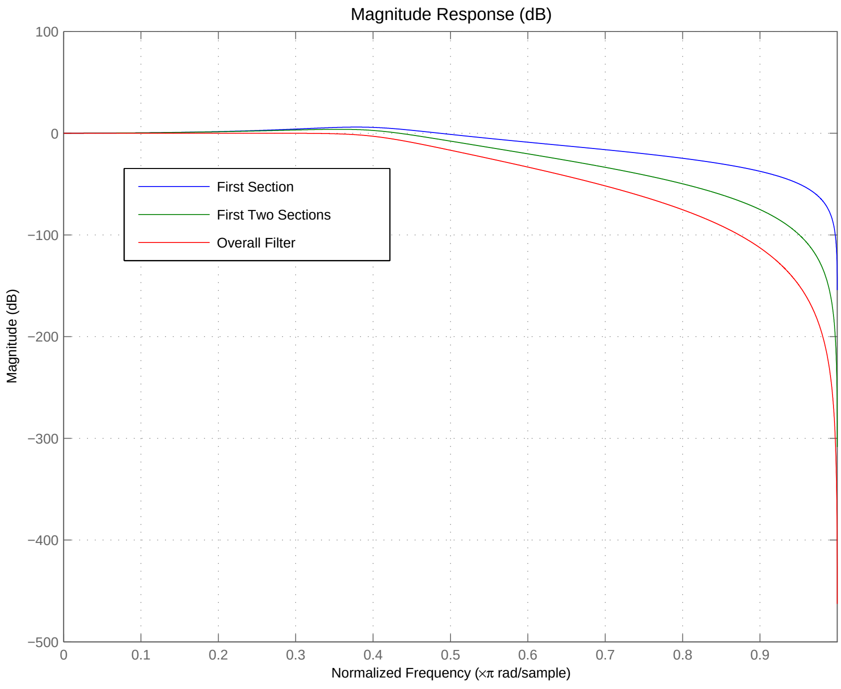
`cumsec(hd,...)` without an output arguments uses FVTool to plot the magnitude response of the cumulative sections.

Examples

To demonstrate how `cumsec` works, this example plots the relative responses of the sections of a sixth-order filter SOS filter with three sections. Each curve adds one more section to form the filter response.

```
hs = fdesign.lowpass('n,fc',6,.4);
hd = butter(hs);
h = cumsec(hd);
hfvt = fvtool(h);
```

```
legend(hfvt,'First Section','First Two Sections','Overall
Filter');
```



See Also

`scale`, `scalecheck`

Purpose

Reverse the filter coefficient and gain changes caused by the function normalize

Syntax

`denormalize(hq)`

Description

`denormalize(hq)` reverses the coefficient changes you make when you use `normalize` with `hq`. The filter coefficients do not change if you call `denormalize(hq)` before you use `normalize(hq)`. Calling `denormalize` more than once on a filter does not change the coefficients after the first `denormalize` call.

Examples

Make a quantized filter `hq` and normalize the filter coefficients. After normalizing the coefficients, restore them to their original values by reversing the effects of the `normalize` function.

```
d=
.highpass('n,fc',14,0.45)

d =

           ResponseType: 'Highpass with cutoff'
      SpecificationType: 'N,Fc'
            Description: {2x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
            FilterOrder: 14
                Fcutoff: 0.4500


hd=butter(d)

hd =

        FilterStructure: 'Direct-Form II, Second-Order Sections'
             Arithmetic: 'double'
              sosMatrix: [7x6 double]
            ScaleValues: [8x1 double]
       PersistentMemory: false
                 States: [2x7 double]
    NumSamplesProcessed: 0
```

```
hd.arithmetic='fixed'

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'fixed'
               sosMatrix: [7x6 double]
             ScaleValues: [8x1 double]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

          CoeffWordLength: 16
           CoeffAutoScale: true
                  Signed: true

          InputWordLength: 16
          InputFracLength: 15

    StageInputWordLength: 16
     StageInputAutoScale: true

   StageOutputWordLength: 16
    StageOutputAutoScale: true

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

         StateWordLength: 16
         StateFracLength: 15

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
```

```
            OverflowMode: 'wrap'

hq=hd;
g=normalize(hq)

g =

     2
     2
     2
     2
     2
     2
     2

hq.SosMatrix

ans =

    0.5000   -1.0000    0.5000    1.0000   -0.2817    0.8008
    0.5000   -1.0000    0.5000    1.0000   -0.2359    0.5081
    0.5000   -1.0000    0.5000    1.0000   -0.2051    0.3110
    0.5000   -1.0000    0.5000    1.0000   -0.1842    0.1776
    0.5000   -1.0000    0.5000    1.0000   -0.1704    0.0892
    0.5000   -1.0000    0.5000    1.0000   -0.1619    0.0350
    0.5000   -1.0000    0.5000    1.0000   -0.1579    0.0093

denormalize(hq)
hq.SosMatrix

ans =

    1.0000   -2.0000    1.0000    1.0000   -0.2817    0.8008
    1.0000   -2.0000    1.0000    1.0000   -0.2359    0.5081
    1.0000   -2.0000    1.0000    1.0000   -0.2051    0.3110
    1.0000   -2.0000    1.0000    1.0000   -0.1842    0.1776
    1.0000   -2.0000    1.0000    1.0000   -0.1704    0.0892
    1.0000   -2.0000    1.0000    1.0000   -0.1619    0.0350
    1.0000   -2.0000    1.0000    1.0000   -0.1579    0.0093
```

See Also `normalize`

Purpose

Return the filter design methods available for designing a filter from a filter design object

Syntax

```
designmethods(d)
```

Description

`designmethods(d)` returns a list of the design methods available for the filter design object `d` with its `SpecificationType`. When you change the `SpecificationType` for a filter design object, the methods available to design filters from the object change.

Examples

Construct a lowpass object and determine the design methods available to design a filter from the object.

```
d=fdesign.lowpass('n,fc',10,12000,48000)

d =

             ResponseType: 'Lowpass with cutoff'
        SpecificationType: 'N,Fc'
              Description: {2x1 cell}
      NormalizedFrequency: false
                       Fs: 48000
              FilterOrder: 10
                  Fcutoff: 12000


>> designmethods(d)


Design Methods for class fdesign.lowpass:


butter

hd=butter(d)

hd =

        FilterStructure: 'Direct-Form II, Second-Order Sections'
             Arithmetic: 'double'
```

```
                sosMatrix: [5x6 double]
              ScaleValues: [6x1 double]
         PersistentMemory: false
                   States: [2x5 double]
      NumSamplesProcessed: 0
```

Now change the `SpecificationType` string for `d` to `'n,fst,ast'` and determine the design methods that apply.

```
set(d,'specificationtype','n,fst,ast');
d

d =

         ResponseType: 'Lowpass with stopband-edge specifications'
    SpecificationType: 'N,Fst,Ast'
          Description: {3x1 cell}
  NormalizedFrequency: false
                   Fs: 48000
          FilterOrder: 10
                Fstop: 12000
                Astop: 60


designmethods(d)


Design Methods for class fdesign.lowpass:


cheby2

hd=cheby2(d)

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [5x6 double]
             ScaleValues: [6x1 double]
        PersistentMemory: false
                  States: [2x5 double]
     NumSamplesProcessed: 0
```

See Also

`butter`, `cheby1`, `cheby2`, `ellip`, `equiripple`, `kaiserwin`

Purpose

Construct a discrete-time, coupled-allpass, lattice filter object—either floating-point or fixed-point

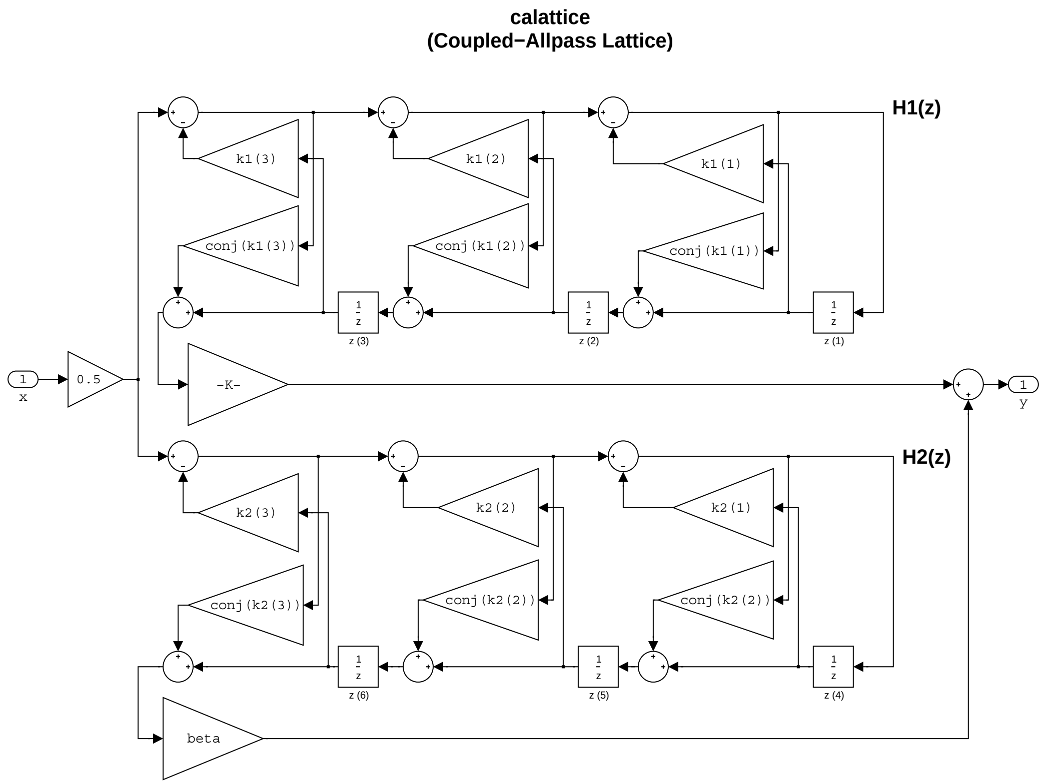
Syntax

```
hd = dfilt.calattice(k1,k2,beta)
hd = dfilt.calattice
```

Description

`hd = dfilt.calattice(k1,k2,beta)` returns a discrete-time, coupled-allpass, lattice filter object `hd`, which is two allpass, lattice filter structures coupled together. The lattice coefficients for each structure are vectors `k1` and `k2`. Input argument `beta` is shown in the diagram below.

`hd = dfilt.calattice` returns a default, discrete-time coupled-allpass, lattice filter object, `hd`. The default values are `k1 = k2 = []`, which is the default value for `dfilt.latticeallpass`, and `beta = 1`. This filter passes the input through to the output unchanged.



Example

Specify a third-order lattice coupled-allpass filter structure for a `dfilt` filter, `hd` with the following code.

```
k1 = [0.9511 + 0.3088i; 0.7511 + 0.1158i]
k2 = 0.7502 - 0.1218i
beta = 0.1385 + 0.9904i
hd = dfilt.calattice(k1,k2,beta)

k1 =

   0.9511 + 0.3088i
   0.7511 + 0.1158i
```

```
k2 =

   0.7502 - 0.1218i


beta =

   0.1385 + 0.9904i


hd =

         FilterStructure: 'Coupled-Allpass Lattice'
                Allpass1: [2x1 double]
                Allpass2: 0.7502- 0.1218i
                    Beta: 0.1385+ 0.9904i
        PersistentMemory: false
                  States: [3x1 double]
     NumSamplesProcessed: 0
```

Notice that the Allpass1 and Allpass2 properties store vectors of coefficients.

```
hd.Allpass1

ans =

   0.9511 + 0.3088i
   0.7511 + 0.1158i
```

See Also

dfilt.calatticepc

dfilt, dfilt.latticeallpass, dfilt.latticear, dfilt.latticearma, dfilt.latticemamax, dfilt.latticemamin in your Signal Processing Toolbox documentation

dfilt.calatticepc

Purpose

Construct a discrete-time, coupled-allpass, power-complementary lattice filter object—either floating-point or fixed-point

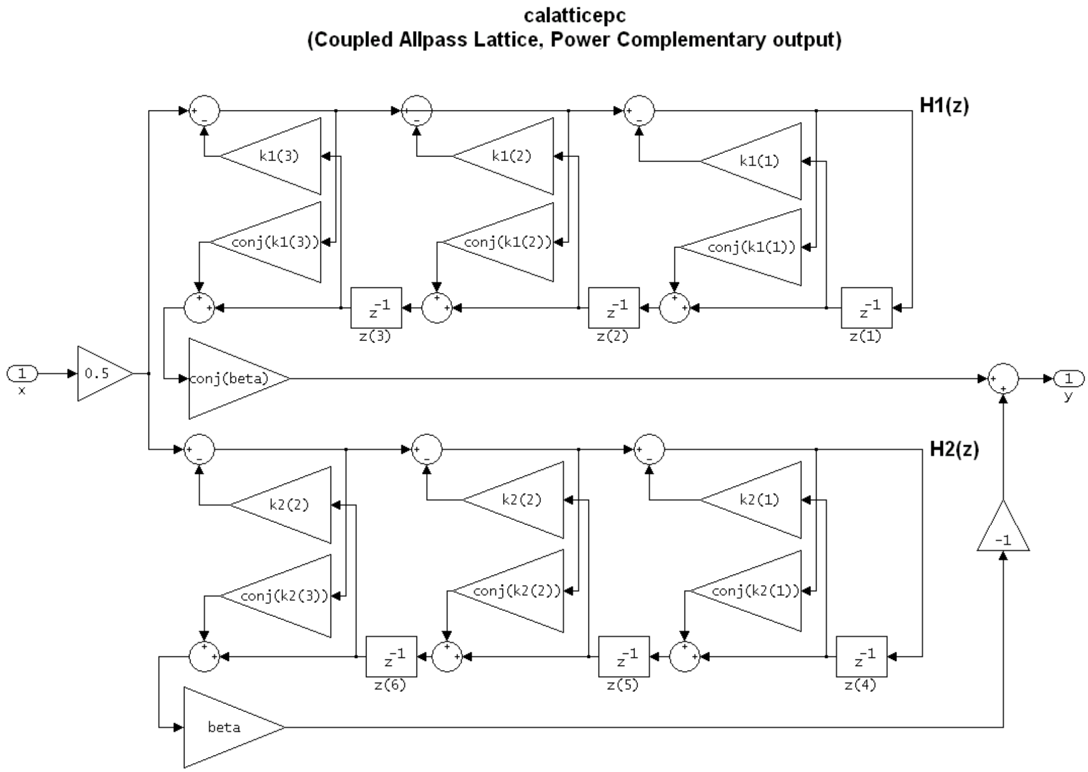
Syntax

```
hd = dfilt.calatticepc(k1,k2,beta)
hd = dfilt.calatticepc
```

Description

`hd = dfilt.calatticepc(k1,k2)` returns a discrete-time, coupled-allpass, lattice filter object `hd`, with power-complementary output. This object is two allpass lattice filter structures coupled together to produce complementary output. The lattice coefficients for each structure are vectors, `k1` and `k2`, respectively. `beta` is shown in the diagram below

`hd = dfilt.calatticepc` returns a default, discrete-time, coupled-allpass, lattice filter object `hd`, with power-complementary output. The default values are `k1=k2=[]`, which is the default value for the `dfilt.latticeallpass`. The default for `beta=1`. This filter passes the input through to the output unchanged.



Example

Specify a third-order lattice coupled-allpass power complementary filter structure for a filter `hd` with the following code. You see from the returned properties that Allpass1 and Allpass2 contain vectors of coefficients for the constituent filters.

```
k1 = [0.9511 + 0.3088i; 0.7511 + 0.1158i]
k2 = 0.7502 - 0.1218i
beta = 0.1385 + 0.9904i
hd = dfilt.calatticepc(k1,k2,beta)
k1 =

   0.9511 + 0.3088i
   0.7511 + 0.1158i
```

```
k2 =

   0.7502 - 0.1218i


beta =

   0.1385 + 0.9904i


hd =

         FilterStructure: 'Coupled-Allpass Lattice, Power
Complementary Output'
                Allpass1: [2x1 double]
                Allpass2: 0.7502- 0.1218i
                    Beta: 0.1385+ 0.9904i
        PersistentMemory: false
                  States: [3x1 double]
     NumSamplesProcessed: 0
```

To see the coefficients for Allpass1, check the property values.

```
get(hd,'Allpass1')

ans =

   0.9511 + 0.3088i
   0.7511 + 0.1158i
```

See Also

dfilt.calattice

dfilt, dfilt.latticeallpass, dfilt.latticear, dfilt.latticearma, dfilt.latticemamax, dfilt.latticemamin in your Signal Processing Toolbox documentation

Purpose

Construct a cascade of discrete-time filter objects

Syntax

Refer to `dfilt.cascade` in the Signal Processing Toolbox for more information.

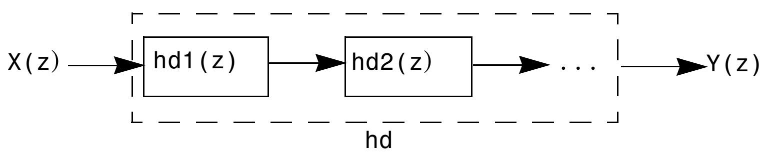
Description

`hd = dfilt.cascade(filterobject1,filterobject2,...)` returns a discrete-time filter object `hd` of type `cascade`, which is a serial interconnection of two or more filter objects `filterobject1`, `filterobject2`, and so on. `dfilt.cascade` accepts any combination of `dfilt` objects (discrete time filters), objects to cascade.

You can use the standard notation to cascade one or more filters:

```
cascade(hd1,hd2,...)
```

where `hd1`, `hd2`, and so on can be mixed types, such as `dfilt` objects and `mfilt` objects.



`hd1`, `hd2`, and so on can be fixed-point filters. All filters in the cascade must be the same arithmetic format—double, single, or fixed. `hd`, the filter object returned, inherits the format of the cascaded filters.

Examples

Cascade a lowpass filter and a highpass filter to produce a bandpass filter.

```
[b1,a1]=butter(8,0.6);          % Lowpass
[b2,a2]=butter(8,0.4,'high');   % Highpass
h1=dfilt.df2t(b1,a1);
h2=dfilt.df2t(b2,a2);
hcas=dfilt.cascade(h1,h2)       % Bandpass with passband 0.4-0.6

hcas =
         Filterstructure: Cascade
              Section(1): Direct Form II Transposed
              Section(2): Direct Form II Transposed
        PersistentMemory: false
```

```
      NumSamplesProcessed: 0
```

To view details of one section, use

```
hcas.section(1)
 ans =
          FilterStructure: 'Direct Form II Transposed'
               Arithmetic: 'double'
                Numerator: [1x9 double]
              Denominator: [1x9 double]
         PersistentMemory: false
                   States: [8x1 double]
      NumSamplesProcessed: 0
```

See Also

`dfilt`, `dfilt.parallel`, `dfilt.scalar`

Purpose

Construct a discrete-time, direct-form I filter object for fixed-point and single-precision floating-point filtering

Syntax

Refer to `dfilt.df1` in the Signal Processing Toolbox.

Description

`hd = dfilt.df1` returns a default discrete-time, direct-form I filter object that uses double-precision arithmetic. By default, the numerator and denominator coefficients b and a are set to 1. With these coefficients the filter passes the input to the output without changes.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

  ```
  set(hd,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

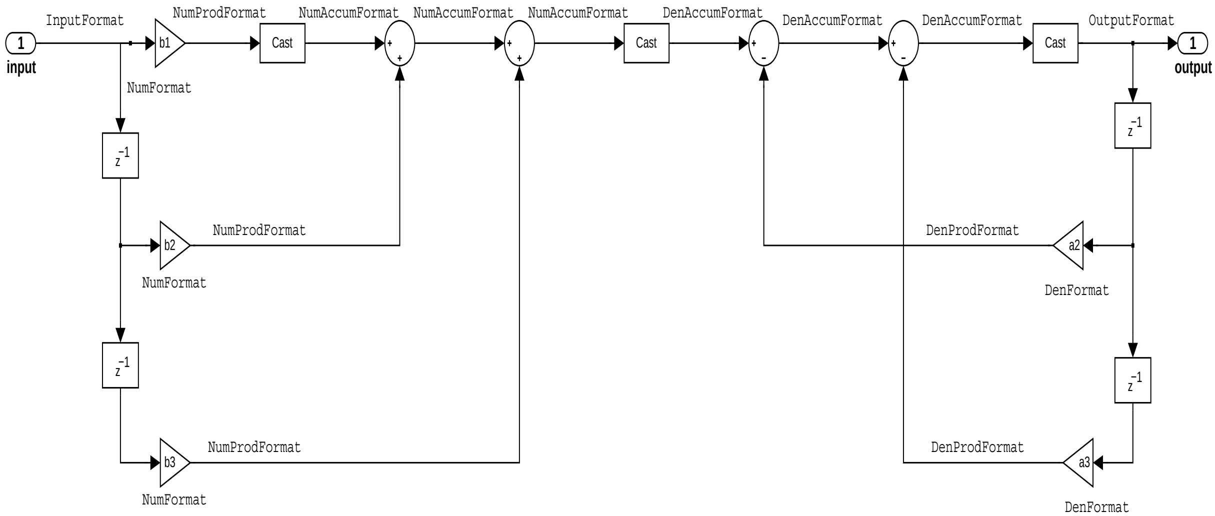
  ```
  set(hd,'arithmetic','fixed');
  ```

For more information about the property `Arithmetic`, refer to “Arithmetic” on page 8-22.

Note `a(1)`, the leading denominator coefficient, cannot be 0. To allow you to change the arithmetic setting to `fixed` or `single`, `a(1)` must be equal to 1.

Fixed-Point Filter Structure

The figure below shows the signal flow for the direct-form I filter implemented by `dfilt.df1`. To help you see how the filter processes the coefficients, input, output, and states of the filter, as well as numerical operations, the figure includes the locations of the arithmetic and data type format elements within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to

the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
DenAccumFormat	`AccumWordLength`	`DenAccumFracLength`	`AccumMode`, `CastBeforeSum`
DenFormat	`CoeffWordLength`	`DenFracLength`	`CoeffAutoScale`, `Signed`, `Denominator`
DenProdFormat	`CoeffWordLength`	`DenProdFracLength`	`ProductMode`, `ProductWordLength`
InputFormat	`InputWordLength`	`InputFracLength`	
NumAccumFormat	`AccumWordLength`	`NumAccumFracLength`	`AccumMode`, `CastBeforeSum`
NumFormat	`CoeffWordLength`	`NumFracLength`	`CoeffAutoScale`, `Signed`, `Numerator`
NumProdFormat	`CoeffWordLength`	`NumProdFracLength`	`ProductWordLength`, `ProductMode`
OutputFormat	`OutputWordLength`	`OutputFracLength`	`OutputMode`

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label DenProdFormat, which always follows a denominator coefficient multiplication element in the signal flow. The label indicates that denominator coefficients leave the multiplication element with the word length and fraction length associated with product operations that include denominator coefficients. From reviewing the table, you see that the DenProdFormat refers to the properties `ProdWordLength`, `ProductMode` and `DenProdFracLength` that fully define the denominator format after multiply (or product) operations.

Properties

In this table you see the properties associated with df1 implementations of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to "Fixed-Point Filter Properties" on page 8-3.

Property Name	Brief Description
`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.
`Arithmetic`	Defines the arithmetic the filter uses. Gives you the options `double`, `single`, and `fixed`. In short, this property defines the operating mode for your filter.
`CastBeforeSum`	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.

Property Name	Brief Description
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength and DenFracLength properties to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
DenAccumFracLength	Specifies the fraction length the filter algorithm uses to interpret the results of product operations involving denominator coefficients. You can change the value for this property when you set AccumMode to SpecifyPrecision.
DenFracLength	Set the fraction length the filter uses to interpret denominator coefficients. DenFracLength is always available, but it is read-only until you set CoeffAutoScale to false.
Denominator	Stores the denominator coefficients for the IIR filter.
DenProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving denominator coefficients. You can change this property value when you set ProductMode to SpecifyPrecision.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.

Property Name	Brief Description
InputWordLength	Specifies the word length applied to interpret input data.
NumAccumFracLength	Specifies how the filter algorithm interprets the results of addition operations involving numerator coefficients. You can change the value of this property after you set AccumMode to SpecifyPrecision.
Numerator	Holds the numerator coefficient values for the filter.
NumFracLength	Sets the fraction length used to interpret the value of numerator coefficients.
NumProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving numerator coefficients. Available to be changed when you set ProductMode to SpecifyPrecision.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.
OutputWordLength	Determines the word length used for the output data.

Property Name	Brief Description
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
`RoundMode`	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • `convergent`—Round up to the next allowable quantized value. • `ceil`—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to `1`. • `fix`—Round negative numbers up and positive numbers down to the next allowable quantized value. • `floor`—Round down to the next allowable quantized value. • `round`—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use `fi` objects, with the associated properties from those objects. For details, refer to `filtstates` in your Signal Processing Toolbox documentation or in the Help system.

Examples

Specify a second-order direct-form I structure for a `dfilt` object, `hd`, with the following code:

```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hd = dfilt.df1(b,a)
hd =

         FilterStructure: 'Direct-Form I'
              Arithmetic: 'double'
               Numerator: [0.3000 0.6000 0.3000]
             Denominator: [1 0 0.2000]
        PersistentMemory: false
                  States: Numerator:  [2x1 double]
                          Denominator:[2x1 double]
     NumSamplesProcessed: 0
```

Now convert `hd` to a fixed-point filter:

```
set(hd,'arithmetic','fixed')
hd

hd =

         FilterStructure: 'Direct-Form I'
```

```
             Arithmetic: 'fixed'
              Numerator: [0.3000 0.6000 0.3000]
            Denominator: [1 0 0.2000]
       PersistentMemory: false
                 States: Numerator:  [2x1 fi]
                         Denominator:[2x1 fi]
    NumSamplesProcessed: 0


        CoeffWordLength: 16
         CoeffAutoScale: true
                 Signed: true

        InputWordLength: 16
        InputFracLength: 15

       OutputWordLength: 16
       OutputFracLength: 15

            ProductMode: 'FullPrecision'

              AccumMode: 'KeepMSB'
        AccumWordLength: 40
          CastBeforeSum: true

              RoundMode: 'convergent'
           OverflowMode: 'wrap'
```

See Also

dfilt, dfilt.df1t, dfilt.df2, dfilt.df2t

Purpose

Construct a fixed-point or single-precision floating-point discrete-time, direct-form I filter object that uses second-order sections

Syntax

Refer to `dfilt.df1sos` in the Signal Processing Toolbox.

Description

`hd = dfilt.df1sos(s)` returns a discrete-time, second-order section, direct-form I filter object `hd`, with coefficients given in the `s` matrix.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

- To change to fixed-point filtering, enter

```
set(hd,'arithmetic','fixed');
```

For more information about the property `Arithmetic`, refer to “Arithmetic” on page 8-22.

`hd = dfilt.df1sos(b1,a1,b2,a2,...)` returns a discrete-time, second-order section, direct-form I filter object `hd`, with coefficients for the first section given in the `b1` and `a1` vectors, for the second section given in the `b2` and `a2` vectors, and so on.

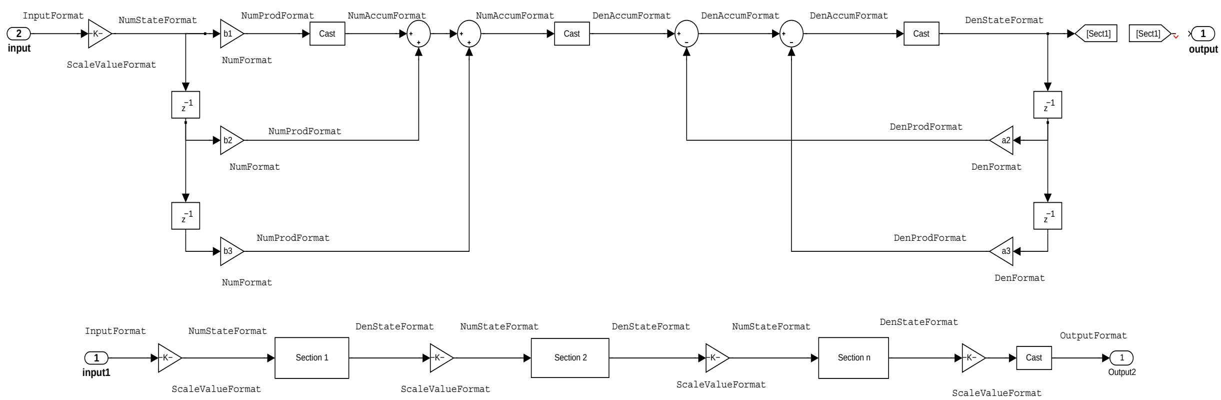
`hd = dfilt.df1sos(...,g)` includes a gain vector `g`. The elements of `g` are the gains for each section. The maximum length of `g` is the number of sections plus one. When you do not specify `g`, all gains default to one.

`hd = dfilt.df1sos` returns a default, discrete-time, second-order section, direct-form I filter object, `hd`. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0. To allow you to change the arithmetic setting to `fixed` or `single`, `a(1)` must be equal to 1.

Fixed-Point Filter Structure

The figure below shows the signal flow for the direct-form I filter implemented in second-order sections by `dfilt.df1sos`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to

the word and fraction lengths (CoeffWordLength, NumFracLength) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
DenAccumFormat	AccumWordLength	DenAccumFracLength	AccumMode, CastBeforeSum
DenFormat	CoeffWordLength	DenFracLength	CoeffAutoScale, Signed, Denominator
DenProdFormat	CoeffWordLength	DenProdFracLength	ProductMode, ProductWordLength
DenStateFormat	DenStateWordLength	DenStateFracLength	CastBeforeSum, States
InputFormat	InputWordLength	InputFracLength	
NumAccumFormat	AccumWordLength	NumAccumFracLength	AccumMode, CastBeforeSum
NumFormat	CoeffWordLength	NumFracLength	CoeffAutoScale, Signed, Numerator
NumProdFormat	CoeffWordLength	NumProdFracLength	ProductWordLength, ProductMode
NumStateFormat	NumStateWordLength	NumStateFracLength	States
OutputFormat	OutputWordLength	OutputFracLength	OutputMode
ScaleValueFormat	CoeffWordLength	ScaleValueFracLength	CoeffAutoScale, ScaleValues

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label DenProdFormat, which always follows a denominator coefficient multiplication element in the signal flow. The label indicates that denominator coefficients leave the multiplication element with the word length and fraction length associated with product operations that include denominator coefficients. From reviewing the table, you see that the

DenProdFormat refers to the properties `ProdWordLength`, `ProductMode` and `DenProdFracLength` that fully define the denominator format after multiply (or product) operations.

Properties

In this table you see the properties associated with SOS implementation of direct-form I `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to "Fixed-Point Filter Properties" on page 8-3.

Property Name	Brief Description
`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.

Property Name	Brief Description
Arithmetic	Defines the arithmetic the filter uses. Gives you the options double, single, and fixed. In short, this property defines the operating mode for your filter.
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength and DenFracLength properties to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
DenAccumFracLength	Specifies the fraction length used to interpret data in the accumulator used to hold the results of sum operations. You can change the value for this property when you set AccumMode to SpecifyPrecision.
DenFracLength	Set the fraction length the filter uses to interpret denominator coefficients. DenFracLength is always available, but it is read-only until you set CoeffAutoScale to false.
DenProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving denominator coefficients. You can change this property value when you set ProductMode to SpecifyPrecision.

Property Name	Brief Description
DenStateFracLength	Specifies the fraction length used to interpret the states associated with denominator coefficients in the filter.
DenStateWordLength	Specifies the word length used to represent the states associated with denominator coefficients in the filter.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
NumAccumFracLength	Specifies how the filter algorithm interprets the results of addition operations involving numerator coefficients. You can change the value of this property after you set `AccumMode` to `SpecifyPrecision`.
NumFracLength	Sets the fraction length used to interpret the value of numerator coefficients.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When `PersistentMemory` is `false`, this property reports the number of samples filtered for each input data set. Setting `PersistentMemory` to `true` causes this property to report the total number of samples processed for all data sets, not each one.

Property Name	Brief Description
NumStateFracLength	Specifies the fraction length used to interpret the states associated with numerator coefficient operations in the filter.
NumWordFracLength	Specifies the word length used to interpret the states associated with numerator coefficient operations in the filter.
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: ▪ AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. ▪ BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. ▪ SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length applied for the output data.

Property Name	Brief Description
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
ScaleValueFracLength	Scale values work with SOS filters. Setting this property controls how your filter interprets the scale values by setting the fraction length. Only available when you disable AutoScaleMode by setting it to false.
ScaleValues	Scaling for the filter objects in SOS filters.

Property Name	Brief Description
`Signed`	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
`SosMatrix`	Holds the filter coefficients as property values. Displays the matrix in the format [sections x coefficients/section datatype]. A `[15x6 double]` SOS matrix represents a filter with 6 coefficients per section and 15 sections, using data type `double` to represent the coefficients.
`States`	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use `fi` objects, with the associated properties from those objects. For details, refer to `filtstates` in your Signal Processing Toolbox documentation or in the Help system.
`StateWordLength`	Sets the word length used to represent the filter states.

Examples

Specify a fixed-point, second-order section, direct-form I `dfilt` object with the following code:

```
b=[0.3 0.6 0.3];
a=[1 0 0.2];
hd=dfilt.df1sos(b,a)

hd =

         FilterStructure: 'Direct-Form I, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [0.3000 0.6000 0.3000 1 0 0.2000]
             ScaleValues: [2x1 double]
```

```
         PersistentMemory: false
                   States: Numerator:  [2x1 double]
                           Denominator:[2x1 double]
      NumSamplesProcessed: 0

hd.arithmetic='fixed'

hd =

          FilterStructure: 'Direct-Form I, Second-Order Sections'
              ScaleValues: [2x1 double]
               Arithmetic: 'fixed'
                sosMatrix: [0.3000 0.6000 0.3000 1 0 0.2000]
         PersistentMemory: false
                   States: Numerator:  [2x1 fi]
                           Denominator:[2x1 fi]
      NumSamplesProcessed: 0


          CoeffWordLength: 16
           CoeffAutoScale: true
                   Signed: true

          InputWordLength: 16
          InputFracLength: 15

         OutputWordLength: 16
               OutputMode: 'AvoidOverflow'

       NumStateWordLength: 16
       NumStateFracLength: 15

       DenStateWordLength: 16
       DenStateFracLength: 15

              ProductMode: 'FullPrecision'

                AccumMode: 'KeepMSB'
          AccumWordLength: 40
            CastBeforeSum: true
```

```
   RoundMode: 'convergent'
OverflowMode: 'wrap'
```

See Also dfilt, dfilt.df2tsos

Purpose

Construct a discrete-time, direct-form I transposed filter object that uses fixed-point or single-precision floating-point arithmetic

Syntax

Refer to `dfilt.df1t` in the Signal Processing Toolbox.

Description

`hd = dfilt.df1t(b,a)` returns a discrete-time, direct-form I transposed filter object `hd`, with numerator coefficients `b` and denominator coefficients `a`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

- To change to fixed-point filtering, enter

```
set(hd,'arithmetic','fixed');
```

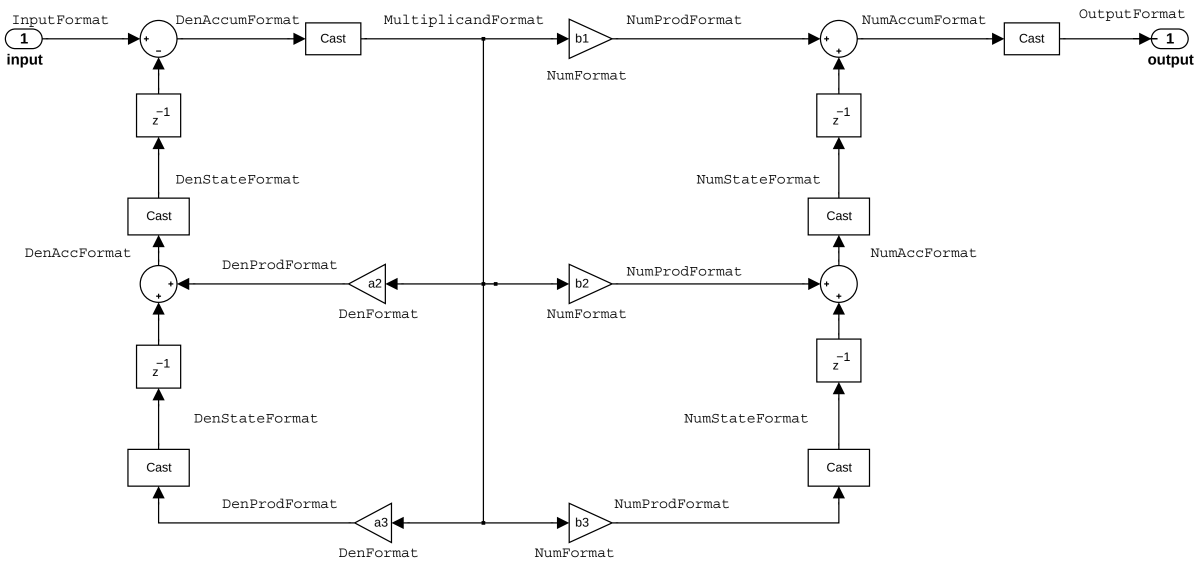
For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.df1t` returns a default, discrete-time, direct-form I transposed filter object `hd`, with `b`=1 and `a`=1. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0. To allow you to change the arithmetic setting to `fixed` or `single`, `a(1)` must be equal to 1.

Fixed-Point Filter Structure

The figure below shows the signal flow for the transposed direct-form I filter implemented by `dfilt.df1t`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to

the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
DenAccumFormat	`AccumWordLength`	`DenAccumFracLength`	`AccumMode`, `CastBeforeSum`
DenFormat	`CoeffWordLength`	`DenFracLength`	`CoeffAutoScale`, `Signed`, `Denominator`
DenProdFormat	`CoeffWordLength`	`DenProdFracLength`	`ProductMode`, `ProductWordLength`
DenStateFormat	`DenStateWordLength`	`DenStateFracLength`	`CastBeforeSum`, `States`
InputFormat	`InputWordLength`	`InputFracLength`	
MultiplicandFormat	`MultiplicandWordLength`	`MultiplicandFracLength`	`CastBeforeSum`
NumAccumFormat	`AccumWordLength`	`NumAccumFracLength`	`AccumMode`, `CastBeforeSum`
NumFormat	`CoeffWordLength`	`NumFracLength`	`CoeffAutoScale`, `Signed`, `Numerator`
NumProdFormat	`CoeffWordLength`	`NumProdFracLength`	`ProductWordLength`, `ProductMode`
NumStateFormat	`NumStateWordLength`	`NumStateFracLength`	`States`
OutputFormat	`OutputWordLength`	`OutputFracLength`	`OutputMode`

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label DenProdFormat, which always follows a denominator coefficient multiplication element in the signal flow. The label indicates that denominator coefficients leave the multiplication element with the word length and fraction length associated with product operations that include denominator coefficients. From reviewing the table, you see that the DenProdFormat refers to the properties `ProdWordLength`, `ProductMode` and `DenProdFracLength` that fully define the denominator format after multiply (or product) operations.

Properties

In this table you see the properties associated with df1t implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to "Fixed-Point Filter Properties" on page 8-3.

Property Name	Brief Description
`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.
`Arithmetic`	Defines the arithmetic the filter uses. Gives you the options `double`, `single`, and `fixed`. In short, this property defines the operating mode for your filter.

Property Name	Brief Description
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength and DenFracLength properties to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
DenAccumFracLength	Specifies the fraction length used to interpret data in the accumulator used to hold the results of sum operations. You can change the value for this property when you set AccumMode to SpecifyPrecision.
DenFracLength	Set the fraction length the filter uses to interpret denominator coefficients. DenFracLength is always available, but it is read-only until you set CoeffAutoScale to false.
Denominator	Holds the denominator coefficients for the filter.
DenProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving denominator coefficients. You can change this property value when you set ProductMode to SpecifyPrecision.

Property Name	Brief Description
DenStateFracLength	Specifies the fraction length used to interpret the states associated with denominator coefficients in the filter.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
MultiplicandFracLength	Sets the fraction length for values (multiplicands) used in multiply operations in the filter.
MultiplicandWordLength	Sets the word length applied to the values input to a multiply operation (the multiplicands).
NumAccumFracLength	Specifies how the filter algorithm interprets the results of addition operations involving numerator coefficients. You can change the value of this property after you set AccumMode to SpecifyPrecision.
Numerator	Holds the numerator coefficient values for the filter.
NumFracLength	Sets the fraction length used to interpret the value of numerator coefficients.

Property Name	Brief Description
NumProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving numerator coefficients. Available to be changed when you set ProductMode to SpecifyPrecision.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.
NumStateFracLength	For IIR filters, this defines the binary point location applied to the numerator states of the filter. Specifies the fraction length used to interpret the states associated with numerator coefficient operations in the filter.
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.

Property Name	Brief Description
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: ▪ AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. ▪ BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. ▪ SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.

Property Name	Brief Description
StateAutoScale	Setting autoscaling for filter states to true reduces the possibility of overflows occurring during fixed-point operations. Set to false, StateAutoScale lets the filter select the fraction length to limit the overflow potential.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions.
StateWordLength	Sets the word length used to represent the filter states.

Examples

Specify a second-order direct-form I transposed filter structure for a dfilt object, hd, with the following code:

```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hd = dfilt.df1t(b,a)

hd =

         FilterStructure: 'Direct-Form I Transposed'
              Arithmetic: 'double'
               Numerator: [0.3000 0.6000 0.3000]
             Denominator: [1 0 0.2000]
        PersistentMemory: false
                  States: Numerator:  [2x1 double]
                          Denominator:[2x1 double]
     NumSamplesProcessed: 0
```

Now convert the filter to single-precision filtering arithmetic.

```
set(hd,'arithmetic','single')
hd
hd =
```

```
        FilterStructure: 'Direct-Form I Transposed'
             Arithmetic: 'fixed'
              Numerator: [0.3000 0.6000 0.3000]
            Denominator: [1 0 0.2000]
       PersistentMemory: false
                 States: Numerator:  [2x1 fi]
                         Denominator:[2x1 fi]
    NumSamplesProcessed: 0


        CoeffWordLength: 16
         CoeffAutoScale: true
                 Signed: true

        InputWordLength: 16
        InputFracLength: 15

       OutputWordLength: 16
             OutputMode: 'AvoidOverflow'

 MultiplicandWordLength: 16
 MultiplicandFracLength: 15

        StateWordLength: 16
         StateAutoScale: true

            ProductMode: 'FullPrecision'

              AccumMode: 'KeepMSB'
        AccumWordLength: 40
          CastBeforeSum: true

              RoundMode: 'convergent'
           OverflowMode: 'wrap'
```

See Also

`dfilt`, `dfilt.df1`, `dfilt.df2`, `dfilt.df2t`

Purpose Construct a discrete-time, second-order section, direct-form I transposed filter object that uses fixed-point or single-precision floating-point arithmetic

Syntax Refer to `dfilt.df1tsos` in the Signal Processing Toolbox.

Description `hd = dfilt.df1tsos(s)` returns a discrete-time, second-order section, direct-form I, transposed filter object `hd`, with coefficients given in the `s` matrix.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

  ```
  set(hd,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hd,'arithmetic','fixed');
  ```

For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.df1tsos(b1,a1,b2,a2,...)` returns a discrete-time, second-order section, direct-form I, transposed filter object `hd`, with coefficients for the first section given in the `b1` and `a1` vectors, for the second section given in the `b2` and `a2` vectors, etc.

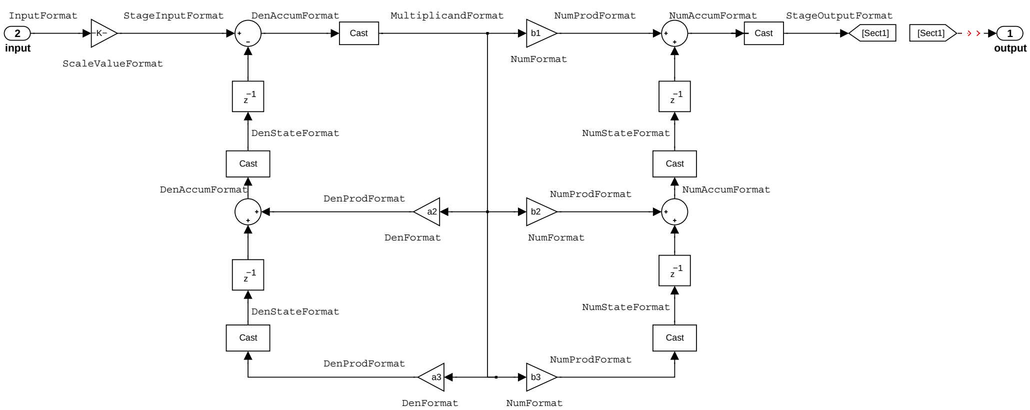
`hd = dfilt.df1tsos(...,g)` includes a gain vector `g`. The elements of `g` are the gains for each section. The maximum length of `g` is the number of sections plus one. If `g` is not specified, all gains default to one.

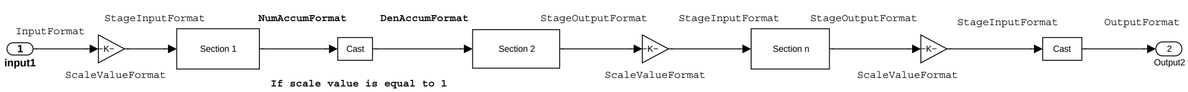
`hd = dfilt.df1tsos` returns a default, discrete-time, second-order section, direct-form I, transposed filter object, `hd`. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0. To allow you to change the arithmetic setting to `fixed` or `single`, `a(1)` must be equal to 1.

Fixed-Point Filter Structure

The figure below shows the signal flow for the direct-form I transposed filter implemented using second-order sections by `dfilt.df1tsos`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.





Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word “format.” In this use, “format” means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
DenAccumFormat	`AccumWordLength`	`DenAccumFracLength`	`AccumMode, CastBeforeSum`
DenFormat	`CoeffWordLength`	`DenFracLength`	`CoeffAutoScale, Signed, Denominator`
DenProdFormat	`CoeffWordLength`	`DenProdFracLength`	`ProductMode, ProductWordLength`
DenStateFormat	`DenStateWordLength`	`DenStateFracLength`	`CastBeforeSum, States`
InputFormat	`InputWordLength`	`InputFracLength`	
MultiplicandFormat	`MultiplicandWordLength`	`MultiplicandFracLength`	`CastBeforeSum`
NumAccumFormat	`AccumWordLength`	`NumAccumFracLength`	`AccumMode, CastBeforeSum`
NumFormat	`CoeffWordLength`	`NumFracLength`	`CoeffAutoScale, Signed, Numerator`
NumProdFormat	`CoeffWordLength`	`NumProdFracLength`	`ProductWordLength, ProductMode`
NumStateFormat	`NumStateWordLength`	`NumStateFracLength`	`States`
OutputFormat	`OutputWordLength`	`OutputFracLength`	`OutputMode`
ScaleValueFormat	CoeffWordLength	ScaleValueFracLength	CoeffAutoScale, ScaleValues
StageInputFormat	StageInputWordLength	StageInputFracLength	StageInputAutoScale
StageOutputFormat	StageOutputWordLength	StageOutputFracLength	StageOutputAutoScale

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label DenProdFormat, which always follows a denominator coefficient multiplication element in the signal flow. The label indicates that denominator coefficients leave the multiplication element with the word length and fraction length associated with product operations that include denominator coefficients. From reviewing the table, you see that the DenProdFormat refers to the properties `ProdWordLength`, `ProductMode` and `DenProdFracLength` that fully define the denominator format after multiply (or product) operations.

Properties

In this table you see the properties associated with SOS implementation of transposed direct-form I `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to "Fixed-Point Filter Properties" on page 8-3.

Property Name	Brief Description
`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.

Property Name	Brief Description
Arithmetic	Defines the arithmetic the filter uses. Gives you the options double, single, and fixed. In short, this property defines the operating mode for your filter.
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength and DenFracLength properties to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
DenAccumFracLength	Specifies the fraction length used to interpret data in the accumulator used to hold the results of sum operations. You can change the value for this property when you set AccumMode to SpecifyPrecision.
DenFracLength	Set the fraction length the filter uses to interpret denominator coefficients. DenFracLength is always available, but it is read-only until you set CoeffAutoScale to false.
DenProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving denominator coefficients. You can change this property value when you set ProductMode to SpecifyPrecision.

Property Name	Brief Description
DenStateFracLength	Specifies the fraction length used to interpret the states associated with denominator coefficients in the filter.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
MultiplicandFracLength	Sets the fraction length for values (multiplicands) used in multiply operations in the filter.
MultiplicandWordLength	Sets the word length applied to the values input to a multiply operation (the multiplicands)
NumAccumFracLength	Specifies how the filter algorithm interprets the results of addition operations involving numerator coefficients. You can change the value of this property after you set AccumMode to SpecifyPrecision.
Numerator	Holds the numerator coefficient values for the filter.
NumProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving numerator coefficients. Available to be changed when you set ProductMode to SpecifyPrecision.

Property Name	Brief Description
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.
NumStateFracLength	For IIR filters, this defines the binary point location applied to the numerator states of the filter. Specifies the fraction length used to interpret the states associated with numerator coefficient operations in the filter.
NumStateWordLength	For IIR filters, this defines the word length applied to the numerator states of the filter. Specifies the word length used to interpret the states associated with numerator coefficient operations in the filter.
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.

Property Name	Brief Description
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: ▪ AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. ▪ BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. ▪ SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
ScaleValueFracLength	Scale values work with SOS filters. Setting this property controls how your filter interprets the scale values by setting the fraction length. Only available when you disable AutoScaleMode by setting it to false.

Property Name	Brief Description
ScaleValues	Scaling for the filter objects in SOS filters.
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
SosMatrix	Holds the filter coefficients as property values. Displays the matrix in the format [sections x coefficients/section datatype]. A [15x6 double] SOS matrix represents a filter with 6 coefficients per section and 15 sections, using data type double to represent the coefficients.
StageInputAutoScale	Tells the filter whether to set the stage input data format to minimize the occurrence of overflow conditions.
StageInputFracLength	Lets you set the fraction length for stage inputs in SOS filters, if you set StageInputAutoScale to false.
StageInputWordLength	Lets you set the word length for stage inputs in SOS filters, if you set StageInputAutoScale to false.
StageOutputAutoScale	Tells the filter whether to set the stage output data format to minimize the occurrence of overflow conditions.
StageOutputFracLength	Lets you set the fraction length for stage outputs in SOS filters, if you set StageOutputAutoScale to false.
StageOutputWordLength	Lets you set the word length for stage outputs in SOS filters, if you set StageOutputAutoScale to false.

Property Name	Brief Description
StateAutoScale	Setting autoscaling for filter states to true reduces the possibility of overflows occurring during fixed-point operations. Set to false, StateAutoScale lets the filter select the fraction length to limit the overflow potential.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions.
StateWordLength	Sets the word length used to represent the filter states.

Examples

With the following code, this example specifies a second-order section, direct-form I transposed dfilt object for a filter. Then we convert the filter to fixed-point operation.

```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hd = dfilt.df1t(b,a)

hd =

         FilterStructure: 'Direct-Form I Transposed'
              Arithmetic: 'double'
               Numerator: [0.3000 0.6000 0.3000]
             Denominator: [1 0 0.2000]
        PersistentMemory: false
                  States: Numerator:  [2x1 double]
                          Denominator:[2x1 double]
     NumSamplesProcessed: 0


set(hd,'arithmetic','fixed')
hd
```

```
hd =

          FilterStructure: 'Direct-Form I Transposed'
               Arithmetic: 'fixed'
                Numerator: [0.3000 0.6000 0.3000]
              Denominator: [1 0 0.2000]
         PersistentMemory: false
                   States: Numerator:  [2x1 fi]
                           Denominator:[2x1 fi]
      NumSamplesProcessed: 0


          CoeffWordLength: 16
           CoeffAutoScale: true
                   Signed: true

          InputWordLength: 16
          InputFracLength: 15

         OutputWordLength: 16
               OutputMode: 'AvoidOverflow'


   MultiplicandWordLength: 16
   MultiplicandFracLength: 15

          StateWordLength: 16
           StateAutoScale: true

              ProductMode: 'FullPrecision'

                AccumMode: 'KeepMSB'
          AccumWordLength: 40
            CastBeforeSum: true

                RoundMode: 'convergent'
             OverflowMode: 'wrap'
```

See Also

`dfilt`, `dfilt.df1sos`, `dfilt.df2sos`, `dfilt.df2tsos`

Purpose

Construct a discrete-time, direct-form II filter object with fixed-point or single-precision floating-point operation

Syntax

Refer to `dfilt.df2` in the Signal Processing Toolbox.

Description

`hd = dfilt.df2(b,a)` returns a discrete-time, direct-form II filter object `hd`, with numerator coefficients `b` and denominator coefficients `a`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

- To change to fixed-point filtering, enter

```
set(hd,'arithmetic','fixed');
```

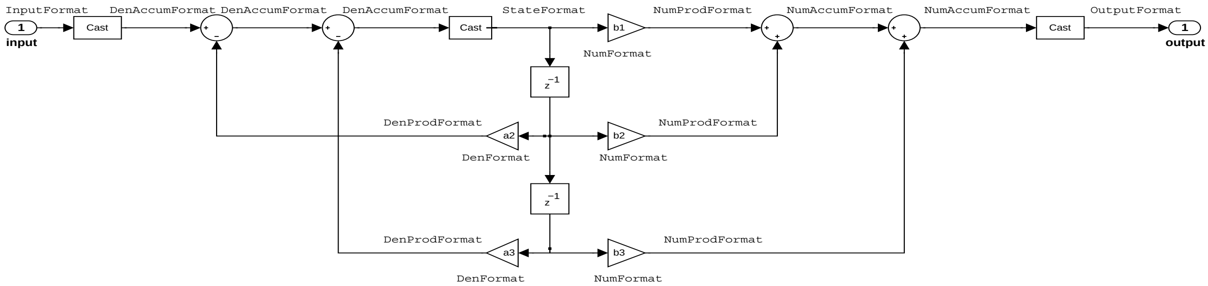
For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.df2` returns a default, discrete-time, direct-form II filter object `hd`, with `b`=1 and `a`=1. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0. To allow you to change the arithmetic setting to `fixed` or `single`, `a(1)` must be equal to 1.

Fixed-Point Filter Structure

The figure below shows the signal flow for the direct-form II filter implemented by `dfilt.df2`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
DenAccumFormat	`AccumWordLength`	`DenAccumFracLength`	`AccumMode`, `CastBeforeSum`
DenFormat	`CoeffWordLength`	`DenFracLength`	`CoeffAutoScale`, `Signed`, `Denominator`

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
DenProdFormat	`CoeffWordLength`	`DenProdFracLength`	`ProductMode`, `ProductWordLength`
InputFormat	`InputWordLength`	`InputFracLength`	
NumAccumFormat	`AccumWordLength`	`NumAccumFracLength`	`AccumMode`, `CastBeforeSum`
NumFormat	`CoeffWordLength`	`NumFracLength`	`CoeffAutoScale`, `Signed`, `Numerator`
NumProdFormat	`CoeffWordLength`	`NumProdFracLength`	`ProductWordLength`, `ProductMode`
OutputFormat	`OutputWordLength`	`OutputFracLength`	`OutputMode`
StateFormat	`StateWordLength`	`StateFracLength`	`States`

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label DenProdFormat, which always follows a denominator coefficient multiplication element in the signal flow. The label indicates that denominator coefficients leave the multiplication element with the word length and fraction length associated with product operations that include denominator coefficients. From reviewing the table, you see that the DenProdFormat refers to the properties `ProdWordLength`, `ProductMode` and `DenProdFracLength` that fully define the denominator format after multiply (or product) operations.

Properties

In this table you see the properties associated with the df2 implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where hd is a filter.

For further information about the properties of this filter or any dfilt object, refer to “Fixed-Point Filter Properties” on page 8-3.

Property Name	Brief Description
AccumMode	Determines how the accumulator outputs stored values. Choose from full precision (FullPrecision), or whether to keep the most significant bits (KeepMSB) or least significant bits (KeepLSB) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set AccumMode to SpecifyPrecision.
AccumWordLength	Sets the word length used to store data in the accumulator/buffer.
Arithmetic	Defines the arithmetic the filter uses. Gives you the options double, single, and fixed. In short, this property defines the operating mode for your filter.
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.

Property Name	Brief Description
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength and DenFracLength properties to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
DenAccumFracLength	Specifies the fraction length used to interpret data in the accumulator used to hold the results of sum operations. You can change the value for this property when you set AccumMode to SpecifyPrecision.
DenFracLength	Set the fraction length the filter uses to interpret denominator coefficients. DenFracLength is always available, but it is read-only until you set CoeffAutoScale to false.
Denominator	Holds the denominator coefficients for IIR filters.
DenProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving denominator coefficients. You can change this property value when you set ProductMode to SpecifyPrecision.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.

Property Name	Brief Description
InputWordLength	Specifies the word length applied to interpret input data.
NumAccumFracLength	Specifies how the filter algorithm interprets the results of addition operations involving numerator coefficients. You can change the value of this property after you set AccumMode to SpecifyPrecision.
Numerator	Holds the numerator coefficient values for the filter.
NumFracLength	Sets the fraction length used to interpret the value of numerator coefficients.
NumProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving numerator coefficients. Available to be changed when you set ProductMode to SpecifyPrecision.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.

Property Name	Brief Description
`OutputMode`	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: ▪ `AvoidOverflow`—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. ▪ `BestPrecision`—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. ▪ `SpecifyPrecision`—lets you set the word and fraction lengths used by the output data from filtering.
`OutputWordLength`	Determines the word length used for the output data.
`OverflowMode`	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either `saturate` (limit the output to the largest positive or negative representable value) or `wrap` (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.

Property Name	Brief Description
StateFracLength	When you set StateAutoScale to false, you enable the StateFracLength property that lets you set the fraction length applied to interpret the filter states.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions.
StateWordLength	Sets the word length used to represent the filter states.

Examples

Specify a second-order direct-form II filter structure for a dfilt object, hd, with the following code:

```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hd = dfilt.df2(b,a)

hd =
             FilterStructure: 'Direct Form II'
                   Numerator: [0.3000 0.6000 0.3000]
                 Denominator: [1 0 0.2000]
    NumberOfSamplesProcessed: 0
                 ResetStates: 'on'
                      States: [2x1 double]
```

To convert the filter to fixed-point arithmetic, change the value of the Arithmetic property

```
set(hd,'arithmetic','fixed')
```

to specify the fixed-point option.

See Also

dfilt, dfilt.df1, dfilt.df1t, dfilt.df2t

Purpose Construct a discrete-time, second-order section, direct-form II filter object that allows fixed-point or single-precision floating-point arithmetic

Syntax Refer to `dfilt.df2sos` in the Signal Processing Toolbox.

Description `hd = dfilt.df2sos(s)` returns a discrete-time, second-order section, direct-form II filter object `hd`, with coefficients given in the `s` matrix.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

- To change to fixed-point filtering, enter

```
set(hd,'arithmetic','fixed');
```

For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.df2sos(b1,a1,b2,a2,...)` returns a discrete-time, second-order section, direct-form II object, `hd`, with coefficients for the first section given in the `b1` and `a1` vectors, for the second section given in the `b2` and `a2` vectors, etc.

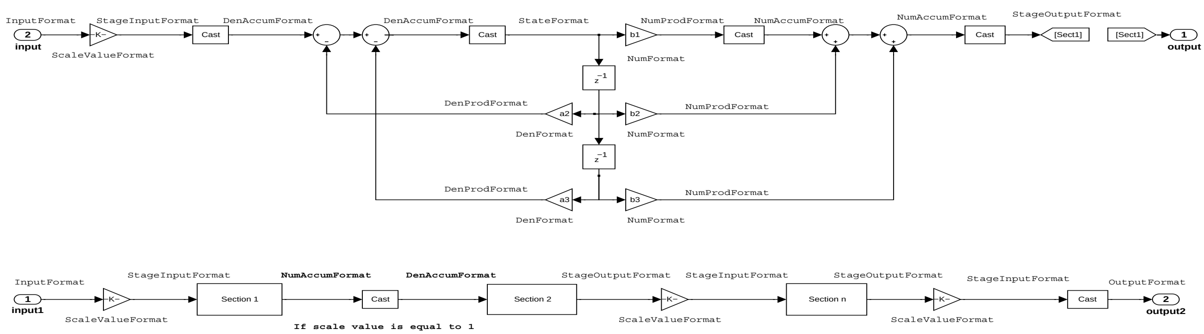
`hd = dfilt.df2sos(...,g)` includes a gain vector `g`. The elements of `g` are the gains for each section. The maximum length of `g` is the number of sections plus one. If `g` is not specified, all gains default to one.

`hd = dfilt.df2sos` returns a default, discrete-time, second-order section, direct-form II filter object, `hd`. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0. To allow you to change the arithmetic setting to `fixed` or `single`, `a(1)` must be equal to 1.

Fixed-Point Filter Structure

The figure below shows the signal flow for the direct-form II filter implemented with second-order sections by `dfilt.df2sos`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to

the word and fraction lengths (CoeffWordLength, NumFracLength) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
DenAccumFormat	AccumWordLength	DenAccumFracLength	AccumMode, CastBeforeSum
DenFormat	CoeffWordLength	DenFracLength	CoeffAutoScale, Signed, sosMatrix
DenProdFormat	CoeffWordLength	DenProdFracLength	ProductMode, ProductWordLength, sosMatrix
InputFormat	InputWordLength	InputFracLength	
NumAccumFormat	AccumWordLength	NumAccumFracLength	AccumMode, CastBeforeSum
NumFormat	CoeffWordLength	NumFracLength	CoeffAutoScale, Signed, sosMatrix
NumProdFormat	CoeffWordLength	NumProdFracLength	ProductWordLength, ProductMode
OutputFormat	OutputWordLength	OutputFracLength	OutputMode
ScaleValueFormat	CoeffWordLength	ScaleValueFracLength	CoeffAutoScale, ScaleValues
StageInputFormat	StageInputWordLength	StageInputFracLength	StageInputAutoScale
StageOutputFormat	StageOutputWordLength	StageOutputFracLength	StageOutputAutoScale
StateFormat	StateWordLength	StateFracLength	CastBeforeSum, States

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label DenProdFormat, which always follows a denominator coefficient multiplication element in the signal flow. The label indicates that denominator coefficients leave the multiplication element with the word length and fraction length associated with product operations that include denominator coefficients. From reviewing the table, you see that the DenProdFormat refers to the properties ProdWordLength, ProductMode and DenProdFracLength that fully define the denominator format after multiply (or product) operations.

Properties

In this table you see the properties associated with second-order section implementation of direct-form II `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to “Fixed-Point Filter Properties” on page 8-3.

Property Name	Brief Description
`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.
`Arithmetic`	Defines the arithmetic the filter uses. Gives you the options `double`, `single`, and `fixed`. In short, this property defines the operating mode for your filter.

Property Name	Brief Description
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to `false` enables you to change the `NumFracLength` and `DenFracLength` properties to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
DenAccumFracLength	Specifies the fraction length used to interpret data in the accumulator used to hold the results of sum operations. You can change the value for this property when you set `AccumMode` to `SpecifyPrecision`.
DenFracLength	Set the fraction length the filter uses to interpret denominator coefficients. `DenFracLength` is always available, but it is read-only until you set `CoeffAutoScale` to `false`.
DenProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving denominator coefficients. You can change this property value when you set `ProductMode` to `SpecifyPrecision`.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.

Property Name	Brief Description
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
NumAccumFracLength	Specifies how the filter algorithm interprets the results of addition operations involving numerator coefficients. You can change the value of this property after you set AccumMode to SpecifyPrecision.
NumFracLength	Sets the fraction length used to interpret the value of numerator coefficients.
NumProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving numerator coefficients. Available to be changed when you set ProductMode to SpecifyPrecision.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.

Property Name	Brief Description
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: ▪ AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. ▪ BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. ▪ SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
ScaleValueFracLength	Scale values work with SOS filters. Setting this property controls how your filter interprets the scale values by setting the fraction length. Only available when you disable AutoScaleMode by setting it to false.
ScaleValues	Scaling for the filter objects in SOS filters.

Property Name	Brief Description
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
SosMatrix	Holds the filter coefficients as property values. Displays the matrix in the format [sections x coefficients/section datatype]. A [15x6 double] SOS matrix represents a filter with 6 coefficients per section and 15 sections, using data type double to represent the coefficients.
StageInputAutoScale	Tells the filter whether to set the stage input data format to minimize the occurrence of overflow conditions.
StageInputFracLength	Lets you set the fraction length for stage inputs in SOS filters, if you set StageInputAutoScale to false.
StageInputWordLength	Lets you set the word length for stage inputs in SOS filters, if you set StageInputAutoScale to false.
StageOutputAutoScale	Tells the filter whether to set the stage output data format to minimize the occurrence of overflow conditions.
StageOutputFracLength	Lets you set the fraction length for stage outputs in SOS filters, if you set StageOutputAutoScale to false.
StageOutputWordLength	Lets you set the word length for stage outputs in SOS filters, if you set StageOutputAutoScale to false.

Property Name	Brief Description
StateFracLength	When you set StateAutoScale to false, you enable the StateFracLength property that lets you set the fraction length applied to interpret the filter states.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions.
StateWordLength	Sets the word length used to represent the filter states.

Examples

Specify a second-order section, direct-form II dfilt object for a Butterworth filter converted to second-order sections, with the following code:

```
[z,p,k] = butter(30,0.5);
[s,g] = zp2sos(z,p,k);
hd = dfilt.df2sos(s,g)

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [15x6 double]
             ScaleValues: [16x1 double]
        PersistentMemory: false
                  States: [2x15 double]
     NumSamplesProcessed: 0
```

With the SOS filter constructed, now change the filter operation to single-precision filtering, and then to fixed-point filtering.

```
set(hd,'arithmetic','single')
hd

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
```

```
                Arithmetic: 'single'
                 sosMatrix: [15x6 double]
               ScaleValues: [16x1 double]
          PersistentMemory: false
                    States: [2x15 single]
       NumSamplesProcessed: 0

hd.arithmetic='fixed'

hd =

          FilterStructure: 'Direct-Form II, Second-Order Sections'
                Arithmetic: 'fixed'
                 sosMatrix: [15x6 double]
               ScaleValues: [16x1 double]
          PersistentMemory: false
                    States: [1x1 embedded.fi]
       NumSamplesProcessed: 0

           CoeffWordLength: 16
            CoeffAutoScale: true
                    Signed: true

           InputWordLength: 16
           InputFracLength: 15

      StageInputWordLength: 16
       StageInputAutoScale: true

     StageOutputWordLength: 16
      StageOutputAutoScale: true

          OutputWordLength: 16
                OutputMode: 'AvoidOverflow'

           StateWordLength: 16
           StateFracLength: 15

               ProductMode: 'FullPrecision'
```

```
        AccumMode: 'KeepMSB'
  AccumWordLength: 40
    CastBeforeSum: true

        RoundMode: 'convergent'
     OverflowMode: 'wrap'
```

See Also `dfilt`, `dfilt.df1sos`, `dfilt.df1tsos`, `dfilt.df2tsos`

Purpose

Construct a discrete-time, direct-form II transposed filter object that can perform single-precision or fixed-point filtering

Syntax

Refer to `dfilt.df2t` in the Signal Processing Toolbox.

Description

`hd = dfilt.df2t(b,a)` returns a discrete-time, direct-form II transposed filter object `hd`, with numerator coefficients `b` and denominator coefficients `a`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

- To change to fixed-point filtering, enter

```
set(hd,'arithmetic','fixed');
```

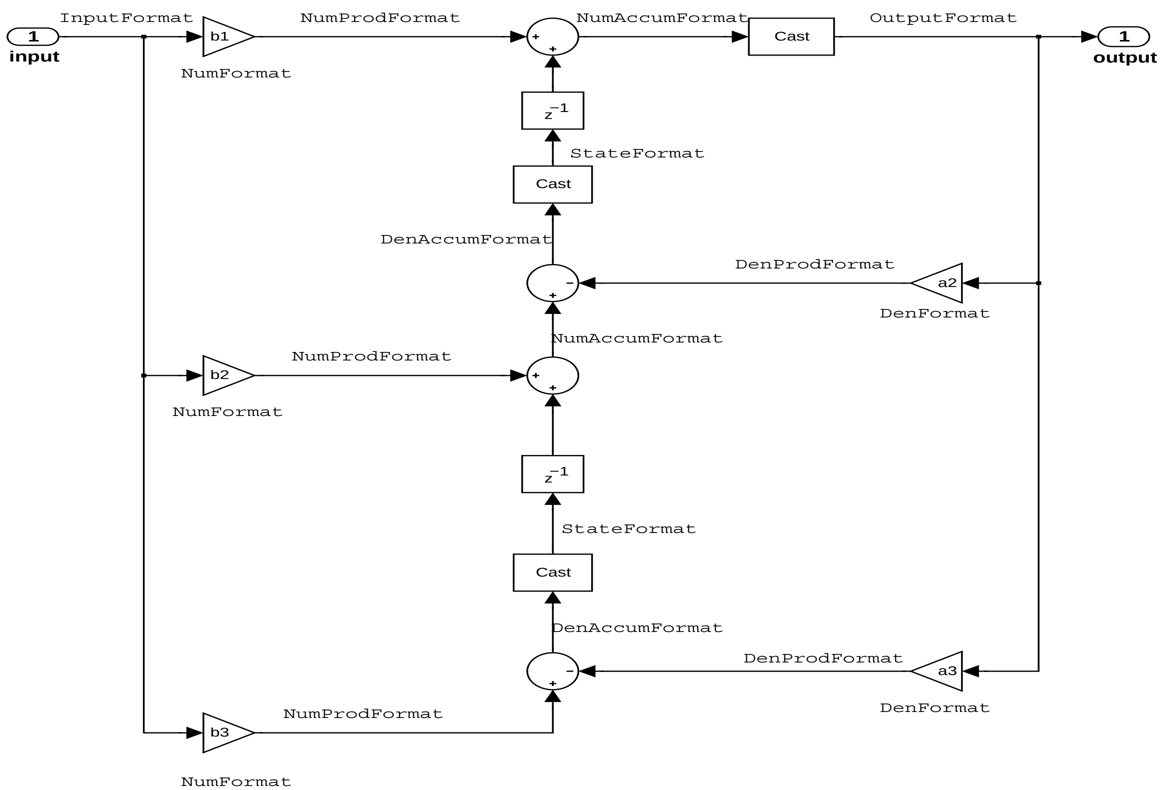
For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.df2t` returns a default, discrete-time, direct-form II transposed filter object `hd`, with `b`=1 and `a`=1. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0. To allow you to change the arithmetic setting to `fixed` or `single`, `a(1)` must be equal to 1.

Fixed-Point Filter Structure

The figure below shows the signal flow for the direct-form II transposed filter implemented by `dfilt.df2t`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word “format.” In this use, “format” means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
DenAccumFormat	`AccumWordLength`	`DenAccumFracLength`	`AccumMode`, `CastBeforeSum`
DenFormat	`CoeffWordLength`	`DenFracLength`	`CoeffAutoScale`, `Signed`, `Denominator`
DenProdFormat	`CoeffWordLength`	`DenProdFracLength`	`ProductMode`, `ProductWordLength`
InputFormat	`InputWordLength`	`InputFracLength`	
NumAccumFormat	`AccumWordLength`	`NumAccumFracLength`	`AccumMode`, `CastBeforeSum`
NumFormat	`CoeffWordLength`	`NumFracLength`	`CoeffAutoScale`, `Signed`, `Numerator`
NumProdFormat	`CoeffWordLength`	`NumProdFracLength`	`ProductWordLength`, `ProductMode`
OutputFormat	`OutputWordLength`	`OutputFracLength`	`OutputMode`
StateFormat	`StateWordLength`	`StateFracLength`	`States`

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label DenProdFormat, which always follows a denominator coefficient multiplication element in the signal flow. The label indicates that denominator coefficients leave the multiplication element with the word length and fraction length associated with product operations that include denominator coefficients. From reviewing the table, you see that the

DenProdFormat refers to the properties `ProdWordLength`, `ProductMode` and `DenProdFracLength` that fully define the denominator format after multiply (or product) operations.

Properties

In this table you see the properties associated with df2t implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to “Fixed-Point Filter Properties” on page 8-3.

Property Name	Brief Description
`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.

Property Name	Brief Description
Arithmetic	Defines the arithmetic the filter uses. Gives you the options double, single, and fixed. In short, this property defines the operating mode for your filter.
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength and DenFracLength properties to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
DenAccumFracLength	Specifies the fraction length used to interpret data in the accumulator used to hold the results of sum operations. You can change the value for this property when you set AccumMode to SpecifyPrecision.
DenFracLength	Set the fraction length the filter uses to interpret denominator coefficients. DenFracLength is always available, but it is read-only until you set CoeffAutoScale to false.
Denominator	Holds the denominator coefficients for IIR filters.

Property Name	Brief Description
DenProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving denominator coefficients. You can change this property value when you set ProductMode to SpecifyPrecision.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
NumAccumFracLength	Specifies how the filter algorithm interprets the results of addition operations involving numerator coefficients. You can change the value of this property after you set AccumMode to SpecifyPrecision.
Numerator	Holds the numerator coefficient values for the filter.
NumFracLength	Sets the fraction length used to interpret the value of numerator coefficients.
NumProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving numerator coefficients. Available to be changed when you set ProductMode to SpecifyPrecision.

Property Name	Brief Description
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: • AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. • BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. • SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.

Property Name	Brief Description
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.

Property Name	Brief Description
StateAutoScale	Setting autoscaling for filter states to true reduces the possibility of overflows occurring during fixed-point operations. Set to false, StateAutoScale lets the filter select the fraction length to limit the overflow potential.
StateFracLength	When you set StateAutoScale to false, you enable the StateFracLength property that lets you set the fraction length applied to interpret the filter states.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions.
StateWordLength	Sets the word length used to represent the filter states.

Examples

Create a fixed-point filter by specifying a second-order direct-form II transposed filter structure for a dfilt object, and then converting the double-precision arithmetic setting to fixed-point.

```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hd = dfilt.df2t(b,a)

hd =

         FilterStructure: 'Direct-Form II Transposed'
              Arithmetic: 'double'
               Numerator: [0.3000 0.6000 0.3000]
             Denominator: [1 0 0.2000]
        PersistentMemory: false
                  States: [2x1 double]
     NumSamplesProcessed: 0
```

```
set(hd,'arithmetic','fixed')
hd

hd =

         FilterStructure: 'Direct-Form II Transposed'
              Arithmetic: 'fixed'
               Numerator: [0.3000 0.6000 0.3000]
             Denominator: [1 0 0.2000]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
        OutputFracLength: 15

         StateWordLength: 16
          StateAutoScale: true

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'
```

See Also

dfilt, dfilt.df1, dfilt.df1t, dfilt.df2

dfilt.df2tsos

Purpose Construct a fixed-point or single-precision floating-point, discrete-time, second-order section direct-form II transposed filter object

Syntax Refer to `dfilt.df2tsos` in the Signal Processing Toolbox.

Description `hd = dfilt.df2sos(s)` returns a discrete-time, second-order section, direct-form II, transposed filter object `hd`, with coefficients given in the matrix `s`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

  ```
  set(hd,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hd,'arithmetic','fixed');
  ```

For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.df2tsos(b1,a1,b2,a2,...)` returns a discrete-time, second-order section, direct-form II, transposed filter object `hd`, with coefficients for the first section given in the `b1` and `a1` vectors, for the second section given in the `b2` and `a2` vectors, etc.

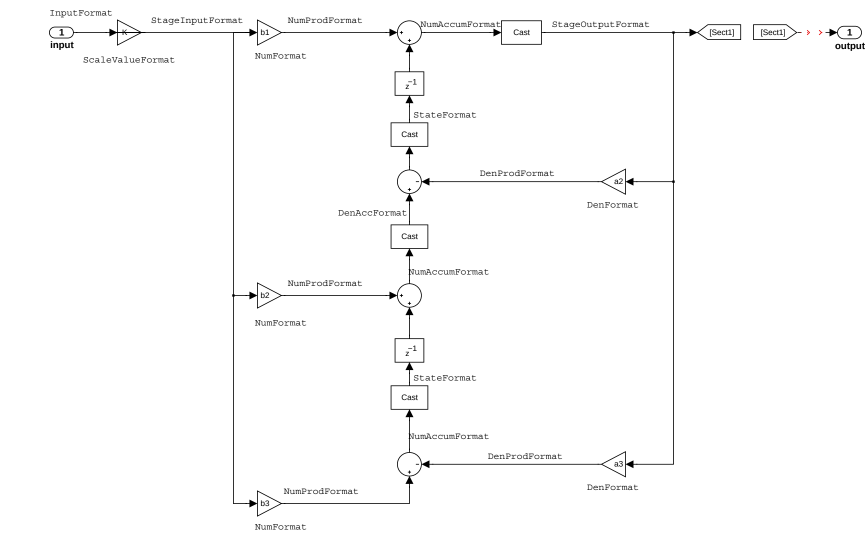
`hd = dfilt.df2tsos(...,g)` includes a gain vector `g`. The elements of `g` are the gains for each section. The maximum length of `g` is the number of sections plus one. If `g` is not specified, all gains default to one.

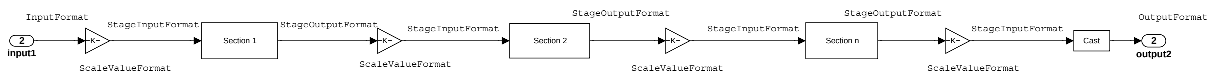
`hd = dfilt.df2tsos` returns a default, discrete-time, second-order section, direct-form II, transposed filter object, `hd`. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0. To allow you to change the arithmetic setting to `fixed` or `single`, `a(1)` must be equal to 1.

Fixed-Point Filter Structure

The figure below shows the signal flow for the second-order section transposed direct-form II filter implemented by `dfilt.dftsos`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.





Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word “format.” In this use, “format” means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
DenAccumFormat	AccumWordLength	DenAccumFracLength	AccumMode, CastBeforeSum
DenFormat	CoeffWordLength	DenFracLength	CoeffAutoScale, Signed, Denominator
DenProdFormat	CoeffWordLength	DenProdFracLength	ProductMode, ProductWordLength
InputFormat	InputWordLength	InputFracLength	
NumAccumFormat	AccumWordLength	NumAccumFracLength	AccumMode, CastBeforeSum
NumFormat	CoeffWordLength	NumFracLength	CoeffAutoScale, Signed, Numerator
NumProdFormat	CoeffWordLength	NumProdFracLength	ProductWordLength, ProductMode
OutputFormat	OutputWordLength	OutputFracLength	OutputMode
ScaleValueFormat	CoeffWordLength	ScaleValueFracLength	CoeffAutoScale, ScaleValues
StageInputFormat	StageInputWordLength	StageInputFracLength	StageInputAutoScale
StageOutputFormat	StageOutputWordLength	StageOutputFracLength	StageOutputAutoScale
StateFormat	StateWordLength	StateFracLength	States

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label DenProdFormat, which always follows a denominator coefficient multiplication element in the signal flow. The label indicates that denominator coefficients leave the multiplication element with the word length and fraction length associated with product operations that include denominator coefficients. From reviewing the table, you see that the DenProdFormat refers to the properties `ProdWordLength`, `ProductMode` and `DenProdFracLength` that fully define the denominator format after multiply (or product) operations.

Properties

In this table you see the properties associated with second-order section implementation of transposed direct-form II `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to "Fixed-Point Filter Properties" on page 8-3.

Property Name	Brief Description
`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.

Property Name	Brief Description
Arithmetic	Defines the arithmetic the filter uses. Gives you the options double, single, and fixed. In short, this property defines the operating mode for your filter.
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength and DenFracLength properties to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
DenAccumFracLength	Specifies the fraction length used to interpret data in the accumulator used to hold the results of sum operations. You can change the value for this property when you set AccumMode to SpecifyPrecision.
DenFracLength	Set the fraction length the filter uses to interpret denominator coefficients. DenFracLength is always available, but it is read-only until you set CoeffAutoScale to false.
DenProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving denominator coefficients. You can change this property value when you set ProductMode to SpecifyPrecision.

Property Name	Brief Description
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
NumAccumFracLength	Specifies how the filter algorithm interprets the results of addition operations involving numerator coefficients. You can change the value of this property after you set AccumMode to SpecifyPrecision.
NumFracLength	Sets the fraction length used to interpret the value of numerator coefficients.
NumProdFracLength	Specifies how the filter algorithm interprets the results of product operations involving numerator coefficients. Available to be changed when you set ProductMode to SpecifyPrecision.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.

Property Name	Brief Description
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: - AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. - BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. - SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
ScaleValueFracLength	Scale values work with SOS filters. Setting this property controls how your filter interprets the scale values by setting the fraction length. Only available when you disable AutoScaleMode by setting it to false.

Property Name	Brief Description
ScaleValues	Scaling for the filter objects in SOS filters.
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
SosMatrix	Holds the filter coefficients as property values—you use set and get to modify them. Displays the matrix in the format [sections x coefficients/section data type]. A [15x6 double] SOS matrix represents a filter with 6 coefficients per section and 15 sections, using data type double to represent the coefficients.
StageInputFracLength	Lets you set the fraction length for stage inputs in SOS filters, if you set StageInputAutoScale to false.
StageInputWordLength	Lets you set the word length for stage inputs in SOS filters, if you set StageInputAutoScale to false.
StageOutputAutoScale	Tells the filter whether to set the stage output data format to minimize the occurrence of overflow conditions.
StageOutputFracLength	Lets you set the fraction length for stage outputs in SOS filters, if you set StageOutputAutoScale to off.
StageOutputWordLength	Lets you set the word length for stage outputs in SOS filters, if you set StageOutputAutoScale to false.

Property Name	Brief Description
StateAutoScale	Setting autoscaling for filter states to true reduces the possibility of overflows occurring during fixed-point operations. Set to false, StateAutoScale lets the filter select the fraction length to limit the overflow potential.
StateFracLength	When you set StateAutoScale to false, you enable the StateFracLength property that lets you set the fraction length applied to interpret the filter states.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions.
StateWordLength	Sets the word length used to represent the filter states.

Examples

Construct a second-order section Butterworth filter for fixed-point filtering. Start by specifying a Butterworth filter, and then convert the filter to second-order sections, with the following code:

```
[z,p,k] = butter(30,0.5);
[s,g] = zp2sos(z,p,k);
hd = dfilt.df2tsos(s,g)

hd =

         FilterStructure: [1x48 char]
              Arithmetic: 'double'
               sosMatrix: [15x6 double]
             ScaleValues: [16x1 double]
        PersistentMemory: false
                  States: [2x15 double]
     NumSamplesProcessed: 0
```

Now change the setting of the property `Arithmetic` to convert the filter to fixed-point operation.

```
hd.arithmetic='fixed'

hd =

         FilterStructure: [1x48 char]
              Arithmetic: 'fixed'
               sosMatrix: [15x6 double]
             ScaleValues: [16x1 double]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

    StageInputWordLength: 16
    StageInputFracLength: 15

   StageOutputWordLength: 16
   StageOutputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

         StateWordLength: 16
          StateAutoScale: true

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true
```

```
           RoundMode: 'convergent'
        OverflowMode: 'wrap'
```

See Also

`dfilt`, `dfilt.df1sos`, `dfilt.df1tsos`, `dfilt.df2sos`

Purpose

Construct a discrete-time, direct-form antisymmetric FIR filter object for fixed-point or single-precision floating-point filtering

Syntax

Refer to `dfilt.dfasymfir` in the Signal Processing Toolbox.

Description

`hd = dfilt.dfasymfir(b)` returns a discrete-time, direct-form, antisymmetric FIR filter object `hd`, with numerator coefficients `b`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

  ```
  set(hd,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hd,'arithmetic','fixed');
  ```

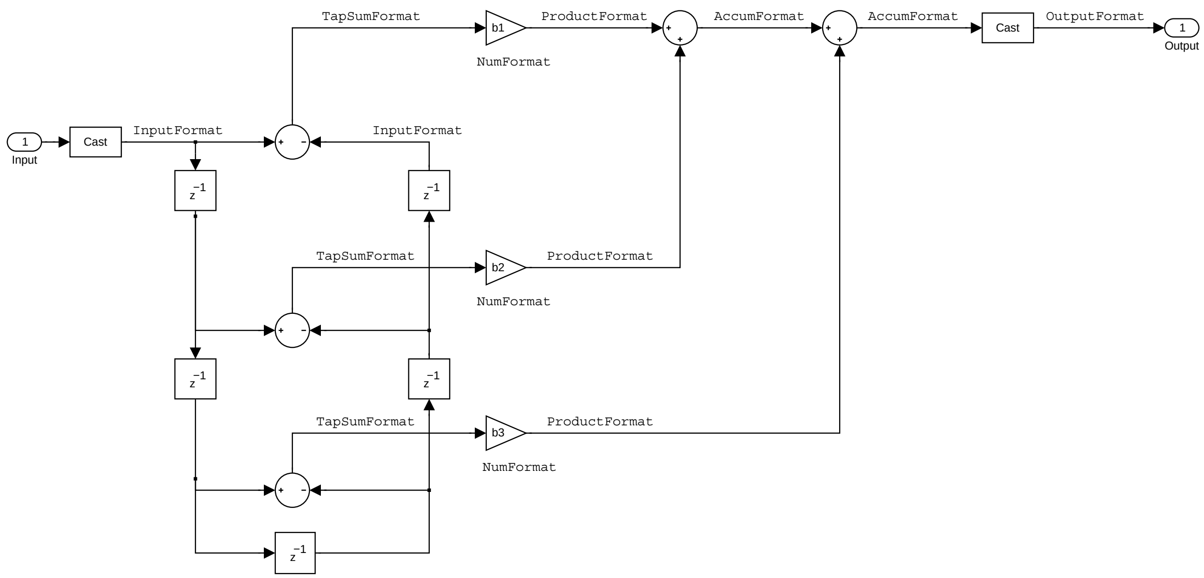
For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.dfasymfir` returns a default, discrete-time, direct-form, antisymmetric FIR filter object `hd`, with `b=1`. This filter passes the input through to the output unchanged.

Note Only the coefficients in the first half of vector `b` are used because `dfilt.dfasymfir` assumes the coefficients in the second half are antisymmetric to those in the first half. For example, in the figure coefficients, b(4) = -b(3), b(5) = -b(2), and b(6) = -b(1).

Fixed-Point Filter Structure

The figure below shows the signal flow for the odd-order antisymmetric FIR filter implemented by `dfilt.dfasymfir`. The even-order filter uses similar flow. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word “format.” In this use, “format” means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to

the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
AccumFormat	`AccumWordLength`	`AccumFracLength`	
InputFormat	`InputWordLength`	`InputFracLength`	
NumFormat	`CoeffWordLength`	`NumFracLength`	`CoeffAutoScale`, `Signed`, `Numerator`
OutputFormat	`OutputWordLength`	`OutputFracLength`	
ProductFormat	`ProductWordLength`	`ProductFracLength`	
TapSumFormat	`InputWordLength`	`InputFracLength`	`InputFormat`

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label ProductFormat, which always follows a coefficient multiplication element in the signal flow. The label indicates that coefficients leave the multiplication element with the word length and fraction length associated with product operations that include coefficients. From reviewing the table, you see that the ProductFormat refers to the properties `ProductFracLength` and `ProductWordLength` that fully define the coefficient format after multiply (or product) operations.

Properties

In this table you see the properties associated with an antisymmetric FIR implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where hd is a filter.

For further information about the properties of this filter or any dfilt object, refer to "Fixed-Point Filter Properties" on page 8-3.

Name	Values	Description
AccumFracLength	Any positive or negative integer number of bits [27]	Specifies the fraction length used to interpret data output by the accumulator.
AccumWordLength	Any integer number of bits[33]	Sets the word length used to store data in the accumulator.
Arithmetic	fixed for fixed-point filters	Setting this to fixed allows you to modify other filter properties to customize your fixed-point filter.
CoeffAutoScale	[true], false	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength property value to specify the precision used.
CoeffWordLength	Any integer number of bits [16]	Specifies the word length to apply to filter coefficients.

Name	Values	Description
FilterInternals	[FullPrecision], SpecifyPrecision	Controls whether the filter automatically sets the output word and fraction lengths, product word and fraction lengths, and the accumulator word and fraction lengths to maintain the best precision results during filtering. The default value, FullPrecision, sets automatic word and fraction length determination by the filter. SpecifyPrecision makes the output and accumulator-related properties available so you can set your own word and fraction lengths for them.
InputFracLength	Any positive or negative integer number of bits [15]	Specifies the fraction length the filter uses to interpret input data. Also controls TapSumFracLength.
InputWordLength	Any integer number of bits [16]	Specifies the word length applied to interpret input data. Also determines TapSumWordLength.
NumFracLength	Any positive or negative integer number of bits [14]	Sets the fraction length used to interpret the numerator coefficients.
OutputFracLength	Any positive or negative integer number of bits [29]	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set FilerInternals to SpecifyPrecision.
OutputWordLength	Any integer number of bits [33]	Determines the word length used for the output data. You make this property editable by setting FilterInternals to SpecifyPrecision.

Name	Values	Description
OverflowMode	saturate, [wrap]	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
ProductFracLength	Any positive or negative integer number of bits [27]	Specifies the fraction length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
ProductWordLength	Any integer number of bits [33]	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.

Name	Values	Description
RoundMode	[convergent], ceil,fix,floor, round	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Name	Values	Description
Signed	[true], false	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
States	fi object to match the filter arithmetic setting	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system.

Examples

Odd Order

Specify a fifth-order direct-form antisymmetric FIR filter structure for a dfilt object, hd, with the following code:

```
b = [-0.008 0.06 -0.44 0.44 -0.06 0.008];
hd = dfilt.dfasymfir(b)

hd =

     FilterStructure: 'Direct-Form Antisymmetric FIR'
          Arithmetic: 'double'
           Numerator: [-0.0080 0.0600 -0.4400 0.4400 -0.0600 0.0080]
    PersistentMemory: false

set(hd,'arithmetic','fixed')
hd =

     FilterStructure: 'Direct-Form Antisymmetric FIR'
          Arithmetic: 'fixed'
           Numerator: [-0.0080 0.0600 -0.4400 0.4400 -0.0600 0.0080]
    PersistentMemory: false

     CoeffWordLength: 16
      CoeffAutoScale: true
              Signed: true
```

```
      InputWordLength: 16
      InputFracLength: 15

      FilterInternals: 'FullPrecision'
```

Now look at the coefficients after converting hd to fixed-point format.

