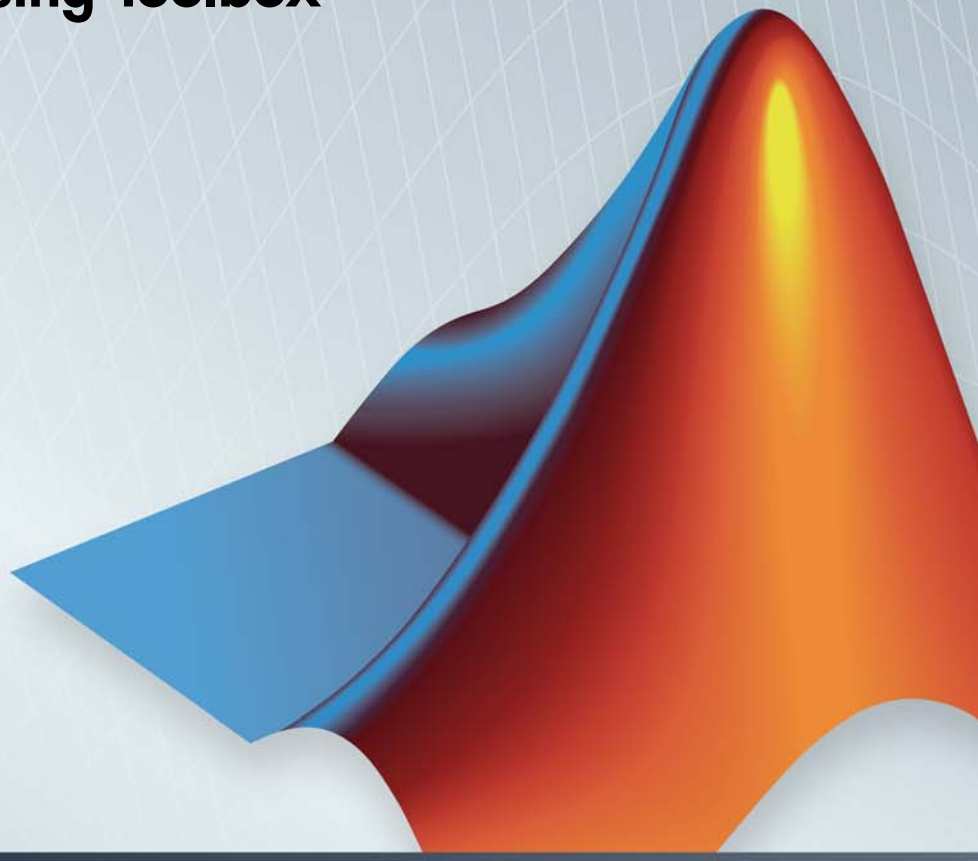
Signal Processing Toolbox™

Reference

R2013a



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Contents

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Functions — Alphabetical List

abs

Purpose

Absolute value (magnitude)

Description

`abs` is a MATLAB® function.

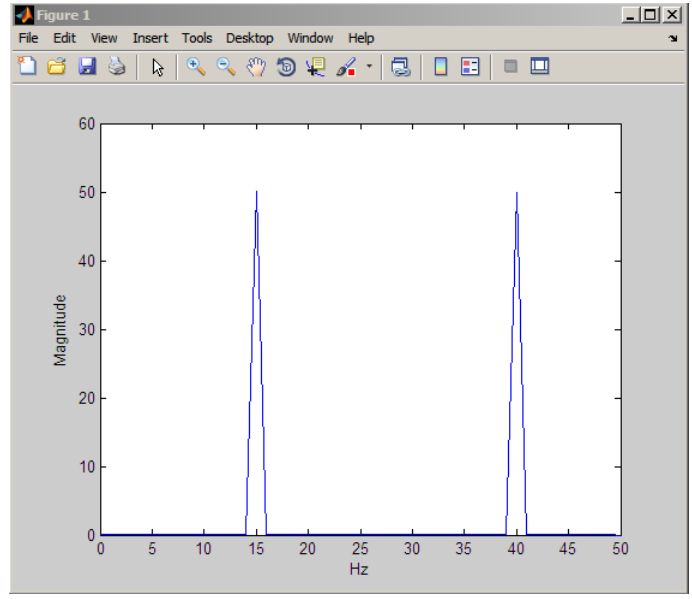
Examples

Calculate the magnitude of the DFT of a sequence:

```
t = (0:99)/100;                          % Time vector
x = sin(2*pi*15*t) + sin(2*pi*40*t);     % Signal
y = fft(x);                              % DFT of x
m = abs(y);                              % Magnitude
m=m(1:51);           % Unique magnitudes
```

Plot the magnitude:

```
f=0:50;     % Frequency vector
plot(f,m);
ylabel('Magnitude'); xlabel('Hz');
```



Purpose

Convert autocorrelation sequence to prediction polynomial

Syntax

```
a = ac2poly(r)
[a,efinal] = ac2poly(r)
```

Description

`a = ac2poly(r)` finds the linear prediction, FIR filter polynomial `a` corresponding to the autocorrelation sequence `r`. `a` is the same length as `r`, and `a(1) = 1`. The prediction filter polynomial represents the coefficients of the prediction filter whose output produces a signal whose autocorrelation sequence is approximately the same as the given autocorrelation sequence `r`.

`[a,efinal] = ac2poly(r)` returns the final prediction error `efinal`, determined by running the filter for `length(r)` steps.

Tips

You can apply this function to real or complex data.