```
get(hd,'numerator')

ans =

   -0.0080    0.0600   -0.4400    0.4400   -0.0600    0.0080
```

Even Order

Specify a fourth-order direct-form antisymmetric FIR filter structure for dfilt object hd, with the following code:

```
b = [-0.01 0.1 0.0 -0.1 0.01];
hd = dfilt.dfasymfir(b)

hd =

     FilterStructure: 'Direct-Form Antisymmetric FIR'
          Arithmetic: 'double'
           Numerator: [-0.0100 0.1000 0 -0.1000 0.0100]
    PersistentMemory: false

hd.arithmetic='fixed'

hd =

     FilterStructure: 'Direct-Form Antisymmetric FIR'
          Arithmetic: 'fixed'
           Numerator: [-0.0100 0.1000 0 -0.1000 0.0100]
    PersistentMemory: false

     CoeffWordLength: 16
      CoeffAutoScale: true
              Signed: true
```

```
          InputWordLength: 16
          InputFracLength: 15

          FilterInternals: 'FullPrecision'

get(hd,'numerator')

ans =

   -0.0100    0.1000         0   -0.1000    0.0100
```

See Also

`dfilt`, `dfilt.dffir`, `dfilt.dffirt`, `dfilt.dfsymfir`

Purpose Construct a discrete-time, fixed-point or single-precision floating-point, direct-form FIR filter object

Syntax Refer to `dfilt.dffir` in the Signal Processing Toolbox.

Description `hd = dfilt.dffir(b)` returns a discrete-time, direct-form finite impulse response (FIR) filter object `hd`, with numerator coefficients `b`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

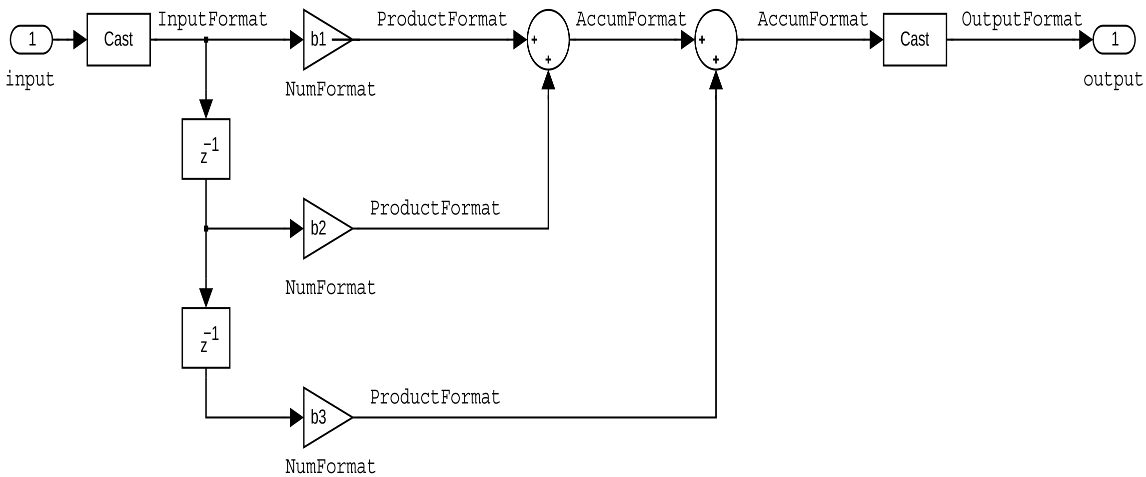
- To change to fixed-point filtering, enter

```
set(hd,'arithmetic','fixed');
```

For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.dffir` returns a default, discrete-time, direct-form FIR filter object `hd`, with `b=1`. This filter passes the input through to the output unchanged.

Fixed-Point Filter Structure The figure below shows the signal flow for the direct-form FIR filter implemented by `dfilt.dffir`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to

the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
AccumFormat	`AccumWordLength`	`AccumFracLength`	
InputFormat	`InputWordLength`	`InputFracLength`	
NumFormat	`CoeffWordLength`	`NumFracLength`	`CoeffAutoScale`, `Signed`, `Numerator`
OutputFormat	`OutputWordLength`	`OutputFracLength`	
ProductFormat	`ProductWordLength`	`ProductFracLength`	

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label ProductFormat, which always follows a coefficient multiplication element in the signal flow. The label indicates that coefficients leave the multiplication element with the word length and fraction length associated with product operations that include coefficients. From reviewing the table, you see that the ProductFormat refers to the properties `ProductFracLength` and `ProductWordLength` that fully define the coefficient format after multiply (or product) operations.

Properties

In this table you see the properties associated with direct-form FIR implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where hd is a filter.

For further information about the properties of this filter or any dfilt object, refer to “Fixed-Point Filter Properties” on page 8-3.

Name	Values	Description
AccumFracLength	Any positive or negative integer number of bits [30]	Specifies the fraction length used to interpret data output by the accumulator.
AccumWordLength	Any integer number of bits[34]	Sets the word length used to store data in the accumulator.
Arithmetic	fixed for fixed-point filters	Setting this to fixed allows you to modify other filter properties to customize your fixed-point filter.
CoeffAutoScale	[true], false	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength property value to specify the precision used.
CoeffWordLength	Any integer number of bits [16]	Specifies the word length to apply to filter coefficients.

Name	Values	Description
FilterInternals	[FullPrecision], SpecifyPrecision	Controls whether the filter automatically sets the output word and fraction lengths, product word and fraction lengths, and the accumulator word and fraction lengths to maintain the best precision results during filtering. The default value, FullPrecision, sets automatic word and fraction length determination by the filter. SpecifyPrecision makes the output and accumulator-related properties available so you can set your own word and fraction lengths for them.
InputFracLength	Any positive or negative integer number of bits [15]	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Any integer number of bits [16]	Specifies the word length applied to interpret input data.
NumFracLength	Any positive or negative integer number of bits [14]	Sets the fraction length used to interpret the numerator coefficients.
OutputFracLength	Any positive or negative integer number of bits [32]	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set FilerInternals to SpecifyPrecision.
OutputWordLength	Any integer number of bits [39]	Determines the word length used for the output data. You make this property editable by setting FilterInternals to SpecifyPrecision.

Name	Values	Description
`OverflowMode`	`saturate, [wrap]`	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either `saturate` (limit the output to the largest positive or negative representable value) or `wrap` (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
`ProductFracLength`	Any positive or negative integer number of bits `[30]`	Specifies the fraction length to use for multiplication operation results. This property becomes writable (you can change the value) when you set `ProductMode` to `SpecifyPrecision`.
`ProductWordLength`	Any integer number of bits `[32]`	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set `ProductMode` to `SpecifyPrecision`.

Name	Values	Description
`RoundMode`	`[convergent]`, `ceil`,`fix`,`floor`, `round`	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • `convergent`—Round up to the next allowable quantized value. • `ceil`—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to `1`. • `fix`—Round negative numbers up and positive numbers down to the next allowable quantized value. • `floor`—Round down to the next allowable quantized value. • `round`—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Name	Values	Description
Signed	[true], false	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
States	fi object to match the filter arithmetic setting	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system.

Examples

Specify a second-order direct-form FIR filter structure for a dfilt object hd, with the following code that constructs the filter in double-precision format and then converts the filter to fixed-point operation:

```
b = [0.05 0.9 0.05];
hd = dfilt.dffir(b)

hd =

     FilterStructure: 'Direct-Form FIR'
          Arithmetic: 'double'
           Numerator: [0.0500 0.9000 0.0500]
    PersistentMemory: false

hd.arithmetic='fixed'

hd =

     FilterStructure: 'Direct-Form FIR'
          Arithmetic: 'fixed'
           Numerator: [0.0500 0.9000 0.0500]
    PersistentMemory: false

     CoeffWordLength: 16
      CoeffAutoScale: true
```

```
              Signed: true

     InputWordLength: 16
     InputFracLength: 15

     FilterInternals: 'FullPrecision'

hd.filterInternals='specifyPrecision'

hd =

     FilterStructure: 'Direct-Form FIR'
          Arithmetic: 'fixed'
           Numerator: [0.0500 0.9000 0.0500]
    PersistentMemory: false

     CoeffWordLength: 16
      CoeffAutoScale: true
              Signed: true

     InputWordLength: 16
     InputFracLength: 15

     FilterInternals: 'SpecifyPrecision'

    OutputWordLength: 34
    OutputFracLength: 30

   ProductWordLength: 32
   ProductFracLength: 30

     AccumWordLength: 34
     AccumFracLength: 30

           RoundMode: 'convergent'
        OverflowMode: 'wrap'
```

See Also

dfilt, dfilt.dfasymfir, dfilt.dffirt, dfilt.dfsymfir

Purpose

Construct a discrete-time, direct-form FIR transposed filter object for fixed-point or single-precision floating-point filtering

Syntax

Refer to `dfilt.dffirt` in the Signal Processing Toolbox.

Description

`hd = dfilt.dffirt(b)` returns a discrete-time, direct-form FIR transposed filter object `hd`, with numerator coefficients `b`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

- To change to fixed-point filtering, enter

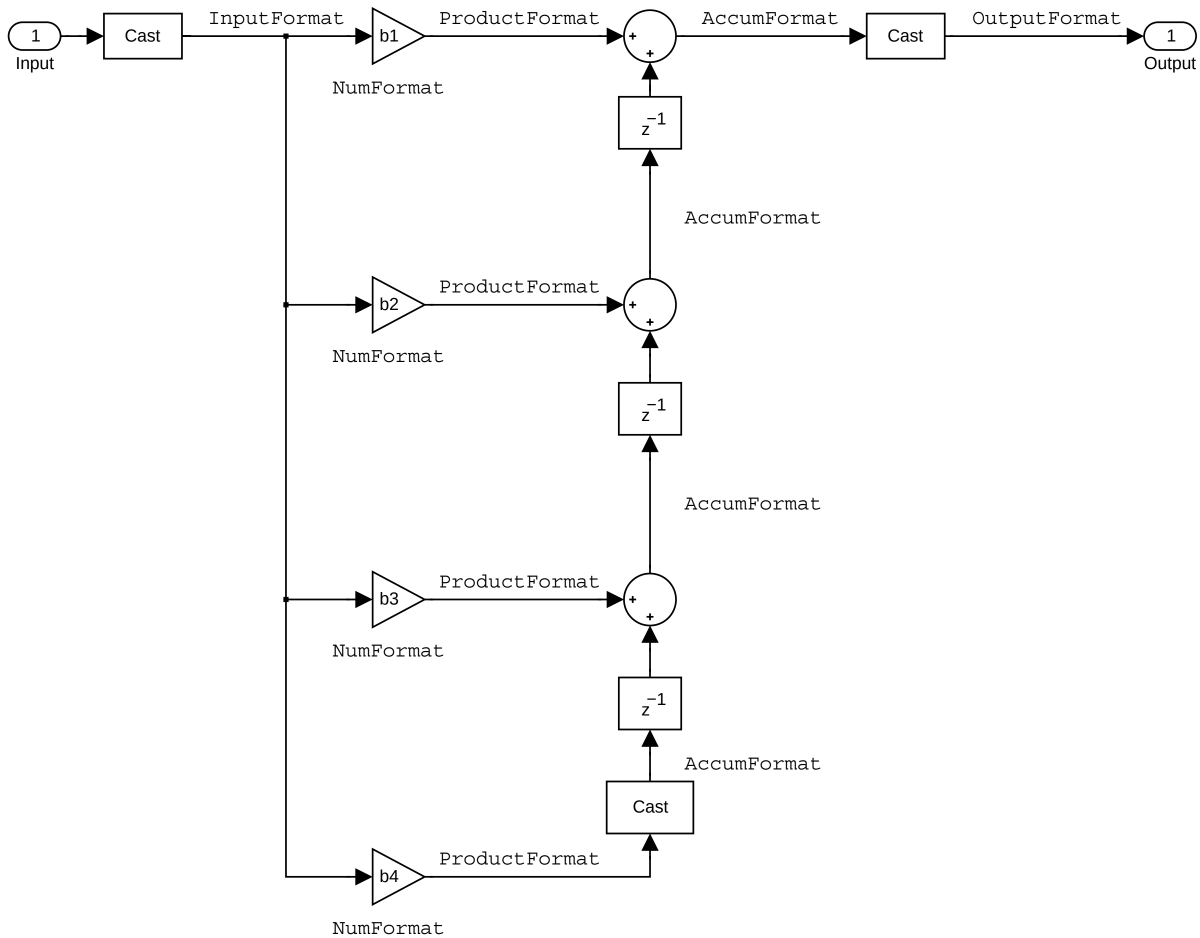
```
set(hd,'arithmetic','fixed');
```

For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.dffirt` returns a default, discrete-time, direct-form FIR transposed filter object `hd`, with `b`=1. This filter passes the input through to the output unchanged.

Fixed-Point Filter Structure

The figure below shows the signal flow for the transposed direct-form FIR filter implemented by `dfilt.dffirt`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
AccumFormat	`AccumWordLength`	`AccumFracLength`	
InputFormat	`InputWordLength`	`InputFracLength`	
NumFormat	`CoeffWordLength`	`NumFracLength`	`CoeffAutoScale`, `Signed`, `Numerator`
OutputFormat	`OutputWordLength`	`OutputFracLength`	
ProductFormat	`ProductWordLength`	`ProductFracLength`	

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label ProductFormat, which always follows a coefficient multiplication element in the signal flow. The label indicates that coefficients leave the multiplication element with the word length and fraction length associated with product operations that include coefficients. From reviewing the table, you see that the ProductFormat refers to the properties `ProductFracLength` and `ProductWordLength` that fully define the coefficient format after multiply (or product) operations.

Properties

In this table you see the properties associated with the transposed direct-form FIR implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to "Fixed-Point Filter Properties" on page 8-3.

Name	Values	Description
`AccumFracLength`	Any positive or negative integer number of bits `[30]`	Specifies the fraction length used to interpret data output by the accumulator.
`AccumWordLength`	Any integer number of bits`[34]`	Sets the word length used to store data in the accumulator.
`Arithmetic`	`fixed` for fixed-point filters	Setting this to `fixed` allows you to modify other filter properties to customize your fixed-point filter.
`CoeffAutoScale`	`[true]`, `false`	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to `false` enables you to change the `NumFracLength` property value to specify the precision used.
`CoeffWordLength`	Any integer number of bits `[16]`	Specifies the word length to apply to filter coefficients.

Name	Values	Description
`FilterInternals`	`[FullPrecision]`, `SpecifyPrecision`	Controls whether the filter automatically sets the output word and fraction lengths, product word and fraction lengths, and the accumulator word and fraction lengths to maintain the best precision results during filtering. The default value, `FullPrecision`, sets automatic word and fraction length determination by the filter. `SpecifyPrecision` makes the output and accumulator-related properties available so you can set your own word and fraction lengths for them.
`InputFracLength`	Any positive or negative integer number of bits `[15]`	Specifies the fraction length the filter uses to interpret input data.
`InputWordLength`	Any integer number of bits `[16]`	Specifies the word length applied to interpret input data.
`NumFracLength`	Any positive or negative integer number of bits `[14]`	Sets the fraction length used to interpret the numerator coefficients.
`OutputFracLength`	Any positive or negative integer number of bits `[30]`	Determines how the filter interprets the filter output data. You can change the value of `OutputFracLength` when you set `FilerInternals` to `SpecifyPrecision`.
`OutputWordLength`	Any integer number of bits `[34]`	Determines the word length used for the output data. You make this property editable by setting `FilterInternals` to `SpecifyPrecision`.

Name	Values	Description
`OverflowMode`	`saturate, [wrap]`	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either `saturate` (limit the output to the largest positive or negative representable value) or `wrap` (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Name	Values	Description
RoundMode	[convergent], ceil,fix,floor, round	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Name	Values	Description
`Signed`	`[true]`, `false`	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
`States`	`fi` object to match the filter arithmetic setting	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use `fi` objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system.

Examples

Specify a second-order direct-form FIR transposed filter structure for a `dfilt` object, `hd`, with the following code:

```
b = [0.05 0.9 0.05];
hd = dfilt.dffirt(b)

hd =

     FilterStructure: 'Direct-Form FIR Transposed'
          Arithmetic: 'double'
           Numerator: [0.0500 0.9000 0.0500]
    PersistentMemory: false
```

Now use the filter property `Arithmetic` to change the filter to fixed-point format.

```
set(hd,'arithmetic','fixed')
hd

hd =

     FilterStructure: 'Direct-Form FIR Transposed'
          Arithmetic: 'fixed'
           Numerator: [0.0500 0.9000 0.0500]
    PersistentMemory: false
```

```
       CoeffWordLength: 16
        CoeffAutoScale: true
                Signed: true

       InputWordLength: 16
       InputFracLength: 15

       FilterInternals: 'FullPrecision'

hd.filterInternals='specifyPrecision'

hd =

       FilterStructure: 'Direct-Form FIR Transposed'
            Arithmetic: 'fixed'
             Numerator: [0.0500 0.9000 0.0500]
      PersistentMemory: false

       CoeffWordLength: 16
        CoeffAutoScale: true
                Signed: true

       InputWordLength: 16
       InputFracLength: 15

       FilterInternals: 'SpecifyPrecision'

      OutputWordLength: 34
      OutputFracLength: 30

     ProductWordLength: 32
     ProductFracLength: 30

       AccumWordLength: 34
       AccumFracLength: 30

             RoundMode: 'convergent'
          OverflowMode: 'wrap'
```

See Also `dfilt`, `dfilt.dffir`, `dfilt.dfasymfir`, `dfilt.dfsymfir`

dfilt.dfsymfir

Purpose Construct a discrete-time, direct-form symmetric FIR filter object that uses fixed-point or single-precision floating-point filtering

Syntax Refer to `dfilt.dfsymfir` in the Signal Processing Toolbox.

Description `hd = dfilt.dfsymfir(b)` returns a discrete-time, direct-form symmetric FIR filter object `hd`, with numerator coefficients `b`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

- To change to fixed-point filtering, enter

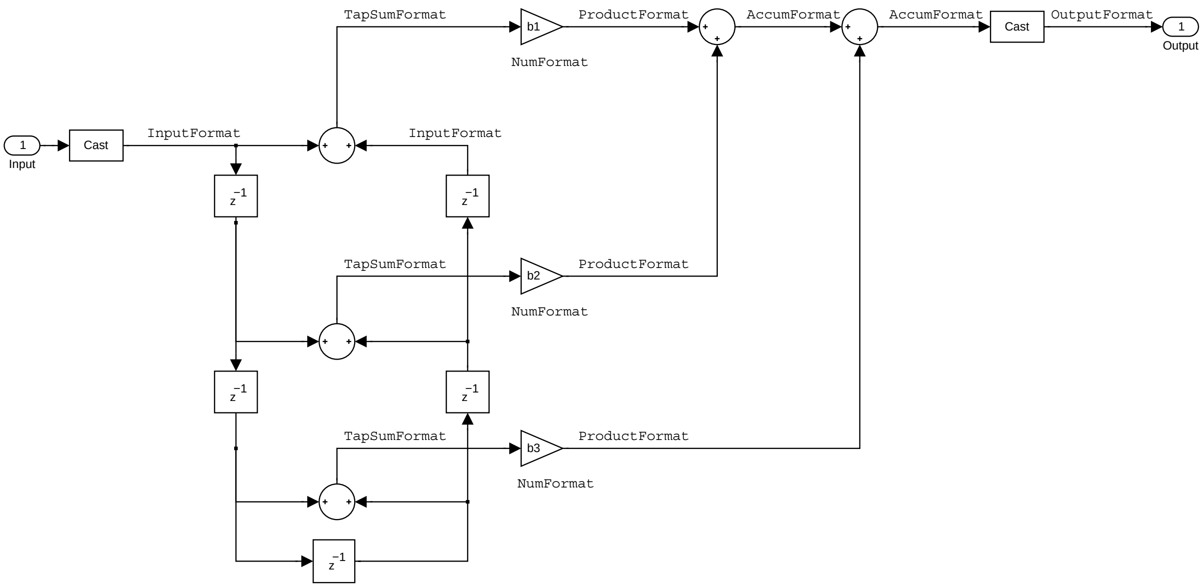
```
set(hd,'arithmetic','fixed');
```

For more information about the property `Arithmetic`, refer to “Arithmetic” on page 8-22.

`hd = dfilt.dfsymfir` returns a default, discrete-time, direct-form symmetric FIR filter object `hd`, with `b=1`. This filter passes the input through to the output unchanged.

Note Only the coefficients in the first half of vector `b` are used because `dfilt.dfsymfir` assumes the coefficients in the second half are symmetric to those in the first half. In the figure below, for example, b(3) = 0, b(4) = b(2) and b(5) = b(1).

Fixed-Point Filter Structure In the following figure you see the signal flow diagram for the symmetric FIR filter that `dfilt.dfsymfir` implements.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to

the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
AccumFormat	`AccumWordLength`	`AccumFracLength`	
InputFormat	`InputWordLength`	`InputFracLength`	
NumFormat	`CoeffWordLength`	`NumFracLength`	`CoeffAutoScale`, `Signed`, `Numerator`
OutputFormat	`OutputWordLength`	`OutputFracLength`	
ProductFormat	`ProductWordLength`	`ProductFracLength`	
TapSumFormat	`InputWordLength`	`InputFracLength`	

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label ProductFormat, which always follows a coefficient multiplication element in the signal flow. The label indicates that coefficients leave the multiplication element with the word length and fraction length associated with product operations that include coefficients. From reviewing the table, you see that the ProductFormat refers to the properties `ProductFracLength` and `ProductWordLength` that fully define the coefficient format after multiply (or product) operations.

Properties

In this table you see the properties associated with the symmetric FIR implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where hd is a filter.

For further information about the properties of this filter or any dfilt object, refer to "Fixed-Point Filter Properties" on page 8-3..

Name	Values	Description
AccumFracLength	Any positive or negative integer number of bits [27]	Specifies the fraction length used to interpret data output by the accumulator.
AccumWordLength	Any integer number of bits[33]	Sets the word length used to store data in the accumulator.
Arithmetic	fixed for fixed-point filters	Setting this to fixed allows you to modify other filter properties to customize your fixed-point filter.
CoeffAutoScale	[true], false	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength property value to specify the precision used.
CoeffWordLength	Any integer number of bits [16]	Specifies the word length to apply to filter coefficients.

Name	Values	Description
FilterInternals	[FullPrecision], SpecifyPrecision	Controls whether the filter automatically sets the output word and fraction lengths, product word and fraction lengths, and the accumulator word and fraction lengths to maintain the best precision results during filtering. The default value, FullPrecision, sets automatic word and fraction length determination by the filter. SpecifyPrecision makes the output and accumulator-related properties available so you can set your own word and fraction lengths for them.
InputFracLength	Any positive or negative integer number of bits [15]	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Any integer number of bits [16]	Specifies the word length applied to interpret input data.
NumFracLength	Any positive or negative integer number of bits [14]	Sets the fraction length used to interpret the numerator coefficients.
OutputFracLength	Any positive or negative integer number of bits [29]	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set FilerInternals to SpecifyPrecision.
OutputWordLength	Any integer number of bits [33]	Determines the word length used for the output data. You make this property editable by setting FilterInternals to SpecifyPrecision.

Name	Values	Description
OverflowMode	saturate, [wrap]	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
ProductFracLength	Any positive or negative integer number of bits [29]	Specifies the fraction length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
ProductWordLength	Any integer number of bits [33]	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.

Name	Values	Description
RoundMode	[convergent], ceil,fix,floor, round	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Name	Values	Description
`Signed`	`[true], false`	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
`States`	`fi` object to match the filter arithmetic setting	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use `fi` objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system.

Examples

Odd Order

Specify a fifth-order direct-form symmetric FIR filter structure for a `dfilt` object `hd`, with the following code:

```
b = [-0.008 0.06 0.44 0.44 0.06 -0.008];
hd = dfilt.dfsymfir(b)

hd =

     FilterStructure: 'Direct-Form Symmetric FIR'
          Arithmetic: 'double'
           Numerator: [-0.0080 0.0600 0.4400 0.4400 0.0600 -0.0080]
    PersistentMemory: false

set(hd,'arithmetic','fixed')
hd

hd =

     FilterStructure: 'Direct-Form Symmetric FIR'
          Arithmetic: 'fixed'
           Numerator: [-0.0080 0.0600 0.4400 0.4400 0.0600 -0.0080]
    PersistentMemory: false
```

```
      CoeffWordLength: 16
       CoeffAutoScale: true
               Signed: true

      InputWordLength: 16
      InputFracLength: 15

      FilterInternals: 'FullPrecision'

hd.filterinternals='specifyPrecision'

hd =

      FilterStructure: 'Direct-Form Symmetric FIR'
           Arithmetic: 'fixed'
            Numerator: [-0.0080 0.0600 0.4400 0.4400 0.0600 -0.0080]
     PersistentMemory: false

      CoeffWordLength: 16
       CoeffAutoScale: true
               Signed: true

      InputWordLength: 16
      InputFracLength: 15

      FilterInternals: 'SpecifyPrecision'

     OutputWordLength: 36
     OutputFracLength: 31

    ProductWordLength: 33
    ProductFracLength: 31

      AccumWordLength: 36
      AccumFracLength: 31

            RoundMode: 'convergent'
         OverflowMode: 'wrap'
```

To use `hd` for fixed-point filtering, change the value of the property `Arithmetic` to `fixed` with the following command:

```
hd.arithmetic = 'fixed'
```

Even Order

Specify a fourth-order, fixed-point, direct-form symmetric FIR filter structure for a `dfilt` object `hd`, with the following code:

```
b = [-0.01 0.1 0.8 0.1 -0.01];
hd = dfilt.dfsymfir(b)

hd =

     FilterStructure: 'Direct-Form Symmetric FIR'
          Arithmetic: 'double'
           Numerator: [-0.0100 0.1000 0.8000 0.1000 -0.0100]
    PersistentMemory: false

set(hd,'arithmetic','fixed')
hd

hd =

     FilterStructure: 'Direct-Form Symmetric FIR'
          Arithmetic: 'fixed'
           Numerator: [-0.0100 0.1000 0.8000 0.1000 -0.0100]
    PersistentMemory: false

     CoeffWordLength: 16
      CoeffAutoScale: true
              Signed: true

     InputWordLength: 16
     InputFracLength: 15

     FilterInternals: 'FullPrecision'

hd.filterinternals='specifyPrecision'
```

```
hd =

         FilterStructure: 'Direct-Form Symmetric FIR'
              Arithmetic: 'fixed'
               Numerator: [-0.0100 0.1000 0.8000 0.1000 -0.0100]
        PersistentMemory: false

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

         FilterInternals: 'SpecifyPrecision'

        OutputWordLength: 36
        OutputFracLength: 30

       ProductWordLength: 33
       ProductFracLength: 30

         AccumWordLength: 36
         AccumFracLength: 30

               RoundMode: 'convergent'
            OverflowMode: 'wrap'
```

See Also

dfilt, dfilt.dfasymfir, dfilt.dffir, dfilt.dffirt

Purpose

Construct a discrete-time, lattice allpass filter object that uses fixed-point or single-precision floating-point arithmetic

Syntax

Refer to `dfilt.latticeallpass` in the Signal Processing Toolbox.

Description

`hd = dfilt.latticeallpass(k)` returns a discrete-time, lattice allpass filter object `hd`, with lattice coefficients, `k`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

- To change to fixed-point filtering, enter

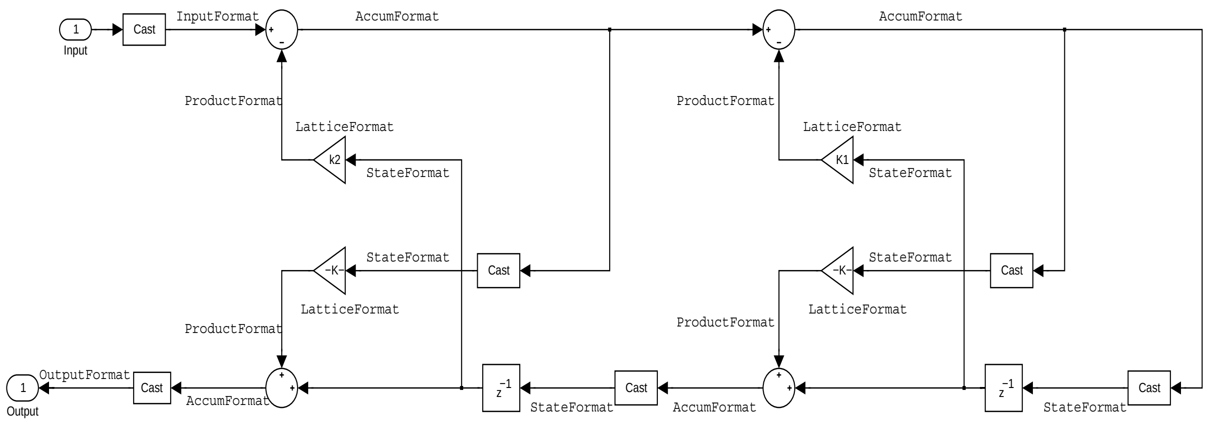
```
set(hd,'arithmetic','fixed');
```

For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.latticeallpass` returns a default, discrete-time, lattice allpass filter object `hd`, with `k=[ ]`. This filter passes the input through to the output unchanged.

Fixed-Point Filter Structure

The figure below shows the signal flow for the allpass lattice filter implemented by `dfilt.latticeallpass`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
AccumFormat	`AccumWordLength`	`AccumFracLength`	`AccumMode`
InputFormat	`InputWordLength`	`InputFracLength`	

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
LatticeFormat	CoeffWordLength	LatticeFracLength	CoeffAutoScale
OutputFormat	OutputWordLength	OutputFracLength	OutputMode
ProductFormat	ProductWordLength	ProductFracLength	ProductMode
StateFormat	StateWordLength	StateFracLength	States

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label ProductFormat, which always follows a coefficient multiplication element in the signal flow. The label indicates that coefficients leave the multiplication element with the word length and fraction length associated with product operations that include coefficients. From reviewing the table, you see that the ProductFormat refers to the properties `ProductFracLength`, `ProductWordLength`, and `ProductMode` that fully define the coefficient format after multiply (or product) operations.

Properties

In this table you see the properties associated with the allpass lattice implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to "Fixed-Point Filter Properties" on page 8-3.

Property Name	Brief Description
`AccumFracLength`	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters and lattice filters. IIR filters have two similar properties—`DenAccumFracLength` and `NumAccumFracLength`—that let you set the precision for numerator and denominator operations separately.
`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.
`Arithmetic`	Defines the arithmetic the filter uses. Gives you the options `double`, `single`, and `fixed`. In short, this property defines the operating mode for your filter.
`CastBeforeSum`	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.

Property Name	Brief Description
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the LatticeFracLength property value to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
Lattice	Any lattice structure coefficients. No default value.
LatticeFracLength	Sets the fraction length applied to the lattice coefficients.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.

Property Name	Brief Description
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: - AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. - BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. - SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
ProductFracLength	For the output from a product operation, this sets the fraction length used to interpret the data. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
`RoundMode`	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • `convergent`—Round up to the next allowable quantized value. • `ceil`—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to `1`. • `fix`—Round negative numbers up and positive numbers down to the next allowable quantized value. • `floor`—Round down to the next allowable quantized value. • `round`—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
`Signed`	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.

Property Name	Brief Description
StateFracLength	When you set StateAutoScale to false, you enable the StateFracLength property that lets you set the fraction length applied to interpret the filter states.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to filtstates in your Signal Processing Toolbox documentation or in the Help system.
StateWordLength	Sets the word length used to represent the filter states.

Examples

Specify a third-order lattice allpass filter structure for a dfilt object hd, with the following code:

```
k = [.66 .7 .44];
hd=dfilt.latticeallpass(k);
```

Now convert hd to fixed-point arithmetic form.

```
hd.arithmetic='fixed'

hd =

         FilterStructure: 'Lattice Allpass'
              Arithmetic: 'fixed'
                 Lattice: [0.6600 0.7000 0.4400]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
```

```
                Signed: true

       InputWordLength: 16
       InputFracLength: 15

      OutputWordLength: 16
            OutputMode: 'AvoidOverflow'

       StateWordLength: 16
       StateFracLength: 15

           ProductMode: 'FullPrecision'

             AccumMode: 'KeepMSB'
       AccumWordLength: 40
         CastBeforeSum: true

             RoundMode: 'convergent'
          OverflowMode: 'wrap'
```

See Also

`dfilt`, `dfilt.latticear`, `dfilt.latticearma`, `dfilt.latticemamax`, `dfilt.latticemamin`

Purpose

Construct a discrete-time, lattice, autoregressive filter object that uses fixed-point or single-precision floating-point arithmetic

Syntax

Refer to `dfilt.latticear` in the Signal Processing Toolbox.

Description

`hd = dfilt.latticear(k)` returns a discrete-time, lattice autoregressive filter object `hd`, with lattice coefficients, `k`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

  ```
  set(hd,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

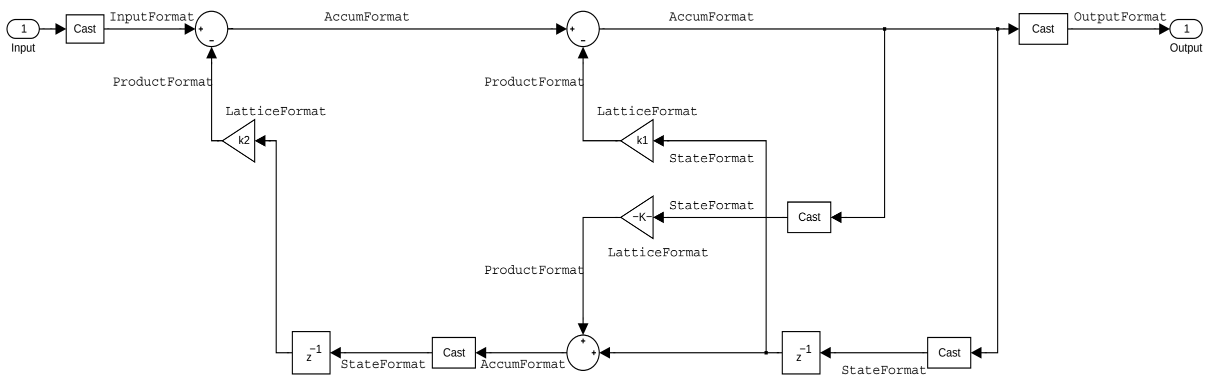
  ```
  set(hd,'arithmetic','fixed');
  ```

For more information about the property `Arithmetic`, refer to “Arithmetic” on page 8-22.

`hd = dfilt.latticear` returns a default, discrete-time, lattice autoregressive filter object `hd`, with `k=[ ]`. This filter passes the input through to the output unchanged.

Fixed-Point Filter Structure

The figure below shows the signal flow for the autoregressive lattice filter implemented by `dfilt.latticear`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
AccumFormat	`AccumWordLength`	`AccumFracLength`	`AccumMode`
InputFormat	`InputWordLength`	`InputFracLength`	

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
LatticeFormat	CoeffWordLength	LatticeFracLength	CoeffAutoScale
OutputFormat	OutputWordLength	OutputFracLength	OutputMode
ProductFormat	ProductWordLength	ProductFracLength	ProductMode
StateFormat	StateWordLength	StateFracLength	States

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label ProductFormat, which always follows a coefficient multiplication element in the signal flow. The label indicates that coefficients leave the multiplication element with the word length and fraction length associated with product operations that include coefficients. From reviewing the table, you see that the ProductFormat refers to the properties `ProductFracLength`, `ProductWordLength`, and `ProductMode` that fully define the coefficient format after multiply (or product) operations.

Properties

In this table you see the properties associated with the autoregressive lattice implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any dfilt object, refer to "Fixed-Point Filter Properties" on page 8-3.

Property Name	Brief Description
AccumFracLength	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters and lattice filters. IIR filters have two similar properties—DenAccumFracLength and NumAccumFracLength—that let you set the precision for numerator and denominator operations separately.
AccumMode	Determines how the accumulator outputs stored values. Choose from full precision (FullPrecision), or whether to keep the most significant bits (KeepMSB) or least significant bits (KeepLSB) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set AccumMode to SpecifyPrecision.
AccumWordLength	Sets the word length used to store data in the accumulator/buffer.
Arithmetic	Defines the arithmetic the filter uses. Gives you the options double, single, and fixed. In short, this property defines the operating mode for your filter.
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.

Property Name	Brief Description
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the LatticeFracLength to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
Lattice	Any lattice structure coefficients.
LatticeFracLength	Sets the fraction length applied to the lattice coefficients.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.

Property Name	Brief Description
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: ▪ AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. ▪ BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. ▪ SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
ProductFracLength	For the output from a product operation, this sets the fraction length used to interpret the data. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.

Property Name	Brief Description
StateFracLength	When you set StateAutoScale to false, you enable the StateFracLength property that lets you set the fraction length applied to interpret the filter states.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to filtstates in your Signal Processing Toolbox documentation or in the Help system.
StateWordLength	Sets the word length used to represent the filter states.

Examples

Specify a third-order lattice autoregressive filter structure for a dfilt object, hd, with the following code that creates a fixed-point filter.

```
k = [.66 .7 .44];
hd1=dfilt.latticear(k)

hd1 =

         FilterStructure: 'Lattice Autoregressive (AR)'
              Arithmetic: 'double'
                 Lattice: [0.6600 0.7000 0.4400]
        PersistentMemory: false
                  States: [3x1 double]
     NumSamplesProcessed: 0

hd1.arithmetic='fixed'

hd1 =

         FilterStructure: 'Lattice Autoregressive (AR)'
```

```
              Arithmetic: 'fixed'
                 Lattice: [0.6600 0.7000 0.4400]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: true
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

         StateWordLength: 16
         StateFracLength: 15

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'

specifyall(hd1)
hd1

hd1 =

         FilterStructure: 'Lattice Autoregressive (AR)'
              Arithmetic: 'fixed'
                 Lattice: [0.6600 0.7000 0.4400]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0
```

```
     CoeffWordLength: 16
      CoeffAutoScale: false
   LatticeFracLength: 15
              Signed: true

     InputWordLength: 16
     InputFracLength: 15

    OutputWordLength: 16
          OutputMode: 'SpecifyPrecision'
    OutputFracLength: 12

     StateWordLength: 16
     StateFracLength: 15

         ProductMode: 'SpecifyPrecision'
   ProductWordLength: 32
   ProductFracLength: 30

           AccumMode: 'SpecifyPrecision'
     AccumWordLength: 40
     AccumFracLength: 30
       CastBeforeSum: true

           RoundMode: 'convergent'
        OverflowMode: 'wrap'
```

See Also

`dfilt`, `dfilt.latticeallpass`, `dfilt.latticearma`, `dfilt.latticemamax`, `dfilt.latticemamin`

Purpose Construct a discrete-time, lattice, autoregressive, moving-average filter object that uses fixed-point or single-precision floating-point arithmetic

Syntax Refer to `dfilt.latticearma` in the Signal Processing Toolbox.

Description `hd = dfilt.latticearma(k)` returns a discrete-time, lattice moving-average autoregressive filter object `hd`, with lattice coefficients, `k`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

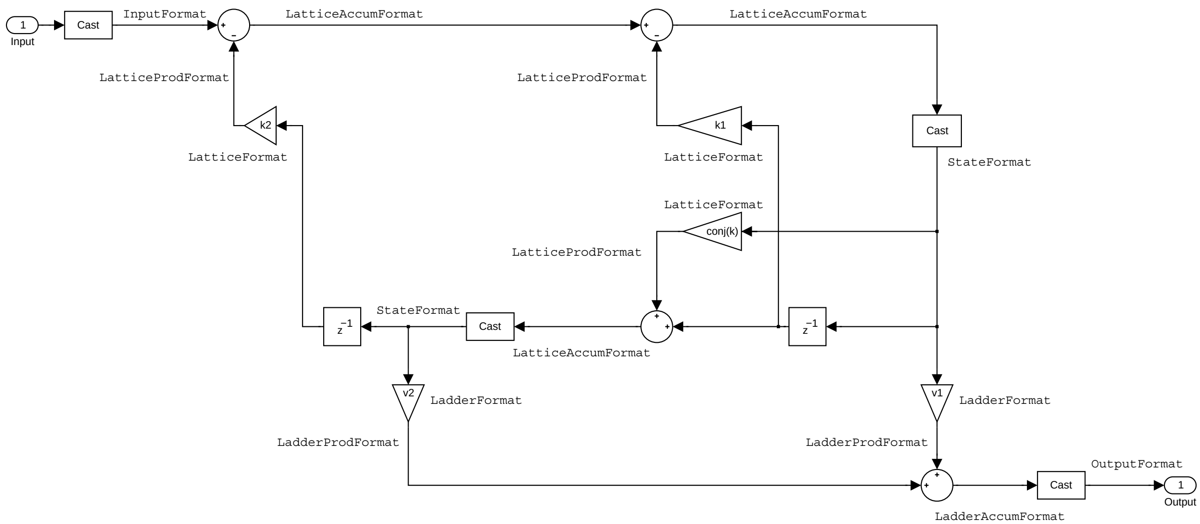
- To change to fixed-point filtering, enter

```
set(hd,'arithmetic','fixed');
```

For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`hd = dfilt.latticearma` returns a default, discrete-time, lattice moving-average, autoregressive filter object `hd`, with `k=[]`. This filter passes the input through to the output unchanged.

Fixed-Point Filter Structure The figure below shows the signal flow for the autoregressive lattice filter implemented by `dfilt.latticearma`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to

the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
InputFormat	`InputWordLength`	`InputFracLength`	
LadderAccumFormat	`AccumWordLength`	`LadderAccumFracLength`	`AccumMode`
LadderFormat	`CoeffWordLength`	`LadderFracLength`	`CoeffAutoScale`
LadderProdFormat	`ProductWordLength`	`LadderProdFracLength`	`ProductMode`
LatticeAccumFormat	`AccumWordLength`	`LatticeAccumFracLength`	`AccumMode`
LatticeFormat	`CoeffWordLength`	`LatticeFracLength`	`CoeffAutoScale`
LatticeProdFormat	`ProductWordLength`	`LatticeProdFracLength`	`ProductMode`
OutputFormat	`OutputWordLength`	`OutputFracLength`	`OutputMode`
StateFormat	`StateWordLength`	`StateFracLength`	`States`

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label LatticeProdFormat, which always follows a coefficient multiplication element in the signal flow. The label indicates that lattice coefficients leave the multiplication element with the word length and fraction length associated with product operations that include coefficients. From reviewing the table, you see that the LatticeProdFormat refers to the properties `ProductWordLength`, `LatticeProdFracLength`, and `ProductMode` that fully define the coefficient format after multiply (or product) operations.

Properties

In this table you see the properties associated with the autoregressive moving-average lattice implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where hd is a filter.

For further information about the properties of this filter or any dfilt object, refer to "Fixed-Point Filter Properties" on page 8-3.

Property Name	Brief Description
AccumFracLength	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters and lattice filters. IIR filters have two similar properties—DenAccumFracLength and NumAccumFracLength—that let you set the precision for numerator and denominator operations separately.
AccumMode	Determines how the accumulator outputs stored values. Choose from full precision (FullPrecision), or whether to keep the most significant bits (KeepMSB) or least significant bits (KeepLSB) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set AccumMode to SpecifyPrecision.
AccumWordLength	Sets the word length used to store data in the accumulator/buffer.

Property Name	Brief Description
Arithmetic	Defines the arithmetic the filter uses. Gives you the options double, single, and fixed. In short, this property defines the operating mode for your filter.
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the LatticeFracLength property to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
Ladder	Stores the ladder coefficients for lattice ARMA (dfilt.latticearma) filters.

Property Name	Brief Description
LadderAccumFracLength	Sets the fraction length used to interpret the output from sum operations that include the ladder coefficients. You can change this property value after you set AccumMode to SpecifyPrecision.
LadderFracLength	Determines the precision used to represent the ladder coefficients in ARMA lattice filters.
Lattice	Stores the lattice structure coefficients.
LatticeFracLength	Sets the fraction length applied to the lattice coefficients.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.

Property Name	Brief Description
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: ▪ AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. ▪ BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. ▪ SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
ProductFracLength	For the output from a product operation, this sets the fraction length used to interpret the data. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.

Property Name	Brief Description
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.

Property Name	Brief Description
StateFracLength	When you set StateAutoScale to false, you enable the StateFracLength property that lets you set the fraction length applied to interpret the filter states.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to filtstates in your Signal Processing Toolbox documentation or in the Help system.
StateWordLength	Sets the word length used to represent the filter states.

See Also

dfilt, dfilt.latticeallpass, dfilt.latticear, dfilt.latticemamin, dfilt.latticemamin

dfilt.latticemamax

Purpose

Construct a discrete-time, lattice, moving-average filter object with maximum phase that uses fixed-point or single-precision floating-point arithmetic

Syntax

Refer to `dfilt.latticemamax` in the Signal Processing Toolbox.

Description

`hd = dfilt.latticemamax(k)` returns a discrete-time, lattice, moving-average filter object `hd`, with lattice coefficients `k`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

  ```
  set(hd,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hd,'arithmetic','fixed');
  ```

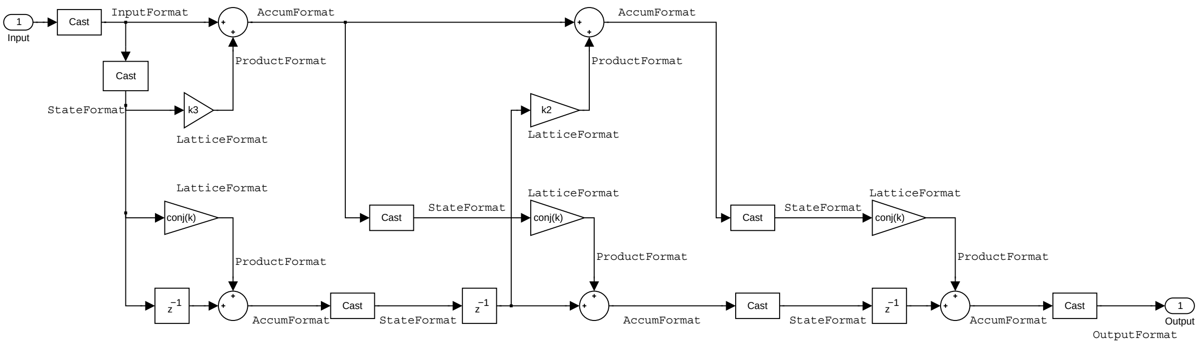
For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

Note When the `k` coefficients define a maximum phase filter, the resulting filter in this structure is maximum phase. When your coefficients do not define a maximum phase filter, placing them in this structure does not produce a maximum phase filter.

`hd = dfilt.latticemamax` returns a default discrete-time, lattice, moving-average filter object `hd`, with `k=[]`. This filter passes the input through to the output unchanged.

Fixed-Point Filter Structure

The figure below shows the signal flow for the maximum phase implementation of a moving-average lattice filter implemented by `dfilt.latticemamax`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word "format." In this use, "format" means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
AccumFormat	`AccumWordLength`	`AccumFracLength`	`AccumMode`
InputFormat	`InputWordLength`	`InputFracLength`	
LatticeFormat			

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
OutputFormat	`OutputWordLength`	`OutputFracLength`	`OutputMode`
ProductFormat	`ProductWordLength`	`ProductFracLength`	`ProductMode`
StateFormat	`StateWordLength`	`StateFracLength`	`States`

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label ProductFormat, which always follows a coefficient multiplication element in the signal flow. The label indicates that coefficients leave the multiplication element with the word length and fraction length associated with product operations that include coefficients. From reviewing the table, you see that the ProductFormat refers to the properties `ProductFracLength`, `ProductWordLength`, and `ProductMode` that fully define the coefficient format after multiply (or product) operations.

Properties

In this table you see the properties associated with the maximum phase, moving average lattice implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any dfilt object, refer to “Fixed-Point Filter Properties” on page 8-3.

Property Name	Brief Description
AccumFracLength	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters and lattice filters. IIR filters have two similar properties—DenAccumFracLength and NumAccumFracLength—that let you set the precision for numerator and denominator operations separately.
AccumMode	Determines how the accumulator outputs stored values. Choose from full precision (FullPrecision), or whether to keep the most significant bits (KeepMSB) or least significant bits (KeepLSB) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set AccumMode to SpecifyPrecision.
AccumWordLength	Sets the word length used to store data in the accumulator/buffer.
Arithmetic	Defines the arithmetic the filter uses. Gives you the options double, single, and fixed. In short, this property defines the operating mode for your filter.
CastBeforeSum	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.

Property Name	Brief Description
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the LatticeFracLength property to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
Lattice	Any lattice structure coefficients.
LatticeFracLength	Sets the fraction length applied to the lattice coefficients.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.

Property Name	Brief Description
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: - AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. - BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. - SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
ProductFracLength	For the output from a product operation, this sets the fraction length used to interpret the data. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.

Property Name	Brief Description
StateFracLength	When you set StateAutoScale to false, you enable the StateFracLength property that lets you set the fraction length applied to interpret the filter states.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to filtstates in your Signal Processing Toolbox documentation or in the Help system.
StateWordLength	Sets the word length used to represent the filter states.

Examples

Specify a fourth-order lattice, moving-average, maximum phase filter structure for a dfilt object, hd, with the following code:

```
k = [.66 .7 .44 .33];
hd = dfilt.latticemamax(k)

hd =
             FilterStructure: 'Lattice maximum phase'
                     Lattice: [1x4 double]
    NumberOfSamplesProcessed: 0
                 ResetStates: 'on'
                      States: [4x1 double]
```

See Also

dfilt, dfilt.latticeallpass, dfilt.latticear, dfilt.latticearma, dfilt.latticemamin

Purpose

Construct a fixed-point or single-precision floating-point discrete-time, lattice, moving-average filter object with minimum

Syntax

Refer to `dfilt.latticemamin` in the Signal Processing Toolbox.

Description

`hd = dfilt.latticemamin(k)` returns a discrete-time, lattice, moving-average, minimum phase, filter object `hd`, with lattice coefficients `k`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

  ```
  set(hd,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hd,'arithmetic','fixed');
  ```

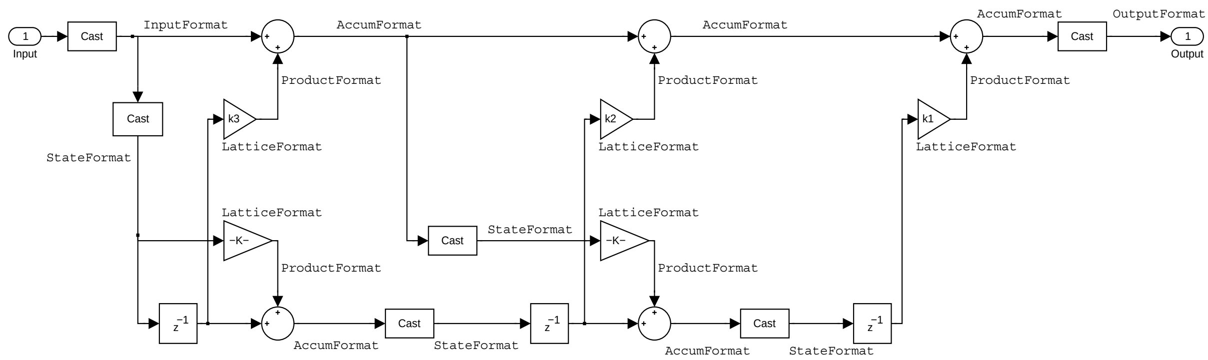
For more information about the property `Arithmetic`, refer to “Arithmetic” on page 8-22.

Note When the `k` coefficients define a minimum phase filter, the resulting filter in this structure is minimum phase. When your coefficients do not define a minimum phase filter, placing them in this structure does not produce a minimum phase filter.

`hd = dfilt.latticemamin` returns a default discrete-time, lattice, moving-average, minimum phase, filter object `hd`, with `k=[]`. This filter passes the input through to the output unchanged.

Fixed-Point Filter Structure

The figure below shows the signal flow for the minimum phase implementation of a moving-average lattice filter implemented by `dfilt.latticemamin`. To help you see how the filter processes the coefficients, input, and states of the filter, as well as numerical operations, the figure includes the locations of the formatting objects within the signal flow.



Notes About the Signal Flow Diagram

To help you understand where and how the filter performs fixed-point arithmetic during filtering, the figure shows various labels associated with data flow and functional elements in the filter. The following table describes each label in the signal flow and relates the label to the filter properties that are associated with it.

You see that the labels use a common format—a prefix followed by the word “format.” In this use, “format” means the word length and fraction length associated with the filter part referred to by the prefix.

For example, the InputFormat label refers to the word length and fraction length used to interpret the data input to the filter. The format properties `InputWordLength` and `InputFracLength` (as shown in the table) store the word length and the fraction length in bits. Or consider NumFormat, which refers to the word and fraction lengths (`CoeffWordLength`, `NumFracLength`) associated with representing filter numerator coefficients.

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
AccumFormat	`AccumWordLength`	`AccumFracLength`	`AccumMode`
InputFormat	`InputWordLength`	`InputFracLength`	
LatticeFormat			

Signal Flow Label	Word Length Property	Fraction Length Property	Related Properties
OutputFormat	OutputWordLength	OutputFracLength	OutputMode
ProductFormat	ProductWordLength	ProductFracLength	ProductMode
StateFormat	StateWordLength	StateFracLength	States

Most important is the label position in the diagram, which identifies where the format applies.

As one example, look at the label ProductFormat, which always follows a coefficient multiplication element in the signal flow. The label indicates that coefficients leave the multiplication element with the word length and fraction length associated with product operations that include coefficients. From reviewing the table, you see that the ProductFormat refers to the properties `ProductFracLength`, `ProductWordLength`, and `ProductMode` that fully define the coefficient format after multiply (or product) operations.

Properties

In this table you see the properties associated with the minimum phase, moving average lattice implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to “Fixed-Point Filter Properties” on page 8-3.

Property Name	Brief Description
`AccumFracLength`	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters and lattice filters. IIR filters have two similar properties—`DenAccumFracLength` and `NumAccumFracLength`—that let you set the precision for numerator and denominator operations separately.
`AccumMode`	Determines how the accumulator outputs stored values. Choose from full precision (`FullPrecision`), or whether to keep the most significant bits (`KeepMSB`) or least significant bits (`KeepLSB`) when output results need shorter word length than the accumulator supports. To let you set the word length and the precision (the fraction length) used by the output from the accumulator, set `AccumMode` to `SpecifyPrecision`.
`AccumWordLength`	Sets the word length used to store data in the accumulator/buffer.
`Arithmetic`	Defines the arithmetic the filter uses. Gives you the options `double`, `single`, and `fixed`. In short, this property defines the operating mode for your filter.
`CastBeforeSum`	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.

Property Name	Brief Description
CoeffAutoScale	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the LatticeFracLength property to specify the precision used.
CoeffWordLength	Specifies the word length to apply to filter coefficients.
FilterStructure	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
Lattice	Any lattice structure coefficients.
LatticeFracLength	Sets the fraction length applied to the lattice coefficients.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.

Property Name	Brief Description
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: ▪ AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. ▪ BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. ▪ SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.
OutputWordLength	Determines the word length used for the output data.
OverflowMode	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Property Name	Brief Description
ProductFracLength	For the output from a product operation, this sets the fraction length used to interpret the data. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
ProductMode	Determines how the filter handles the output of product operations. Choose from full precision (FullPrecision), or whether to keep the most significant bit (KeepMSB) or least significant bit (KeepLSB) in the result when you need to shorten the data words. For you to be able to set the precision (the fraction length) used by the output from the multiplies, you set ProductMode to SpecifyPrecision.
ProductWordLength	Specifies the word length to use for multiplication operation results. This property becomes writable (you can change the value) when you set ProductMode to SpecifyPrecision.
PersistentMemory	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. on is the default setting.

<table>
<tr><th>Property Name</th><th>Brief Description</th></tr>
<tr><td><code>RoundMode</code></td><td>Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths).

• <code>convergent</code>—Round up to the next allowable quantized value.
• <code>ceil</code>—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to <code>1</code>.
• <code>fix</code>—Round negative numbers up and positive numbers down to the next allowable quantized value.
• <code>floor</code>—Round down to the next allowable quantized value.
• <code>round</code>—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up.

The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.</td></tr>
<tr><td><code>Signed</code></td><td>Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.</td></tr>
</table>

Property Name	Brief Description
StateFracLength	When you set StateAutoScale to false, you enable the StateFracLength property that lets you set the fraction length applied to interpret the filter states.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to filtstates in your Signal Processing Toolbox documentation or in the Help system.
StateWordLength	Sets the word length used to represent the filter states.

Examples

Specify a third-order lattice, moving-average, minimum phase, filter structure for a dfilt object, hd, with the following code:

```
k = [.66 .7 .44];
hd = dfilt.latticemamin(k)

hd =

       FilterStructure: 'Lattice Moving-Average (MA) For Minimum
Phase'
              Arithmetic: 'double'
                 Lattice: [0.6600 0.7000 0.4400]
        PersistentMemory: false
                  States: [3x1 double]
     NumSamplesProcessed: 0

set(hd,'arithmetic','fixed')
specifyall(hd)
hd
```

```
hd =

       FilterStructure: 'Lattice Moving-Average (MA) For Minimum
Phase'
            Arithmetic: 'fixed'
               Lattice: [0.6600 0.7000 0.4400]
      PersistentMemory: false
                States: [1x1 embedded.fi]
   NumSamplesProcessed: 0

       CoeffWordLength: 16
        CoeffAutoScale: false
     LatticeFracLength: 15
                Signed: true

       InputWordLength: 16
       InputFracLength: 15

      OutputWordLength: 16
            OutputMode: 'SpecifyPrecision'
      OutputFracLength: 12

       StateWordLength: 16
       StateFracLength: 15

           ProductMode: 'SpecifyPrecision'
     ProductWordLength: 32
     ProductFracLength: 30

             AccumMode: 'SpecifyPrecision'
       AccumWordLength: 40
       AccumFracLength: 30
         CastBeforeSum: true

             RoundMode: 'convergent'
          OverflowMode: 'wrap'
```

See Also

dfilt, dfilt.latticeallpass, dfilt.latticear, dfilt.latticearma, dfilt.latticemamax

Purpose Construct a discrete-time, parallel structure filter object

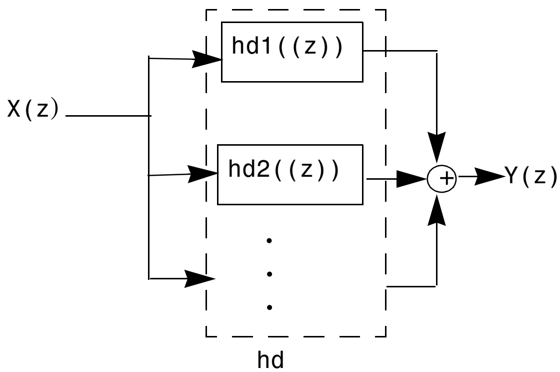
Syntax Refer to `dfilt.parallel` in the Signal Processing Toolbox.

Description `hd = dfilt.parallel(hd1,hd2,...)` returns a discrete-time filter object `hd`, which is a structure of two or more `dfilt` filter objects, `hd1`, `hd2`, and so on arranged in parallel.

You can also use the standard notation to combine filters into a parallel structure.

```
parallel(hd1,hd2,...)
```

In this syntax, `hd1`, `hd2`, and so on can be a mix of `dfilt` objects, `mfilt` objects, and `adaptfilt` objects.



`hd1`, `hd2`, and so on can be fixed-point filters. All filters in the parallel structure must be the same arithmetic format—double, single, or fixed. `hd`, the filter returned, inherits the format of the individual filters.

See Also `dfilt`, `dfilt.cascade`

`dfilt.cascade`, `dfilt.parallel` in your Signal Processing Toolbox documentation

Purpose

Construct a discrete-time, scalar filter object that uses fixed-point or single-precision floating-point arithmetic

Syntax

Refer to `dfilt.scalar` in the Signal Processing Toolbox.

Description

`dfilt.scalar(g)` returns a discrete-time, scalar filter object with gain `g`, where `g` is a scalar.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hd` as follows:

- To change to single-precision filtering, enter

```
set(hd,'arithmetic','single');
```

- To change to fixed-point filtering, enter

```
set(hd,'arithmetic','fixed');
```

For more information about the property `Arithmetic`, refer to "Arithmetic" on page 8-22.

`dfilt.scalar` returns a default, discrete-time scalar gain filter object `hd`, with gain `1`.

Properties

In this table you see the properties associated with the scalar implementation of `dfilt` objects.

Note The table lists all the properties that a filter can have. Many of the properties are dynamic, meaning they exist only in response to the settings of other properties. You might not see all of the listed properties all the time.

To view all the properties for a filter at any time, use

```
get(hd)
```

where `hd` is a filter.

For further information about the properties of this filter or any `dfilt` object, refer to "Fixed-Point Filter Properties" on page 8-3.

Property Name	Brief Description
`Arithmetic`	Defines the arithmetic the filter uses. Gives you the options `double`, `single`, and `fixed`. In short, this property defines the operating mode for your filter.
`CastBeforeSum`	Specifies whether to cast numeric data to the appropriate accumulator format (as shown in the signal flow diagrams) before performing sum operations.
`CoeffAutoScale`	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to `false` enables you to change the `CoeffFracLength` property to specify the precision used.
`CoeffFracLength`	Set the fraction length the filter uses to interpret coefficients. `CoeffFracLength` is always available, but it is read-only until you set `CoeffAutoScale` to `false`.
`CoeffWordLength`	Specifies the word length to apply to filter coefficients.
`FilterStructure`	Describes the signal flow for the filter object, including all of the active elements that perform operations during filtering—gains, delays, sums, products, and input/output.
`Gain`	Returns the gain for the scalar filter. Scalar filters do not alter the input data except by adding gain.

Property Name	Brief Description
InputFracLength	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Specifies the word length applied to interpret input data.
NumSamplesProcessed	Reports the number of samples actually processed by the filter in a filtering operation. When PersistentMemory is false, this property reports the number of samples filtered for each input data set. Setting PersistentMemory to true causes this property to report the total number of samples processed for all data sets, not each one.
OutputFracLength	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set OutputMode to SpecifyPrecision.
OutputMode	Sets the mode the filter uses to scale the filtered data for output. You have the following choices: ▪ AvoidOverflow—directs the filter to set the output data word length and fraction length to avoid causing the data to overflow. ▪ BestPrecision—directs the filter to set the output data word length and fraction length to maximize the precision in the output data. ▪ SpecifyPrecision—lets you set the word and fraction lengths used by the output data from filtering.

Property Name	Brief Description
`OutputWordLength`	Determines the word length used for the output data.
`OverflowMode`	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either `saturate` (limit the output to the largest positive or negative representable value) or `wrap` (set overflowing values to the nearest representable value using modular arithmetic). The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
`PersistentMemory`	Specifies whether to reset the filter states and memory before each filtering operation. Lets you decide whether your filter retains states from previous filtering runs. `on` is the default setting.

Property Name	Brief Description
RoundMode	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

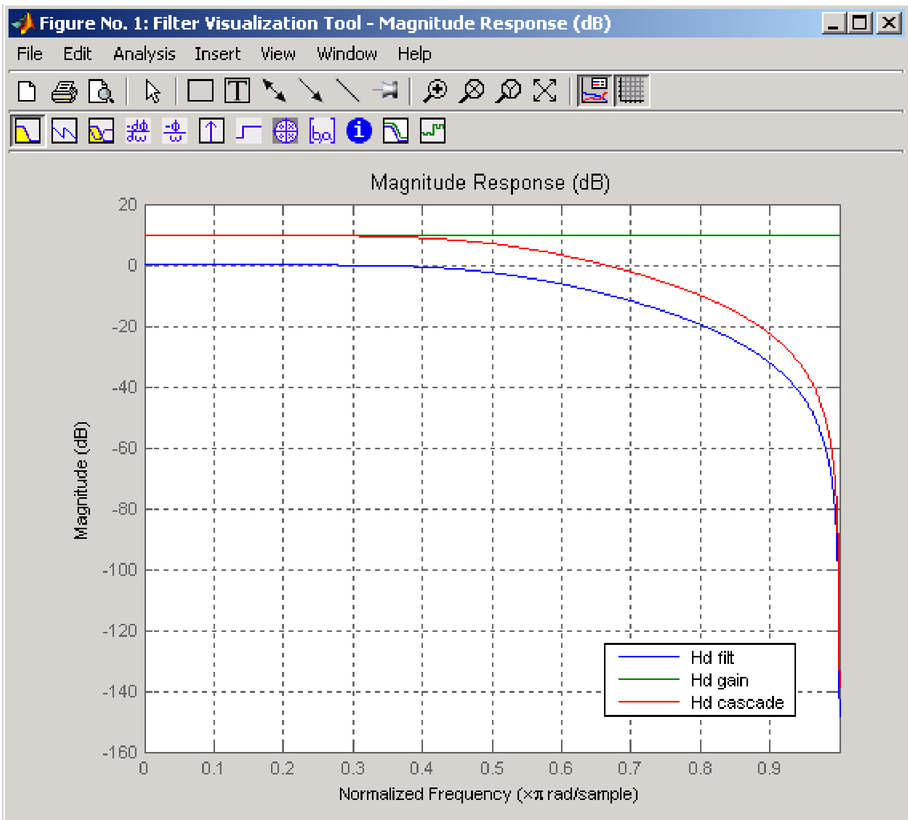
Property Name	Brief Description
Signed	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
States	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to filtstates in your Signal Processing Toolbox documentation or in the Help system.

Example

Create a direct-form I filter object hd_filt and a scalar object with a gain of 3 hd_gain and cascade them together.

```
b = [0.3 0.6 0.3];
a = [1 0 0.2];
hd_filt = dfilt.df1(b,a)
hd_gain = dfilt.scalar(3)
hd=cascade(hd_gain,hd_filt)
fvtool(hd_filt,hd_gain,hd)

hd_filt =
         FilterStructure: 'direct-form I'
              Arithmetic: 'double'
               Numerator: [0.3000 0.6000 0.3000]
             Denominator: [1 0 0.2000]
        PersistentMemory: false
                  States: [4x1 double]
     NumSamplesProcessed: 0

hd_gain =
         FilterStructure: 'Scalar'
              Arithmetic: 'double'
                    Gain: 3
        PersistentMemory: false
```

```
                   States: []
      NumSamplesProcessed: 0

hd =
           FilterStructure: Cascade
                Section(1): Scalar
                Section(2): Direct Form I
          PersistentMemory: false
       NumSamplesProcessed: 0
```



To view the sections of the cascaded filter, use

```
hd.section(1)
```

```
ans =
         FilterStructure: 'Scalar'
              Arithmetic: 'double'
                    Gain: 3
        PersistentMemory: false
                  States: []
     NumSamplesProcessed: 0
```

and

```
hd.section(2)

ans =
         FilterStructure: 'Direct Form I'
              Arithmetic: 'double'
               Numerator: [0.3000 0.6000 0.3000]
             Denominator: [1 0 0.2000]
        PersistentMemory: false
                  States: [4x1 double]
     NumSamplesProcessed: 0
```

See Also

`dfilt`, `dfilt.cascade`

Purpose Display a filter object with properties and values

Syntax

```
disp(q)
disp(f)
disp(hq)
```

Description Similar to omitting the closing semicolon from an expression on the command line, except that `disp` does not display the variable name. `disp` lists the property names and property values for any filter object, such as a `dfilt` object or `adaptfilt` object.

The following examples illustrate the default display for an adaptive filter `ha` and a multirate filter `hm`.

```
ha=adaptfilt.rls

ha =

              Algorithm: 'Direct Form FIR RLS Adaptive Filter'
           FilterLength: 10
           Coefficients: [0 0 0 0 0 0 0 0 0 0]
                 States: [9x1 double]
       ForgettingFactor: 1
             KalmanGain: []
                 InvCov: [10x10 double]
       PersistentMemory: false
    NumSamplesProcessed: 0

disp(ha)
              Algorithm: 'Direct-Form FIR RLS Adaptive Filter'
           FilterLength: 10
           Coefficients: [0 0 0 0 0 0 0 0 0 0]
                 States: [9x1 double]
       ForgettingFactor: 1
             KalmanGain: []
                 InvCov: [10x10 double]
       PersistentMemory: false
    NumSamplesProcessed: 0
```

```
hm=mfilt.cicdecim(6)

hm =

          FilterStructure: 'Cascaded Integrator-Comb Decimator'
        DifferentialDelay: 1
         NumberOfSections: 2
            InputBitWidth: 16
           OutputBitWidth: 16
             BitsPerStage: 16
         DecimationFactor: 6
         PersistentMemory: false
                   States: [2x2 int32]
     NumSamplesProcessed: 0

disp(hm)
          FilterStructure: 'Cascaded Integrator-Comb Decimator'
        DifferentialDelay: 1
         NumberOfSections: 2
            InputBitWidth: 16
           OutputBitWidth: 16
             BitsPerStage: 16
         DecimationFactor: 6
         PersistentMemory: false
                   States: [2x2 int32]
     NumSamplesProcessed: 0
```

See Also

`set`

double

Purpose Cast a fixed-point filter to a version that uses double-precision arithmetic

Syntax

```
hd = double(h)
```

Description

`hd = double(h)` returns a new filter `hd` that has the same structure and coefficients as `h`, but whose arithmetic property is set to `double` to use double-precision arithmetic for filtering. `double(h)` is not the same as the `reffilter(h)` function:

- `hd`, the filter returned by `double` has the quantized coefficients of `h` represented in double-precision floating-point format
- The reference filter returned by `reffilter` has double-precision, floating-point coefficients that have not been quantized.

You might find `double(h)` useful to isolate the effects of quantizing the coefficients of a filter by using `double` to create a filter `hd` that operates in double-precision but uses the quantized filter coefficients.

Examples

Use the same filter, once with fixed-point arithmetic and once with floating-point, to compare fixed-point filtering with double-precision floating-point filtering.

```
h = dfilt.dffir(firgr(27,[0 .4 .6 1],
[1 1 0 0]));              % Lowpass filter.
h.arithmetic = 'fixed';   % Set h to use fixed-point arithmetic
                          % to filter. Quantize the coeffs.
hd = double(h);           % Cast h to double-precision
                          % floating-point coefficients.
n = 0:99; x = sin(0.7*pi*n(:)); % Set up an input signal.
y = filter(h,x);                % Fixed-point output.
yd = filter(hd,x);              % Floating-point output.
norm(yd-double(y),inf)
ans =

  9.2014e-004
```

`norm` shows that the largest difference (maximum error) between the output values from the fixed versus floating filtering comparison is about 0.0009—either good or less good depending on your application.

See Also reffilter

Purpose

Design elliptical or Cauer digital filters using the specifications in a filter design object

Syntax

```
hd  = ellip(d)
hd = ellip(d,'matchexactly',match)
```

Description

`hd = ellip(d)` designs a Butterworth IIR digital filter using the specifications supplied in the object `h`.

`hd = ellip(d,'matchexactly',match)` returns an elliptical IIR filter where the filter response matches the specified response exactly for one filter band. `match`, which specifies which filter band to match, is one of

- `passband`—match the passband specification exactly in the final filter.
- `stopband`—match the specified stopband performance exactly in the final filter.
- `both`—match both the passband and stopband specifications. This is the default setting.

Use the **`matchexactly`** input option only when your filter object designs minimum order elliptical filters. This condition applies when you do not specify the filter order in your filter design object. Lowpass, highpass, bandpass, and bandstop filter design objects support a `SpecificationType` string that does not include the filter order as an input specification.

Examples

These example demonstrate using ellip to design filters based on filter design objects.

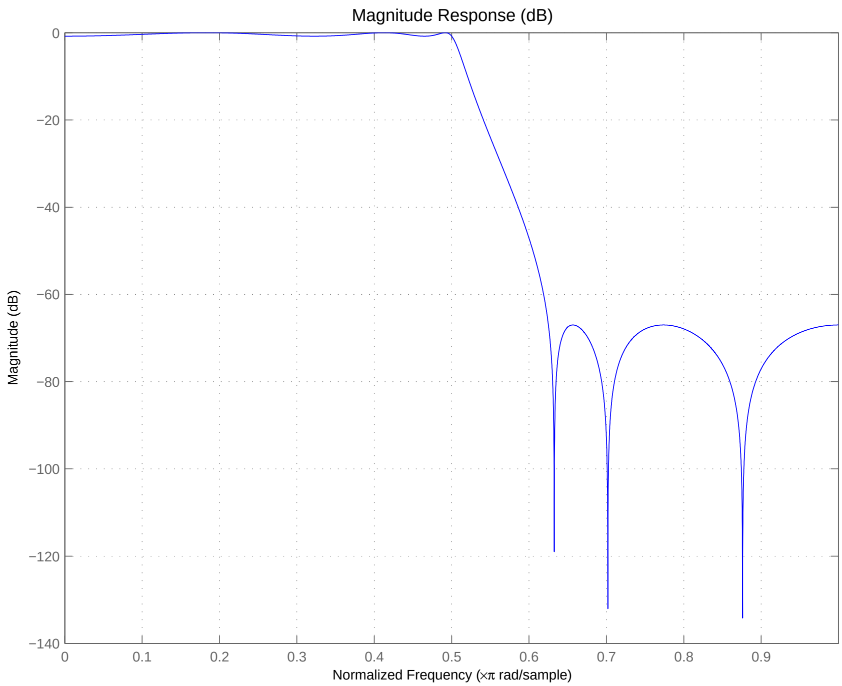
Example 1—construct the default bandpass filter design object and design an elliptic filter.

```
d = fdesign.bandpass;
hd = ellip(d,'matchexactly','passband');

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [4x6 double]
             ScaleValues: [5x1 double]
```

```
     PersistentMemory: false
               States: [2x4 double]
 NumSamplesProcessed: 0
```

Example 2—construct a lowpass object with order, passband-edge frequency, stopband-edge frequency, and passband ripple specifications, and then design an elliptic filter.

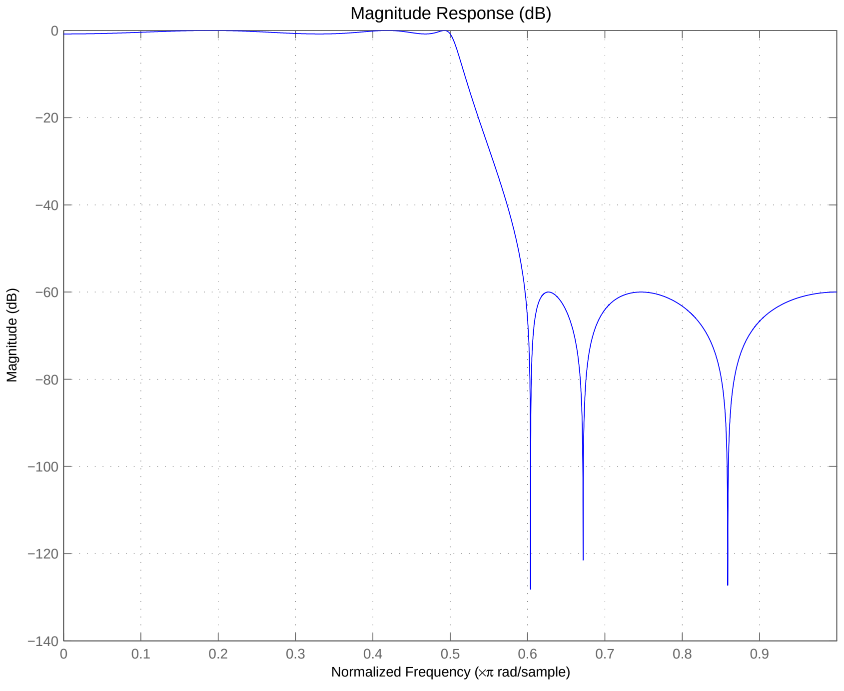
```
d = fdesign.lowpass('n,fp,fst,ap',6,20,25,.8,80);
ellip(d); % Opens fvtool to display the filter.
```



Example 3—construct a lowpass object with filter order, passband edge frequency, passband ripple, and stopband attenuation specifications, and then design an elliptic filter.

```
d = fdesign.lowpass('n,fp,ap,ast',6,20,.8,60,80);
```

```
ellip(d); % Automatically starts fvtool to display the filter.
```



See Also

`butter`, `cheby1`, `cheby2`

Purpose

Use Euclid's theorem to return the integer factors for a multirate filter

Syntax

```
[lo,mo] = euclidfactors(hm)
```

Description

`[lo,mo] = euclidfactors(hm)` returns integer factors `lo` and `mo` such that (`lo`*L)-(`mo`*M) = -1. L and M are relatively prime and represent the interpolation and decimation factors of the multirate filter `hm`.

`euclidfactors` works with multirate filters that have both decimation and interpolation factors, such as `mfilt.firfracdecim`, `mfilt.firfracinterp`, or `mfilt.firsrc`. You cannot return the factors for plain decimators or interpolators

Examples

Use an FIR fractional decimator, with L = 5 and M = 7, to show what `euclidfactors` does.

```
hm=mfilt.firfracdecim(5,7)

hm =

        FilterStructure: 'Direct-Form FIR Polyphase Fractional Decimator'
              Numerator: [1x168 double]
      RateChangeFactors: [5 7]
       PersistentMemory: false
                 States: [62x1 double]
    NumSamplesProcessed: 0

[lo,mo]=euclidfactors(hm)

lo =

     4


mo =

     3
```

Indeed, (`lo`*L)-(`mo`*M) = (4*5)-(3*7) = -1.

See Also

`polyphase`, `nstates`

equiripple

Purpose

Design equiripple single-rate filters or multirate FIR filters from the specifications in a filter design object

Syntax

```
hd = equiripple(d)
hd = equiripple(d,'densityfactor',df)
```

Description

`hd = equiripple(d)` designs an equiripple FIR digital filter or multirate filter using the specifications supplied in the object `d`. Equiripple filter designs minimize the maximum ripple in the pass- and stopbands.

`hd` is either a `dfilt` object (a single-rate digital filter) or an `mfilt` object (a multirate digital filter) depending on the `SpecificationType` property of the filter design object `d` and the design object type—halfband or interp.

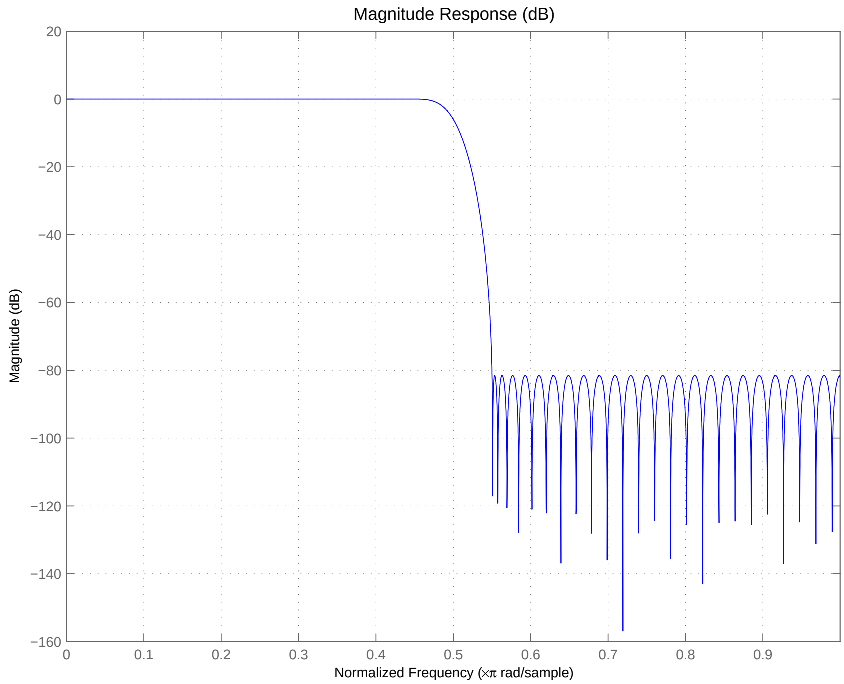
`hd = equiripple(d,'`**`densityfactor`**`',df)` returns an equiripple FIR filter where you specify the density factor, `df`, as an input argument.

Examples

Here is an example of designing a single-rate equiripple filter from a halfband filter design object.

```
d = fdesign.halfband
hd = equiripple(d);
fvtool(hd);
```

Displaying the filter in FVTool shows the equiripple nature of the filter.



`equiripple` also designs multirate filters. This example generates a halfband interpolator filter.

```
d = fdesign.interp(2);
hd = equiripple(d);
```

See Also

`firls`, `kaiserwin`

Purpose

Write a file that contains filter coefficients for multirate filters, adaptive filters, or discrete-time filter objects

Syntax

```
fcfwrite(h)
fcfwrite(h,filename)
fcfwrite(...,'fmt')
```

Description

`fcfwrite(h)` writes a filter coefficient ASCII file to a directory you choose, or your current MATLAB working directory. `h` can be a single filter object or a vector of filter objects. On execution, `fcfwrite` opens the **Export Filter Coefficients to .FCF File** dialog to let you assign a file name for the output file. You can choose the destination directory within this dialog as well.

The default file name is `untitled.fcf`. When you have the Filter Design Toolbox, you can use `fcfwrite(h)` to write filter coefficient files for multirate filters, adaptive filters, and discrete-time filters.

`fcfwrite(h,filename)` writes the filter coefficients and general information to a text file called `filename` in your present MATLAB working directory and opens the file in the MATLAB editor for you to review or modify.

If you do not include a file extension in `filename`, `fcfwrite` adds the extension `fcf` to `filename`.

`fcfwrite(...,'fmt')` writes the filter coefficients in the format specified by the input argument `fmt`. Valid `fmt` values are `hex` for hexadecimal, `dec` for decimal, or `bin` for binary representation of the filter coefficients.

Examples

To demonstrate `fcfwrite`, create a fixed-point IIR filter at the command line, and then write the filter coefficients to a file named `iirfilter.fcf`.

```
d=fdesign.lowpass

d =

           ResponseType: 'Minimum-order lowpass'
      SpecificationType: 'Fp,Fst,Ap,Ast'
            Description: {4x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
```

```
                   Fpass: 0.4500
                   Fstop: 0.5500
                   Apass: 1
                   Astop: 60


hd=butter(d)

hd =

        FilterStructure: 'Direct-Form II, Second-Order Sections'
             Arithmetic: 'double'
              sosMatrix: [13x6 double]
            ScaleValues: [14x1 double]
       PersistentMemory: false
                 States: [2x13 double]
    NumSamplesProcessed: 0

set(hd,'arithmetic','fixed');

fcfwrite(hd,'iirfilter.fcf');
```

Here is the output from `fcfwrite` as it appears in the MATLAB editor. Not shown here is the filename—`iirfilter.fcf` as specified.

```
%
% Generated by MATLAB(R) 7.0 and the Signal Processing Toolbox
% 6.2.
%
% Generated on: 15-Jun-2004 14:05:48
%

File Format: Decimal

Discrete-Time IIR Filter (real)
-------------------------------
Filter Structure   : Direct-Form II, Second-Order Sections
Filter Order       : 25
Stable             : Yes
Linear Phase       : No
Arithmetic         : fixed
```

```
Numerator           : S16Q13
Denominator         : S16Q14
Scale Values        : S16Q14
Input               : S16Q15
Stage Input         : S16Q8
Stage Output        : S16Q10
Output              : S16Q10
State               : S16Q15
Numerator Prod      : S32Q28
Denominator Prod    : S32Q29
Numerator Accum     : S40Q28
Denominator Accum   : S40Q29
Round Mode          : convergent
Overflow Mode       : wrap
Cast Before Sum     : true


SOS matrix:
1  2  1  1  -0.22222900390625  0.88262939453125
1  2  1  1  -0.19903564453125  0.68621826171875
1  2  1  1  -0.18060302734375  0.5303955078125
1  2  1  1  -0.1658935546875   0.40570068359375
1  2  1  1  -0.154052734375    0.305419921875
1  2  1  1  -0.14453125        0.22479248046875
1  2  1  1  -0.136962890625    0.16015625
1  2  1  1  -0.13092041015625  0.10906982421875
1  2  1  1  -0.126220703125    0.06939697265625
1  2  1  1  -0.12274169921875  0.0399169921875
1  2  1  1  -0.12030029296875  0.01947021484375
1  2  1  1  -0.118896484375    0.0074462890625
1  1  0  1  -0.0592041015625   0

Scale Factors:
0.41510009765625
0.371826171875
0.33746337890625
0.3099365234375
0.287841796875
0.27008056640625
0.25579833984375
```

```
0.2445068359375
0.23577880859375
0.22930908203125
0.22479248046875
0.22216796875
0.47039794921875
```

To write two or more filters out to one file, provide the filters as a vector to `fcfwrite`:

```
fcfwrite([hd hd1 hd2])
```

See Also

`adaptfilt`, `dfilt`, `mfilt`

`dfilt`, `fcfwrite` in the Signal Processing Toolbox documentation

fdatool

Purpose Open the Filter Design and Analysis Tool

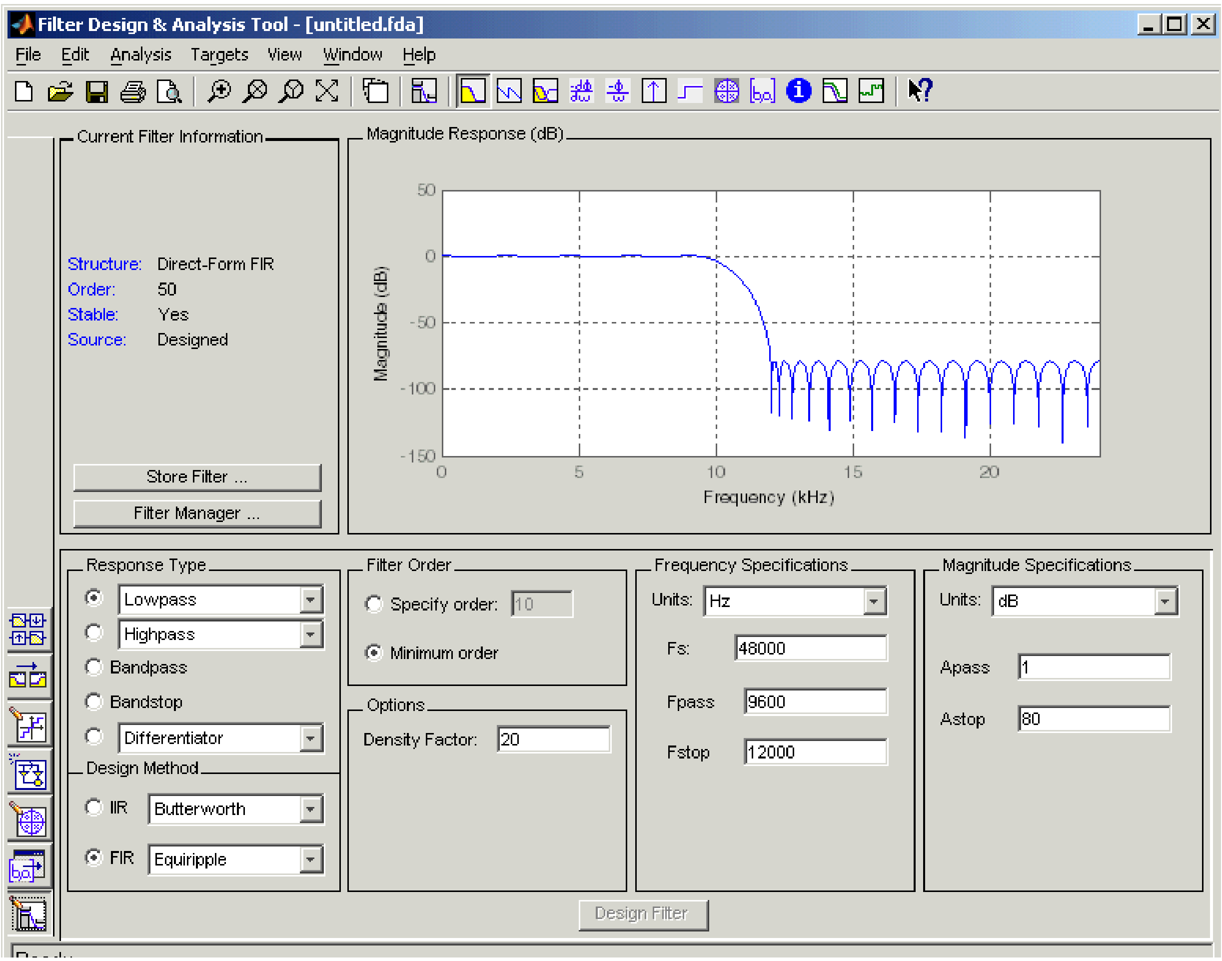
Syntax `fdatool`

Description `fdatool` opens the Filter Design and Analysis Tool (FDATool). Use this tool to:

- Design filters
- Quantize filters
- Analyze filters
- Modify existing filter designs
- Create multirate filters
- Realize Simulink models of quantized, direct-form, FIR filters
- Perform digital frequency transformations of filters

Refer to "Using FDATool with the Filter Design Toolbox" for more information about using the analysis, design, and quantization features of FDATool. For general information about using FDATool, refer to "Filter Design and Analysis Tool" in your Signal Processing Toolbox documentation.

When you open FDATool and you have Filter Design Toolbox installed, FDATool incorporates additional features that are provided by Filter Design Toolbox. With Filter Design Toolbox installed, FDATool lets you design and analyze quantized filters, as well as convert quantized filters to various filter structures, transform filters, design multirate filters, and realize models of filters.



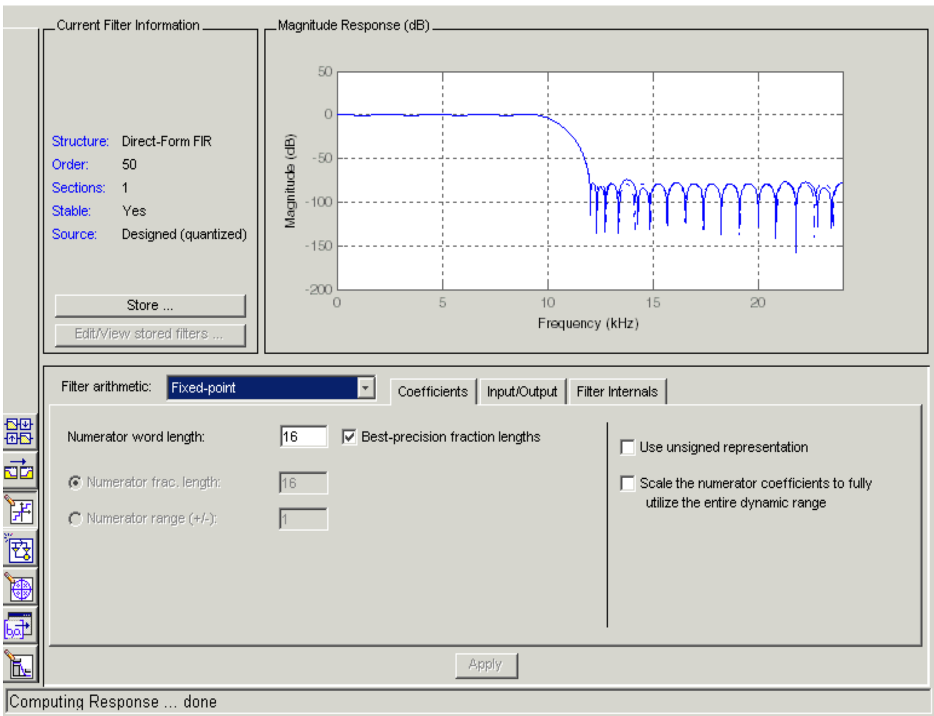
Use the **Set quantization parameters** option to configure the quantization settings for a quantized filter.

Set Quantization Parameters—provides access to the properties of the quantizers that compose a quantized filter. When you click **Set Quantization Parameters**, you see FDATool displaying the quantization options at the bottom of the dialog, as shown in the figure.

Transform Filter—clicking this button opens the **Frequency Transformations** pane so you can use digital frequency transformations to change the magnitude response of your filter.

Create a multirate filter—clicking this button switches FDATool to multirate filter design mode so you can design interpolators, decimators, and fractional rate change filters.

Realize Model—starting from your quantized, direct-form, FIR filter, clicking this button creates a Simulink model of your filter structure in new model window.



Other options in the menu bar let you convert the filter structure to a new structure, change the order of second-order sections in a filter, or change the scaling applied to the filter, among many possibilities.

Remarks

By incorporating many advanced filter design methods from Filter Design Toolbox, FDATool provides more design methods than the SPTool Filter Designer.

See Also

`fdatool`, `fvtool`, `sptool` in your Signal Processing Toolbox documentation

fdesign

Purpose

Create a filter design object for designing discrete-time filters

Syntax

```
d = fdesign.type
d = fdesign.type(specificationtype)
d = fdesign.type(specificationtype,specvalue1,specvalue2,...)
d = fdesign.type(...,fs)
d = fdesign.type(...,magunits)
```

Description

Filter Design Objects

`d = fdesign.`*`type`* returns a filter design object `h`, of filter type *`type`*. To create filters from `d`, use one of the design methods listed in "Using Filter Design Methods With Design Objects" on page 9-505.

`type` can be one of the entries in the following table that specify the filter response desired, such as a bandstop filter or an interpolator.

fdesign Type String	Description
`bandpass`	`fdesign.bandpass` creates an object to design bandpass filters.
`bandstop`	`fdesign.bandstop` creates an object to design bandstop filters.
`decim`	`fdesign.decim` creates an object to design decimators.
`halfband`	`fdesign.halfband` creates an object to design halfband filters.
`highpass`	`fdesign.highpass` creates an object to design highpass filters.
`interp`	`fdesign.interp` creates an object to design interpolators.
`lowpass`	`fdesign.lowpass` creates an object to design lowpass filters.

fdesign Type String	Description
`nyquist`	`fdesign.nyquist` creates an object to design nyquist filters.
`src`	`fdesign.src` creates an object to design sample-rate converters.

Note `fdesign` does not design filters. It returns a filter object that you use with a filter design method to design a filter.

Use the `doc fdesign.type` syntax at the MATLAB prompt to get help on a specific structure. Using `doc` in a syntax like

```
doc fdesign.lowpass
doc fdesign.bandstop
```

gets more information about the lowpass or bandstop structure objects.

Each `type` has a property `SpecificationType` that defines the specifications to use to design your filter. You can use defaults or specify the `SpecificationType` property when you construct the design object. With the `SpecificationType` property, you provide filter constraints such as the filter order or the passband attenuation to use when you construct your filter from the design object.

Properties

`fdesign` returns an object. Every filter design object has the following properties.

Property Name	Default Value	Description
`ResponseType`	Depends on the chosen type	Defines the type of filter to design, such as an interpolator or bandpass filter. This is a read-only value.
`SpecificationType`	Depends on the chosen type	Defines the filter characteristics used to define the desired filter performance, such as the cutoff frequency `Fstop` or the filter order `N`.
`Description`	Depends on the filter type you choose	Contains descriptions of the filter specifications used to define the object, and the filter specifications you use when you create a filter from the object. This is a read-only value.
`NormalizedFrequency`	Logical `true`	Determines whether the filter calculation uses normalized frequency from 0 to 1, or the frequency band from 0 to Fs/2, the sampling frequency. Accepts either `true` or `false` without single quotation marks.

In addition to these properties, filter design objects may have other properties as well, depending on whether they design `dfilt` objects or `mfilt` objects.

Added Properties for mfilt Objects	Description
`DecimationFactor`	Specifies the amount to decrease the sampling rate. Always a positive integer.
`InterpolationFactor`	Specifies the amount to increase the sampling rate. Always a positive integer.
`PolyphaseLength`	Polyphase length is the length of each polyphase subfilter that composes the decimator or interpolator or src filters. Total filter length is the product of `pl` and the rate change factors. `pl` must be an even integer.

`d = fdesign.`*`type`*`(specificationtype)` In `specificationtype`, you specify the variables to use that define your filter design, such as the passband frequency or the stopband attenuation. These variables are applied to the filter design method you choose to design your filter.

For example, when you create a default lowpass filter design object `d`, `fdesign` sets the passband frequency `Fpass`, the stopband frequency `Fstop`, the stopband attenuation `Astop`, and the passband attenuation `Apass` (ripple in the passband) for `d`:

```
d = fdesign.lowpass

d =

          ResponseType: 'Minimum-order lowpass'
     SpecificationType: 'Fp,Fst,Ap,Ast'
           Description: {4x1 cell}
   NormalizedFrequency: true
                    Fs: 'Normalized'
```

```
Fpass: 0.4500
Fstop: 0.5500
Apass: 1
Astop: 60
```

For a lowpass filter, `SpecificationType` must be one of the following strings:

SpecificationType String	Description
`Fp,Fst,Ap,Ast`	Define the filter by specifying the passband cutoff, the stopband cutoff, the ripple in the passband, and the attenuation in the stopband. This is the default string for a lowpass filter.
`N,Fc`	Set the filter order and the cutoff frequency to define the filter.
`N,Fp,Ap`	Set the filter order, passband cutoff frequency, and passband ripple.
`N,Fst,Ast`	Define the filter by setting the order, stopband frequency, and stopband attenuation.
`N,Fp,Ap,Ast`	Set the order, passband cutoff frequency, passband ripple, and stopband attenuation.
`N,Fp,Fst,Ap`	Set the filter order, passband cutoff frequency, stopband frequency, and passband ripple.

Other filter object types, such as Nyquist or highpass, accept a different set of strings for `SpecificationType`. Refer to the Help system for details about the strings for each filter type.

One important note is that the `SpecificationType` string you choose controls which design method works for the design object. For the lowpass filter design object `d` from earlier, you can use `butter`, `cheby1`, `cheby2`, or `ellip` to design a filter. However, if the `SpecificationType` string had been `'n,fp,fst,ap'`, you could only use the `ellip` design method to design your filter.

When you implement this lowpass filter `hd` using a filter design method such as Butterworth (the butter design function), the constraints in `fp`, `fst`, `ap`, and ast (the default string and filter specification) define the response of the final minimum-order lowpass filter:

```
hd = butter(d)

hd =

         FilterStructure: 'Direct-Form II, Second-Order Sections'
              Arithmetic: 'double'
               sosMatrix: [13x6 double]
             ScaleValues: [14x1 double]
        PersistentMemory: false
                  States: [2x13 double]
     NumSamplesProcessed: 0
```

FVTool shows that hd is a lowpass filter that meets the design specification.

d = fdesign.*type*(...,fs) adds the argument fs, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

d = fdesign.*type*(...,magunits) specifies the units for any magnitude specification you provide in the input arguments. magunits can be one of

- linear—specify the magnitude in linear units
- dB—specify the magnitude in dB (decibels)
- squared—specify the magnitude in power units

When you omit the magunits argument, fdesign assumes that all magnitudes are in dB. Note that fdesign stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

Using Filter Design Methods With Design Objects

After you create a filter design object, you use a filter design method to implement your filter with a selected algorithm. The following methods are available for filter design objects, but all methods do not apply to all object types. Also, the specification string you use to define the object changes the algorithms available to design a filter. Enter doc butter, for example, to get more information about using the Butterworth design method with your filter design object.

Design Function	Description
`butter`	Implement a Butterworth filter resulting in an SOS filter with direct-form II structure.
`cheby1`	Implement a Chebyshev Type I filter, resulting in a direct-form II second-order filter.
`cheby2`	Implement a Chebyshev Type II filter, resulting in an SOS filter with direct-form II structure
`ellip`	Implement an elliptic filter resulting in an SOS filter with direct-form II structure
`equiripple`	Implement an equiripple filter.
`firls`	Implement a least-squares filter.
`kaiserwin`	Implement a filter that uses a Kaiser window.

When you use any of the design methods without providing an output argument, the resulting filter design appears in FVTool by default.

Along with filter design methods, `fdesign` works with supporting methods that help you create filter design objects or determine which design methods work for a given design object.

Supporting Function	Description
`setspecs`	Set all of the specifications simultaneously.
`designmethods`	Return the design methods.

You can set filter specification values by passing them after the SpecificationType argument, or by passing the values without the SpecificationType string.

Filter object constructors take the input arguments in the same order as setspecs and the order in the strings for SpecificationType. Enter doc setspecs at the prompt for more information about using setspecs.

When the first input to fdesign is not a valid SpecificationType string like 'n,fc', fdesign assumes that the input argument is a filter specification and applies it using the default SpecificationType string—fp,fst,ap,ast for a lowpass object, for example.

Examples

These examples show a few default filter objects constructed from the MATLAB command prompt, and how to design a Butterworth filter.

Example 1—Halfband filter object with filter order and stopband attenuation provided as input arguments. Add the linear magunits option so you specify the attenuation in decimal—0.0001.

```
n = 80;
ast = 1e-4;
fs = 48000
d=fdesign.halfband('n,ast',n,ast,fs,'linear')

d =

           ResponseType: [1x51 char]
      SpecificationType: 'N,Ast'
            Description: {2x1 cell}
    NormalizedFrequency: false
                     Fs: 48000
            FilterOrder: 80
                  Astop: 80

d.description

ans =


    'Filter Order'
```

```
    'Stopband Attenuation (dB)'
```

Example 2—Interpolator Filter Object

```
d = fdesign.interp

d =

           ResponseType: 'Minimum-order halfband'
      SpecificationType: 'TW,Ast'
            Description: {2x1 cell}
    InterpolationFactor: 2
    NormalizedFrequency: true
                     Fs: 'Normalized'
        TransitionWidth: 0.1000
                  Astop: 80


d.Description

ans =

    'Transition Width'
    'Stopband Attenuation (dB)'
```

Example 3—Highpass Filter Object

```
d=fdesign.highpass

d =

           ResponseType: 'Minimum-order highpass'
      SpecificationType: 'Fst,Fp,Ast,Ap'
            Description: {4x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
                  Fstop: 0.4500
                  Fpass: 0.5500
                  Astop: 60
                  Apass: 1
```

```
d.Description

ans =

    'Stopband Frequency'
    'Passband Frequency'
    'Stopband Attenuation (dB)'
    'Passband Ripple (dB)'
```

Notice the correspondence between the properties `SpecificationType` and `Description`—in `Description` you see in words the definitions of the variables shown in `SpecificationType`.

Example 4—Lowpass Butterworth Filter Design

Use a filter design object to construct a lowpass Butterworth filter with default `SpecificationType` `fp,fst,ap,ast`—the edge frequencies of the passband and stopband, the attenuation in the passband, and the attenuation in the stopband. Start by creating the design object `d` and providing the filter order and cutoff frequency values.

```
d = fdesign.lowpass(0.4,0.5,1,80);
d

d =

           ResponseType: 'Minimum-order lowpass'
      SpecificationType: 'Fp,Fst,Ap,Ast'
            Description: {4x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
                  Fpass: 0.4000
                  Fstop: 0.5000
                  Apass: 1
                  Astop: 80
```

Determine which design methods apply to `d`.

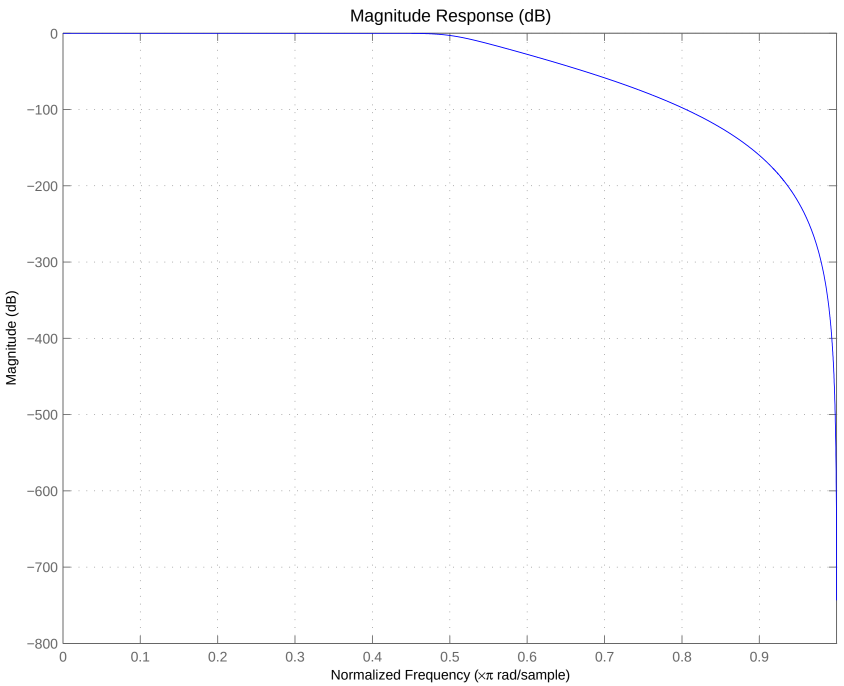
```
designmethods(d)


Design Methods for class fdesign.lowpass:
```

```
butter
cheby1
cheby2
ellip
```

Now use d and the butter design method to design a Butterworth filter.

```
hd = butter(d,'matchexactly','passband');
fvtool(hd);
```

The resulting filter magnitude response shown by FVTool appears below.



If you had a default Nyquist filter design object d

```
d = fdesign.nyquist
```

you could find out which design methods apply to d by entering

```
designmethods(d)


Design methods for class fdesign.nyquist:


kaiserwin
```

Notice that only the Kaiser window-based design method applies to default Nyquist filter objects.

See Also

butter, cheby1, cheby2, designmethods, ellip, equiripple, fdatool, fdesign.bandpass, fdesign.bandstop, fdesign.decim, fdesign.halfband, fdesign.highpass, fdesign.interp, fdesign.lowpass, fdesign.nyquist, fdesign.src, firls, fvtool, kaiserwin, setspecs

fdesign.bandpass

Purpose Construct a filter design object that has bandpass filter specifications

Syntax

```
d = fdesign.bandpass
d = fdesign.bandpass(spectype)
d = fdesign.bandpass(spectype,specvalue1,specvalue2,...)
d = fdesign.bandpass(fstop1,fpass1,fpass2,fstop2,astop1,...
   apass,astop2)
d = fdesign.bandpass(...,fs)
d = fdesign.bandpass(...,magunits)
```

Description

`d = fdesign.bandpass` constructs a bandpass filter design object `d`, applying default values for the properties `Fstop1`, `Fpass1`, `Fpass2`, `Fstop2`, `Astop1`, `Apass`, and `Astop2`—one possible set of values you use to specify a bandpass filter.

Using `fdesign.bandpass` with a design method generates a `dfilt` object.

`d = fdesign.bandpass(`*`spectype`*`)` constructs object `d` and sets its `SpecificationType` property to `spectype`. Entries in the `spectype` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for *`spectype`* are shown below and used to define the bandpass filter. The strings are not case sensitive.

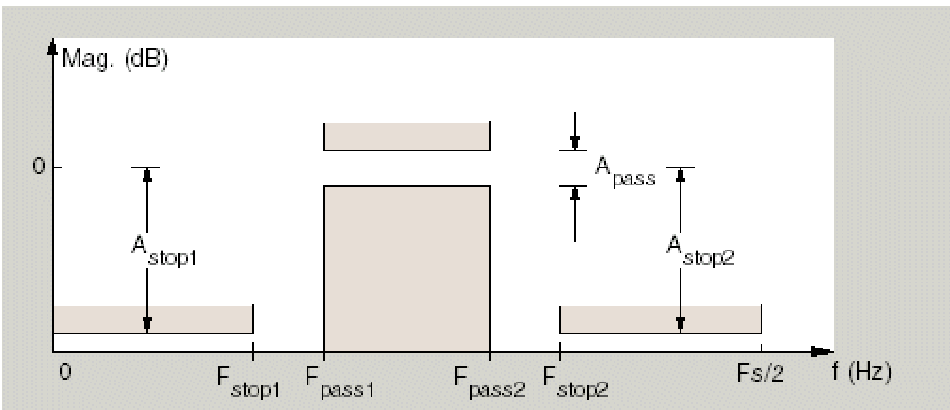
- `fst1,fp1,fp2,fst2,ast1,ap,ast2` (default `spectype`)
- `n,fc1,fc2`
- `n,fp1,fp2,ap`
- `n,fst1,fst2,ast`
- `n,fp1,fp2,ast1,ap,ast2`
- `n,fst1,fp1,fp2,fst2,ap`

The string entries are defined as follows:

- `ap`—amount of ripple allowed in the pass band. Also called Apass.
- `ast1`—attenuation in the first stop band in dB (the default units). Also called Astop1.
- `ast2`—attenuation in the second stop band in dB (the default units). Also called Astop2.

- `fc1`—cutoff frequency for the point 3dB point below the passband value for the first cutoff. Specified in normalized frequency units.
- `fc2`—cutoff frequency for the point 3dB point below the passband value for the second cutoff. Specified in normalized frequency units.
- `fp1`—frequency at the edge of the start of the pass band. Specified in normalized frequency units. Also called Fpass1.
- `fp2`—frequency at the edge of the end of the pass band. Specified in normalized frequency units. Also called Fpass2.
- `fst1`—frequency at the edge of the start of the first stop band. Specified in normalized frequency units. Also called Fstop1.
- `fst2`—frequency at the edge of the start of the second stop band. Specified in normalized frequency units. Also called Fstop2.
- `n`—filter order.

Graphically, the filter specifications look like this.



Regions between specification values like `fst1` and `fp1` are transition regions where the filter response is not explicitly defined.

The filter design methods that apply to a bandpass filter design object change depending on the `SpecificationType` string. Paired with each string in the following table are the design methods for bandpass design objects that use that string.

SpecificationType String	Applicable Design Method
fst1,fp1,fp2,fst2,ast1,ap,ast2	butter,cheby1,cheby2,ellip
n,fc1,fc2	butter
n,fp1,fp2,ap	cheby1
n,fst1,fst2,ast	cheby2
n,fp1,fp2,ast1,ap,ast2	ellip
n,fst1,fp1,fp2,fst2,ap	ellip

`d = fdesign.bandpass(spectype,specvalue1,specvalue2,...)` constructs an object `d` and sets its specifications at construction time.

`d = fdesign.bandpass(fstop1,fpass1,fpass2,fstop2,astop1,... apass,astop2)` constructs `d`, an object with the default `SpecificationType` property string, using the specifications provided as input arguments.

`d = fdesign.bandpass(...,fs)` adds the argument `fs`, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

`d = fdesign.bandpass(...,magunits)` specifies the units for any magnitude specification you provide in the input arguments. `magunits` can be one of

- `linear`—specify the magnitude in linear units
- `dB`—specify the magnitude in dB (decibels)
- `squared`—specify the magnitude in power units

When you omit the `magunits` argument, `fdesign` assumes that all magnitudes are in dB. Note that `fdesign` stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

As you change the specifications, the `ResponseType` property changes as well. You see the `ResponseType` property when you get the filter object information.

```
get(d)
            ResponseType: 'Minimum-order bandpass'
             Description: {7x1 cell}
       SpecificationType: 'Fst1,Fp1,Fp2,Fst2,Ast1,Ap,Ast2'
     NormalizedFrequency: 0
                      Fs: 48
                  Fstop1: 10
                  Fpass1: 12
                  Fpass2: 14
                  Fstop2: 16
                  Astop1: 80
                   Apass: 0.5000
                  Astop2: 60
```

Examples

These examples show how to construct a bandpass filter design object. First, create a default object without using input arguments.

```
d = fdesign.bandpass
d =

            ResponseType: 'Minimum-order bandpass'
       SpecificationType: 'Fst1,Fp1,Fp2,Fst2,Ast1,Ap,Ast2'
             Description: {7x1 cell}
     NormalizedFrequency: true
                      Fs: 'Normalized'
                  Fstop1: 0.3500
                  Fpass1: 0.4500
                  Fpass2: 0.5500
                  Fstop2: 0.6500
                  Astop1: 60
                   Apass: 1
                  Astop2: 60
```

Now, pass the filter specifications that correspond to the default `SpecificationType`—`fst1,fp1,fp2,fst2,ast1,ap,ast2`—without specifying the `SpecificationType` string. Notice that we add `fs` as the final input argument to specify the sampling frequency of 48 Hz.

```
d = fdesign.bandpass(10, 12, 14, 16, 80, .5, 60, 48)
d =
```

```
          ResponseType: 'Minimum-order bandpass'
     SpecificationType: 'Fst1,Fp1,Fp2,Fst2,Ast1,Ap,Ast2'
           Description: {7x1 cell}
   NormalizedFrequency: false
                    Fs: 48
                Fstop1: 10
                Fpass1: 12
                Fpass2: 14
                Fstop2: 16
                Astop1: 80
                 Apass: 0.5000
              Astop2: 60
```

Next create a design object by passing a specification type string 'n,fc1,fc2'—the resulting object uses default values for `n`, `fc1`, and `fc2`.

```
d = fdesign.bandpass('n,fc1,fc2')

d =

          ResponseType: 'Bandpass with cutoff'
     SpecificationType: 'N,Fc1,Fc2'
           Description: {3x1 cell}
   NormalizedFrequency: true
                    Fs: 'Normalized'
           FilterOrder: 10
              Fcutoff1: 0.4000
              Fcutoff2: 0.6000
```

Create the same filter, passing the specification values to the object rather than accepting the default values for `n`, `fc1`, and `fc2`. Notice that you can include the sampling frequency `fs` as the final input argument, and that you specify the cutoff frequencies in Hz since `fs` is in Hz.

```
d = fdesign.bandpass('n,fc1,fc2', 10, 9600, 14400, 48000)
d =

          ResponseType: 'Bandpass with cutoff'
     SpecificationType: 'N,Fc1,Fc2'
           Description: {3x1 cell}
   NormalizedFrequency: false
                    Fs: 48000
```

```
FilterOrder: 10
   Fcutoff1: 9600
   Fcutoff2: 14400
```

See Also

`fdesign`, `fdesign.bandstop`, `fdesign.highpass`, `fdesign.lowpass`

fdesign.bandstop

Purpose Construct a filter design object that has bandstop filter specifications

Syntax

```
d = fdesign.bandstop
d = fdesign.bandstop(spectype)
d = fdesign.bandstop(spectype,specvalue1,specvalue2,...)
d = fdesign.bandstop(fstop1,fpass1,fpass2,fstop2,astop1,...
   apass,astop2)
d = fdesign.bandstop(...,fs)
d = fdesign.bandstop(...,magunits)
```

Description

`d = fdesign.bandstop` constructs a bandstop filter design object `d`, applying default values for the properties `Fpass1`, `Fstop1`, `Fstop2`, `Fpass2`, `Apass1`, `Astop1` and `Apass2`.

Using `fdesign.bandstop` with a design method generates a `dfilt` object.

`d = fdesign.bandstop(`*`spectype`*`)` constructs object `d` and sets its '`SpecificationType`' to *`spectype`*. Entries in the `spectype` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for *`spectype`* are shown below. The strings are not case sensitive.

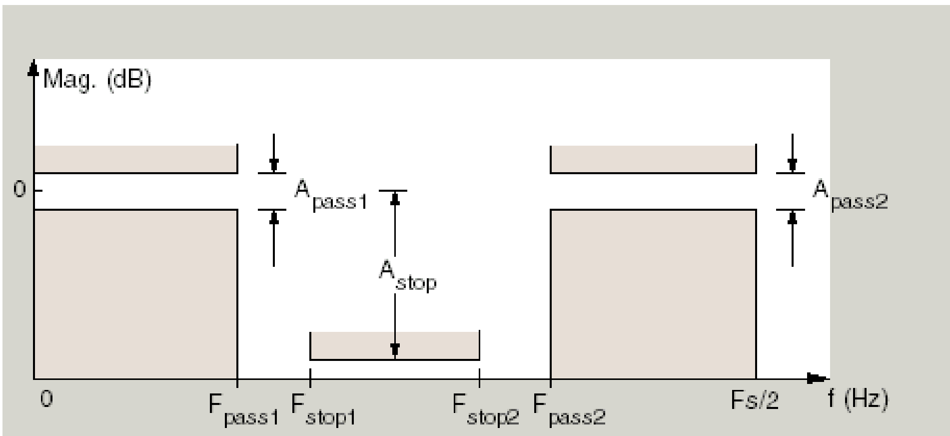
- `fp1,fst1,fst2,fp2,ap1,ast,ap2` (default `spectype`)
- `n,fc1,fc2`
- `n,fp1,fp2,ap`
- `n,fst1,fst2,ast`
- `n,fp1,fp2,ast,ap`
- `n,fp1,fst1,fst2,fp2,ap`

The string entries are defined as follows:

- `ap`—amount of ripple allowed in the pass band in dB (the default units). Also called Apass.
- `ast`—attenuation in the first stop band in dB (the default units). Also called Astop1.
- `fc1`—cutoff frequency for the point 3dB point below the passband value for the first cutoff. Specified in normalized frequency units.

- `fc2`—cutoff frequency for the point 3dB point below the passband value for the second cutoff. Specified in normalized frequency units.
- `fp1`—frequency at the start of the pass band. Specified in normalized frequency units. Also called Fpass1.
- `fp2`—frequency at the end of the pass band. Specified in normalized frequency units. Also called Fpass2.
- `fst1`—frequency at the end of the first stop band. Specified in normalized frequency units. Also called Fstop1.
- `fst2`—frequency at the start of the second stop band. Specified in normalized frequency units. Also called Fstop2.
- `n`—filter order.

Graphically, the filter specifications look like this:



Regions between specification values like `fp1` and `fst1` are transition regions where the filter response is not explicitly defined.

The filter design methods that apply to a bandstop filter design object change depending on the `SpecificationType` string. Paired with each string in the following table are the design methods for bandstop design objects that use that string.

SpecificationType String	Applicable Design Method
`fp1,fst1,fst2,fp2,ap1,ast,ap2`	`butter`, `cheby1`, `cheby2`, `ellip`
`n,fc1,fc2`	`butter`
`n,fp1,fp2,ap`	`cheby1`
`n,fst1,fst2,ast`	`cheby2`
`n,fp1,fp2,ap,ast`	`ellip`
`n,fp1,fst1,fst2,fp2,ap`	`ellip`

`d = fdesign.bandstop(spectype,spec1,spec2,...)` constructs an object `d` and sets its specifications at construction time.

`d = fdesign.bandstop(fpass1,fstop1,fstop2,fpass2,apass1,... astop,apass2)` constructs an object `d` with the default 'SpecificationType' property string, using the specifications you provide as input arguments.

`d = fdesign.bandstop(...,fs)` adds the argument `fs`, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

`d = fdesign.bandstop(...,magunits)` specifies the units for any magnitude specification you provide in the input arguments. `magunits` can be one of

- `linear`—specify the magnitude in linear units
- `dB`—specify the magnitude in dB (decibels)
- `squared`—specify the magnitude in power units

When you omit the `magunits` argument, `fdesign` assumes that all magnitudes are in dB. Note that `fdesign` stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

As you change the specifications, the `ResponseType` property changes as well. You see the `ResponseType` property when you get the filter object information.

```
get(d)

             ResponseType: 'Minimum-order bandstop'
              Description: {7x1 cell}
        SpecificationType: 'Fp1,Fst1,Fst2,Fp2,Ap1,Ast,Ap2'
      NormalizedFrequency: 1
                       Fs: 'Normalized'
                   Fpass1: 0.3500
                   Fstop1: 0.4500
                   Fstop2: 0.5500
                   Fpass2: 0.6500
                   Apass1: 1
                    Astop: 60
                   Apass2: 1
```

Examples

These examples show how to construct a bandpass filter design object. First, create a default object without using input arguments.

```
d = fdesign.bandstop
d =

             ResponseType: 'Minimum-order bandstop'
              Description: {7x1 cell}
        SpecificationType: 'Fp1,Fst1,Fst2,Fp2,Ap1,Ast,Ap2'
      NormalizedFrequency: 1
                       Fs: 'Normalized'
                   Fpass1: 0.3500
                   Fstop1: 0.4500
                   Fstop2: 0.5500
                   Fpass2: 0.6500
                   Apass1: 1
                    Astop: 60
                   Apass2: 1
```

Now create an object by passing a specification type string 'n,fc1,fc2'—the resulting object uses default values for n, fc1, and fc2.

```
d=fdesign.bandstop('n,fc1,fc2')

d =

             ResponseType: 'Bandstop with cutoff'
```

```
        SpecificationType: 'N,Fc1,Fc2'
              Description: {3x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
              FilterOrder: 10
                 Fcutoff1: 0.4000
                 Fcutoff2: 0.6000

designmethods(d)

Design Methods for class fdesign.bandstop:


butter
cheby1
cheby2
ellip
```

Create another bandstop filter, passing the specification values to the object rather than accepting the default values for `n`, `fc1`, and `fc2`. Notice that you can add `fs` as the final input argument to specify the sampling frequency of 48 kHz.

```
d = fdesign.bandstop('n,fc1,fc2', 10, 9600, 14400, 48000)

d =

             ResponseType: 'Bandstop with cutoff'
        SpecificationType: 'N,Fc1,Fc2'
              Description: {3x1 cell}
      NormalizedFrequency: false
                       Fs: 48000
              FilterOrder: 10
                 Fcutoff1: 9600
                 Fcutoff2: 14400
```

For this bandstop filter, pass the filter specifications that correspond to the default `SpecificationType`—`fp1,fst1,fst2,fp2,ap1,ast,ap2`.

```
d = fdesign.bandstop(0.3,0.4,0.6,0.7,0.5,60,1)
```

```
d =

             ResponseType: 'Minimum-order bandstop'
        SpecificationType: 'Fp1,Fst1,Fst2,Fp2,Ap1,Ast,Ap2'
              Description: {7x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
                   Fpass1: 0.3000
                   Fstop1: 0.4000
                   Fstop2: 0.6000
                   Fpass2: 0.7000
                   Apass1: 0.5000
                    Astop: 60
                   Apass2: 1
```

And for the final example, pass the magnitude specifications in squared units, using the `magunits` option `squared`.

```
d = fdesign.bandstop(0.4,0.5,0.6,0.7,0.98,0.01,0.99,'squared')
d =

             ResponseType: 'Minimum-order bandstop'
        SpecificationType: 'Fp1,Fst1,Fst2,Fp2,Ap1,Ast,Ap2'
              Description: {7x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
                   Fpass1: 0.4000
                   Fstop1: 0.5000
                   Fstop2: 0.6000
                   Fpass2: 0.7000
                   Apass1: 0.0877
                    Astop: 20
                   Apass2: 0.0436
```

See Also

`fdesign`, `fdesign.bandpass`, `fdesign.highpass`, `fdesign.lowpass`

fdesign.decim

Purpose Construct a filter design object that has decimator filter specifications

Syntax

```
d = fdesign.decim(m)
d = fdesign.decim(m,spectype)
d = fdesign.decim(m,spectype,specvalue1,specvalue2,...
d = fdesign.decim(m,tw,ast)
d = fdesign.decim(...,fs)
d = fdesign.decim(...,magunits)
```

Description

`d = fdesign.decim(m)` constructs a decimating filter design object `d`, applying default values for the properties `tw` and `ast`. Specify `m`, the decimation factor as an integer. When you omit the input argument `m`, `fdesign, decim` sets the decimation factor `m` to 2.

Using `fdesign.decim` with a design method generates an `mfilt` object.

`d = fdesign.decim(m,`*`spectype`*`)` constructs object `d` and sets its '`SpecificationType`' to *`spectype`*. Entries in the `spectype` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for *`spectype`* are shown below. The strings are not case sensitive. Notice that the decimation factor `m` is not in the specification strings.

- `tw,ast` (default `spectype`)
- `pl,tw`
- `pl`
- `pl,ast`

The string entries are defined as follows:

- `ast`—attenuation in the stop band in dB (the default units).
- `pl`—polyphase filter length. Polyphase length is the length of each polyphase subfilter that composes the decimator. Total filter length is the product of `pl` and the decimation factor. `pl` must be an even integer.
- `tw`—width of the transition region between the pass and stop bands.

The filter design methods that apply to a decimating filter design object change depending on the `SpecificationType` string. Paired with each string in the following table are the design methods for decimating filter design objects that use that string.

SpecificationType String	Applicable Design Method
tw,ast	equiripple, kaiserwin
pl,tw	kaiserwin
pl	window
pl,ast	kaiserwin

d = fdesign.decim(m,spectype,spec1,spec2,...) constructs an object d and sets its specifications at construction time.

d = fdesign.decim(m,'tw,ast') constructs an object d with the default 'SpecificationType' property string, using the default specification values.

d = fdesign.decim(...,fs) adds the argument fs, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

d = fdesign.decim(...,magunits) specifies the units for any magnitude specification you provide in the input arguments. magunits can be one of

- linear—specify the magnitude in linear units
- dB—specify the magnitude in dB (decibels)
- squared—specify the magnitude in power units

When you omit the magunits argument, fdesign assumes that all magnitudes are in dB. Note that fdesign stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

As you change the specifications, the ResponseType property changes as well. You see the ResponseType property when you get the filter object information.

```
get(d)
           ResponseType: 'Minimum-order halfband'
            Description: {2x1 cell}
      SpecificationType: 'TW,Ast'
```

```
    DecimationFactor: 2
 NormalizedFrequency: 1
                  Fs: 'Normalized'
     TransitionWidth: 0.1000
               Astop: 80
```

Examples

These examples show how to construct a decimation filter design object. First, create a default object without using input arguments.

```
d = fdesign.decim(2)
d =

        ResponseType: 'Minimum-order halfband'
         Description: {2x1 cell}
   SpecificationType: 'TW,Ast'
    DecimationFactor: 2
 NormalizedFrequency: 1
                  Fs: 'Normalized'
     TransitionWidth: 0.1000
               Astop: 80
```

Now create an object by passing a specification type string 'pl,ast'—the resulting object uses default values for `pl` and `ast`.

```
d=fdesign.decim(3,'pl,ast')

d =

        ResponseType: [1x50 char]
   SpecificationType: 'PL,Ast'
         Description: {2x1 cell}
    DecimationFactor: 3
 NormalizedFrequency: true
                  Fs: 'Normalized'
     PolyphaseLength: 24
               Astop: 80
```

Create another decimating filter object, passing the specification values to the object rather than accepting the default values for `tw` and `ast`.

```
d=fdesign.decim(3,.01,80)
```

```
d =

          ResponseType: 'Minimum-order nyquist'
     SpecificationType: 'TW,Ast'
           Description: {2x1 cell}
      DecimationFactor: 3
   NormalizedFrequency: true
                    Fs: 'Normalized'
       TransitionWidth: 0.0100
                 Astop: 80
```

Now pass the filter specifications that correspond to the `SpecificationType`— `pl,ast`.

```
d=fdesign.decim(3,'pl,ast',42,80)
```

```
d =

          ResponseType: [1x50 char]
     SpecificationType: 'PL,Ast'
           Description: {2x1 cell}
      DecimationFactor: 3
   NormalizedFrequency: true
                    Fs: 'Normalized'
       PolyphaseLength: 42
                 Astop: 80
```

Now design a decimator using the `kaiserwin` design method.

```
hm = kaiserwin(d)
```

Pass a new specification type for the filter, specifying the filter order.

```
 hs = fdesign.decim(5,'pl,ast',12)
 hs =

          ResponseType: [1x50 char]
     SpecificationType: 'PL,Ast'
           Description: {2x1 cell}
      DecimationFactor: 5
   NormalizedFrequency: true
                    Fs: 'Normalized'
       PolyphaseLength: 12
```

```
                   Astop: 80
```

In this example, you specify a sampling frequency as the last input argument.

```
hs = fdesign.decim(2,'PL,TW',18,.1,5)

hs =

          ResponseType: [1x47 char]
     SpecificationType: 'PL,TW'
           Description: {2x1 cell}
      DecimationFactor: 2
   NormalizedFrequency: false
                    Fs: 5
       PolyphaseLength: 18
       TransitionWidth: 0.1000
```

In this last example, use the linear option for the filter and specify the stopband ripple attenuation in linear format.

```
hs = fdesign.decim(4,'TW,Ast',.1,1e-3,5,'linear') % 1e-3 = 60dB

hs =

          ResponseType: 'Minimum-order nyquist'
     SpecificationType: 'TW,Ast'
           Description: {2x1 cell}
      DecimationFactor: 4
   NormalizedFrequency: false
                    Fs: 5
       TransitionWidth: 0.1000
                 Astop: 60
```

See Also

`fdesign`, `fdesign.halfband`, `fdesign.interp`, `fdesign.nyquist`,

Purpose

Construct a filter design object that has halfband filter specifications

Syntax

```
d = fdesign.halfband
d = fdesign.halfband(spectype)
d = fdesign.halfband(spectype,specvalue1,specvalue2,...)
d = fdesign.halfband(tw,ast)
d = fdesign.halfband(...,fs)
d = fdesign.halfband(...,magunits)
```

Description

`d = fdesign.halfband` constructs a halfband filter design object `d`, applying default values for the properties `tw` and `ast`.

Using `fdesign.halfband` with a design method generates a `dfilt` object.

`d = fdesign.halfband(`*`spectype`*`)` constructs object `d` and sets its '`SpecificationType`' to *`spectype`*. Entries in the `spectype` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for *`spectype`* are shown below. The strings are not case sensitive.

- `tw,ast` (default `spectype`)
- `n,tw`
- `n`
- `n,ast`

The string entries are defined as follows:

- `ast`—attenuation in the stop band in dB (the default units).
- `n`—filter order.
- `tw`—width of the transition region between the pass and stop bands. Specified in normalized frequency units.

The filter design methods that apply to a halfband filter design object change depending on the `SpecificationType` string. Paired with each string in the following table are the design methods for halfband filter design objects that use that string.

SpecificationType String	Applicable Design Method
tw,ast	equiripple, kaiserwin
n,tw	equiripple, kaiserwin, firls
n	window
n,ast	equiripple, kaiserwin

`d = fdesign.halfband(spectype,spec1,spec2,...)` constructs an object `d` and sets its specifications at construction time.

`d = fdesign.halfband(tw,ast)` constructs an object `d` assuming the default '`SpecificationType`' property string `'tw,ast'`, using the default specification values for the input arguments `tw` and `ast`.

`d = fdesign.halfband(...,fs)` adds the argument `fs`, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

`d = fdesign.halfband(...,magunits)` specifies the units for any magnitude specification you provide in the input arguments. `magunits` can be one of

- `linear`—specify the magnitude in linear units
- `dB`—specify the magnitude in dB (decibels)
- `squared`—specify the magnitude in power units

When you omit the `magunits` argument, `fdesign` assumes that all magnitudes are in dB. Note that `fdesign` stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

As you change the specifications, the `ResponseType` property changes as well. You see the `ResponseType` property when you get the filter object information.

```
get(d)
            ResponseType: 'Minimum-order halfband'
             Description: {2x1 cell}
```

```
      SpecificationType: 'TW,Ast'
       DecimationFactor: 2
    NormalizedFrequency: 1
                     Fs: 'Normalized'
        TransitionWidth: 0.1000
                  Astop: 80
```

Examples

These examples show how to construct a halfband filter design object. First, create a default object without using input arguments.

```
d=fdesign.halfband

d =

           ResponseType: 'Minimum-order halfband'
      SpecificationType: 'TW,Ast'
            Description: {2x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
        TransitionWidth: 0.1000
                  Astop: 80
```

Now create an object by passing a specification type string 'n,ast'—the resulting object uses default values for n and ast.

```
d=fdesign.halfband('n,ast')

d =

          ResponseType: 'Halfband with filter order and stopband
attenuation'
      SpecificationType: 'N,Ast'
            Description: {2x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
            FilterOrder: 10
                  Astop: 80
```

Create another halfband filter object, passing the specification values to the object rather than accepting the default values for n and ast.

```
d = fdesign.halfband('n,ast', 42, 80)
```

```
d =

         ResponseType: 'Halfband with filter order and stopband attenuation'
    SpecificationType: 'N,Ast'
          Description: {2x1 cell}
  NormalizedFrequency: true
                   Fs: 'Normalized'
          FilterOrder: 42
                Astop: 80
```

For another example, pass the filter values that correspond to the default SpecificationType—n,ast.

```
d = fdesign.halfband(.01, 80)

d =

         ResponseType: 'Minimum-order halfband'
    SpecificationType: 'TW,Ast'
          Description: {2x1 cell}
  NormalizedFrequency: true
                   Fs: 'Normalized'
      TransitionWidth: 0.0100
                Astop: 80%
```

This example designs an equiripple FIR filter, starting by passing a new specification type and specification values to fdesign.halfband.

```
hs = fdesign.halfband('n,ast',80,70);
hs

hs =

         ResponseType: [1x51 char]
    SpecificationType: 'N,Ast'
          Description: {2x1 cell}
  NormalizedFrequency: true
                   Fs: 'Normalized'
          FilterOrder: 80
                Astop: 70

equiripple(h); % Opens FVTool automatically.
```

In the final example, pass the for the filter, and then design a least-squares FIR filter from the object, using `firls` as the design method.

```
hs = fdesign.halfband('n,tw', 42, .04)

hs =

            ResponseType: [1x47 char]
       SpecificationType: 'N,TW'
             Description: {2x1 cell}
     NormalizedFrequency: true
                      Fs: 'Normalized'
             FilterOrder: 42
         TransitionWidth: 0.0400


designmethods(hs)


Design Methods for class fdesign.halfband:


equiripple
kaiserwin
firls

hd=firls(hs)

hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'double'
               Numerator: [1x43 double]
        PersistentMemory: false
                  States: [42x1 double]
     NumSamplesProcessed: 0
```

See Also

`fdesign`, `fdesign.decim`, `fdesign.interp`, `fdesign.nyquist`, `fdesign.src`

fdesign.highpass

Purpose Construct a filter design object that has highpass filter specifications

Syntax

```
d = fdesign.highpass
d = fdesign.highpass(spectype)
d = fdesign.highpass(spectype,specvalue1,specvalue2,...)
d = fdesign.highpass(fst,fp,ast,ap)
d = fdesign.highpass(...,fs)
d = fdesign.highpass(...,magunits)
```

Description

`d = fdesign.highpass` constructs a bandstop filter design object `d`, applying default values for the properties `fst`, `fp`, `ast` and `ap`.

Using `fdesign.highpass` with a design method generates a `dfilt` object.

`d = fdesign.highpass(`*`spectype`*`)` constructs object `d` and sets its '`SpecificationType`' to *`spectype`*. Entries in the `spectype` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for *`spectype`* are shown below. The strings are not case sensitive.

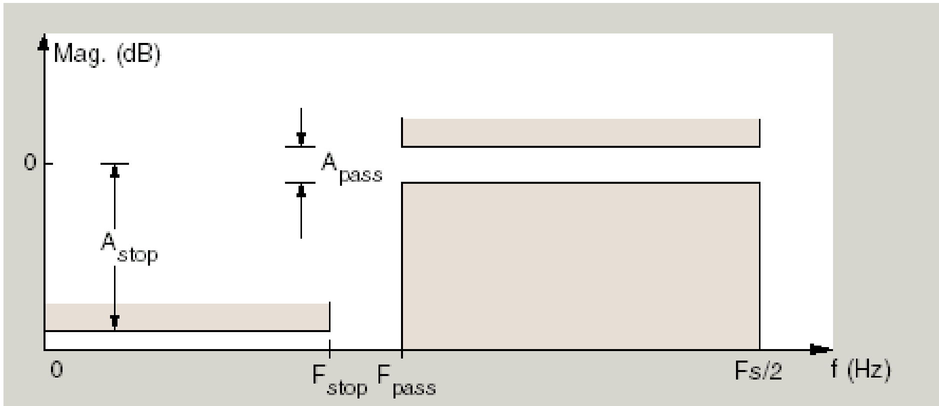
- `fst,fp,ast,ap` (default `spectype`)
- `n,fc`
- `n,fp,ap`
- `n,fst,ast`
- `n,fp,ast,ap`
- `n,fst,fp,ap`

The string entries are defined as follows:

- `ap`—amount of ripple allowed in the pass band in dB (the default units). Also called Apass.
- `ast`—attenuation in the stop band in dB (the default units). Also called Astop.
- `fc`—cutoff frequency for the point 3dB point below the passband value. Specified in normalized frequency units.
- `fp`—frequency at the start of the pass band. Specified in normalized frequency units. Also called Fpass.
- `fst`—frequency at the end of the stop band. Specified in normalized frequency units. Also called Fstop.

- n—filter order.

Graphically, the filter specifications look like this:



Regions between specification values like fst1 and fp are transition regions where the filter response is not explicitly defined.

The filter design methods that apply to a highpass filter design object change depending on the SpecificationType string. Paired with each string in the following table are the design methods for highpass filter design objects that use that string.

SpecificationType String	Applicable Design Method
fst,fp,ast,ap	butter, cheby1, cheby2, ellip
n,fc	butter
n,fp,ap	cheby1
n,fst,ast	cheby2
n,fp,ast,ap	ellip
n,fst,fp,ap	ellip

`d = fdesign.highpass(spectype,spec1,spec2,...)` constructs an object `d` and sets its specification values at construction time.

`d = fdesign.highpass(fst,fp,ast,ap)` constructs an object `d` with the values for the default '`SpecificationType`' property string, using the specifications you provide as input arguments.

`d = fdesign.highpass(...,fs)` adds the argument `fs`, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

`d = fdesign.highpass(...,magunits)` specifies the units for any magnitude specification you provide in the input arguments. `magunits` can be one of

- `linear`—specify the magnitude in linear units
- `dB`—specify the magnitude in dB (decibels)
- `squared`—specify the magnitude in power units

When you omit the `magunits` argument, `fdesign` assumes that all magnitudes are in dB. Note that `fdesign` stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

As you change the specifications, the `ResponseType` property changes as well. You see the `ResponseType` property when you get the filter object information.

```
get(d)
             ResponseType: 'Minimum-order highpass'
              Description: {4x1 cell}
        SpecificationType: 'Fst,Fp,Ast,Ap'
      NormalizedFrequency: 1
                       Fs: 'Normalized'
                    Fstop: 0.4500
                    Fpass: 0.5500
                    Astop: 60
                    Apass: 1
```

Examples

These examples how to construct a highpass filter design object. First, create a default without using input arguments.

```
d=fdesign.highpass
```

```
d =

             ResponseType: 'Minimum-order highpass'
        SpecificationType: 'Fst,Fp,Ast,Ap'
              Description: {4x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
                    Fstop: 0.4500
                    Fpass: 0.5500
                    Astop: 60
                    Apass: 1
```

This time, pass the specifications that correspond to the default `SpecificationType` string.

```
hs = fdesign.highpass(.4,.5,80,1);

hs =

             ResponseType: 'Minimum-order highpass'
        SpecificationType: 'Fst,Fp,Ast,Ap'
              Description: {4x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
                    Fstop: 0.4000
                    Fpass: 0.5000
                    Astop: 80
                    Apass: 1
```

Now create an object by passing a specification type string `'n,fc'`—the resulting object uses default values for `n` and `fc`.

```
d=fdesign.highpass('n,fc')

d =

             ResponseType: 'Highpass with cutoff'
        SpecificationType: 'N,Fc'
              Description: {2x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
```

```
              FilterOrder: 10
                  Fcutoff: 0.5000
```

Create the same filter, passing the values for `n` and `fc` rather than accepting the default values. Notice that you can add include the sampling frequency `fs` as the final input argument. Adding fs puts all the frequency specifications into linear frequency format, rather than normalized frequency.

```
d=fdesign.highpass('n,fc',10,9600,48000)

d =

             ResponseType: 'Highpass with cutoff'
        SpecificationType: 'N,Fc'
              Description: {2x1 cell}
      NormalizedFrequency: false
                       Fs: 48000
              FilterOrder: 10
                  Fcutoff: 9600
```

Finally, pass values for the filter specifications that match the default `SpecificationType` string—`fp = 10`, `fst = 12`, `ast = 80` and `ap = 0.5`. Add the sampling frequency on the end.

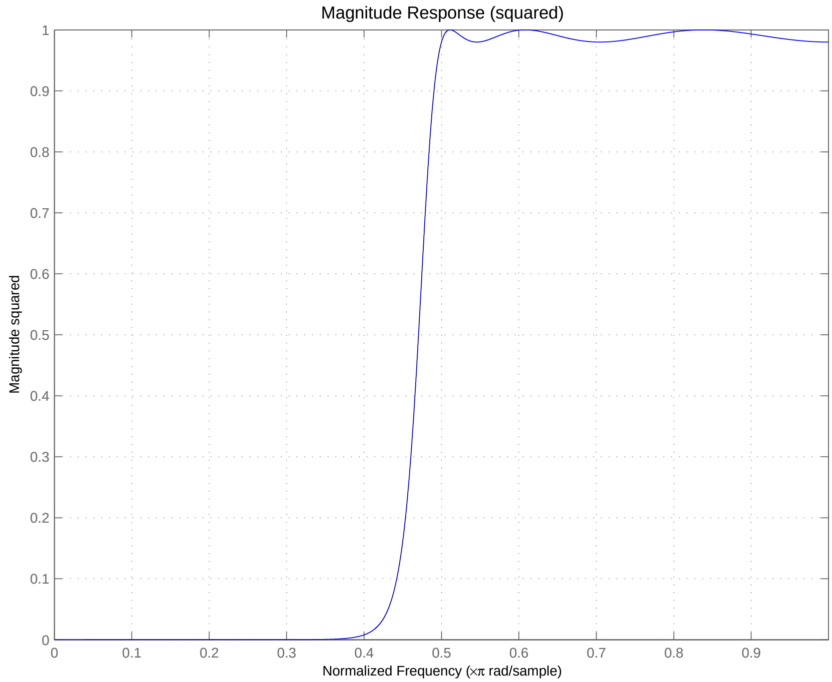
```
d=fdesign.highpass(10,12,80,0.5,48000)

d =

             ResponseType: 'Minimum-order highpass'
        SpecificationType: 'Fst,Fp,Ast,Ap'
              Description: {4x1 cell}
      NormalizedFrequency: false
                       Fs: 48000
                    Fstop: 10
                    Fpass: 12
                    Astop: 80
```

To demonstrate the `magunits` input option, pass the magnitude specifications in squared units and include the `squared` input argument for `magunits`.

```
hs = fdesign.highpass(.4, .5, .02, .98, 'squared');
hd = cheby1(hs);
```

```
fvtool(hd,'MagnitudeDisplay','Magnitude Squared');
```

The figure below show the filter response.



See Also

`fdesign`, `fdesign.bandpass`, `fdesign.bandstop`, `fdesign.lowpass`

fdesign.interp

Purpose Construct a filter design object that has interpolator filter specifications

Syntax

```
d = fdesign.interp(l)
d = fdesign.interp(l,spectype)
d = fdesign.interp(l,spectype,specvalue1,specvalue2,  )
d = fdesign.interp(l,transitionwidth,astop)
d = fdesign.interp(...,fs)
d = fdesign.interp(...,magunits)
```

Description

`d = fdesign.interp(l)` constructs an interpolating filter design object `d`, applying default values for the properties `tw` and `ast`. Specify `l`, the interpolation factor as an integer. When you omit the input argument `l`, `fdesign.interp` sets the interpolation factor `l` to 2. By default, the filter object designs a minimum-order halfband filter.

Using `fdesign.interp` with a design method generates an `mfilt` object.

`d = fdesign.interp(l,`*`spectype`*`)` constructs object `d` and sets its '`SpecificationType`' to *`spectype`*. Entries in the `spectype` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for *`spectype`* are shown below. The strings are not case sensitive. Notice that the interpolation factor `l` is not in the specification strings.

- `tw`,`ast` (default `spectype`)
- `pl`,`tw`
- `pl`
- `pl`,`ast`

The string entries are defined as follows:

- `ast`—attenuation in the stop band in dB (the default units).
- `pl`—polyphase filter length. Polyphase length is the length of each polyphase subfilter that composes the interpolator. Total filter length is the product of `pl` and the interpolation factor. `pl` must be an even integer.
- `tw`—width of the transition region between the pass and stop bands. Specified in normalized frequency units.

The filter design methods that apply to an interpolating filter design object change depending on the `SpecificationType` string. Paired with each string in the following table are the design methods for interpolating filter design objects that use that string.

SpecificationType String	Applicable Design Method
tw,ast	equiripple, kaiserwin
pl,tw	kaiserwin
pl	window
pl,ast	kaiserwin

d = fdesign.interp(l,spectype,spec1,spec2,...) constructs an object d and sets its specifications at construction time.

d = fdesign.interp(l,'tw,ast') constructs an object d with the default 'SpecificationType' property string, using the default specification values.

d = fdesign.interp(...,fs) adds the argument fs, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

d = fdesign.interp(...,magunits) specifies the units for any magnitude specification you provide in the input arguments. magunits can be one of

- linear—specify the magnitude in linear units
- dB—specify the magnitude in dB (decibels)
- squared—specify the magnitude in power units

When you omit the magunits argument, fdesign assumes that all magnitudes are in dB. Note that fdesign stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

As you change the specifications, the ResponseType property changes as well. You see the ResponseType property when you get the filter object information.

```
get(d)

         ResponseType: 'Minimum-order halfband'
    SpecificationType: 'TW,Ast'
```

```
          Description: {2x1 cell}
  InterpolationFactor: 2
  NormalizedFrequency: true
                   Fs: 'Normalized'
      TransitionWidth: 0.1000
                Astop: 80
```

Examples

These examples show how to construct an interpolator filter design object. First, create a default object without using input arguments.

```
d = fdesign.interp
d =

         ResponseType: 'Minimum-order halfband'
    SpecificationType: 'TW,Ast'
          Description: {2x1 cell}
  InterpolationFactor: 2
  NormalizedFrequency: true
                   Fs: 'Normalized'
      TransitionWidth: 0.1000
                Astop: 80
```

Now create an object by passing a specification type string 'n,ast'—the resulting object uses default values for n and ast.

```
d=fdesign.interp(5,'pl,ast')

d =

         ResponseType: [1x50 char]
    SpecificationType: 'PL,Ast'
          Description: {2x1 cell}
  InterpolationFactor: 5
  NormalizedFrequency: true
                   Fs: 'Normalized'
      PolyphaseLength: 24
                Astop: 80


d.responseType
```

```
ans =

Nyquist with filter order and stopband attenuation
```

Create another interpolating filter object, passing the specification values to the object rather than accepting the default values for `n` and `ast`.

```
d = fdesign.interp(4, 'pl,ast', 42, 80)

d =

          ResponseType: 'Nyquist with filter order and stopband attenuation'
     SpecificationType: 'Pl,Ast'
           Description: {2x1 cell}
   InterpolationFactor: 4
   NormalizedFrequency: true
                    Fs: 'Normalized'
       PolyphaseLength: 42
                 Astop: 80
```

Now pass the filter specifications that correspond to the default `SpecificationType`—`n,ast`.

```
d = fdesign.interp(5, .01, 80)

d =

          ResponseType: 'Minimum-order nyquist'
     SpecificationType: 'TW,Ast'
           Description: {2x1 cell}
   InterpolationFactor: 5
   NormalizedFrequency: true
                    Fs: 'Normalized'
       TransitionWidth: 0.0100
                 Astop: 80
```

Use `d` to design an interpolator using the `kaiserwin` design method.

```
hm=kaiserwin(d)

hm =

        FilterStructure: 'Direct-Form FIR Polyphase Interpolator'
              Numerator: [1x1010 double]
    InterpolationFactor: 5
```

```
    PersistentMemory: false
              States: [201x1 double]
NumSamplesProcessed: 0
```

Finally, include the sampling frequency input argument when you create the interpolator design object. Specify the `magunits` option with the value `linear`. In this case, you enter the attenuation in linear units—1e-3.

```
hs = fdesign.interp(4,'tw,ast',0.1,1e-3,5,'linear') %
1e-3 = 60dB
```

See Also

`fdesign`, `fdesign.decim`, `fdesign.halfband`, `fdesign.nyquist`, `fdesign.src`

Purpose Construct a filter design object that has lowpass filter specifications

Syntax

```
d = fdesign.lowpass
d = fdesign.lowpass(spectype)
d = fdesign.lowpass(spectype,specvalue1,specvalue2,  )
d = fdesign.lowpass(fp,fst,ap,ast)
d = fdesign.lowpass(...,fs)
d = fdesign.lowpass(...,magunits)
```

Description

`d = fdesign.lowpass` constructs a bandstop filter design object `d`, applying default values for the properties `fp`, `fst`, `ap`, and `ast`.

Using `fdesign.lowpass` with a design method generates a `dfilt` object.

`d = fdesign.lowpass(`*`spectype`*`)` constructs object `d` and sets its '`SpecificationType`' to *`spectype`*. Entries in the `spectype` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for *`spectype`* are shown below. The strings are not case sensitive.

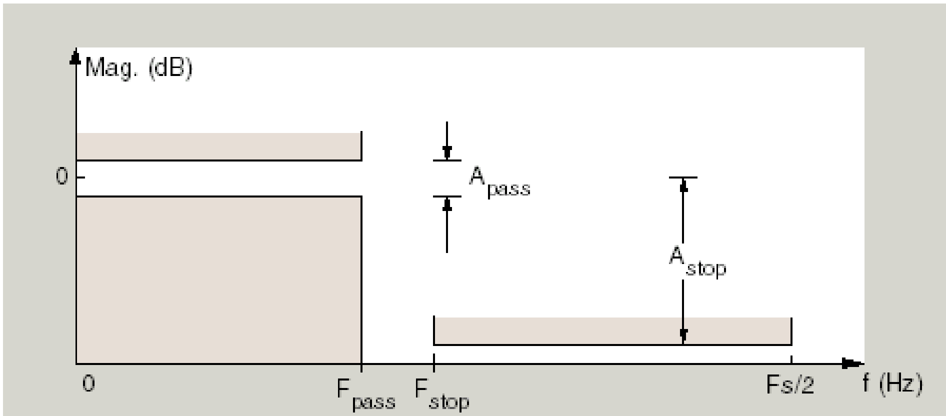
- `fp,fst,ap,ast` (default `spectype`)
- `n,fc`
- `n,fp,ap`
- `n,fst,ast`
- `n,fp,ap,ast`
- `n,fp,fst,ap`

The string entries are defined as follows:

- `ap`—amount of ripple allowed in the pass band in dB (the default units). Also called Apass.
- `ast`—attenuation in the stop band in dB (the default units). Also called Astop.
- `fc`—cutoff frequency for the point 3dB point below the passband value. Specified in normalized frequency units.
- `fp`—frequency at the start of the pass band. Specified in normalized frequency units. Also called Fpass.
- `fst`—frequency at the end of the stop band. Specified in normalized frequency units. Also called Fstop.

- n—filter order.

Graphically, the filter specifications look like this:



Regions between specification values like `fp` and `fst` are transition regions where the filter response is not explicitly defined.

The filter design methods that apply to a lowpass filter design object change depending on the `SpecificationType` string. Paired with each string in the following table are the design methods for lowpass filter design objects that use that string.

SpecificationType String	Applicable Design Method
`fp,fst,ap,ast`	`butter`, `cheby1`, `cheby2`, `ellip`
`n,fc`	`butter`
`n,fp,ap`	`cheby1`
`n,fst,ast`	`cheby2`
`n,fp,ap,ast`	`ellip`
`n,fp,fst,ap`	`ellip`

`d = fdesign.lowpass(`*`spectype`*`,spec1,spec2,...)` constructs an object `d` and sets its specification values at construction time.

`d = fdesign.lowpass(fp,fst,ap,ast)` constructs an object `d` with values for the default '`SpecificationType`' property string options, using the specifications you provide as input arguments.

`d = fdesign.lowpass(...,fs)` adds the argument `fs`, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

`d = fdesign.lowpass(...,magunits)` specifies the units for any magnitude specification you provide in the input arguments. `magunits` can be one of

- `linear`—specify the magnitude in linear units
- `dB`—specify the magnitude in dB (decibels)
- `squared`—specify the magnitude in power units

When you omit the `magunits` argument, `fdesign` assumes that all magnitudes are in dB. Note that `fdesign` stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

As you change the specifications, the `ResponseType` property changes as well. You see the `ResponseType` property when you get the filter object information.

```
get(d)
            ResponseType: 'Minimum-order lowpass'
             Description: {4x1 cell}
       SpecificationType: 'Fp,Fst,Ap,Ast'
     NormalizedFrequency: 1
                      Fs: 'Normalized'
                   Fpass: 0.4500
                   Fstop: 0.5500
                   Apass: 1
                   Astop: 60
```

Examples

These examples how to construct a lowpass filter design object. First, create a default lowpass filter object without using input arguments.

```
d=fdesign.lowpass
```

```
d =

             ResponseType: 'Minimum-order lowpass'
        SpecificationType: 'Fp,Fst,Ap,Ast'
              Description: {4x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
                    Fpass: 0.4500
                    Fstop: 0.5500
                    Apass: 1
                    Astop: 60
```

Now create an object by passing specifications for the passband and stopband edge frequencies and the passband and stopband attenuations—the resulting object uses the input values for `fp`, `fst`, `ap`, and `ast`.

```
hs = fdesign.lowpass(.4,.5,1,80);
hs

hs =

             ResponseType: 'Minimum-order lowpass'
        SpecificationType: 'Fp,Fst,Ap,Ast'
              Description: {4x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
                    Fpass: 0.4000
                    Fstop: 0.5000
                    Apass: 1
                    Astop: 80
```

Create another filter object, passing the values for `n` and `fc` rather than accepting the default values. Notice that you can add include the sampling frequency `fs` as the final input argument.

```
d=fdesign.lowpass('n,fc',10, 9600,48000)

d =

             ResponseType: 'Lowpass with cutoff'
        SpecificationType: 'N,Fc'
```

```
             Description: {2x1 cell}
     NormalizedFrequency: false
                      Fs: 48000
             FilterOrder: 10
                 Fcutoff: 9600
```

Finally, pass values for the filter specifications that match the default `SpecificationType` string entries—`fp = 0.4`, `fst = 0.5`, `ast = 80` and `ap = 1.0`. Add the sampling frequency on the end.

```
hs = fdesign.lowpass(.4,.5,1,80)

hs =

            ResponseType: 'Minimum-order lowpass'
       SpecificationType: 'Fp,Fst,Ap,Ast'
             Description: {4x1 cell}
     NormalizedFrequency: true
                      Fs: 'Normalized'
                   Fpass: 0.4000
                   Fstop: 0.5000
                   Apass: 1
                   Astop: 80
```

Finally, the next examples add the sampling frequency specification in Hz, and then the `magunits` option.

```
hs = fdesign.lowpass('N,Fp,Ap', 10, 9600, .5, 48000);
```

and

```
hs = fdesign.lowpass(.4, .5, .98, .02, 'squared');
```

Using the last example filter object, create a highpass filter.

```
hd = cheby1(hs);
```

See Also

`fdesign`, `fdesign.bandpass`, `fdesign.bandstop`, `fdesign.highpass`

fdesign.nyquist

Purpose Construct a filter design object that has Nyquist filter specifications

Syntax

```
d = fdesign.nyquist
d = fdesign.nyquist(spectype)
d = fdesign.nyquist(spectype,specvalue1,specvalue2,  )
d = fdesign.nyquist('tw,ast')
d = fdesign.nyquist(...,fs)
d = fdesign.nyquist(...,magunits)
```

Description

`d = fdesign.nyquist` constructs a Nyquist filter design object `d`, applying default values for the properties `tw` and `ast`. By default, the filter object designs a minimum-order Nyquist filter.

Using `fdesign.nyquist` with a design method generates a `dfilt` object.

`d = fdesign.nyquist(`*spectype*`)` constructs object `d` and sets its '`SpecificationType`' to *spectype*. Entries in the `spectype` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for *spectype* are shown below. The strings are not case sensitive.

- `tw,ast` (default `spectype`)
- `n,tw`
- `n`
- `n,ast`

The string entries are defined as follows:

- `ast`—attenuation in the stop band in dB (the default units).
- `n`—filter order.
- `tw`—width of the transition region between the pass and stop bands. Specified in normalized frequency units.

The filter design methods that apply to an interpolating filter design object change depending on the `SpecificationType` string. Paired with each string in the following table are the design methods for interpolating filter design objects that use that string.

SpecificationType String	Applicable Design Method
tw,ast	kaiserwin
n,tw	kaiserwin
n	window
n,ast	kaiserwin

`d = fdesign.nyquist(`*`spectype`*`,spec1,spec2,...)` constructs an object `d` and sets its specifications at construction time.

`d = fdesign.nyquist('tw,ast')` constructs an object `d` with the default '`SpecificationType`' property string, using the default specification values.

`d = fdesign.nyquist(...,fs)` adds the argument `fs`, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

`d = fdesign.nyquist(...,magunits)` specifies the units for any magnitude specification you provide in the input arguments. `magunits` can be one of

- `linear`—specify the magnitude in linear units
- `dB`—specify the magnitude in dB (decibels)
- `squared`—specify the magnitude in power units

When you omit the `magunits` argument, `fdesign` assumes that all magnitudes are in dB. Note that `fdesign` stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

As you change the specifications, the `ResponseType` property changes as well. You see the `ResponseType` property when you get the filter object information.

```
get(d)

          ResponseType: 'Minimum-order halfband'
     SpecificationType: 'TW,Ast'
```

```
          Description: {2x1 cell}
  InterpolationFactor: 2
  NormalizedFrequency: true
                   Fs: 'Normalized'
      TransitionWidth: 0.1000
                Astop: 80
```

Examples

These examples show how to construct a Nyquist filter design object. First, create a default object without using input arguments.

```
d=fdesign.nyquist

d =

         ResponseType: 'Minimum-order nyquist'
    SpecificationType: 'TW,Ast'
          Description: {'Stopband Attenuation (dB)'}
                 Band: 2
  NormalizedFrequency: true
                   Fs: 'Normalized'
      TransitionWidth: 0.1000
                Astop: 80
```

Now create an object by passing a specification type string 'n,ast'—the resulting object uses default values for n and ast.

```
d=fdesign.nyquist('n,ast')

d =

         ResponseType: 'Nyquist with filter order and stopband attenuation'
    SpecificationType: 'N,Ast'
          Description: {'Stopband Attenuation (dB)'}
                 Band: 2
  NormalizedFrequency: true
                   Fs: 'Normalized'
          FilterOrder: 10
                Astop: 80
```

Create another Nyquist filter object, passing the specification values to the object rather than accepting the default values for n and ast.

```
d=fdesign.nyquist('n,ast',42,80)

d =
```

```
      ResponseType: 'Nyquist with filter order and stopband attenuation'
 SpecificationType: 'N,Ast'
       Description: {'Stopband Attenuation (dB)'}
              Band: 2
NormalizedFrequency: true
                Fs: 'Normalized'
       FilterOrder: 42
             Astop: 80
```

Finally, pass the filter specifications that correspond to the default `SpecificationType`—`tw,ast`. When you pass only the values, `fdesign.nyquist` assumes the default `SpecificationType` string.

```
hs = fdesign.nyquist(4,.01, 80)

hs =

           ResponseType: 'Minimum-order nyquist'
      SpecificationType: 'TW,Ast'
            Description: {2x1 cell}
                   Band: 4
    NormalizedFrequency: true
                     Fs: 'Normalized'
        TransitionWidth: 0.0100
                  Astop: 80
```

Now design a Nyquist filter using the `kaiserwin` design method.

```
hd=kaiserwin(hs)

hd =

        FilterStructure: 'Direct-Form FIR'
             Arithmetic: 'double'
              Numerator: [1x103 double]
       PersistentMemory: false
                 States: [102x1 double]
    NumSamplesProcessed: 0
```

See Also

`fdesign`, `fdesign.decim`, `fdesign.halfband`, `fdesign.interp`, `fdesign.src`

fdesign.src

Purpose

Construct a filter design object that has sample rate converter filter specifications

Syntax

```
d = fdesign.src(l,m)
d = fdesign.src(l,m,spectype)
d = fdesign.src(l,m,spectype,specvalue1,specvalue2,...)
d = fdesign.src(l,m,'tw,ast')
d = fdesign.src(...,fs)
d = fdesign.src(...,magunits)
```

Description

`d = fdesign.src(l,m)` constructs a sample rate change filter design object `d`, applying default values for the properties `tw` and `ast`. Specify `l`, the interpolation factor as an integer. When you omit the input argument `l`, `fdesign.src` sets the interpolation factor `l` to 3. When you omit `m`, the decimation factor, `fdesign.src` sets `m` to 2. By default, the filter object designs a minimum-order Nyquist filter to perform the fractional sample rate change.

Using `fdesign.src` with a design method generates an `mfilt` object.

`d = fdesign.src(l,m,`*`spectype`*`)` constructs object `d` and sets its '`SpecificationType`' to *`spectype`*. Entries in the `spectype` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for *`spectype`* are shown below. The strings are not case sensitive. Notice that the interpolation factor `l` and the decimation factor `m` are not in the specification strings.

- `tw,ast` (default `spectype`)
- `pl,tw`
- `pl`
- `pl,ast`

The string entries are defined as follows:

- `ast`—attenuation in the stop band in dB (the default units).
- `pl`—polyphase filter length. Polyphase length is the length of each polyphase subfilter that composes the sample rate changer. Total filter length is whichever of the following is larger—the product of `pl` and the decimation factor, or the product of `pl` and the interpolation factor. `pl` must be an even integer.

- `tw`—width of the transition region between the pass and stop bands. Specified in normalized frequency units.

The filter design methods that apply to an interpolating filter design object change depending on the `SpecificationType` string. Paired with each string in the following table are the design methods for interpolating filter design objects that use that string.

SpecificationType String	Applicable Design Method
`tw,ast`	`kaiserwin`
`pl,tw`	`kaiserwin`
`pl`	`window`
`pl,ast`	`kaiserwin`

`d = fdesign.src(l,m,spectype,spec1,spec2,...)` constructs an object `d` and sets its specifications at construction time.

`d = fdesign.src(l,m,'tw,ast')` constructs an object `d` with the default `SpecificationType` property string, using the default specification values as well.

`d = fdesign.src(...,fs)` adds the argument `fs`, specified in Hz to define the sampling frequency to use. In this case, all frequencies in the specifications are in Hz as well.

`d = fdesign.src(...,magunits)` specifies the units for any magnitude specification you provide in the input arguments. `magunits` can be one of

- `linear`—specify the magnitude in linear units
- `dB`—specify the magnitude in dB (decibels)
- `squared`—specify the magnitude in power units

When you omit the `magunits` argument, `fdesign` assumes that all magnitudes are in dB. Note that `fdesign` stores all magnitude specifications in dB (converted to dB if necessary) regardless of how you specify the magnitudes.

As you change the specifications, the `ResponseType` property changes as well. You see the `ResponseType` property when you get the filter object information.

```
get(d)

           ResponseType: 'Minimum-order halfband'
      SpecificationType: 'TW,Ast'
            Description: {2x1 cell}
    InterpolationFactor: 2
    NormalizedFrequency: true
                     Fs: 'Normalized'
        TransitionWidth: 0.1000
                  Astop: 80
```

Examples

These examples show how to construct a filter design object that changes the sample rate of an input signal. First, create a default object without using input arguments.

```
d=fdesign.src

d =

           ResponseType: 'Minimum-order nyquist'
      SpecificationType: 'TW,Ast'
            Description: {2x1 cell}
    InterpolationFactor: 3
       DecimationFactor: 2
    NormalizedFrequency: true
                     Fs: 'Normalized'
        TransitionWidth: 0.1000
                  Astop: 80
```

Now create an object by passing the specifications that match the default type string 'tw,ast'—the resulting object uses your input values for `tw` and `ast`. In the absence of a `SpecificationType` string, `fdesign.src` assumes the default string and assigns the input arguments accordingly.

```
hs = fdesign.src(5, 3, .05, 40)

hs =
```

```
        ResponseType: 'Minimum-order nyquist'
   SpecificationType: 'TW,Ast'
         Description: {2x1 cell}
 InterpolationFactor: 5
    DecimationFactor: 3
 NormalizedFrequency: true
                  Fs: 'Normalized'
     TransitionWidth: 0.0500
               Astop: 40
```

Create another sample rate changing filter object, passing the specification values to the object rather than accepting the default values for `pl`. By not entering both `pl` and `ast` you let `fdesign` use the default `ast` value while using the filter polyphase length you provided.

```
hs = fdesign.src(2, 3,'pl,ast',12)

hs =

            ResponseType: [1x50 char]
       SpecificationType: 'PL,Ast'
             Description: {2x1 cell}
     InterpolationFactor: 2
        DecimationFactor: 3
     NormalizedFrequency: true
                      Fs: 'Normalized'
         PolyphaseLength: 12
                   Astop: 80
```

For this example, pass the sampling frequency input argument to `fdesign.src`.

```
hs = fdesign.src(3, 2,'pl,tw',14,.1,5)

hs =

            ResponseType: [1x46 char]
       SpecificationType: 'PL,TW'
             Description: {2x1 cell}
     InterpolationFactor: 3
        DecimationFactor: 2
```

```
     NormalizedFrequency: false
                      Fs: 5
         PolyphaseLength: 14
         TransitionWidth: 0.1000
```

Now design a sample rate changer using the `kaiserwin` design method.

```
hm=kaiserwin(hs)
Warning: Filter order is too low. Design may be inaccurate.

hm =

         FilterStructure: [1x47 char]
               Numerator: [1x42 double]
       RateChangeFactors: [3 2]
        PersistentMemory: false
                  States: [13x1 double]
     NumSamplesProcessed: 0
```

Notice that MATLAB warns that your filter may not meet your requirements. Changing such features as the polyphase length or the transition width can help eliminate the warning and improve the filter design.

You can specify the stopband ripple in linear units, rather than in dB, by including the `magunits` input option set to `linear`, and providing the ripple decimal form.

```
hs = fdesign.src(4, 7,'tw,ast',.1,1e-3,5,'linear') % 1e-3 = 60dB

hs =

            ResponseType: 'Minimum-order nyquist'
       SpecificationType: 'TW,Ast'
             Description: {2x1 cell}
     InterpolationFactor: 4
        DecimationFactor: 7
     NormalizedFrequency: false
                      Fs: 5
         TransitionWidth: 0.1000
                   Astop: 60
```

See Also

`fdesign`, `fdesign.decim`, `fdesign.halfband`, `fdesign.interp`, `fdesign.nyquist`

fftcoeffs

Purpose

Return the frequency-domain coefficients used when filtering with discrete-time and adaptive filter objects

Syntax

```
c = fftcoeffs(hd)
c = fftcoeffs(ha)
```

Description

`c = fftcoeffs(hd)` Return the frequency-domain coefficients used when filtering with the `dfilt.fftfir` object. c contains the coefficients

`c = fftcoeffs(ha)` Return the frequency-domain coefficients used when filtering with `adaptfilt` objects.

`fftcoeffs` applies to the following adaptive filter algorithms:

- `adaptfilt.fdaf`
- `adaptfilt.pbfdaf`
- `adaptfilt.pbufdaf`
- `adaptfilt.ufdaf`

Examples

This example demonstrates returning the FFT coefficients from the discrete-time filter `hd`.

```
b = [0.05 0.9 0.05];
len = 50;
hd = dfilt.fftfir(b,len)

hd =

        FilterStructure: 'Overlap-Add FIR'
              Numerator: [0.0500 0.9000 0.0500]
            BlockLength: 50
    NonProcessedSamples: []
       PersistentMemory: false

c=fftcoeffs(hd)

c =

  Columns 1 through 4

   1.0000             0.9920 - 0.1204i   0.9681 - 0.2386i   0.9289 - 0.3523i

  Columns 5 through 8

   0.8753 - 0.4594i   0.8084 - 0.5580i   0.7297 - 0.6464i   0.6408 - 0.7233i

  Columns 9 through 12
```

```
  0.5435 - 0.7874i   0.4398 - 0.8381i   0.3317 - 0.8747i   0.2211 - 0.8971i

 Columns 13 through 16

  0.1099 - 0.9054i        0 - 0.9000i  -0.1070 - 0.8815i  -0.2097 - 0.8506i

 Columns 17 through 20

 -0.3066 - 0.8084i  -0.3967 - 0.7558i  -0.4790 - 0.6939i  -0.5528 - 0.6240i

 Columns 21 through 24

 -0.6176 - 0.5472i  -0.6730 - 0.4645i  -0.7185 - 0.3771i  -0.7541 - 0.2860i

 Columns 25 through 28

 -0.7796 - 0.1921i  -0.7949 - 0.0965i  -0.8000            -0.7949 + 0.0965i

 Columns 29 through 32

 -0.7796 + 0.1921i  -0.7541 + 0.2860i  -0.7185 + 0.3771i  -0.6730 + 0.4645i

 Columns 33 through 36

 -0.6176 + 0.5472i  -0.5528 + 0.6240i  -0.4790 + 0.6939i  -0.3967 + 0.7558i

 Columns 37 through 40

 -0.3066 + 0.8084i  -0.2097 + 0.8506i  -0.1070 + 0.8815i        0 + 0.9000i

 Columns 41 through 44

  0.1099 + 0.9054i   0.2211 + 0.8971i   0.3317 + 0.8747i   0.4398 + 0.8381i

 Columns 45 through 48

  0.5435 + 0.7874i   0.6408 + 0.7233i   0.7297 + 0.6464i   0.8084 + 0.5580i

 Columns 49 through 52

  0.8753 + 0.4594i   0.9289 + 0.3523i   0.9681 + 0.2386i   0.9920 + 0.1204i
```

Similarly, you can use fftcoeffs with the adaptive filters algorithms listed above. Start by constructing an adaptive filter `ha`.

```
d  = 16;                      % Number of samples of delay.
b  = exp(j*pi/4)*[-0.7 1];  % Numerator coefficients of channel.
a  = [1 -0.7];             % Denominator coefficients of channel.
ntr= 1000;                    % Number of iterations.
```

```
s  = sign(randn(1,ntr+d)) +...
j*sign(randn(1,ntr+d));     % Baseband QPSK signal.
n  = 0.1*(randn(1,ntr+d) + j*randn(1,ntr+d));  % Noise signal.
r  = filter(b,a,s)+n;       % Received signal.
x  = r(1+d:ntr+d);          % Input signal (received signal).
d  = s(1:ntr);             % Desired signal (delayed QPSK signal).
del = 1;                    % Initial FFT input powers.
mu  = 0.1;                  % Step size.
lam = 0.9;                  % Averaging factor.
d   = 8;                    % Block size.
ha = adaptfilt.pbufdaf(32,mu,1,del,lam,n);
```

Here are the coefficients before you filter a signal.

```
c=fftcoeffs(ha)

c =

  Columns 1 through 13

    0    0    0    0    0    0    0    0    0    0    0    0    0
    0    0    0    0    0    0    0    0    0    0    0    0    0
    0    0    0    0    0    0    0    0    0    0    0    0    0
    0    0    0    0    0    0    0    0    0    0    0    0    0

  Columns 14 through 16

     0     0     0
     0     0     0
     0     0     0
     0     0     0
```

Filtering a signal y produces complex nonzero coefficients that you use `fftcoeffs` to see.

```
[y,e] = filter(ha,x,d);
c=fftcoeffs(ha)

c =

  Columns 1 through 4

   0.1425 - 0.0957i   0.0487 - 0.0503i  -0.0479 + 0.0315i   0.0769 - 0.0435i
```

```
 0.7264 - 0.7605i  -0.7423 - 0.6382i   0.1758 + 0.6679i   0.2018 - 0.6544i
-0.1604 + 0.0747i  -0.0709 + 0.2610i  -0.1634 + 0.2929i  -0.1488 + 0.3610i
-0.0396 + 0.0416i   0.0985 + 0.0095i   0.0733 + 0.0011i   0.0700 + 0.0348i

Columns 5 through 8

-0.0604 + 0.1767i   0.0732 - 0.0648i  -0.0870 + 0.0383i   0.0298 - 0.0852i
-0.1665 + 0.3741i   0.3174 - 0.5234i  -0.1990 + 0.4150i   0.3657 - 0.4760i
-0.2198 + 0.4273i  -0.2690 + 0.3981i  -0.2820 + 0.3095i  -0.3633 + 0.3517i
-0.0537 - 0.0855i  -0.0190 + 0.0336i   0.0091 - 0.0061i  -0.0299 + 0.0001i

Columns 9 through 12

-0.0437 + 0.0676i   0.0499 - 0.0164i  -0.0397 + 0.0165i   0.0455 - 0.0085i
-0.3293 + 0.3076i   0.4986 - 0.3949i  -0.3300 + 0.3448i   0.5492 - 0.2633i
-0.2671 + 0.3238i  -0.3813 + 0.2999i  -0.4130 + 0.2333i  -0.2910 + 0.2823i
-0.0300 + 0.0236i  -0.0103 + 0.0438i   0.0244 + 0.0476i   0.1043 + 0.0359i

Columns 13 through 16

-0.0602 + 0.1189i  -0.0227 - 0.1076i  -0.0282 + 0.0634i   0.0170 - 0.0464i
-0.4385 + 0.0549i   0.5232 - 0.1904i  -0.6414 - 0.1717i   0.5580 + 0.6477i
-0.4511 + 0.3217i  -0.4301 + 0.1765i  -0.2805 + 0.1270i  -0.2531 + 0.0299i
 0.1076 - 0.0383i  -0.0166 + 0.0020i   0.0004 - 0.0376i   0.0071 - 0.0714i
```

See Also

`adaptfilt.fdaf`, `adaptfilt.pbfdaf`, `adaptfilt.pbufdaf`, `adaptfilt.ufdaf`

filtmsb

Purpose

Return the most significant bit (MSB) of a cascaded integrator-comb (CIC) filter

Syntax

`filtmsb(hm)`

Description

`filtmsb(hm)` returns the most significant bit (MSB) of the filter output and is a function of the parameters

- R—the interpolation factor
- M—the differential delay
- N—the number of stages in the filter
- `InputBitWidth`—the width of the input in bits

of filter `hm`. Since the output of the integrators can grow without bound, the MSB returned represents the maximum number of bits that can propagate through the filter without losing data. This MSB is not only the MSB at the filter output; it is also the MSB for all stages.

Examples

Using the `mfilt.cicinterp` filter constructor, create a multirate filter and determine the most significant bit.

```
hm=mfilt.cicinterp

hm =

        FilterStructure: 'Cascaded Integrator-Comb Interpolator'
      DifferentialDelay: 1
       NumberOfSections: 2
          InputBitWidth: 16
         OutputBitWidth: 16
    InterpolationFactor: 2
       PersistentMemory: false
                 States: [2x2 int32]
    NumSamplesProcessed: 0

filtmsb(hm)

ans =
```

```
18
```

See Also `gain`, `mfilt`

Purpose Apply filter objects to data and access states and filtering information

Syntax

Fixed-Point Filter Syntaxes

```
y = filter(hq,x)
[y,zf] = filter(hq,x)
[...] = filter(hq,x,zi)
[...] = filter(hq,x,zi,dim)
[y,zf,s,z,v] = filter(hq,x...)
```

Adaptive Filter Syntax

```
[y,e] = filter(ha,x,d)
```

Multirate Filter Syntax

```
[y] = filter(hm,x,dim)
```

Description This reference page contains three sections that describe the syntaxes for the filter objects:

- Fixed-Point Filter Syntaxes
- “Adaptive Filter Syntaxes” on page 9-568
- “Multirate Filter Syntaxes” on page 9-568

Fixed-Point Filter Syntaxes

`y = filter(hq,x)` filters a vector of real or complex input data `x` through a fixed-point filter `hq`, producing filtered output data `y`. The vectors `x` and `y` have the same length.

If `x` is a matrix, `y = filter(hq,x)` filters each column of `x` to produce a matrix `y`. If `x` is a multidimensional array, `y = filter(hq,x)` filters `x` along the first nonsingleton dimension of `x`.

`[y,zf] = filter(hq,x)` produces an additional output argument `zf`. `zf` contains the final values for the state vector calculated from zero initial conditions for the state. The form `zf` takes depends on the data to be filtered and the number of stages in the filter, as detailed in Table 9-3, Final State Form Depends on Filtered Data and Filter Structure.

`[...] = filter(hq,x,zi)` specifies the initial conditions for the state vector in `zi`. The form for specifying `zi` is described in Table 9-2, Initial State Format Depends on the Filter Structure. To specify the same initial condition for all state components, enter `zi` as a scalar. You can set `zi` to zero, `[]`, or `{}` to specify zero (the default) initial conditions.

The form of the initial and final states associated with a fixed-point filter `hq` depends on the filter structure and the data to be filtered. The following tables give the form for either entering the initial states or retrieving the final states of the quantized filter.

Table 9-2: Initial State Format Depends on the Filter Structure

Number of Filter Sections	Format of the Initial State
1	A column vector of length s_1
n	A 1-by-n cell array of vectors of length s_i, i=1, 2,...,n

Table 9-3: Final State Form Depends on Filtered Data and Filter Structure

Filtered Data	Number of Filter Sections	Form of the Final State
Vector	1	A column vector of length s_1
Vector	n	A 1-by-n cell array of vectors of length s_i, i=1, 2,...,n
Multidimensional array	1	An s_1-by-c matrix
Multidimensional array	n	1-by-n cell array of s_i-by-c matrices, i=1, 2,...,n

The variables in these tables are described as follows:

- s_i is the number of states in the ith section of the filter.
- c is `prod(size(x))/size(x,dim)`, where `dim` is the first nonsingleton dimension into which you are filtering.

To figure out the dimensions of the initial or final conditions, run the filter once with empty initial conditions (IC). After you run with empty ICs, the final conditions are the right size for the initial conditions:

```
[y,zf] = filter(hq,x);
```

Look at the size and data type of `zf`. The initial conditions, `zi`, will be the same size as `zf`.

`[...] = filter(hq,x,zi,dim)` applies the quantized filter `hq` to the input data located along the specific dimension of `x` specified by `dim`. When you provide the `dim` input argument, the filter operates along the dimension specified by `dim`. When your input data `x` is a vector or matrix and `dim` is 1, each column of `x` is treated as a one input channel. When `dim` is 2, the filter treats each row of the input `x` as a channel.

`[y,zf,s,z,v] = filter(hq,x...)` returns `s`, a MATLAB structure containing quantization information; `z`, the filter state sequence; and `v`, the number of overflows at each time step of the filter. When you include four or five output arguments, the input argument `x` must be a vector. `z` is a cell array containing the sequence of states at each time step, having 1 element per filter and 1 column per time step. The initial conditions of the k-th filter section are in the first column of `z{k}:zi{k}=z{k}(:,1)`. The final conditions of the k-th filter section are in the last column of `z{k}:zf{k} = z{k}(:,end)`. Overflows for the k-th section are in `v{k}`.

Adaptive Filter Syntaxes

`[y,e] = filter(ha,x,d)` filters a vector of real or complex input data `x` through an adaptive filter object `ha`, producing the estimated desired response data `y` and the prediction error `e`, both resulting from the process of adapting the filter. The vectors `x` and `y` have the same length. Use `d` for the desired signal. Note that `d` and `x` must be the same length signal chains.

Multirate Filter Syntaxes

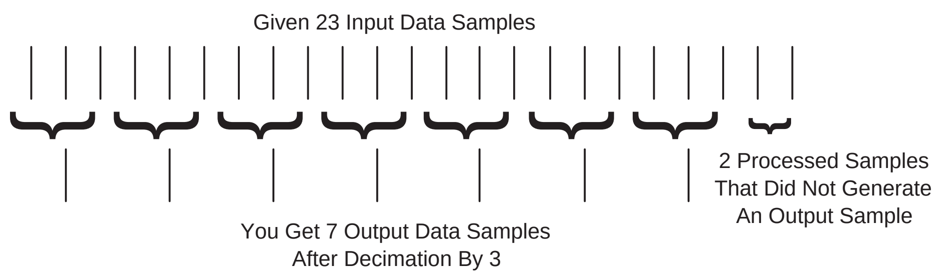
`[y] = filter(hm,x,dim)` filters a vector of real or complex input data `x` through a multirate filter object `hm`, returning `y`, the results of the filtering operation.

When you are filtering multichannel data, dim lets you specify which dimension of the input matrix to filter along—whether a row represents a channel or a column represents a channel. When you provide the dim input argument, the filter operates along the dimension specified by dim. When your input data x is a vector or matrix and dim is 1, each column of x is treated as a one input channel. When dim is 2, the filter treats each row of the input x as a channel.

To filter multichannel data in a loop environment, you must use the dim input argument to set the processing dimension.

You specify the initial conditions for each channel individually, when needed, by setting hm.states to a matrix of nstates(hm) rows (one row containing the states for one channel of input data) and size(x,2) columns (one column containing the filter states for each channel).

The number of data samples in your input data set x does not need to be a multiple of the rate change factor r for the object. When the rate change factor is not an even divisor of the number of input samples x, filter processes the samples as shown in the following figure, where the rate change factor is 3 and the number of input samples is 23. Decimators always take the first input sample to generate the first output sample. After that, the next output sample comes after each r number of input samples.



Examples

Filter a signal using a filter with various initial conditions (IC).

```
x = randn(100,1);        % Original signal.
b = fir1(50,.4);         % 50th-order linear-phase FIR filter.
hd = dfilt.dffir(b);     % Direct-form FIR implementation.

% Do not set specific initial conditions.
```

```
y1 = filter(hd,x);     % 'PersistentMemory' is 'false' (default).
zf = hd.States;         % Final conditions.

% Use nonzero initial conditions by setting ICs.

hd.persistentmemory = true;
hd.States = 1;          % Uses scalar expansion.
y2 = filter(hd,x);
stem([y1 y2])           % Different sequences at the beginning.

% Filter streaming data
reset(hd);              % Clear filter history.
y3 = filter(hd,x);      % Filter the entire signal in one block.
reset(hd);              % Clear filter history.
yloop = [];
xblock = reshape(x,[20 5]);

% Filtering the signal section-by-section is equivalent to
% filtering the entire signal at once.

for i=1:5,
yloop = [yloop; filter(hd,xblock(:,i))];
```

Algorithm

Quantized Filters

The `filter` command implements fixed- or floating-point arithmetic on the quantized filter structure you specify. The *state vector z* associated with the filter is a vector whose components are derived from the values of each of the input signals to each delay in the filter. The length of z is the same as the number of delays in the filter.

The implementation of `filter` depends on the filter structure. For example, the operation of `filter` at sample m for a direct-form II transposed filter is given by the quantized time domain difference equations for y and the states z_i

shown below. Square brackets denote the quantization that takes place for the input data x, the output data y, the coefficients, the products, and the sums.

$$
\begin{aligned}
y(m) &= \left[\frac{[[[b(1)][x(m)]] + z_1(m-1)]}{[a(1)]}\right] \\
z_1(m) &= [[[b(2)][x(m)]] + z_2(m-1) - [[a(2)]y(m)]] \\
\vdots &= \vdots \\
z_{n-2}(m) &= [[[[b(n-1)][x(m)]] + z_{n-1}(m-1)] - [[a(n-1)]y(m)]] \\
z_{n-1}(m) &= [[[b(n)][x(m)]] - [[a(n)]y(m)]]
\end{aligned}
$$

Notice that for this `df2t` filter structure, you divide by $a(1)$. For efficient computation, choose $a(1)$ to be a power of 2.

Note `dfilt/filter` does not normalize the filter coefficients automatically. Function `filter` supplied by MATLAB does normalize the coefficients.

Adaptive Filters

The algorithm used by `filter` when you apply an adaptive filter object to a signal depends on the algorithm you chose for your adaptive filter. To learn more about each adaptive filter algorithm, refer to the literature reference provided on the appropriate `adaptfilt.`*`algorithm`* reference page.

Multirate Filters

The algorithm applied by `filter` when you apply a multirate filter objects to signals depends on the algorithm you chose for the filter—the form of the multirate filter, such as decimator or interpolator. To learn more about each filter algorithm, refer to the literature reference provided on the appropriate multirate filter reference page.

See Also

`adaptfilt`, `impz`, `mfilt`, `nstates`
`dfilt` in the Signal Processing Toolbox

References

[1] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, 1989.

filtstates.cic

Purpose

Object for storing the states of CIC filters

Description

`filtstates.cic` objects hold the states information for CIC filters. Once you create a CIC filter, the states for the filter are stored in `filtstates.cic` objects, and you can access them and change them as you would any property of the filter. This arrangement parallels that of the `filtstates` object that IIR direct-form I filters use (refer to `filtstates` for more information).

Each `States` property in the CIC filter comprises two properties—`Numerator` and `Comb`—that hold `filtstates.cic` objects.Within the `filtstates.cic` objects are the numerator-related and comb-related filter states. The states are column vectors, where each column represents the states for one section of the filter. For example, a CIC filter with four decimator sections and four interpolator sections has `filtstates.cic` objects that contain four columns of states each.