Examples

Consider the autocorrelation sequence:

```
r = [5.0000 -1.5450 -3.9547 3.9331 1.4681 -4.7500];
```

The corresponding prediction filter polynomial is

```
[a,efinal] = ac2poly(r)
a =
    1.0000  0.6147  0.9898  0.0004  0.0034 -0.0077
efinal =
    0.1791
```

References

[1] Kay, S.M. *Modern Spectral Estimation.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

See Also

`ac2rc` | `poly2ac` | `rc2poly`

ac2rc

Purpose

Convert autocorrelation sequence to reflection coefficients

Syntax

```
[k,r0] = ac2rc(r)
```

Description

`[k,r0] = ac2rc(r)` finds the reflection coefficients `k` corresponding to the autocorrelation sequence `r`. `r0` contains the zero-lag autocorrelation. If `r` is a matrix where the columns are separate channels of autocorrelation sequences, `r0` contains the zero-lag autocorrelation coefficient for each channel. These reflection coefficients can be used to specify the lattice prediction filter that produces a sequence with approximately the same autocorrelation sequence as the given sequence `r`.

Tips

You can apply this function to real or complex data.

References

[1] Kay, S.M. *Modern Spectral Estimation.* Englewood Cliffs, NJ: Prentice-Hall, 1988.

See Also

`ac2poly` | `poly2rc` | `rc2ac`

Purpose

Phase angle

Description

`angle` is a MATLAB function.

Signal-specific Example

Calculate the phase of the FFT of a sequence.

```
t = (0:99)/100;                          % Time vector
x = sin(2*pi*15*t) + sin(2*pi*40*t);     % Signal
y = fft(x);                              % Compute DFT of x
p = unwrap(angle(y));                    % Phase
```

Plot the phase:

```
f = (0:length(y)-1)'/length(y)*100;      % Frequency vector
plot(f,p)
```

arburg

Purpose

Autoregressive (AR) all-pole model parameters estimated using Burg method

Syntax

```
ar_coeffs = arburg(data,order)
[ar_coeffs,NoiseVariance] = arburg(data,order)
[ar_coeffs,NoiseVariance,reflect_coeffs] = arburg(data,order)
```

Description

`ar_coeffs = arburg(data,order)` returns the AR coefficients for the input `data` and model `order`. The elements of `ar_coeffs` are normalized by `ar_coeffs(1)`. The model `order` requires an integer value less than the length of the input `data`.

`[ar_coeffs,NoiseVariance] = arburg(data,order)` returns the estimated variance `NoiseVariance` of the white noise input.

`[ar_coeffs,NoiseVariance,reflect_coeffs] = arburg(data,order)` returns the reflection coefficients `reflect_coeffs`.

Definitions

AR(p) Model

In an AR model of order p, the current output is a linear combination of the past p outputs plus a white noise input. The weights on the p past outputs minimize the mean-square prediction error of the autoregression. If *y[n]* is the current value of the output and *x[n]* is a zero mean white noise input, the AR(p) model is:

$$y[n] + \sum_{k=1}^{p} a(k)y[n-k] = x[n]$$

Reflection Coefficients

The *reflection coefficients* are the partial autocorrelation coefficients scaled by (–1). The reflection coefficients indicate the time dependence between *y[n]* and *y[n-k]* after subtracting the prediction based on the intervening *k-1* time steps.

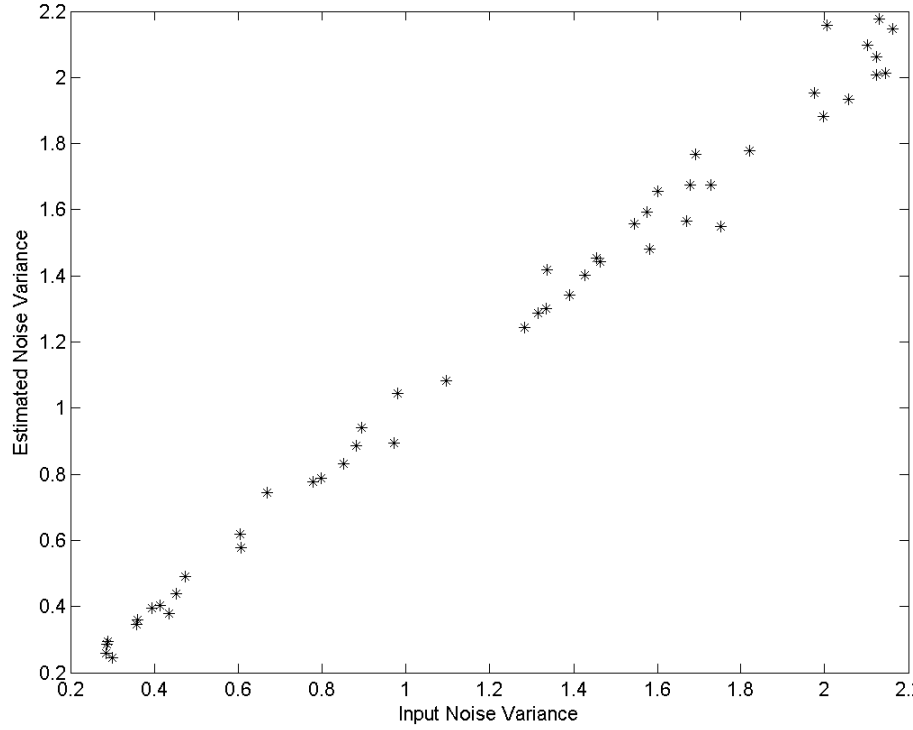
Examples

Generate AR(4) process and estimate coefficients:

```
A=[1 -2.7607 3.8106 -2.6535 0.9238];
% AR(4) coefficients
y=filter(1,A,0.2*randn(1024,1));
% Filter a white noise input to create AR(4) process
ar_coeffs=arburg(y,4);
%compare the results in ar_coeffs to the vector A.
```

Estimate input noise variance for AR(4) model:

```
A=[1 -2.7607 3.8106 -2.6535 0.9238];
% Generate noise standard deviations
% Seed random number generator for reproducible results
rng default;
noise_stdz=rand(50,1)+0.5;
for j=1:50
y=filter(1,A,noise_stdz(j)*randn(1024,1));
[ar_coeffs,NoiseVariance(j)]=arburg(y,4);
end
%Compare actual vs. estimated variances
plot(noise_stdz.^2,NoiseVariance,'k*');
xlabel('Input Noise Variance');
ylabel('Estimated Noise Variance');
```



Algorithms

The Burg method estimates the reflection coefficients and uses the reflection coefficients to estimate the AR coefficients recursively. You can find the recursion and lattice filter relations describing the update of the forward and backward prediction errors in [1].

References

[1] Kay, S.M. *Modern Spectral Estimation: Theory and Application*. Englewood Cliffs, NJ: Prentice Hall, 1988, pp. 228–230.

See Also

`arcov` | `armcov` | `aryule` | `levinson` | `lpc`

How To

- “Parametric Modeling”

Purpose

Estimate AR model parameters using covariance method

Syntax

```
a = arcov(x,p)
[a,e] = arcov(x,p)
```

Description

`a = arcov(x,p)` uses the covariance method to fit a `p`th order autoregressive (AR) model to the input signal, `x`, which is assumed to be the output of an AR system driven by white noise. This method minimizes the forward prediction error in the least-squares sense. The vector `a` contains the normalized estimate of the AR system parameters, $A(z)$, in descending powers of z. Let *y(n)* be a wide-sense stationary random process obtained by filtering a white noise input with variance e with the system function $A(z)$. If $P_y(e^{j\omega})$ is the power spectral density of *y(n)*, then:

$$P_y(e^{j\omega}) = \frac{e}{|A(e^{j\omega})|^2} = \frac{e}{|1+\sum_{k=1}^{P} a(k)e^{-j\omega k}|^2}$$

Because the method characterizes the input data using an all-pole model, the correct choice of the model order `p` is important.

`[a,e] = arcov(x,p)` returns the variance estimate, `e`, of the white noise input to the AR model.

See Also

`arburg` | `armcov` | `aryule` | `lpc` | `pcov` | `prony`

Purpose

Estimate AR model parameters using modified covariance method

Syntax

```
a = armcov(x,p)
[a,e] = armcov(x,p)
```

Description

`a = armcov(x,p)` uses the modified covariance method to fit a `p`th order autoregressive (AR) model to the input signal, `x`, which is assumed to be the output of an AR system driven by white noise. This method minimizes the forward and backward prediction errors in the least-squares sense. The vector `a` contains the normalized estimate of the AR system parameters, $A(z)$, in descending powers of z. Let *y(n)* be a wide-sense stationary random process obtained by filtering a white noise input with variance e with the system function $A(z)$. If $P_y(e^{j\omega})$ is the power spectral density of *y(n)*:

$$P_y(e^{j\omega}) = \frac{e}{|A(e^{j\omega})|^2} = \frac{e}{|1+\sum_{k=1}^{P} a(k)e^{-j\omega k}|^2}$$

Because the method characterizes the input data using an all-pole model, the correct choice of the model order `p` is important.

`[a,e] = armcov(x,p)` returns the variance estimate, `e`, of the white noise input to the AR model.

See Also

`arburg` | `arcov` | `aryule` | `lpc` | `pmcov` | `prony`

Purpose

Estimate autoregressive (AR) all-pole model using Yule-Walker method

Syntax

```
ar_coeffs = aryule(data,order)
[ar_coeffs,NoiseVariance] = aryule(data,order)
[ar_coeffs,NoiseVariance,reflect_coeffs] = aryule(data,order)
```

Description

`ar_coeffs = aryule(data,order)` returns the AR coefficients for the input `data` and model `order`. The elements of `ar_coeffs` are normalized by `ar_coeffs(1)`. `order` is a positive integer that cannot exceed the length of the input data.

`[ar_coeffs,NoiseVariance] = aryule(data,order)` returns the estimated variance `NoiseVariance` of the white noise input.

`[ar_coeffs,NoiseVariance,reflect_coeffs] = aryule(data,order)` returns the reflection coefficients `reflect_coeffs`.

Definitions

AR(p) Model

In an AR model of order p, the current output is a linear combination of the past p outputs plus a white noise input. The weights on the p past outputs minimize the mean-square prediction error of the autoregression. If *y[n]* is the current value of the output and *x[n]* is a zero-mean white noise input, the AR(p) model is:

$$\sum_{k=0}^{p} a[k]y[n-k] = x[n]$$

Reflection Coefficients

The reflection coefficients are the partial autocorrelation coefficients scaled by (–1). The reflection coefficients indicate the time dependence between *y[n]* and *y[n-k]* after subtracting the prediction based on the intervening *k-1* time steps.