Examples

To show you the filtstates.cic object, create a CIC decimator and filter a signal.

```
hm=mfilt.cicdecim(5,2,4)

hm =

          FilterStructure: 'Cascaded Integrator-Comb Decimator'
               Arithmetic: 'fixed'
        DifferentialDelay: 2
         NumberOfSections: 4
         DecimationFactor: 5
         PersistentMemory: false

          InputWordLength: 16
          InputFracLength: 15

    SectionWordLengthMode: 'MinWordLengths'

hm.persistentMemory=true

hm =

          FilterStructure: 'Cascaded Integrator-Comb Decimator'
               Arithmetic: 'fixed'
```

```
          DifferentialDelay: 2
           NumberOfSections: 4
           DecimationFactor: 5
           PersistentMemory: true
                     States: Integrator: [4x1 States]
                                   Comb: [4x1 States]
                InputOffset: 0

            InputWordLength: 16
            InputFracLength: 15

     SectionWordLengthMode: 'MinWordLengths'
```

Use hm to filter some input data.

```
fs = 44.1e3;             % Original sampling frequency: 44.1kHz.
n = 0:10239;           % 10240 samples, 0.232 second long signal.
x  = sin(2*pi*1e3/fs*n); % Original signal, sinusoid at 1kHz.
y=filter(hm,x)
```

hm has nonzero states now.

```
s=hm.states

s =

    Integrator: [4x1 States]
          Comb: [4x1 States]

s.Integrator

ans =

  1.0e+003 *

    0.0043
   -2.0347
   -0.4175
    0.8206

s.Comb
```

```
ans =

   1.0e+003 *

    -3.1301
    -0.8493
    -2.5474
     1.7888
    -1.6253
     3.1981
     0.4729
     3.4559
```

You can use int to see the states as 32-bit integers.

```
int(s.Integrator)

ans =

       142435
     -8334019
      -427469
       210081
```

whos shows you the `filtstates.cic` object.

```
whos
  Name      Size                      Bytes  Class

  Fs        1x1                           8  double array
  ans       4x1                          16  int32 array
  hm        1x1                              mfilt.cicdecim
  n         1x10240                   81920  double array
  s         1x1                              filtstates.cic
  x         1x10240                   81920  double array
  y         1x2048                           embedded.fi

Grand total is 20488 elements using 163864 bytes
```

See Also

`mfilt`, `mfilt.cicdecim`, `mfilt.cicinterp` `filtstates` in the Signal Processing Toolbox documentation

fircband

Purpose

Perform constrained-band equiripple FIR filter design

Syntax

```
b = fircband(n,f,a,w,c)
b = fircband(n,f,a,s)
b = fircband(...,'1')
b = fircband(...,'minphase')
b = fircband(..., 'check')
b = fircband(...,{lgrid})
[b,err] = fircband(...)
[b,err,res] = fircband(...)
```

Description

`fircband` is a minimax filter design algorithm that you use to design the following types of real FIR filters:

- Types 1-4 linear phase
  - Type 1 is even order, symmetric
  - Type 2 is odd order, symmetric
  - Type 3 is even order, antisymmetric
  - Type 4 is odd order, antisymmetric
- Minimum phase
- Maximum phase,
- Minimum order (even or odd), extra ripple
- Maximal ripple
- Constrained ripple
- Single-point band (notching and peaking)
- Forced gain
- Arbitrary shape frequency response curve filters

`b = fircband(n,f,a,w,c)` designs filters having constrained error magnitudes (ripples). `c` is a cell array of strings of the same length as `w`. The entries of `c` must be either `'c'` to indicate that the corresponding element in `w` is a constraint (the ripple for that band cannot exceed that value) or `'w'` indicating that the corresponding entry in `w` is a weight. There must be at least one unconstrained band—`c` must contain at least one **`w`** entry. For instance, Example 1 below uses a weight of one in the passband, and constrains the stopband ripple not to exceed 0.2 (about 14 dB).

A hint about using constrained values: if your constrained filter does not touch the constraints, increase the error weighting you apply to the unconstrained bands.

Notice that, when you work with constrained stopbands, you enter the stopband constraint according to the standard conversion formula for power—the resulting filter attenuation or constraint equals 20*log(*constraint*) where *constraint* is the value you enter in the function. For example, to set 20 dB of attenuation, use a value for the constraint equal to 0.1. This applies to constrained stopbands only.

`b = fircband(n,f,a,s)` is used to design filters with special properties at certain frequency points. `s` is a cell array of strings and must be the same length as `f` and `a`. Entries of `s` must be one of:

- `'`**`n`**`'`—normal frequency point.
- `'`**`s`**`'`—single-point band. The frequency band is given by a single point. You must specify the corresponding gain at this frequency point in `a`.
- `'`**`f`**`'`—forced frequency point. Forces the gain at the specified frequency band to be the value specified.
- `'`**`i`**`'`—indeterminate frequency point. Use this argument when bands abut one another (no transition region).

`b = fircband(...,'1')` designs a type 1 filter (even-order symmetric). You could also specify type 2 (odd-order symmetric), type 3 (even-order antisymmetric), or type 4 (odd-order antisymmetric) filters. Note there are restrictions on `a` at `f` = 0 or `f`=1 for types 2, 3, and 4.

`b = fircband(...,'`**`minphase`**`')` designs a minimum-phase FIR filter. There is also '**`maxphase`**'.

`b = fircband(..., '`**`check`**`')` produces a warning when there are potential transition-region anomalies in the filter response.

`b = fircband(...,{lgrid})`, where `{lgrid}` is a scalar cell array containing an integer, controls the density of the frequency grid.

[b,err] = fircband(...) returns the unweighted approximation error magnitudes. err has one element for each independent approximation error.

[b,err,res] = fircband(...) returns a structure res of optional results computed by fircband, and contains the following fields:.

Structure Field	Contents
res.fgrid	Vector containing the frequency grid used in the filter design optimization
res.des	Desired response on fgrid
res.wt	Weights on fgrid
res.h	Actual frequency response on the frequency grid
res.error	Error at each point (desired response - actual response) on the frequency grid
res.iextr	Vector of indices into fgrid of extremal frequencies
res.fextr	Vector of extremal frequencies
res.order	Filter order
res.edgecheck	Transition-region anomaly check. One element per band edge. Element values have the following meanings: 1 = OK 0 = probable transition-region anomaly -1 = edge not checked Computed when you specify the 'check' input option in the function syntax.

Structure Field	Contents
`res.iterations`	Number of Remez iterations for the optimization
`res.evals`	Number of function evaluations for the optimization

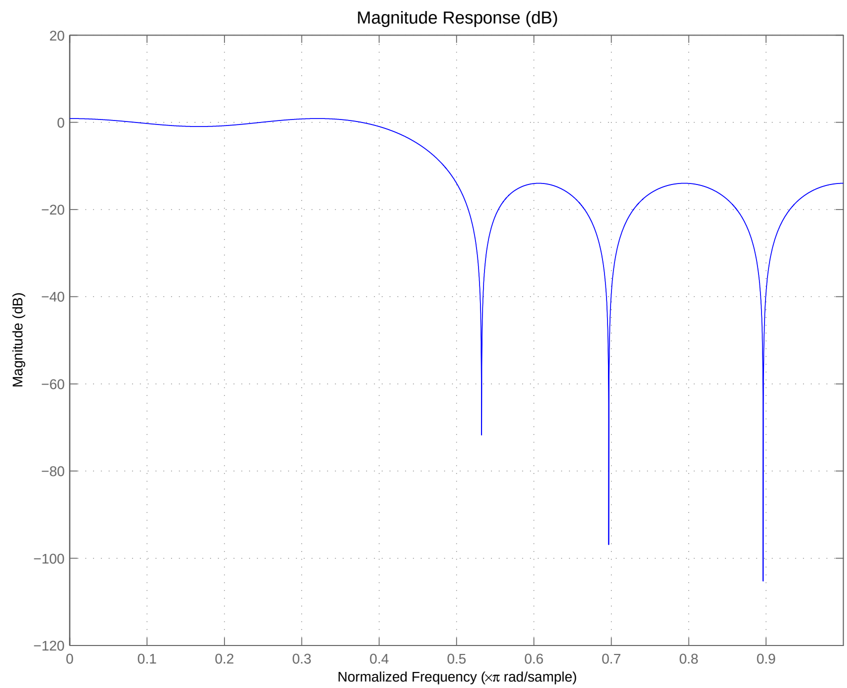
Examples

Two examples of designing filters with constrained bands.

Example 1—design a 12th-order lowpass filter with a constraint on the filter response.

```
b = fircband(12,[0 0.4 0.5 1], [1 1 0 0], [1 0.2], {'w' 'c'});
```

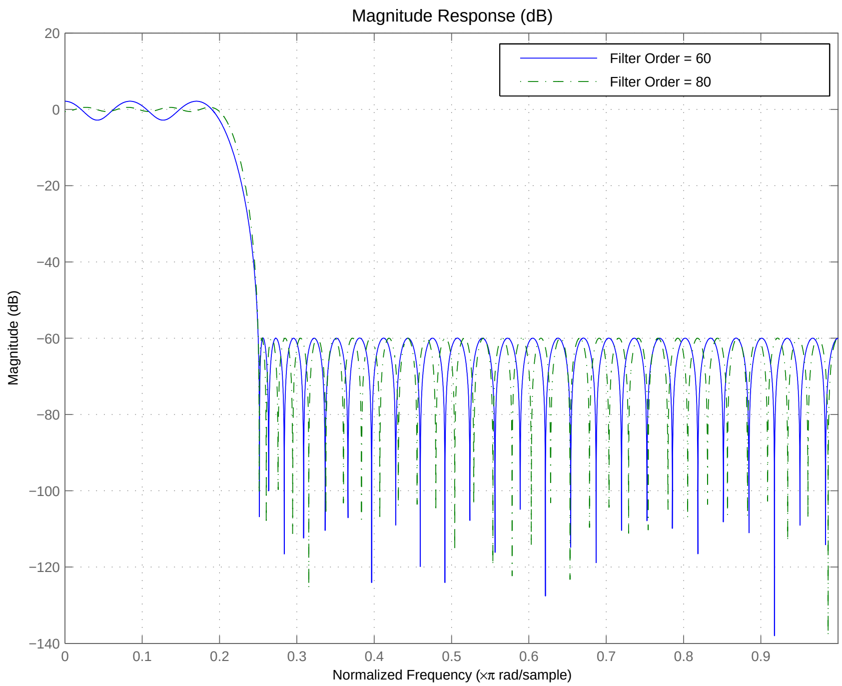
Using `fvtool` to display the result `b` shows you the response of the filter you designed.



Example 2—design two filters of different order with the stopband constrained to 60 dB. Use excess order (80) in the second filter to improve the passband ripple.

```
b1=fircband(60,[0 .2 .25 1],[1 1 0 0],[1 .001],{'w','c'});
b2=fircband(80,[0 .2 .25 1],[1 1 0 0],[1 .001],{'w','c'});
fvtool(b1,1,b2,1)
```

To set the stopband constraint to 60 dB, enter `0.001`, since 20*log(0.001) = -60, or 60 dB of signal attenuation.



See Also

`firceqrip`, `firgr`, `firls`

`firpm` in the Signal Processing Toolbox
Also refer to "Constrained Band Equiripple FIR Filter Design" in Demos

firceqrip

Purpose Design constrained, equiripple, finite impulse response (FIR) filters

Syntax

```
hd = firceqrip(n,wo,del)
hd = firceqrip(...,'slope',r)
hd = firceqrip(...,'passedge')
hd = firceqrip(...,'stopedge')
hd = firceqrip(...,'high')
hd = firceqrip(...,'min')
hd = firceqrip(...,'invsinc',c)
```

Description

`hd = firceqrip(n,wo,del)` design an order `n` filter (filter length equal `n+1`) lowpass FIR filter with linear phase.

`firceqrip` produces the same equiripple lowpass filters that `firpm` produces using the Parks-McClellan algorithm. The difference is how you specify the filter characteristics for the function.

Input argument `wo` specifies the cutoff frequency. The two-element vector `del` specifies the peak or maximum error allowed in the passband and stopbands. Enter `[d1 d2]` for `del` where `d1` sets the passband error and `d2` sets the stopband error. Since `firceqrip` works in the normalized frequency domain, you must set `wo` to be between 0 and 1 (0 < `wo` < 1).

`hd = firceqrip(...,'slope',r)` uses the input keyword '**slope**' and input argument `r` to design a filter with a stopband that does not demonstrate equiripple characteristics. `r` determines the slope of the stopband in dB when `r` > 0. Try setting `r` to `10` to see the effect on the filter frequency response. In the Examples section, Example 3 designs a filter with `r` equal to 20.

`hd = firceqrip(...,'passedge')` designs a filter where `wo` specifies the frequency at which the passband starts to roll off.

`hd = firceqrip(...,'stopedge')` designs a filter where `wo` specifies the frequency at which the stopband begins.

`hd = firceqrip(...,'high')` designs a high pass FIR filter instead of a lowpass filter.

`hd = firceqrip(...,'min')` designs an FIR filter with minimum phase.

hd = firceqrip(...,'**invsinc**',c)) designs a lowpass filter whose passband has the shape of the inverse sinc function. For this syntax, keyword **invsinc** applies the inverse sinc function as defined by whether c is a scalar or a two-element vector:

- When you use c as a scalar with the **invsinc** keyword, firceqrip applies the function 1/sinc(c*w), where w is the normalized frequency, to the passband.
- When you use c as a two-element vector entered as [c p], with the **invsinc** keyword, firceqrip applies the function $1/\text{sinc}(c*w)^p$ to the passband, where w is the normalized frequency.

In both cases, c must meet the condition $c < 1/w_o$.

When you use a cascaded-integrated comb (CIC) filter in series with this FIR filter, argument p lets you compensate for the droop in the passband of the CIC filter. Setting p equal to the number of stages in your CIC generally produces an FIR filter whose passband neatly compensates for the CIC passband shape.

To let you specify precisely the FIR filter to design, use any or all of the optional input arguments together. Any ordering of the optional arguments works—order is not important in the function call. Refer to Examples 3 and 4 to see multiple optional input arguments being used.

Note If the wo you specify is too small or too large, or if either c or p is too large, your filter specifications may be unfeasible, causing the design algorithm to fail to generate your filter.

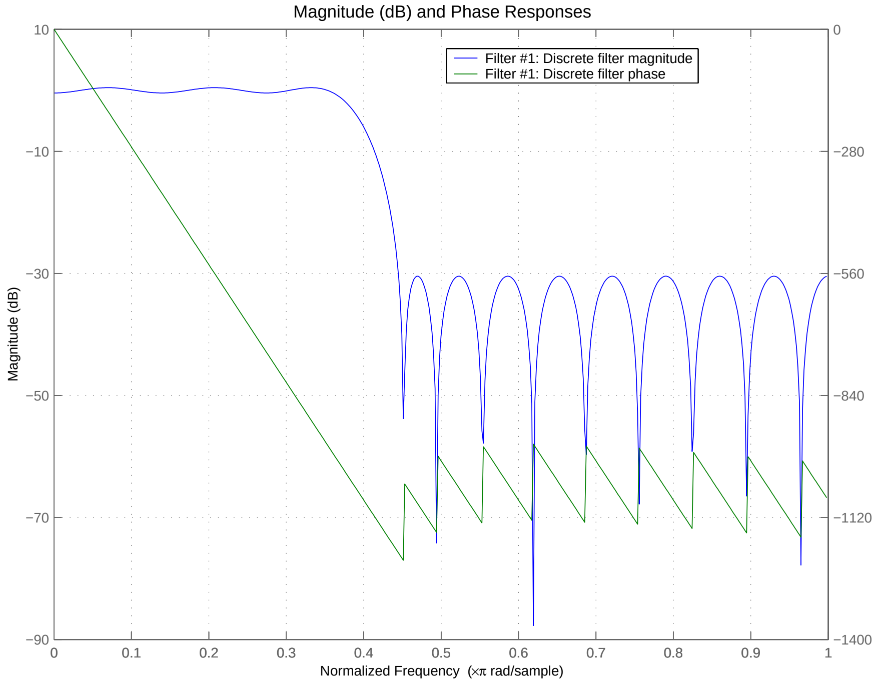
Examples

To introduce a few of the variations on FIR filters that you design with firceqrip, these five examples cover both the default syntax hd = firceqrip(n,wo,del) and some of the optional input arguments. For each example, the input arguments n, wo, and del remain the same.

Example 1—Design an order = 30 FIR filter without using optional input arguments or keywords.

```
hd = firceqrip(n,wo,del); fvtool(hd)
```

Both the phase and magnitude response for the resulting lowpass filter appear in the plot shown here.



Example 2—Design an order = 30 FIR filter with the **stopedge** keyword to define the response at the edge of the filter stopband.

```
hd = firceqrip(n,wo,del,'stopedge'); fvtool(hd,1)
```

Example 3—Design an order = 30 FIR filter with the **slope** keyword and r = 20.

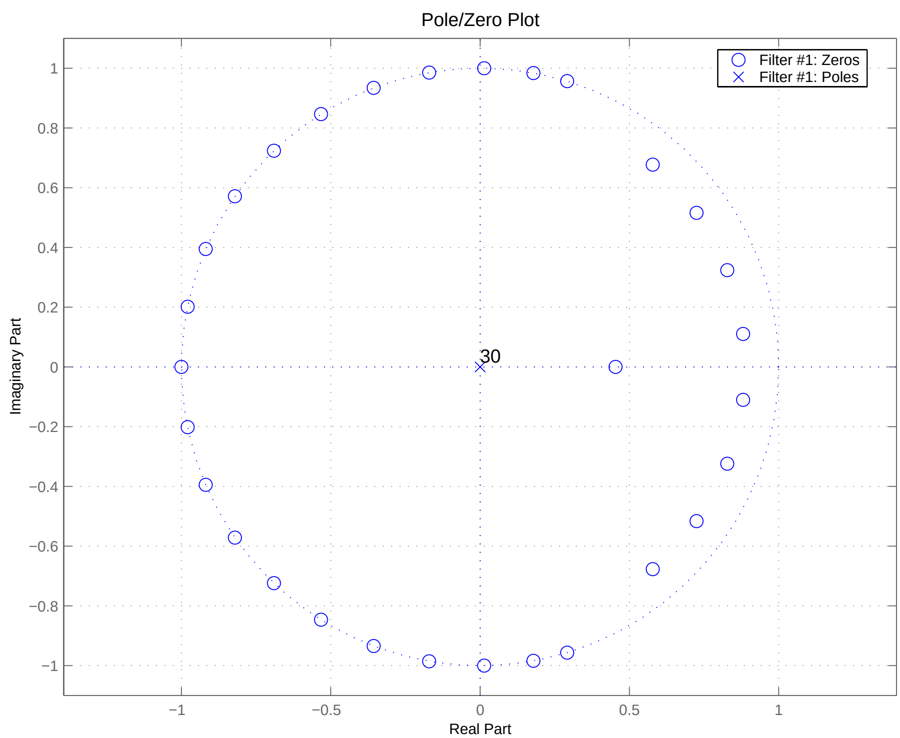
```
hd = firceqrip(n,wo,del,'slope',20,'stopedge'); fvtool(hd)
```

Example 4—Design an order = 30 FIR filter defining the stopband and specifying that the resulting filter is minimum phase with the **min** keyword.

```
hd = firceqrip(n,wo,del,'stopedge','min'); fvtool(hd)
```

Comparing this filter to the filter in Example 1, notice that the cutoff frequency `wo = 0.4` now applies to the edge of the stopband rather than the point at which the frequency response magnitude is 0.5.

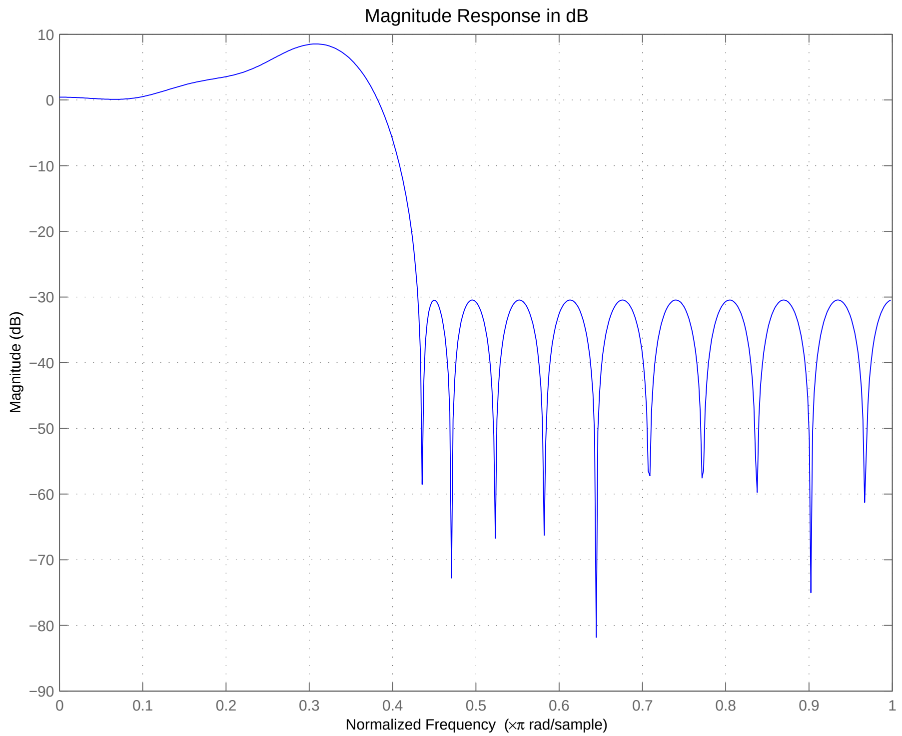
Viewing the zero-pole plot shown here reveals this is a minimum phase FIR filter—the zeros lie on or inside the unit circle, z = 1.



Example 5—Design an order = 30 FIR filter with the **`invsinc`** keyword to shape the filter passband with an inverse sinc function.

```
hd = firceqrip(n,wo,del,'invsinc',[2 1.5]); fvtool(hd,1)
```

With the inverse sinc function being applied defined as 1/sinc(2*w)$^{1.5}$, the figure shows the reshaping of the passband that results from using the **invsinc** keyword option, and entering c as the two-element vector [2 1.5].



See Also

firhalfband, firnyquist, firgr, ifir, iirgrpdelay, iirlpnorm, iirlpnormc

fircls, firls, firpm in your Signal Processing Toolbox documentation

Purpose

Use the Parks-McClellan technique to design digital FIR filters

Syntax

```
b = firgr(n,f,a,w)
b = firgr(n,f,a,'hilbert')
b = firgr(n,f,a,'differentiator')
b = firgr(m,f,a,r)
b = firgr({m,ni},f,a,r)
b = firgr(n,f,a,w,e)
b = firgr(n,f,a,s)
b = firgr(n,f,a,s,w,e)
```

Description

`firgr` is a minimax filter design algorithm you use to design the following types of real FIR filters:

- Types 1-4 linear phase:
  - Type 1 is even order, symmetric
  - Type 2 is odd order, symmetric
  - Type 3 is even order, antisymmetric
  - Type 4 is odd order, antisymmetric
- Minimum phase
- Maximum phase
- Minimum order (even or odd)
- Extra ripple
- Maximal ripple
- Constrained ripple
- Single-point band (notching and peaking)
- Forced gain
- Arbitrary shape frequency response curve filters

`b = firgr(n,f,a,w)` returns a length `n+1` linear phase FIR filter which has the best approximation to the desired frequency response described by `f` and `a` in the minimax sense. `w` is a vector of weights, one per band. When you omit `w`, all bands are weighted equally. For more information on the input arguments, refer to `firpm` in *Signal Processing Toolbox User's Guide*.

`b = firgr(n,f,a,'hilbert')` and `b = firgr(n,f,a,'differentiator')` design FIR Hilbert transformers and differentiators. For more information on designing these filters, refer to `firpm` in *Signal Processing Toolbox User's Guide*.

`b = firgr(m,f,a,r)`, where `m` is one of `'minorder'`, `'mineven'` or `'minodd'`, designs filters repeatedly until the minimum order filter, as specified in `m`, that meets the specifications is found. `r` is a vector containing the peak ripple per frequency band. You must specify `r`. When you specify `'mineven'` or `'minodd'`, the minimum even or odd order filter is found.

`b = firgr({m,ni},f,a,r)` where `m` is one of `'minorder'`, `'mineven'` or `'minodd'`, uses `ni` as the initial estimate of the filter order. `ni` is optional for common filter designs, but it must be specified for designs in which `firpmord` cannot be used, such as while designing differentiators or Hilbert transformers.

`b = firgr(n,f,a,w,e)` specifies independent approximation errors for different bands. Use this syntax to design extra ripple or maximal ripple filters. These filters have interesting properties such as having the minimum transition width. `e` is a cell array of strings specifying the approximation errors to use. Its length must equal the number of bands. Entries of `e` must be in the form `'e#'` where `#` indicates which approximation error to use for the corresponding band. For example, when `e = {'e1','e2','e1'}`, the first and third bands use the same approximation error `'e1'` and the second band uses a different one `'e2'`. Note that when all bands use the same approximation error, such as `{'e1','e1','e1',...}`, it is equivalent to omitting `e`, as in `b = firgr(n,f,a,w)`.

`b = firgr(n,f,a,s)` is used to design filters with special properties at certain frequency points. `s` is a cell array of strings and must be the same length as `f` and `a`. Entries of `s` must be one of:

- `'n'` - normal frequency point.
- `'s'` - single-point band. The frequency "band" is given by a single point. The corresponding gain at this frequency point must be specified in `a`.
- `'f'` - forced frequency point. Forces the gain at the specified frequency band to be the value specified.
- `'i'` - indeterminate frequency point. Use this argument when adjacent bands abut one another (no transition region).

For example, the following command designs a bandstop filter with zero-valued single-point stop bands (notches) at 0.25 and 0.55.

```
b = firgr(42,[0 0.2 0.25 0.3 0.5 0.55 0.6 1],[1 1 0 1 1 0 1 1],...
{'n' 'n' 's' 'n' 'n' 's' 'n' 'n'})
```

```
b = firgr(82,[0 0.055 0.06 0.1 0.15 1],[0 0 0 0 1 1],...
{'n' 'i' 'f' 'n' 'n' 'n'})
```

designs a highpass filter with the gain at 0.06 forced to be zero. The band edge at 0.055 is indeterminate since the first two bands actually touch. The other band edges are normal.

`b = firgr(n,f,a,s,w,e)` specifies weights and independent approximation errors for filters with special properties. The weights and properties are included in vectors `w` and `e`. Sometimes, you may need to use independent approximation errors to get designs with forced values to converge. For example,

```
b = firgr(82,[0 0.055 0.06 0.1 0.15 1], [0 0 0 0 1 1],...
{'n' 'i' 'f' 'n' 'n' 'n'}, [10 1 1] ,{'e1' 'e2' 'e3'});
```

`b = firgr(...,'1')` designs a type 1 filter (even-order symmetric). You can specify type 2 (odd-order symmetric), type 3 (even-order antisymmetric), and type 4 (odd-order antisymmetric) filters as well. Note that restrictions apply to `a` at f=0 or f=1 for FIR filter types 2, 3, and 4.

`b = firgr(...,'minphase')` designs a minimum-phase FIR filter. You can use the argument `'maxphase'` to design a maximum phase FIR filter.

`b = firgr(..., 'check')` returns a warning when there are potential transition-region anomalies.

`b = firgr(...,{lgrid})`, where `{lgrid}` is a scalar cell array. The value of the scalar controls the density of the frequency grid by setting the number of samples used along the frequency axis.

`[b,err] = firgr(...)` returns the unweighted approximation error magnitudes. `err` contains one element for each independent approximation error returned by the function.

[b,err,res] = firgr(...) returns the structure res comprising optional results computed by firgr. res contains the following fields.

Structure Field	Contents
res.fgrid	Vector containing the frequency grid used in the filter design optimization
res.des	Desired response on fgrid
res.wt	Weights on fgrid
res.h	Actual frequency response on the frequency grid
res.error	Error at each point (desired response - actual response) on the frequency grid
res.iextr	Vector of indices into fgrid of extremal frequencies
res.fextr	Vector of extremal frequencies
res.order	Filter order
res.edgecheck	Transition-region anomaly check. One element per band edge. Element values have the following meanings: 1 = OK 0 = probable transition-region anomaly -1 = edge not checked Computed when you specify the 'check' input option in the function syntax.
res.iterations	Number of s iterations for the optimization
res.evals	Number of function evaluations for the optimization

`firgr` is also a "function function", allowing you to write a function that defines the desired frequency response.

`b = firgr(n,f,fresp,w)` returns a length N+1 FIR filter which has the best approximation to the desired frequency response as returned by the user-defined function `fresp`. Use the following `firgr` syntax to call `fresp`:

```
[dh,dw] = fresp(n,f,gf,w)
```

where:

- `fresp` is the string variable that identifies the function that you use to define your desired filter frequency response.
- `n` is the filter order.
- `f` is the vector of frequency band edges which must appear monotonically between 0 and 1, where 1 is one-half of the sampling frequency. The frequency bands span `f(k)` to `f(k+1)` for `k` odd. The intervals `f(k+1)` to `f(k+2)` for `k` odd are "transition bands" or "don't care" regions during optimization.
- `gf` is a vector of grid points that have been chosen over each specified frequency band by `firgr`, and determines the frequencies at which `firgr` evaluates the response function.
- `w` is a vector of real, positive weights, one per band, for use during optimization. `w` is optional in the call to `firgr`. If you do not specify `w`, it is set to unity weighting before being passed to `fresp`.
- `dh` and `dw` are the desired frequency response and optimization weight vectors, evaluated at each frequency in grid `gf`.

`firgr` includes a predefined frequency response function named `'firpmfrf2'`. You can write your own based on the simpler `'firpmfrf'`. See the help for `private/firpmfrf` for more information.

`b = firgr(n,f,{`*`fresp`*`,p1,p2,...},w)` specifies optional arguments `p1`, `p2`,..., `pn` to be passed to the response function `fresp`.

`b = firgr(n,f,a,w)` is a synonym for `b = firgr(n,f,{'firpmfrf2',a},w)`, where `a` is a vector containing your specified response amplitudes at each band edge in `f`. By default, `firgr` designs symmetric (even) FIR filters. `'firpmfrf2'` is the predefined frequency response function. If you do not specify your own frequency response function (the `fresp` string variable), `firgr` uses 'firpmfrf2'.

`b = firgr(...,'h')` and `b = firgr(...,'d')` design antisymmetric (odd) filters. When you omit the `'h'` or `'d'` arguments from the `firgr` command syntax, each frequency response function `fresp` can tell `firgr` to design either an even or odd filter. Use the command syntax `sym = *fresp*('defaults',{n,f,[],w,p1,p2,...})`.

`firgr` expects `fresp` to return `sym = 'even'` or `sym = 'odd'`. If `fresp` does not support this call, `firgr` assumes even symmetry.

For more information about the input arguments to `firgr`, refer to `firpm`.

Examples

These examples demonstrate some filters you might design using `firgr`.

Example 1—design an FIR filter with two single-band notches at 0.25 and 0.55

```
b1 = firgr(42,[0 0.2 0.25 0.3 0.5 0.55 0.6 1],[1 1 0 1 1 0 1 1],...
{'n' 'n' 's' 'n' 'n' 's' 'n' 'n'});
```

Example 2—design a highpass filter whose gain at 0.06 is forced to be zero. The gain at 0.055 is indeterminate since it should abut the band.

```
b2 = firgr(82,[0 0.055 0.06 0.1 0.15 1],[0 0 0 0 1 1],...
{'n' 'i' 'f' 'n' 'n' 'n'});
```

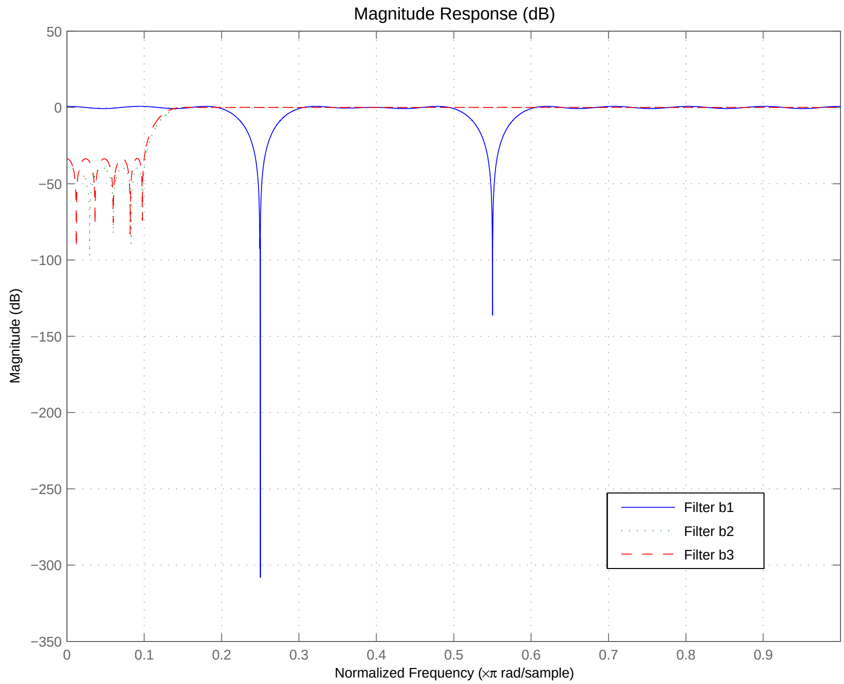
Example 3—design a second highpass filter with forced values and independent approximation errors.

```
b3 = firgr(82,[0 0.055 0.06 0.1 0.15 1], [0 0 0 0 1 1], ...
{'n' 'i' 'f' 'n' 'n' 'n'}, [10 1 1] ,{'e1' 'e2' 'e3'});
```

Use the filter visualization tool to view the results of the filters created in these examples.

```
fvtool(b1,1,b2,1,b3,1)
```

Here is the figure from fvtool.



See Also

`butter`, `cheby1`, `cheby2`, `ellip`, `freqz`, `filter`, `firls`, `fircls`, and `firpm` in your Signal Processing Toolbox documentation

Reference

Shpak, D.J. and A. Antoniou, “A generalized Remez method for the design of FIR digital filters,” *IEEE Trans. Circuits and Systems*, pp. 161-174,Feb. 1990.

firhalfband

Purpose

Design a halfband FIR filter

Syntax

```
b = firhalfband(n,fp)
b = firhalfband(n,win)
b = firhalfband(n,dev,'dev')
b = firhalfband('minorder',fp,dev)
b = firhalfband('minorder',fp,dev,'kaiser')
b = firhalfband(...,'high')
b = firhalfband(...,'minphase')
```

Description

`b = firhalfband(n,fp)` designs a lowpass halfband FIR filter of order `n` with an equiripple characteristic. `n` must be an even integer. `fp` determines the passband edge frequency, and it must satisfy 0 < `fp` < 1/2, where 1/2 corresponds to $\pi/2$ rad/sample.

`b = firhalfband(n,win)` designs a lowpass Nth-order filter using the truncated, windowed-impulse response method instead of the equiripple method. `win` is an `n+1` length vector. The ideal impulse response is truncated to length `n + 1`, and then multiplied point-by-point with the window specified in `win`.

`b = firhalfband(n,dev,'dev')` designs an Nth-order lowpass halfband filter with an equiripple characteristic. Input argument `dev` sets the value for the maximum passband and stopband ripple allowed.

`b = firhalfband('minorder',fp,dev)` designs a lowpass minimum-order filter, with passband edge `fp`. The peak ripple is constrained by the scalar `dev`. This design uses the equiripple method.

`b = firhalfband('minorder',fp,dev,'kaiser')` designs a lowpass minimum-order filter, with passband edge `fp`. The peak ripple is constrained by the scalar `dev`. This design uses the Kaiser window method.

`b = firhalfband(...,'high')` returns a highpass halfband FIR filter.

`b = firhalfband(...,'minphase')` designs a minimum-phase FIR filter such that the filter is a spectral factor of a halfband filter (recall that `h = conv(b,fliplr(b))` is a halfband filter). This can be useful for designing perfect reconstruction, two-channel FIR filter banks. The **minphase** option for

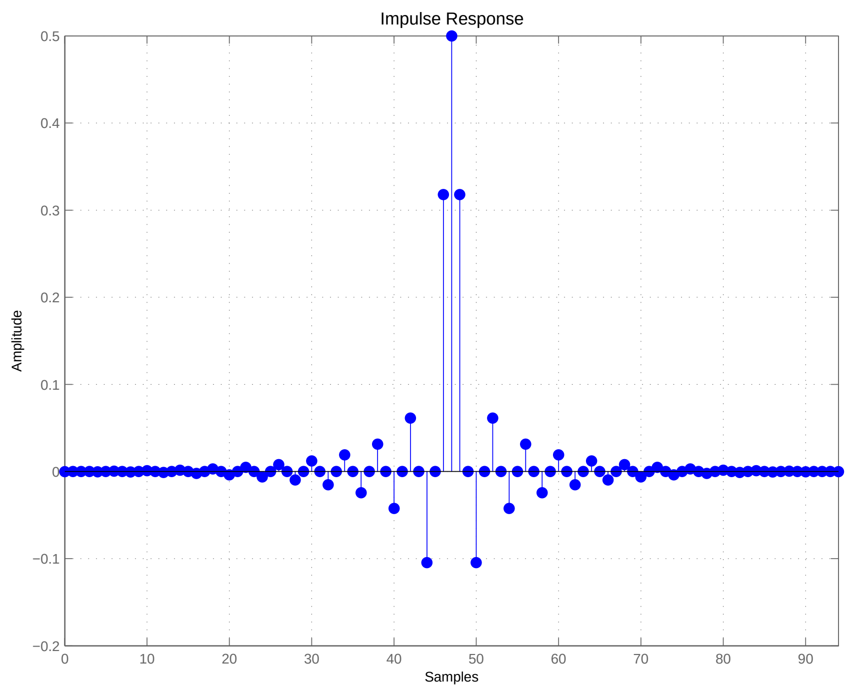
`firhalfband` is not available for the window-based halfband filter designs—
`b = firhalfband(n,win)` and
`b = firhalfband('minorder',fp,dev,'kaiser')`.

In the minimum phase cases, the filter order must be odd.

Examples

This example designs a minimum order halfband filter with specified maximum ripple:

```
b = firhalfband('minorder',.45,0.0001);
h = dfilt.dfsymfir(b);
impz(b) % Impulse response is zero for every other sample
```



The next example designs a halfband filter with specified maximum ripple of 0.0001 dB in the pass and stop bands.

```
b = firhalfband(98,0.0001,'dev');
h = mfilt.firdecim(2,b); % Create a polyphase decimator
freqz(h); % 80 dB attenuation in the stopband
```

See Also

`firnyquist`, `firgr`
`fir1`, `firls`, `firpm` in your Signal Processing Toolbox documentation

References

Saramaki, T, “Finite Impulse Response Filter Design,” *Handbook for Digital Signal Processing*. S.K. Mitra and J.F. Kaiser Eds. Wiley-Interscience, N.Y., 1993, Chapter 4.

Purpose

Convert FIR Type I lowpass to FIR Type 1 lowpass with inverse band width

Syntax

```
g = firlp2lp(b)
```

Description

`g = firlp2lp(b)` transforms the Type I lowpass FIR filter `b` with zero-phase response Hr(w) to a Type I lowpass FIR filter `g` with zero-phase response [1 - Hr(π-w)].

When `b` is a narrowband filter, `g` will be a wideband filter and vice versa. The passband and stopband ripples of `g` will be equal to the stopband and passband ripples of `b`.

Examples

Overlay the original narrowband lowpass and the resulting wideband lowpass

```
b = firgr(36,[0 .2 .25 1],[1 1 0 0],[1 5]);
zerophase(b);
hold on
h = firlp2lp(b);
zerophase(h); hold off
```

See Also

`firlp2hp`

`zerophase` in your Signal Processing Toolbox documentation

References

[1] Saramaki, T, Finite Impulse Response Filter Design, *Handbook for Digital Signal Processing*. S.K. Mitra and J.F. Kaiser Eds. Wiley-Interscience, N.Y., 1993, Chapter 4.

firlp2hp

Purpose

Convert FIR lowpass filter to Type I FIR highpass filter

Syntax

```
g = firlp2hp(b)
g  = firlp2hp(b,'narrow')
g = firlp2hp(b,'wide')
```

Description

`g = firlp2hp(b)` transforms the lowpass FIR filter `b` into a Type I highpass FIR filter `g` with zero-phase response Hr(π-w). Filter `b` can be any FIR filter, including a nonlinear-phase filter.

The passband and stopband ripples of `g` will be equal to the passband and stopband ripples of `b`.

`g  = firlp2hp(b,'narrow')` transforms the lowpass FIR filter `b` into a Type I narrow band highpass FIR filter `g` with zero-phase response Hr(π-w). `b` can be any FIR filter, including a nonlinear-phase filter.

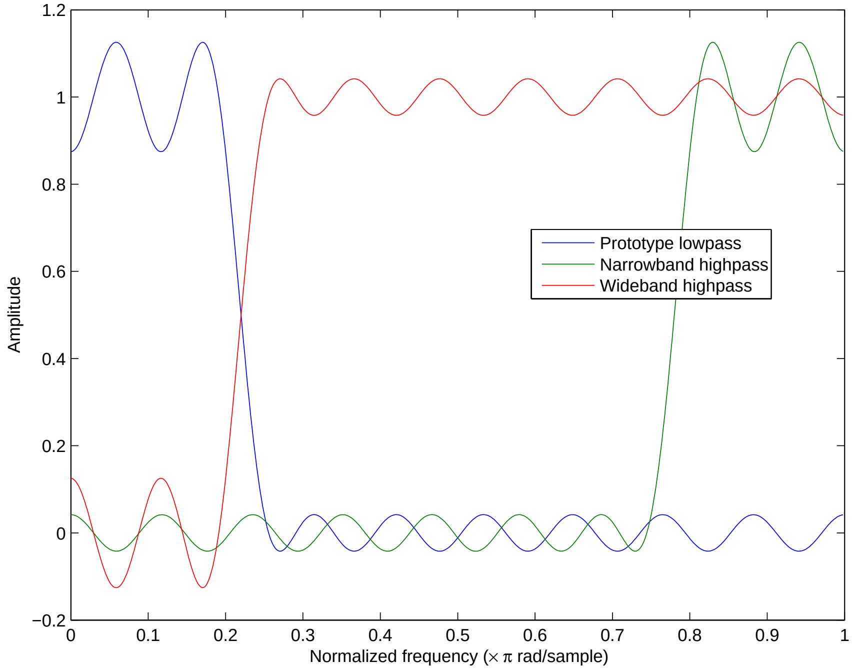
`g = firlp2hp(b,'wide')` transforms the Type I lowpass FIR filter `b` with zero-phase response Hr(w) into a Type I wide band highpass FIR filter `g` with zero-phase response 1 - Hr(w). Note the restriction that `b` must be a Type I linear-phase filter.

For this case, the passband and stopband ripples of `g` will be equal to the stopband and passband ripples of `b`.

Examples

Overlay the original narrowband lowpass (the prototype filter) and the post-conversion narrowband highpass and wideband highpass filters to compare and assess the conversion. The plot below shows the results.

```
b = firgr(36,[0 .2 .25 1],[1 1 0 0],[1 3]);
zerophase(b); hold on;
h = firlp2hp(b);
zerophase(h);
g = firlp2hp(b,'wide');
zerophase(g); hold off
```



See Also

`firlp2lp`

`zerophase` in your Signal Processing Toolbox documentation

References

[1] Saramaki, T, Finite Impulse Response Filter Design, *Handbook for Digital Signal Processing*. S.K. Mitra and J.F. Kaiser Eds. Wiley-Interscience, N.Y., 1993, Chapter 4.

firlpnorm

Purpose Least P-norm optimal FIR filter design

Syntax

```
b = firlpnorm(n,f,edges,a)
b = firlpnorm(n,f,edges,a,w)
b = firlpnorm(n,f,edges,a,w,p)
b = firlpnorm(n,f,edges,a,w,p,dens)
b = firlpnorm(n,f,edges,a,w,p,dens,initnum)
b = firlpnorm(...,'minphase')
[b,err] = firlpnorm(...)
```

Description

`b = firlpnorm(n,f,edges,a)` returns a filter of numerator order `n` which represents the best approximation to the frequency response described by `f` and `a` in the least-Pth norm sense. P is set to 128 by default, which essentially equivalent to the infinity norm. Vector `edges` specifies the band-edge frequencies for multiband designs. `firlpnorm` uses an unconstrained quasi-Newton algorithm to design the specified filter.

`f` and `a` must have the same number of elements, which can exceed the number of elements in `edges`. This lets you specify filters with any gain contour within each band. However, the frequencies in `edges` must also be in vector `f`. Always use `freqz` to check the resulting filter.

Note `firlpnorm` uses a nonlinear optimization routine that may not converge in some filter design cases. Furthermore the algorithm is not well-suited for certain large-order (order > 100) filter designs.

`b = firlpnorm(n,f,edges,a,w)` uses the weights in `w` to weight the error. `w` has one entry per frequency point (the same length as `f` and `a`) which tells `firlpnorm` how much emphasis to put on minimizing the error in the vicinity of each frequency point relative to the other points. For example,

```
b = firlpnorm(20,[0 .15 .4 .5 1],[0 .4 .5 1],...
[1 1.6 1 0 0],[1 1 1 10 10])
```

designs a lowpass filter with a peak of 1.6 within the passband, and with emphasis placed on minimizing the error in the stopband.

b = firlpnorm(n,f,edges,a,w,p) where p is a two-element vector [pmin pmax] lets you specify the minimum and maximum values of p used in the least-pth algorithm. Default is [2 128] which essentially yields the L-infinity, or Chebyshev, norm. pmin and pmax should be even numbers. The design algorithm starts optimizing the filter with pmin and moves toward an optimal filter in the pmax sense.When p is the string '**inspect**', firlpnorm does not optimize the resulting filter. You might use this feature to inspect the initial zero placement.

b = firlpnorm(n,f,edges,a,w,p,dens) specifies the grid density dens used in the optimization. The number of grid points is [dens*(n+1)]. The default is 20. You can specify dens as a single-element cell array. The grid is equally spaced.

b = firlpnorm(n,f,edges,a,w,p,dens,initnum) lets you determine the initial estimate of the filter numerator coefficients in vector initnum. This can prove helpful for difficult optimization problems. The pole-zero editor in the Signal Processing Toolbox can be used for generating initnum.

b = firlpnorm(...,'minphase') where string 'minphase' is the last argument in the argument list generates a minimum-phase FIR filter. By default, firlpnorm design mixed-phase filters. Specifying input option 'minphase' causes firlpnorm to use a different optimization method to design the minimum-phase filter. As a result of the different optimization used, the minimum-phase filter can yield slightly different results.

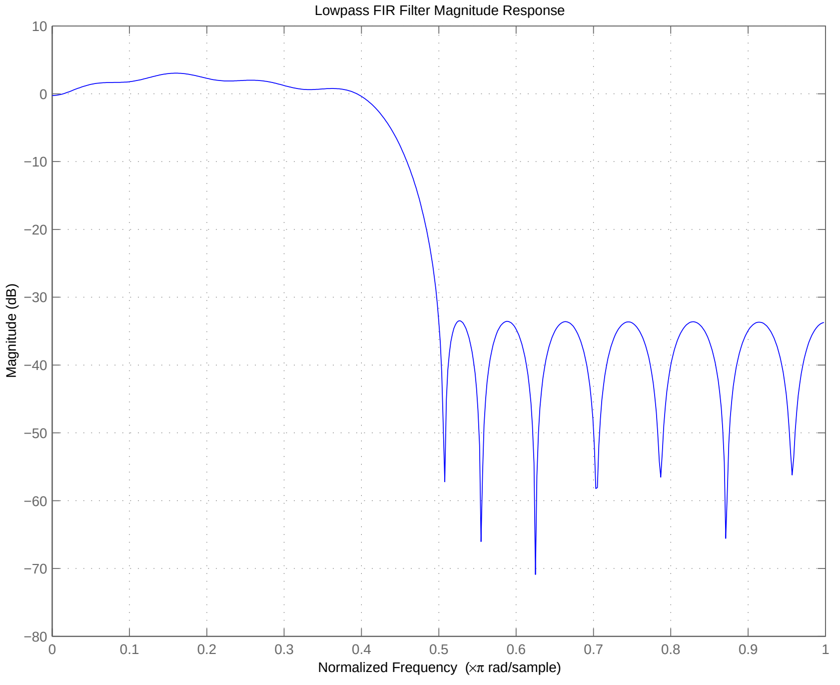
[b,err] = firlpnorm(...) returns the least-pth approximation error err.

Examples

To demonstrate firlpnorm, here are two examples — the first designs a lowpass filter and the second a highpass, minimum-phase filter.

```
% Lowpass filter with a peak of 1.4 in the passband.
b = firlpnorm(22,[0 .15 .4 .5 1],[0 .4 .5 1],[1 1.4 1 0 0],...
[1 1 1 2 2]);
fvtool(b)
```

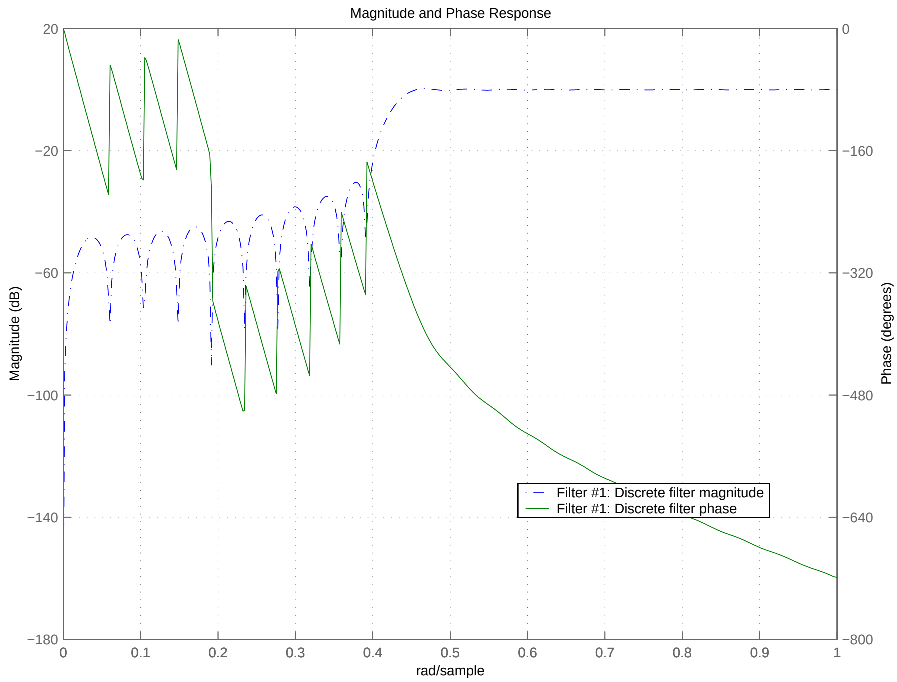
From the figure you see the resulting filter is lowpass, with the desired 1.4 peak in the passband (notice the 1.4 specified in vector a).



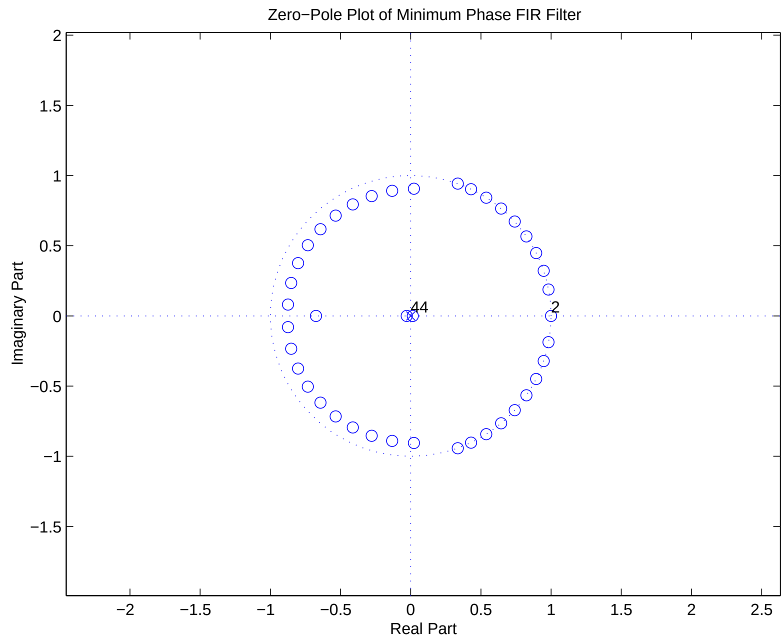
Now for the minimum-phase filter.

```
% Highpass minimum-phase filter optimized for the 4-norm.
b = firlpnorm(44,[0 .4 .45 1],[0 .4 .45 1],[0 0 1 1],[5 1 1 1],...
[2 4],'minphase');
fvtool(b)
```

As shown in the next figure, this is a minimum-phase, highpass filter.



The next zero-pole plot shows the minimum phase nature more clearly.



See Also

`firgr`, `iirgrpdelay`, `iirlpnorm`, `iirlpnormc`
`filter`, `fvtool`, `freqz`, `zplane` in your Signal Processing Toolbox documentation

References

[1] Saramaki, T, Finite Impulse Response Filter Design, *Handbook for Digital Signal Processing*. S.K. Mitra and J.F. Kaiser Eds. Wiley-Interscience, N.Y., 1993, Chapter 4.

firls

Purpose

Design filters using filter specifications from a filter design object and a least-square minimization technique

Syntax

`hd = firls(d)`

Description

`hd = firls(d)` designs a discrete-time FIR filter using a least-squares error minimization method. Only halfband and interpolation design objects with `SpecificationType` of `'n,tw'` or `'pl,tw'` work as design objects for `firls`.

`hd` is either a `dfilt` object (a single-rate digital filter) or an `mfilt` object (a multirate digital filter) depending on the `SpecificationType` property of the filter design object `d` and the design object type—halfband or interp.

Examples

Here are two examples of using `firls` to design filters. The first example returns a single-rate halfband filter.

```
d = fdesign.halfband('n,tw',120,.04); % 120 is the filter order
hd = firls(d);
```

Now use `firls` to design a multirate halfband interpolator filter.

```
d = fdesign.interp(2,'pl,tw',60,.04); % 60 is the polyphase
                                      % length
hm = firls(d);
```

See Also

`equiripple`, `kaiserwin`

firminphase

Purpose

Compute the minimum-phase FIR spectral factor

Syntax

```
h = firminphase(b)
h = firminphase(b,nz)
```

Description

`h = firminphase(b)` computes the minimum-phase FIR spectral factor `h` of a linear-phase FIR filter `b`. Filter `b` must be real, have even order, and have nonnegative zero-phase response.

`h = firminphase(b,nz)` specifies the number of zeros, `nz`, of `b` that lie on the unit circle. You must specify `nz` as an even number to compute the minimum-phase spectral factor because every root on the unit circle must have even multiplicity. Including `nz` can help `firminphase` calculate the required FIR spectral factor. Zeros with multiplicity greater than two on the unit circle cause problems in the spectral factor determination.

Note You can find the maximum-phase spectral factor, `g`, by reversing `h`, such that $g = fliplr(h)$, and $b = conv(h, g)$.

Example

This example designs a constrained least squares filter with a nonnegative zero-phase response, and then uses `firminphase` to compute the minimum-phase spectral factor.

```
f   = [0 0.4 0.8 1];
a   = [0 1 0];
up  = [0.02 1.02  0.01];
lo  = [0 0.98 0]; % The zeros insure nonnegative zero-phase resp.
n   = 32;
b   = fircls(n,f,a,up,lo);
h   = firminphase(b);
```

See Also

`firgr`
`fircls`, `zerophase` in your Signal Processing Toolbox documentation

References

[1] Saramaki, T, Finite Impulse Response Filter Design, *Handbook for Digital Signal Processing*. S.K. Mitra and J.F. Kaiser Eds. Wiley-Interscience, N.Y., 1993, Chapter 4.

Purpose

Design a Lowpass Nyquist (Lth-band) FIR filter

Syntax

`firnyquist(n,l,r,varargin)`

Description

`b = firnyquist(n,l,r)` designs an Nth order, Lth band, Nyquist FIR filter with a roll-off factor `r` and an equiripple characteristic.

The rolloff factor `r` is related to the normalized transition width `tw` by $tw = 2\pi(r/l)$ (rad/sample). The order, `n`, must be even. `l` must be an integer greater than one. If `l` is not specified, it defaults to `4`. `r` must satisfy `0< r < 1`. If `r` is not specified, it defaults to `0.5`.

`b = firnyquist('`**`minorder`**`',l,r,dev)` designs a minimum-order, Lth band Nyquist FIR filter with a rolloff factor `r` using the Kaiser window. The peak ripple is constrained by the scalar `dev`.

`b = firnyquist(n,l,r,decay)` designs an Nth order, Lth band, Nyquist FIR filter where the scalar `decay`, specifies the rate of decay in the stopband. `decay` must be nonnegative. If omitted or left empty, `decay` defaults to `0` which yields an equiripple stopband. `A` nonequiripple stopband may be desirable for decimation purposes.

`b = firnyquist(n,l,r,'`**`nonnegative`**`')` returns an FIR filter with nonnegative zero-phase response. This filter can be spectrally factored into minimum-phase and maximum-phase “square-root” filters. This allows using the spectral factors in applications such as matched-filtering.

`b = firnyquist(n,l,r,'`**`minphase`**`')` returns the minimum-phase spectral factor `bmin` of order `n`. `bmin` meets the condition `b=conv(bmin,bmax)` so that `b` is an Lth band FIR Nyquist filter of order 2n with rolloff factor `r`. Obtain `bmax`, the maximum phase spectral factor by reversing the coefficients of `bmin`. For example, `bmax = bmin(end:-1:1)`.

Example

Example 1: This example designs a minimum phase factor of a Nyquist filter.

```
bmin = firnyquist(47,10,.45,'minphase');
b = firnyquist(2*47,10,.45,'nonnegative');
[h,w,s] = freqz(b); hmin = freqz(bmin);
fvtool(b,1,bmin,1);
```

Example 2: This example compares filters with different decay rates.

```
b1 = firnyquist(72,8,.3,0); % Equiripple
b2 = firnyquist(72,8,.3,.5);
b3 = firnyquist(72,8,.3,1);
fvtool(b1,1,b2,1,b3,1);
```

See Also

`firhalfband`, `firgr`, `firls`, `firminphase`
`firrcos`, `firls` in your Signal Processing Toolbox documentation

References

[1] T. Saramaki, Finite Impulse Response Filter Design, *Handbook for Digital Signal Processing*. S.K. Mitra and J.F. Kaiser Eds. Wiley-Interscience, N.Y., 1993, Chapter 4.

Purpose

Design an two-channel, FIR filter bank for perfect reconstruction

Syntax

```
[h0,h1,g0,g1] = firpr2chfb(n,fp)
[h0,h1,g0,g1] = firpr2chfb(n,dev,'dev')
[h0,h1,g0,g1] = firpr2chfb('minorder',fp,dev)
```

Description

`[h0,h1,g0,g1] = firpr2chfb(n,fp)` designs four FIR filters for the analysis sections (`h0` and `h1`) and synthesis section is (`g0` and `g1`) of a two-channel perfect reconstruction filter bank. The design corresponds to the orthogonal filter banks also known as power-symmetric filter banks.

`n` is the order of all four filters. It must be an odd integer. `fp` is the passband-edge for the lowpass filters `h0` and `g0`. The passband-edge argument `fp` must be less than 0.5. `h1` and `g1` are highpass filters with the passband-edge given by (1-`fp`).

`[h0,h1,g0,g1] = firpr2chfb(n,dev,'dev')` designs the four filters such that the maximum stopband ripple of `h0` is given by the scalar `dev`. The stopband-ripple of `h1` is also be given by `dev`, while the maximum stopband-ripple for both `g0` and `g1` is (2*`dev`).

`[h0,h1,g0,g1] = firpr2chfb('minorder',fp,dev)` designs the four filters such that `h0` meets the passband-edge specification `fp` and the stopband-ripple `dev` using minimum order filters to meet the specification.

Algorithm

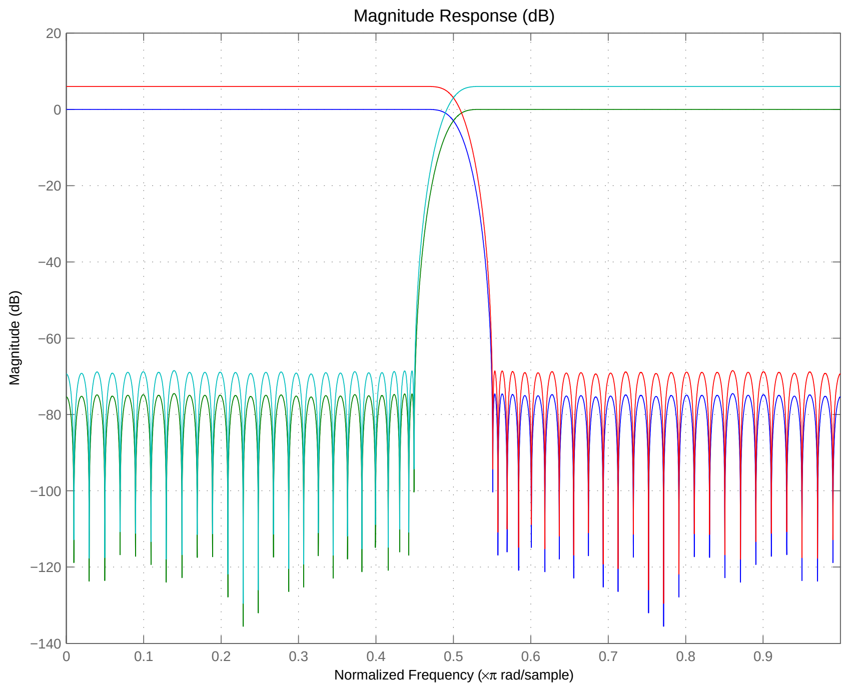
For perfect reconstruction, filters that compose the filter bank must fulfill these conditions.

Examples

Design a filter bank with filters of order `n` equal to 99 and passband edges of 0.45 and 0.55.

```
n = 99;
[h0,h1,g0,g1] = firpr2chfb(n,.45);
fvtool(h0,1,h1,1,g0,1,g1,1);
```

Here are the filters, showing clearly the passband edges.



Use the following stem plots to verify perfect reconstruction using the filter bank created by `firpr2chfb`.

```
stem(1/2*conv(g0,h0)+1/2*conv(g1,h1))
n=0:n;
stem(1/2*conv((-1).^n.*h0,g0)+1/2*conv((-1).^n.*h1,g1))
stem(1/2*conv((-1).^n.*g0,h0)+1/2*conv((-1).^n.*g1,h1))
stem(1/2*conv((-1).^n.*g0,(-1).^n.*h0)+1/2*conv((-1).^n.*g1,...
(-1).^n.*h1))
stem(conv((-1).^n.*h1,h0)-conv((-1).^n.*h0,h1))
```

See Also

`firceqrip`, `firgr`, `firhalfband`, `firnyquist`

Purpose Determine the type of a linear phase FIR filter, either discrete-time or multirate

Syntax

```
t = firtype(hd)
t = firtype(hm)
```

Description The next sections describe common `firtype` operation with discrete-time and multirate filters.

Discrete-Time Filters

`t = firtype(hd)` determines the type (1 through 4) of a discrete-time FIR filter object `hd`, returning the type number in `t`. Filter `hd` must be both real and have linear phase.

Filter types 1 through 4 are defined as follows:

- Type 1—even order symmetric coefficients
- Type 2—odd order symmetric coefficients
- Type 3—even order antisymmetric coefficients
- Type 4—odd order antisymmetric coefficients

When `hd` is a cascade or parallel filter and therefore has multiple stages, each stage must be a real FIR filter with linear phase. In this case, `t` is a cell array containing the filter type of each stage.

Multirate Filters

`t = firtype(hm)` determines the type (1 through 4) of the multirate filter object `hm`. The filter must be real and have linear phase.

Filter types 1 through 4 are defined as follows:

- Type 1—even order symmetric coefficients
- Type 2—odd order symmetric coefficients
- Type 3—even order antisymmetric coefficients
- Type 4—odd order antisymmetric coefficients

When hm has multiple sections, all sections must be real FIR filters with linear phase. In this case, t is a cell array containing the filter type of each section.

Examples

Determine the type of the default interpolator for L=4.

```
l = 4;
hm = mfilt.firinterp(l);
firtype(hm)
ans =

     1
```

See Also

islinphase

Purpose

Compute the frequency response of discrete-time filters, adaptive filters, and multirate filters

Syntax

```
[h,w] = freqz(ha)
[h,w] = freqz(ha,n)
freqz(ha)
[h,w] = freqz(hd)
[h,w] = freqz(hd,n)
freqz(hd)
[h,w] = freqz(hm)
[h,w] = freqz(hm,n)
freqz(hm)
```

Description

The next sections describe common `freqz` operation with adaptive, discrete-time, and multirate filters. For more input options, refer to `freqz` in the Signal Processing Toolbox.

Adaptive Filters

For adaptive filters, `freqz` returns the instantaneous frequency response based on the current filter coefficients.

`[h,w] = freqz(ha)` returns the frequency response vector `h` and the corresponding frequency vector `w` for the adaptive filter `ha`. When `ha` is a vector of adaptive filters, `freqz` returns the matrix `h`. Each column of `h` corresponds to one filter in the vector `ha`.

`[h,w] = freqz(ha,n)` returns the frequency response vector `h` and the corresponding frequency vector `w` for the adaptive filter `ha`. `freqz` uses the transfer function associated with the adaptive filter to calculate the frequency response of the filter with the current coefficient values. The vectors `h` and `w` are both of length `n`. The frequency vector `w` has values ranging from 0 to π radians per sample. If you do not specify the integer `n`, or you specify it as the empty vector `[]`, the frequency response is calculated using the default value of 8192 samples for the FFT.

`freqz(ha)` uses FVTool to plot the magnitude and unwrapped phase of the frequency response of the adaptive filter `ha`. If `ha` is a vector of filters, `freqz` plots the magnitude response and phase for each filter in the vector.

Discrete-Time Filters

[h,w] = freqz(hd) returns the frequency response vector h and the corresponding frequency vector w for the discrete-time filter hd. When hd is a vector of discrete-time filters, freqz returns the matrix h. Each column of h corresponds to one filter in the vector hd.

[h,w] = freqz(hd,n) returns the frequency response vector h and the corresponding frequency vector w for the discrete-time filter hd. freqz uses the transfer function associated with the discrete-time filter to calculate the frequency response of the filter with the current coefficient values. The vectors h and w are both of length n. The frequency vector w has values ranging from 0 to π radians per sample. If you do not specify the integer n, or you specify it as the empty vector [], the frequency response is calculated using the default value of 8192 samples for the FFT.

freqz(hd) uses FVTool to plot the magnitude and unwrapped phase of the frequency response of the adaptive filter hd. If hd is a vector of filters, freqz plots the magnitude response and phase for each filter in the vector.

Multirate Filters

[h,w] = freqz(hm) returns the frequency response vector h and the corresponding frequency vector w for the multirate filter hd. When hd is a vector of multirate filters, freqz returns the matrix h. Each column of h corresponds to one filter in the vector hd.

[h,w] = freqz(hd,n) returns the frequency response vector h and the corresponding frequency vector w for the multirate filter hd. freqz uses the transfer function associated with the multirate filter to calculate the frequency response of the filter with the current coefficient values. The vectors h and w are both of length n. The frequency vector w has values ranging from 0 to π radians per sample. If you do not specify the integer n, or you specify it as the empty vector [], the frequency response is calculated using the default value of 8192 samples for the FFT.

freqz(hd) uses FVTool to plot the magnitude and unwrapped phase of the frequency response of the adaptive filter hd. If hd is a vector of filters, freqz plots the magnitude response and phase for each filter in the vector.

Remarks

There are several ways of analyzing the frequency response of filters. `freqz` accounts for quantization effects in the filter coefficients, but does not account for quantization effects in filtering arithmetic. To account for the quantization effects in filtering arithmetic, refer to function `noisepsd`.

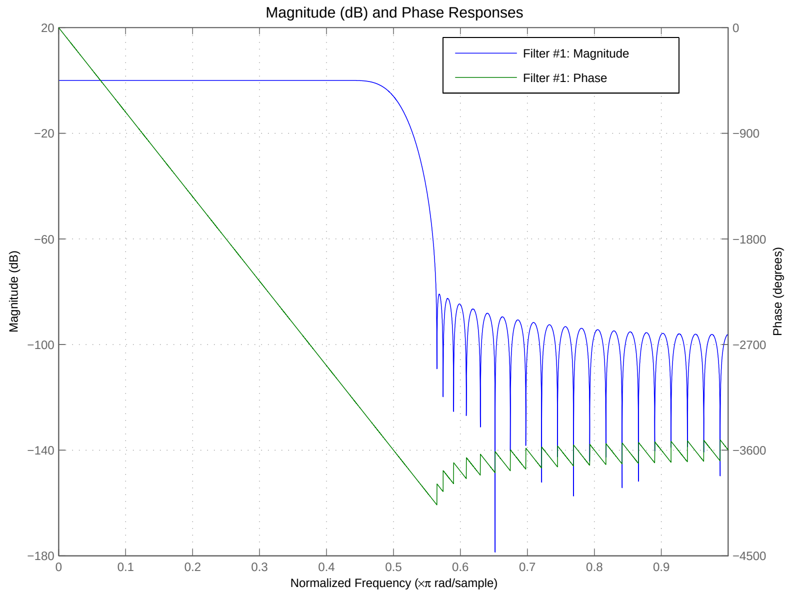
Algorithm

`freqz` calculates the frequency response for a filter from the filter transfer function $Hq(z)$. The complex-valued frequency response is calculated by evaluating $Hq(e^{j\omega})$ at discrete values of w specified by the syntax you use. The integer input argument `n` determines the number of equally-spaced points around the upper half of the unit circle at which `freqz` evaluates the frequency response. The frequency ranges from 0 to π radians per sample when you do not supply a sampling frequency as an input argument. When you supply the scalar sampling frequency `fs` as an input argument to `freqz`, the frequency ranges from 0 to `fs`/2 Hz.

Examples

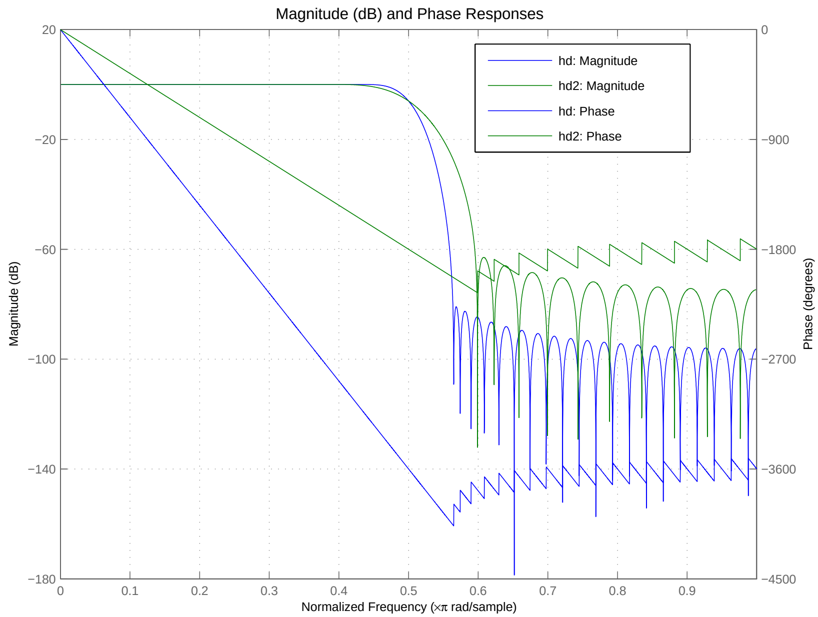
Plot the estimated frequency response of a filter. This example uses discrete-time filters, but any `adaptfilt`, `dfilt`, or `mfilt` object would work. First plot the results for one filter.

```
b = fir1(80,0.5,kaiser(81,8));
hd = dfilt.dffir(b);
freqz(hd);
```



If you have more than one filter, you can plot them on the same figure using a vector of filters.

```
b = fir1(40,0.5,kaiser(41,6));
hd2 = dfilt.dffir(b);
h = [hd hd2];
freqz(h);
```



See Also

`adaptfilt`, `dfilt`, `mfilt`
`fvtool` in your Signal Processing Toolbox documentation

gain

Purpose

Return the gain of a cascaded integrator-comb (CIC) filter

Syntax

```
gain(hm)
gain(hm,j)
```

Description

`gain(hm)` returns the gain of `hm` from the first section up to and including the last section of the CIC decimation filter.

`gain(hm,j)` returns the gain of the `j`th section of a CIC interpolation filter. When you omit `j`, `gain` assumes that `j` is 2*N, where N is the number of sections, and returns the gain of the last section of the filter.

Examples

Determine the overall gain and the gain for each section of a CIC interpolator with zero latency.

```
hm=mfilt.cicinterpzerolat(6)

hm =

        FilterStructure: 'Zero-Latency Cascaded Integrator-Comb
Interpolator'
      DifferentialDelay: 1
       NumberOfSections: 2
          InputBitWidth: 16
         OutputBitWidth: 16
    InterpolationFactor: 6
   ResetBeforeFiltering: 'on'
                 States: [2x2 int32]
    NumSamplesProcessed: 0

gain(hm)

ans =

     6

gain(hm,1)

ans =
```

```
     2

gain(hm,2)

ans =

     4
```

See Also filtmsb

grpdelay

Purpose

Return the group delay for adaptive filters, discrete-time filters, and multirate filters

Syntax

```
[gd,w] = grpdelay(ha)
[gd,w] = grpdelay(ha,n)
[gd,w] = grpdelay(...,f)
grpdelay(ha)
[gd,w] = grpdelay(hd)
[gd,w] = grpdelay(hd,n)
[gd,w] = grpdelay(...,f)
grpdelay(hd)
[gd,w] = grpdelay(hm)
[gd,w] = grpdelay(hm,n)
[gd,w] = grpdelay(...,f)
grpdelay(hm)
```

Description

The next sections describe common `grpdelay` operation with adaptive, discrete-time, and multirate filters. For more input options, refer to `grpdelay` in the Signal Processing Toolbox.

Adaptive Filters

For adaptive filters, `grpdelay` returns the instantaneous group delay based on the current filter coefficients.

`[gd,w] = grpdelay(ha)` returns the group delay vector `gd` and the corresponding frequency vector `w` for the adaptive filter `ha`. When `ha` is a vector of adaptive filters, `grpdelay` returns the matrix `gd`. Each column of `gd` corresponds to one filter in the vector `ha`. If you provide a row vector of frequency points `f` as an input argument, each row of `gd` corresponds to one filter in the vector.

Function `grpdelay` uses the transfer function associated with the adaptive filter to calculate the group delay of the filter with the current coefficient values. The vectors `gd` and `w` are both of length `n`. The frequency vector `w` has values ranging from 0 to π radians per sample. If you do not specify the integer `n`, or you specify it as the empty vector `[]`, the frequency response is calculated using the default value of 8192 samples for the FFT.

[gd,w] = grpdelay(h,n) returns length n vectors vector gd containing the current group delay for the adaptive filter ha and the vector w which contains the frequencies in radians at which grpdelay calculated the delay. Group delay is

$$-\frac{d}{dw}(angle(w))$$

The frequency response is evaluated at n points equally spaced around the upper half of the unit circle. For FIR filters where n is a power of two, the computation is done faster using FFTs. When you do not specify n, it defaults to 8192.

grpdelay(ha) uses FVTool to plot the group delay of the adaptive filter ha. If ha is a vector of filters, grpdelay plots the magnitude response and phase for each filter in the vector.

Discrete-Time Filters

[gd,w] = grpdelay(hd) returns the group delay vector gd and the corresponding frequency vector w for the discrete-time filter hd. When hd is a vector of discrete-time filters, grpdelay returns the matrix gd. Each column of gd corresponds to one filter in the vector hd. If you provide a row vector of frequency points f as an input argument, each row of gd corresponds to each filter in the vector.

Function grpdelay uses the transfer function associated with the discrete-time filter to calculate the group delay of the filter. The vectors gd and w are both of length n. The frequency vector w has values ranging from 0 to π radians per sample. If you do not specify the integer n, or you specify it as the empty vector [], the frequency response is calculated using the default value of 8192 samples for the FFT.

[gd,w] = grpdelay(hd,n) returns length n vectors vector gd containing the current group delay for the discrete-time filter hd and the vector w which contains the frequencies in radians at which grpdelay calculated the delay. Group delay is

$$-\frac{d}{dw}(angle(w))$$

The frequency response is evaluated at n points equally spaced around the upper half of the unit circle. For FIR filters where n is a power of two, the computation is done faster using FFTs. When you do not specify n, it defaults to 8192.

`grpdelay(hd)` uses FVTool to plot the group delay of the discrete-time filter hd. If hd is a vector of filters, grpdelay plots the magnitude response and phase for each filter in the vector.

Multirate Filters

`[gd,w] = grpdelay(hm)` returns the group delay vector gd and the corresponding frequency vector w for the multirate filter hm. When hm is a vector of multirate filters, grpdelay returns the matrix gd. Each column of gd corresponds to one filter in the vector hm. If you provide a row vector of frequency points f as an input argument, each row of gd corresponds to one filter in the vector.

Function grpdelay uses the transfer function associated with the multirate filter to calculate the group delay of the filter. The vectors gd and w are both of length n. The frequency vector w has values ranging from 0 to π radians per sample. If you do not specify the integer n, or you specify it as the empty vector [], the frequency response is calculated using the default value of 8192 samples for the FFT.

`[gd,w] = grpdelay(hm,n)` returns length n vectors vector gd containing the group delay for the multirate filter hm and the vector w which contains the frequencies in radians at which grpdelay calculated the delay. Group delay is

$$-\frac{d}{dw}(angle(w))$$

The frequency response is evaluated at n points equally spaced around the upper half of the unit circle. For FIR filters where n is a power of two, the computation is done faster using FFTs. When you do not specify n, it defaults to 8192.

`grpdelay(hm)` uses FVTool to plot the magnitude and unwrapped phase of the group delay of the multirate filter hm. If ha is a vector of filters, grpdelay plots the group delay for each filter in the vector.

See Also phasez, zerophase

ifir

Purpose

Design interpolated FIR filters

Syntax

```
h = ifir(l,type,f,dev)
h = ifir(l,type,f,dev,str)
```

Description

`h = ifir(l,type,f,dev)` finds a periodic filter $f(z^l)$ and an image-suppressor filter $G(z)$ such that

$$h = f(z^l)G(z)$$

where l is the interpolation factor.

`h` represents the optimal minimax FIR approximation to the desired response specified by the string `type`. Specify the filter band edge frequencies in vector `f`. With `ifir`, you designs a filter that meets the response defined by `type` which does not exceed the peak ripple specified in vector `dev`.

`type` must be a string with either '**`low`**' to generate lowpass filters or '**`high`**' for highpass filters. `f` must be a two-element vector containing two values — the first defining the passband edge frequency and the second that defines the stopband edge frequency. Vector `dev` must contain two values that specify the peak ripple or deviation allowed in the passband and stopband.

`h = ifir(l,type,f,dev,str)` uses the string specified in `str` to select the degree of optimization the interpolation algorithm uses. `str` can be one of three allowed strings:

str String Value	Description
'**`simple`**'	
'**`intermediate`**'	
'**`advanced`**'	

`str` lets you direct the filter design algorithm to trade between the time it takes to design the filter and optimizing the filter order. The '**`advanced`**' option can substantially reduce the filter order, especially for *g(z)*.

Examples

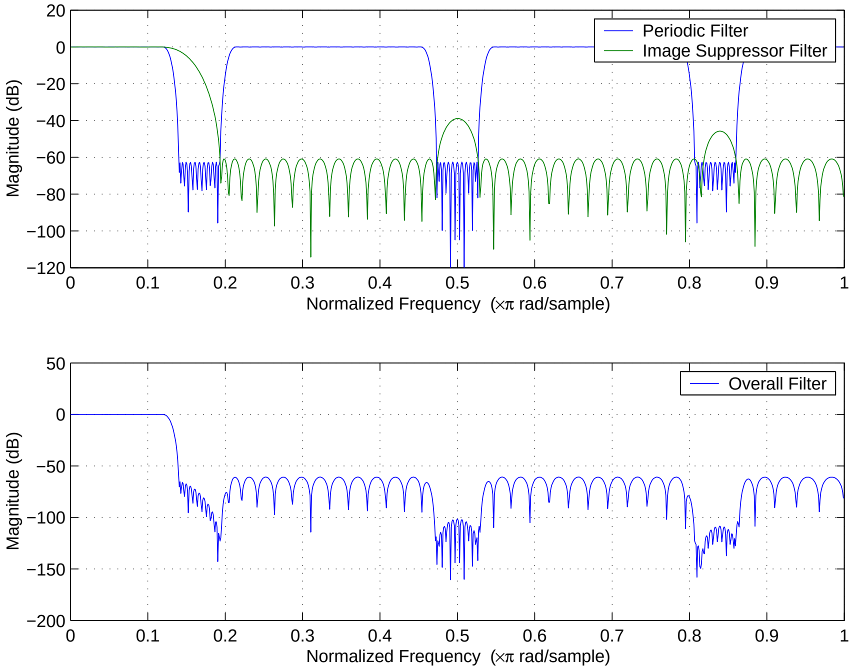
The first example creates a lowpass filter using `ifir` with an interpolation factor of 6. In example 2, the code designs a wideband highpass filter with the same interpolation factor. You can see the plots of the examples after the code sections.

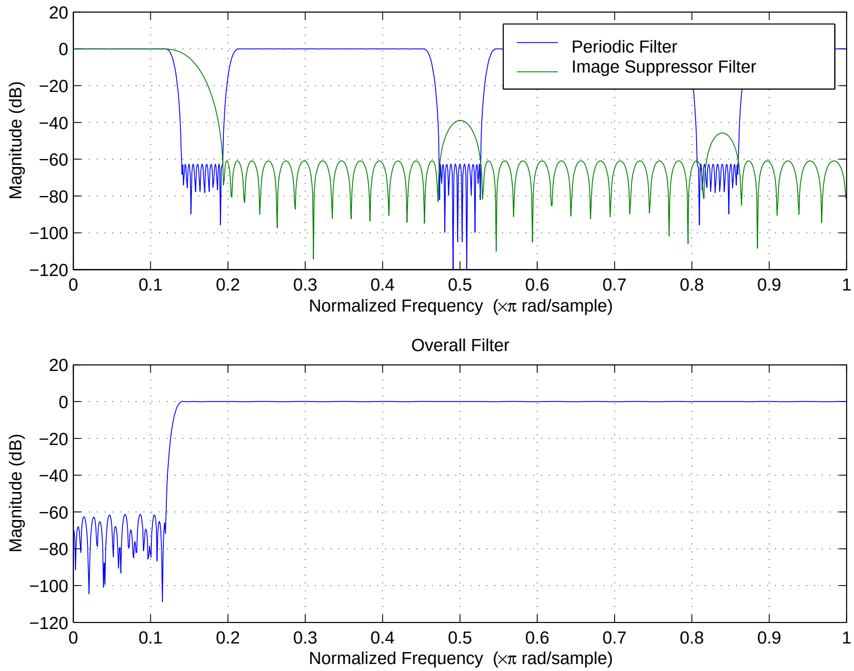
Create a narrowband lowpass design using an interpolation factor of 6.

```
[h,g]=ifir(6,'low',[.12 .14],[.01 .001]);
[Hh,w]=freqz(h,1,1024); Hg=freqz(g,1,1024);
h = hh.*hg; % Compounded response
subplot(2,1,1), freqzplot([hh,hg],w,'mag');
legend('Periodic Filter','Image Suppressor Filter');
subplot(2,1,2), freqzplot(h,w,'mag');
legend('Overall Filter');
```

Use the '**`high`**' option to create a wideband highpass design using an interpolation factor of 6.

```
[h,g,d]=ifir(6,'high',[.12 .14],[.001 .01]);
[hh,w]=freqz(h,1,1024); hg=freqz(g,1,1024);
h = hh.*hg; % Branch 1 compounded response
hd = freqz(d,1,1024); % Branch 2 response
hoverall = h+hd;
freqzplot(hoverall,w,'mag');
title('Overall Filter');
```


Periodic Filter
Image Suppressor Filter
Magnitude (dB)
Normalized Frequency (×π rad/sample)
Overall Filter
Magnitude (dB)
Normalized Frequency (×π rad/sample)



See Also

`firgr`

`fir1`, `firls`, `firpm` in your Signal Processing Toolbox documentation

References

[1] Saramaki, T., Finite Impulse Response Filter Design, *Handbook for Digital Signal Processing*. S.K. Mitra and J.F. Kaiser Eds. Wiley-Interscience, N.Y., 1993, Chapter 4.

iirbpc2bpc

Purpose Transform an IIR complex bandpass filter to an IIR complex bandpass filter with different frequency response characteristics

Syntax

```
[Num,Den,AllpassNum,AllpassDen] = iirbpc2bpc(B,A,Wo,Wt)
```

Description `[Num,Den,AllpassNum,AllpassDen] = iirbpc2bpc(B,A,Wo,Wt)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the complex bandpass prototype by applying a first-order complex bandpass to complex bandpass frequency transformation.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with the numerator specified by `B` and the denominator specified by `A`.

This transformation effectively places two features of an original filter, located at frequencies W_{o1} and W_{o2}, at the required target frequency locations, W_{t1}, and W_{t2} respectively. It is assumed that W_{t2} is greater than W_{t1}. In most of the cases the features selected for the transformation are the band edges of the filter passbands. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

This transformation can also be used for transforming other types of filters; e.g., complex notch filters or resonators can be repositioned at two distinct desired frequencies at any place around the unit circle; e.g., in the adaptive system.

Examples Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

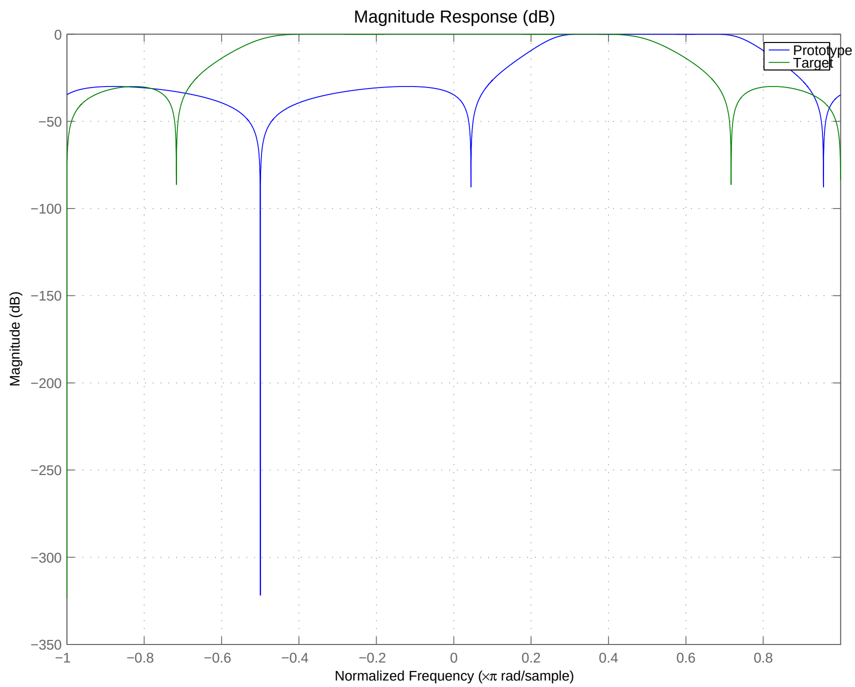
Create a complex passband from 0.25 to 0.75:

```
[b, a] = iirlp2bpc (b, a, 0.5, [0.25,0.75]);
[num, den] = iirbpc2bpc(b, a, [0.25, 0.75], [-0.5, 0.5]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```

Using FVTool to plot the filters shows you the comparison, presented in this figure.



Arguments

B

Numerator of the prototype lowpass filter

A

Denominator of the prototype lowpass filter

Wo

Frequency values to be transformed from the prototype filter

Wt

Desired frequency locations in the transformed target filter

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

See Also

`iirftransf`, `allpassbpc2bpc`, `zpkbpc2bpc`

Purpose

Design an IIR comb notching or peaking digital filter

Syntax

```
[num,den] = iircomb(n,bw)
[num,den] = iircomb(n,bw,ab)
[num,den] = iircomb( , 'type')
```

Description

`[num,den] = iircomb(n,bw)` returns a digital notching filter with order `n` and with the width of the filter notch at -3dB set to `bw`, the filter bandwidth. The filter order must be a positive integer. `n` also defines the number of notches in the filter across the frequency range from 0 to 2π—the number of notches equals `n+1`.

For the notching filter, the transfer function takes the form

$$H(z) = b \times \frac{1 - z^{-n}}{1 - az^{-n}}$$

where a and b are the filter coefficients and n is the filter order or the number of notches in the filter minus 1.

The quality factor (Q factor) q for the filter is related to the filter bandwidth by q = ω_0/bw where ω_0 is the frequency to remove from the signal.

`[num,den] = iircomb(n,bw,ab)` returns a digital notching filter whose bandwidth, `bw`, is specified at a level of -`ab` decibels. Including the optional input argument `ab` lets you specify the magnitude response bandwidth at a level that is not the default -3dB point, such as -6 dB or 0 dB.

`[num,den] = iircomb( ,'type')` returns a digital filter of the specified type. The input argument `type` can be either

- `'notch'` to design an IIR notch filter. Notch filters attenuate the response at the specified frequencies. This is the default type. When you omit the `type` input argument, `iircomb` returns a notch filter.
- `'peak'` to design an IIR peaking filter. Peaking filters boost the signal at the specified frequencies.

The transfer function for peaking filters is

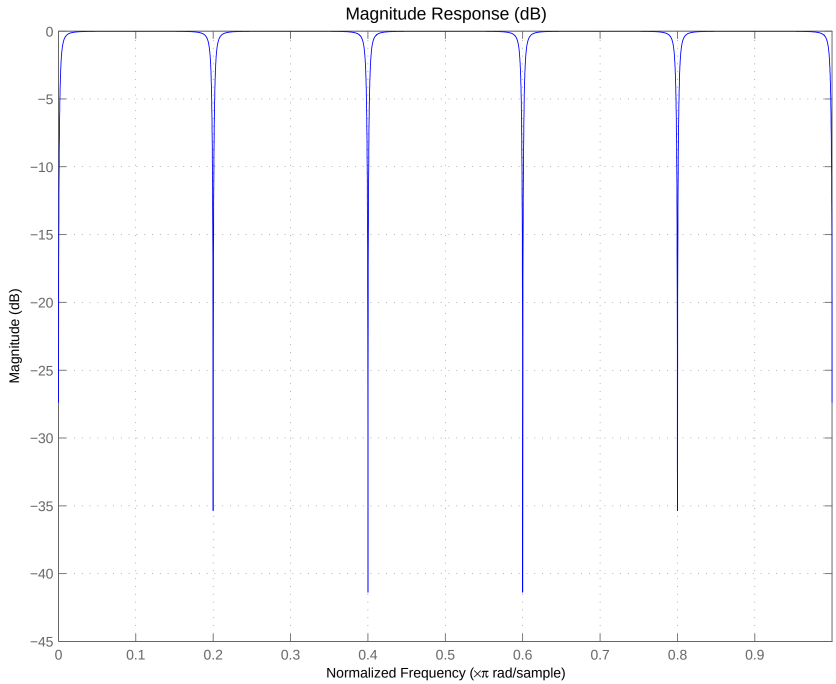
$$H(z) = b \times \frac{1 + z^{-n}}{1 - az^{-n}}$$

Examples

Design and plot an IIR notch filter with 11 notches (equal to filter order plus 1) that removes a 60 Hz tone (`f0`) from a signal at 600 Hz (`fs`). For this example, set the Q factor for the filter to 35 and use it to specify the filter bandwidth.