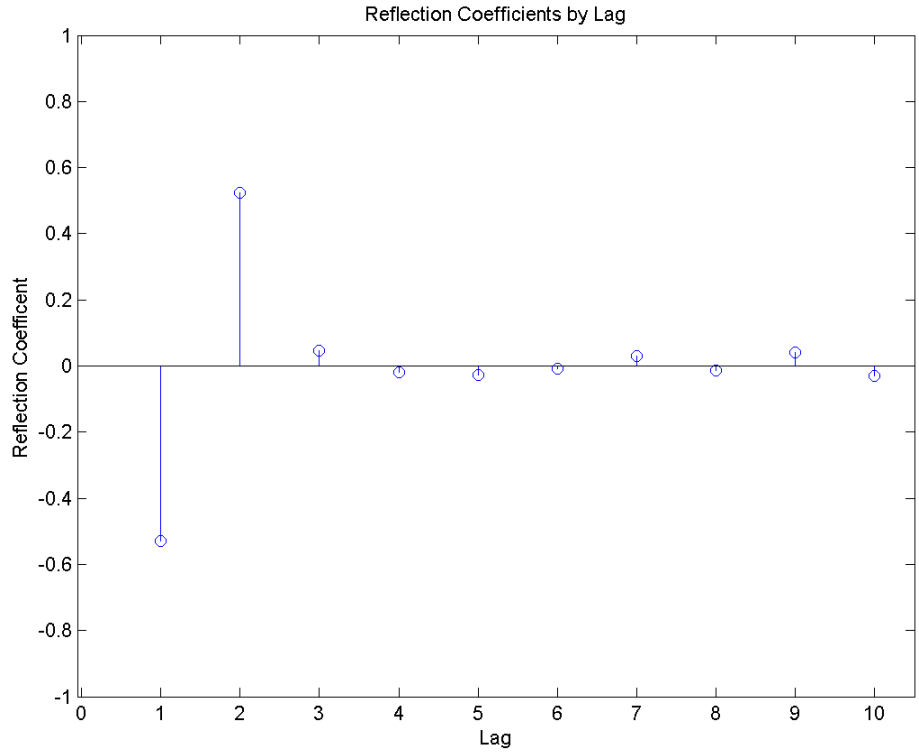
Examples

Create an AR(4) process and estimate the coefficients:

```
A=[1 -2.7607 3.8106 -2.6535 0.9238];
% AR(4) coefficients
y=filter(1,A,0.2*randn(1024,1));
%filter a white noise input to create AR(4) process
ar_coeffs=aryule(y,4);
%compare the results in ar_coeffs to the vector A.
```

Estimate model order using decay of reflection coefficients:

```
rng default;
y=filter(1,[1 -0.75 0.5],0.2*randn(1024,1));
%create AR(2) process
[ar_coeffs,NoiseVariance,reflect_coeffs]=aryule(y,10);
% Fit AR(10) model
stem(reflect_coeffs); axis([-0.05 10.5 -1 1]);
title('Reflection Coefficients by Lag'); xlabel('Lag');
ylabel('Reflection Coefficent');
```



The reflection coefficients decay to zero after lag 2, which indicates that an AR(10) model significantly overestimates the time dependence in the data.

Algorithms

`aryule` uses the Levinson-Durbin recursions on the biased estimate of the sample autocorrelation sequence to compute the coefficients.

References

Monson,H. *Statistical Digital Signal Processing and Modeling*, John Wiley & Sons, 1996.

See Also

`arburg` | `arcov` | `armcov` | `levinson` | `lpc`

How To

- “Parametric Modeling”

bandpower

Purpose

Band power

Syntax

```
p = bandpower(x)
p = bandpower(x,fs,freqrange)

p = bandpower(pxx,f,psdflag)
p = bandpower(pxx,f,freqrange,psdflag)
```

Description

`p = bandpower(x)` returns the average power in the input signal, `x`.

`p = bandpower(x,fs,freqrange)` returns the average power in the frequency range, `freqrange`, specified as a two-element vector. You must input the sampling frequency, `fs`, to return the power in a specified frequency range. `bandpower` uses a modified periodogram to determine the average power in `freqrange`.

`p = bandpower(pxx,f,psdflag)` returns the average power computed by integrating the power spectral density (PSD) estimate, `pxx`. The integral is approximated by the rectangle method. The input, `f`, is a vector of frequencies corresponding to the PSD estimates in `pxx`. `psdflag` is the string `'psd'`, which indicates the input is a PSD estimate and not time series data.

`p = bandpower(pxx,f,freqrange,psdflag)` returns the average power contained in the frequency interval, `freqrange`. If the frequencies in `freqrange` do not match values in `f`, the closest values are used. The average power is computed by integrating the power spectral density (PSD) estimate, `pxx`. The integral is approximated by the rectangle method.

Input Arguments

x - Time series input

row or column vector

Input time series data, specified as a row or column vector

Example: `cos(pi/4*(0:159))+randn(1,160)`

Data Types
`double | single`
Complex Number Support: Yes

fs - Sampling frequency
1 (default) | positive scalar

Sampling frequency for the input time series data, specified as a positive scalar.

Data Types
`double | single`

freqrange - Frequency range for band power computation
two-element real-valued row or column vector

Frequency range for the band power computation, specified as a two-element real-valued row or column vector. If the input signal, `x`, contains N samples, `freqrange` must be within the following intervals.

- [0, `fs`/2] if `x` is real-valued and N is even
- [0, (N-1)`fs`/(2N)] if `x` is real-valued and N is odd
- [-(N-2)`fs`/(2N), `fs`/2] if `x` is complex-valued and N is even
- [-(N-1)`fs`/(2N), (N-1)`fs`/(2N)] if `x` is complex-valued and N is odd

Data Types
`double | single`

pxx - PSD estimates
real-valued column vector with nonnegative elements

One- or two-sided PSD estimate, specified as a column vector with nonnegative elements.