```
fs = 600; fo = 60;  q = 35; bw = (fo/(fs/2))/q;
[b,a] = iircomb(fs/fo,bw,'notch'); % Note the type flag 'notch'
fvtool(b,a);
```

Using the Filter Visualization Tool (FVTool) generates the following plot showing the filter notches. Note the notches are evenly spaced and one falls at exactly 60 Hz.



See Also

`firgr`, `iirnotch`, `iirpeak`

iirftransf

Purpose

IIR frequency transformation of the digital filter

Syntax

```
[OutNum,OutDen] = iirftransf(OrigNum,OrigDen,FTFNum,FTFDen)
```

Description

`[OutNum,OutDen] = iirftransf(OrigNum,OrigDen,FTFNum,FTFDen)` returns the numerator and denominator vectors, `OutNum` and `OutDen`, of the target filter, which is the result of transforming the prototype filter specified by the numerator, `OrigNum`, and denominator, `OrigDen`, with the mapping filter given by the numerator, `FTFNum`, and the denominator, `FTFDen`. If the allpass mapping filter is not specified, then the function returns an original filter.

Examples

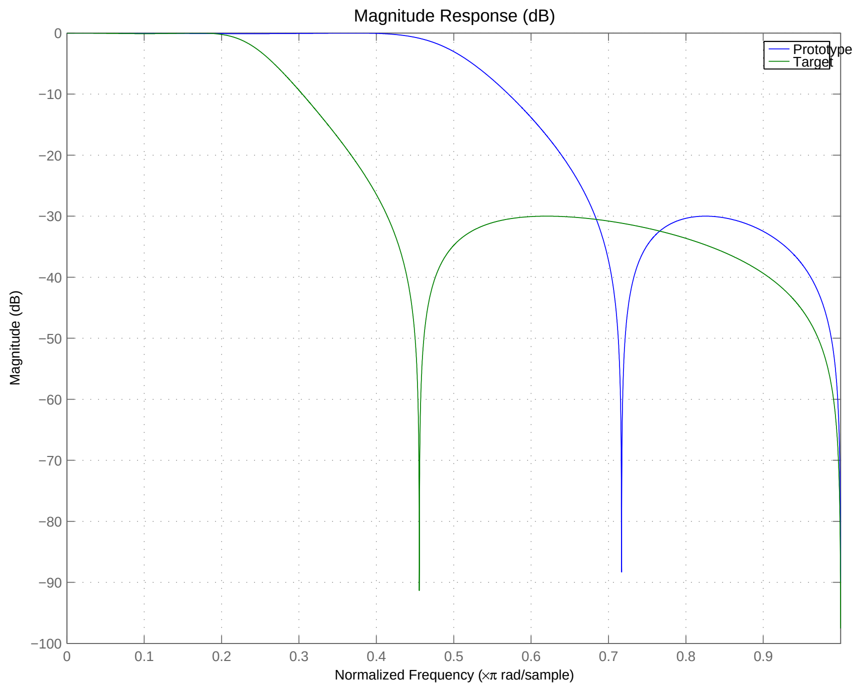
Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
[AlpNum, AlpDen] = allpasslp2lp(0.5, 0.25);
[num, den] = iirftransf(b, a, AlpNum, AlpDen);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```

Here's the comparison between the filters.



Arguments

`OrigNum`
Numerator of the prototype lowpass filter

`OrigDen`
Denominator of the prototype lowpass filter

`FTFNum`
Numerator of the mapping filter

`FTFDen`
Denominator of the mapping filter

`OutNum`
Numerator of the target filter

`OutDen`
Denominator of the target filter

See Also

`zpkftransf`

Purpose

Optimal IIR filter design with prescribed group-delay

Syntax

```
[num,den] = iirgrpdelay(n,f,edges,a)
[num,den] = iirgrpdelay(n,f,edges,a,w)
[num,den] = iirgrpdelay(n,f,edges,a,w,radius)
[num,den] = iirgrpdelay(n,f,edges,a,w,radius,p)
[num,den] = iirgrpdelay(n,f,edges,a,w,radius,p,dens)
[num,den] = iirgrpdelay(n,f,edges,a,w,radius,p,dens,initden)
[num,den] = iirgrpdelay(n,f,edges,a,w,radius,p,dens,initden,tau)
[num,den,tau] = iirgrpdelay(n,f,edges,a,w)
```

Description

`[num,den] = iirgrpdelay(n,f,edges,a)` returns an allpass IIR filter of order `n` (`n` must be even) which is the best approximation to the relative group-delay response described by `f` and `a` in the least-pth sense. `f` is a vector of frequencies between 0 and 1 and `a` is specified in samples. The vector `edges` specifies the band-edge frequencies for multi-band designs. `iirgrpdelay` uses a constrained Newton-type algorithm. Always check your resulting filter using `grpdelay` or `freqz`.

`[num,den] = iirgrpdelay(n,f,edges,a,w)` uses the weights in `w` to weight the error. `w` has one entry per frequency point and must be the same length length as `f` and `a`). Entries in `w` tell `iirgrpdelay` how much emphasis to put on minimizing the error in the vicinity of each specified frequency point relative to the other points.

`f` and `a` must have the same number of elements. `f` and `a` can contains more elements than the vector `edges` contains. This lets you use `f` and `a` to specify a filter that has any group-delay contour within each band.

`[num,den] = iirgrpdelay(n,f,edges,a,w,radius)` returns a filter having a maximum pole radius equal to `radius`, where 0<`radius`<1. `radius` defaults to 0.999999. Filters whose pole radius you constrain to be less than 1.0 can better retain transfer function accuracy after quantization.

`[num,den] = iirgrpdelay(n,f,edges,a,w,radius,p)`, where `p` is a two-element vector `[pmin pmax]`, lets you determine the minimum and maximum values of `p` used in the least-pth algorithm. `p` defaults to [2 128] which yields filters very similar to the L-infinity, or Chebyshev, norm. `pmin` and pmax should be even. If p is the string 'inspect', no optimization occurs. You might use this feature to inspect the initial pole/zero placement.

[num,den] = iirgrpdelay(n,f,edges,a,w,radius,p,dens) specifies the grid density dens used in the optimization process. The number of grid points is (dens*(n+1)). The default is 20. dens can be specified as a single-element cell array. The grid is not equally spaced.

[num,den] = iirgrpdelay(n,f,edges,a,w,radius,p,dens,initden) allows you to specify the initial estimate of the denominator coefficients in vector initden. This can be useful for difficult optimization problems. The pole-zero editor in the Signal Processing Toolbox can be used for generating initden.

[num,den] = iirgrpdelay(n,f,edges,a,w,radius,p,dens,initden,tau) allows the initial estimate of the group delay offset to be specified by the value of tau, in samples.

[num,den,tau] = iirgrpdelay(n,f,edges,a,w) returns the resulting group delay offset. In all cases, the resulting filter has a group delay that approximates [a + tau]. Allpass filters can have only positive group delay and a non-zero value of tau accounts for any additional group delay that is needed to meet the shape of the contour specified by (f,a). The default for tau is max(a).

Hint: If the zeros or poles cluster together, your filter order may be too low or the pole radius may be too small (overly constrained). Try increasing n or radius.

For group-delay equalization of an IIR filter, compute a by subtracting the filter's group delay from its maximum group delay. For example,

```
[be,ae] = ellip(4,1,40,0.2);
f = 0:0.001:0.2;
g = grpdelay(be,ae,f,2);   % Equalize only the passband.
a = max(g)-g;
[num,den]=iirgrpdelay(8, f, [0 0.2], a);
```

See Also

freqz, filter, grpdelay, iirlpnorm, iirlpnormc, zplane

References

Antoniou, A., *Digital Filters: Analysis, Design, and Applications*, Second Edition, McGraw-Hill, Inc. 1993.

Purpose

Transform an IIR real lowpass filter to an IIR real bandpass filter frequency response

Syntax

`[Num,Den,AllpassNum,AllpassDen] = iirlp2bp(B,A,Wo,Wt)`

`[G,AllpassNum,AllpassDen] = iirlp2bp(Hd,Wo,Wt)`, where `Hd` is a `dfilt` object

Description

`[Num,Den,AllpassNum,AllpassDen] = iirlp2bp(B,A,Wo,Wt)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the real lowpass prototype by applying a second-order real lowpass to real bandpass frequency mapping.

It also returns the numerator, `AllpassNum`, and the denominator `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with a numerator specified by `B` and a denominator specified by `A`.

This transformation effectively places one feature of an original filter, located at frequency -W_o, at the required target frequency location, W_{t1}, and the second feature, originally at +W_o, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. This transformation implements the "DC Mobility," meaning that the Nyquist feature stays at Nyquist, but the DC feature moves to a location dependent on the selection of W_ts.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandpass transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature: the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Real lowpass to bandpass transformation can also be used for transforming other types of filters; e.g., real notch filters or resonators can be doubled and positioned at two distinct desired frequencies.

`[G,AllpassNum,AllpassDen] = iirlp2bp(Hd,Wo,Wt)` returns transformed `dfilt` object `G` with a real bandpass magnitude response. The coefficients `AllpassNum` and `AllpassDen` represent the allpass mapping filter for mapping the prototype filter frequency Wo and target frequencies vector Wt. Note that in this syntax Hd is a dfilt object with a lowpass magnitude response.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b,a] = ellip(3, 0.1, 30, 0.409);
```

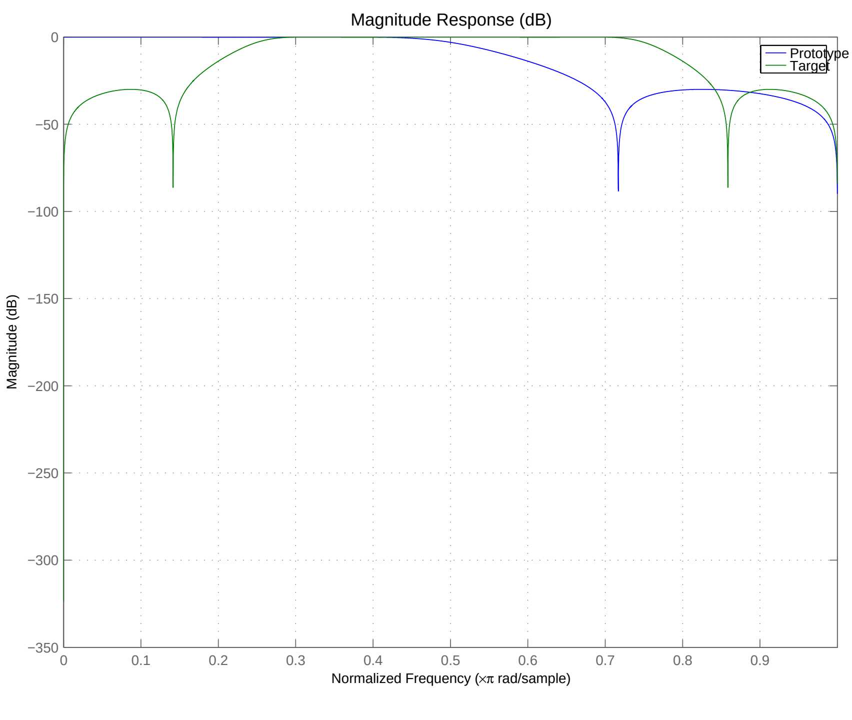
Create the real bandpass filter by placing the cutoff frequencies of the prototype filter at the band edge frequencies W_{t1}=0.25 and W_{t2}=0.75:

```
[num,den] = iirlp2bp(b,a,0.5,[0.25,0.75]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b,a,num,den);
```

You can compare the results in this figure to verify the transformation.



Arguments

`B`
Numerator of the prototype lowpass filter

`A`
Denominator of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirftransf`, `allpasslp2bp`, `zpklp2bp`

References

[1] Constantinides, A.G., “Spectral transformations for digital filters,” *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, “Prototype reference transfer function parameters in the discrete-time frequency transformations,” *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, “Closed-form solutions for discrete-time elliptic transfer functions,” *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

Constantinides, A.G., “Design of bandpass digital filters,” *IEEE Proceedings*, vol. 1, pp. 1129-1231, June 1969.

iirlp2bpc

Purpose

IIR lowpass to complex bandpass frequency transformation frequency response

Syntax

```
[Num,Den,AllpassNum,AllpassDen] = iirlp2bpc(B,A,Wo,Wt)
```

`[G,AllpassNum,AllpassDen] = iirlp2bpc(Hd,Wo,Wt)`, where `Hd` is a `dfilt` object

Description

`[Num,Den,AllpassNum,AllpassDen] = iirlp2bpc(B,A,Wo,Wt)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the real lowpass prototype by applying a first-order real lowpass to complex bandpass frequency transformation.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with a numerator specified by `B` and a denominator specified by `A`.

This transformation effectively places one feature of an original filter, located at frequency $-W_o$, at the required target frequency location, W_{t1}, and the second feature, originally at $+W_o$, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandpass transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Lowpass to bandpass transformation can also be used for transforming other types of filters; for example real notch filters or resonators can be doubled and positioned at two distinct desired frequencies at any place around the unit circle forming a pair of complex notches/resonators. This transformation can be used for designing bandpass filters for radio receivers from the high-quality prototype lowpass filter.

`[G,AllpassNum,AllpassDen] = iirlp2bpc(Hd,Wo,Wt)` returns transformed `dfilt` object `G` with a bandpass magnitude response. The coefficients `AllpassNum` and `AllpassDen` represent the allpass mapping filter for mapping the prototype filter frequency Wo and the target frequencies vector Wt. Note that in this syntax Hd is a dfilt object with a lowpass magnitude response.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

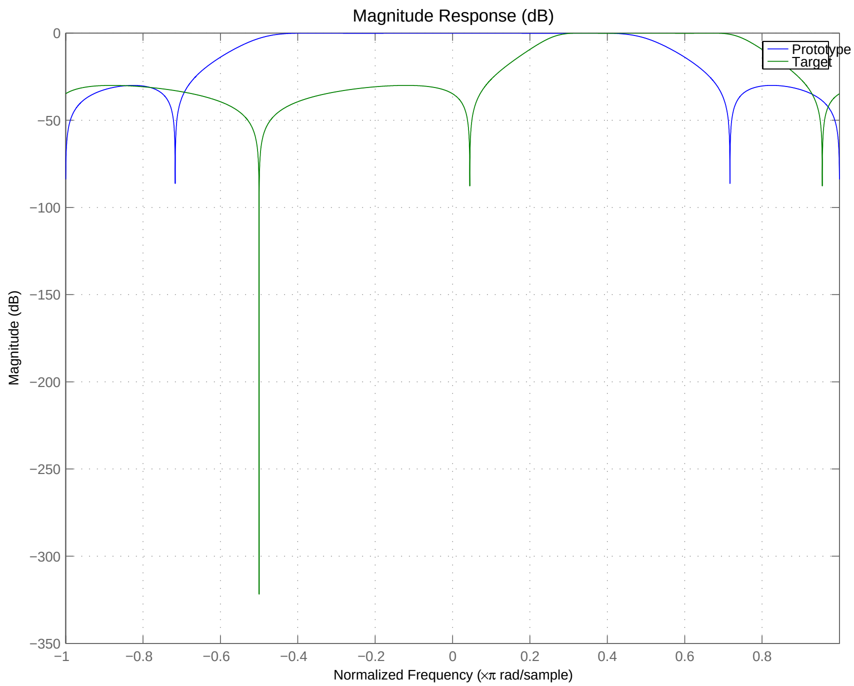
Move the cutoffs of the prototype filter to the new locations W_{t1}=0.25 and W_{t2}=0.75 creating a complex bandpass filter:

```
[num, den] = iirlp2bpc(b, a, 0.5, [0.25, 0.75]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```

PLotting the prototype and target filters together in FVTool lets you compare the filters.



Arguments

`B`

Numerator of the prototype lowpass filter

`A`

Denominator of the prototype lowpass filter

`Wo`

Frequency value to be transformed from the prototype filter. It should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wt`

Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`

Denominator of the mapping filter

See Also

`iirftransf`, `allpasslp2bpc`, `zpklp2bpc`

iirlp2bs

Purpose

Transform an IIR real lowpass filter to an IIR real bandstop filter frequency response

Syntax

```
[Num,Den,AllpassNum,AllpassDen] = iirlp2bs(B,A,Wo,Wt)
[G,AllpassNum,AllpassDen] = iirlp2bs(Hd,Wo,Wt)
```
, where Hd is a dfilt object

Description

[Num,Den,AllpassNum,AllpassDen] = iirlp2bs(B,A,Wo,Wt) returns the numerator and denominator vectors, Num and Den respectively, of the target filter transformed from the real lowpass prototype by applying a second-order real lowpass to real bandstop frequency mapping.

It also returns the numerator, AllpassNum, and the denominator, AllpassDen, of the allpass mapping filter. The prototype lowpass filter is given with a numerator specified by B and a denominator specified by A.

This transformation effectively places one feature of an original filter, located at frequency $-W_o$, at the required target frequency location, W_{t1}, and the second feature, originally at $+W_o$, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. This transformation implements the "Nyquist Mobility," which means that the DC feature stays at DC, but the Nyquist feature moves to a location dependent on the selection of W_o and W_ts.

Relative positions of other features of an original filter change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. After the transformation feature F_2 will precede F_1 in the target filter. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandstop transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

[G,AllpassNum,AllpassDen] = iirlp2bs(Hd,Wo,Wt) returns transformed dfilt object G with a bandstop magnitude response. The coefficients AllpassNum and AllpassDen represent the allpass mapping filter for mapping the prototype filter frequency Wo and the target frequencies vector Wt. Note that in this syntax Hd is a dfilt object with a lowpass magnitude response.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

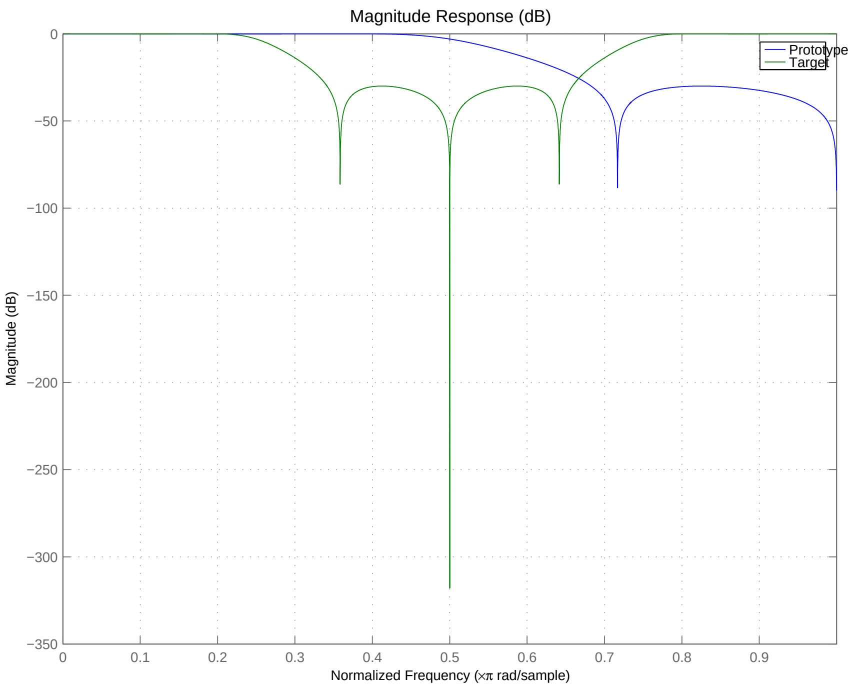
```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

Create the real bandstop filter by placing the cutoff frequencies of the prototype filter at the band edge frequencies W_{t1}=0.25 and W_{t2}=0.75:

```
[num, den] = iirlp2bs(b, a, 0.5, [0.25, 0.75]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```



With both filters plotted in the figure, you see clearly the results of the transformation.

Arguments

B

Numerator of the prototype lowpass filter

`A`
Denominator of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirftransf`, `allpasslp2bs`, `zpklp2bs`

References

[1] Constantinides, A.G., "Spectral transformations for digital filters," *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, "Prototype reference transfer function parameters in the discrete-time frequency transformations," *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, "Closed-form solutions for discrete-time elliptic transfer functions," *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

Constantinides, A.G., "Design of bandpass digital filters," *IEEE Proceedings*, vol. 1, pp. 1129-1231, June 1969.

Purpose

Transform an IIR real lowpass filter to an IIR complex bandstop filter frequency response

Syntax

`[Num,Den,AllpassNum,AllpassDen] = iirlp2bsc(B,A,Wo,Wt)`

`[G,AllpassNum,AllpassDen] = iirlp2bsc(Hd,Wo,Wt)`, where `Hd` is a `dfilt` object

Description

`[Num,Den,AllpassNum,AllpassDen] = iirlp2bsc(B,A,Wo,Wt)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the real lowpass prototype by applying a first-order real lowpass to complex bandstop frequency transformation.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with a numerator specified by `B` and the denominator specified by `A`.

This transformation effectively places one feature of an original filter, located at frequency -W_o, at the required target frequency location, W_{t1}, and the second feature, originally at +W_o, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. Additionally the transformation swaps passbands with stopbands in the target filter.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandstop transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Lowpass to bandpass transformation can also be used for transforming other types of filters; e.g., real notch filters or resonators can be doubled and positioned at two distinct desired frequencies at any place around the unit circle forming a pair of complex notches/resonators. This transformation can be used for designing bandstop filters for band attenuation or frequency equalizers, from the high-quality prototype lowpass filter.

`[G,AllpassNum,AllpassDen] = iirlp2bsc(Hd,Wo,Wt)` returns transformed `dfilt` object `G` with a bandstop magnitude response. The coefficients

`AllpassNum` and `AllpassDen` represent the allpass mapping filter for mapping the prototype filter frequency `Wo` and the target frequencies vector `Wt`. Note that in this syntax `Hd` is a `dfilt` object with a lowpass magnitude response.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

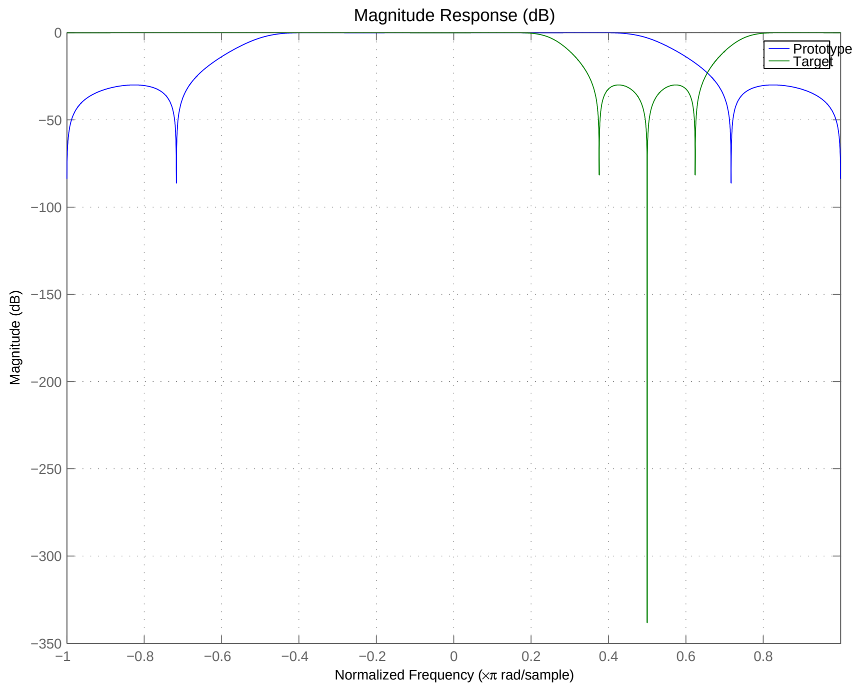
Move the cutoffs of the prototype filter to the new locations W_{t1}=0.25 and W_{t2}=0.75 creating a complex bandstop filter:

```
[num, den] = iirlp2bsc(b, a, 0.5, [0.25, 0.75]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```

The last command in the example plots both filters in the same window so you can compare the results.



Arguments

`B`

Numerator of the prototype lowpass filter

`A`

Denominator of the prototype lowpass filter

`Wo`

Frequency value to be transformed from the prototype filter. It should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wt`

Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also

`iirftransf`, `allpasslp2bsc`, `zpklp2bsc`.

Purpose Transform a discrete time lowpass IIR filter to a highpass filter

Syntax

```
[num,den] = iirlp2hp(b,a,wc,wd)
[G,AllpassNum,AllpassDen] = iirlp2hp(Hd,Wo,Wt), where Hd is a dfilt object
```

Description

`[num,den] = iirlp2hp(b,a,wc,wd)` with input arguments `b` and `a`, the numerator and denominator coefficients (zeros and poles) for a lowpass IIR filter, `iirlp2bp` transforms the magnitude response from lowpass to highpass. `num` and `den` return the coefficients for the transformed highpass filter. For `wc`, enter a selected frequency from your lowpass filter. You use the chosen frequency to define the magnitude response value you want in the highpass filter. Enter one frequency for the highpass filter — the value that defines the location of the transformed point — in `wd`. Note that all frequencies are normalized between zero and one. Notice also that the filter order does not change when you transform to a highpass filter.

When you select `wc` and designate `wd`, the transformation algorithm sets the magnitude response at the `wd` values of your bandstop filter to be the same as the magnitude response of your lowpass filter at `wc`. Filter performance between the values in `wd` is not specified, except that the stopband retains the ripple nature of your original lowpass filter and the magnitude response in the stopband is equal to the peak response of your lowpass filter. To accurately specify the filter magnitude response across the stopband of your bandpass filter, use a frequency value from within the stopband of your lowpass filter as `wc`. Then your bandstop filter response is the same magnitude and ripple as your lowpass filter stopband magnitude and ripple.

The fact that the transformation retains the shape of the original filter is what makes this function useful. If you have a lowpass filter whose characteristics, such as rolloff or passband ripple, particularly meet your needs, the transformation function lets you create a new filter with the same characteristic performance features, but in a highpass version. Without designing the highpass filter from the beginning.

In some cases transforming your filter may cause numerical problems, resulting in incorrect conversion to the highpass filter. Use `fvtool` to verify the response of your converted filter.

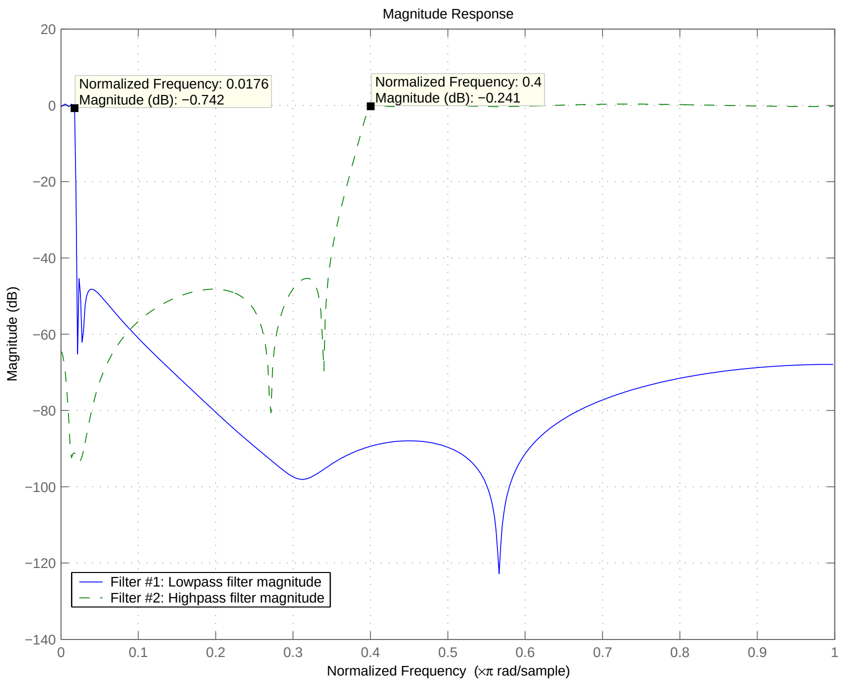
`[G,AllpassNum,AllpassDen] = iirlp2hp(Hd,Wo,Wt)` returns transformed `dfilt` object `G` with a highpass magnitude response. The coefficients

AllpassNum and AllpassDen represent the allpass mapping filter for mapping the prototype filter frequency Wo and the target frequencies vector Wt. Note that in this syntax Hd is a dfilt object with a lowpass magnitude response.

Examples

This example transforms an IIR filter from lowpass to high pass by moving the magnitude response at one frequency in the source filter to a new location in the transformed filter. To generate a highpass filter whose passband flattens out at 0.4, we select the frequency in the lowpass filter where the passband starts to rolloff (wc = 0.0175) and move it to the new location at wd = 0.4.

```
[b,a] = iirlpnorm(10,6,[0 0.0175 0.02 0.0215 0.025 1],...
[0 0.0175 0.02 0.0215 0.025 1],[1 1 0 0 0 0],[1 1 1 1 10 10]);
wc = 0.0175;
wd = 0.4;
[num,den] = iirlp2hp(b,a,wc,wd);
fvtool(b,a,num,den);
```



In the figure showing the magnitude responses for the two filters, the transition band for the highpass filter is essentially the mirror image of the transition for the lowpass filter from 0.0175 to 0.025, stretched out over a wider frequency range. In the passbands, the filter share common ripple characteristics and magnitude.

See Also

`iirlp2bp`, `iirlp2bs`, `iirlp2lp`, `firlp2lp`, `firlp2hp`

References

Sanjit K. Mitra, *Digital Signal Processing. A Computer-Based Approach*, Second Edition, McGraw-Hill, 2001.

iirlp2lp

Purpose

Transform a discrete time lowpass IIR filter to a different lowpass filter

Syntax

`[num,den] = iirlp2lp(b,a,wc,wd)`

`[G,AllpassNum,AllpassDen] = iirlp2lp(Hd,Wo,Wt)`, where `Hd` is a `dfilt` object

Description

`[num,den] = iirlp2hp(b,a,wc,wd)` with input arguments `b` and `a`, the numerator and denominator coefficients (zeros and poles) for a lowpass IIR filter, `iirlp2bp` transforms the magnitude response from lowpass to highpass. `num` and `den` return the coefficients for the transformed highpass filter. For `wc`, enter a selected frequency from your lowpass filter. You use the chosen frequency to define the magnitude response value you want in the highpass filter. Enter one frequency for the highpass filter — the value that defines the location of the transformed point — in `wd`. Note that all frequencies are normalized between zero and one. Notice also that the filter order does not change when you transform to a highpass filter.

When you select `wc` and designate `wd`, the transformation algorithm sets the magnitude response at the `wd` values of your bandstop filter to be the same as the magnitude response of your lowpass filter at `wc`. Filter performance between the values in `wd` is not specified, except that the stopband retains the ripple nature of your original lowpass filter and the magnitude response in the stopband is equal to the peak response of your lowpass filter. To accurately specify the filter magnitude response across the stopband of your bandpass filter, use a frequency value from within the stopband of your lowpass filter as `wc`. Then your bandstop filter response is the same magnitude and ripple as your lowpass filter stopband magnitude and ripple.

The fact that the transformation retains the shape of the original filter is what makes this function useful. If you have a lowpass filter whose characteristics, such as rolloff or passband ripple, particularly meet your needs, the transformation function lets you create a new filter with the same characteristic performance features, but in a highpass version. Without designing the highpass filter from the beginning.

In some cases transforming your filter may cause numerical problems, resulting in incorrect conversion to the highpass filter. Use `fvtool` to verify the response of your converted filter.

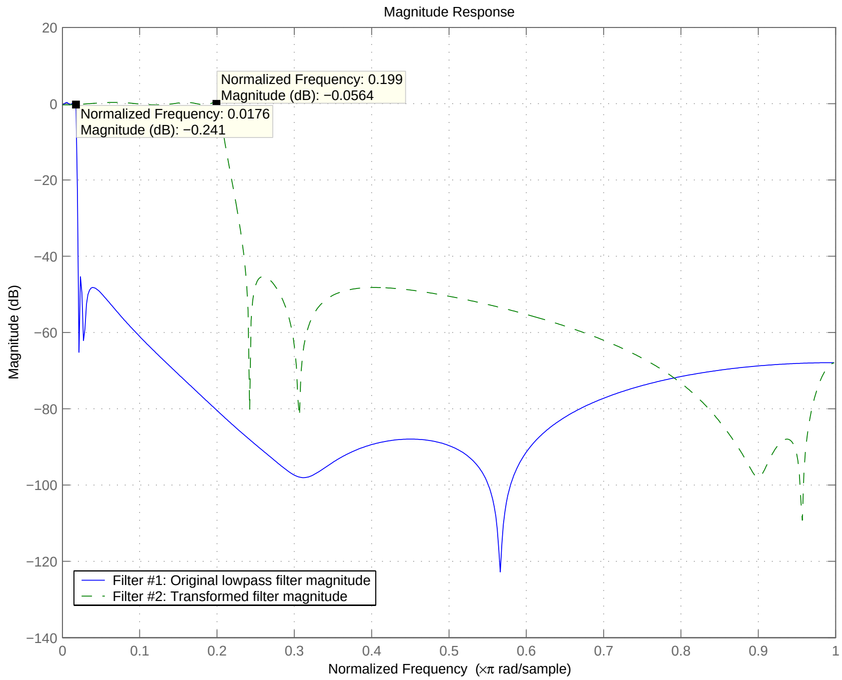
`[G,AllpassNum,AllpassDen] = iirlp2lp(Hd,Wo,Wt)` returns transformed `dfilt` object `G` with a lowpass magnitude response. The coefficients `AllpassNum` and AllpassDen represent the allpass mapping filter for mapping the prototype filter frequency Wo and the target frequencies vector Wt. Note that in this syntax Hd is a dfilt object with a lowpass magnitude response.

Examples

This example transforms an IIR filter from lowpass to high pass by moving the magnitude response at one frequency in the source filter to a new location in the transformed filter. To generate a lowpass filter whose passband extends out to 0.2, we select the frequency in the lowpass filter where the passband starts to rolloff (wc = 0.0175) and move it to the new location at wd = 0.2.

```
[b,a] = iirlpnorm(10,6,[0 0.0175 0.02 0.0215 0.025 1],...
[0 0.0175 0.02 0.0215 0.025 1],[1 1 0 0 0 0],[1 1 1 1 10 10]);
wc = 0.0175;
wd = 0.2;
[num,den] = iirlp2lp(b,a,wc,wd);
fvtool(b,a,num,den);
```

Moving the edge of the passband from 0.0175 to 0.2 results in a new lowpass filter whose peak response in-band is the same as the original filter: same ripple, same absolute magnitude.



Notice that the rolloff is slightly less steep and the stopband profiles are the same for both filters; the new filter stopband is a "stretched" version of the original, as is the passband of the new filter.

See Also

`iirlp2bp`, `iirlp2bs`, `iirlp2hp`, `firlp2lp`, `firlp2hp`

References

Sanjit K. Mitra, *Digital Signal Processing. A Computer-Based Approach*, Second Edition, McGraw-Hill, 2001.

Purpose

Transform an IIR real lowpass filter to an IIR real M-band filter frequency response

Syntax

```
[Num,Den,AllpassNum,AllpassDen] = iirlp2mb(B,A,Wo,Wt)
[Num,Den,AllpassNum,AllpassDen] = iirlp2mb(B,A,Wo,Wt,Pass)
```

`[G,AllpassNum,AllpassDen] = iirlp2bpc(Hd,Wo,Wt)`, where `Hd` is a `dfilt` object

```
[G,AllpassNum,AllpassDen] = iirlp2mb(...,Pass)
```

Description

`[Num,Den,AllpassNum,AllpassDen] = iirlp2mb(B,A,Wo,Wt)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the real lowpass prototype by applying an `M`th-order real lowpass to real multibandpass frequency mapping. By default the DC feature is kept at its original location.

`[Num,Den,AllpassNum,AllpassDen]=iirlp2mb(B,A,Wo,Wt,Pass)` allows you to specify an additional parameter, `Pass`, which chooses between using the "DC Mobility" and the "Nyquist Mobility." In the first case the Nyquist feature stays at its original location and the DC feature is free to move. In the second case the DC feature is kept at an original frequency and the Nyquist feature is movable.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with a numerator specified by `B` and a denominator specified by `A`.

This transformation effectively places one feature of an original filter, located at frequency W_o, at the required target frequency locations, W_{t1},...,W_{tM}.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

`[G,AllpassNum,AllpassDen] = iirlp2bs(Hd,Wo,Wt)` returns transformed `dfilt` object `G` with an IIR real M-band filter frequency response. The coefficients `AllpassNum` and `AllpassDen` represent the allpass mapping filter for mapping the prototype filter frequency `Wo` and the target frequencies vector `Wt`. Note that in this syntax `Hd` is a `dfilt` object with a lowpass magnitude response.

`[G,AllpassNum,AllpassDen] = iirlp2mb(Hd,Wo,Wt)` returns transformed `dfilt` object `G` with an IIR real M-band filter frequency response. The coefficients `AllpassNum` and `AllpassDen` represent the allpass mapping filter for mapping the prototype filter frequency `Wo` and the target frequencies vector `Wt`. Note that in this syntax `Hd` is a `dfilt` object with a lowpass magnitude response.

`[G,AllpassNum,AllpassDen] = iirlp2mb(...,Pass)` returns transformed `dfilt` object `G` with an IIR real M-band filter frequency response. This syntax allows you to specify an additional parameter, `Pass`, which chooses between using the "DC Mobility" and the "Nyquist Mobility." In the first case the Nyquist feature stays at its original location and the DC feature is free to move. In the second case the DC feature is kept at an original frequency and the Nyquist feature is allowed to move.

The coefficients `AllpassNum` and `AllpassDen` represent the allpass mapping filter for mapping the prototype filter frequency `Wo` and the target frequencies vector `Wt`. Note that in this syntax `Hd` is a `dfilt` object with a lowpass magnitude response.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

Example 1: Create the real multiband filter with two passbands:

```
[num1, den1] = iirlp2mb(b, a, 0.5, [2 4 6 8]/10);
[num2, den2] = iirlp2mb(b, a, 0.5, [2 4 6 8]/10, 'pass');
```

The second code snippet uses the `pass` option to select the Nyquist mobility option. In this case the resulting filter is the same.

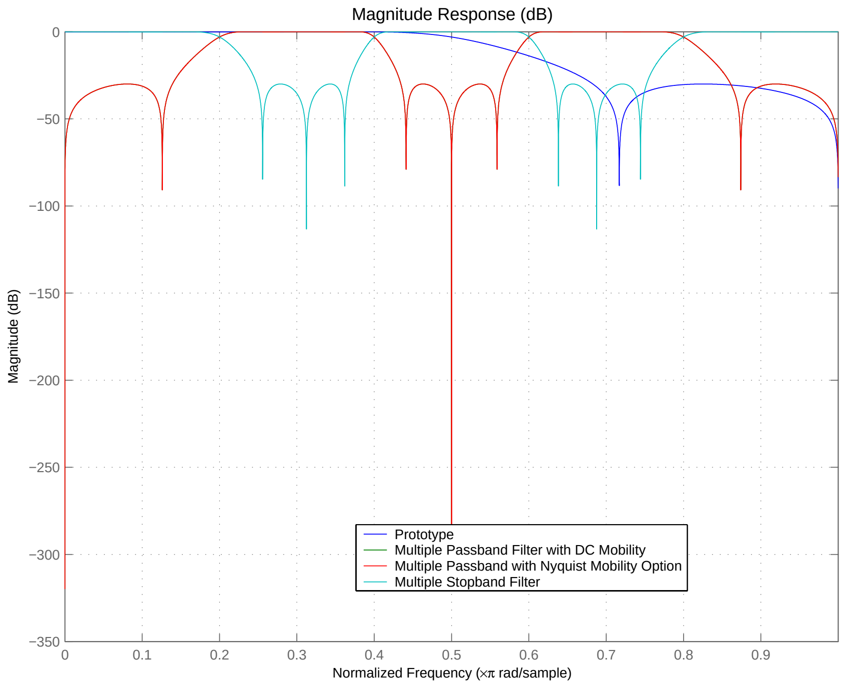
Example 2: Create the real multiband filter with two stopbands:

```
[num3, den3] = iirlp2mb(b, a, 0.5, [2 4 6 8]/10, 'stop');
```

Verify the result by comparing the prototype filter with target filters:

```
fvtool(b, a, num1, den1, num2, den2, num3, den3);
```

Combining all of the filters, prototypes and targets, on one figure makes comparing them straightforward. Passbands for the filters in example 1 appear separately in the figure, although they overlap to a degree that makes them hard to identify—they have identical coefficients.



Arguments

`B`

Numerator of the prototype lowpass filter

`A`
Denominator of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`Pass`
Choice (`'pass'`/`'stop'`) of passband/stopband at DC, `'pass'` being the default

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirftransf`, `allpasslp2mb`, `zpklp2mb`

References

[1] Franchitti, J.C., "All-pass filter interpolation and frequency transformation problems," MSc Thesis, Dept. of Electrical and Computer Engineering, University of Colorado, 1985.

[2] Feyh, G., J.C. Franchitti and C.T. Mullis, "All-pass filter interpolation and frequency transformation problem," *Proceedings 20th Asilomar Conference on Signals, Systems and Computers*, Pacific Grove, California, pp. 164-168, November 1986.

[3] Mullis, C.T. and R. A. Roberts, *Digital Signal Processing*, section 6.7, Reading, Mass., Addison-Wesley, 1987.

[4] Feyh, G., W.B. Jones and C.T. Mullis, "An extension of the Schur Algorithm for frequency transformations," *Linear Circuits, Systems and Signal Processing: Theory and Application*, C. J. Byrnes et al Eds, Amsterdam: Elsevier, 1988.

Purpose

Transform an IIR real lowpass filter to an IIR complex M-band filter frequency response

Syntax

`[Num,Den,AllpassNum,AllpassDen] = iirlp2mbc(B,A,Wo,Wc)`

`[G,AllpassNum,AllpassDen] = iirlp2mbc(Hd,Wo,Wt)`, where `Hd` is a `dfilt` object

Description

`[Num,Den,AllpassNum,AllpassDen] = iirlp2mbc(B,A,Wo,Wc)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the real lowpass prototype by applying an `M`th-order real lowpass to complex multibandpass frequency transformation.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with a numerator specified by `B` and a denominator specified by `A`.

This transformation effectively places one feature of an original filter, located at frequency W_o, at the required target frequency locations, W_{t1},...,W_{tM}.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

`[G,AllpassNum,AllpassDen] = iirlp2mbc(Hd,Wo,Wt)` returns transformed `dfilt` object `G` with an IIR complex M-band filter frequency response. The coefficients `AllpassNum` and `AllpassDen` represent the allpass mapping filter for mapping the prototype filter frequency `Wo` and the target frequencies vector `Wt`. Note that in this syntax `Hd` is a `dfilt` object with a lowpass magnitude response.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

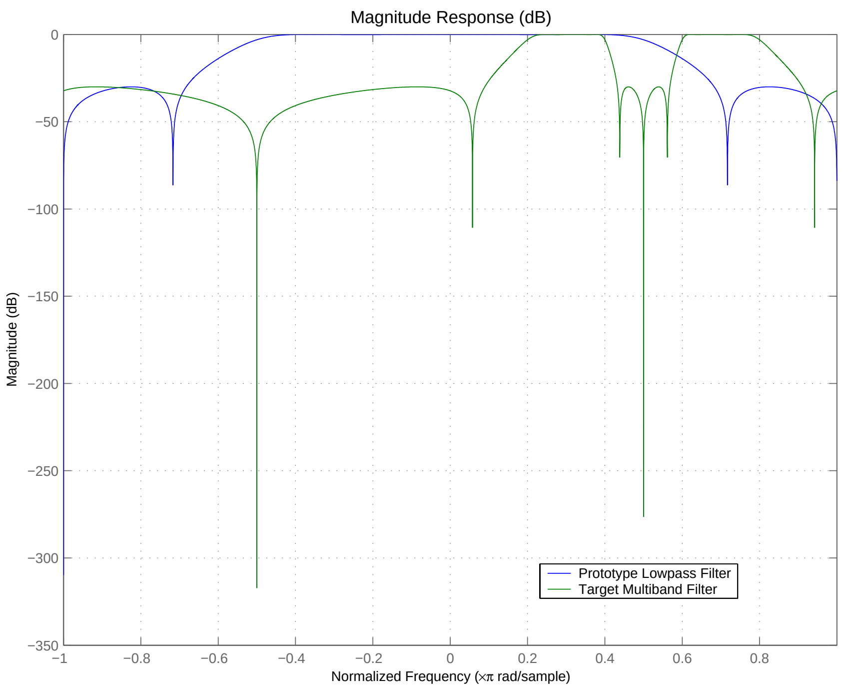
```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

Now create a complex multiband filter with two passbands:

```
[num1, den1] = iirlp2mbc(b, a, 0.5, [2 4 6 8]/10);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num1, den1);
```



You see in the figure that iirlp2mbc replicates the desired feature at 0.5 in the lowpass filter at four locations in the multiband filter.

Arguments

`B`
Numerator of the prototype lowpass filter.

`A`
Denominator of the prototype lowpass filter.

`Wo`
Frequency value to be transformed from the prototype filter. It should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wc`
Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`Num`
Numerator of the target filter.

`Den`
Denominator of the target filter.

`AllpassNum`
Numerator of the mapping filter.

`AllpassDen`
Denominator of the mapping filter.

See Also

`iirftransf`, `allpasslp2mbc`, `zpklp2mbc`

iirlp2xc

Purpose

Transform an IIR real lowpass filter to an IIR complex N-point filter frequency response

Syntax

```
[Num,Den,AllpassNum,AllpassDen] = iirlp2xc(B,A,Wo,Wt)
[G,AllpassNum,AllpassDen] = iirlp2xc(Hd,Wo,Wt), where Hd is a dfilt object
```

Description

`[Num,Den,AllpassNum,AllpassDen] = iirlp2xc(B,A,Wo,Wt)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the real lowpass prototype by applying an `N`th-order real lowpass to complex multipoint frequency transformation.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with a numerator specified by `B` and a denominator specified by `A`.

Parameter `N` also specifies the number of replicas of the prototype filter created around the unit circle after the transformation. This transformation effectively places `N` features of an original filter, located at frequencies $W_{o1},\ldots,W_{oN}$, at the required target frequency locations, $W_{t1},\ldots,W_{tM}$.

Relative positions of other features of an original filter are the same in the target filter for the Nyquist mobility and are reversed for the DC mobility. For the Nyquist mobility this means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation. For DC mobility feature F_2 will precede F_1 after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., a stopband edge, DC, the deep minimum in the stopband, or other ones. The only condition is that the features must be selected in such a way that when creating `N` bands around the unit circle, there will be no band overlap.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

[G,AllpassNum,AllpassDen] = iirlp2xc(Hd,Wo,Wt) returns transformed dfilt object G with an IIR complex N-point filter frequency response. The coefficients AllpassNum and AllpassDen represent the allpass mapping filter for mapping the prototype filter frequency Wo and the target frequencies vector Wt. Note that in this syntax Hd is a dfilt object with a lowpass magnitude response.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

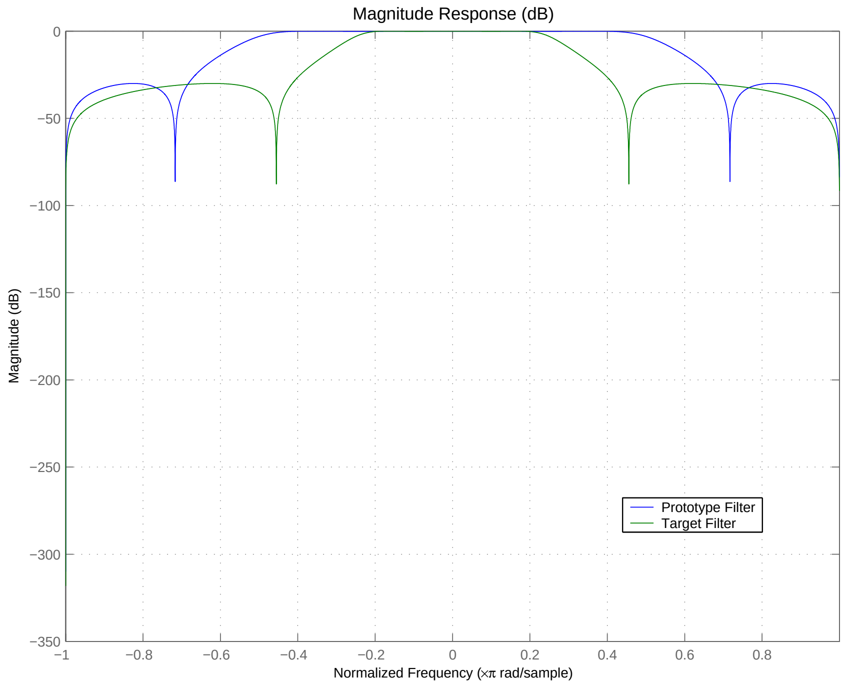
Create the complex bandpass filter from the real lowpass filter:

```
[num, den] = iirlp2xc(b, a, [-0.5 0.5], [-0.25 0.25]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```

REviewing the coefficients and the figure produced by the example shows that the target filter has complex coefficients and is indeed a bandpass filter as expected.



Arguments

`B`

Numerator of the prototype lowpass filter

`A`

Denominator of the prototype lowpass filter

`Wo`

Frequency values to be transformed from the prototype filter. They should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wt`

Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also

`iirftransf`, `allpasslp2xc`, `zpklp2xc`

iirlp2xn

Purpose

Transform an IIR real lowpass filter to an IIR real N-point filter frequency response

Syntax

```
[Num,Den,AllpassNum,AllpassDen] = iirlp2xn(B,A,Wo,Wt)
[Num,Den,AllpassNum,AllpassDen] = iirlp2xn(B,A,Wo,Wt,Pass)
```

`[G,AllpassNum,AllpassDen] = iirlp2bpc(Hd,Wo,Wt)`, where `Hd` is a `dfilt` object

```
[G,AllpassNum,AllpassDen] = iirlp2bpc(...,Pass)
```

Description

`[Num,Den,AllpassNum,AllpassDen] = iirlp2xn(B,A,Wo,Wt)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the real lowpass prototype by applying an `N`th-order real lowpass to real multipoint frequency transformation, where `N` is the number of features being mapped. By default the DC feature is kept at its original location.

`[Num,Den,AllpassNum,AllpassDen]=iirlp2xn(B,A,Wo,Wt,Pass)` allows you to specify an additional parameter, `Pass`, which chooses between using the "DC Mobility" and the "Nyquist Mobility." In the first case the Nyquist feature stays at its original location and the DC feature is free to move. In the second case the DC feature is kept at an original frequency and the Nyquist feature is allowed to move.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with the numerator specified by `B` and the denominator specified by `A`.

Parameter `N` also specifies the number of replicas of the prototype filter created around the unit circle after the transformation. This transformation effectively places `N` features of an original filter, located at frequencies $W_{o1},...,W_{oN}$, at the required target frequency locations, $W_{t1},...,W_{tM}$.

Relative positions of other features of an original filter are the same in the target filter for the Nyquist mobility and are reversed for the DC mobility. For the Nyquist mobility this means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation. For DC mobility feature F_2 will precede F_1 after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones. The only condition is that the features must be selected in such a way that when creating `N` bands around the unit circle, there will be no band overlap.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

`[G,AllpassNum,AllpassDen] = iirlp2xn(Hd,Wo,Wt)` returns transformed `dfilt` object `G` with an IIR real N-point filter frequency response. The coefficients `AllpassNum` and `AllpassDen` represent the allpass mapping filter for mapping the prototype filter frequency `Wo` and the target frequencies vector `Wt`. Note that in this syntax `Hd` is a `dfilt` object with a lowpass magnitude response.

`[G,AllpassNum,AllpassDen] = iirlp2xn(...,Pass)` returns transformed `dfilt` object `G` with an IIR real N-point filter frequency response. This syntax allows you to specify an additional parameter, `Pass`, which chooses between using the "DC Mobility" and the "Nyquist Mobility." In the first case the Nyquist feature stays at its original location and the DC feature is free to move. In the second case the DC feature is kept at an original frequency and the Nyquist feature is allowed to move.

The coefficients `AllpassNum` and `AllpassDen` represent the allpass mapping filter for mapping the prototype filter frequency `Wo` and the target frequencies vector `Wt`. Note that in this syntax `Hd` is a `dfilt` object with a lowpass magnitude response.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

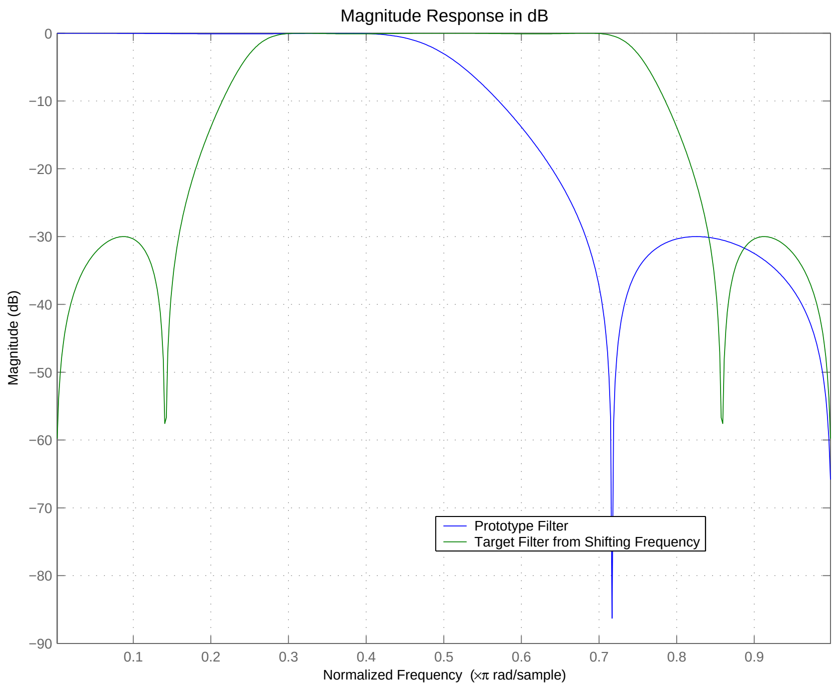
Move the cutoffs of the prototype filter to the new locations W_{t1}=0.25 and W_{t2}=0.75 creating a real bandpass filter:

```
[num, den] = iirlp2xn(b, a, [-0.5 0.5], [0.25 0.75], 'pass');
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```

`iirlp2xn` has created the desired bandpass filter with the cutoff locations specified in the command.



Arguments

`B`
Numerator of the prototype lowpass filter

`A`
Denominator of the prototype lowpass filter

`Wo`
Frequency values to be transformed from the prototype filter

`Wt`
Desired frequency locations in the transformed target filter

`Pass`
Choice (`'pass'`/`'stop'`) of passband/stopband at DC, `` `pass' `` being the default

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirftransf`, `allpasslp2xn`, `zpklp2xn`

References

[1] Cain, G.D., A. Krukowski and I. Kale, "High Order Transformations for Flexible IIR Filter Design," *VII European Signal Processing Conference (EUSIPCO'94)*, vol. 3, pp. 1582-1585, Edinburgh, United Kingdom, September 1994.

[2] Krukowski, A., G.D. Cain and I. Kale, "Custom designed high-order frequency transformations for IIR filters," *38th Midwest Symposium on Circuits and Systems (MWSCAS'95)*, Rio de Janeiro, Brazil, August 1995.

iirlpnorm

Purpose

Least P-norm optimal IIR filter design

Syntax

```
[num,den] = iirlpnorm(n,d,f,edges,a)
[num,den] = iirlpnorm(n,d,f,edges,a,w)
[num,den] = iirlpnorm(n,d,f,edges,a,w,p)
[num,den] = iirlpnorm(n,d,f,edges,a,w,p,dens)
[num,den] = iirlpnorm(n,d,f,edges,a,w,p,dens,initnum,initden)
```

Description

`[num,den] = iirlpnorm(n,d,f,edges,a)` returns a filter having a numerator order `n` and denominator order `d` which is the best approximation to the desired frequency response described by `f` and `a` in the least-pth sense. The vector `edges` specifies the band-edge frequencies for multi-band designs. An unconstrained quasi-Newton algorithm is employed and any poles or zeros that lie outside of the unit circle are reflected back inside. `n` and `d` should be chosen so that the zeros and poles are used effectively. See the "Hints" section. Always use `freqz` to check the resulting filter.

`[num,den] = iirlpnorm(n,d,f,edges,a,w)` uses the weights in `w` to weight the error. `w` has one entry per frequency point (the same length as `f` and `a`) which tells `iirlpnorm` how much emphasis to put on minimizing the error in the vicinity of each frequency point relative to the other points. `f` and `a` must have the same number of elements, which may exceed the number of elements in `edges`. This allows for the specification of filters having any gain contour within each band. The frequencies specified in `edges` must also appear in the vector `f`. For example,

```
[num,den] = iirlpnorm(5,12,[0 .15 .4 .5 1],[0 .4 .5 1],...
[1 1.6 1 0 0],[1 1 1 10 10])
```

is a lowpass filter with a peak of 1.6 within the passband.

`[num,den] = iirlpnorm(n,d,f,edges,a,w,p)` where `p` is a two-element vector `[pmin pmax]` allows for the specification of the minimum and maximum values of `p` used in the least-pth algorithm. Default is [2 128] which essentially yields the L-infinity, or Chebyshev, norm. `Pmin` and `pmax` should be even. If `p` is the string `'inspect'`, no optimization will occur. This can be used to inspect the initial pole/zero placement.

`[num,den] = iirlpnorm(n,d,f,edges,a,w,p,dens)` specifies the grid density `dens` used in the optimization. The number of grid points is `(dens*(n+d+1))`. The default is 20. `dens` can be specified as a single-element cell array. The grid is not equally spaced.

`[num,den] = iirlpnorm(n,d,f,edges,a,w,p,dens,initnum,initden)` allows for the specification of the initial estimate of the filter numerator and denominator coefficients in vectors `initnum` and `initden`. This may be useful for difficult optimization problems. The pole-zero editor in the Signal Processing Toolbox can be used for generating `initnum` and `initden`.

Hints

- This is a weighted least-pth optimization.
- Check the radii and locations of the poles and zeros for your filter. If the zeros are on the unit circle and the poles are well inside the unit circle, try increasing the order of the numerator or reducing the error weighting in the stopband.
- Similarly, if several poles have a large radii and the zeros are well inside of the unit circle, try increasing the order of the denominator or reducing the error weighting in the passband.

See Also

`iirlpnormc`, `filter`, `freqz`, `iirgrpdelay`, `zplane`

References

Antoniou, A., *Digital Filters: Analysis, Design, and Applications*, Second Edition, McGraw-Hill, Inc. 1993.

iirlpnormc

Purpose

Design a constrained least P-norm optimal IIR filter

Syntax

```
[num,den] = iirlpnormc(n,d,f,edges,a)
[num,den] = iirlpnormc(n,d,f,edges,a,w)
[num,den] = iirlpnormc(n,d,f,edges,a,w,radius)
[num,den] = iirlpnormc(n,d,f,edges,a,w,radius,p)
[num,den] = iirlpnormc(n,d,f,edges,a,w,radius,p,dens)
[num,den] = iirlpnormc(n,d,f,edges,a,w,radius,p,...
  dens,initnum,initden)
[num,den,err] = iirlpnormc(...)
[num,den,err,sos,g] = iirlpnormc(...)
```

Description

`[num,den] = iirlpnormc(n,d,f,edges,a)` returns a filter having a numerator order `n` and denominator order `d` which is the best approximation to the desired frequency response described by `f` and `a` in the least-pth sense. The vector `edges` specifies the band-edge frequencies for multi-band designs. A constrained Newton-type algorithm is employed. `n` and `d` should be chosen so that the zeros and poles are used effectively. See the “Hints” section. Always check the resulting filter using `fvtool`.

`[num,den] = iirlpnormc(n,d,f,edges,a,w)` uses the weights in `w` to weight the error. `w` has one entry per frequency point (the same length as `f` and `a`) which tells `iirlpnormc` how much emphasis to put on minimizing the error in the vicinity of each frequency point relative to the other points. `f` and `a` must have the same number of elements, which can exceed the number of elements in `edges`. This allows for the specification of filters having any gain contour within each band. The frequencies specified in `edges` must also appear in the vector `f`. For example,

```
[num,den] = iirlpnormc(5,5,[0 .15 .4 .5 1],[0 .4 .5 1],...
[1 1.6 1 0 0],[1 1 1 10 10])
```

designs a lowpass filter with a peak of 1.6 within the passband.

`[num,den] = iirlpnormc(n,d,f,edges,a,w,radius)` returns a filter having a maximum pole radius of `radius` where 0<`radius`<1. `radius` defaults to 0.999999. Filters that have a reduced pole radius may retain better transfer function accuracy after you quantize them.

`[num,den] = iirlpnormc(n,d,f,edges,a,w,radius,p)` where `p` is a two-element vector `[pmin pmax]` allows for the specification of the minimum and maximum values of `p` used in the least-pth algorithm. Default is [2 128] which essentially yields the L-infinity, or Chebyshev, norm. `pmin` and `pmax` should be even. If `p` is the string `'inspect'`, no optimization will occur. This can be used to inspect the initial pole/zero placement.

`[num,den] = iirlpnormc(n,d,f,edges,a,w,radius,p,dens)` specifies the grid density `dens` used in the optimization. The number of grid points is `(dens*(n+d+1))`. The default is 20. `dens` can be specified as a single-element cell array. The grid is not equally spaced.

`[num,den] = iirlpnormc(n,d,f,edges,a,w,radius,p,dens,...`
`initnum,initden)` allows for the specification of the initial estimate of the filter numerator and denominator coefficients in vectors `initnum` and `initden`. This may be useful for difficult optimization problems. The pole-zero editor in the Signal Processing Toolbox can be used for generating `initnum` and `initden`.

`[num,den,err] = iirlpnormc(...)` returns the least-Pth approximation error `err`.

`[num,den,err,sos,g] = iirlpnormc(...)` returns the second-order section representation in the matrix SOS and gain G. For numerical reasons you may find SOS and G beneficial in some cases.

Hints

- This is a weighted least-pth optimization.
- Check the radii and location of the resulting poles and zeros.
- If the zeros are all on the unit circle and the poles are well inside of the unit circle, try increasing the order of the numerator or reducing the error weighting in the stopband.
- Similarly, if several poles have a large radius and the zeros are well inside of the unit circle, try increasing the order of the denominator or reducing the error weight in the passband.
- If you reduce the pole radius, you might need to increase the order of the denominator.

The message

```
Poorly conditioned matrix. See the "help" file.
```

indicates that iirlpnormc cannot accurately compute the optimization because either:

a The approximation error is extremely small (try reducing the number of poles or zeros—refer to the hints above).

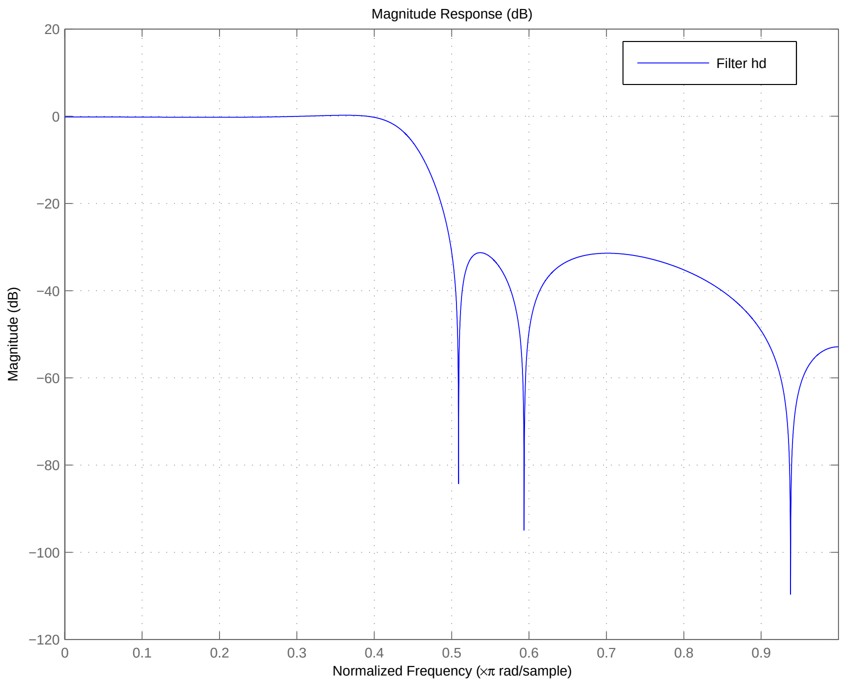
b The filter specifications have huge variation, such as a=[1 1e9 0 0].

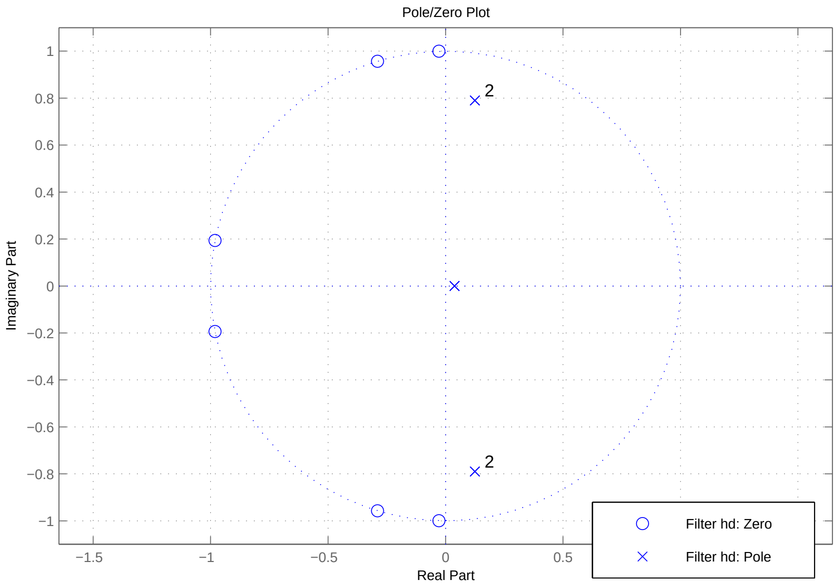
Examples

This example returns a lowpass filter whose pole radius is constrained to 0.8

```
[b,a,err,s,g] = iirlpnormc(6,6,[0 .4 .5 1],[0 .4 .5 1],...
[1 1 0 0],[1 1 1 1],.8);
hd = dfilt.df1sos(s,g); % Construct second-order sections filter.
fvtool(hd); % View filter's magnitude response
```

From the magnitude response shown here you see the lowpass nature of the filter. The pole/zero plot following shows that the poles are constrained to 0.8 as specified in the command.


Magnitude Response (dB)
Filter hd
Magnitude (dB)
20
0
−20
−40
−60
−80
−100
−120
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
Normalized Frequency (×π rad/sample)



See Also

`freqz`, `filter`, `iirgrpdelay`, `iirlpnorm`, `zplane`

References

Antoniou, A., *Digital Filters: Analysis, Design, and Applications*, Second Edition, McGraw-Hill, Inc. 1993.

Purpose

Design a second-order IIR notch digital filter

Syntax

```
[num,den] = iirnotch(w0,bw)
[num,den] = iirnotch(w0,bw,ab)
```

Description

`[num,den] = iirnotch(w0,bw)` turns a digital notching filter with the notch located at `w0`, and with the bandwidth at the -3 dB point set to `bw`. To design the filter, `w0` must meet the condition 0.0 < `w0` < 1.0, where 1.0 corresponds to π radians per sample in the frequency range.

The quality factor (Q factor) q for the filter is related to the filter bandwidth by $q = \omega_0/bw$ where ω_0 is `w0`, the frequency to remove from the signal.

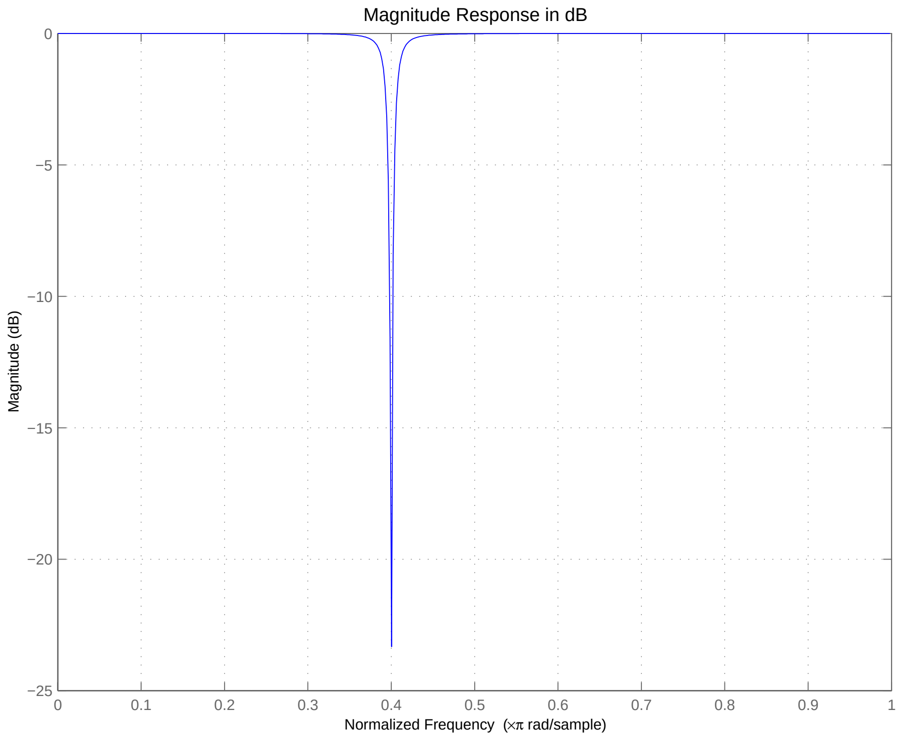
`[num,den] = iirnotch(w0,bw,ab)` returns a digital notching filter whose bandwidth, `bw`, is specified at a level of -`ab` decibels. Including the optional input argument `ab` lets you specify the magnitude response bandwidth at a level that is not the default -3dB point, such as -6 dB or 0 dB.

Examples

Design and plot an IIR notch filter that removes a 60 Hz tone (`f0`) from a signal at 300 Hz (`fs`). For this example, set the Q factor for the filter to 35 and use it to specify the filter bandwidth:

```
wo = 60/(300/2);  bw = wo/35;
[b,a] = iirnotch(wo,bw);
fvtool(b,a);
```

Shown in the next plot, the notch filter has the desired bandwidth with the notch located at 60 Hz, or 0.4π radians per sample. Compare this plot to the comb filter plot shown on the reference page for `iircomb`.



See Also

`firgr`, `iircomb`, `iirpeak`

Purpose

Design a second-order IIR peak or resonator digital filter

Syntax

```
[num,den] = iirpeak(w0,bw)
[num,den] = iirpeak(w0,bw,ab)
```

Description

`[num,den] = iirpeak(w0,bw)` turns a second-order digital peaking filter with the peak located at `w0`, and with the bandwidth at the +3dB point set to `bw`. To design the filter, `w0` must meet the condition 0.0 < `w0` < 1.0, where 1.0 corresponds to π radians per sample in the frequency range.

The quality factor (Q factor) q for the filter is related to the filter bandwidth by $q = \omega_0/bw$ where ω_0 is `w0` the signal frequency to boost.

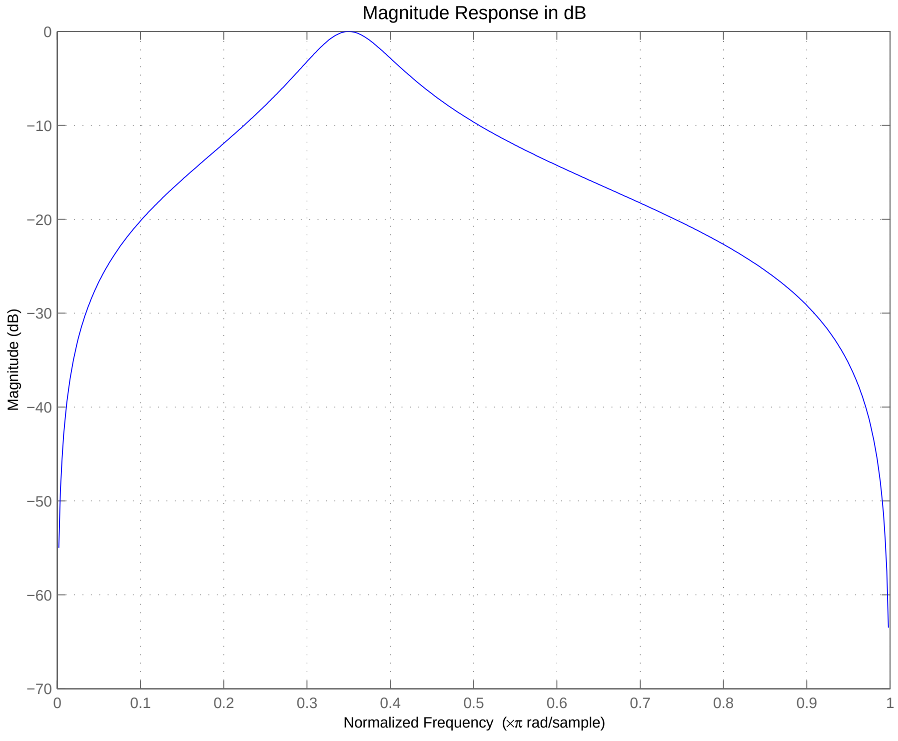
`[num,den] = iirpeak(w0,bw,ab)` returns a digital peaking filter whose bandwidth, `bw`, is specified at a level of +`ab` decibels. Including the optional input argument `ab` lets you specify the magnitude response bandwidth at a level that is not the default +3dB point, such as +6 dB or 0 dB.

Examples

Design and plot an IIR peaking filter that boosts the frequency at 1.75 Khz in a signal and has bandwidth of 500 Hz at the -3 dB point:

```
fs = 10000; wo = 1750/(fs/2);  bw = 500/(fs/2);
[b,a] = iirpeak(wo,bw);
fvtool(b,a);
```

Shown in the next plot, the peak filter has the desired gain and bandwidth at 1.75 KHz.



See Also

`firgr`, `iircomb`, `iirnotch`

Purpose

Compute power complementary filter.

Syntax

```
[bp,ap] = iirpowcomp(b,a)
[bp,ap,c] = iirpowcomp(b,a)
```

Description

`[bp,ap] = iirpowcomp(b,a)` returns the coefficients of the power complementary IIR filter g(z) = bp(z)/ap(z) in vectors `bp` and `ap`, given the coefficients of the IIR filter h(z) = b(z)/a(z) in vectors `b` and `a`. `b` must be symmetric (Hermitian) or antisymmetric (antihermitian) and of the same length as `a`. The two power complementary filters satisfy the relation

$$|H(w)|^2 + |G(w)|^2 = 1.$$

`[bp,ap,c] = iirpowcomp(b,a)` where `c` is a complex scalar of magnitude =1, forces `bp` to satisfy the generalized hermitian property

`conj(bp(end:-1:1)) = c*bp`.

When `c` is omitted, it is chosen as follows:

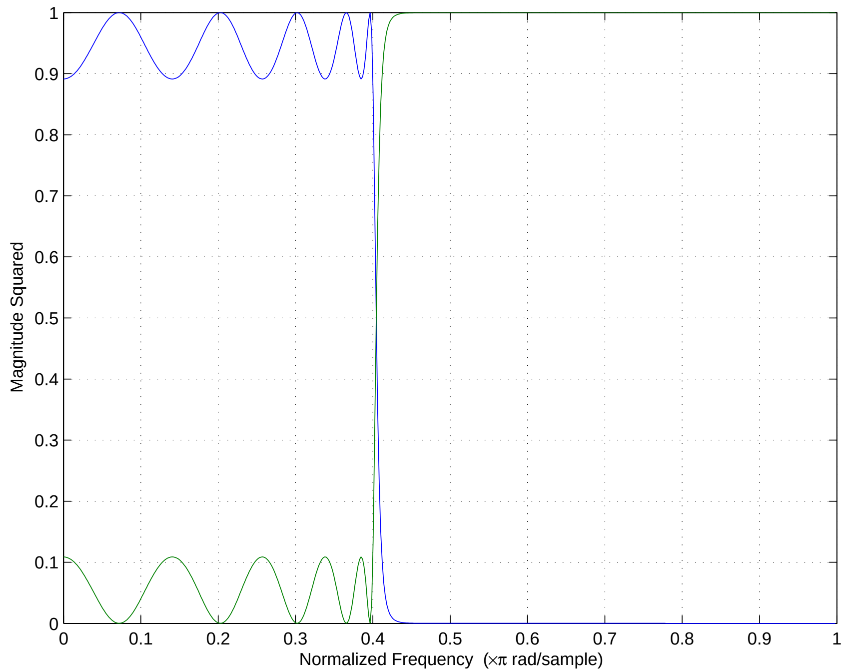
- When `b` is real, chooses C as 1 or -1, whichever yields `bp` real
- When `b` is complex, C defaults to 1

`ap` is always equal to `a`.

Examples

```
[b,a]=cheby1(10,.5,.4);
[bp,ap]=iirpowcomp(b,a);
[h,w,s]=freqz(b,a); [h1,w,s]=freqz(bp,ap);
s.plot='mag'; s.yunits='sq';freqzplot([h h1],w,s)
```

The next figure presents the results of applying iirpowcomp to the Chebyshev filter—the power complementary version of the original filter.



See Also

`tf2ca`, `tf2cl`, `ca2tf`, `cl2tf`

Purpose Upsample an IIR filter by an integer factor

Syntax

```
[Num,Den,AllpassNum,AllpassDen] = iirrateup(B,A,N)
```

Description `[Num,Den,AllpassNum,AllpassDen] = iirrateup(B,A,N)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter being transformed from any prototype by applying an `N`th-order rateup frequency transformation, where `N` is the upsample ratio. Transformation creates `N` equal replicas of the prototype filter frequency response.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with a numerator specified by `B` and a denominator specified by `A`.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

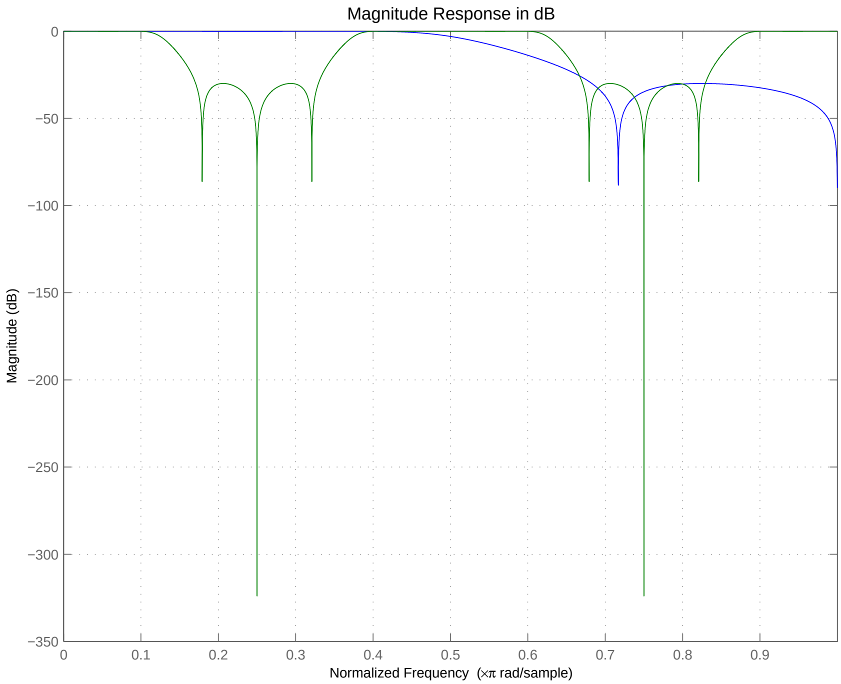
Examples Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
[num, den] = iirrateup(b, a, 4);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```

As shown in the figure produced by FVTool, the transformed filter appears as expected.



Arguments

`B`

Numerator of the prototype lowpass filter

`A`

Denominator of the prototype lowpass filter

`N`

Frequency multiplication ratio

`Num`

Numerator of the target filter

`Den`

Denominator of the target filter

`AllpassNum`

Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also

`iirftransf`, `allpassrateup`, `zpkrateup`

iirshift

Purpose Shift the frequency response of an IIR real filter

Syntax

```
[Num,Den,AllpassNum,AllpassDen] = iirshift(B,A,Wo,Wt)
```

Description `[Num,Den,AllpassNum,AllpassDen] = iirshift(B,A,Wo,Wt)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the real lowpass prototype by applying a second-order real shift frequency mapping.

It also returns the numerator, `AllpassNum`, and the denominator of the allpass mapping filter, `AllpassDen`. The prototype lowpass filter is given with the numerator specified by `B` and the denominator specified by `A`.

This transformation places one selected feature of an original filter located at frequency W_o to the required target frequency location, W_t. This transformation implements the "DC Mobility," which means that the Nyquist feature stays at Nyquist, but the DC feature moves to a location dependent on the selection of W_o and W_t.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the real shift transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can change their position in a simple way without designing them from the beginning.

Examples Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

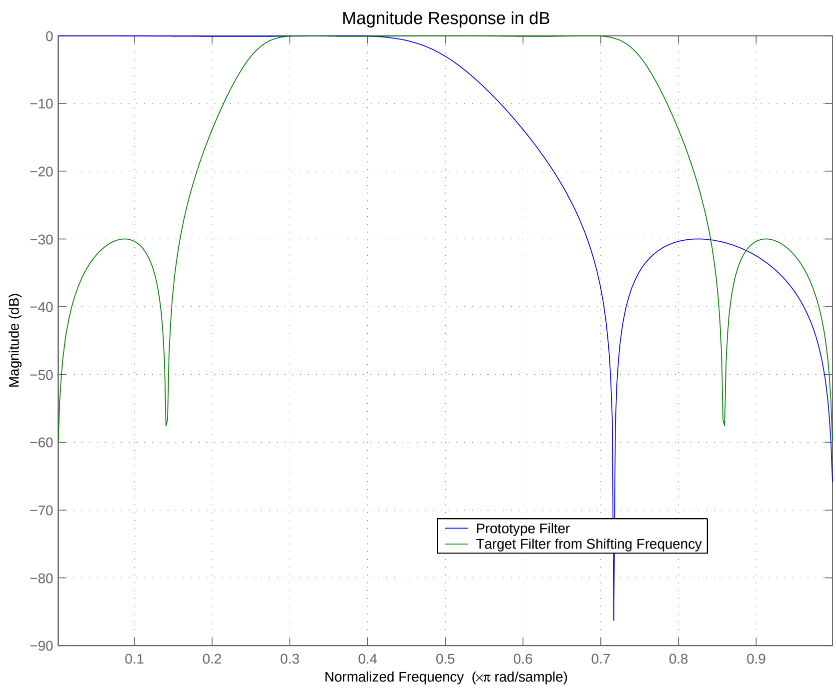
Perform the real frequency shift by defining where the selected feature of the prototype filter, originally at W_o=0.5, should be placed in the target filter, W_t=0.75:

```
Wo = 0.5; Wt = 0.75;
[num, den] = iirshift(b, a, Wo, Wt);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```

Shifting the specified feature from the prototype to the target generates the response shown in the figure.



Arguments

B

Numerator of the prototype lowpass filter

A

Denominator of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`iirftransf`, `allpassshift`, `zpkshift`.

Purpose Shift the frequency response of an IIR complex filter

Syntax

```
[Num,Den,AllpassNum,AllpassDen] = iirshiftc(B,A,Wo,Wt)
```

Description

`[Num,Den,AllpassNum,AllpassDen] = iirshiftc(B,A,Wo,Wc)` returns the numerator and denominator vectors, `Num` and `Den` respectively, of the target filter transformed from the real lowpass prototype by applying a first-order complex frequency shift transformation. This transformation rotates all the features of an original filter by the same amount specified by the location of the selected feature of the prototype filter, originally at W_o, placed at W_t in the target filter.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with the numerator specified by `B` and the denominator specified by `A`.

`[Num,Den,AllpassNum,AllpassDen] = iirshiftc(B,A,0,0.5)` calculates the allpass filter for doing the Hilbert transformation, i.e. a 90 degree counterclockwise rotation of an original filter in the frequency domain.

`[Num,Den,AllpassNum,AllpassDen] = iirshiftc(B,A,0,-0.5)` calculates the allpass filter for doing an inverse Hilbert transformation, i.e. a 90 degree clockwise rotation of an original filter in the frequency domain.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3, 0.1, 30, 0.409);
```

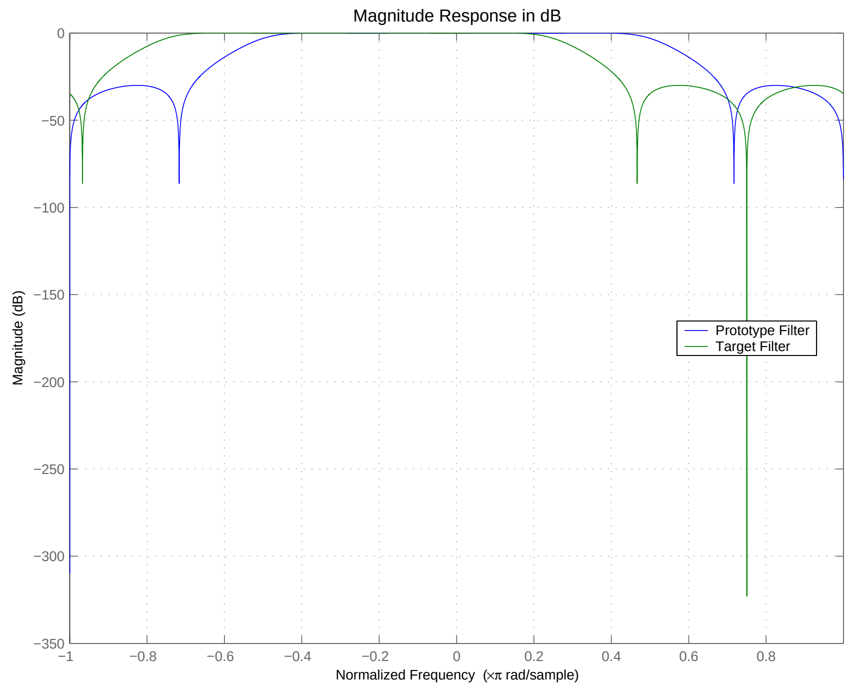
Rotate all features of the prototype filter in the frequency domain by the same amount by specifying where the selected feature of an original filter, W_o=0.5, should appear in the target filter, W_t=0.25:

```
[num, den] = iirshiftc(b, a, 0.5, 0.25);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, num, den);
```

After applying the shift, the selected feature from the original filter is just where it should be, at W_t = 0.25.



Arguments

`B`
Numerator of the prototype lowpass filter

`A`
Denominator of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Num`
Numerator of the target filter

`Den`
Denominator of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

See Also

`iirftransf`, `allpassshiftc`, `zpkshiftc`

References

[1] Oppenheim, A.V., R.W. Schafer and J.R. Buck, *Discrete-Time Signal Processing*, Prentice-Hall International Inc., 1989.

[2] Dutta-Roy, S.C. and B. Kumar, “On digital differentiators, Hilbert transformers, and half-band low-pass filters,” *IEEE Transactions on Education*, vol. 32, pp. 314-318, August 1989.

Purpose

Compute the impulse response of adaptive filter, discrete-time filters, and multirate filters

Syntax

```
[h,t] = impz(ha)
[h,t] = impz(...,fs)
impz(ha,...)
[h,t] = impz(hd)
[h,t] = impz(...,fs)
impz(hd,...)
[h,t] = impz(hm)
[h,t] = impz(...,fs)
impz(hm,...)
```

Description

The next sections describe common `impz` operation with adaptive, discrete-time, and multirate filters. For more input options, refer to `impz` in the Signal Processing Toolbox.

Adaptive Filters

For adaptive filters, `impz` returns the instantaneous impulse response based on the current filter coefficients.

`[h,t] = impz(ha)` computes the instantaneous impulse response of the adaptive filter `ha` choosing the number of samples for you, and returns the response in column vector `h` and a vector of times or sample intervals in `t` where (`t` = [0 1 2...]').

`[h,t] = impz(...,fs)` returns a matrix `h` if `ha` is a vector. Each column of the matrix corresponds to one filter in the vector. When `ha` is a vector of adaptive filters, `impz` returns the matrix `h`. Each column of `h` corresponds to one filter in the vector `ha`. If you provide a sampling frequency `fs` as an input argument, `impz` uses `fs` in when determining the impulse response.

`impz(ha,...)` uses FVTool to plot the impulse response of the adaptive filter `ha`. If `ha` is a vector of filters, `impz` plots the response and for each filter in the vector.

Discrete-Time Filters

[h,t] = impz(hd) computes the instantaneous impulse response of the discrete-time filter hd choosing the number of samples for you, and returns the response in column vector h and a vector of times or sample intervals in t where (t = [0 1 2...]'). impz returns a matrix h if hd is a vector. Each column of the matrix corresponds to one filter in the vector. When hd is a vector of discrete-time filters, impz returns the matrix h. Each column of h corresponds to one filter in the vector hd.

impz(hd) uses FVTool to plot the impulse response of the discrete-time filter hd. If hd is a vector of filters, impz plots the response and for each filter in the vector.

Multirate Filters

[h,t] = impz(hm) computes the instantaneous impulse response of the multirate filter hm choosing the number of samples for you, and returns the response in column vector h and a vector of times or sample intervals in t where (t = [0 1 2...]'). [h,t] = impz(hm) returns a matrix h if hm is a vector. Each column of the matrix corresponds to one filter in the vector. When hm is a vector of multirate filters, impz returns the matrix h. Each column of h corresponds to one filter in the vector ha.

impz(hm) uses FVTool to plot the impulse response of the multirate filter hm. If ha is a vector of filters, impz plots the response and for each filter in the vector.

Note that the multirate filter impulse response is computed relative to the rate at which the filter is running. When you specify fs (the sampling rate) as an input argument, impz assumes the filter is running at that rate.

For multistage cascades, impz forms a single-stage multirate filter that is equivalent to the cascade and computes the response relative to the rate at which the equivalent filter is running. impz does not support all multistage cascades. Only cascades for which it is possible to derive an equivalent single-stage filter are allowed for analysis.

As an example, consider a 2-stage interpolator where the first stage has an interpolation factor of 2 and the second stage has an interpolation factor of 4. An equivalent single-stage filter with an overall interpolation factor of 8 can be found. `impz` uses the equivalent filter for the analysis. If a sampling frequency `fs` is specified as an input argument to `impz`, the function interprets `fs` as the rate at which the equivalent filter is running.

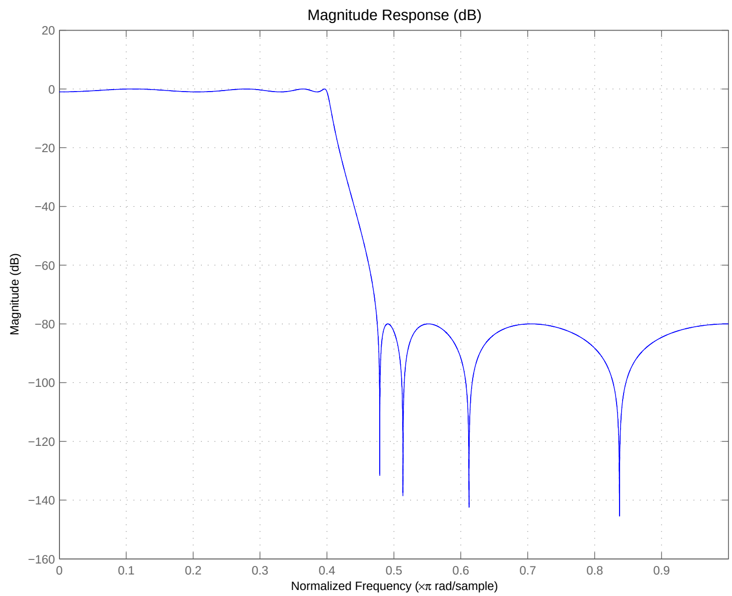
Note `impz` works for both real and complex filters. When you omit the output arguments, `impz` plots only the real part of the impulse response.

Examples

Create a discrete-time filter for a fourth-order, low-pass elliptic filter with a cutoff frequency of 0.4 times the Nyquist frequency. Use a second-order sections structure to resist quantization errors. Plot the first 50 samples of the impulse response, along with the reference impulse response.

```
% Create a design object for the prototype filter.

d = fdesign.lowpass(.4,.5,1,80)

d =

           ResponseType: 'Minimum-order lowpass'
      SpecificationType: 'Fp,Fst,Ap,Ast'
            Description: {4x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
                  Fpass: 0.4000
                  Fstop: 0.5000
                  Apass: 1
                  Astop: 80

% Use ellip to design the discrete-time filter in second-order.
% section form, order.

hd=ellip(d)

hd =

        FilterStructure: 'Direct-Form II, Second-Order Sections'
             Arithmetic: 'double'
```

```
              sosMatrix: [4x6 double]
            ScaleValues: [5x1 double]
  ResetBeforeFiltering: 'on'
                 States: [2x4 double]
   NumSamplesProcessed: 0
% Convert hd to fixed-point and check the impulse response
hd.arithmetic='fixed';
impz(hd)
```



See Also

`filter`

Purpose Return information about an adaptive, discrete-time, or multirate filter

Syntax

```
s = info(ha)
s = info(hd)
s = info(hm)
```

Description The next sections describe common `info` operation with adaptive, discrete-time, and multirate filters.

Adaptive Filters

`s = info(ha)` returns a string matrix with information about the filter `ha`.

Generally, `info` returns more information than the default display for the filter.

Discrete-Time Filters

`s = info(hd)` returns a string matrix with information about the filter `hd`.

Generally, `info` returns more information than the default display for the filter.

Multirate Filters

`s = info(hm)` returns a string matrix with information about the filter `hm`.

Generally, `info` returns more information than the default display for the filter.

In all instances, when the filter object uses fixed-point arithmetic, `info` returns additional information about the filter, including the arithmetic setting and details about the filter internals.

Examples Given two filters—`hd` and `hm`, use `info` to learn more about each filter. Here is `hd`, a discrete-time direct-form FIR filter

```
hd

hd =
```

```
      FilterStructure: 'Direct-Form FIR'
           Arithmetic: 'double'
            Numerator: [1x81 double]
     PersistentMemory: false
```

Similarly, here is a multirate CIC filter `hm`.

```
hm

hm =

        FilterStructure: 'Zero-Latency Cascaded Integrator-Comb
Interpolator'
      DifferentialDelay: 1
       NumberOfSections: 4
          InputBitWidth: 16
         OutputBitWidth: 16
    InterpolationFactor: 10
   ResetBeforeFiltering: 'on'
                 States: [2x4 int32]
    NumSamplesProcessed: 0
```

Now use `info` to get more details about both filters.

```
s=info(hd)

s =

Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure  : Direct-Form FIR
Filter Order      : 32
Stable            : Yes
Linear Phase      : Yes (Type 1)

s=info(hm)

s =

Discrete-Time FIR Multirate Filter (real)
-----------------------------------------
```

```
Filter Structure      : Zero-Latency Cascaded Integrator-Comb
Interpolator
Differential Delay    : 1
Number of Stages      : 4
Input Bit Width       : 16
Output Bit Width      : 16
Interpolation Factor  : 10
Filter Order          : 36
Stable                : Yes
Linear Phase          : Yes (Type 1)
```

If you convert your filter object, such as a `dfilt` or `mfilt`, to a fixed-point filter, `info` returns more information about the ranges provided by the fixed-point formats in the filter. After converting `hd` to fixed arithmetic, `info` returns this display:

```
info(hd)
Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure  : Direct-Form FIR
Filter Length     : 81
Stable            : Yes
Linear Phase      : Yes (Type 1)
Arithmetic        : fixed
Numerator         : s16,15 -> [-1 1)
Input             : s16,15 -> [-1 1)
Filter Internals  : Full Precision
  Output          : s39,30 -> [-256 256)  (auto determined)
  Product         : s32,30 -> [-2 2)  (auto determined)
  Accumulator     : s39,30 -> [-256 256)  (auto determined)
  Round Mode      : No rounding
  Overflow Mode   : No overflow
```

Similarly, the information displayed for a CIC decimator `hm` looks like this:

```
info(hm)

Discrete-Time FIR Multirate Filter (real)
-----------------------------------------
Filter Structure        : Cascaded Integrator-Comb Decimator
Decimation Factor       : 4
```

```
Differential Delay      : 1
Number of Sections      : 2
Stable                  : Yes
Linear Phase            : Yes (Type 1)

Input                   : s16,15
Output                  : s16,12
Filter Internals        : Minimum Word Lengths
  Integrator Section 1  : s19,15
  Integrator Section 2  : s18,14
  Comb Section 1        : s18,14
  Comb Section 2        : s17,13
```

See Also

`coefficients`, `isfir`, `isstable`, `islinphase`

`dfilt` in the Signal Processing Toolbox documentation

Purpose

Return the states from a CIC filter as a vector containing the numerator and denominator states for all sections of the filter

Syntax

```
integerstates = int(hm.states)
```

Description

`integerstates = int(hm.states)` returns the states of a CIC filter in vector form, rather than as the native `filtstates` object. An important point about int is that it quantizes the state values to the smallest number of bits possible while maintaining the state values.

Examples

For many users, the states of multirate filters are most useful as a vector, but the CIC filters store the states as objects. Here is how you get the states from you CIC filter as a vector.

hm = mfilt.cicdecim(

See Also

`filtstates.cic`, `mfilt.cicdecim`, `mfilt.cicinterp`

Purpose

Syntax

Description

Examples

See Also

isallpass

Purpose Test quantized filters to determine whether they are allpass structures

Syntax

```
flag = isallpass(f)
flag = isallpass(f,k)
```

Description `flag = isallpass(f)` determines whether the filter object `f` is an allpass filter, returning `1` if true and `0` if false.

`flag = isallpass(f,k)` determines whether the `k`-th section of the filter object `f` is an allpass section and returns `1` if true and `0` if false.

Since lattice coupled allpass filters always have allpass sections, this function always returns `1` for filters whose structure is `latticeca`.

See Also `isfir`, `islinphase`, `ismaxphase`, `isminphase`, `isreal`, `issos`, `isstable`

Purpose

Test quantized filters to see whether they are FIR filters

Syntax

```
flag = isfir(hq)
flag = isfir(hq,k)
```

Description

`flag = isfir(hq)` determines whether quantized filter `hq` is an FIR filter, returning `flag` equal to `1` when the quantized filter is an FIR filter, and `0` when it is IIR.

`flag = isfir(hq,k)` determines whether the `k`th-section of quantized filter `hq` is an FIR filter, returning `flag` equal to `1` when the `k`th-section is an FIR filter and `0` when it is IIR.

`isfir(hq)` looks at filter `hq` and determines whether the filter, in transfer function form, has a scalar for its denominator. If it does, it is an FIR filter.

Examples

```
hq = dfilt.dffir;
isfir(hq)
ans =

      1
```

returns `1` for the status of filter `hq`; the filter is an FIR structure with denominator reference coefficient equal to one.

See Also

`isallpass`, `islinphase`, `ismaxphase`, `isminphase`, `isreal`, `issos`, `isstable`

islinphase

Purpose Test quantized filters to see whether they are linear phase

Syntax

```
flag = islinphase(hq)
flag = islinphase(hq,k)
```

Description

`flag = islinphase(hq)` determines if the quantized filter `hq` is linear phase, and returns `1` if true and `0` if false.

`flag = islinphase(hq,k)` determines if the `k`th-section of the filter `hq` is a linear phase section and returns `1` if true and `0` if false.

The determination is based on the reference coefficients. A filter has linear phase if it is FIR and its transfer function coefficients are symmetric or antisymmetric. If it is IIR and it has poles on or outside the unit circle and both numerator and denominator are symmetric or antisymmetric, it is linear phase also.

Examples

This IIR filter has linear phase.

```
num=[1 0 0 0 0 -1];
den=[1 -1];
hq = dfilt.df2(num,den);
islinphase(hq)
ans =

     1
```

See Also

`isallpass`, `isfir`, `ismaxphase`, `isminphase`, `isreal`, `issos`, `isstable`

Purpose Test quantized filters to see whether they are maximum phase filters

Syntax

```
flag = ismaxphase(hq)
flag = ismaxphase(hq,k)
```

Description

`flag = ismaxphase(hq)` determines whether filter `hq` is maximum phase, returning `1` if true and `0` if false.

`flag = ismaxphase(hq,k)` determines if the `k`th-section of filter `hq` is a maximum phase section and returns `1` if true and `0` if false.

The determination is based on the reference coefficients. A filter is maximum phase when the zeros of its transfer function are on or outside the unit circle, or when the numerator is a scalar.

Examples

```
hq = dfilt.dffirt;
ismaxphase(hq)
```

returns `1` so this is a maximum phase quantized filter. Notice that the filter coefficients (zeros) are 1.0 before quantization. Compare to `isminphase`.

See Also `isallpass`, `isfir`, `islinphase`, `isminphase`, `isreal`, `issos`, `isstable`

isminphase

Purpose Test filters to see if they are minimum phase

Syntax

```
flag = isminphase(hq)
flag = isminphase(hq,k)
```

Description

`flag = isminphase(hq)` determines if the filter `hq` is minimum phase and returns `1` if true and `0` if false.

`flag = isminphase(hq,k)` determines if the `k`-th section of the filter `hq` is a minimum phase section and returns `1` if true and `0` if false.

The determination is based on the reference coefficients. A filter is minimum phase when the zeros of its transfer function are on or inside the unit circle, or the numerator is a scalar.

Examples

This example creates a minimum phase quantized filter.

```
hq = dfilt.dffir;
isminphase(hq)
```

If you look at the example in `ismaxphase`, you may notice that this filter is also maximum phase. Since both the poles and zeros of the filter lie on the unit circle, it passes the tests for minimum and maximum phase designation.

See Also

`isallpass`, `isfir`, `islinphase`, `ismaxphase`, `isreal`, `issos`, `isstable`,

Purpose Test quantized filters for purely real coefficients

Syntax `isreal(hq)`

Description `isreal(hq)` returns `1` (or true) if all reference filter coefficients for the quantized filter `hq` are real, and returns `0` (or false) otherwise.

`isreal(hq)` returns `1` if all filter coefficients in quantized filter `hq` have zero imaginary part. Otherwise, `isreal(hq)` returns a `0` indicating that the filter is complex. Complex quantized filters have one or more coefficients with nonzero imaginary parts.

Note Quantizing a filter cannot make a real filter into a complex filter.

Examples

```
% Create a reference filter.
[b,a] = ellip(2,0.5,20,0.4);

% Create a quantized filter from the reference filter.

hq = dfilt.df2t(b,a);
hq.arithmetic='fixed';
% Test if all filter coefficients are real.

r = isreal(hq)

r =
     1
```

See Also `isfir`, `islinphase`, `ismaxphase`, `isminphase`, `issos`, `isstable`, `isallpass`

Purpose Test whether quantized filters are composed of second-order sections

Syntax

```
flag = issos(hq)
```

Description `flag = issos(hq)` determines whether quantized filter `hq` consists of second-order sections. Returns `1` if all sections of quantized filter `hq` have order less than or equal to two, and `0` otherwise.

Examples

```
[b,a] = butter(8,.5);
hd = dfilt.df2(b,a);
hd.arithmetic='fixed';
hdsos = sos(Hd,'up',inf)
issos(hdsos)
v =

     1
```

Quantized filter `hdsos` is in second-order section form.

See Also `isallpass`, `isfir`, `islinphase`, `ismaxphase`, `isminphase`, `isreal`, `isstable`

Purpose Test whether a quantized filter is stable

Syntax

```
r = isstable(hq)
r = isstable(hq,k)
```

Description r = isstable(hq) tests quantized filter hq to determine whether its poles are inside the unit circle. If the poles lie on or outside the circle, isstable returns r = 0. If the poles are inside the circle, isstable returns r = 1.

r = isstable(hq,k) returns the stability of the kth-section of a multiple section quantized filter. Based on the locations of the poles of the specified section, isstable returns r = 1 if the filter section is stable, and 0 otherwise.

To determine the filter stability, isstable checks the quantized filter coefficients. When the poles lie on or inside the unit circle, the quantized filter is stable. FIR filters are stable by design since the defining transfer functions do not have denominator polynomials.

Examples Since filter stability is very important in your design process, use isstable to determine whether your quantized IIR filter is indeed stable:

```
hq = dfilt.df2t;
isstable(hq)
ans =
   1
```

See Also isallpass, isfir, islinphase, ismaxphase, isminphase, isreal, issos, zplane

kaiserwin

Purpose

Design digital filters or multirate filters using filter specifications from a filter design object and a Kaiser window

Syntax

```
hd = kaiserwin(d)
```

Description

`hd = kaiserwin(d)` designs a digital filter hd, or a multirate filter hm that uses a Kaiser window. For `kaiserwin` to work, the filter order in the design object must be even. In addition, higher order filters (filter order greater than 120) tend to be more accurate for smaller transition widths. `kaiserwin` returns a warning when your filter order may be too low to design your filter accurately.

Examples

This example designs a direct form FIR filter from a halfband filter design object.

```
d=fdesign.halfband('n,tw',100,0.004)

d =

          ResponseType: 'Halfband with filter order and transition width'
      SpecificationType: 'N,TW'
            Description: {2x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
            FilterOrder: 100
        TransitionWidth: 0.0040


hd=kaiserwin(d)
Warning: Filter order is too low. Design may be inaccurate.

hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'double'
               Numerator: [1x101 double]
    ResetBeforeFiltering: 'on'
                  States: [100x1 double]
     NumSamplesProcessed: 0
```

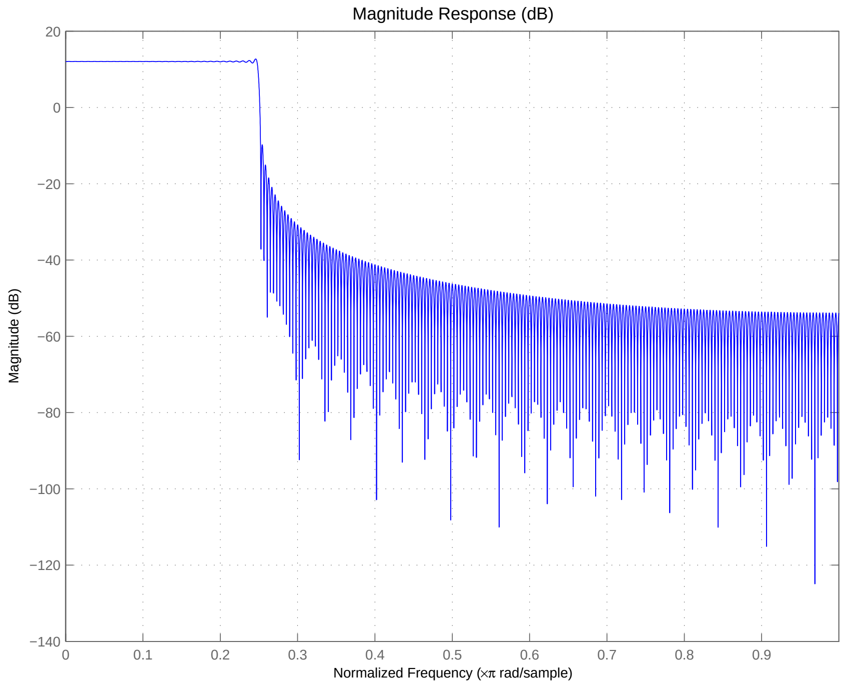
In this example, `kaiserwin` uses an interpolating filter design object to implement a multirate filter.

```
d=fdesign.interp(4,'pl,tw',120,0.004)
```

```
d =

          ResponseType: [1x46 char]
     SpecificationType: 'PL,TW'
           Description: {2x1 cell}
   InterpolationFactor: 4
   NormalizedFrequency: true
                    Fs: 'Normalized'
       PolyphaseLength: 120
       TransitionWidth: 0.0040


hm=kaiserwin(d)

hm =

        FilterStructure: 'Direct-Form FIR Polyphase Interpolator'
              Numerator: [1x480 double]
    InterpolationFactor: 4
   ResetBeforeFiltering: 'on'
                 States: [119x1 double]
    NumSamplesProcessed: 0
```

With the polyphase length of 120 you do not see the warning about the filter accuracy. Increasing the transition width `tw` can also reduce the possible inaccuracies.

FVTool shows clearly the multirate filter `hm`.



See Also

equiripple, firls

Purpose

Return the maximum step size that allows an adaptive filter to converge

Syntax

```
mumax = maxstep(ha,x)
[mumax,mumaxmse] = maxstep(ha,x)
```

Description

`mumax = maxstep(ha,x)` predicts a bound on the step size to provide convergence of the mean values of the adaptive filter coefficients. The columns of the matrix `x` contain individual input signal sequences. The signal set is assumed to have zero mean or nearly so.

`[mumax,mumaxmse] = maxstep(ha,x)` predicts a bound on the adaptive filter step size to provide convergence of the LMS adaptive filter coefficients in the mean-square sense. `maxstep` issues a warning when `ha.stepsize` is outside of the range 0 < `ha.stepsize` < `mumaxmse`/2.

Note `maxstep` is available for the following adaptive filter objects:
—`adaptfilt.blms`
—`adaptfilt.blmsfft`
—`adaptfilt.lms`
—`adaptfilt.nlms` (uses a different syntax. Refer to the text below.)
—`adaptfilt.se`

For `adaptfilt.nlms` filter objects, `maxstep` uses a slightly different syntax:

```
mumax = maxstep(ha)
[mumax,mumaxmse] = maxstep(ha)
```

The maximum step size for convergence is fully defined by the filter object `ha`. Matrix `x` is not necessary. If you include an `x` input matrix, MATLAB returns an error.

Examples

Analyze and simulate a 32-coefficient (31st-order) LMS adaptive filter object. To demonstrate the process, run 2000 iterations and 50 trials.

```
% Specify [numiterations,numexamples] = size(x);
x = zeros(2000,50);
d = x;
ha  = fir1(31,0.5);  % FIR system to be identified
```

```
for k=1:size(x,2);   % Create input and desired response signal
                     % matrices
% Set the (k)th input to the filter
x(:,k) = filter(sqrt(0.75),[1 -0.5],sign(randn(size(x,1),1)));
n = 0.1*randn(size(x,1),1);     % (k)th observation noise signal
d(:,k) = filter(ha,1,x(:,k))+n; % (k)th desired signal end
mu = 0.1;                       % LMS step size
ha = adaptfilt.lms(32,mu);
[mumax,mumaxmse] = maxstep(ha,x);
```

See Also `msepred`, `msesim`, `filter`

Purpose Construct a multirate filter object

Syntax

```
hm = mfilt.structure(input1,input2, )
```

Description

`hm = mfilt.`*`structure`*`(input1,input2, )` returns the object `hm` of type *`structure`*. As with `dfilt` and `adaptfilt` objects, you must include the *`structure`* string to construct a multirate filter object. You can, however, construct a default multirate filter object of a given structure by not including input arguments in your calling syntax.

Multirate filters include decimators and interpolators, and fractional decimators and fractional interpolators, meaning the resulting interpolation or decimation factor is not an integer.

Structures

Each of the following multirate filter structures has a reference page of its own.

Filter Structure String	Description of Resulting Multirate Filter
`mfilt.cascade`	Cascade multirate filters to form another filter
`mfilt.cicdecim`	Cascaded integrator-comb decimator
`mfilt.cicinterp`	Cascaded integrator-comb interpolator
`mfilt.fftfirinterp`	Overlap-add FIR polyphase interpolator
`mfilt.firdecim`	Direct-form FIR polyphase decimator
`mfilt.firfracdecim`	Direct-form FIR polyphase fractional decimator
`mfilt.firfracinterp`	Direct-form FIR polyphase fractional interpolator
`mfilt.firinterp`	Direct-form FIR polyphase interpolator
`mfilt.firsrc`	Direct-form FIR polyphase sample rate converter

Filter Structure String	Description of Resulting Multirate Filter
`mfilt.firtdecim`	Direct-form transposed FIR polyphase decimator
`mfilt.holdinterp`	FIR hold interpolator
`mfilt.linearinterp`	FIR Linear interpolator

Examples

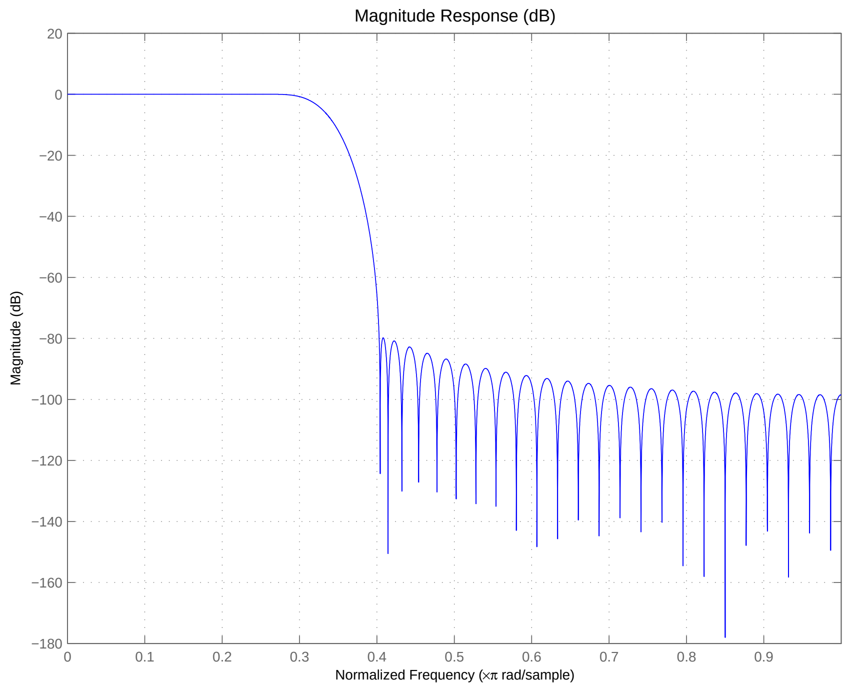
Create an FIR decimator that uses a decimation factor equal to three. In this case, the only input argument needed is m, the decimation factor. Other input arguments are available to you—refer to the reference page for the structure that interests you for more information.

m=3;

```
hm=mfilt.firdecim(m)

hm =

           FilterStructure: 'Direct-Form FIR Polyphase Decimator'
                 Numerator: [1x73 double]
          DecimationFactor: 3
  NumberOfSamplesProcessed: 0
               ResetStates: 'on'
                    States: [72x1 double]
```

To demonstrate a few of the methods that apply to multirate filters, here are two examples of using `hm`, your FIR decimator.

Use the Filter Visualization tool to review the magnitude response of your decimator.



Now check to see if your filter is stable.

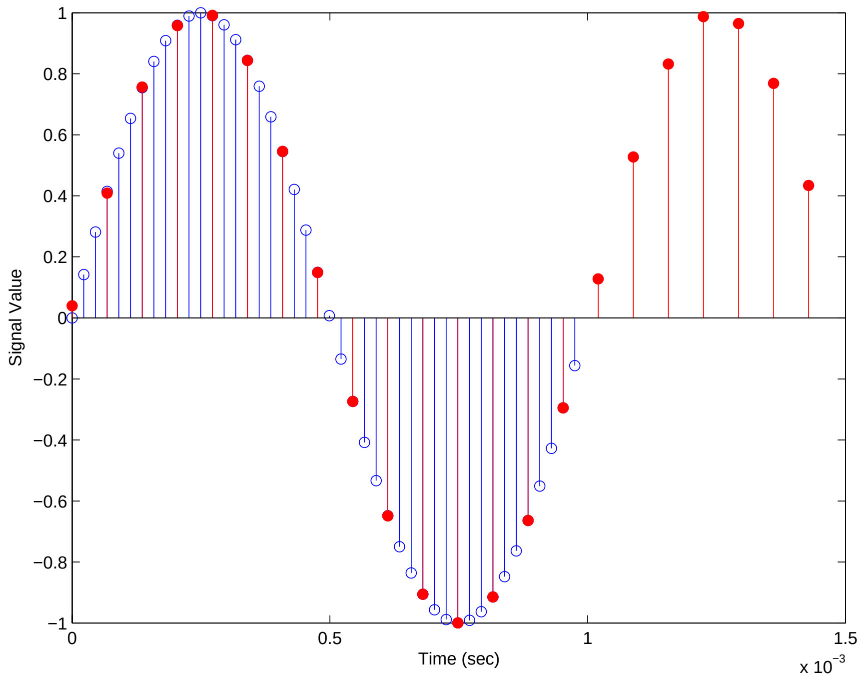
```
isstable(hm)

ans =

     1
```

Finally, pass a signal through the filter to see if it indeed decimates by three.

```
m = 3;                        % Decimation factor
hm = mfilt.firdecim(m);       % We use the default filter
fs = 44.1e3;                  % Original sample freq: 44.1kHz.
n = 0:10239;                 % 10240 samples, 0.232 second long
                              % signal
x  = sin(2*pi*1e3/fs*n);     % Original signal, sinusoid at 1 kHz
```

```
y = filter(hm,x);              % 5120 samples, still 0.232 seconds
stem(n(1:44)/fs,x(1:44))       % Plot original sampled at 44.1kHz
hold on                      % Plot decimated signal (22.05kHz) in red
stem(n(1:22)/(fs/m),y(13:34),'r','filled')
xlabel('Time (sec)');ylabel('Signal Value')
```

Here is the stem plot that shows the result of the decimation process.




```
hm =

         FilterStructure: 'Direct-Form FIR Polyphase Decimator'
               Numerator: [1x73 double]
        DecimationFactor: 3
        PersistentMemory: 'on'
                  States: [72x1 double]
```

```
        NumSamplesProcessed: 10239
```

Notice that the filter processed 10239 samples with 1 nonprocessed sample whose value is 0.8963. One nonprocessed sample results from dividing the number of samples, 10240, by the decimation factor, 3, to get 3413 output samples and one left over.

See Also

`mfilt.firfracdecim`, `mfilt.firfracinterp`, `mfilt.firinterp`, `mfilt.firsrc`, `mfilt.firtdecim`

mfilt.cascade

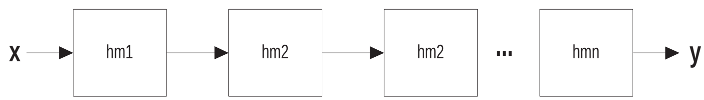
Purpose Cascade one or more dfilt and mfilt objects into a filter

Syntax

```
hm = cascade(hm1,hm2,...,hmn)
```

Description `hm = cascade(hm1,hm2,...,hmn)` creates filter object `hm` by cascading (connecting in series) the individual filter objects `hm1`, `hm2`, and so on to `hmn`.

In block diagram form, the cascade looks like this, with x as the input to the filter `hm` and y the output from the cascade filter `hm`:



Examples Create a variety of `mfilt` objects and cascade them together.

```
hm(1) = mfilt.firdecim(12);
hm(2) = mfilt.firdecim(4);
h1 = mfilt.cascade(hm(1),hm(2));

hm(3) = mfilt.firinterp(4);
hm(4) = mfilt.firinterp(12);
h2 = mfilt.cascade(hm(3),hm(4));
```

Now cascade `h1` and `h2` together to get another multirate filter.

```
h3 = mfilt.cascade(h1,h2,9600);
```

See Also `dfilt.cascade` in your Signal Processing Toolbox documentation

Purpose

Construct a fixed-point cascaded integrator-comb decimation filter object

Syntax

```
hm = mfilt.cicdecim(r,m,n,iwl,owl,wlps)
```

Description

`hm = mfilt.cicdecim(r,m,n,iwl,owl,wlps)` returns a cascaded integrator-comb (CIC) decimation filter object. All of the input arguments are optional. When you omit one or more input options, the object applies default values for the omitted input argument as shown in the next table.

The following table describes the input arguments for creating `hm`.

Input Arguments	Description
`r`	Decimation factor applied to the input signal. Sharpens the response curve to let you change the shape of the response. Default value is 2.
`m`	Differential delay. Changes both the shape and number of nulls in the filter response. Also affects the null locations. Increasing `m` increases the number and sharpness of the nulls and response between nulls. Generally, one or two work best as values for `m`. Default is 1.
`n`	Number of sections. Deepens the nulls in the response curve. Note that this is the number of either comb or integrator sections, not the total section count. 2 is the default value.
`iwl`	Word length of the input signal. Use any integer number of bits. The default value is 16 bits.

Input Arguments	Description
owl	Word length of the output signal. It can be any positive integer number of bits. By default, owl is 16 bits.
wlps	Defines the number of bits per word in each filter section while accumulating the data in the integrator sections or while subtracting the data during the comb sections (using 'wrap' arithmetic). Enter wlps as a scalar or vector of length 2*n, where n is the number of sections. When wlps is a scalar, the scalar value is applied to each filter section. The default is 16 for each section in the decimator. When you elect to specify wlps as an input argument, the SectionWordLengthMode property automatically switches from the default value of MinWordLengths to SpecifyWordLengths.

Constraints and Word Length Considerations

CIC decimators have the following constraint—the word lengths of the filter section must be monotonically decreasing. The word length of each filter section must be the same size as, or smaller than, the word length of the previous filter section.

The formula for B_{max}, the most significant bit at the filter output, is given in the Hogenauer paper in the References below.

$$B_{max} = (N\log_2 RM + B_{in} - 1)$$

where B_{in} is the number of bits of the input.

The cast operations shown in the diagram in "Algorithm" on page 9-737perform the changes between the word lengths of each section. When you specify word lengths that do not follow the constraints above, the constructor returns an error.

When you specify the word lengths correctly, the most significant bit B_{max} stays the same throughout the filter, while the word length of each section either decreases or stays the same. This can cause the fraction length to change throughout the filter as least significant bits are truncated to decrease the word length, as shown in "Algorithm" on page 9-737.

Properties of the Object

Objects have properties that control the way the object behaves. This table lists all the properties for the filter, with a description of each.

Name	Values	Default	Description
`Arithmetic`	`fixed`	`fixed`	Reports the kind of arithmetic the filter uses. CIC decimators are always fixed-point filters.
`DecimationFactor`	Any positive integer	2	Amount to reduce the input sampling rate.
`DifferentialDelay`	Any integer	1	Sets the differential delay for the filter. Usually a value of one or two is appropriate.
`FilterStructure`	`mfilt` structure string	None	Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of `mfilt` objects.
`InputFracLength`	Any positive integer	15	The number of bits applied to the fraction length to interpret the input data to the filter.

Name	Values	Default	Description
InputOffset	0 -> r.	0	Indicates the length of the output signal given the length of the input signal. InputOffset starts at zero and cycles through the phases as follows for each input sample: 0->r->(r-1)->(r-2)->(r-p)->0, where p = r-1.
InputWordLength	Any positive integer	16	The number of bits applied to the word length to interpret the input data to the filter.
NumSamplesProcessed (not part of the default display)	Any integer	0	Returns the number of samples processed during filtering. Use get to see this property. When PersistentMemory is false, this value resets to zero each time you use the filter, even in a loop. To accumulate the total number of samples processed for all the times you use the filter, set PersistentMemory to true.
NumberOfSections	Any positive integer	2	Number of sections used in the decimator. Generally called n. Reflects either the number of decimator or comb sections, not the total number of sections in the filter.

Name	Values	Default	Description
OutputFracLength	Any positive integer	15	The number of bits applied to the fraction length to interpret the output data from the filter. Read-only.
OutputWordLength	Any positive integer	16	The number of bits applied to the word length to interpret the output data from the filter.
PersistentMemory	false or true	false	Determines whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. When PersistentMemory is false, you cannot access the filter states. Setting PersistentMemory to true reveals the States property so you can modify the filter states.

Name	Values	Default	Description
SectionWordLengths	Any integer or a vector of length 2*n.	16	Defines the bits per section used while accumulating the data in the integrator sections or while subtracting the data during the comb sections (using 'wrap' arithmetic). Enter SectionWordLengths as a scalar or vector of length 2*n, where n is the number of sections. When SectionWordLengths is a scalar, the scalar value is applied to each filter section. When SectionWordLengths is a vector of values, the values apply to the sections in order. The default is 16 for each section in the decimator. Available when SectionWordLengthMode is SpecifyWordLengths.

Name	Values	Default	Description
`SectionWordLengthMode`	`MinWordLengths` or `SpecifyWordLengths`	`MinWordLength`	Determines whether the filter object sets the section word lengths or you provide the word lengths explicitly. By default, the filter uses the input and output word lengths in the command to determine the optimal word lengths for each section, according to the information in [1]. When you choose `SpecifyWordLengths`, you provide the word length for each section. In addition, choosing `SpecifyWordLengths` exposes the SectionWordLengths property for you to modify as needed.

Name	Values	Default	Description
States	filtstates.cic object	m+1-by-n matrix of zeros, after you call function int.	Stored conditions for the filter, including values for the integrator and comb sections before and after filtering. m is the differential delay of the comb section and n is the number of sections in the filter. The integrator states are stored in the first matrix row. States for the comb section fill the remaining rows in the matrix. Available for modification when PersistentMemory is true. Refer to the filtstates object in the Signal Processing Toolbox for more general information about the filtstates object.

About the States of the Filter

In the states property you find the states for both the integrator and comb portions of the filter. states is a matrix of dimensions m+1-by-n, with the states apportioned as follows:

- States for the integrator portion of the filter are stored in the first row of the state matrix.
- States for the comb portion fill the remaining rows in the state matrix..

To review the states of a CIC filter, use int to assign the states to a variable in MATLAB. As an example, here are the states for a CIC decimator hm before and after filtering a data set.

```
x = fi(ones(1,10),true,16,0); % Fixed-point input data.
hm = mfilt.cicdecim(2,1,2,16,16,16);
```

```
sts=int(hm.states)

sts =

     0     0
     0     0

set(hm,'InputFracLength',0); % Integer input specified.
y=filter(hm,x)

sts=int(hm.states)

sts =

    10    45
    28    13
```

STS is an integer matrix that `int` returns from the contents of the `filtstates.cic` object in `.

Design Considerations

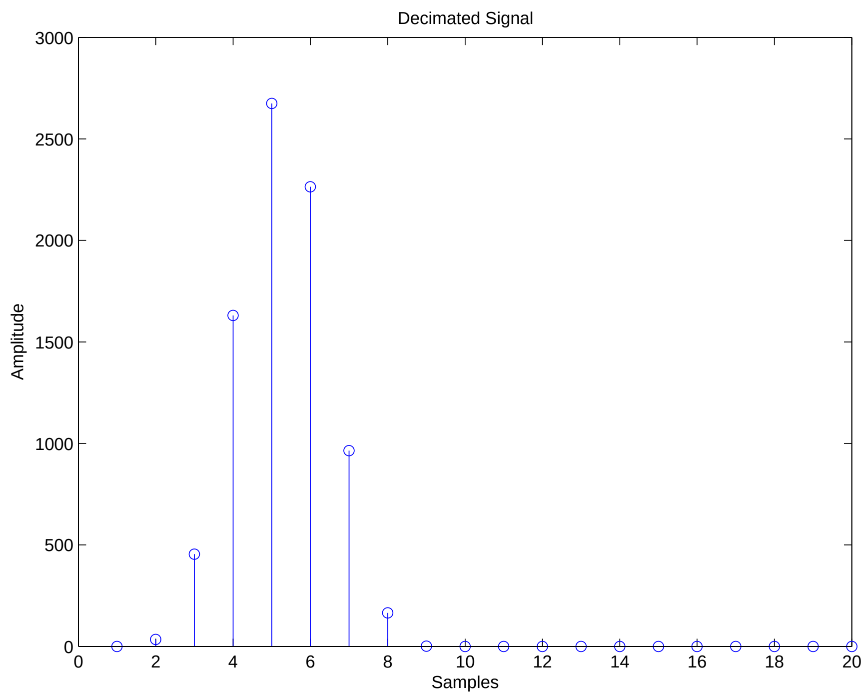
When you design your CIC decimation filter, remember the following general points:

- The filter output spectrum has nulls at $\omega = k * 2\pi/rm$ radians, $k = 1,2,3\ldots.$
- Aliasing and imaging occur in the vicinity of the nulls.
- `n`, the number of sections in the filter, determines the passband attenuation. Increasing `n` improves the filter ability to reject aliasing and imaging, but it also increases the droop (or rolloff) in the filter passband. Using an appropriate FIR filter in series after the CIC decimation filter can help you compensate for the induced droop.
- The DC gain for the filter is a function of the decimation factor. Raising the decimation factor increases the DC gain.

Examples

This example applies a decimation factor `r` equal to `8` to a 160-point impulse signal. The signal output from the filter has 160/`r`, or 20, points or samples. Choosing 10 bits for the word length represents a fairly common setting for analog to digital converters. The plot shown after the code presents the stem plot of the decimated signal, with 20 samples remaining after decimation:

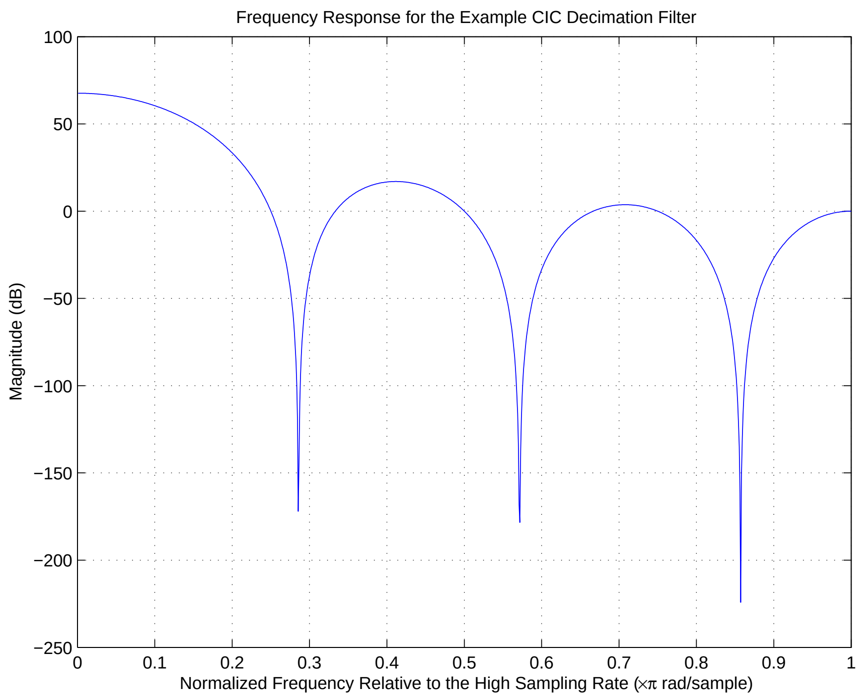
```
m = 2;  % Differential delays in the filter.
n = 4;  % Filter sections
r = 8   % Decimation factor
x = int16(zeros(160,1)); x(1) = 1;    % Create a 160-point
                                      % impulse signal.
hm = mfilt.cicdecim(r,m,n); % Expects 16-bit input by default.
y = filter(hm,x);
stem(double(y));                      % Plot the output as ...
                                      % a stem plot.
xlabel('Samples'); ylabel('Amplitude');
title('Decimated Signal');
```



The next example demonstrates one way to compute the filter frequency response, using a 4-section decimation filter with the decimation factor set to 7:

```
hm = mfilt.cicdecim(7,1,4);
fvtool(hm)
```

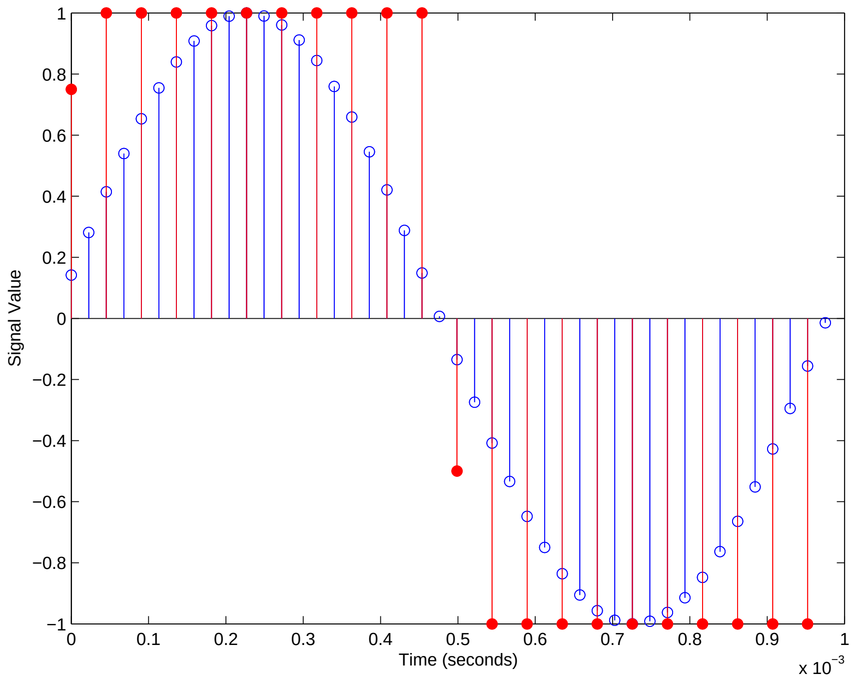
FVTool provides ways for you to change the title and x labels to match the figure shown. Here's the frequency response plot for the filter. For details about the transfer function used to produce the frequency response, refer to [1] in the References section.



This final example demonstrates the decimator for converting from 44.1 kHz audio to 22.05 kHz—decimation by two. To overlay the before and after signals, scale the output and plot the signals on a stem plot.

```
r = 2;                      % Decimation factor.
```

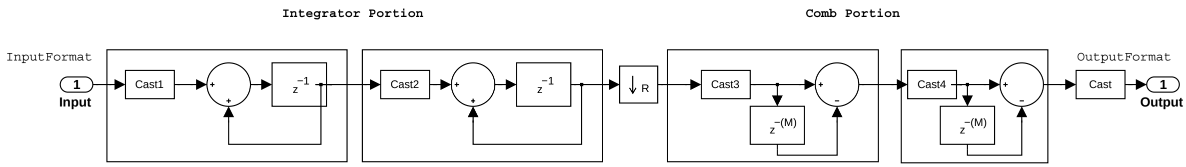
```
hm = mfilt.cicdecim(r); % Use default NumberOfSections &
                        % DifferentialDelay.
fs = 44.1e3;            % Original sample freq: 44.1kHz.
n = 0:10239;           % 10240 samples, 0.232 second long signal.
x  = sin(2*pi*1e3/fs*n);% Original signal, sinusoid at 1kHz.

% Scale input to use the full dynamic range of the int16 data type.
x = int16((2^15-1)*x/gain(hm));

y_int = filter(hm,x); % 5120 samples, still 0.232 seconds.

% Scale the input and output to overlay plots.
x = double(x); x = x/max(abs(x));
y = double(y_int); y = y/max(abs(y));
stem(n(1:44)/fs,x(2:45)); hold on;  % Plot original signal
                                      % sampled at 44.1kHz.
stem(n(1:22)/(fs/r),y(3:24),'r','filled'); % Plot decimated
                                           % signal (22.05kHz)
                                           % in red.
xlabel('Time (seconds)');ylabel('Signal Value');
```



Algorithm

To show how the CIC decimation filter is constructed, the following figure presents a block diagram of the filter structure for a two-section CIC decimation filter (n = 2). *fs* is the high sampling rate, the input to the decimation process.

For details about the bits that are removed in the Comb section, refer to [1] in References.



The word length and fraction length at each section of the decimator are shown in the following table. Either you specify the word length for each filter section in the `SectionWordLengths` property as a vector of integers, or you let the filter constructor set the word lengths by making `MinWordLengths` the value for `SectionWordLengthMode`. The calculation for each fraction length is shown below:

Decimator Word Lengths and Fraction Lengths

Position in the Signal Flow	Word Length	Fraction Length
Filter Input	`InputWL`	`InputFL`
1st section output	`sectionOneWL`	`InputFL`
2nd section output	`sectionTwoWL`	`InputFL + (sectionTwoWL - sectionOneWL)`
3rd section output	`sectionThreeWL`	`sectionTwoFL + (sectionThreeWL - sectionTwoWL)`
4th section output	`sectionFourWL`	`sectionThreeFL + (sectionFourWL - sectionThreeWL)`
Nth section output	`section(N)WL`	`section(N-1)FL + (section(N)WL - section(N-1)WL)`
Filter Output	`OutputWL`	`FinalsectionFL + (OutputWL - FinalsectionWL)`

See Also

`mfilt`, `mfilt.cicinterp`

References

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

[2] Meyer-Baese, Uwe, "Hogenauer CIC Filters," in *Digital Signal Processing with Field Programmable Gate Arrays*, Springer, 2001, pp. 155-172

mfilt.cicinterp

Purpose

Construct a fixed-point cascaded integrator-comb interpolation filter object

Syntax

```
hm = mfilt.cicinterp(r,m,n,ilw,owl,wlps)
```

Description

`hm = mfilt.cicinterp(r,m,n,ilw,owl,wlps)` constructs a cascaded integrator-comb (CIC) interpolation filter object that uses fixed-point arithmetic. All of the input arguments are optional. When you omit one or more input options, the omitted option applies default values shown in the table below.

The following table describes the input arguments for creating `hm`.

Input Arguments	Description
`r`	Interpolation factor applied to the input signal. Sharpens the response curve to let you change the shape of the response. 2 is the default value.
`m`	Differential delay. Changes both the shape and number of nulls in the filter response. Also affects the null locations. Increasing `m` increases the number and sharpness of the nulls and response between nulls. Generally, one or two work as values for `m`. 1 is the default.
`n`	Number of sections. Deepens the nulls in the response curve. Note that this is the number of either comb or integrator sections, not the total section count. By default, the filter has two sections.
`iwl`	Word length of the input signal. Use any integer number of bits. The default value is 16 bits.

Input Arguments	Description
`owl`	Word length of the output signal. It can be any positive integer number of bits. By default, `owl` is 16 bits.
`wlps`	Defines the number of bits per word in each filter section while accumulating the data in the integrator sections or while subtracting the data during the comb sections (using 'wrap' arithmetic). Enter `wlps` as a scalar or vector of length 2*`n`, where `n` is the number of sections. When `wlps` is a scalar, the scalar value is applied to each filter section. The default is 16 for each section in the integrator. When you elect to specify `wlps` as an input argument, the `SectionWordLengthMode` property automatically switches from the default value of `MinWordLengths` to `SpecifyWordLengths`.

Constraints and Conversions

In Hogenauer [1], the author describes the constraints on CIC interpolator filters. `mfilt.cicinterp` enforces a constraint—the word lengths of the filter sections must be nondecreasing. That is, the word length of each filter section must be the same size as, or greater than, the word length of the previous filter section.

The formula for W_j, the minimum register width, is derived in [1]. The formula for W_j is given by

$$W_j = \mathtt{ceil}(B_{in} + \log_2 G_j)$$

where G_j, the maximum register growth up to the jth section, is given by

$$G_j = \begin{cases} 2^j, & j = 1, 2, \ldots, N \\ \dfrac{2^{2N-j}(RM)^{j-N}}{R}, & j = N+1, \ldots, 2N \end{cases}$$

When the differential delay, M, is 1, there is also a special condition for the register width of the last comb, W_N, that is given by

$$W_N = B_{in} + N - 1 \qquad \text{if } M = 1$$

The conversions denoted by the cast blocks in the integrator diagrams in "Algorithm" on page 9-751 perform the changes between the word lengths of each section. When you specify word lengths that do not follow the constraints described in this section, `mfilt.cicinterp` returns an error.

The fraction lengths and scalings of the filter sections do not change. At each section the word length is either staying the same or increasing. The signal scaling can change at the output after the final filter section if you choose the output word length to be less than the word length of the final filter section.

Properties of the Object

Objects have properties that control the way the object behaves. This table lists all the properties for the filter, with a description of each.

Name	Values	Default	Description
`Arithmetic`	`fixed`	`fixed`	Reports the kind of arithmetic the filter uses. CIC interpolators are always fixed-point filters.
`InterpolationFactor`	Any positive integer	2	Amount to increase the input sampling rate.
`DifferentialDelay`	Any integer	1	Sets the differential delay for the filter. Usually a value of one or two is appropriate.

Name	Values	Default	Description
FilterStructure	mfilt structure string	None	Reports the type of filter object, such as a interpolator or fractional integrator. You cannot set this property—it is always read only and results from your choice of mfilt objects.
InputFracLength	Any positive integer	16	The number of bits applied as the fraction length to interpret the input data to the filter.
InputWordLength	Any positive integer	16	The number of bits applied to the word length to interpret the input data to the filter.
NumSamplesProcessed	Any integer	0	Returns the number of samples processed during filtering. When PersistentMemory is false, this value resets to zero each time you use the filter, even in a loop. To accumulate the total number of samples processed for all the times you use the filter, set PersistentMemory to true.

Name	Values	Default	Description
NumberOfSections	Any positive integer	2	Number of sections used in the interpolator. Generally called n. Reflects either the number of interpolator or comb sections, not the total number of sections in the filter.
OutputFracLength	Any positive integer	15	The number of bits applied to the fraction length to interpret the output data from the filter. Read-only.
OutputWordLength	Any positive integer	16	The number of bits applied to the word length to interpret the output data from the filter.

Name	Values	Default	Description
PersistentMemory	false or true	false	Determines whether the filter states get restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to zero any state that the filter changes during processing. States that the filter does not change are not affected. When PersistentMemory is false, you cannot access the filter states. Setting PersistentMemory to true reveals the States property so you can modify the filter states.

Name	Values	Default	Description
SectionWordLengths	Any integer or a vector of length 2*n.	16	Defines the bits per section used while accumulating the data in the integrator sections or while subtracting the data during the comb sections (using 'wrap' arithmetic). Enter SectionWordLengths as a scalar or vector of length 2*n, where n is the number of sections. When SectionWordLengths is a scalar, the scalar value is applied to each filter section. When SectionWordLengths is a vector of values, the values apply to the sections in order. The default is 16 for each section in the interpolator. Available when SectionWordLengthMode is SpecifyWordLengths.

Name	Values	Default	Description
SectionWordLengthMode	MinWordLengths or SpecifyWordLengths	MinWordLength	Determines whether the filter object sets the section word lengths or you provide the word lengths explicitly. By default, the filter uses the input and output word lengths in the command to determine the proper word lengths for each section, according to the information in [1]. When you choose SpecifyWordLengths, you provide the word length for each section. In addition, choosing SpecifyWordLengths exposes the SectionWordLengths property for you to modify as needed.

Name	Values	Default	Description
States	filtstates.cic object	m+1-by-n matrix of zeros, after you call function int.	Stored conditions for the filter, including values for the integrator and comb sections before and after filtering. m is the differential delay of the comb section and n is the number of sections in the filter. The integrator states are stored in the first matrix row. States for the comb section fill the remaining rows in the matrix. Available for modification when PersistentMemory is true. Refer to the filtstates object in the Signal Processing Toolbox for more general information about the filtstates object.

About the States of the Filter

In the states property you find the states for both the integrator and comb portions of the filter. states is a matrix of dimensions m+1-by-n, with the states apportioned as follows:

- States for the integrator portion of the filter are stored in the first row of the state matrix.
- States for the comb portion fill the remaining rows in the state matrix..

To review the states of a CIC filter, or any filter object states, use int to assign the states to a variable in MATLAB. As an example, here are the states for a CIC interpolator hm before and after filtering a data set.

```
x = fi(ones(1,10),true,16,0); % Fixed-point input data.
hm = mfilt.cicinterp(2,1,2,16,16,16);
```

```
sts=int(hm.states)

sts =

     0     0
     0     0

set(hm,'InputFracLength',0); % Integer input specified.
y=filter(hm,x)

sts=int(hm.states)

sts =

    10    45
    28    13
```

Design Considerations

When you design your CIC interpolation filter, remember the following general points:

- The filter output spectrum has nulls at $\omega = k * 2\pi/rm$ radians, $k = 1,2,3\ldots$.
- Aliasing and imaging occur in the vicinity of the nulls.
- n, the number of sections in the filter, determines the passband attenuation. Increasing n improves the filter ability to reject aliasing and imaging, but it also increases the droop or rolloff in the filter passband. Using an appropriate FIR filter in series after the CIC interpolation filter can help you compensate for the induced droop.
- The DC gain for the filter is a function of the interpolation factor. Raising the interpolation factor increases the DC gain.

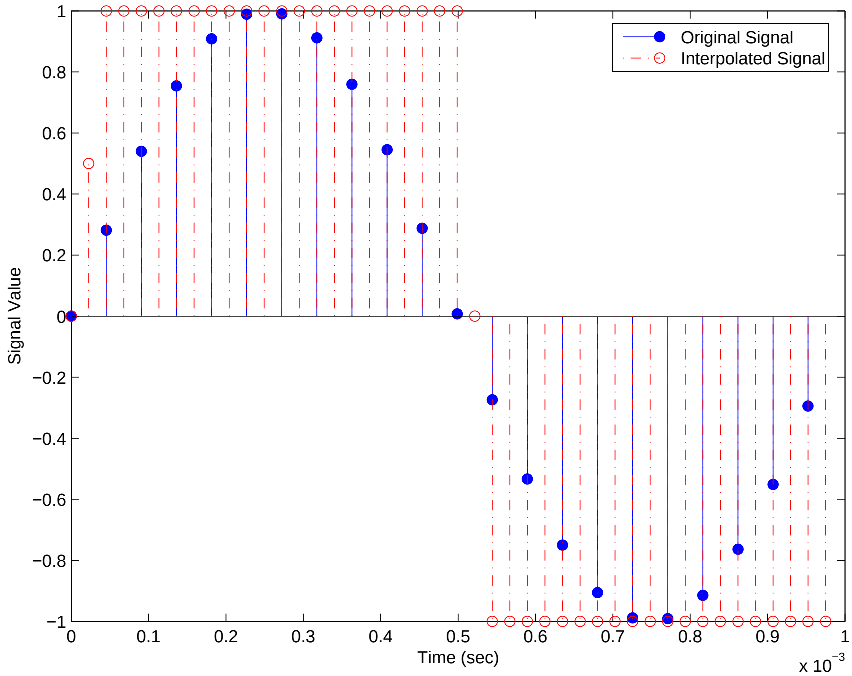
Examples

Demonstrate interpolation by a factor of two, in this case from 22.05 kHz to 44.1 kHz. Note the scaling required to see the results in the stem plot and to use the full range of the int16 data type.

```
R = 2;                     % Interpolation factor
hm = mfilt.cicinterp(R);   % Use default NumberOfSections and
                           % DifferentialDelay
fs = 22.05e3;              % Original sample freq:22.05 kHz.
```

```
n = 0:5119;              % 5120 samples, 0.232 second long signal
x  = sin(2*pi*1e3/fs*n);    % Original signal, sinusoid at 1 kHz

% Scale input to use the full dynamic range of the INT16 data type
x = int16((2^15-1)*x/gain(hm));

y_int = filter(hm,x); % 5120 samples, still 0.232 seconds

% Scale the input and output to overlay plots
x = double(x); x = x/max(abs(x));
y = double(y_int); y = y/max(abs(y));
stem(n(1:22)/fs,x(1:22),'filled'); % Plot original signal sampled
                                   % at 22.05 kHz
hold on;
stem(n(1:44)/(fs*R),y(4:47),'r');  % Plot interpolated signal
                                   % (44.1 kHz) in red
xlabel('Time (sec)');ylabel('Signal Value');
```

As you expect, the plot shows that the interpolated signal matches the input sine shape, with additional samples between each original sample.



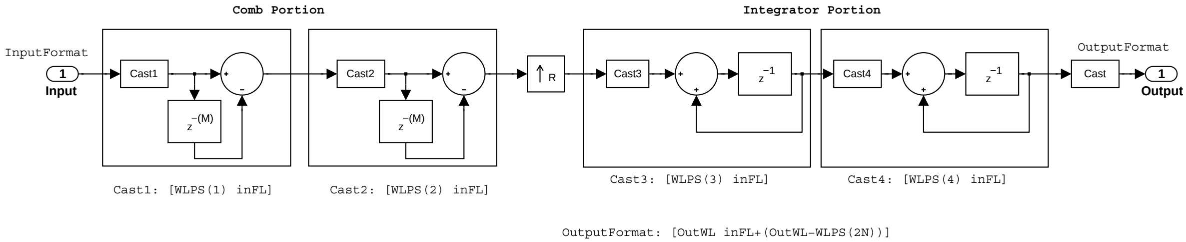
Use the filter visualization tool (FVTool) to plot the response of the interpolator object. For example, to plot the response of an interpolator with an interpolation factor of 7, 4 sections, and 1 differential delay, do something like the following:

```
hm = mfilt.cicinterp(7,1,4)
fvtool(hm)
```

Algorithm

To show how the CIC interpolation filter is constructed, the following figure presents a block diagram of the filter structure for a two-section CIC interpolation filter (n = 2). *fs* is the high sampling rate, the output from the interpolation process.

For details about the bits that are removed in the integrator section, refer to [1] in References.



The word length and fraction length at each section of the interpolator are shown in the following table. When you select `SpecifyWordLengths` for the `SectionWordLengthMode`, you specify the word length for each filter stage in the `SectionWordLengths` property. When you select `MinWordLengths`, the filter section word lengths are automatically set to the minimum number of bits possible in a valid CIC interpolator, in accordance with the formula for W_j in Constraints and Conversions below. The calculation for each fraction length is shown below:

Interpolator Word Lengths and Fraction Lengths

	Word Length	Fraction Length
Filter Input	`InputWL`	`InputFL`
1^{st} Section Output	`SectionOneWL`	`InputFL`
N^{th} Section Output	`Section(N)WL`	`InputFL`
Filter Output	`OutputWL`	`InputFL + (OutputWL - FinalsectionWL)`

References

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

[2] Meyer-Baese, Uwe, "Hogenauer CIC Filters," in Digital Signal Processing with Field Programmable Gate Arrays, Springer, 2001, pp. 155-172

Purpose Construct a an overlap-add FIR polyphase interpolator filter object

Syntax

```
hm = mfilt.fftfirinterp(l,num,bl)
```

Description

`hm = mfilt.fftfirinterp(l,num,bl)` returns a discrete-time FIR filter object that uses the overlap-add method for filtering input data.

The number of FFT points is given by `[bl+ceil(length(num)/l)-1]`. It is to your advantage to choose `bl` such that the number of FFT points is a power of two—using powers of two can improve the efficiency of the FFT and the associated interpolation process.

Input Arguments

The following table describes the input arguments for creating `hm`.

Input Argument	Description
`l`	Interpolation factor for the filter. `l` specifies the amount to increase the input sampling rate. It must be an integer. When you do not specify a value for `l` it defaults to 2.
`num`	Vector containing the coefficients of the FIR lowpass filter used for interpolation. When `num` is not provided as an input, `fftfirinterp` uses a lowpass Nyquist filter with gain equal to `l` and cutoff frequency equal to π`/l` by default.
`bl`	Length of each block of input data used in the filtering. `bl` must be an integer. When you omit input `bl`, it defaults to 100

mfilt.fftfirinterp Object Properties

Every multirate filter object has properties that govern the way it behaves when you use it. Note that many of the properties are also input arguments for creating `mfilt.fftfirinterp` objects. The next table describes each property for an `mfilt.fftfirinterp` filter object.

Name	Values	Description
`FilterStructure`		Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of `mfilt` object.
`Numerator`		Vector containing the coefficients of the FIR lowpass filter used for interpolation.
`InterpolationFactor`		Interpolation factor for the filter. It specifies the amount to increase the input sampling rate. It must be an integer.
`BlockLength`		Length of each block of input data used in the filtering.
`PersistentMemory`	`false` or `true`	Determines whether the filter states are restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. `PersistentMemory` returns to zero any state that the filter changes during processing. States that the filter does not change are not affected.

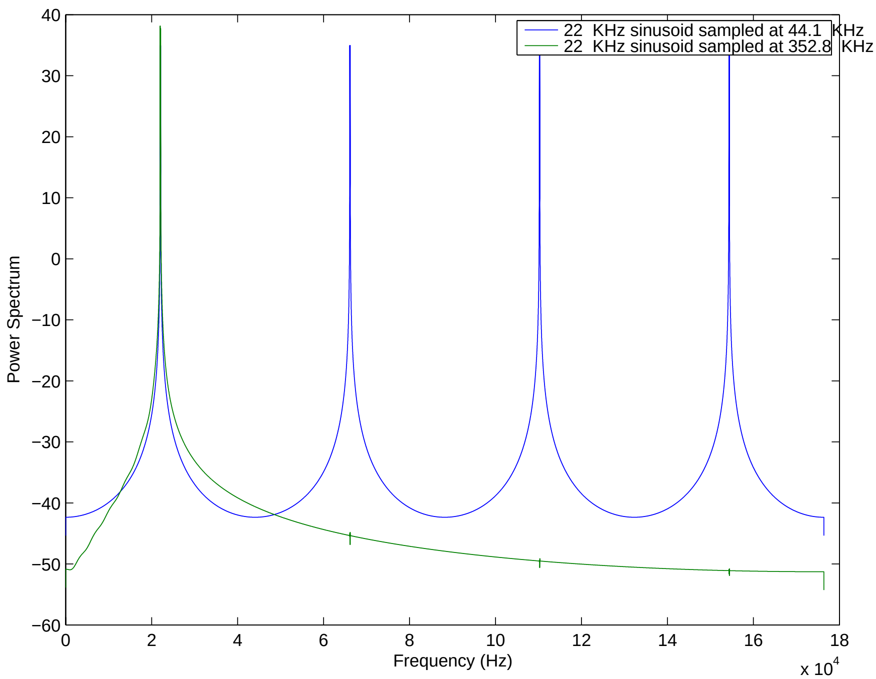
Name	Values	Description
States		Stored conditions for the filter, including values for the interpolator states.
NumSamplesProcessed		Returns the number of samples processed during filtering. As a check, the number of samples reported processed plus the number of nonprocessed samples should be the total number of input samples.

Examples

Interpolation by a factor of 8. Notice that this object removes the spectral replicas in the signal after interpolation.

```
l = 8;                          % Interpolation factor
hm = mfilt.fftfirinterp(l);     % We use the default filter
n = 8192;                       % Number of points
hm.blocklength = n;           % Set block length to number of points
fs = 44.1e3;                    % Original sample freq: 44.1 kHz.
n = 0:n-1;                      % 0.1858 secs of data
x = sin(2*pi*n*22e3/fs);      % Original signal, sinusoid at 22 kHz
y = filter(hm,x);               % Interpolated sinusoid
xu = l*upsample(x,8);         % Upsample to compare--the spectrum
                                % does not change
[px,f]=periodogram(xu,[],65536,l*fs);% Power spectrum of original
                                       % signal
[py,f]=periodogram(y,[],65536,l*fs); % Power spectrum of
                                       % interpolated signal
plot(f,10*log10(([fs*px,l*fs*py])))
legend('22  kHz sinusoid sampled at 44.1  kHz',...
'22  kHz sinusoid sampled at 352.8  kHz')
xlabel('Frequency (Hz)'); ylabel('Power Spectrum');
```

To see the results of the example, look at this figure.



See Also

`mfilt.firinterp`, `mfilt.holdinterp`, `mfilt.linearinterp`, `mfilt.firfracinterp`, `mfilt.cicinterp`

Purpose

Construct floating-point or fixed-point direct-form FIR polyphase decimator filters

Syntax

```
hm = mfilt.firdecim(m)
hm = mfilt.firdecim(m,num)
```

Description

`hm = mfilt.firdecim(m)` returns a direct-form FIR polyphase decimator object `hm` with a decimation factor of `m`. A lowpass Nyquist filter of gain 1 and cutoff frequency of `π/m` is designed by default.

`hm = mfilt.firdecim(m,num)` uses the coefficients specified by `num` for the decimation filter. This lets you specify more completely the FIR filter to use for the decimator.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hm` as follows:

- To change to single-precision filtering, enter

  ```
  set(hm,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hm,'arithmetic','fixed');
  ```

Input Arguments

The following table describes the input arguments for creating `hm`.

Input Argument	Description
`m`	Decimation factor for the filter. `m` specifies the amount to reduce the sampling rate of the input signal. It must be an integer. When you do not specify a value for `m` it defaults to 2.
`num`	Vector containing the coefficients of the FIR lowpass filter used for decimation. When `num` is not provided as an input, `mfilt.firdecim` constructs a lowpass Nyquist filter with gain of 1 and cutoff frequency equal to π/`m` by default. The default length for the Nyquist filter is 24*`m`. Therefore, each polyphase filter component has length 24.

Object Properties

This section describes the properties for both floating-point filters (double-precision and single-precision) and fixed-point filters.

Floating-Point Filter Properties

Every multirate filter object has properties that govern the way it behaves when you use it. Note that many of the properties are also input arguments for creating `mfilt.firdecim` objects. The next table describes each property for an `mfilt.firdecim` filter object.

Name	Values	Description
`Arithmetic`	`Double`, `single`, `fixed`	Defines the arithmetic the filter uses. Gives you the options `double`, `single`, and `fixed`. In short, this property defines the operation mode for your filter.
`DecimationFactor`	Integer	Decimation factor for the filter. `m` specifies the amount to reduce the sampling rate of the input signal. It must be an integer.

Name	Values	Description
FilterStructure	String	Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of mfilt object. Describes the signal flow for the filter object.
InputOffset	Integers	Contains a value derived from the number of input samples and the decimation factor— InputOffset = mod(length(nx),m) where nx is the number of input samples that have been processed so far and m is the decimation factor.
Numerator	Vector	Vector containing the coefficients of the FIR lowpass filter used for decimation.
PersistentMemory	false, true	Determines whether the filter states get restored to zeros for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory set to false returns filter states to the default values after filtering. States that the filter does not change are not affected. Setting this to true allows you to modify the States, InputOffset, and PolyphaseAccum properties.

Name	Values	Description
PolyphaseAccum	0 in double, single, or fixed for the different filter arithmetic settings.	Differentiates between the adders in the filter that work in full precision at all times (PolyphaseAccum) and the adders in the filter that the user controls and that may introduce quantization effects when FilterInternals is set to SpecifyPrecision.
States	Double, single, or fi matching the filter arithmetic setting.	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Double is the default setting for floating-point filters in double arithmetic.

Fixed-Point Filter Properties

This table shows the properties associated with the fixed-point implementation of the filter. You see one or more of these properties when you set Arithmetic to fixed. Notice that some of the properties have different default values when they refer fixed point filters. One example is the property PolyphaseAccum which stores data as doubles when you use your filter in double-precision mode, but stores a fi object in fixed-point mode.

Note The table lists all of the properties that a fixed-point filter can have. Many of the properties listed are dynamic, meaning they exist only in response to the settings of other properties.

To view all of the characteristics for a filter at any time, use

```
info(hm)
```

where hm is a filter.

For further information about the properties of this filter or any mfilt object, refer to "Multirate Filter Properties" on page 8-122.

Name	Values	Description
AccumFracLength	Any positive or negative integer number of bits [32]	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters.
AccumWordLength	Any integer number of bits [39]	Sets the word length used to store data in the accumulator.
Arithmetic	fixed for fixed-point filters	Setting this to fixed allows you to modify other filter properties to customize your fixed-point filter.
CoeffAutoScale	[true], false	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength property value to specify the precision used.
CoeffWordLength	Any integer number of bits [16]	Specifies the word length to apply to filter coefficients.

Name	Values	Description
FilterInternals	[FullPrecision], SpecifyPrecision	Controls whether the filter automatically sets the output word and fraction lengths, product word and fraction lengths, and the accumulator word and fraction lengths to maintain the best precision results during filtering. The default value, FullPrecision, sets automatic word and fraction length determination by the filter. SpecifyPrecision makes the output and accumulator-related properties available so you can set your own word and fraction lengths for them.
InputFracLength	Any positive or negative integer number of bits [15]	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Any integer number of bits[16]	Specifies the word length applied to interpret input data.
OutputFracLength	Any positive or negative integer number of bits [32]	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set FilterInternals to SpecifyPrecision.
OutputWordLength	Any integer number of bits [39]	Determines the word length used for the output data. You make this property editable by setting FilterInternals to SpecifyPrecision.

Name	Values	Description
OverflowMode	saturate, [wrap]	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic.) The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

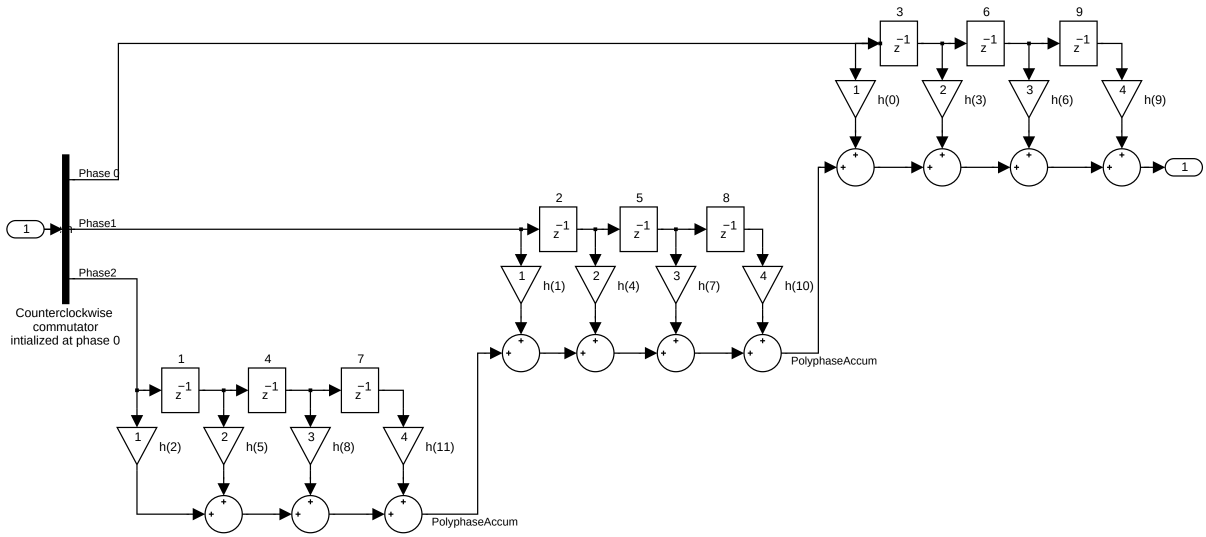
Name	Values	Description
RoundMode	[convergent], ceil,fix,floor, round	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Name	Values	Description
`Signed`	`[true]`, `false`	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
`States`	`fi` object	This property contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use `fi` objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system. For information about the ordering of the states, refer to the filter structure section.

Filter Structure

To provide decimation, `mfilt.firdecim` uses the following structure. At the input you see a commutator that operates counterclockwise, moving from position 0 to position 2, position 1, and back to position 0 as input samples enter the filter.

The figure below details the signal flow for the direct form FIR filter implemented by `mfilt.firdecim`.



Notice the order of the states in the filter flow diagram. States 1 through 9 appear in the diagram above each delay element. State 1 applies to the first delay element in phase 2. State 2 applies to the first delay element in phase 1. State 3 applies to the first delay element in phase 0. State 4 applies to the second delay in phase 2, and so on. When you provide the states for the filter as a vector to the `States` property, the above description explains how the filter assigns the states you specify.

In property value form, the states for a filter `hm` are

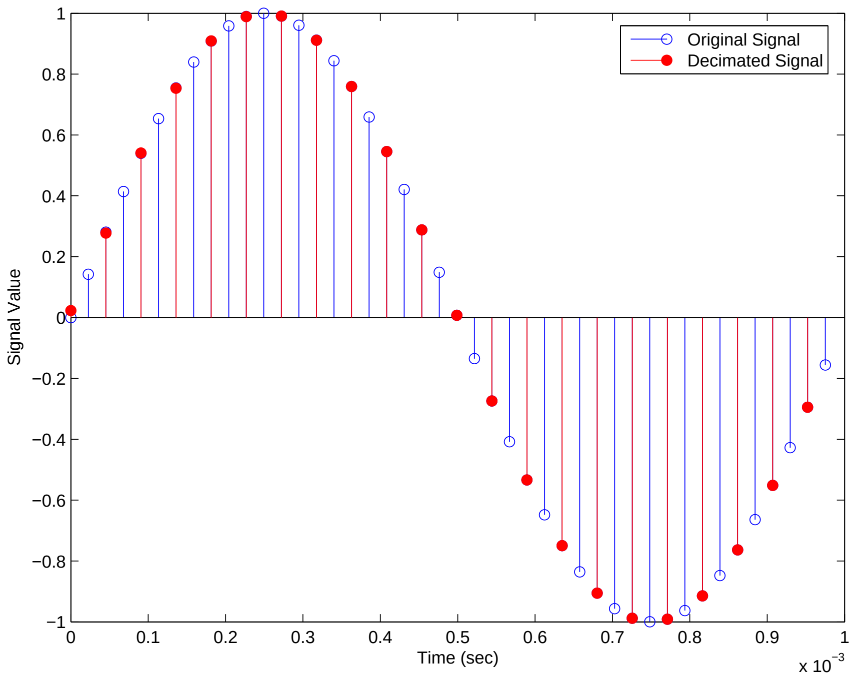
```
hm.states=[1:9];
```

Examples

Convert an input signal from 44.1 kHz to 22.05 kHz using decimation by a factor of 2. In the figure that appears after the example code, you see the results of the decimation.

```
m = 2;                        % Decimation factor.
hm = mfilt.firdecim(m);       % Use the default filter.
fs = 44.1e3;                  % Original sample freq: 44.1kHz.
n = 0:10239;                 % 10240 samples, 0.232 second long
                              % signal.
```

```
x  = sin(2*pi*1e3/fs*n);      % Original signal--sinusoid at 1kHz.
y = filter(hm,x);              % 5120 samples, 0.232 seconds.
stem(n(1:44)/fs,x(1:44))      % Plot original sampled at 44.1 kHz.
hold on                       % Plot decimated signal (22.05 kHz)
                               % in red.
stem(n(1:22)/(fs/m),y(13:34),'r','filled')
xlabel('Time (sec)');ylabel('Signal Value')
```



See Also

`mfilt.firtdecim`, `mfilt.firfracdecim`, `mfilt.cicdecim`

mfilt.firfracdecim

Purpose Construct a direct-form FIR polyphase fractional decimator filter object

Syntax `hm = mfilt.firfracdecim(l,m,num)`

Description `hm = mfilt.firfracdecim(l,m,num)` returns a direct-form FIR polyphase fractional decimator. Input argument `l` is the interpolation factor. `l` must be an integer. When you omit `l` in the calling syntax, it defaults to 2. `m` is the decimation factor. It must be an integer. If not specified, it defaults to l+1.

`num` is a vector containing the coefficients of the FIR lowpass filter used for decimation. If omitted, a lowpass Nyquist filter of gain l and cutoff frequency of π/`max(l,m)` is used by default.

By specifying both a decimation factor and an interpolation factor, you can decimate your input signal by noninteger amounts. The fractional decimator first interpolates the input, then decimates to result in an output signal whose sample rate is `l/m` of the input rate. By default, the resulting decimation factor is 3/2 when you do not provide `l` and `m` in the calling syntax. Specify `l` smaller than `m` for proper decimation.

Input Arguments

The following table describes the input arguments for creating `hm`.

Input Argument	Description
`l`	Interpolation factor for the filter. It must be an integer. When you do not specify a value for `l` it defaults to 2.
`num`	Vector containing the coefficients of the FIR lowpass filter used for interpolation. When `num` is not provided as an input, `firfracdecim` uses a lowpass Nyquist filter with gain equal to `l` and cutoff frequency equal to π/`max(l,m)` by default.
`m`	Decimation factor for the filter. `m` specifies the amount to reduce the sampling rate of the input signal. It must be an integer. When you do not specify a value for `m` it defaults to `l` + 1.

mfilt.firfracdecim Object Properties

Every multirate filter object has properties that govern the way it behaves when you use it. Note that many of the properties are also input arguments for creating `mfilt.firfracdecim` objects. The next table describes each property for an `mfilt.firfracdecim` filter object.

Name	Values	Description
`FilterStructure`	String	Reports the type of filter object, such as a decimator or fractional decimator. You cannot set this property—it is always read only and results from your choice of `mfilt` object.
`Numerator`	Vector	Vector containing the coefficients of the FIR lowpass filter used for interpolation.
`RateChangeFactors`	`[l,m]`	Reports the decimation (`m`) and interpolation (`l`) factors for the filter object. Combining these factors results in the final rate change for the signal.
`PersistentMemory`	`false` or `true`	Determines whether the filter states are restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. `PersistentMemory` returns to zero any state that the filter changes during processing. States that the filter does not change are not affected.

Name	Values	Description
`States`	Matrix	Stored conditions for the delays between each interpolator phase, the filter states, and the states at the output of each phase in the filter. The number of states is (`lh`-1)*`m`+(l-1)*(`lo`+`mo`) where `lh` is the length of each subfilter, and `l` and `m` are the interpolation and decimation factors. `lo` and `mo`, the input and output delays between each interpolation phase, are integers from Euclid's theorem such that `lo`*l-`mo`*`m` = -1 (refer to the reference for more details). Use `euclidfactors` to get `lo` and `mo` for an `mfilt.firfracdecim` object
`NumSamplesProcessed`	Integer	Returns the number of samples processed during filtering. As a check, the number of samples reported processed plus the number of nonprocessed samples should be the total number of input samples.

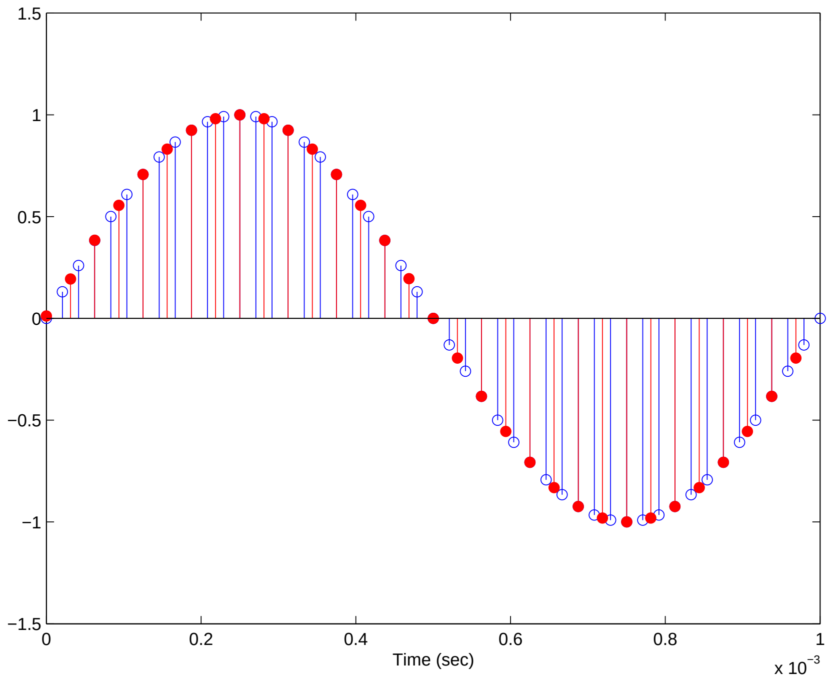
Example

To demonstrate `firfracdecim`, perform a fractional decimation by a factor of 2/3. This is one way to downsample a 48 kHz signal to 32 kHz, commonly done in audio processing.

```
l  = 2; m = 3;                % Interpolation/decimation factors.
hm = mfilt.firfracdecim(l,m); % We use the default
fs = 48e3;                    % Original sample freq: 48 kHz.
n  = 0:10239;                 % 10240 samples, 0.213 second long
                              % signal
```

```
x  = sin(2*pi*1e3/fs*n);      % Original signal, sinusoid at 1 kHz
y  = filter(hm,x);            % 9408 samples, still 0.213 seconds
stem(n(1:49)/fs,x(1:49)); hold on; % Plot original signal sampled
                                   % at 48 kHz
stem(n(1:32)/(fs*l/m),y(13:44),'r','filled') % Plot decimated
                                             % signal at 32 kHz
xlabel('Time (sec)');
```

As shown, the plot clearly demonstrates the reduced sampling frequency of 32 kHz.



See Also

`mfilt.firsrc`, `mfilt.firfracinterp`, `mfilt.firinterp`, `mfilt.firdecim`

References

Fliege, N.J., *Multirate Digital Signal Processing*, John Wiley & Sons, Ltd., 1994

mfilt.firfracinterp

Purpose

Construct a direct-form FIR polyphase fractional interpolator filter object

Syntax

```
hm = mfilt.firfracinterp(l,m,num)
```

Description

`hm = mfilt.firfracinterp(l,m,num)` returns a direct-form FIR polyphase fractional interpolator mfilt object. `l` is the interpolation factor. It must be an integer. If not specified, `l` defaults to 3.

`m` is the decimation factor. Like `l`, it must be an integer. If you do not specify `m` in the calling syntax, it defaults to 1. If you also do not specify a value for `l`, `m` defaults to 2.

`num` is a vector containing the coefficients of the FIR lowpass filter used for interpolation. If omitted, a lowpass Nyquist filter of gain `l` and cutoff frequency of π/`max(l,m)` is used by default.

By specifying both a decimation factor and an interpolation factor, you can interpolate your input signal by noninteger amounts. The fractional interpolator first interpolates the input, then decimates to result in an output signal whose sample rate is `l/m` of the input rate. For proper interpolation, you specify `l` to be greater than `m`. By default, the resulting interpolation factor is 3/2 when you do not provide `l` and `m` in the calling syntax.

Input Arguments

The following table describes the input arguments for creating `hm`.

Input Argument	Description
`l`	Interpolation factor for the filter. `l` specifies the amount to increase the input sampling rate. It must be an integer. When you do not specify a value for `l` it defaults to 3.

Input Argument	Description
num	Vector containing the coefficients of the FIR lowpass filter used for interpolation. When num is not provided as an input, firfracinterp uses a lowpass Nyquist filter with gain equal to l and cutoff frequency equal to π/max(l,m) by default.
m	Decimation factor for the filter. m specifies the amount to reduce the sampling rate of the input signal. It must be an integer. When you do not specify a value for m it defaults to 1. When you do not specify l as well, m defaults to 2.

mfilt.firfracinterp Object Properties

Every multirate filter object has properties that govern the way it behaves when you use it. Note that many of the properties are also input arguments for creating mfilt.firfracinterp objects. The next table describes each property for an mfilt.firfracinterp filter object.

Name	Values	Description
FilterStructure		Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of mfilt object.
Numerator		Vector containing the coefficients of the FIR lowpass filter used for interpolation.
RateChangeFactors	[l,m]	Reports the decimation (m) and interpolation (l) factors for the filter object. Combining these factors results in the final rate change for the signal.

Name	Values	Description
PersistentMemory	false or true	Determines whether the filter states are restored to their starting values for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory returns to the default values any state that the filter changes during processing. States that the filter does not change are not affected.
States	Matrix	Stored conditions for the filter, including values for the interpolator and comb states.
NumSamplesProcessed	Integer	Returns the number of samples processed during filtering. As a check, the number of samples reported processed plus the number of nonprocessed samples should be the total number of input samples.

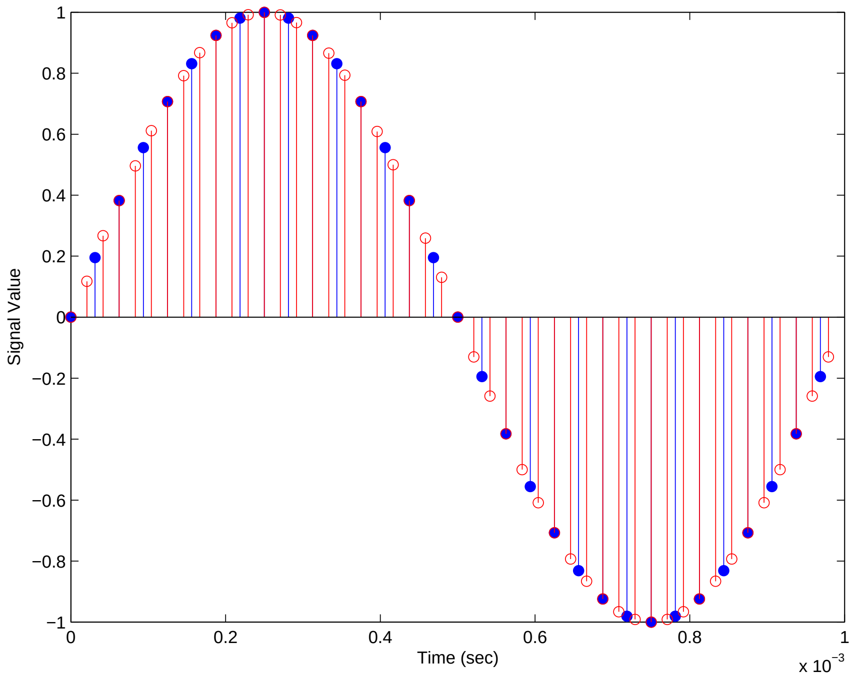
Examples

To convert a signal from 32 kHz to 48 kHz requires fractional interpolation. This example uses the mfilt.firfracinterp object to upsample an input signal. Setting l = 3 and m = 2 returns the same mfilt object as the default mfilt.firfracinterp object.

```
l  = 3; m = 2;              % Interpolation/decimation factors.
hm = mfilt.firfracinterp(l,m);    % We use the default filter
fs = 32e3;                  % Original sample freq: 32 kHz.
n = 0:6799;              % 6800 samples, 0.212 second long signal
x  = sin(2*pi*1e3/fs*n);     % Original signal, sinusoid at 1 kHz
y = filter(hm,x);           % 10200 samples, still 0.212 seconds
```

```
stem(n(1:32)/fs,x(1:32),'filled') % Plot original sampled at
                                  % 32 kHz
hold on;
% Plot fractionally interpolated signal (48 kHz) in red
stem(n(1:48)/(fs*l/m),y(20:67),'r')
xlabel('Time (sec)');ylabel('Signal Value')
```

Having the ability to interpolate by fractional amounts lets us raise the sampling rate from 32 to 48 kHz, something you cannot do with integral interpolators. Both signals appear in the following figure.



See Also

`mfilt.firsrc`, `mfilt.firfracdecim`, `mfilt.firinterp`, `mfilt.firdecim`

mfilt.firinterp

Purpose

Construct floating-point or fixed-point FIR filter-based interpolators

Syntax

```
hm = mfilt.firinterp(l)
hm = mfilt.firinterp(l,num)
```

Description

`hm = mfilt.firinterp(l)` returns an FIR-based interpolator object `hm` with an interpolation factor of `l`. A lowpass Nyquist filter of gain `l` and cutoff frequency of π/`l` is the default if you do not include `l` as an input.

`hm = mfilt.firinterp(l,num)` uses the coefficients specified by `num` for the numerator coefficients of the interpolation filter.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hm` as follows:

- To change to single-precision filtering, enter

  ```
  set(hm,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hm,'arithmetic','fixed');
  ```

Input Arguments

The following table describes the input arguments for creating `hm`.

Input Argument	Description
`l`	Interpolation factor for the filter. `l` specifies the amount to increase the input sampling rate. It must be an integer. When you do not specify a value for `l` it defaults to 2.
`num`	Vector containing the coefficients of the FIR lowpass filter used for interpolation. When `num` is not provided as an input, `firinterp` uses a lowpass Nyquist filter with gain equal to `l` and cutoff frequency equal to π/`l` by default. The default length for the Nyquist filter is 24*`l`. Therefore, each polyphase filter component has length 24.

Object Properties

This section describes the properties for both floating-point filters (double-precision and single-precision) and fixed-point filters.

Floating-Point Filter Properties

Every multirate filter object has properties that govern the way it behaves when you use it. Note that many of the properties are also input arguments for creating `mfilt.firinterp` objects. The next table describes each property for an `mfilt.firinterp` filter object.

Name	Values	Description
`Arithmetic`	`Double`, `single`, `fixed`	Defines the arithmetic the filter uses. Gives you the options `double`, `single`, and `fixed`. In short, this property defines the operation mode for your filter.
`FilterStructure`	String	Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of `mfilt` object. Describes the signal flow for the filter object.
`InterpolationFactor`	Integer	Interpolation factor for the filter. `l` specifies the amount to increase the sampling rate of the input signal. It must be an integer.
`Numerator`	Vector	Vector containing the coefficients of the FIR lowpass filter used for decimation.

Name	Values	Description
`PersistentMemory`	`[false]`, `true`	Determines whether the filter states get restored to zeros for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. `PersistentMemory` set to `false` returns filter states to the default values after filtering. States that the filter does not change are not affected. Setting this to `true` allows you to modify the `States` property.
`States`	`Double`, `single`, matching the filter arithmetic setting.	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions.

Fixed-Point Filter Properties

This table shows the properties associated with the fixed-point implementation of the `mfilt.firinterp` filter.

Note The table lists all of the properties that a fixed-point filter can have. Many of the properties listed are dynamic, meaning they exist only in response to the settings of other properties.

To view all of the characteristics for a filter at any time, use

```
info(hm)
```

where `hm` is a filter.

For further information about the properties of this filter or any mfilt object, refer to "Multirate Filter Properties" on page 8-122.

Name	Values	Description
AccumFracLength	Any positive or negative integer number of bits. [32]	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters and lattice filters. IIR filters have two similar properties—DenAccumFracLength and NumAccumFracLength—that let you set the precision for numerator and denominator operations separately.
AccumWordLength	Any integer number of bits[39]	Sets the word length used to store data in the accumulator.
Arithmetic	fixed for fixed-point filters	Setting this to fixed allows you to modify other filter properties to customize your fixed-point filter.
CoeffAutoScale	[true], false	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength property value to specify the precision used.
CoeffWordLength	Any integer number of bits [16]	Specifies the word length to apply to filter coefficients.

Name	Values	Description
FilterInternals	[FullPrecision], SpecifyPrecision	Controls whether the filter automatically sets the output word and fraction lengths, product word and fraction lengths, and the accumulator word and fraction lengths to maintain the best precision results during filtering. The default value, FullPrecision, sets automatic word and fraction length determination by the filter. SpecifyPrecision makes the output and accumulator-related properties available so you can set your own word and fraction lengths for them.
InputFracLength	Any positive or negative integer number of bits [15]	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Any integer number of bits [16]	Specifies the word length applied to interpret input data.
NumFracLength	Any positive or negative integer number of bits [14]	Sets the fraction length used to interpret the numerator coefficients.
OutputFracLength	Any positive or negative integer number of bits [32]	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set FilterInternals to SpecifyPrecision.
OutputWordLength	Any integer number of bits [39]	Determines the word length used for the output data. You make this property editable by setting FilterInternals to SpecifyPrecision.

Name	Values	Description
OverflowMode	saturate, [wrap]	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic.) The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

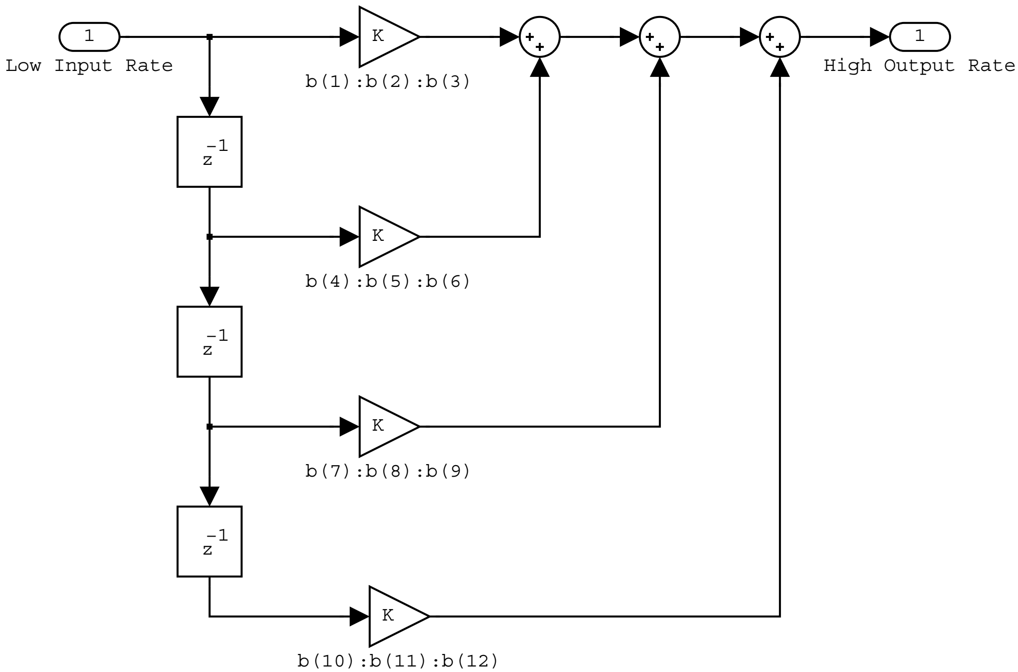
Name	Values	Description
RoundMode	[convergent], ceil,fix,floor, round	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Name	Values	Description
`Signed`	`[true]`, `false`	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
`States`	`fi` object to match the filter arithmetic setting.	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use `fi` objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system.

Filter Structure

To provide interpolation, `mfilt.firinterp` uses the following structure.

The figure below details the signal flow for the direct form FIR filter implemented by `mfilt.firinterp`. In the figure, the delay line updates happen at the lower input rate. The remainder of the filter— the sums and coefficients—operate at the higher output rate.



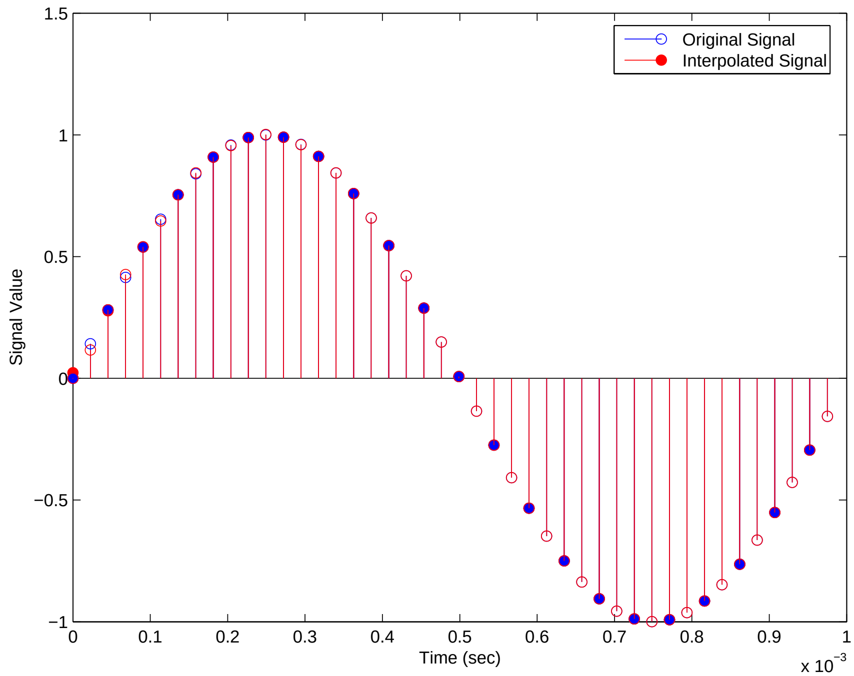
Examples

This example uses `mfilt.firinterp` to double the sample rate of a 22.05 kHz input signal. The output signal ends up at 44.1 kHz. Although `l` is set explicitly to 2, this represents the default interpolation value for `mfilt.firinterp` objects.