Data Types
`double | single`

f - Frequency vector for PSD estimates
column vector with real-valued elements

Frequency vector, specified as a column vector. The frequency vector, `f`, contains the frequencies corresponding to the PSD estimates in `pxx`.

Data Types
`double | single`

psdflag - Power spectrum input flag
`'psd'`

Flag indicating that the input data is a PSD estimate, specified as the string `'psd'`.

Output Arguments

p - Average band power
nonnegative scalar

Average band power, specified as a nonnegative scalar.

Data Types
`double | single`

Examples

Comparison with $\bullet_2$ Norm

Create a signal consisting of a 100-Hz sine wave in additive N(0,1) white Gaussian noise. The sampling frequency is 1 kHz. Determine the average power compare against the ℓ_2 norm.

```
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t)+randn(size(t));
p = bandpower(x)
norm(x,2)^2/numel(x)
```

Percentage of Total Power in Frequency Interval

Determine the percentage of the total power in a specified frequency interval.

Create a signal consisting of a 100-Hz sine wave in additive N(0,1) white Gaussian noise. The sampling frequency is 1 kHz. Determine the percentage of the total power in the [50,150] Hz interval.

```
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t)+randn(size(t));
pband = bandpower(x,1000,[50 100]);
ptot = bandpower(x,1000,[0 500]);
per_power = 100*(pband/ptot)
```

Periodogram Input

Determine the average power by first computing a PSD estimate using the periodogram. Input the PSD estimate to `bandpower`.

Create a signal consisting of a 100-Hz sine wave in additive N(0,1) white Gaussian noise. The sampling frequency is 1 kHz. Obtain the periodogram and use the `psdflag`, `'psd'`, to compute the average power using the PSD estimate. Compare the result against the average power computed in the time domain.

```
t = 0:0.001:1-0.001;
Fs = 1000;
x = cos(2*pi*100*t)+randn(size(t));
[Pxx,F] = periodogram(x,rectwin(length(x)),length(x),Fs);
p = bandpower(Pxx,F,'psd')
norm(x,2)^2/numel(x)
```

Percentage of Power in Frequency Band (Periodogram)

Determine the percentage of the total power in a specified frequency interval using the periodogram as the input.

Create a signal consisting of a 100-Hz sine wave in additive N(0,1) white Gaussian noise. The sampling frequency is 1 kHz. Obtain the periodogram and corresponding frequency vector. Using the PSD estimate, determine the percentage of the total power in the interval [50,150] Hz.

```
Fs = 1000;
t = 0:1/Fs:1-0.001;
x = cos(2*pi*100*t)+randn(size(t));
[Pxx,F] = periodogram(x,rectwin(length(x)),length(x),Fs);
pband = bandpower(Pxx,F,[50 100],'psd');
```

```
ptot = bandpower(Pxx,F,'psd');
per_power = 100*(pband/ptot)
```

See Also

periodogram | sfdr

Purpose

Modified Bartlett-Hann window

Syntax

```
w = barthannwin(L)
```

Description

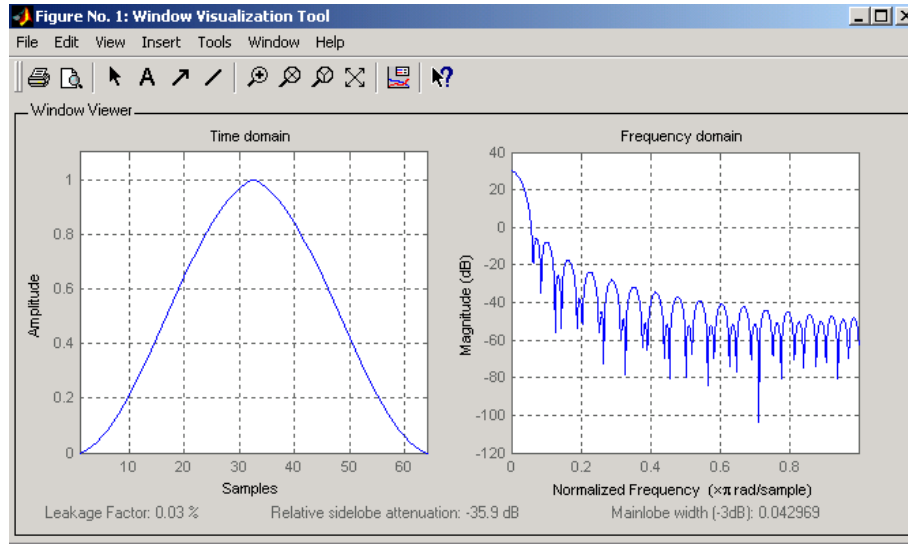
`w = barthannwin(L)` returns an `L`-point modified Bartlett-Hann window in the column vector `w`. Like Bartlett, Hann, and Hamming windows, this window has a mainlobe at the origin and asymptotically decaying sidelobes on both sides. It is a linear combination of weighted Bartlett and Hann windows with near sidelobes lower than both Bartlett and Hann and with far sidelobes lower than both Bartlett and Hamming windows. The mainlobe width of the modified Bartlett-Hann window is not increased relative to either Bartlett or Hann window mainlobes.

Note The Hann window is also called the Hanning window.

Examples

Create a 64-point Bartlett-Hann window and display the result using WVTool:

```
L=64;
wvtool(barthannwin(L))
```



Algorithms

The equation for computing the coefficients of a Modified Bartlett-Hanning window is

$$w(n) = 0.62 - 0.48\left|\left(\frac{n}{N} - 0.5\right)\right| + 0.38\cos\left(2\pi\left(\frac{n}{N} - 0.5\right)\right)$$

where $0 \le n \le N$ and the window length is $L = N + 1$.

References

[1] Ha, Y.H., and J.A. Pearce. "A New Window and Comparison to Standard Windows." *IEEE® Transactions on Acoustics, Speech, and Signal Processing*. Vol. 37, No. 2, (February 1999). pp. 298-301.

[2] Oppenheim, A.V., and R.W. Schafer. *Discrete-Time Signal Processing*. Upper Saddle River, NJ: Prentice-Hall, 1999, p. 468.

See Also

`bartlett` | `blackmanharris` | `bohmanwin` | `nuttallwin` | `parzenwin` | `rectwin` | `triang` | `window` | `wintool` | `wvtool`

Purpose

Bartlett window

Syntax

```
w = bartlett(L)
```

Description

`w = bartlett(L)` returns an `L`-point Bartlett window in the column vector `w`, where `L` must be a positive integer. The coefficients of a Bartlett window are computed as follows:

$$w(n) = \begin{cases} \dfrac{2n}{N}, & 0 \le n \le \dfrac{N}{2} \\ 2 - \dfrac{2n}{N}, & \dfrac{N}{2} \le n \le N \end{cases}$$

The window length $L = N + 1$.

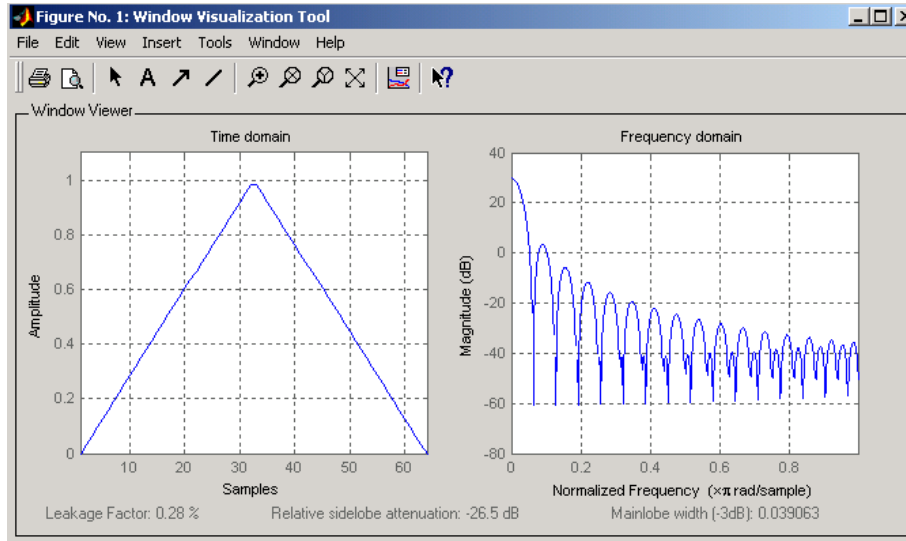
The Bartlett window is very similar to a triangular window as returned by the `triang` function. The Bartlett window always ends with zeros at samples 1 and `n`, however, while the triangular window is nonzero at those points. For `L` odd, the center `L-2` points of `bartlett(L)` are equivalent to `triang(L-2)`.

Note If you specify a one-point window (set `L=1`), the value `1` is returned.

Examples

Create a 64-point Bartlett window and display the result using WVTool:

```
L=64;
wvtool(bartlett(L))
```



References

[1] Oppenheim, A.V., and R.W. Schafer. *Discrete-Time Signal Processing*. Upper Saddle River, NJ: Prentice-Hall, 1999, pp. 468-471.

See Also

`barthannwin` | `blackmanharris` | `bohmanwin` | `nuttallwin` | `parzenwin` | `rectwin` | `triang` | `window` | `wintool` | `wvtool`

Purpose

Bessel analog lowpass filter prototype

Syntax

```
[z,p,k] = besselap(n)
```

Description

`[z,p,k] = besselap(n)` returns the poles and gain of an order `n` Bessel analog lowpass filter prototype. `n` must be less than or equal to 25. The function returns the poles in the length `n` column vector `p` and the gain in scalar `k`. `z` is an empty matrix because there are no zeros. The transfer function is

$$H(s) = \frac{k}{(s - p(1))(s - p(2))\cdots(s - p(n))}$$

`besselap` normalizes the poles and gain so that at low frequency and high frequency the Bessel prototype is asymptotically equivalent to the Butterworth prototype of the same order [1]. The magnitude of the filter is less than $\sqrt{1/2}$ at the unity cutoff frequency $\Omega_c = 1$.

Analog Bessel filters are characterized by a group delay that is maximally flat at zero frequency and almost constant throughout the passband. The group delay at zero frequency is

$$\left(\frac{(2n)!}{2^n n!}\right)^{1/n}$$

Algorithms

`besselap` finds the filter roots from a lookup table constructed using Symbolic Math Toolbox™ software.

References

[1] Rabiner, L.R., and B. Gold. *Theory and Application of Digital Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1975. Pgs. 228-230.

See Also

`besself` | `buttap` | `cheb1ap` | `cheb2ap` | `ellipap`

besself

Purpose

Bessel analog filter design

Syntax

```
[b,a] = besself(n,Wo)
[z,p,k] = besself(...)
[A,B,C,D] = besself(...)
```

Description

`besself` designs lowpass, analog Bessel filters, which are characterized by almost constant group delay across the entire passband, thus preserving the wave shape of filtered signals in the passband. `besself` does not support the design of digital Bessel filters.