```
l = 2;                          % Interpolation factor.
hm = mfilt.firinterp(l);        % Use the default filter.
fs = 22.05e3;                    % Original sample freq: 22.05 kHz.
n = 0:5119;                      % 5120 samples, 0.232s long signal.
x  = sin(2*pi*1e3/fs*n);        % Original signal, sinusoid at 1 kHz.
y = filter(hm,x);               % 10240 samples, still 0.232s.
stem(n(1:22)/fs,x(1:22),'filled') % Plot original sampled at
                                  % 22.05 kHz.
hold on;
```

```
% Plot interpolated signal (44.1 kHz) in red
stem(n(1:44)/(fs*l),y(25:68),'r')
xlabel('Time (sec)');ylabel('Signal Value')
```

With interpolation by 2, the resulting signal perfectly matches the original, but with twice as many samples—one between each original sample, as shown in the following figure.



See Also

`mfilt.holdinterp`, `mfilt.linearinterp`, `mfilt.fftfirinterp`, `mfilt.firfracinterp`, `mfilt.cicinterp`

mfilt.firsrc

Purpose Construct floating-point or fixed-point direct form FIR polyphase sample rate converters

Syntax

```
hm = mfilt.firsrc(l,m,num)
```

Description `hm = mfilt.firsrc(l,m,num)` returns a direct-form FIR polyphase sample rate converter. `l` specifies the interpolation factor. It must be an integer and when omitted in the calling syntax, it defaults to 2.

`m` is the decimation factor. It must be an integer. If not specified, `m` defaults to 1. If `l` is also not specified, `m` defaults to 3 and the overall rate change factor is 2/3.

You specify the coefficients of the FIR lowpass filter used for sample rate conversion in `num`. If omitted, a lowpass Nyquist filter with gain `l` and cutoff frequency of π/`max(l,m)` is the default.

Combining an interpolation factor and a decimation factor lets you use `mfilt.firsrc` to perform fractional interpolation or decimation on an input signal. Using an `mfilt.firsrc` object applies a rate change factor defined by `l/m` to the input signal. For proper rate changing to occur, `l` and `m` must be relatively prime—meaning the ratio `l/m` cannot be reduced to a ratio of smaller integers.

When you are doing sample-rate conversion with large values of `l` or `m`, such as `l` or `m` greater than 20, using the `mfilt.firsrc` structure is the most effective approach. Other possible fractional rate change structures, such as `mfilt.firfracinterp` (where `l` > `m`) or `mfilt.firfracdecim` (where `l < m`) may have prohibitively large memory requirements for applications that require large rate changes.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hm` as follows:

- To change to single-precision filtering, enter

  ```
  set(hm,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hm,'arithmetic','fixed');
  ```

Input Arguments

The following table describes the input arguments for creating `hm`.

Input Argument	Description
`l`	Interpolation factor for the filter. `l` specifies the amount to increase the input sampling rate. It must be an integer. When you do not specify a value for `l`, it defaults to 2.
`num`	Vector containing the coefficients of the FIR lowpass filter used for interpolation. When num is not provided as an input, `mfilt.firsrc` uses a lowpass Nyquist filter with gain equal to `l` and cutoff frequency equal to `π/max(l,m)` by default. The default length for the Nyquist filter is 24*`m`. Therefore, each polyphase filter component has length 24.
`m`	Decimation factor for the filter. `m` specifies the amount to reduce the sampling rate of the input signal. It must be an integer. When you do not specify a value for `m`, it defaults to 1. When `l` is unspecified as well, `m` defaults to 3.

Object Properties

This section describes the properties for both floating-point filters (double-precision and single-precision) and fixed-point filters.

Floating-Point Filter Properties

Every multirate filter object has properties that govern the way it behaves when you use it. Note that many of the properties are also input arguments for creating `mfilt.firsrc` objects. The next table describes each property for an `mfilt.firsrc` filter object.

Name	Values	Description
Arithmetic	[Double], single, fixed	Defines the arithmetic the filter uses. Gives you the options double, single, and fixed. In short, this property defines the operation mode for your filter.
FilterStructure	String	Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of mfilt object. Describes the signal flow for the filter object.
InputOffset	Integers	Contains a value derived from the number of input samples and the decimation factor—InputOffset = mod(length(nx),m) where nx is the number of input samples and m is the decimation factor.
Numerator	Vector	Vector containing the coefficients of the FIR lowpass filter used for decimation.

Name	Values	Description
PersistentMemory	false, true	Determines whether the filter states get restored to zeros for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory set to false returns filter states to the default values after filtering. States that the filter does not change are not affected. Setting this to true allows you to modify the States, InputOffset, and PolyphaseAccum properties.
RateChangeFactors	Positive integers. [2 3]	Specifies the interpolation and decimation factors [l m] (the rate change factors) for changing the input sample rate by nonintegral amounts.
States	Double, single, matching the filter arithmetic setting.	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions.

Fixed-Point Filter Properties

This table shows the properties associated with the fixed-point implementation of the mfilt.firsrc filter.

Note The table lists all of the properties that a fixed-point filter can have. Many of the properties listed are dynamic, meaning they exist only in response to the settings of other properties.

To view all of the characteristics for a filter at any time, use

```
info(hm)
```

where `hm` is a filter.

For further information about the properties of this filter or any `mfilt` object, refer to “Multirate Filter Properties” on page 8-122.

Name	Values	Description
`AccumFracLength`	Any positive or negative integer number of bits. `[32]`	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters.
`AccumWordLength`	Any integer number of bits `[39]`	Sets the word length used to store data in the accumulator.
`Arithmetic`	`fixed` for fixed-point filters	Setting this to `fixed` allows you to modify other filter properties to customize your fixed-point filter.
`CoeffAutoScale`	`[true]`, `false`	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to `false` enables you to change the `NumFracLength` property value to specify the precision used.
`CoeffWordLength`	Any integer number of bits `[16]`	Specifies the word length to apply to filter coefficients.

Name	Values	Description
FilterInternals	[FullPrecision], SpecifyPrecision	Controls whether the filter automatically sets the output word and fraction lengths, product word and fraction lengths, and the accumulator word and fraction lengths to maintain the best precision results during filtering. The default value, FullPrecision, sets automatic word and fraction length determination by the filter. SpecifyPrecision makes the output and accumulator-related properties available so you can set your own word and fraction lengths for them.
InputFracLength	Any positive or negative integer number of bits [15]	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Any integer number of bits [16]	Specifies the word length applied to interpret input data.
NumFracLength	Any positive or negative integer number of bits [14]	Sets the fraction length used to interpret the numerator coefficients.
OutputFracLength	Any positive or negative integer number of bits [32]	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set FilterInternals to SpecifyPrecision.
OutputWordLength	Any integer number of bits [39]	Determines the word length used for the output data. You make this property editable by setting FilterInternals to SpecifyPrecision.

Name	Values	Description
OverflowMode	saturate, [wrap]	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic.) The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
RateChangeFactors	Positive integers [2 3]	Specifies the interpolation and decimation factors [l m] (the rate change factors) for changing the input sample rate by nonintegral amounts.

Name	Values	Description
RoundMode	[convergent], ceil,fix,floor, round	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

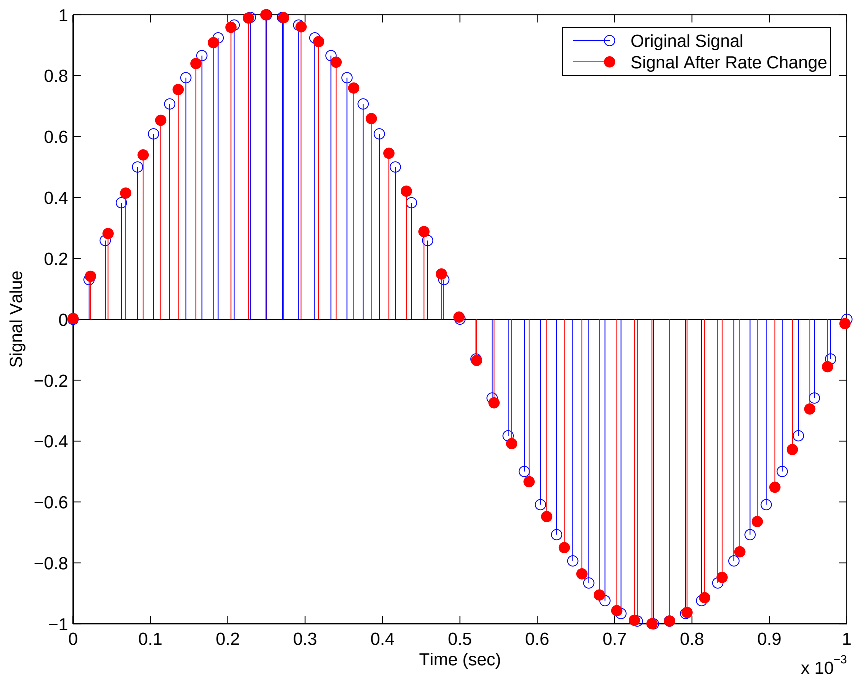
Name	Values	Description
Signed	[true], false	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
States	fi object	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system. For information about the ordering of the states, refer to the filter structure section.

Examples

This is an example of a common audio rate change process—changing the sample rate of a high end audio (48 kHz) signal to the compact disc sample rate (44.1 kHz). This conversion requires a rate change factor of 0.91875, or l = 147 and m = 160.

```
l  = 147; m = 160;            % Interpolation/decimation factors.
hm = mfilt.firsrc(l,m);        % Use the default FIR filter.
fs = 48e3;                     % Original sample freq: 48 kHz.
n = 0:10239;                  % 10240 samples, 0.213 seconds long.
x  = sin(2*pi*1e3/fs*n);      % Original signal, sinusoid at 1 kHz.
y = filter(hm,x);             % 9408 samples, still 0.213 seconds.
stem(n(1:49)/fs,x(1:49))       % Plot original sampled at 48 kHz.
hold on

% Plot fractionally decimated signal (44.1 kHz) in red
stem(n(1:45)/(fs*l/m),y(13:57),'r','filled')
xlabel('Time (sec)');ylabel('Signal Value')
```

Fractional decimation provides you the flexibility to pick and choose the sample rates you want by carefully selecting l and m, the interpolation and decimation factors, that result in the final fractional decimation. The following figure shows the signal after applying the rate change filter hm to the original signal.



See Also

`mfilt.firfracinterp`, `mfilt.firfracdecim`, `mfilt.firinterp`, `mfilt.firdecim`

mfilt.firtdecim

Purpose Construct fixed-point or floating-point direct-form transposed FIR filters

Syntax

```
hm = mfilt.firtdecim(m)
hm = mfilt.firtdecim(m,num)
```

Description

`hm = mfilt.firtdecim(m)` returns a polyphase decimator `mfilt` object `hm` based on a direct-form transposed FIR structure with a decimation factor of `m`. A lowpass Nyquist filter of gain 1 and cutoff frequency of π/`m` is the default.

`hm = mfilt.firtdecim(m,num)` uses the coefficients specified by `num` for the decimation filter. `num` is a vector containing the coefficients of the transposed FIR lowpass filter used for decimation. If omitted, a lowpass Nyquist filter with gain of 1 and cutoff frequency of π/`m` is the default.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hm` as follows:

- To change to single-precision filtering, enter

  ```
  set(hm,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hm,'arithmetic','fixed');
  ```

Input Arguments

The following table describes the input arguments for creating `hm`.

Input Argument	Description
num	Vector containing the coefficients of the FIR lowpass filter used for interpolation. When num is not provided as an input, firtdecim uses a lowpass Nyquist filter with gain equal to 1 and cutoff frequency equal to π/m by default. The default length for the Nyquist filter is 24*m. Therefore, each polyphase filter component has length 24.
m	Decimation factor for the filter. m specifies the amount to reduce the sampling rate of the input signal. It must be an integer. When you do not specify a value for m it defaults to 2.

Object Properties

This section describes the properties for both floating-point filters (double-precision and single-precision) and fixed-point filters.

Floating-Point Filter Properties

Every multirate filter object has properties that govern the way it behaves when you use it. Note that many of the properties are also input arguments for creating mfilt.firtdecim objects. The next table describes each property for an mfilt.firtdecim filter object.

Name	Values	Description
Arithmetic	Double, single, fixed	Specifies the arithmetic the filter uses to process data while filtering.
DecimationFactor	Integer	Decimation factor for the filter. m specifies the amount to reduce the sampling rate of the input signal. It must be an integer.

Name	Values	Description
FilterStructure	String	Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of mfilt object. Also describes the signal flow for the filter object.
InputOffset	Integers	Contains a value derived from the number of input samples and the decimation factor—InputOffset = mod(length(nx),m) where nx is the number of input samples that have been processed so far and m is the decimation factor.
Numerator	Vector	Vector containing the coefficients of the FIR lowpass filter used for decimation.
PersistentMemory	[false], true	Determines whether the filter states get restored to zeros for each filtering operation. The starting values are the values in place when you create the filter if you have not changed the filter since you constructed it. PersistentMemory set to false returns filter states to the default values after filtering. States that the filter does not change are not affected. Setting this to true allows you to modify the States, InputOffset, and PolyphaseAccum properties.

Name	Values	Description
PolyphaseAccum	Double, single [0]	The idea behind having both PolyphaseAccum and Accum is to differentiate between the adders in the filter that work in full precision at all times (PolyphaseAccum) from the adders in the filter that the user controls and that may introduce quantization effects when FilterInternals is set to SpecifyPrecision.
States	Double, single matching the filter arithmetic setting.	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions.

Fixed-Point Filter Properties

This table shows the properties associated with the fixed-point implementation of the mfilt.firtdecim filter.

Note The table lists all of the properties that a fixed-point filter can have. Many of the properties listed are dynamic, meaning they exist only in response to the settings of other properties.

To view all of the characteristics for a filter at any time, use

```
info(hm)
```

where hm is a filter.

For further information about the properties of this filter or any mfilt object, refer to "Multirate Filter Properties" on page 8-122.

Name	Values	Description
AccumFracLength	Any positive or negative integer number of bits. [32]	Specifies the fraction length used to interpret data output by the accumulator. This is a property of FIR filters and lattice filters. IIR filters have two similar properties—DenAccumFracLength and NumAccumFracLength—that let you set the precision for numerator and denominator operations separately.
AccumWordLength	Any integer number of bits [39]	Sets the word length used to store data in the accumulator.
Arithmetic	fixed for fixed-point filters	Setting this to fixed allows you to modify other filter properties to customize your fixed-point filter.
CoeffAutoScale	[true], false	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength property value to specify the precision used.
CoeffWordLength	Any integer number of bits [16]	Specifies the word length to apply to filter coefficients.

Name	Values	Description
FilterInternals	[FullPrecision], SpecifyPrecision	Controls whether the filter automatically sets the output word and fraction lengths, product word and fraction lengths, and the accumulator word and fraction lengths to maintain the best precision results during filtering. The default value, FullPrecision, sets automatic word and fraction length determination by the filter. SpecifyPrecision makes the output and accumulator-related properties available so you can set your own word and fraction lengths for them.
InputFracLength	Any positive or negative integer number of bits [15]	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Any integer number of bits [16]	Specifies the word length applied to interpret input data.
NumFracLength	Any positive or negative integer number of bits [14]	Sets the fraction length used to interpret the numerator coefficients.
OutputFracLength	Any positive or negative integer number of bits [32]	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set FilterInternals to SpecifyPrecision.
OutputWordLength	Any integer number of bits [39]	Determines the word length used for the output data. You make this property editable by setting FilterInternals to SpecifyPrecision.

Name	Values	Description
OverflowMode	saturate, [wrap]	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic.)) The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.
PolyphaseAccum	fi object with zeros to start	Differentiates between the adders in the filter that work in full precision at all times (PolyphaseAccum) and the adders in the filter that the user controls and that may introduce quantization effects when FilterInternals is set to SpecifyPrecision.

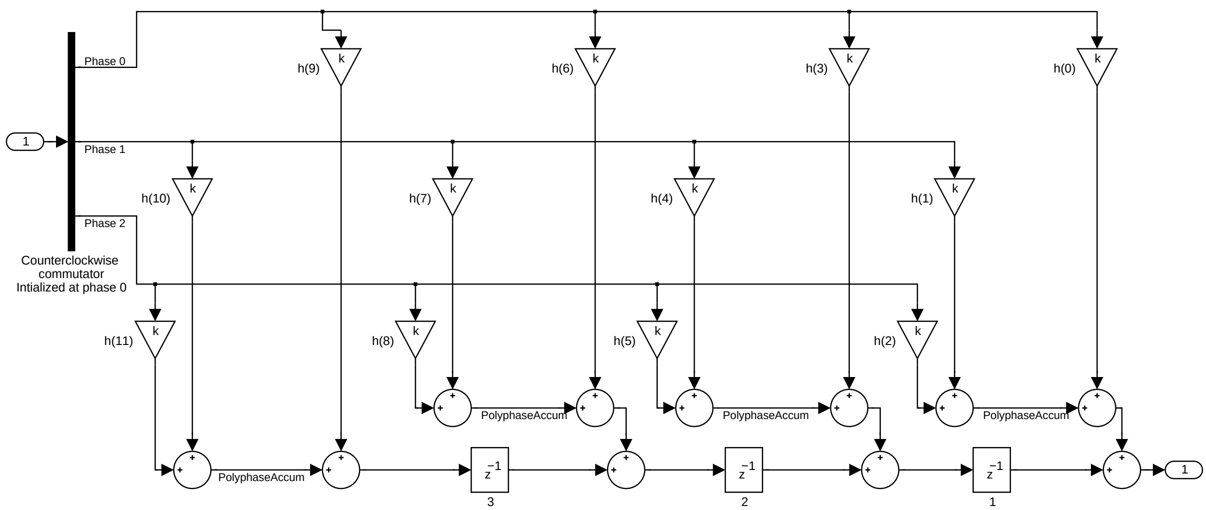
Name	Values	Description
RoundMode	[convergent], ceil,fix,floor, round	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Name	Values	Description
Signed	[true], false	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
States	fi object	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system. For information about the ordering of the states, refer to the filter structure section.

Filter Structure

To provide sample rate changes, `mfilt.firtdecim` uses the following structure. At the input you see a commutator that operates counterclockwise, moving from position 0 to position 2, position 1, and back to position 0 as input samples enter the filter. To keep track of the position of the commutator, the `mfilt` object uses the property `InputOffset` which reports the current position of the commutator in the filter.

The figure below details the signal flow for the direct form FIR filter implemented by `mfilt.firtdecim`.



Notice the order of the states in the filter flow diagram. States 1 through 3 appear in the diagram below each delay element. State 1 applies to the third delay element in phase 2. State 2 applies to the second delay element in phase 2. State 3 applies to the first delay element in phase 2. When you provide the states for the filter as a vector to the `States` property, the above description explains how the filter assigns the states you specify.

In property value form, the states for a filter `hm` are

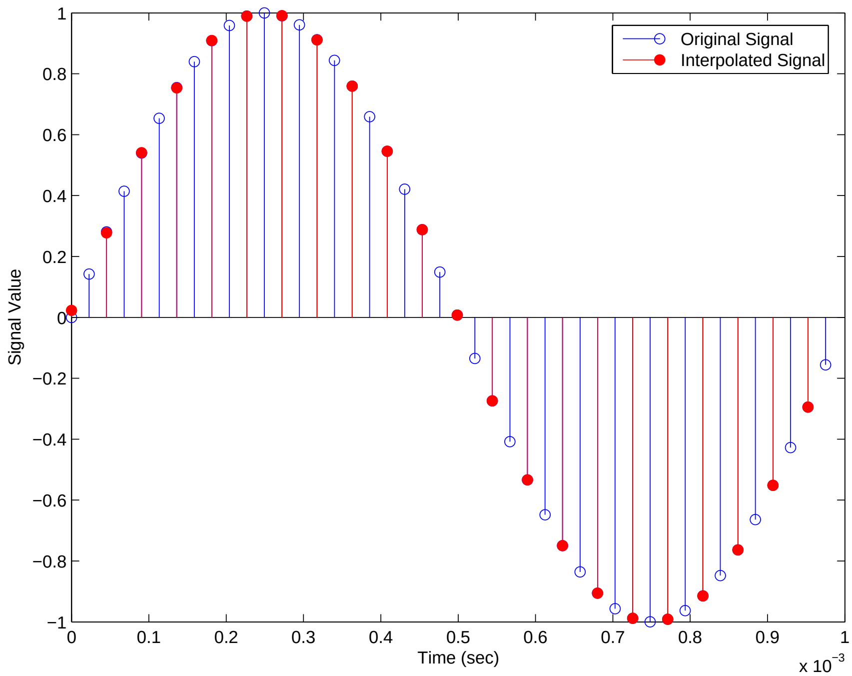
```
hm.states=[1:3];
```

Examples

Demonstrate decimating an input signal by a factor of 2, in this case converting from 44.1 kHz down to 22.05 kHz. In the figure shown following the code, you see the results of decimating the signal.

```
m = 2;                    % Decimation factor.
hm = mfilt.firtdecim(m); % Use the default filter coeffs.
fs = 44.1e3;              % Original sample freq: 44.1 kHz.
n = 0:10239;             % 10240 samples, 0.232 second long signal
x  = sin(2*pi*1e3/fs*n); % Original signal--sinusoid at 1 kHz.
y = filter(hm,x);         % 5120 samples, 0.232 seconds.
stem(n(1:44)/fs,x(1:44)) % Plot original sampled at 44.1 kHz.
hold on                 % Plot decimated signal (22.05 kHz) in red
```

```
stem(n(1:22)/(fs/m),y(13:34),'r','filled')
xlabel('Time (sec)');ylabel('Signal Value')
```



See Also

`mfilt.firdecim`, `mfilt.firfracdecim`, `mfilt.cicdecim`

Purpose

Construct fixed-point or floating-point FIR hold interpolators

Syntax

```
hm = mfilt.holdinterp(l)
```

Description

`hm = mfilt.holdinterp(l)` returns the object `hm` that represents a hold interpolator with the interpolation factor `l`. To work, `l` must be an integer. When you do not include `l` in the calling syntax, it defaults to 2. To perform interpolation by noninteger amounts, use one of the fractional interpolator objects, such as `mfilt.firsrc` or `mfilt.firfracinterp`.

When you use this hold interpolator, each sample added to the input signal between existing samples has the value of the most recent sample from the original signal. Thus you see something like a staircase profile where the interpolated samples form a plateau between the previous and next original samples. The example demonstrates this profile clearly. Compare this to the interpolation process for other interpolators in the toolbox, such as `mfilt.linearinterp`.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hm` as follows:

- To change to single-precision filtering, enter

  ```
  set(hm,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hm,'arithmetic','fixed');
  ```

Input Arguments

The following table describes the input arguments for creating `hm`.

Input Argument	Description
`l`	Interpolation factor for the filter. `l` specifies the amount to increase the input sampling rate. It must be an integer. When you do not specify a value for `l` it defaults to 2.

Object Properties

This section describes the properties for both floating-point filters (double-precision and single-precision) and fixed-point filters.

Floating-Point Filter Properties

Every multirate filter object has properties that govern the way it behaves when you use it. Note that many of the properties are also input arguments for creating `mfilt.holdinterp` objects. The next table describes each property for an `mfilt.interp` filter object.

Name	Values	Description
`Arithmetic`	`Double`, `single`, `fixed`	Specifies the arithmetic the filter uses to process data while filtering.
`FilterStructure`	String	Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of `mfilt` object.
`InterpolationFactor`	Integer	Interpolation factor for the filter. `l` specifies the amount to increase the input sampling rate. It must be an integer.
`PersistentMemory`	'`false`' or '`true`'	Determines whether the filter states are restored to zero for each filtering operation.
`States`	Double or single array	Filter states. `states` defaults to a vector of zeros that has length equal to `nstates(hm)`. Always available, but visible in the display only when `PersistentMemory` is true.

Fixed-Point Filter Properties

This table shows the properties associated with the fixed-point implementation of the `mfilt.holdinterp` filter.

Note The table lists all of the properties that a fixed-point filter can have. Many of the properties listed are dynamic, meaning they exist only in response to the settings of other properties.

To view all of the characteristics for a filter at any time, use

```
info(hm)
```

where `hm` is a filter.

For further information about the properties of this filter or any `mfilt` object, refer to “Multirate Filter Properties” on page 8-122.

Name	Values	Description
`Arithmetic`	`Double`, `single`, `fixed`	Specifies the arithmetic the filter uses to process data while filtering.
`FilterStructure`	String	Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of `mfilt` object.
`InputFracLength`	Any positive or negative integer number of bits `[15]`	Specifies the fraction length the filter uses to interpret input data.
`InputWordLength`	Any integer number of bits `[16]`	Specifies the word length applied to interpret input data.

Name	Values	Description
`InterpolationFactor`	Integer	Interpolation factor for the filter. `l` specifies the amount to increase the input sampling rate. It must be an integer.
`PersistentMemory`	`'false'` or `'true'`	Determine whether the filter states get restored to zero for each filtering operation
`States`	`fi` object	Contains the filter states before, during, and after filter operations. For hold interpolators, the states are always empty—hold interpolators do not have states. Notice that the states use `fi` objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system.

Filter Structure

Hold interpolators do not have structures or filter coefficients.

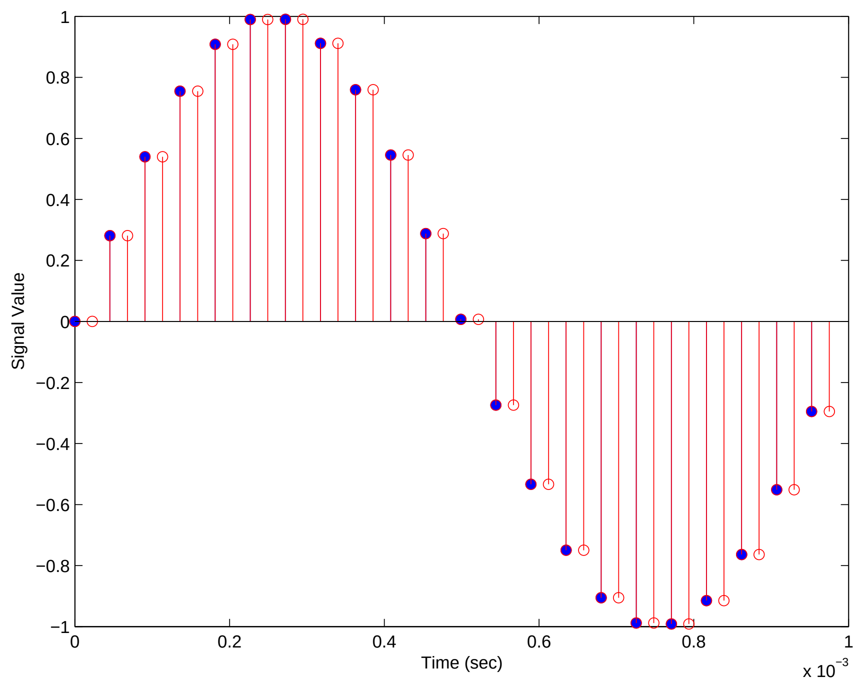
Examples

To see the effects of hold-based interpolation, interpolate an input sine wave from 22.05 to 44.1 kHz. Note that each added sample retains the value of the most recent original sample.

```
l = 2;                         % Interpolation factor
hm = mfilt.holdinterp(l);
fs = 22.05e3;                  % Original sample freq: 22.05 kHz.
n = 0:5119;                 % 5120 samples, 0.232 second long signal
x  = sin(2*pi*1e3/fs*n);     % Original signal, sinusoid at 1 kHz
y = filter(hm,x);            % 10240 samples, still 0.232 seconds
stem(n(1:22)/fs,x(1:22),'filled') % Plot original sampled at
                                  % 22.05 kHz
```

```
hold on                        % Plot interpolated signal (44.1 kHz)
in red
stem(n(1:44)/(fs*l),y(1:44),'r')
xlabel('Time (sec)');ylabel('Signal Value')
```

The following figure shows clearly the step nature of the signal that comes from interpolating the signal using the hold algorithm approach. Compare the output to the linear interpolation used in `mfilt.linearinterp`.



See Also

`mfilt.linearinterp`, `mfilt.firinterp`, `mfilt.firfracinterp`, `mfilt.cicinterp`

mfilt.linearinterp

Purpose Construct floating-point or fixed-point linear interpolator filters

Syntax

```
hm = mfilt.linearinterp(l)
```

Description

`hm = mfilt.linearinterp(l)` returns an FIR linear interpolator `hm` with an integer interpolation factor `l`. Provide `l` as a positive integer. The default value for the interpolation factor is 2 when you do not include the input argument `l`.

When you use this linear interpolator, the samples added to the input signal have values between the values of adjacent samples in the original signal. Thus you see something like a smooth profile where the interpolated samples continue a line between the previous and next original samples. The example demonstrates this smooth profile clearly. Compare this to the interpolation process for `mfilt.holdinterp`, which creates a stairstep profile.

Make this filter a fixed-point or single-precision filter by changing the value of the `Arithmetic` property for the filter `hm` as follows:

- To change to single-precision filtering, enter

  ```
  set(hm,'arithmetic','single');
  ```

- To change to fixed-point filtering, enter

  ```
  set(hm,'arithmetic','fixed');
  ```

Input Arguments

The following table describes the input argument for `mfilt.linearinterp`.

Input Argument	Description
`l`	Interpolation factor for the filter. `l` specifies the amount to increase the input sampling rate. It must be an integer. When you do not specify a value for `l` it defaults to 2.

Object Properties

This section describes the properties for both floating-point filters (double-precision and single-precision) and fixed-point filters.

Floating-Point Filter Properties

Every multirate filter object has properties that govern the way it behaves when you use it. Note that many of the properties are also input arguments for creating `mfilt.linearinterp` objects. The next table describes each property for an `mfilt.linearinterp` filter object.

Name	Values	Description
`Arithmetic`	`Double`, `single`, `fixed`	Specifies the arithmetic the filter uses to process data while filtering.
`FilterStructure`	String	Reports the type of filter object. You cannot set this property—it is always read only and results from your choice of `mfilt` object.
`InterpolationFactor`	Integer	Interpolation factor for the filter. `l` specifies the amount to increase the input sampling rate. It must be an integer.
`PersistentMemory`	'`false`' or '`true`'	Determine whether the filter states get restored to zero for each filtering operation
`States`	Double or single array	Filter states. `states` defaults to a vector of zeros that has length equal to `nstates(hm)`. Always available, but visible in the display only when `PersistentMemory` is `true`.

Fixed-Point Filter Properties

This table shows the properties associated with the fixed-point implementation of the `mfilt.holdinterp` filter.

Note The table lists all of the properties that a fixed-point filter can have. Many of the properties listed are dynamic, meaning they exist only in response to the settings of other properties.

To view all of the characteristics for a filter at any time, use

```
info(hm)
```

where hm is a filter.

For further information about the properties of this filter or any mfilt object, refer to “Multirate Filter Properties” on page 8-122.

Name	Values	Description
AccumFracLength	Any positive or negative integer number of bits. Depends on L. [29 when L=2]	Specifies the fraction length used to interpret data output by the accumulator.
AccumWordLength	Any integer number of bits [33]	Sets the word length used to store data in the accumulator.
Arithmetic	fixed for fixed-point filters	Setting this to fixed allows you to modify other filter properties to customize your fixed-point filter.
CoeffAutoScale	[true], false	Specifies whether the filter automatically chooses the proper fraction length to represent filter coefficients without overflowing. Turning this off by setting the value to false enables you to change the NumFracLength property value to specify the precision used.
CoeffWordLength	Any integer number of bits [16]	Specifies the word length to apply to filter coefficients.

Name	Values	Description
FilterInternals	[FullPrecision], SpecifyPrecision	Controls whether the filter automatically sets the output word and fraction lengths, product word and fraction lengths, and the accumulator word and fraction lengths to maintain the best precision results during filtering. The default value, FullPrecision, sets automatic word and fraction length determination by the filter. SpecifyPrecision makes the output and accumulator-related properties available so you can set your own word and fraction lengths for them.
InputFracLength	Any positive or negative integer number of bits [15]	Specifies the fraction length the filter uses to interpret input data.
InputWordLength	Any integer number of bits [16]	Specifies the word length applied to interpret input data.
NumFracLength	Any positive or negative integer number of bits [14]	Sets the fraction length used to interpret the numerator coefficients.
OutputFracLength	Any positive or negative integer number of bits [29]	Determines how the filter interprets the filter output data. You can change the value of OutputFracLength when you set FilterInternals to SpecifyPrecision.
OutputWordLength	Any integer number of bits [33]	Determines the word length used for the output data. You make this property editable by setting FilterInternals to SpecifyPrecision.

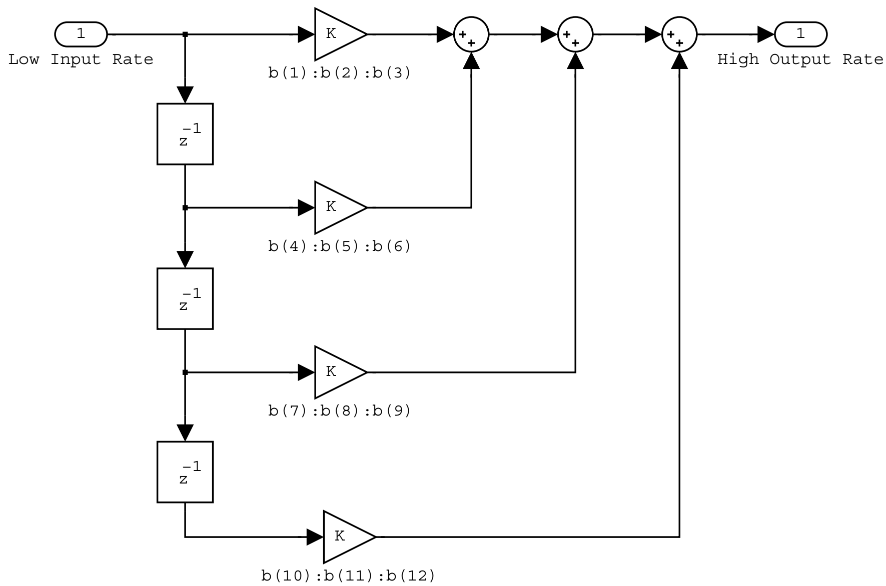
Name	Values	Description
OverflowMode	saturate, [wrap]	Sets the mode used to respond to overflow conditions in fixed-point arithmetic. Choose from either saturate (limit the output to the largest positive or negative representable value) or wrap (set overflowing values to the nearest representable value using modular arithmetic.)) The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always saturates. Finally, products never overflow—they maintain full precision.

Name	Values	Description
RoundMode	[convergent], ceil,fix,floor, round	Sets the mode the filter uses to quantize numeric values when the values lie between representable values for the data format (word and fraction lengths). • convergent—Round up to the next allowable quantized value. • ceil—Round to the nearest allowable quantized value. Numbers that are exactly halfway between the two nearest allowable quantized values are rounded up only if the least significant bit (after rounding) would be set to 1. • fix—Round negative numbers up and positive numbers down to the next allowable quantized value. • floor—Round down to the next allowable quantized value. • round—Round to the nearest allowable quantized value. Numbers that are halfway between the two nearest allowable quantized values are rounded up. The choice you make affects only the accumulator and output arithmetic. Coefficient and input arithmetic always round. Finally, products never overflow—they maintain full precision.

Name	Values	Description
Signed	[true], false	Specifies whether the filter uses signed or unsigned fixed-point coefficients. Only coefficients reflect this property setting.
States	fi object	Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Notice that the states use fi objects, with the associated properties from those objects. For details, refer to fixed-point objects in your Fixed-Point Toolbox documentation or in the online Help system. For information about the ordering of the states, refer to the filter structure below.

Filter Structure

Linear interpolator structures depend on the FIR filter you use to implement the filter. By default, the structure is direct-form FIR.

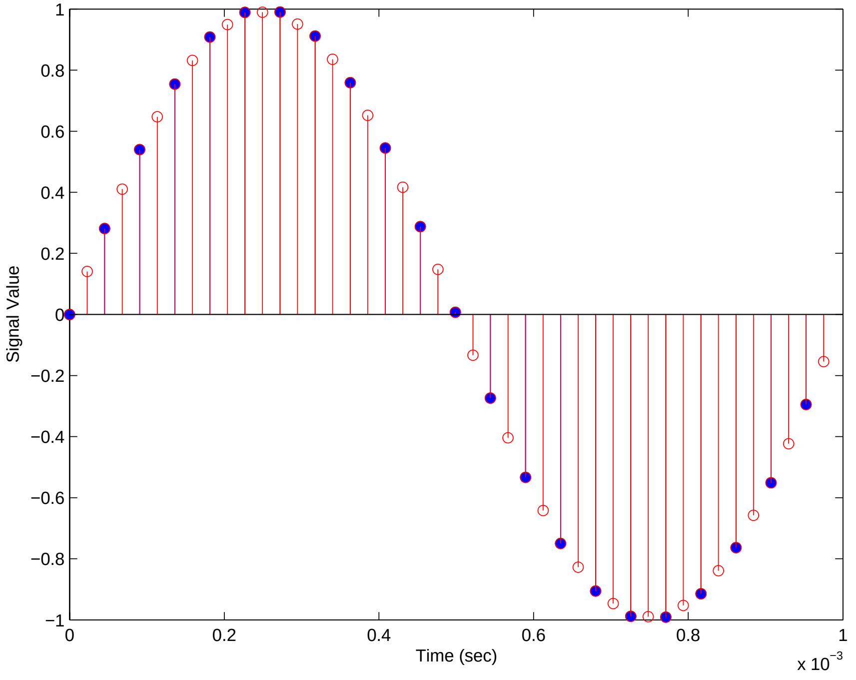


Examples

Interpolation by a factor of 2 (used to convert the input signal sampling rate from 22.05 kHz to 44.1 kHz).

```
l = 2;                         % Interpolation factor
hm = mfilt.linearinterp(l);
fs = 22.05e3;                  % Original sample freq: 22.05 kHz.
n = 0:5119;                % 5120 samples, 0.232 second long signal
x  = sin(2*pi*1e3/fs*n);       % Original signal, sinusoid at 1 kHz
y = filter(hm,x);              % 10240 samples, still 0.232 seconds
stem(n(1:22)/fs,x(1:22),'filled') % Plot original sampled at
                                  % 22.05 kHz
hold on                        % Plot interpolated signal (44.1
                               % kHz) in red
stem(n(1:44)/(fs*l),y(2:45),'r')
xlabel('Time (s)');ylabel('Signal Value')
```

Using linear interpolation, as compared to the hold approach of `mfilt.holdinterp`, provides greater fidelity to the original signal.



See Also

`mfilt.holdinterp`, `mfilt.firinterp`, `mfilt.firfracinterp`, `mfilt.cicinterp`

Purpose Calculate and return the predicted mean-squared error for selected adaptive filters

Syntax

```
[mmse,emse] = msepred(ha,x,d)
[mmse,emse,meanw,mse,tracek] = msepred(ha,x,d)
[mmse,emse,meanw,mse,tracek] = msepred(ha,x,d,m)
```

Description `[mmse,emse] = msepred(ha,x,d)` predicts the steady-state values at convergence of the minimum mean-squared error (`mmse`) and the excess mean-squared error (`emse`) given the input and desired response signal sequences in `x` and `d` and the property values in the adaptfilt object `ha`.

`[mmse,emse,meanw,mse,tracek] = msepred(ha,x,d)` calculates three sequences corresponding to the analytical behavior of the LMS adaptive filter defined by `ha`:

- `meanw`—contains the sequence of coefficient vector means. The columns of matrix `meanw` contain predictions of the mean values of the LMS adaptive filter coefficients at each time instant. The dimensions of `meanw` are (`size`(`x`,1))-by-(`ha.length`).
- `mse`—contains the sequence of mean-square errors. This column vector contains predictions of the mean-square error of the LMS adaptive filter at each time instant. The length of `mse` is equal to `size`(`x`,1).
- `tracek`—contains the sequence of total coefficient error powers. This column vector contains predictions of the total coefficient error power of the LMS adaptive filter at each time instant. The length of `tracek` is equal to `size`(`x`,1).

`[mmse,emse,meanw,mse,tracek] = msepred(ha,x,d,m)` specifies an optional input argument `m` that is the decimation factor for computing `meanw`, `mse`, and `tracek`. When `m` > 1, `msepred` saves every `m`th predicted value of each of these sequences. When you omit the optional argument `m`, it defaults to one.

Note `msepred` is available for the following adaptive filters only:
—`adaptfilt.blms`
—`adaptfilt.blmsfft`
—`adaptfilt.lms`

—adaptfilt.nlms
—adaptfilt.se

Using msepred is the same for any adaptfilt object constructed by the supported filters.

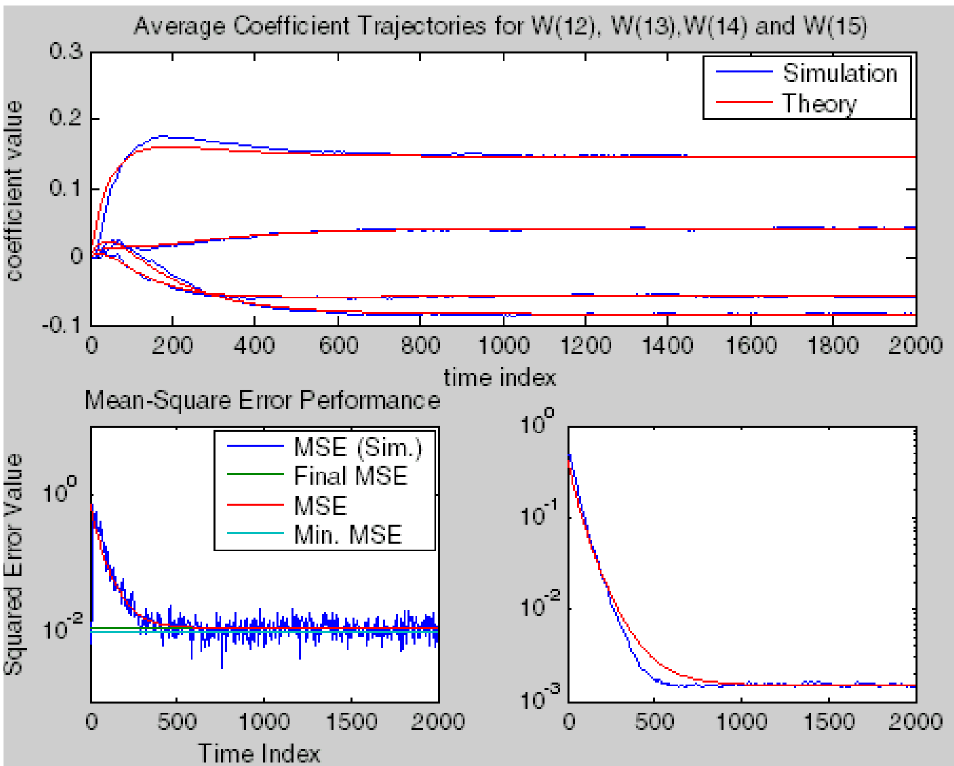
Examples

Analyze and simulate a 32-coefficient adaptive filter using 25 trials of 2000 iterations each.

```
x = zeros(2000,25); d = x;     % Initialize variables
ha = fir1(31,0.5);             % FIR system to be identified
x = filter(sqrt(0.75),[1 -0.5],sign(randn(size(x))));
n = 0.1*randn(size(x));        % observation noise signal
d = filter(ha,1,x)+n;          % desired signal
l = 32;                        % Filter length
mu = 0.008;                    % LMS step size.
m  = 5;                        % Decimation factor for analysis
                               % and simulation results
ha = adaptfilt.lms(l,mu);
[mmse,emse,meanW,mse,traceK] = msepred(ha,x,d,m);
[simmse,meanWsim,Wsim,traceKsim] = msesim(ha,x,d,m);
nn = m:m:size(x,1);
subplot(2,1,1);
plot(nn,meanWsim(:,12),'b',nn,meanW(:,12),'r',nn,...
meanWsim(:,13:15),'b',nn,meanW(:,13:15),'r');
title('Average Coefficient Trajectories for W(12), W(13),...
 W(14) and W(15)');
legend('Simulation','Theory');
xlabel('Time Index'); ylabel('Coefficient Value');
subplot(2,2,3);
semilogy(nn,simmse,[0 size(x,1)],[(emse+mmse)...
(emse+mmse)],nn,mse,[0 size(x,1)],[mmse mmse]);
title('Mean-Square Error Performance');
axis([0 size(x,1) 0.001 10]);
legend('MSE (Sim.)','Final MSE','MSE','Min. MSE');
xlabel('Time Index'); ylabel('Squared Error Value');
subplot(2,2,4);
semilogy(nn,traceKsim,nn,traceK,'r');
```

```
title('Sum-of-Squared Coefficient Errors'); axis([0 size(x,1)...
0.0001 1]);
legend('Simulation','Theory');
xlabel('Time Index'); ylabel('Squared Error Value');
```

Viewing the plots in this figure you see the various error values plotted in both simulation and theory. Each subplot reveals more information about the results as the simulation converges with the theoretical performance.



See Also

`filter`, `maxstep`, `msesim`

msesim

Purpose

Calculate and return the measured mean-squared error for an adaptive filter

Syntax

```
mse = msesim(ha,x,d)
[mse,meanw,w,tracek] = msesim(ha,x,d)
[mse,meanw,w,tracek] = msesim(ha,x,d,m)
```

Description

`mse = msesim(ha,x,d)` returns the sequence of mean-square errors in column vector `mse`. The vector contains estimates of the mean-square error of the adaptive filter at each time instant during adaptation. The length of `mse` is equal to `size(x,1)`. The columns of matrix `x` contain individual input signal sequences, and the columns of the matrix `d` contain corresponding desired response signal sequences.

`[mse,meanw,w,tracek] = msesim(ha,x,d)` calculates three parameters that correspond to the simulated behavior of the adaptive filter defined by `ha`:

- `meanw`—sequence of coefficient vector means. The columns of this matrix contain estimates of the mean values of the LMS adaptive filter coefficients at each time instant. The dimensions of `meanw` are (`size(x,1)`)-by-(`ha.length`).
- `w`—estimate of the final values of the adaptive filter coefficients for the algorithm corresponding to `ha`.
- `tracek`—sequence of total coefficient error powers. This column vector contains estimates of the total coefficient error power of the LMS adaptive filter at each time instant. The length of `tracek` is equal to `size(X,1)`.

`[mse,meanw,w,tracek] = msesim(ha,x,d,m)` specifies an optional input argument `m` that is the decimation factor for computing `meanw`, `mse`, and `tracek`. When `m` > 1, `msepsim` saves every `m`th predicted value of each of these sequences. When you omit the optional argument `m`, it defaults to one.

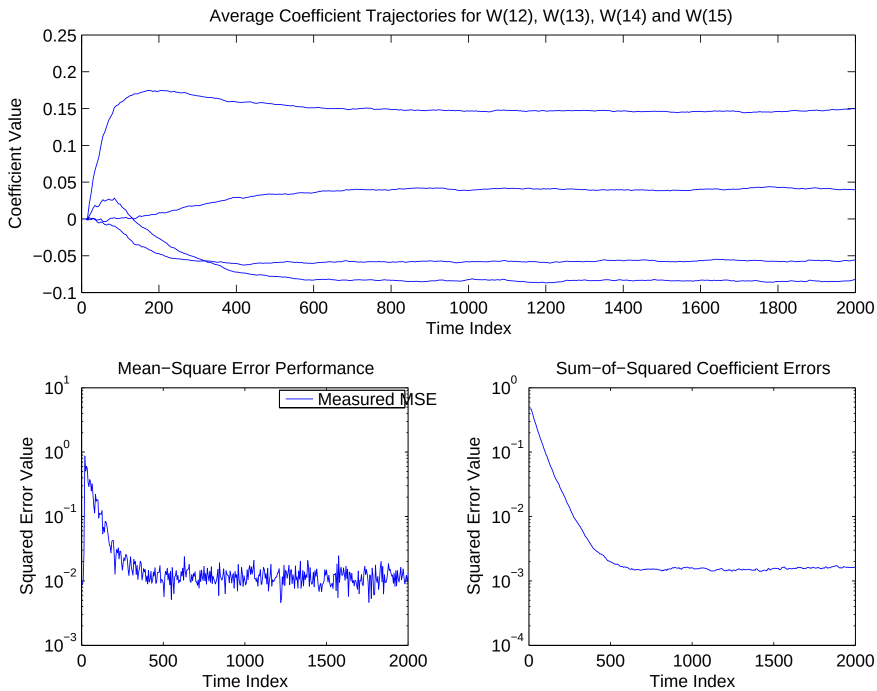
Examples

Simulation of a 32-coefficient FIR filter using 25 trials, each trial having 2000 iterations of the adaptation process.

```
x = zeros(2000,25); d = x;          % Initialize variables
ha = fir1(31,0.5);                  % FIR system to be identified
x = filter(sqrt(0.75),[1 -0.5],sign(randn(size(x))));
n = 0.1*randn(size(x));             % Observation noise signal
d = filter(ha,1,x)+n;               % Desired signal
```

```
l = 32;                               % Filter length
mu = 0.008;                           % LMS Step size.
m  = 5;                           % Decimation factor for analysis
                                      % and simulation results
ha = adaptfilt.lms(l,mu);
[simmse,meanWsim,Wsim,traceKsim] = msesim(ha,x,d,m);
nn = m:m:size(x,1);
subplot(2,1,1);
plot(nn,meanWsim(:,12),'b',nn,meanWsim(:,13:15),'b');
title('Average Coefficient Trajectories for W(12), W(13),
W(14) and W(15)');
xlabel('Time Index'); ylabel('Coefficient Value');
subplot(2,2,3);
semilogy(nn,simmse);
title('Mean-Square Error Performance'); axis([0 size(x,1) 0.001
10]);
legend('Measured MSE');
xlabel('Time Index'); ylabel('Squared Error Value');
subplot(2,2,4);
semilogy(nn,traceKsim);
title('Sum-of-Squared Coefficient Errors'); axis([0 size(x,1)
0.0001 1]);
xlabel('Time Index'); ylabel('Squared Error Value');
```

Calculating the mean squared error for an adaptive filter is one measure of the performance of the adapting algorithm. In this figure, you see a variety of measures of the filter, including the error values.



See Also

`filter`, `msepred`

Purpose

Compute the power spectral density (PSD) of filter output caused by roundoff noise during the quantization process

Syntax

```
hpsd = noisepsd(hd,l);
hpsd = noisepsd(hd,l, propertyname1, propertyvalue1,
  propertyname2,propertyvalue2, );
hpsd = noisepsd(hd,l,opts);
```

Description

`hpsd = noisepsd(hd,l)` computes the power spectral density (PSD) at the output of filter `hd` due to roundoff noise produced by quantization errors within the filter. `l` is the number of trials used to compute the average. The PSD is computed from the average over the `l` trials. The more trials you specify, the better the estimate, but at the expense of longer computation time. When you do not explicitly set `l`, it defaults to 10 trials.

`hpsd` is a `psd` data object.To extract the PSD vector (the data from the PSD) from `hpsd`, enter

```
get(hpsd,'data')
```

at the prompt. Plot the PSD data with `plot(hpsd)`. The average power of the output noise (the integral of the PSD) can be computed with `avgpower`, a method of `dspdata` objects:

```
avgpwr = avgpower(hpsd).
```

`hpsd = noisepsd(hd,l,p1,v1,p2,v2,...)` specifies optional parameters via propertyname/propertyvalue pairs. The properties of the `psd` object, and the valid entries are:

Property Name	Default Value	Description and Valid Entries
`Nfft`	512	Specifies the number of FFT points to use to calculate the PSD.
`NormalizedFrequency`	`true`	Determines whether to use normalized frequency. Enter one of the logical `true` or `false`. Note that you do not use single quotations around this property value because it is a logical, not a string.
`Fs`	`normalized`	Specifies the sampling frequency to use when you set `NormalizedFrequency` to `false`. Any integer value greater than 1 works. Enter the value in Hz.

Property Name	Default Value	Description and Valid Entries
`SpectrumType`	`onesided`	Tells `noisepsd` whether to generate a one-sided PSD or two-sided. Options are `onesided` or `twosided`. If you choose a two-sided computation, you can also choose `centerdc` = `true`. Otherwise, `centerdc` must be `false`. • `onesided` converts the spectrum to a spectrum calculated over half the Nyquist interval. All properties affected by the new frequency range are adjusted automatically. • `twosided` converts the spectrum to a spectrum calculated over the whole Nyquist interval. All properties affected by the new frequency range are adjusted automatically.

Property Name	Default Value	Description and Valid Entries
CenterDC	false	Shifts the zero-frequency component to the center of a two-sided spectrum. • When you set SpectrumType to onesided, it is changed to twosided and the data is converted to a two-sided spectrum. • Setting CenterDC to false shifts the data and the frequency values in the object so that DC is in the left edge of the spectrum. This operation does not effect the SpectrumType property setting.

Note If the spectrum data you specify is calculated over half the Nyquist interval and you do not specify a corresponding frequency vector, the default frequency vector assumes that the number of points in the whole FFT was even. Also, the plot option to convert to a whole or two-sided spectrum assumes the original whole FFT length was even.

`noisepsd(hd,l,opts)` uses an options object `opts` to specify the optional input arguments instead of specifying property-value pairs in the command. Use `opts = noisepsdopts(hd)` to create the object. `opts` then has the `noisepsd` settings from `hd`. After creating `opts`, you change the property values before calling `noisepsd`:

```
set(opts,'fs',48e3); % Set Fs to 48 kHz.
```

Examples

Compute the PSD of the output noise caused by the quantization processes in a fixed-point, direct form FIR filter.

```
b = firgr(27,[0 .4 .6 1],[1 1 0 0]);
h = dfilt.dffir(b); % Create the filter object.
h.arithmetic = 'fixed'; % Quantize the filter to fixed-point.
hpsd = noisepsd(h);
plot(hpsd)
```

`hpsd` looks like this—the data resulting from the noise PSD calculation. You can review the data in `hpsd.data'`.

Here is the specification for hpsd.

```
hpsd =

                    Name: 'Power Spectral Density'
                    Data: [257x1 double]
```

```
           SpectrumType: 'Onesided'
            Frequencies: [257x1 double]
    NormalizedFrequency: true
                     Fs: 'Normalized'
```

See Also

filter, noisepsdopts, norm, reorder, scale

spectrum.welch in the Signal Processing Toolbox

References

McClellan, et al., *Computer-Based Exercises for Signal Processing Using MATLAB 5*, Prentice-Hall, 1998.

Purpose

Create an object that contains options for running the output noise PSD computation `noisepsd` on a filter

Syntax

```
opts = noisepsdopts(hd)
```

Description

`opts = noisepsdopts(hd)` uses the current settings in the filter `hd` to create an options object `opts` that contains specified options for computing the output noise PSD for a filter `hd`. You can pass `opts` to the `scale` method as an input argument to apply scaling settings to a second-order filter.

Within `opts`, the `noisepsd` options object returned by `noisepsdopts`, you can set the following properties:

Property Name	Default Value	Description and Valid Entries
`Nfft`	512	Specifies the number of FFT points to use to calculate the PSD.
`NormalizedFrequency`	`true`	Determines whether to use normalized frequency. Enter one of the logical `true` or `false`. Note that you do not use single quotations around this property value because it is a logical value, not a string.
`Fs`	`normalized`	Specifies the sampling frequency to use when you set `NormalizedFrequency` to `false`. Any integer value greater than 1 works. Enter the value in Hz.

Property Name	Default Value	Description and Valid Entries
SpectrumType	onesided	Tells noisepsd whether to generate a one-sided PSD or two-sided. Options are onesided or twosided. If you choose a two-sided computation, you can also choose centerdc = true. Otherwise, centerdc must be false. • onesided converts the spectrum to a spectrum calculated over half the Nyquist interval. All properties affected by the new frequency range are adjusted automatically. • twosided converts the spectrum to a spectrum calculated over the whole Nyquist interval. All properties affected by the new frequency range are adjusted automatically.

Property Name	Default Value	Description and Valid Entries
`CenterDC`	`false`	Shifts the zero-frequency component to the center of a two-sided spectrum. • When you set `SpectrumType` to `onesided`, it is changed to `twosided` and the data is converted to a two-sided spectrum. • Setting `CenterDC` to `false` shifts the data and the frequency values in the object so that DC is in the left edge of the spectrum. This operation does not effect the `SpectrumType` property setting.

See Also

`noisepsd`

Purpose

Return the p-norm of adaptfilt, dfilt, or mfilt objects

Syntax

```
l = norm(ha)
l = norm(ha,pnorm)
l = norm(hd)
l = norm(hd,pnorm)
l = norm(hd,'L2',tol)
l = norm(hm)
l = norm(hm,pnorm)
```

Description

All of the variants of norm return the filter p-norm for the object in the syntax, either an adaptive filter, a digital filter, or a multirate filter. When you omit the pnorm argument, norm returns the L2-norm for the object.

Note that by Parseval's theorem, the L2-norm of a filter is equal to the l2 norm. This equality is not true for the other norm variants.

For adaptfilt Objects

l = norm(ha)returns the L2-norm of an adaptive filter.

l = norm(ha,pnorm) adds the input argument pnorm to let you specify the norm returned. pnorm can be either

- Frequency-domain norms specified by one of L1, L2, or Linf
- Discrete-time domain norms specified by one of l1, l2, or linf

For dfilt Objects

l = norm(hd) returns the L2-norm of a discrete-time filter.

l = norm(hd,pnorm) includes input argument pnorm that lets you specify the norm returned. pnorm can be either

- Frequency-domain norms specified by one of L1, L2, or Linf
- Discrete-time domain norms specified by one of l1, l2, or linf

By Parseval's theorem, the L2-norm of a filter is equal to the l2 norm. This equality is not true for the other norm variants.

IIR filters respond slightly differently to norm. When you compute the l2, linf, L1, and L2 norms for an IIR filter, norm(...,L2,tol) lets you specify the tolerance for the accuracy in the computation. For l1, l2, L2, and linf, norm uses the tolerance to truncate the infinite impulse response that it uses to calculate the norm. For L1, norm passes the tolerance to the numerical integration algorithm. Refer to Examples to see this in use. You cannot specify Linf for the norm and include the tol option.

For mfilt Objects

l = norm(hm) returns the L2-norm of a multirate filter.

l = norm(hm,pnorm) includes argument pnorm to let you specify the norm returned. pnorm can be either

- Frequency-domain norms specified by one of L1, L2, or Linf
- Discrete-time domain norms specified by one of l1, l2, or linf

Note that, by Parseval's theorem, the L2-norm of a filter is equal to the l2 norm. This equality is not true for the other norm variants.

Examples

Adaptfilt Objects

For the adaptive filter example, compute the 2-norm of an adaptfilt object, here an LMS-based adaptive filter.

```
ha = adaptfilt.lms; % norm(ha) is zero because all coeffs are zero
% Create some data to filter to generate filter coeffs
x = randn(100,1);
d = x + randn(100,1);
[y,e] = filter(ha,x,d);
l2 = norm(ha); % Now norm(ha) is nonzero
l2 =

    1.1231
```

Dfilt Objects

To demonstrate the tolerance option used with an IIR filter (dfilt object), compute the 2-norm of filter hd with a tolerance of 1e-10.

```
d=fdesign.lowpass('n,fc',5,0.4)
```

```
d =

             ResponseType: 'Lowpass with cutoff'
        SpecificationType: 'N,Fc'
              Description: {2x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
              FilterOrder: 5
                  Fcutoff: 0.4000


hd = butter(d);
l2=norm(hd,'l2',1e-10)

l2 =

    0.6336
```

Mfilt Objects

In this example, compute the infinity norm of an FIR interpolator, which is an `mfilt` object.

```
hm = mfilt.firinterp;
linf = norm(hm,inf);
linf =

    2.0002
```

See Also

`reorder`, `scale`, `scalecheck`

Purpose

Normalize filter numerator or feed-forward coefficients to be between -1 and 1

Syntax

```
normalize(hq)
g = normalize(hq)
```

Description

`normalize(hq)` normalizes the filter numerator coefficients for a quantized filter to have values between -1 and 1. Notice that the coefficients of `hq` change—`normalize` does not copy `hq` and return the copy. To restore the coefficients of `hq` to the original values, use `denormalize`.

Note that for lattice filters, the feed-forward coefficients stored in the property `lattice` are normalized.

`g = normalize(hd)` normalizes the numerator coefficients for the filter `hq` to between -1 and 1 and returns the gain `g` due to the normalization operation. Calling `normalize` again does not change the coefficients. `g` always returns the gain returned by the first call to normalize the filter.

Examples

Create a direct form II quantized filter that uses second-order sections. Then use `normalize` to maximize the use of the range of representable coefficients.

```
d=fdesign.lowpass('n,fp,ap,ast',8,.5,2,40);

hd=ellip(d);

hd =

        FilterStructure: 'Direct-Form II, Second-Order Sections'
             Arithmetic: 'double'
              sosMatrix: [4x6 double]
            ScaleValues: [5x1 double]
       PersistentMemory: 'on'
                 States: [2x4 double]
    NumSamplesProcessed: 0

hd.arithmetic='fixed'

hd =

        FilterStructure: 'Direct-Form II, Second-Order Sections'
```

```
                Arithmetic: 'fixed'
                 sosMatrix: [4x6 double]
               ScaleValues: [5x1 double]
          PersistentMemory: 'on'
                    States: [1x1 embedded.fi]
       NumSamplesProcessed: 0

           CoeffWordLength: 16
            CoeffAutoScale: true
                    Signed: true

           InputWordLength: 16
           InputFracLength: 15

      StageInputWordLength: 16
       StageInputAutoScale: true

     StageOutputWordLength: 16
      StageOutputAutoScale: true

          OutputWordLength: 16
                OutputMode: 'AvoidOverflow'

           StateWordLength: 16
           StateFracLength: 15

               ProductMode: 'FullPrecision'

                 AccumMode: 'KeepMSB'
           AccumWordLength: 40
             CastBeforeSum: true

                 RoundMode: 'convergent'
              OverflowMode: 'wrap'

           InheritSettings: false
```

Check the filter coefficients to see that some of them are greater than 1.

```
hd.sosMatrix
```

```
ans =

    1.0000    1.5132    1.0000    1.0000   -0.9207    0.4373
    1.0000    0.3867    1.0000    1.0000   -0.2779    0.8242
    1.0000    0.0929    1.0000    1.0000   -0.0514    0.9610
    1.0000    0.0339    1.0000    1.0000   -0.0020    0.9934
```

Use `normalize` to modify the coefficients into the range between -1 and 1. A quick check of the SOS matrix shows all of the numerator coefficients now within the limits. You see that `g` contains the gains applied to each section of the SOS filter.

```
g = normalize(hd)

g =

    1.5132
    1.0000
    1.0000
    1.0000

hd.sosMatrix

ans =

    0.6608    1.0000    0.6608    1.0000   -0.9207    0.4373
    1.0000    0.3867    1.0000    1.0000   -0.2779    0.8242
    1.0000    0.0929    1.0000    1.0000   -0.0514    0.9610
    1.0000    0.0339    1.0000    1.0000   -0.0020    0.9934
```

Notice that none of the numerator coefficients exceed -1 or 1.

See Also

`denormalize`

nstates

Purpose

Return the number of filter states in a discrete-time filter or multirate filter

Syntax

```
n = nstates(hd)
n = nstates(hm)
```

Description

Discrete-Time Filters

`n = nstates(hd)` returns the number of states `n` in the discrete-time filter `hd`. The number of states depends on the filter structure and the coefficients.

Multirate Filters

`n = nstates(hm)` returns the number of states `n` in the multirate filter `hm`. The number of states depends on the filter structure and the coefficients.

Examples

Check the number of states for two different filters, one a direct form FIR filter, the other a multirate filter.

```
h=firls(30,[0 .1 .2 .5]*2,[1 1 0 0])

hd=dfilt.dffir(h)

hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'double'
               Numerator: [1x31 double]
    PersistentMemory: 'on'
                  States: [30x1 double]
     NumSamplesProcessed: 0

n=nstates(hd)

n =

    30


hm=mfilt.firfracdecim(2,3)
```

```
hm =

          FilterStructure: [1x46 char]
                Numerator: [1x72 double]
        RateChangeFactors: [2 3]
         PersistentMemory: false
                   States: [35x1 double]
      NumSamplesProcessed: 0

n=nstates(hm)

n =

    35
```

See Also

mfilt

order

Purpose

Return the filter order of a quantized filter

Syntax

```
n=order(hq)
```

Description

`n = order(hq)` returns the order `n` of the quantized filter `hq`. When `hq` is a single-section filter, `n` is the number of delays required for a minimum realization of the filter.

When `hq` has more than one section, `n` is the number of delays required for a minimum realization of the overall filter.

Examples

Create a discrete-time filter. Quantize the filter and convert to second-order section form. Then use `order` to check the order of the filter.

```
[b,a] = ellip(4,3,20,.6); % Create the reference filter.
hq = dfilt.df2(b,a);
% Quantize the filter and convert to second-order sections.
set(hq,'arithmetic','fixed');

n=order(hq) % Check the order of the overall filter.
n = 4
```

Purpose Return the phase delay of a discrete-time filter or multirate filter

Syntax

```
phasedelay(hd)
[phi,w] = phasedelay(hd,n)
[phi,w] = phasedelay(...,f)
phasedelay(hm)
[phi,w] = phasedelay(hm,n)
[phi,w] = phasedelay(...,f)
[phi,w] = phasedelay(...,fs)
```

Description The following sections describe `phasedelay` operation for discrete-time filters and multirate filters. For more information about optional input arguments for `phasedelay`, refer to `phasez` in the Signal Processing Toolbox.

Discrete-Time Filters

`phasedelay(hd)` displays the phase delay response of `hd` in the Filter Visualization Tool (FVTool).

`[phi,w]=phasedelay(hd,n)` returns vectors `phi` and `w` containing the instantaneous phase delay response of the adaptive filter `hd`, and the frequencies in radians at which it is evaluated. The response is evaluated at `n` points equally spaced around the upper half of the unit circle. When you do not specify `n`, it defaults to 8192.

If `hd` is a vector of filter objects, `phasedelay` returns `phi` as a matrix. Each column of `phi` corresponds to one filter in the vector. If you provide a row vector of frequency points `f` as an input argument, each row of `phi` corresponds to each filter in the vector. You can provide `fs`, the sampling frequency, as an input as well. `phasedelay` uses `fs` to calculate the delay response and plots the response to `fs`/2.

Multirate Filters

`phasedelay(hm)` displays the phase response of `hm` in the Filter Visualization Tool (FVTool).

`[phi,w]=phasedelay(hm,n)` returns vectors `phi` and `w` containing the instantaneous phase delay response of the adaptive filter `hm`, and the frequencies in radians at which it is evaluated. The response is evaluated at `n` points equally spaced around the upper half of the unit circle. When you do not specify `n`, it defaults to 8192.

If `hm` is a vector of filter objects, `phasedelay` returns `phi` as a matrix. Each column of `phi` corresponds to one filter in the vector. If you provide a row vector of frequency points `f` as an input argument, each row of `phi` corresponds to each filter in the vector.

Note that the multirate filter delay response is computed relative to the rate at which the filter is running. When you specify `fs` (the sampling rate) as an input argument, `phasedelay` assumes the filter is running at that rate.

For multistage cascades, `phasedelay` forms a single-stage multirate filter that is equivalent to the cascade and computes the response relative to the rate at which the equivalent filter is running. `phasedelay` does not support all multistage cascades. Only cascades for which it is possible to derive an equivalent single-stage filter are allowed for analysis.

As an example, consider a 2-stage interpolator where the first stage has an interpolation factor of 2 and the second stage has an interpolation factor of 4. An equivalent single-stage filter with an overall interpolation factor of 8 can be found. `phasedelay` uses the equivalent filter for the analysis. If a sampling frequency `fs` is specified as an input argument to `phasedelay`, the function interprets `fs` as the rate at which the equivalent filter is running.

See Also

`freqz`, `grpdelay`, `phasez`, `zerophase`, `zplane`

`freqz`, `fvtool`, `phasez`, `zerophase` in the Signal Processing Toolbox documentation

Purpose

Return the unwrapped phase response for an adaptive filter, discrete-time filter, or multirate filter

Syntax

```
phasez(ha)
[phi,w] = phasez(ha,n)
[phi,w] = phasez(...,f)
phasez(hd)
[phi,w] = phasez(hd,n)
[phi,w] = phasez(...,f)
phasez(hm)
[phi,w] = phasez(hm,n)
[phi,w] = phasez(...,f)
[phi,w] = phasez(...,fs)
```

Description

The following sections describe `phasez` operation for adaptive filters, discrete-time filters, and multirate filters. For more information about optional input arguments for `phasez`, refer to `phasez` in the Signal Processing Toolbox.

Adaptive Filters

For adaptive filters, `phasez` returns the instantaneous unwrapped phase response based on the current filter coefficients.

`phasez(ha)` displays the phase response of `ha` in the Filter Visualization Tool (FVTool).

`[phi,w]=phasez(ha,n)` returns vectors `phi` and `w` containing the instantaneous phase response of the adaptive filter `ha`, and the frequencies in radians at which it is evaluated. The phase response is evaluated at `n` points equally spaced around the upper half of the unit circle. When you do not specify `n`, it defaults to 8192.

If `ha` is a vector of filter objects, `phasez` returns `phi` as a matrix. Each column of `phi` corresponds to one filter in the vector. If you provide a row vector of frequency points `f` as an input argument, each row of `phi` corresponds to each filter in the vector.

Discrete-Time Filters

phasez(hd) displays the phase response of hd in the Filter Visualization Tool (FVTool).

[phi,w]=phasez(hd,n) returns vectors phi and w containing the instantaneous phase response of the adaptive filter hd, and the frequencies in radians at which it is evaluated. The phase response is evaluated at n points equally spaced around the upper half of the unit circle. When you do not specify n, it defaults to 8192.

If hd is a vector of filter objects, phasez returns phi as a matrix. Each column of phi corresponds to one filter in the vector. If you provide a row vector of frequency points f as an input argument, each row of phi corresponds to each filter in the vector.

Multirate Filters

phasez(hm) displays the phase response of hm in the Filter Visualization Tool (FVTool).

[phi,w]=phasez(hm,n) returns vectors phi and w containing the instantaneous phase response of the adaptive filter hm, and the frequencies in radians at which it is evaluated. The phase response is evaluated at n points equally spaced around the upper half of the unit circle. When you do not specify n, it defaults to 8192.

If hm is a vector of filter objects, phasez returns phi as a matrix. Each column of phi corresponds to one filter in the vector. If you provide a row vector of frequency points f as an input argument, each row of phi corresponds to each filter in the vector.

Note that the multirate filter response is computed relative to the rate at which the filter is running. When you specify fs (the sampling rate) as an input argument, phasez assumes the filter is running at that rate.

For multistage cascades, phasez forms a single-stage multirate filter that is equivalent to the cascade and computes the response relative to the rate at which the equivalent filter is running. phasez does not support all multistage cascades. Only cascades for which it is possible to derive an equivalent single-stage filter are allowed for analysis.

As an example, consider a 2-stage interpolator where the first stage has an interpolation factor of 2 and the second stage has an interpolation factor of 4. An equivalent single-stage filter with an overall interpolation factor of 8 can be found. `phasez` uses the equivalent filter for the analysis. If a sampling frequency `fs` is specified as an input argument to `phasez`, the function interprets `fs` as the rate at which the equivalent filter is running.

See Also

`freqz`, `grpdelay`, `phasedelay`, `zerophase`, `zplane`

`freqz`, `fvtool`, `phasez` in the Signal Processing Toolbox documentation

polyphase

Purpose

Return the polyphase decomposition of multirate filters

Syntax

```
p = polyphase(hm)
polyphase(hm)
```

Description

`p = polyphase(hm)` returns the polyphase matrix `p` of the multirate filter `hm`. Each row in the matrix represents one subfilter of the multirate filter. The first row of matrix `p` represents the first subfilter, the second row the second subfilter, and so on to the last subfilter.

`polyphase(hm)` called with no output argument launches the Filter Visualization Tool (FVTool) with all the polyphase subfilters to allow you to analyze each component subfilter individually.

Examples

When you create a multirate filter that uses polyphase decomposition, polyphase lets you analyze the component filters indiviually by returning the components as rows in a matrix.

This example creates an interpolate by eight filter.

```
hm=mfilt.firinterp(8)

hm =

         FilterStructure: 'Direct-Form FIR Polyphase Interpolator'
               Numerator: [1x192 double]
     InterpolationFactor: 8
        PersistentMemory: false
                  States: [23x1 double]
     NumSamplesProcessed: 0
```

In this syntax, the matrix p contains all of the subfilters for hm, one filter per matrix row.

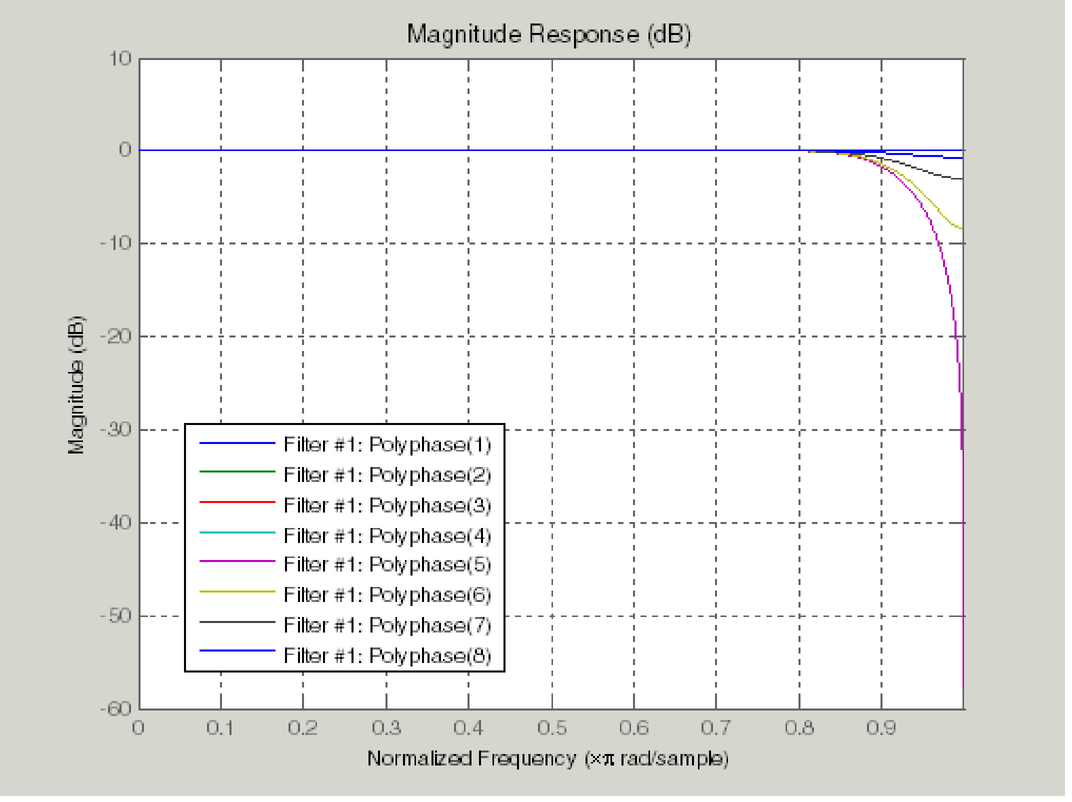
```
p=polyphase(hm)

p =

  Columns 1 through 8

         0         0         0         0         0         0         0         0
   -0.0000    0.0002   -0.0006    0.0013   -0.0026    0.0048   -0.0081    0.0133
   -0.0001    0.0004   -0.0012    0.0026   -0.0052    0.0094   -0.0160    0.0261
   -0.0001    0.0006   -0.0017    0.0038   -0.0074    0.0132   -0.0223    0.0361
```

```
  -0.0002    0.0008   -0.0020    0.0045   -0.0086    0.0153   -0.0257    0.0415
  -0.0002    0.0008   -0.0021    0.0045   -0.0086    0.0151   -0.0252    0.0406
  -0.0002    0.0007   -0.0018    0.0038   -0.0071    0.0124   -0.0205    0.0330
  -0.0001    0.0004   -0.0011    0.0022   -0.0041    0.0072   -0.0118    0.0189

 Columns 9 through 16

        0         0         0         0    1.0000         0         0         0
  -0.0212    0.0342   -0.0594    0.1365    0.9741   -0.1048    0.0511   -0.0303
  -0.0416    0.0673   -0.1189    0.2958    0.8989   -0.1730    0.0878   -0.0527
  -0.0576    0.0938   -0.1691    0.4659    0.7814   -0.2038    0.1071   -0.0648
  -0.0661    0.1084   -0.2003    0.6326    0.6326   -0.2003    0.1084   -0.0661
  -0.0648    0.1071   -0.2038    0.7814    0.4659   -0.1691    0.0938   -0.0576
  -0.0527    0.0878   -0.1730    0.8989    0.2958   -0.1189    0.0673   -0.0416
  -0.0303    0.0511   -0.1048    0.9741    0.1365   -0.0594    0.0342   -0.0212

 Columns 17 through 24

        0         0         0         0         0         0         0         0
   0.0189   -0.0118    0.0072   -0.0041    0.0022   -0.0011    0.0004   -0.0001
   0.0330   -0.0205    0.0124   -0.0071    0.0038   -0.0018    0.0007   -0.0002
   0.0406   -0.0252    0.0151   -0.0086    0.0045   -0.0021    0.0008   -0.0002
   0.0415   -0.0257    0.0153   -0.0086    0.0045   -0.0020    0.0008   -0.0002
   0.0361   -0.0223    0.0132   -0.0074    0.0038   -0.0017    0.0006   -0.0001
   0.0261   -0.0160    0.0094   -0.0052    0.0026   -0.0012    0.0004   -0.0001
   0.0133   -0.0081    0.0048   -0.0026    0.0013   -0.0006    0.0002   -0.0000
```

Finally, using `polyphase` without an output argument opens the Filter Visualization Tool, ready for you to use the analysis capabilities of the tool to investigate the interpolator `hm`.

```
polyphase(hm)
```

In this figure, we switch FVTool to show the magnitude responses for the subfilters.



See Also

`mfilt`

Purpose Directly realize a Simulink subsystem block for a direct-form quantized filter

Syntax

```
realizemdl(hq)
realizemdl(hq, propertyname1, propertyvalue1,...)
```

Description

`realizemdl(hq)` generates a model of filter `hq` in a Simulink subsystem block using sum, gain, and delay blocks from Simulink. The properties and values of `hq` define the resulting subsystem block parameters.

`realizemdl` requires Simulink. To accurately realize models of quantized filters, use Simulink Fixed-Point.

`realizemdl(hq,propertyname1,propertyvalue1,...)` generates the model or `hq` with the associated `propertyname`/`propertyvalue` pairs, and any other values you set in `hq`.

Note Subsystem filter blocks that you use `realizemdl` to create support sample-based input and output only. You cannot input or output frame-based signals with the block.

Using the optional propertyname/propertyvalue pairs lets you control more fully the way the block subsystem model gets built, such as where the block goes, what the name is, or how to optimize the block structure. Valid properties and values for `realizemdl` are listed in this table, with the default value noted and descriptions of what the properties do.

Property Name	Property Values	Description
`Destination`	`'current'` (default) or `'new'`	Specify whether to add the block to your current Simulink model or create a new model to contain the block.
`Blockname`	`'filter'` (default)	Provides the name for the new subsystem block. By default the block is named `'filter'`. To enter a name for the block, use the `propertyvalue` set to a string '*`blockname`*'.

Property Name	Property Values	Description
`OverwriteBlock`	`'off'` or `'on'`	Specify whether to overwrite an existing block with the same name or create a new block.
`OptimizeZeros`	`'off'` (default) or `'on'`	Specify whether to remove zero-gain blocks.
`OptimizeOnes`	`'off'` (default) or `'on'`	Specify whether to replace unity-gain blocks with direct connections.
`OptimizeNegOnes`	`'off'` (default) or `'on'`	Specify whether to replace negative unity-gain blocks with a sign change at the nearest sum block.
`OptimizeDelayChains`	`'off'` (default) or `'on'`	Specify whether to replace cascaded chains of delay blocks with a single integer delay block to provide an equivalent delay.

Examples

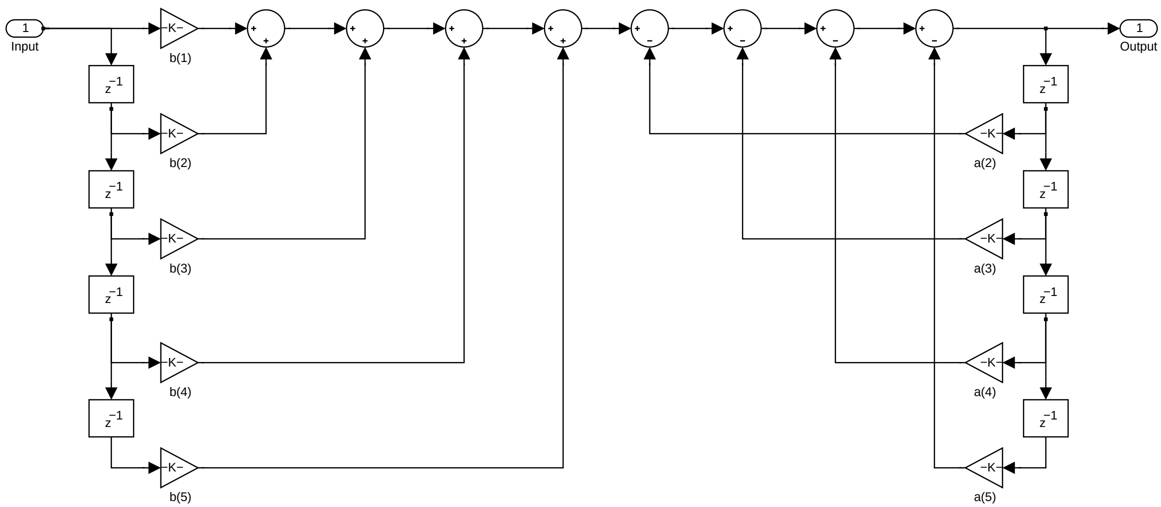
To demonstrate how `realizemdl` works to create models, these two examples show the default and optional syntaxes in use. Both examples begin from a quantized filter designed by `butter` in the Signal Processing Toolbox.

```
[b,a] = butter(4,.5);
hq = dfilt.df1(b,a);
```

Example 1—Using the default syntax to realize a model of your quantized filter `hq`. When you use this syntax, `realizemdl` uses blocks from Simulink and Simulink Fixed-Point to realize the subsystem in your current Simulink model.

```
realizemdl(hq);
```

Look at the figure to see the model as realized by `realizemdl`.



Example 2—Using propertyname/propertyvalue pairs to specify the features of the subsystem block model created by `realizemdl`.

First, convert the filter to fixed-point arithmetic to ensure a few zero valued coefficients:

```
hq.arithmetic = 'fixed';
```

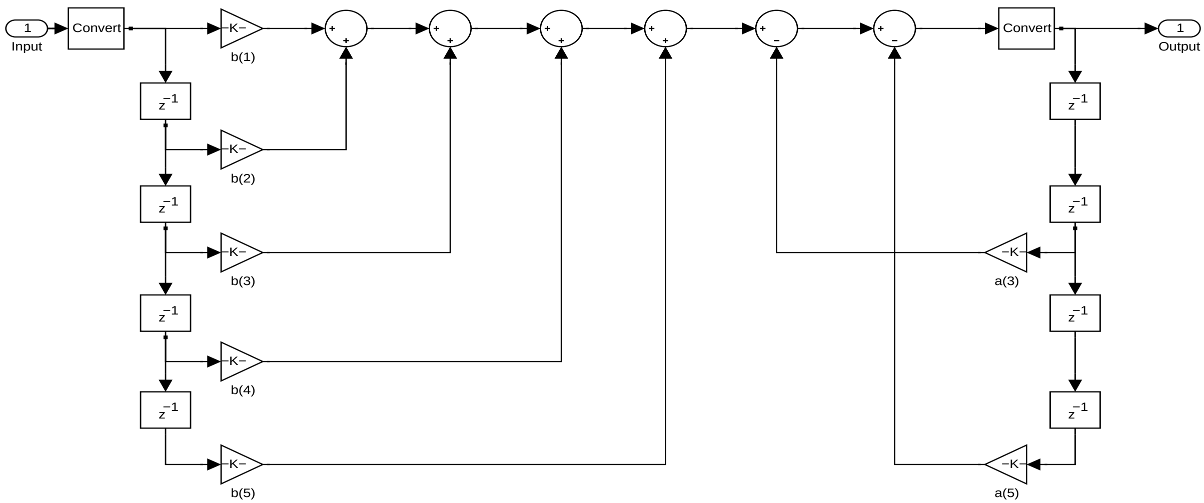
Your filter has two zero value denominators, a(2) and a(4):

```
FilterStructure: 'Direct-Form I'
              Arithmetic: 'fixed'
               Numerator: [0.0940 0.3759 0.5639 0.3759 0.0940]
             Denominator: [1 0 0.4860 0 0.0176]
        PersistentMemory: false
                  States: Numerator:  [4x1 fi]
                          Denominator:[4x1 fi]
     NumSamplesProcessed: 0
```

Now realize the model implementation.

```
realizemdl(hq,'optimizezeros','on',...
'blockname','newfiltermodel');
```

Since this example uses the optional property name `optimizezeros`, set to 'on', the resulting block subsystem is slightly different—the zero-gain blocks for coefficients `a(2)` and `a(4)` are not included in the subsystem.



See Also

`realizemdl` under the methods for `dfilt` in the Signal Processing Toolbox

Purpose

Return the double-precision floating-point reference filter that can correspond to a fixed-point or single-precision floating-point filter

Syntax

```
href = reffilter(hd)
```

Description

`href = reffilter(hd)` returns a new filter `href` that has the same structure as `hd`, but uses the reference coefficients and has its arithmetic property set to `double`. Note that `hd` can be either a fixed-point filter (arithmetic property set to '`fixed`', or a single-precision floating-point filter whose arithmetic property is '`single`').

`reffilter(hd)` differs from `double(hd)` in that

- the filter `href` returned by `reffilter` has the reference coefficients of `hd`.
- `double(hd)` returns the quantized coefficients of `hd` represented in double-precision.

To check the performance of your fixed-point filter, use `href = reffilter(hd)` to quickly have the floating-point, double-precision version of `hd` available for comparison.

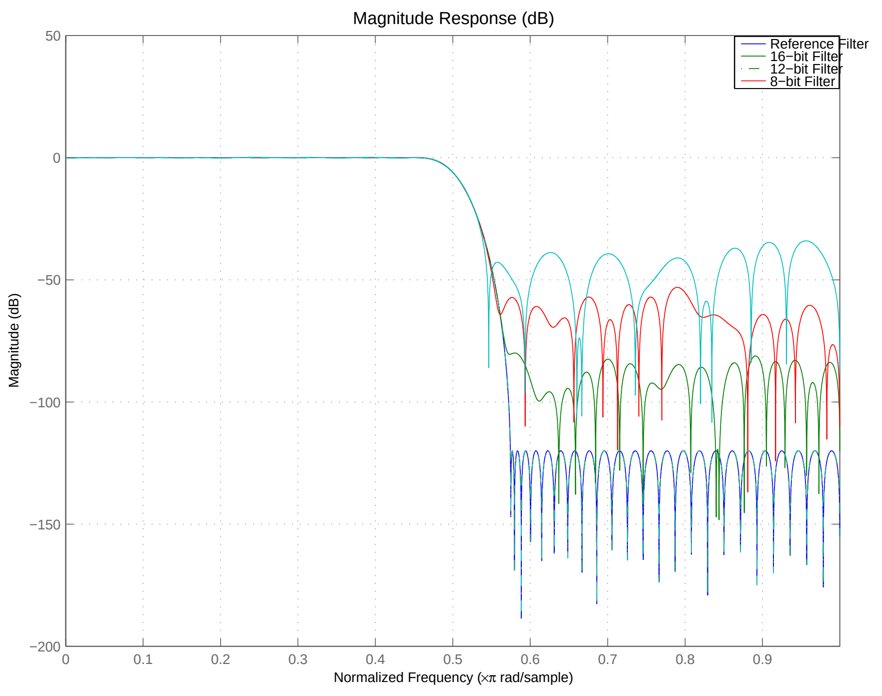
Examples

Compare several fixed-point quantizations of a filter with the same double-precision floating-point version of the filter.