`[b,a] = besself(n,Wo)` designs an order `n` lowpass analog Bessel filter, where Wo is the frequency up to which the filter's group delay is approximately constant. Larger values of the filter order (`n`) produce a group delay that better approximates a constant up to frequency `Wo`.

`besself` returns the filter coefficients in the length `n+1` row vectors `b` and `a`, with coefficients in descending powers of *s,* derived from this transfer function:

$$H(s) = \frac{B(s)}{A(s)} = \frac{b(1)s^n + b(2)s^{n-1} + \ldots + b(n+1)}{s^n + a(2)s^{n-1} + \ldots + a(n+1)}$$

`[z,p,k] = besself(...)` returns the zeros and poles in length `n` or `2*n` column vectors `z` and `p` and the gain in the scalar `k`.

`[A,B,C,D] = besself(...)` returns the filter design in state-space form, where `A`, `B`, `C`, and `D` are

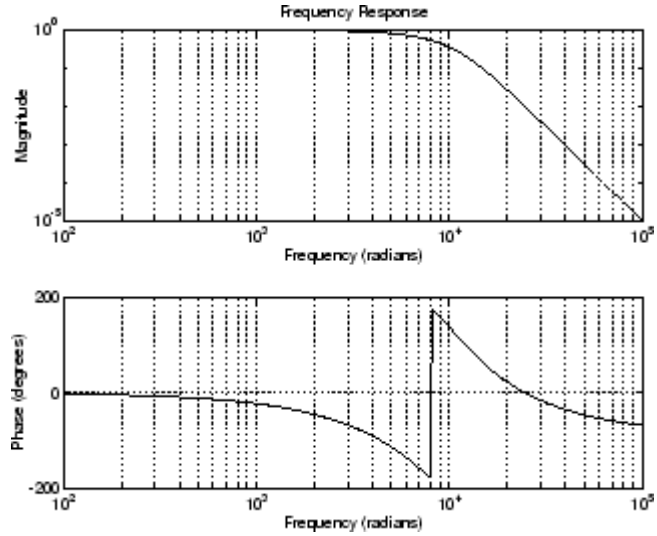
$$x = Ax + Bu$$
$$y = Cx + Du$$

and u is the input, x is the state vector, and y is the output.

Examples

Design a fifth-order analog lowpass Bessel filter with an approximate constant group delay up to 10,000 rad/s and plot the frequency response of the filter using `freqs`:

```
[b,a] = besself(5,10000);
freqs(b,a)                    % Plot frequency response
```



Limitations

Lowpass Bessel filters have a monotonically decreasing magnitude response, as do lowpass Butterworth filters. Compared to the Butterworth, Chebyshev, and elliptic filters, the Bessel filter has the slowest rolloff and requires the highest order to meet an attenuation specification.

For high order filters, the state-space form is the most numerically accurate, followed by the zero-pole-gain form. The transfer function coefficient form is the least accurate; numerical problems can arise for filter orders as low as 15.

Algorithms

`besself` performs a four-step algorithm:

1. It finds lowpass analog prototype poles, zeros, and gain using the `besselap` function.

2. It converts the poles, zeros, and gain into state-space form.

3 It transforms the lowpass prototype into a lowpass filter that meets the design specifications.

4 It converts the state-space filter back to transfer function or zero-pole-gain form, as required.

See Also

`besselap` | `butter` | `cheby1` | `cheby2` | `ellip`

Purpose

Bilinear transformation method for analog-to-digital filter conversion

Syntax

```
[zd,pd,kd] = bilinear(z,p,k,fs)
[zd,pd,kd] = bilinear(z,p,k,fs,fp)
[numd,dend] = bilinear(num,den,fs)
[numd,dend] = bilinear(num,den,fs,fp)
[Ad,Bd,Cd,Dd] = bilinear(A,B,C,D,fs)
[Ad,Bd,Cd,Dd] = bilinear(A,B,C,D,fs,fp)
```

Description

The *bilinear transformation* is a mathematical mapping of variables. In digital filtering, it is a standard method of mapping the *s* or analog plane into the *z* or digital plane. It transforms analog filters, designed using classical filter design techniques, into their discrete equivalents.

The bilinear transformation maps the *s*-plane into the *z*-plane by

$$H(z) = H(s)\Big|_{s=2f_z\frac{z-1}{z+1}}$$

This transformation maps the $j\Omega$ axis (from $\Omega = -\infty$ to $+\infty$) repeatedly around the unit circle (e^{jw}, from $\omega = -\pi$ to π) by

$$\omega = 2\tan^{-1}\left(\frac{\Omega}{2f_s}\right)$$

`bilinear` can accept an optional parameter `Fp` that specifies prewarping. `fp`, in hertz, indicates a "match" frequency, that is, a frequency for which the frequency responses before and after mapping match exactly. In prewarped mode, the bilinear transformation maps the *s*-plane into the *z*-plane with

$$H(z) = H(s)\Big|_{s=\frac{2\pi f_p}{\tan\left(\pi\frac{f_p}{f_z}\right)}\frac{(z-1)}{(z+1)}}$$

With the prewarping option, `bilinear` maps the $j\Omega$ axis (from $\Omega = -\infty$ to $+\infty$) repeatedly around the unit circle (e^{jw}, from $\omega = -\pi$ to π) by

$$\omega = 2\tan^{-1}\left(\frac{\Omega\tan\left(\pi\frac{f_p}{f_s}\right)}{2\pi f_p}\right)$$

In prewarped mode, `bilinear` matches the frequency $2\pi f_p$ (in radians per second) in the *s*-plane to the normalized frequency $2\pi f_p/f_s$ (in radians per second) in the *z*-plane.