```
h = dfilt.dffir(firceqrip(87,.5,[1e-3,1e-6])); % Lowpass filter.
h1 = copy(h); h2 = copy(h); % Create copies of h.
h.arithmetic = 'fixed';   % Set h to filter using fixed-point
                          % arithmetic.
h1.arithmetic = 'fixed';  % Same for h1.
h2.arithmetic = 'fixed';  % Same for h2.
h.CoeffWordLength  = 16;  % Use 16 bits to represent the
                          % coefficients.
h1.CoeffWordLength = 12;  % Use 12 bits to represent the
                          % coefficients.
h2.CoeffWordLength =  8;  % Use 8 bits to represent the
                          % coefficients.
href = reffilter(h);
hfvt = fvtool(href,h,h1,h2);
set(hfvt,'ShowReference','off'); % Reference displayed once
                                 % already.
```

```
legend(hfvt,'Reference filter','16-bits','12-bits','8-bits');
```

The following plot, taken from FVTool, shows `href`, the reference filter, and the effects of using three different word lengths to represent the coefficients.



As expected, the fidelity of the fixed-point filters suffers as you change the representation of the coefficients. With `href` available, it is easy to see just how the fixed-point filter compares to the ideal.

See Also

`double`

Purpose

Rearrange the order of the sections in a second-order section filter

Syntax

```
reorder(hd,order)
reorder(hd,numorder,denorder)
reorder(hd,numorder,denorder,svorder)
reorder(hd,filter_type)
reorder(hd,dir_flag)
reorder(hd,dir_flag,sv))
```

Description

`reorder(hd,order)` rearranges the sections of filter `hd` using the vector of indices provided in `order`.

`order` does not need to contain all of the indices of the filter. Omitting one or more filter section indices removes the omitted sections from the filter. You can use a logical array to remove sections from the filter, but not to reorder it (refer to the Examples to see this done).

`reorder(hd,numorder,denorder)` reorders the numerator and denominator separately using the vectors of indices in `numorder` and `denorder`. These two vectors must be the same length.

`reorder(hd,numorder,denorder,svorder)` the scale values can be independently reordered. When `svorder` is not specified, the scale values are reordered with the numerator. The output scale value always remains on the end when you use the argument `numorder` to reorder the scale values.

`reorder(hd,filter_type)` where `filter_type` is one of `auto`, `lowpass`, `highpass`, `bandpass`, or `bandstop`, reorders `hd` in a way suitable for the filter type you specify by `filter_type`. This reordering mode can be especially helpful for fixed-point implementations where the order of the filter sections can significantly affect your filter performance.

The `auto` option and automatic ordering only apply to filters that you used `fdesign` to create.With the `auto` option as an input argument, `reorder` automatically rearranges the filter sections depending on the specification response type of the design, such as lowpass, or bandstop. This technique appears in the first example.

`reorder(hd,dir_flag)` if `dir_flag` is `up`, the first filter section contains the poles closest to the origin, and the last section contains the poles closest to the unit circle. When dir_flag is down, the sections are ordered in the opposite direction. reorder always pairs zeros with the poles closest to them.

reorder(hd,dir_flag,sv) sv is either the string poles or zeros and describes how to reorder the scale values. By default the scale values are not reordered when you use the dir_flag option.

Examples

Being able to rearrange the order of the sections in a filter can be a powerful tool for controlling the filter process. This example uses reorder to change the sections of a df2sos filter, and displays the results of the changes in Filter Visualization Tool using setfilter to select the filter to display.

First use the automatic reordering option on a lowpass filter.

```
d = fdesign.lowpass('n,fc',15,0.5)
hd = butter(d);
d =

           ResponseType: 'Lowpass with cutoff'
      SpecificationType: 'N,Fc'
            Description: {2x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
            FilterOrder: 15
                Fcutoff: 0.5000

reorder(hd,'auto')
hd

hd =

        FilterStructure: 'Direct-Form II, Second-Order Sections'
             Arithmetic: 'double'
              sosMatrix: [8x6 double]
            ScaleValues: [9x1 double]
       PersistentMemory: false
                 States: [2x8 double]
    NumSamplesProcessed: 0
```

Create an SOS filter in the direct form II implementation.

```
[z,p,k] = butter(15,.5);
[sos, g] = zp2sos(z,p,k);
hd = dfilt.df2sos(sos,g);
```

Reorder the sections by moving the second section to be between the seventh and eighth sections.

```
reorder(hd, [1 3:7 2 8]);
hfvt = fvtool(hd, 'analysis', 'coefficients');
```

Remove the third, fourth and seventh sections.

```
hd1 = copy(hd);
reorder(hd1, logical([1 1 0 0 1 1 0 1]));
setfilter(hfvt, hd1);
```

Move the first filter to the end and remove the eighth section

```
hd2 = copy(hd);
reorder(hd2, [2:7 1]);
setfilter(hfvt, hd2);
```

Move the numerator and denominator independently.

```
hd3 = copy(hd);
reorder(hd3, [1 3:8 2], [1:8]);
setfilter(hfvt, hd3);
```

See Also

`cumsec`, `scale`, `scaleopts`

Reference

Schlichthärle, Dietrich, *Digital Filters Basics and Design*, Springer-Verlag Berlin Heidelberg, 2000.

reset

Purpose

Reset the properties of adaptive filters, discrete-time filters, and multirate filters to their initial conditions

Syntax

```
reset(ha)
reset(hd)
reset(hm)
```

Description

`reset(ha)` resets all the properties of the adaptive filter `ha` that are updated when filtering to the value specified at construction. If you do not specify a value for any particular property when you construct an adaptive filter, the property value for that property is reset to the default value for the property.

`reset(hd)` resets all the properties of the discrete-time filter `hd` to their factory values that are modifed when you run the filter. In particular, the `NumSamplesProcessed` and `States` properties are reset to zero.

`reset(hm)` resets all the properties of the multirate filter `hm` to their factory value that are modifed when the filter is run. In particular, the `NumSamplesProcessed` and `States` properties are reset to zero when `hm` is a decimator. Additionally, the filter internal properties are also reset to their factory values.

Examples

Denoise a sinusoid and reset the filter after filtering with it.

```
h = adaptfilt.lms(5,.05,1,[0.5,0.5,0.5,0.5,0.5]);
n = filter(1,[1 1/2 1/3],.2*randn(1,2000));
d = sin((0:1999)*2*pi*0.005) + n; % Noisy sinusoid
x = n;
[y,e]= filter(h,x,d);             % e has denoised signal
disp(h)
reset(h); % Reset the coefficients, states, and the
          % NumSamplesProcessed property.
disp(h)
```

See Also

`quantizer`, `set`

Purpose Scale the sections of second-order section filters

Syntax

```
scale(hd)
scale(hd,pnorm)
scale(hd,pnorm,p1v,p2,v2, )
scale(hd,pnorm,opts)
```

Description `scale(hd)` scales the second-order section filter `hd` using peak magnitude response scaling (L-infinity, Linf), to reduce the possibility of overflows when your filter `hd` operates in fixed-point arithmetic mode.

`scale(hd,pnorm)` specifies the norm used to scale the filter. `pnorm` can be either a discrete-time-domain norm or a frequency-domain norm.

Valid time-domain norm values for `pnorm` are `l1`, `l2`, and `linf`. Valid frequency-domain norm values are `L1`, `L2`, and `Linf`. Note that `L2` norm is equal to `l2` norm (by Parseval's theorem) but this is not true for other norms—`l1` is not the same as `L1` and `Linf` is not the same as `linf`.

Filter norms can be ordered in terms of how stringent they are, as follows from most stringent to least:

`l1 >= Linf >= L2 = l2 >= L1 >= linf`

Using `l1`, the most stringent scaling, produces a filter that is least likely to overflow, but has the worst signal-to-noise ratio performance. `Linf` scaling, the least stringent, and the default scaling, is the most commonly used scaling norm.

scale(hd,pnorm,p1,v1,p2,v2,...) uses parameter name/parameter value pair input arguments to specify optional scaling parameters. Valid parameter names and options values appear in the table.

Parameter	Default	Description and Valid Value
MaxNumerator	2	Maximum allowed value for numerator coefficients.
MaxScaleValue	Not Used	Maximum allowed scale values. The filter applies the MaxScaleValue limit only when you set ScaleValueConstraint to a value other than unit (the default setting). Setting MaxScaleValue to any numerical value automatically changes the ScaleValueConstraint setting to none.
NumeratorConstraint	none	Specifies whether and how to constrain numerator coefficient values. Options are none, normalize, po2, and unit
OverflowMode	wrap	Sets the way the filter handles arithmetic overflow situations during scaling. Choose from wrap, saturate or satall.

Parameter	Default	Description and Valid Value
ScaleValueConstraint	unit	Specify whether to constrain the filter scale values, and how to constrain them. Valid options are none, po2, and unit. Choosing unit for the constraint disables the MaxScaleValue property setting. po2 constrains the scale values to be powers of 2, while none removes any constraint on the scale values.
sosReorder	auto	Reorder filter sections prior to applying scaling. Select one of auto, none, up, or down.

If your device does not have guard bits available and you are using saturation arithmetic for filtering, use the satall setting for OverFlowMode instead of saturate.

With the Arithmetic property of hd set to double or single, the filter uses the default values for all options that you do not specify explicitly. When you set Arithmetic to fixed, the values used for the scaling options are set according to the settings in filter hd. However, if you specify a scaling option different from the settings in hd, the filter uses your explicit option selection for scaling purposes, but does not change the property setting in hd.

scale(hd,pnorm,opts) uses an input scale options object opts to specify the optional scaling parameters in lieu of specifying parameter-value pairs. You can create the opts object using

```
opts = scaleopts(hd)
```

For more information about scaling objects, refer to scaleopts in the Help system.

Examples

Demonstrate the Linf-norm scaling of a lowpass elliptic filter with second-order sections. Start by creating a lowpass elliptical filter in zero, pole, gain (z,p,k) form.

```
[z,p,k] = ellip(5,1,50,.3);
[sos,g] = zp2sos(z,p,k);
hd = dfilt.df2sos(sos,g);
scale(hd,'linf','scalevalueconstraint','none','maxscalevalue',2)
```

See Also

cumsec, norm, reorder, scalecheck, scaleopts

Purpose Check the scaling of a second-order sections filter

Syntax

```
s = scalecheck(hd,pnorm)
```

Description

For df1sos and df2tsos Filters

`s = scalecheck(hd,pnorm)` returns a row vector `s` that reports the p-norm of the filter computed from the filter input to the output of each second-order section. Therefore, the number of elements in `s` is one less than the number of sections in the filter. Note that this p-norm computation does not include the trailing scale value of the filter (which you can find by entering

```
hd.scalevalue(end)
```

at the MATLAB prompt.

`pnorm` can be either frequency-domain norms specified by `L1`, `L2`, or `Linf` or discrete-time-domain norms—`l1`, `l2`, `linf`. Note that the L2-norm of a filter is equal to the l2-norm (Parseval's theorem). This is not true for other norms.

For df2sos and df1tsos Filters

`s = scalecheck(hd,pnorm)` returns `s`, a row vector whose elements contain the p-norm from the filter input to the input of the recursive part of each second-order section. This computation of the p-norm corresponds to the input to the multipliers in these filter structures, and are the locations in the signal flow where overflow should be avoided.

When `hd` has nontrivial scale values, that is, if any scale values are not equal to one, `s` is a two-row matrix, rather than a vector. The first row elements of `s` report the p-norm of the filter computed from the filter input to the output of each second-order section. The elements of the second row of `s` contain the p-norm computed from the input of the filter to the input of each scale value between the sections. Note that for `df2sos` and `df1tsos` filter structures, the last numerator and the trailing scale value for the filter are not included when `scalecheck` checks the scale.

For a given p-norm, an optimally scaled filter has partial norms equal to one, so matrix `s` contain all ones.

Examples Check the Linf-norm scaling of a filter.

```
hs = fdesign.lowpass; % Create a filter design specifications
object.
hd = ellip(hs);       % Design an elliptic sos filter
scale(hd,'Linf');
s = scalecheck(hd,'Linf')
```

Or, in another form:

```
[b,a]=ellip(10,.5,20,0.5);
[s,g]=tf2sos(b,a);
hd=dfilt.df1sos(s,g)

hd =

          FilterStructure: 'Direct-Form I, Second-Order Sections'
               Arithmetic: 'double'
                sosMatrix: [5x6 double]
              ScaleValues: [6x1 double]
         PersistentMemory: false
                   States: [1x1 filtstates.dfiir]
      NumSamplesProcessed: 0

1x1 struct array with no fields.

scalecheck(hd,'Linf')

ans =

    0.7631    0.9627    0.9952    0.9994    1.0000
```

See Also

norm, reorder, scale, scaleopts

Purpose Create an object that contains scaling options for second-order section scaling

Syntax

```
opts = scaleopts(hd)
```

Description

`opts = scaleopts(hd)` uses the current settings in the filter `hd` to create an options object `opts` that contains specified scaling options for second-order section scaling. You can pass `opts` to the `scale` method as an input argument to apply scaling settings to a second-order filter.

Within `opts`, the scaling options object returned by `scaleopts`, you can set the following properties:

Parameter	Default	Description and Valid Value
`MaxNumerator`	2	Maximum allowed value for numerator coefficients.
`MaxScaleValue`	No default value	Maximum allowed scale values. The filter applies the `MaxScaleValue` limit only when you set `ScaleValueConstraint` to a value other than `unit`. Setting `MaxScaleValue` to a numerical value automatically changes the `ScaleValueConstraint` setting to `none`.
`NumeratorConstraint`	`none`	Specifies whether and how to constrain numerator coefficient values. Options are `none`, `normalize`, `po2`, and `unit`,

Parameter (Continued)	Default	Description and Valid Value
OverflowMode	wrap	Sets the way the filter handles arithmetic overflow situations during scaling. Choose either wrap or saturate
ScaleValueConstraint	unit	Specify whether to constrain the filter scale values, and how to constrain them. Valid options are none, po2, and unit

When you set the properties of opts and then use opts as an input argument to scale(hd,opts), scale applies the settings in opts to scale hd.

Examples

From a filter hd, you can create an options scaling object that contains the scaling options settings you require.

```
[b,a]=ellip(10,.5,20,0.5);
[s,g]=tf2sos(b,a);
hd=dfilt.df1sos(s,g)
opts=scaleopts(hd)

opts =

            MaxNumerator: 2
     NumeratorConstraint: 'none'
            OverflowMode: 'wrap'
    ScaleValueConstraint: 'unit'
           MaxScaleValue: 'Not used'
```

See Also

cumsec, norm, reorder, scale, scalecheck

Purpose Set the specifications for an existing filter design (`fdesign`) object

Syntax

```
setspecs(d,specvalue1,specvalue2,...)
setspecs(d,SpecificationType,specvalue1,specvalue2,...)
setspecs(...,fs)
setspecs(...,inputunits)
```

Description

`setspecs(d,specvalue1,specvalue2,...)` Set the specifications in the order that they appear in the `SpecificationType` property for the design object `d`.

`setspecs(d,SpecificationType,specvalue1,specvalue2,...)` lets you change the specifications for the object and set values for the new specifiers. When you already have a filter design object, this syntax lets you change the `SpecificationType` string and the associated specification values for the object, rather than recreating the object to change it.

`setspecs(...fs)` Set the `fs`. If you choose to specify the `fs`, it must be immediately after you provide all of the specifications for the current `SpecificationType`. Refer to Examples to see this being used.

`setspecs(...,`*`inputunits`*`)` Specifying the *`inputunits`* option allows you to specify your filter magnitude specification values in different units. *`inputunits`* can be either of these strings:

- '**`linear`**'—to indicate that your input specification values represent linear units, such as decimal values for the filter feature locations when you select normalized sampling frequency.
- '**`squared`**'—indicating that your input specification values represent squared magnitude values, usually dB. This is the default value. When you omit the *`inputunits`* argument, `setspecs` assumes all specification values are in square magnitude form.

You are not required to provide `fs`, the sampling frequency, as an input when you use the `inputunits` option. As you see from the syntax options, the `inputunits` option must be the rightmost input argument in the syntax—`inputunits` must be passed as the final input.

Examples

To demonstrate using `setspecs`, the following examples show how to use various syntax forms to set the values in filter design objects.

Example 1

Create a lowpass design object d using filter order and a cutoff value for the location of the edge of the passband. Then change the cutoff and order specifications of d.

```
d = fdesign.lowpass('n,fc')

d =

             ResponseType: 'Lowpass with cutoff'
        SpecificationType: 'N,Fc'
              Description: {2x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
              FilterOrder: 10
                  Fcutoff: 0.5000

setspecs(d, 20, .4);

d =

             ResponseType: 'Lowpass with cutoff'
        SpecificationType: 'N,Fc'
              Description: {2x1 cell}
      NormalizedFrequency: true
                       Fs: 'Normalized'
              FilterOrder: 20
                  Fcutoff: 0.4000
```

Example 2

Now specify a sampling frequency after you make d.

```
d = fdesign.lowpass('n,fc')

d =

             ResponseType: 'Lowpass with cutoff'
        SpecificationType: 'N,Fc'
              Description: {2x1 cell}
      NormalizedFrequency: true
```

```
                 Fs: 'Normalized'
        FilterOrder: 10
            Fcutoff: 0.5000


setspecs(d, 20, 4, 20);

d

d =

           ResponseType: 'Lowpass with cutoff'
      SpecificationType: 'N,Fc'
            Description: {2x1 cell}
    NormalizedFrequency: false
                     Fs: 20
            FilterOrder: 20
                Fcutoff: 4
```

Example 3

This example uses the `inputunits` argument to change from the default setting of square to linear unit. Start with the default lowpass design object that specifies the edge locations for the passband and stopband, and the desired attenuation in the pass- and stopbands.

```
d=fdesign.lowpass

d =

           ResponseType: 'Minimum-order lowpass'
      SpecificationType: 'Fp,Fst,Ap,Ast'
            Description: {4x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
                  Fpass: 0.4500
                  Fstop: 0.5500
                  Apass: 1
                  Astop: 60
```

Convert to linear input values and reset the filter spec for d at the same time. With the linear argument included, the inputs for the response features now need to be in linear units.

```
setspecs(d,.4,.5,.1,.05,'linear')
d

d =

           ResponseType: 'Minimum-order lowpass'
      SpecificationType: 'Fp,Fst,Ap,Ast'
            Description: {4x1 cell}
    NormalizedFrequency: true
                     Fs: 'Normalized'
                  Fpass: 0.4000
                  Fstop: 0.5000
                  Apass: 1.7430
                  Astop: 26.0206
```

Example 4

Finally, use `setspecs` to change the `SpecificationType` string and apply new filter specifications to `d`.

```
d=fdesign.decim(3)

d =

           ResponseType: 'Minimum-order nyquist'
      SpecificationType: 'TW,Ast'
            Description: {2x1 cell}
       DecimationFactor: 3
    NormalizedFrequency: true
                     Fs: 'Normalized'
        TransitionWidth: 0.1000
                  Astop: 80


setspecs(d,'n,ast',16,70)
d

d =

           ResponseType: 'Nyquist with filter order and stopband attenuation'
      SpecificationType: 'N,Ast'
            Description: {2x1 cell}
       DecimationFactor: 3
```

```
NormalizedFrequency: true
                 Fs: 'Normalized'
  PolyphaseLength: 16
            Astop: 70
```

See Also

designmethods, fdesign.bandpass, fdesign.bandstop, fdesign.decim, fdesign.halfband, fdesign.highpass, fdesign.interp, fdesign.lowpass, fdesign.nyquist, fdesign.src

Purpose

Provide access to all the fixed-point scaling modes and features in direct-form FIR filter objects

Syntax

```
specifyall(hd)
specifyall(hd,false)
specifyall(hd,true)
```

Description

`specifyall` sets all of the `autoscale` property values of direct-form FIR filters to `false` and all `*modes` of the filters to `SpecifyPrecision`. In this table, you see the results of using `specifyall` with direct-form FIR filters.

Property Name	Default	Setting After Applying specifyall
CoeffAutoScale	true	false
OutputMode	AvoidOverflow	SpecifyPrecision
ProductMode	FullPrecision	SpecifyPrecision
AccumMode	KeepMSB	SpecifyPrecision
RoundMode	convergent	convergent
OverflowMode	wrap	wrap

`specifyall(hd)` gives you maximum control over all settings in a filter `hd` by setting all of the autoscale options that are `true` to `false`, turning off all autoscaling and resetting all modes—`OutputMode`, `ProductMode`, and `AccumMode`—to `SpecifyPrecision`. After you use `specifyall`, you must supply the property values for the mode- and scaling related properties.

`specifyall` provides an alternative to changing all these properties individually. Do note that `specifyall` changes all of the settings; to set some but not all of the modes, set each property as you require.

`specifyall(hd,false)` performs the opposite operation of `specifyall(hd)` by setting all of the autoscale options to `true`; all of the modes to their default values; and hiding the fraction length properties in the display, meaning you cannot access them to set them or view them.

`specifyall(hd,true)` is equivalent to `specifyall(hd)`.

Examples

This examples demonstrates using specifyall to provide access to all of the fixed-point settings of an FIR filter implemented with the direct-form structure. Notice the displayed property values shown after you change the filter to fixed-point arithmetic, then after you use specifyall to disable all of the automatic filter scaling and reset the mode values.

```
b = fircband(12,[0 0.4 0.5 1], [1 1 0 0], [1 0.2], {'w' 'c'});
hd = dfilt.dffir(b);
hd.arithmetic = 'fixed'
hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'fixed'
               Numerator: [1x13 double]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: 'true'
                  Signed: 'on'

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'AvoidOverflow'

             ProductMode: 'FullPrecision'

               AccumMode: 'KeepMSB'
         AccumWordLength: 40
           CastBeforeSum: 'on'

               RoundMode: 'convergent'
            OverflowMode: 'wrap'

         InheritSettings: 'off'

specifyall(hd)
```

```
hd

hd =

         FilterStructure: 'Direct-Form FIR'
              Arithmetic: 'fixed'
               Numerator: [1x13 double]
        PersistentMemory: false
                  States: [1x1 embedded.fi]
     NumSamplesProcessed: 0

         CoeffWordLength: 16
          CoeffAutoScale: false
           NumFracLength: 16
                  Signed: true

         InputWordLength: 16
         InputFracLength: 15

        OutputWordLength: 16
              OutputMode: 'SpecifyPrecision'
        OutputFracLength: 11

             ProductMode: 'SpecifyPrecision'
       ProductWordLength: 32
       ProductFracLength: 31

               AccumMode: 'SpecifyPrecision'
         AccumWordLength: 40
         AccumFracLength: 31
           CastBeforeSum: true

               RoundMode: 'convergent'
            OverflowMode: 'wrap'

         InheritSettings: false
```

The mode properties `InputMode`, `ProductMode`, and `AccumMode` now have the value `SpecifyPrecision` and the fraction length properties appear in the display. Now you use the properties (`InputFracLength`, `ProdFracLength`, AccumFracLength) to set the precision the filter applies to the input, product, and accumulator operations. CoeffAutoScale switches to false, meaning autoscaling of the filter coefficients will not be done to prevent overflows. None of the other filter properties change when you apply specifyall.

See Also

double, reffilter
fi, fimath in the Fixed-Point Toolbox

Purpose

Convert a quantized filter to second-order section form, order, and scale

Syntax

```
Hq2 = sos(Hq)
Hq2 = sos(Hq, order)
Hq2 = sos(Hq, order, scale)
```

Description

`Hq2 = sos(Hq)` returns a quantized filter `Hq2` that has second-order sections and the `dft2` structure. Use the same optional arguments used in `tf2sos`.

`Hq2 = sos(Hq, order)` specifies the order of the sections in `Hq2`, where `order` is either of the following strings:

- `'down'` — to order the sections so the first section of `Hq2` contains the poles closest to the unit circle (L_∞ norm scaling)
- `'up'` — to order the sections so the first section of `Hq2` contains the poles farthest from the unit circle (L_2 norm scaling and the default)

`Hq2 = sos(Hq, order, scale)` also specifies the desired scaling of the gain and numerator coefficients of all second-order sections, where `scale` is one of the following strings:

- `'none'` — to apply no scaling (default)
- `'inf'` — to apply infinity-norm scaling
- `'two'` — to apply 2-norm scaling

Use infinity-norm scaling in conjunction with up-ordering to minimize the probability of overflow in the filter realization. Consider using 2-norm scaling in conjunction with down-ordering to minimize the peak round-off noise.

When `Hq` is a fixed-point filter, the filter coefficients are normalized so that the magnitude of the maximum coefficient in each section is 1. The gain of the filter is applied to the first scale value of `Hq2`.

`sos` uses the direct form II transposed (`dft2`) structure to implement second-order section filters.

Examples

```
[b,a]=butter(8,.5);
Hq = dfilt.df2t(b,a);
Hq.arithmetic = 'fixed';
Hq1 = sos(Hq)
```

See Also `convert`, `dfilt`

`tf2sos` in your Signal Processing Toolbox documentation

stepz

Purpose

Return the step response for adaptive, discrete-time, or multirate filters

Syntax

```
[h,t] = stepz(ha)
stepz(ha)
[h,t] = stepz(hm)
stepz(hm)
```

Description

The next sections describe common `stepz` operation with adaptive and multirate filters. For more input options and for information about using `stepz` with discrete-time filters, refer to `stepz` in the Signal Processing Toolbox.

Adaptive Filters

For adaptive filters, `stepz` returns the instantaneous zero-phase response based on the current filter coefficients.

`[h,t] = stepz(ha)` returns the step response `h` of the multirate filter `ha`. The length of column vector `h` is the length of the impulse response of `ha`. Returned vector `t` contains the time samples at which `stepz` evaluated the step response. `stepz` returns `h` as a matrix when `ha` is a vector of filters. Each column of the matrix corresponds to one filter in the vector.

`stepz(ha)` displays the filter step response in the Filter Visualization Tool (FVTool).

Multirate Filters

`[h,t] = stepz(hm)` returns the step response `h` of the multirate filter `hm`. The length of column vector `h` is the length of the impulse response of `hm`. The vector `t` contains the time samples at which `stepz` evaluated the step response. `stepz` returns `h` as a matrix when `hm` is a vector of filters. Each column of the matrix corresponds to one filter in the vector.

`stepz(hm)` displays the step response in the Filter Visualization Tool (FVTool).

Note that the response is computed relative to the rate at which the filter is running. If a sampling frequency is specified, it is assumed that the filter is running at that rate.

Note that the multirate filter delay response is computed relative to the rate at which the filter is running. When you specify `fs` (the sampling rate) as an input argument, `stepz` assumes the filter is running at that rate.

For multistage cascades, `stepz` forms a single-stage multirate filter that is equivalent to the cascade and computes the response relative to the rate at which the equivalent filter is running. `stepz` does not support all multistage cascades. Only cascades for which it is possible to derive an equivalent single-stage filter are allowed for analysis.

As an example, consider a two-stage interpolator where the first stage has an interpolation factor of 2 and the second stage has an interpolation factor of 4. An equivalent single-stage filter with an overall interpolation factor of 8 can be found. `stepz` uses the equivalent filter for the analysis. If you specify a sampling frequency `fs` as an input argument to `stepz`, the function interprets `fs` as the rate at which the equivalent filter is running.

See Also

`freqz`, `impz`

tf2ca

Purpose Transfer function to coupled allpass conversion

Syntax

```
[d1,d2] = tf2ca(b,a)
[d1,d2] = tf2ca(b,a)
[d1,d2,beta] = tf2ca(b,a)
```

Description

`[d1,d2] = tf2ca(b,a)` where `b` is a real, symmetric vector of numerator coefficients and `a` is a real vector of denominator coefficients, corresponding to a stable digital filter, returns real vectors `d1` and `d2` containing the denominator coefficients of the allpass filters *H1(z)* and *H2(z)* such that

$$H(z) = \frac{B(z)}{A(z)} = \frac{1}{2[H1(z) + H2(z)]}$$

representing a coupled allpass decomposition.

`[d1,d2] = tf2ca(b,a)` where `b` is a real, antisymmetric vector of numerator coefficients and `a` is a real vector of denominator coefficients, corresponding to a stable digital filter, returns real vectors `d1` and `d2` containing the denominator coefficients of the allpass filters *H1(z)* and *H2(z)* such that

$$H(z) = \frac{B(z)}{A(z)} = \left(\frac{1}{2}\right)[H1(z) - H2(z)]$$

In some cases, the decomposition is not possible with real *H1(z)* and *H2(z)*. In those cases a generalized coupled allpass decomposition may be possible, whose syntax is

```
[d1,d2,beta] = tf2ca(b,a)
```

to return complex vectors `d1` and `d2` containing the denominator coefficients of the allpass filters *H1(z)* and *H2(z)*, and a complex scalar `beta`, satisfying `|beta| = 1`, such that

$$H(z) = \frac{B(z)}{A(z)} = \left(\frac{1}{2}\right)[\bar{\beta} \bullet H1(z) + \beta \bullet H2(z)]$$

representing the generalized allpass decomposition.

In the above equations, *H1(z)* and *H2(z)* are real or complex allpass IIR filters given by

$$H1(z) = \frac{fliplr((\overline{D1(z)}))}{D1(z)}, H2(1)(z) = \frac{fliplr((\overline{D2(1)(z)}))}{D2(1)(z)}$$

where *D1(z)* and *D2(z)* are polynomials whose coefficients are given by `d1` and `d2`.

Note A coupled allpass decomposition is not always possible. Nevertheless, Butterworth, Chebyshev, and Elliptic IIR filters, among others, can be factored in this manner. For details, refer to *Signal Processing Toolbox User's Guide*.

Examples

```
[b,a]=cheby1(9,.5,.4);
[d1,d2]=tf2ca(b,a); % TF2CA returns denominators of the allpass.
num = 0.5*conv(fliplr(d1),d2)+0.5*conv(fliplr(d2),d1);
den = conv(d1,d2); % Reconstruct numerator and denonimator.
max([max(b-num),max(a-den)]) % Compare original and reconstructed
                             % numerator and denominators.
```

See Also

`ca2tf`, `cl2tf`, `iirpowcomp`, `latc2tf`, `tf2latc`

Purpose

Transfer function to coupled allpass lattice conversion

Syntax

```
[k1,k2] = tf2cl(b,a)
[k1,k2] = tf2cl(b,a)
```

Description

`[k1,k2] = tf2cl(b,a)` where `b` is a real, symmetric vector of numerator coefficients and `a` is a real vector of denominator coefficients, corresponding to a stable digital filter, will perform the coupled allpass decomposition

$$H(z) = \frac{B(z)}{A(z)} = \frac{1}{2[H1(z) + H2(z)]}$$

of a stable IIR filter *H(z)* and convert the allpass transfer functions *H1(z)* and H2(z) to a coupled lattice allpass structure with coefficients given in vectors `k1` and `k2`.

`[k1,k2] = tf2cl(b,a)` where `b` is a real, antisymmetric vector of numerator coefficients and `a` is a real vector of denominator coefficients, corresponding to a stable digital filter, performs the coupled allpass decomposition

$$H(z) = \frac{B(z)}{A(z)} = \left(\frac{1}{2}\right)[H1(z) - H2(z)]$$

of a stable IIR filter *H(z)* and converts the allpass transfer functions *H1(z)* and *H2(z)* to a coupled lattice allpass structure with coefficients given in vectors `k1` and `k2`.

In some cases, the decomposition is not possible with real *H1(z)* and *H2(z)*. In those cases, a generalized coupled allpass decomposition may be possible, using the command syntax

```
[k1,k2,beta] = tf2cl(b,a)
```

to perform the generalized allpass decomposition of a stable IIR filter *H(z)* and convert the complex allpass transfer functions *H1(z)* and *H2(z)* to corresponding lattice allpass filters

$$H(z) = \frac{B(z)}{A(z)} = \left(\frac{1}{2}\right)[\bar{\beta} \bullet H1(z) + \beta \bullet H2(z)]$$

where `beta` is a complex scalar of magnitude equal to 1.

Note Coupled allpass decomposition is not always possible. Nevertheless, Butterworth, Chebyshev, and Elliptic IIR filters, among others, can be factored in this manner. For details, refer to *Signal Processing Toolbox User's Guide*.

Examples

```
[b,a]=cheby1(9,.5,.4);
[k1,k2]=tf2cl(b,a); % Get the reflection coeffs. for the lattices.
[num1,den1]=latc2tf(k1,'allpass'); % Convert each allpass lattice
[num2,den2]=latc2tf(k2,'allpass'); % back to transfer function.
num = 0.5*conv(num1,den2)+0.5*conv(num2,den1);
den = conv(den1,den2); % Reconstruct numerator and denonimator.
max([max(b-num),max(a-den)]) % Compare original and reconstructed
                             % numerator and denominators.
```

See Also

`ca2tf`, `cl2tf`, `iirpowcomp`
`latc2tf`, `tf2ca`, `tf2latc` in Signal Processing Toolbox

window

Purpose

Design an FIR filter using the windowed impulse response method

Syntax

```
h = window(d,fcnhndl,fcnarg)
h = window(d, win)
```

description

`h = window(d,fcnhndl,fcnarg)` designs an FIR filter using the specifications in filter design object `d`. Depending on the specification type of `d`, the returned filter is either a single-rate digital filter—a `dfilt`, or a multirate digital filter—an `mfilt`.

`fcnhndl` is a handle to a filter design function that returns a window vector, such as the `hamming` or `blackman` functions. `fcnarg` is an optional argument that returns a window. You pass the function to `window`. Refer to example 1 below to see the function argument used to design the filter.

`h = window(d,win)` designs a filter using the vector you supply in `win`. The length of vector `win` must be the same as the impulse response of the filter, which is equal to the filter order plus one. Example 2 shows this being done.

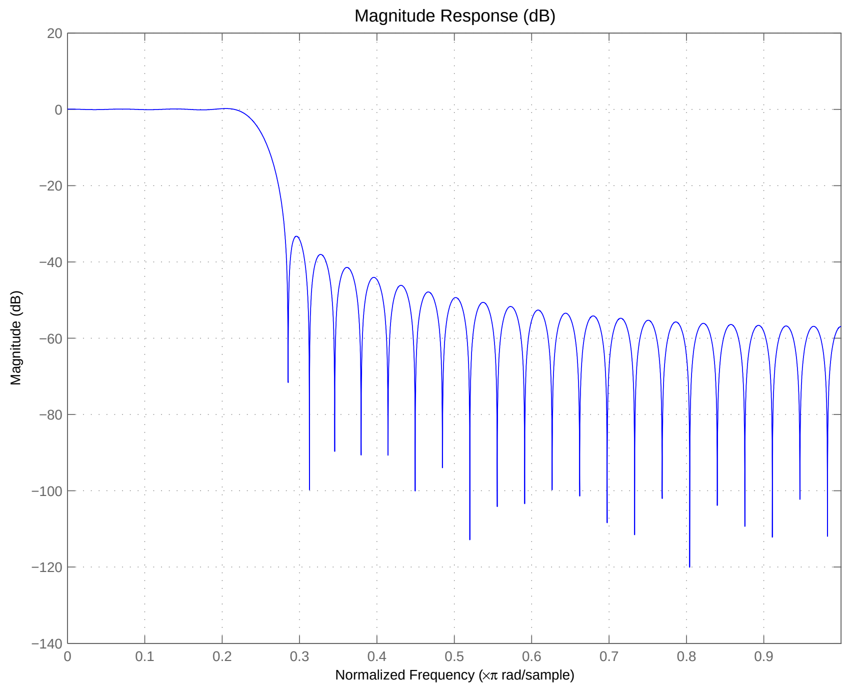
Examples

These examples design filters using the two design techniques of specifying a function handle or passing a window vector as an input argument.

Example 1

Use a function handle and optional input arguments to design a multirate filter. We use a function handle to the function Kaiser to provide the window. Since this example creates a decimating filter design object, `window` returns a multirate filter.

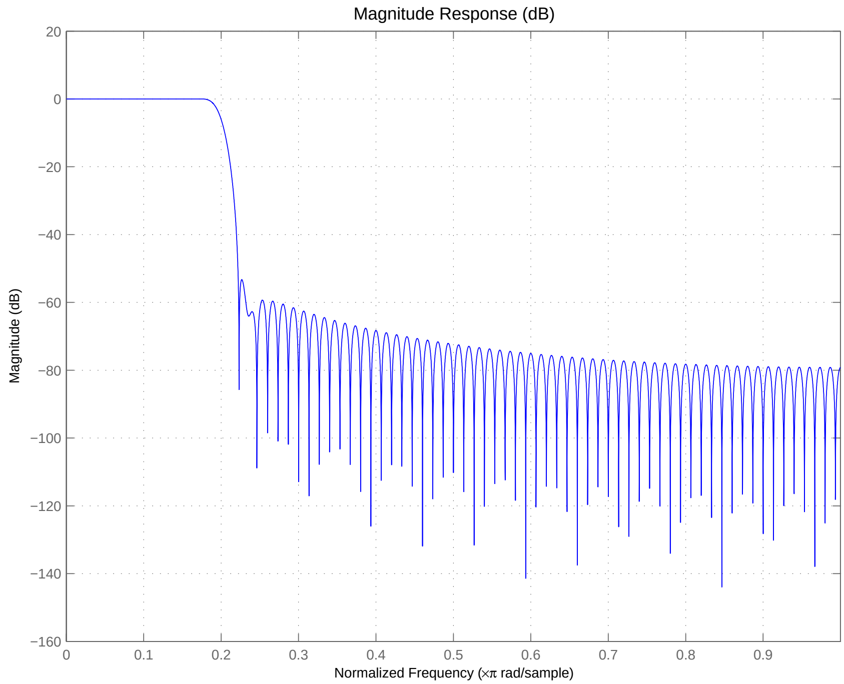
```
d = fdesign.decim(4,'pl',14);
hm = window(d,@kaiser,2.5);
fvtool(hm)
```



Example 2

Use a window vector provided by the `hamming` window design function. For this example, the design object is a Nyquist filter, thus window returns `hd` as a discrete-time filter.

```
d = fdesign.nyquist(5,'n',150);
hd = window(d,hamming(151));
fvtool(hd)
```



See Also

`firls`, `kaiserwin`

Purpose

Return the zero-phase response for adaptive, discrete-time, and multirate filters

Syntax

```
zerophase(ha)
[hr,w] = zerophase(ha,n)
[hr,w] = zerophase(...,f)
zerophase(hd)
[hr,w] = zerophase(hd,n)
[hr,w] = zerophase(...,f)
zerophase(hm)
[hr,w] = zerophase(hm,n)
[hr,w] = zerophase(...,f)
[hr,w] = zerophase(...,fs)
```

Description

The next sections describe common `zerophase` operation with adaptive, discrete-time, and multirate filters. For more input options, refer to `zerophase` in the Signal Processing Toolbox.

Adaptive Filters

For adaptive filters, `zerophase` returns the instantaneous zero-phase response based on the current filter coefficients.

`zerophase(ha)` displays the zero-phase response of `ha` in the Filter Visualization Tool (FVTool).

`[hr,w] = zerophase(ha,n)` returns length `n` vectors `hr` and `w` containing the instantaneous zero-phase response of the adaptive filter `ha`, and the frequencies in radians at which `zerophase` evaluated the response. The zero-phase response is evaluated at `n` points equally spaced around the upper half of the unit circle. For an FIR filter where `n` is a power of two, the computation is done faster using FFTs. If `n` is not specified, it defaults to 8192.

`[hr,w] = zerophase(ha)` returns a matrix `hr` if `ha` is a vector of filters. Each column of the matrix corresponds to each filter in the vector. If you provide a row vector of frequency points `f` as an input argument, each row of `hr` corresponds to one filter in the vector.

Discrete-Time Filters

`zerophase(hd)` displays the zero-phase response of `hd` in the Filter Visualization Tool (FVTool).

`[hr,w] = zerophase(hd,n)` returns length `n` vectors `hr` and `w` containing the instantaneous zero-phase response of the adaptive filter `hd`, and the frequencies in radians at which `zerophase` evaluated the response. The zero-phase response is evaluated at `n` points equally spaced around the upper half of the unit circle. For an FIR filter where `n` is a power of two, the computation is done faster using FFTs. If `n` is not specified, it defaults to 8192.

`[hr,w] = zerophase(hd)` returns a matrix `hr` if `hd` is a vector of filters. Each column of the matrix corresponds to each filter in the vector. If you provide a row vector of frequency points `f` as an input argument, each row of `hr` corresponds to one filter in the vector.

Multirate Filters

`zerophase(hm)` displays the zero-phase response of `hd` in the Filter Visualization Tool (FVTool).

`[hr,w] = zerophase(hm,n)` returns length `n` vectors `hr` and `w` containing the instantaneous zero-phase response of the adaptive filter `hm`, and the frequencies in radians at which `zerophase` evaluated the response. The zero-phase response is evaluated at `n` points equally spaced around the upper half of the unit circle. For an FIR filter where `n` is a power of two, the computation is done faster using FFTs. If `n` is not specified, it defaults to 8192.

`[hr,w] = zerophase(hm)` returns a matrix `hr` if `hm` is a vector of filters. Each column of the matrix corresponds to each filter in the vector. If you provide a row vector of frequency points `f` as an input argument, each row of `hr` corresponds to one filter in the vector.

Note that the response is computed relative to the rate at which the filter is running. If a sampling frequency is specified, it is assumed that the filter is running at that rate.

Note that the multirate filter delay response is computed relative to the rate at which the filter is running. When you specify `fs` (the sampling rate) as an input argument, `zerophase` assumes the filter is running at that rate.

For multistage cascades, `zerophase` forms a single-stage multirate filter that is equivalent to the cascade and computes the response relative to the rate at which the equivalent filter is running. `zerophase` does not support all multistage cascades. Only cascades for which it is possible to derive an equivalent single-stage filter are allowed for analysis.

As an example, consider a two-stage interpolator where the first stage has an interpolation factor of 2 and the second stage has an interpolation factor of 4. An equivalent single-stage filter with an overall interpolation factor of 8 can be found. `zerophase` uses the equivalent filter for the analysis. If a sampling frequency `fs` is specified as an input argument to `zerophase`, the function interprets `fs` as the rate at which the equivalent filter is running.

See Also

`freqz`, `fvtool`, `grpdelay`, `impz`, `mfilt`, `phasez`, `zerophase`, `zplane`

zpkbpc2bpc

Purpose

Zero-pole-gain complex bandpass frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpkbpc2bpc(Z,P,K,Wo,Wt)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpkbpc2bpc(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the complex bandpass prototype by applying a first-order complex bandpass to complex bandpass frequency transformation.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The original lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

This transformation effectively places two features of an original filter, located at frequencies W_{o1} and W_{o2}, at the required target frequency locations, W_{t1}, and W_{t2} respectively. It is assumed that W_{t2} is greater than W_{t1} . In most of the cases the features selected for the transformation are the band edges of the filter passbands. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

This transformation can also be used for transforming other types of filters; e.g., complex notch filters or resonators can be repositioned at two distinct desired frequencies at any place around the unit circle; e.g., in the adaptive system.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
```

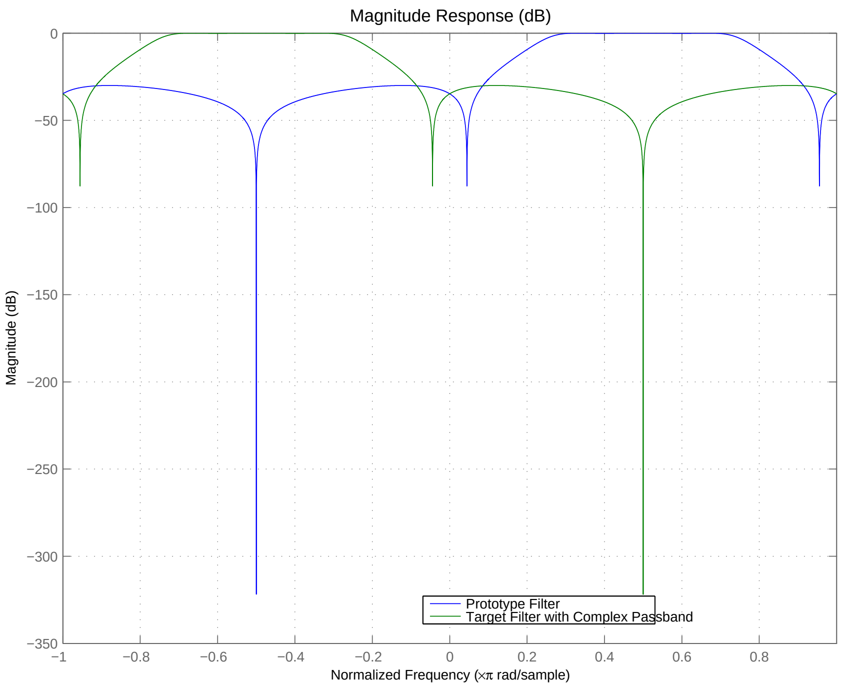
Create a complex passband from 0.25 to 0.75:

```
[b, a] = iirlp2bpc(b,a,0.5,[0.25,0.75]);
z = roots(b);
p = roots(a);
k = b(1);
[z2,p2,k2] = zpkbpc2bpc(z, p, k, [0.25, 0.75], [-0.75, -0.25]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

Comparing the filters in FVTool shows the example results. Use the features in FVTool to check the filter coefficients, or other filter analyses.



Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

See Also

`zpkftransf`, `allpassbpc2bpc`, `iirbpc2bpc`

Purpose Zero-pole-gain frequency transformation of the digital filter

Syntax

```
[Z2,P2,K2] = zpkftransf(Z,P,K,AllpassNum,AllpassDen)
```

Description `[Z2,P2,K2] = zpkftransf(Z,P,K,AllpassNum,AllpassDen)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the transformed lowpass digital filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`. If `AllpassDen` is not specified it will default to 1. If neither `AllpassNum` nor `AllpassDen` is specified, then the function returns the input filter.

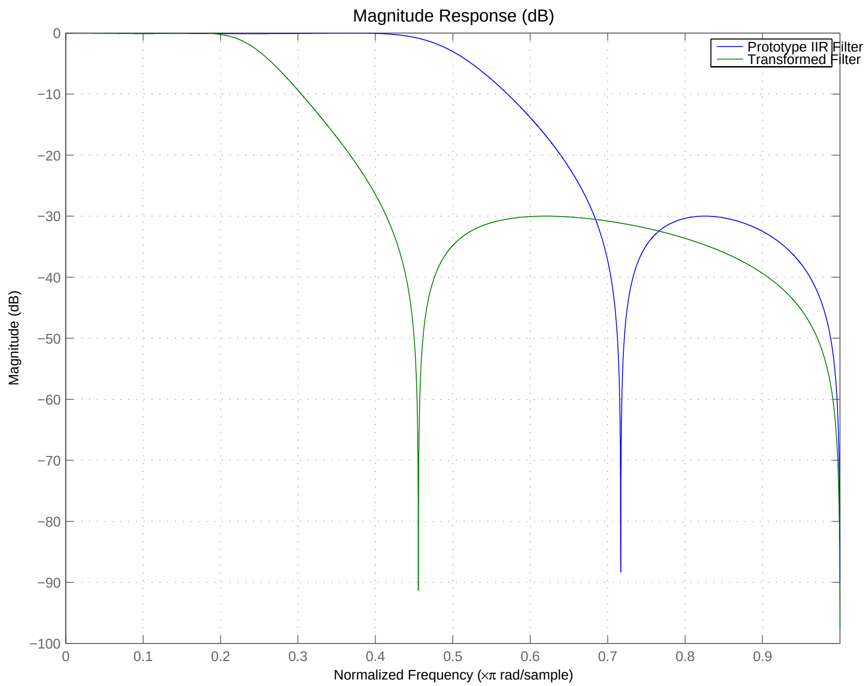
Examples Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
[AlpNum, AlpDen] = allpasslp2lp(0.5, 0.25);
[z2, p2, k2] = zpkftransf(roots(b),roots(a),b(1),AlpNum,AlpDen);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

After transforming the filter, you get the response shown in the figure, where the passband has been shifted towards zero.



Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`FTFNum`
Numerator of the mapping filter

`FTFDen`
Denominator of the mapping filter

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

See Also

`iirftransf`

zpklp2bp

Purpose Zero-pole-gain lowpass to bandpass frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2bp(Z,P,K,Wo,Wt)
```

Description `[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2bp(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying a second-order real lowpass to real bandpass frequency mapping.

It also returns the numerator, `AllpassNum`, and the denominator `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

This transformation effectively places one feature of an original filter, located at frequency $-W_o$, at the required target frequency location, W_{t1}, and the second feature, originally at $+W_o$, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. This transformation implements the “DC Mobility,” which means that the Nyquist feature stays at Nyquist, but the DC feature moves to a location dependent on the selection of W_t.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandpass transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Real lowpass to bandpass transformation can also be used for transforming other types of filters; e.g., real notch filters or resonators can be easily doubled and positioned at two distinct, desired frequencies.

Examples Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
```

```
[z2,p2,k2] = zpklp2bp(z, p, k, 0.5, [0.2 0.3]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`zpkftransf`, `allpasslp2bp`, `iirlp2bp`

References

[1] Constantinides, A.G., "Spectral transformations for digital filters," *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, “Prototype reference transfer function parameters in the discrete-time frequency transformations,” *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, “Closed-form solutions for discrete-time elliptic transfer functions,” *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

[4] Constantinides, A.G., “Design of bandpass digital filters,” *IEEE Proceedings*, vol. 1, pp. 1129-1231, June 1969.

Purpose Zero-pole-gain lowpass to complex bandpass frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2bpc(Z,P,K,Wo,Wt)
```

Description `[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2bpc(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying a first-order real lowpass to complex bandpass frequency transformation.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

This transformation effectively places one feature of an original filter, located at frequency $-W_o$, at the required target frequency location, W_{t1}, and the second feature, originally at $+W_o$, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandpass transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Lowpass to bandpass transformation can also be used for transforming other types of filters; e.g., real notch filters or resonators can be doubled and positioned at two distinct desired frequencies at any place around the unit circle forming a pair of complex notches/resonators. This transformation can be used for designing bandpass filters for radio receivers from the high-quality prototype lowpass filter.

Examples Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
```

```
k = b(1);
[z2,p2,k2] = zpklp2bpc(z, p, k, 0.5, [0.2 0.3]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter. It should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`Wt`
Desired frequency locations in the transformed target filter. They should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also

`zpkftransf`, `allpasslp2bpc`, `iirlp2bpc`

Purpose

Zero-pole-gain lowpass to bandstop frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2bs(Z,P,K,Wo,Wt)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2bs(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying a second-order real lowpass to real bandstop frequency mapping.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

This transformation effectively places one feature of an original filter, located at frequency -W_o, at the required target frequency location, W_{t1}, and the second feature, originally at +W_o, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. This transformation implements the "Nyquist Mobility," which means that the DC feature stays at DC, but the Nyquist feature moves to a location dependent on the selection of W_o and W_ts.

Relative positions of other features of an original filter change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. After the transformation feature F_2 will precede F_1 in the target filter. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandstop transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
[z2,p2,k2] = zpklp2bs(z, p, k, 0.5, [0.2 0.3]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`zpkftransf`, `allpasslp2bs`, `iirlp2bs`

References

[1] Constantinides, A.G., “Spectral transformations for digital filters,” *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, “Prototype reference transfer function parameters in the discrete-time frequency transformations,” *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, “Closed-form solutions for discrete-time elliptic transfer functions,” *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

[4] Constantinides, A.G., “Design of bandpass digital filters,” *IEEE Proceedings*, vol. 1, pp. 1129-1231, June 1969.

zpklp2bsc

Purpose

Zero-pole-gain lowpass to complex bandstop frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2bsc(Z,P,K,Wo,Wt)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2bsc(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying a first-order real lowpass to complex bandstop frequency transformation.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

This transformation effectively places one feature of an original filter, located at frequency $-W_o$, at the required target frequency location, W_{t1}, and the second feature, originally at $+W_o$, at the new location, W_{t2}. It is assumed that W_{t2} is greater than W_{t1}. Additionally the transformation swaps passbands with stopbands in the target filter.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to bandstop transformation is not restricted only to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Lowpass to bandpass transformation can also be used for transforming other types of filters; e.g., real notch filters or resonators can be doubled and positioned at two distinct desired frequencies at any place around the unit circle forming a pair of complex notches/resonators.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
```

```
[z2,p2,k2] = zpklp2bsc(z, p, k, 0.5, [0.2, 0.3]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter. It should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wt`
Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also

`zpkftransf`, `allpasslp2bsc`, `iirlp2bsc`

zpklp2hp

Purpose

Zero-pole-gain lowpass to highpass frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2hp(Z,P,K,Wo,Wt)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2hp(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying a first-order real lowpass to real highpass frequency mapping. This transformation effectively places one feature of an original filter, located at frequency W_o, at the required target frequency location, W_t, at the same time rotating the whole frequency response by half of the sampling frequency. Result is that the DC and Nyquist features swap places.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and the gain factor, `K`.

Relative positions of other features of an original filter change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. After the transformation feature F_2 will precede F_1 in the target filter. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to highpass transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, or the deep minimum in the stopband, or other ones.

Lowpass to highpass transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can change their position in a simple way without designing them again.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
[z2,p2,k2] = zpklp2hp(z, p, k, 0.5, 0.25);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`zpkftransf`, `allpasslp2hp`, `iirlp2hp`

References

[1] Constantinides, A.G., “Spectral transformations for digital filters,” *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, “Prototype reference transfer function parameters in the discrete-time frequency transformations,” *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, “Closed-form solutions for discrete-time elliptic transfer functions,” *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

[4] Constantinides, A.G., “Frequency transformations for digital filters,” *Electronics Letters*, vol. 3, no. 11, pp. 487-489, November 1967.

zpklp2lp

Purpose

Zero-pole-gain lowpass to lowpass frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2lp(Z,P,K,Wo,Wt)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2lp(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying a first-order real lowpass to real lowpass frequency mapping. This transformation effectively places one feature of an original filter, located at frequency W_o, at the required target frequency location, W_t.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the lowpass to lowpass transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Lowpass to lowpass transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can change their position in a simple way without designing them again.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

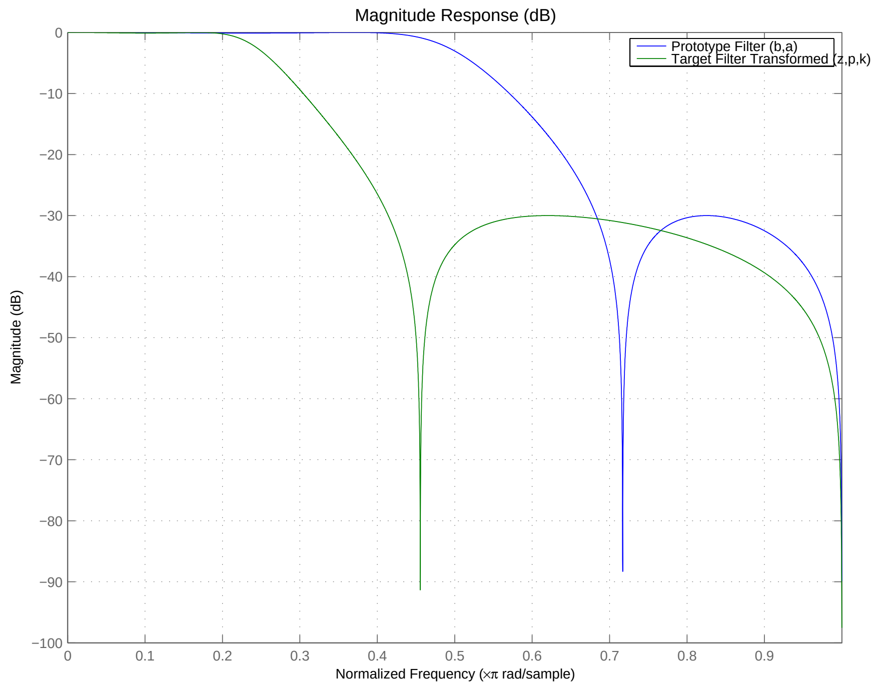
```
[b, a] = ellip(3, 0.1, 30, 0.409);
z = roots(b);
p = roots(a);
k = b(1);
[z2,p2,k2] = zpklp2lp(z, p, k, 0.5, 0.25);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

zpklp2lp

Using zpklp2lp creates the desired half band IIR filter with the transformed features that you specify in the transformation function. This figure shows the results.



Arguments

Z
Zeros of the prototype lowpass filter

P
Poles of the prototype lowpass filter

K
Gain factor of the prototype lowpass filter

Wo
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`zpkftransf`, `allpasslp2lp`, `iirlp2lp`

References

[1] Constantinides, A.G., “Spectral transformations for digital filters,” *IEE Proceedings*, vol. 117, no. 8, pp. 1585-1590, August 1970.

[2] Nowrouzian, B. and A.G. Constantinides, “Prototype reference transfer function parameters in the discrete-time frequency transformations,” *Proceedings 33rd Midwest Symposium on Circuits and Systems*, Calgary, Canada, vol. 2, pp. 1078-1082, August 1990.

[3] Nowrouzian, B. and L.T. Bruton, “Closed-form solutions for discrete-time elliptic transfer functions,” *Proceedings of the 35th Midwest Symposium on Circuits and Systems*, vol. 2, pp. 784-787, 1992.

[4] Constantinides, A.G., “Frequency transformations for digital filters,” *Electronics Letters*, vol. 3, no. 11, pp. 487-489, November 1967.

zpklp2mb

Purpose

Zero-pole-gain lowpass to M-band frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2mb(Z,P,K,Wo,Wt)
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2mb(Z,P,K,Wo,Wt,Pass)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2mb(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying an `M`th-order real lowpass to real multibandpass frequency mapping. By default the DC feature is kept at its original location.

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2mb(Z,P,K,Wo,Wt,Pass)` allows you to specify an additional parameter, `Pass`, which chooses between using the "DC Mobility" and the "Nyquist Mobility". In the first case the Nyquist feature stays at its original location and the DC feature is free to move. In the second case the DC feature is kept at an original frequency and the Nyquist feature is allowed to move.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

This transformation effectively places one feature of an original filter, located at frequency W_o, at the required target frequency locations, W_{t1},...,W_{tM}.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

Examples

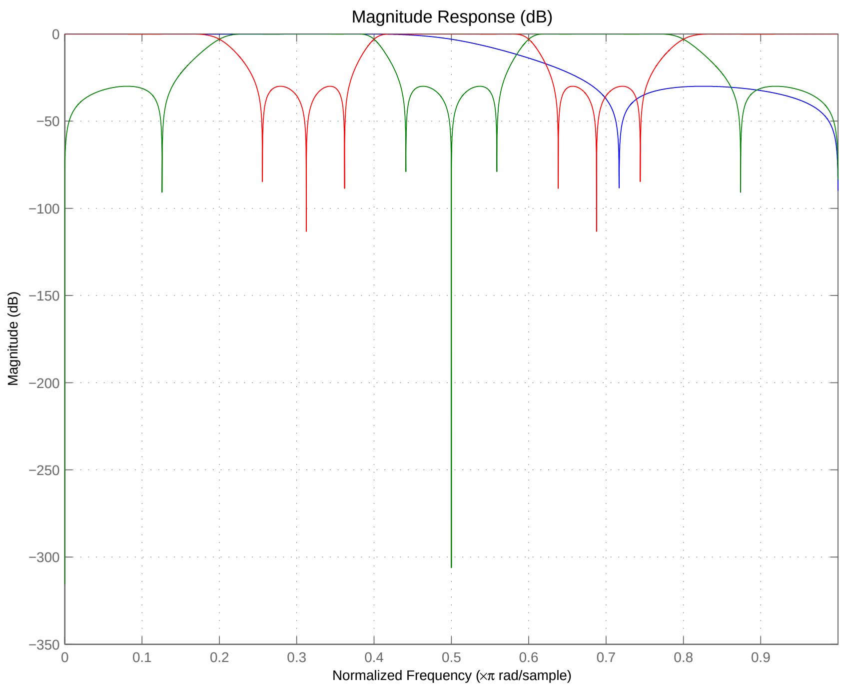
Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
[z1,p1,k1] = zpklp2mb(z, p, k, 0.5, [2 4 6 8]/10, 'pass');
[z2,p2,k2] = zpklp2mb(z, p, k, 0.5, [2 4 6 8]/10, 'stop');
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k1*poly(z1), poly(p1), k2*poly(z2), poly(p2));
```

The resulting multiband filter that replicates features from the prototype appears in the figure shown. Note the accuracy of the replication process.



Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Pass`
Choice (`'pass'`/`'stop'`) of passband/stopband at DC, `` `pass' `` being the default

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`zpkftransf`, `allpasslp2mb`, `iirlp2mb`

References

[1] Franchitti, J.C., "All-pass filter interpolation and frequency transformation problems," MSc Thesis, Dept. of Electrical and Computer Engineering, University of Colorado, 1985.

[2] Feyh, G., J.C. Franchitti and C.T. Mullis, "All-pass filter interpolation and frequency transformation problem," *Proceedings 20th Asilomar Conference on*

Signals, Systems and Computers, Pacific Grove, California, pp. 164-168, November 1986.

[3] Mullis, C.T. and R.A. Roberts, *Digital Signal Processing*, section 6.7, Reading, Massachusetts, Addison-Wesley, 1987.

[4] Feyh, G., W.B. Jones and C.T. Mullis, *An extension of the Schur Algorithm for frequency transformations, Linear Circuits, Systems and Signal Processing: Theory and Application*, C. J. Byrnes et al Eds, Amsterdam: Elsevier, 1988.

zpklp2mbc

Purpose

Zero-pole-gain lowpass to complex M-band frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklpmbc(Z,P,K,Wo,Wt)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpklpmbc(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying an `M`th-order real lowpass to complex multibandpass frequency transformation.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

This transformation effectively places one feature of an original filter, located at frequency W_o, at the required target frequency locations, W_{t1},...,W_{tM}.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature, for example, the stopband edge, the DC, the deep minimum in the stopband, or other ones.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

This transformation can also be used for transforming other types of filters; e.g., to replicate notch filters and resonators at any required location.

Examples

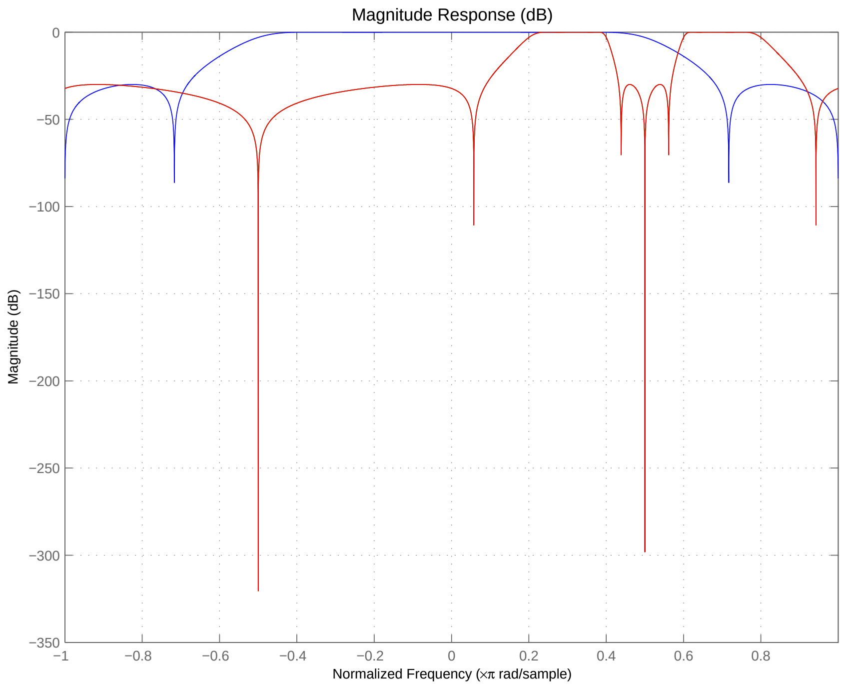
Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
[z1,p1,k1] = zpklp2mbc(z, p, k, 0.5, [2 4 6 8]/10);
[z2,p2,k2] = zpklp2mbc(z, p, k, 0.5, [2 4 6 8]/10);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k1*poly(z1), poly(p1), k2*poly(z2), poly(p2));
```

You could review the coefficients to compare the filters, but the graphical comparison shown here is quicker and easier.



However, looking at the coefficients in FVTool shows the complex nature desired.

Arguments

Z
Zeros of the prototype lowpass filter

P
Poles of the prototype lowpass filter

K
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter. It should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wt`
Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

See Also

`zpkftransf`, `allpasslp2mbc`, `iirlp2mbc`

Purpose Zero-pole-gain lowpass to complex N-point frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2xc(Z,P,K,Wo,Wt)
```

Description [Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2xc(Z,P,K,Wo,Wt) returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying an Nth-order real lowpass to complex multipoint frequency transformation.

It also returns the numerator, AllpassNum, and the denominator, AllpassDen, of the allpass mapping filter. The prototype lowpass filter is given with zeros, Z, poles, P, and gain factor, K.

Parameter N also specifies the number of replicas of the prototype filter created around the unit circle after the transformation. This transformation effectively places N features of an original filter, located at frequencies W_{o1},...,W_{oN}, at the required target frequency locations, W_{t1},...,W_{tM}.

Relative positions of other features of an original filter are the same in the target filter for the Nyquist mobility and are reversed for the DC mobility. For the Nyquist mobility this means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation. For DC mobility feature F_2 will precede F_1 after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones. The only condition is that the features must be selected in such a way that when creating N bands around the unit circle, there will be no band overlap.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

Examples Design a prototype real IIR halfband filter using a standard elliptic approach:

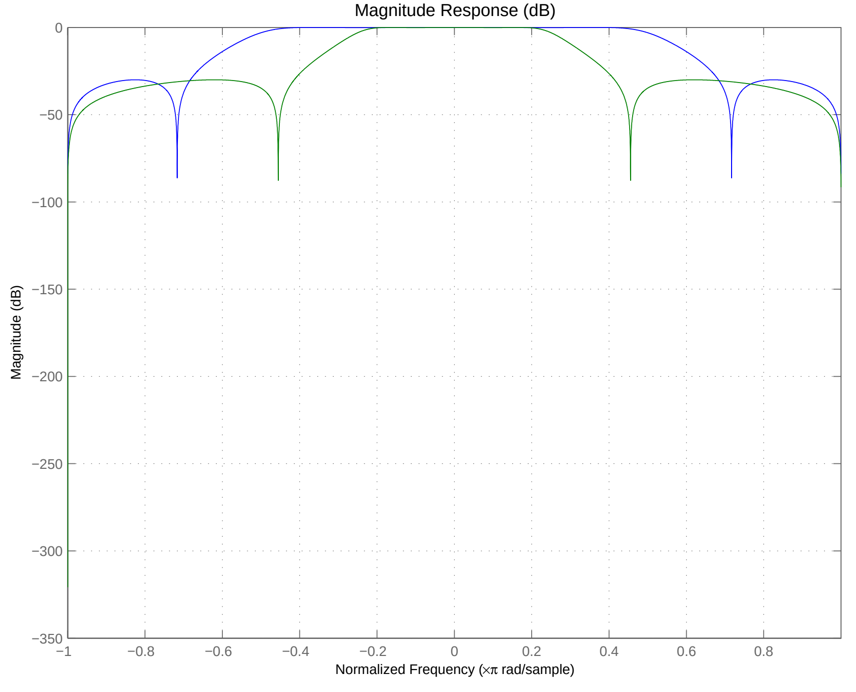
```
[b, a] = ellip(3,0.1,30,0.409);
```

```
z = roots(b);
p = roots(a);
k = b(1);
[z2,p2,k2] = zpklp2xc(z, p, k, [-0.5 0.5], [-0.25 0.25]);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

Plotting the filters on the same axes lets you compare the results graphically, shown here.



Arguments

z

Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`Wo`
Frequency values to be transformed from the prototype filter. They should be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

`Wt`
Desired frequency locations in the transformed target filter. They should be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`

Denominator of the mapping filter

See Also

`zpkftransf`, `allpasslp2xc`, `iirlp2xc`

zpklp2xn

Purpose

Zero-pole-gain lowpass to N-point frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2xn(Z,P,K,Wo,Wt)
[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2xn(Z,P,K,Wo,Wt,Pass)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2xn(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying an `N`th-order real lowpass to real multipoint frequency transformation, where `N` is the number of features being mapped. By default the DC feature is kept at its original location.

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpklp2xn(Z,P,K,Wo,Wt,Pass)` allows you to specify an additional parameter, `Pass`, which chooses between using the "DC Mobility" and the "Nyquist Mobility". In the first case the Nyquist feature stays at its original location and the DC feature is free to move. In the second case the DC feature is kept at an original frequency and the Nyquist feature is allowed to move.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

Parameter `N` also specifies the number of replicas of the prototype filter created around the unit circle after the transformation. This transformation effectively places `N` features of an original filter, located at frequencies $W_{o1},\ldots,W_{oN}$, at the required target frequency locations, $W_{t1},\ldots,W_{tM}$.

Relative positions of other features of an original filter are the same in the target filter for the Nyquist mobility and are reversed for the DC mobility. For the Nyquist mobility this means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation. For DC mobility feature F_2 will precede F_1 after the transformation.

Choice of the feature subject to this transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones. The only condition is that the features must be selected in such a way that when creating N bands around the unit circle, there will be no band overlap.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can be easily replicated at a number of required frequency locations. A good application would be an adaptive tone cancellation circuit reacting to the changing number and location of tones.

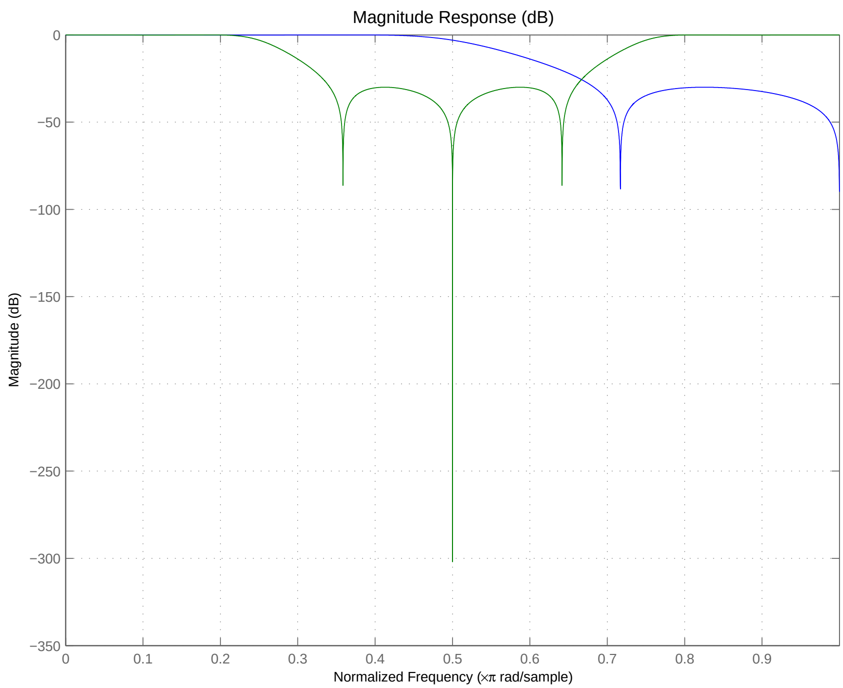
Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
[z2,p2,k2] = zpklp2xn(z, p, k, [-0.5 0.5], [-0.25 0.25], 'pass');
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```



As demonstrated by the figure, the target filter has the desired response shape and values replicated from the prototype.

Arguments

`Z`

Zeros of the prototype lowpass filter

`P`

Poles of the prototype lowpass filter

`K`

Gain factor of the prototype lowpass filter

`Wo`

Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Pass`
Choice (`'pass'`/`'stop'`) of passband/stopband at DC, `` `pass' `` being the default

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassDen`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also

`zpkftransf`, `allpasslp2xn`, `iirlp2xn`

References

[1] Cain, G.D., A. Krukowski and I. Kale, "High Order Transformations for Flexible IIR Filter Design," *VII European Signal Processing Conference (EUSIPCO'94)*, vol. 3, pp. 1582-1585, Edinburgh, United Kingdom, September 1994.

[2] Krukowski, A., G.D. Cain and I. Kale, "Custom designed high-order frequency transformations for IIR filters," *38th Midwest Symposium on Circuits and Systems (MWSCAS'95)*, Rio de Janeiro, Brazil, August 1995.

zpkrateup

Purpose Zero-pole-gain complex bandpass frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpkrateup(Z,P,K,N)
```

Description `[Z2,P2,K2,AllpassNum,AllpassDen] = zpkrateup(Z,P,K,N)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter being transformed from any prototype by applying an `N`th-order rateup frequency transformation, where `N` is the upsample ratio. Transformation creates `N` equal replicas of the prototype filter frequency response.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The original lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Examples Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
```

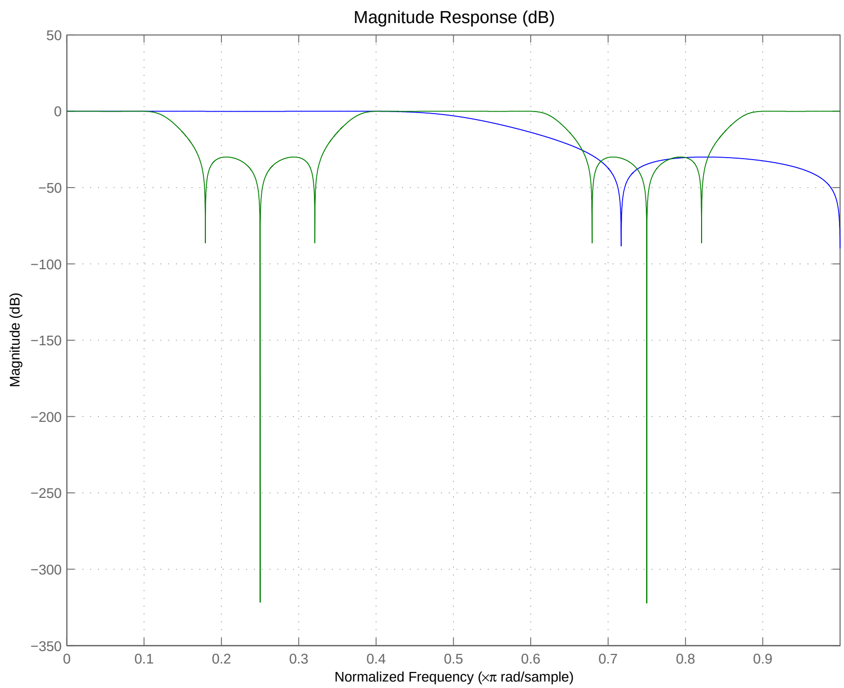
Upsample the prototype filter four times:

```
[z2,p2,k2] = zpkrateup(z, p, k, 4);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

Applying the upsample process creates a bandpass filter, as shown here.



Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`N`
Integer upsampling ratio

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

See Also

`zpkrateup`, `allpassrateup`, `iirrateup`

Purpose

Zero-pole-gain real shift frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpkshift(Z,P,K,Wo,Wt)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpkshift(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying a second-order real shift frequency mapping.

It also returns the numerator, `AllpassNum`, and the denominator of the allpass mapping filter, `AllpassDen`. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and gain factor, `K`.

This transformation places one selected feature of an original filter, located at frequency W_o, at the required target frequency location, W_t. This transformation implements the "DC Mobility," which means that the Nyquist feature stays at Nyquist, but the DC feature moves to a location dependent on the selection of W_o and W_t.

Relative positions of other features of an original filter do not change in the target filter. This means that it is possible to select two features of an original filter, F_1 and F_2, with F_1 preceding F_2. Feature F_1 will still precede F_2 after the transformation. However, the distance between F_1 and F_2 will not be the same before and after the transformation.

Choice of the feature subject to the real shift transformation is not restricted to the cutoff frequency of an original lowpass filter. In general it is possible to select any feature; e.g., the stopband edge, the DC, the deep minimum in the stopband, or other ones.

This transformation can also be used for transforming other types of filters; e.g., notch filters or resonators can change their position in a simple way without the need to design them again.

Examples

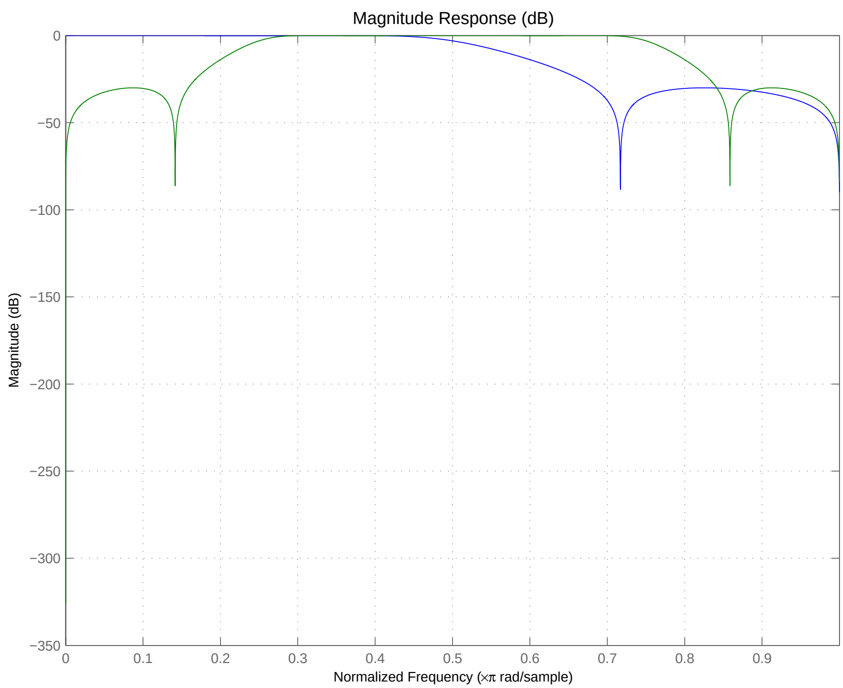
Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
[z2,p2,k2] = zpkshift(z, p, k, 0.5, 0.25);
```

Verify the result by comparing the prototype filter with the target filter:

```
fvtool(b, a, k2*poly(z2), poly(p2));
```

It is clear from the following figure that the shift process has taken the response value at 0.5 in the prototype and replicated it in the target at 0.25, as specified.



Arguments

Z
Zeros of the prototype lowpass filter

P
Poles of the prototype lowpass filter

K
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

K2

Gain factor of the target filter

`AllpassNum`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between 0 and 1, with 1 corresponding to half the sample rate.

See Also `zpkftransf`, `allpassshift`, `iirshift`

zpkshiftc

Purpose Zero-pole-gain complex shift frequency transformation

Syntax

```
[Z2,P2,K2,AllpassNum,AllpassDen] = zpkshiftc(Z,P,K,Wo,Wt)
```

Description

`[Z2,P2,K2,AllpassNum,AllpassDen] = zpkshiftc(Z,P,K,Wo,Wt)` returns zeros, Z_2, poles, P_2, and gain factor, K_2, of the target filter transformed from the real lowpass prototype by applying a first-order complex frequency shift transformation. This transformation rotates all the features of an original filter by the same amount specified by the location of the selected feature of the prototype filter, originally at W_o, placed at W_t in the target filter.

It also returns the numerator, `AllpassNum`, and the denominator, `AllpassDen`, of the allpass mapping filter. The prototype lowpass filter is given with zeros, `Z`, poles, `P`, and the gain factor, `K`.

`[Num,Den,AllpassNum,AllpassDen] = zpkshiftc(Z,P,K,0,0.5)` performs the Hilbert transformation, i.e. a 90 degree counterclockwise rotation of an original filter in the frequency domain.

`[Num,Den,AllpassNum,AllpassDen] = zpkshiftc(Z,P,K,0,-0.5)` performs the inverse Hilbert transformation, i.e. a 90 degree clockwise rotation of an original filter in the frequency domain.

Examples

Design a prototype real IIR halfband filter using a standard elliptic approach:

```
[b, a] = ellip(3,0.1,30,0.409);
z = roots(b);
p = roots(a);
k = b(1);
```

Example 1: Rotation by -0.25:

```
[z2,p2,k2] = zpkshiftc(z, p, k, 0.5, 0.25);
fvtool(b, a, k2*poly(z2), poly(p2));
```

Example 2: Hilbert transform:

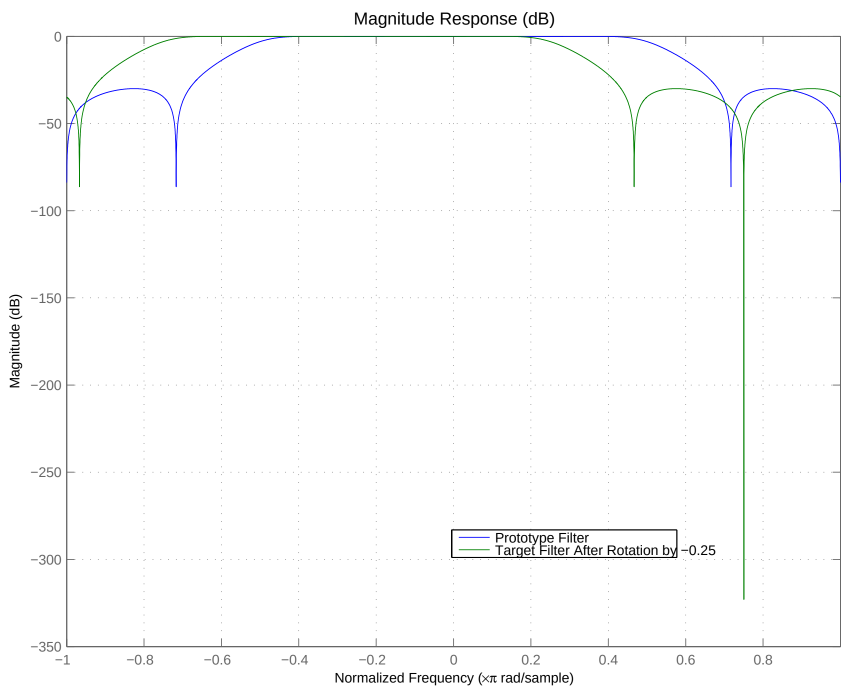
```
[z2,p2,k2] = zpkshiftc(z, p, k, 0, 0.5);
fvtool(b, a, k2*poly(z2), poly(p2));
```

Example 3: Inverse Hilbert transform:

```
[z2,p2,k2] = zpkshiftc(z, p, k, 0, -0.5);
fvtool(b, a, k2*poly(z2), poly(p2));
```

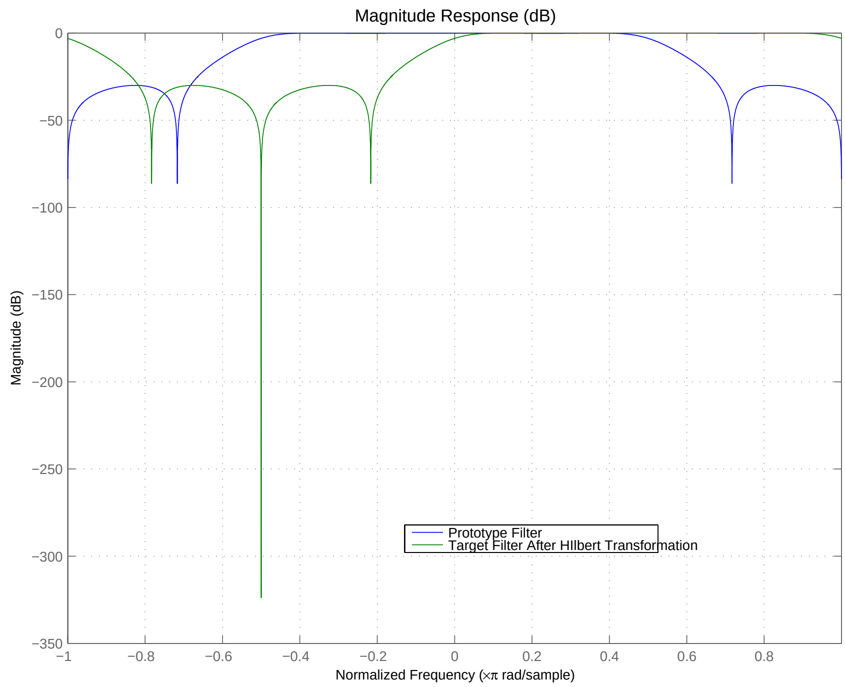
Result of Example 1

After performing the rotation, the resulting filter shows the features desired.



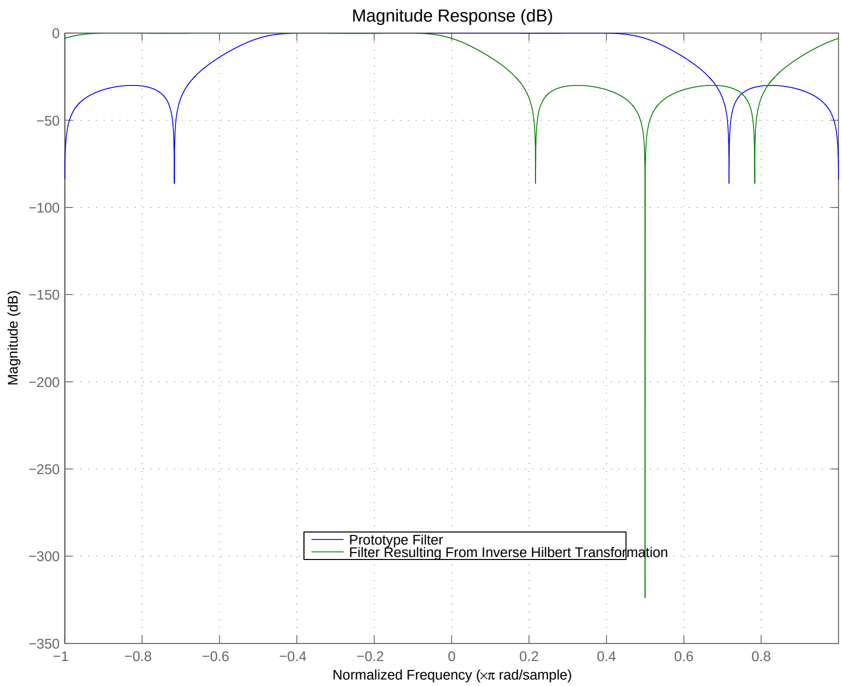
Result of Example 2

Similar to the first example, performing the Hilbert transformation generates the desired target filter, shown here.



Result of Example 3

Finally, using the inverse Hilbert transformation creates yet a third filter, as the figure shows.



Arguments

`Z`
Zeros of the prototype lowpass filter

`P`
Poles of the prototype lowpass filter

`K`
Gain factor of the prototype lowpass filter

`Wo`
Frequency value to be transformed from the prototype filter

`Wt`
Desired frequency location in the transformed target filter

`Z2`
Zeros of the target filter

`P2`
Poles of the target filter

`K2`
Gain factor of the target filter

`AllpassDen`
Numerator of the mapping filter

`AllpassDen`
Denominator of the mapping filter

Frequencies must be normalized to be between -1 and 1, with 1 corresponding to half the sample rate.

See Also

`zpkftransf`, `allpassshiftc`, `iirshiftc`

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Purpose Compute a zero-pole plot for quantized filters

Syntax

```
zplane(Hq)
zplane(Hq,'plotoption')
zplane(Hq,'plotoption','plotoption2')
[zq,pq,kq] = zplane(Hq)
[zq,pq,kq,zr,pr,kr] = zplane(Hq)
```

Description This function displays the poles and zeros of quantized filters, as well as the poles and zeros of the associated unquantized reference filter.

`zplane(Hq)` plots the zeros and poles of a quantized filter `Hq` in the current figure window. The poles and zeros of the quantized and unquantized filters are plotted by default. The symbol `o` represents a zero of the unquantized reference filter, and the symbol `x` represents a pole of that filter. The symbols □ and + are used to plot the zeros and poles of the quantized filter `Hq`. The plot includes the unit circle for reference.

`zplane(Hq,'`*`plotoption`*`')` plots the poles and zeros associated with the quantized filter `Hq` according to one specified plot option. The string `'`*`plotoption`*`'` can be either of the following reference filter display options:

- **`'on'`** to display the poles and zeros of both the quantized filter and the associated reference filter (default)
- **`'off'`** to display the poles and zeros of only the quantized filter

`zplane(Hq,'`*`plotoption`*`','`*`plotoption2`*`')` plots the poles and zeros associated with the quantized filter `Hq` according to two specified plot options. The string `'`*`plotoption`*`'` can be selected from the reference filter display options listed in the previous syntax. The string `'`*`plotoption2`*`'` can be selected from the section-by-section plotting style options described below:

- **`'individual'`** to display the poles and zeros of each section of the filter in a separate figure window
- **`'overlay'`** to display the poles and zeros of all sections of the filter on the same plot
- **`'tile'`** to display the poles and zeros of each section of the filter in a separate plot in the same figure window

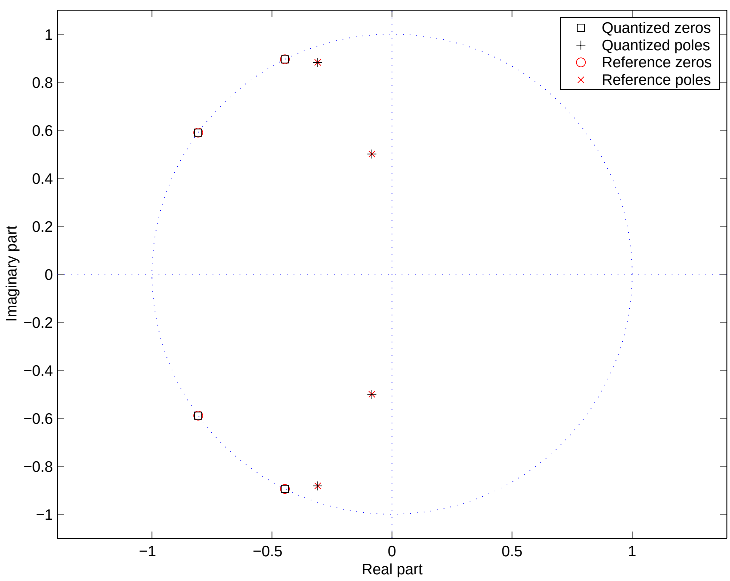
`[zq,pq,kq] = zplane(Hq)` returns the vectors of zeros `zq`, poles `pq`, and gains `kq`. If `Hq` has *n* sections, `zq`, `pq`, and `kq` are returned as 1-by-*n* cell arrays. If there are no zeros (or no poles), `zq` (or `pq`) is set to the empty matrix `[]`.

`[zq,pq,kq,zr,pr,kr] = zplane(Hq)` returns the vectors of zeros `zr`, poles `pr`, and gains `kr` of the reference filter associated with the quantized filter `Hq`, and returns the vectors of zeros `zq`, poles `pq`, and gains `kq` for the quantized filter `Hq`.

Examples

Create a quantized filter `Hq` from a fourth-order digital filter with cutoff frequency of 0.6. Scale the transfer function parameters to avoid overflows due to coefficient quantization. Plot the quantized and unquantized poles and zeros associated with this quantized filter.

```
[b,a] = ellip(4,.5,20,.6);
Hq = dfilt.df2(b/2 a/2);
Hq.arithmetic = 'fixed';
zplane(Hq);
```



See Also

`freqz`, `impz`

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[6] McClellan, J.H., C.S. Burrus, A.V. Oppenheim, T.W. Parks, R.W. Schafer, and H.W. Schuessler, *Computer-Based Exercises for Signal Processing Using MATLAB 5*, Prentice-Hall, 1998.

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