The `bilinear` function works with three different linear system representations: zero-pole-gain, transfer function, and state-space form.

Zero-Pole-Gain

`[zd,pd,kd] = bilinear(z,p,k,fs)` and

`[zd,pd,kd] = bilinear(z,p,k,fs,fp)` convert the *s*-domain transfer function specified by `z`, `p`, and `k` to a discrete equivalent. Inputs `z` and `p` are column vectors containing the zeros and poles, `k` is a scalar gain, and `fs` is the sampling frequency in hertz. `bilinear` returns the discrete equivalent in column vectors `zd` and `pd` and scalar `kd`. The optional match frequency, `fp` is in hertz and is used for prewarping.

Transfer Function

`[numd,dend] = bilinear(num,den,fs)` and

`[numd,dend] = bilinear(num,den,fs,fp)` convert an *s*-domain transfer function given by `num` and `den` to a discrete equivalent. Row vectors `num` and `den` specify the coefficients of the numerator and denominator, respectively, in descending powers of *s*. Let *B(s)* be the numerator polynomial and *A(s)* be the denominator polynomial. The transfer function is:

$$\frac{B(s)}{A(s)} = \frac{B(1)s^n + \cdots + B(n)s + B(n+1)}{A(1)s^m + \cdots + A(m)s + A(m+1)}$$

`fs` is the sampling frequency in hertz. `bilinear` returns the discrete equivalent in row vectors `numd` and `dend` in descending powers of *z*

(ascending powers of z^{-1}). `fp` is the optional match frequency, in hertz, for prewarping.

State-Space

`[Ad,Bd,Cd,Dd] = bilinear(A,B,C,D,fs)` and

`[Ad,Bd,Cd,Dd] = bilinear(A,B,C,D,fs,fp)` convert the continuous-time state-space system in matrices `A`, `B`, `C`, `D`

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

to the discrete-time system:

$$x[n+1] = A_d x[n] + B_d u[n]$$
$$y[n] \quad = C_d x[n] + D_d u[n]$$

`fs` is the sampling frequency in hertz. `bilinear` returns the discrete equivalent in matrices `Ad`, `Bd`, `Cd`, `Dd`. The optional match frequency, `fp` is in hertz and is used for prewarping.

Algorithms

`bilinear` uses one of two algorithms depending on the format of the input linear system you supply. One algorithm works on the zero-pole-gain format and the other on the state-space format. For transfer function representations, `bilinear` converts to state-space form, performs the transformation, and converts the resulting state-space system back to transfer function form.

Zero-Pole-Gain Algorithm

For a system in zero-pole-gain form, `bilinear` performs four steps:

1. If `fp` is present, it prewarps:

   ```
   fp = 2*pi*fp;
   fs = fp/tan(fp/fs/2)
   ```

   otherwise, `fs = 2*fs`.

2 It strips any zeros at ±∞ using

```
z = z(finite(z));
```

3 It transforms the zeros, poles, and gain using

```
pd = (1+p/fs)./(1-p/fs);    % Do bilinear transformation
zd = (1+z/fs)./(1-z/fs);
kd = real(k*prod(fs-z)./prod(fs-p));
```

4 It adds extra zeros at -1 so the resulting system has equivalent numerator and denominator order.

State-Space Algorithm

For a system in state-space form, `bilinear` performs two steps:

1 If `fp` is present, let

$$\lambda = \frac{\pi f_p}{\tan(\pi f_p / f_s)}$$

If `fp` is not present, let λ=*fs*.

2 Compute `Ad`, `Bd`, `Cd`, and `Dd` in terms of `A`, `B`, `C`, and `D` using

$$Ad = (I - A\tfrac{1}{2\lambda})^{-1}(I + A\tfrac{1}{2\lambda})$$
$$Bd = \tfrac{1}{\sqrt{\lambda}}(I - A\tfrac{1}{2\lambda})^{-1}B$$
$$Cd = \tfrac{1}{\sqrt{\lambda}}C(I - A\tfrac{1}{2\lambda})^{-1}$$
$$Dd = \tfrac{1}{2\lambda}C(I - A\tfrac{1}{2\lambda})^{-1}B + D$$

Diagnostics

`bilinear` requires that the numerator order be no greater than the denominator order. If this is not the case, `bilinear` displays

```
Numerator cannot be higher order than denominator.
```

For `bilinear` to distinguish between the zero-pole-gain and transfer function linear system formats, the first two input parameters must be vectors with the same orientation in these cases. If this is not the case, `bilinear` displays

```
First two arguments must have the same orientation.
```

References

[1] Parks, T.W., and C.S. Burrus. *Digital Filter Design.* New York: John Wiley & Sons, 1987. Pgs. 209-213.

[2] Oppenheim, A.V., and R.W. Schafer. *Discrete-Time Signal Processing.* Upper Saddle River, NJ: Prentice-Hall, 1999, pp. 450-454.

See Also

`impinvar` | `lp2bp` | `lp2bs` | `lp2hp` | `lp2lp`

bitrevorder

Purpose

Permute data into bit-reversed order

Syntax

```
y = bitrevorder(x)
[y,i] = bitrevorder(x)
```

Description

`bitrevorder` is useful for pre-arranging filter coefficients so that bit-reversed ordering does not have to be performed as part of an `fft` or inverse FFT (`ifft`) computation. This can improve run-time efficiency for external applications or for Simulink® blockset models. Both MATLAB `fft` and `ifft` functions process linear input and output.

Note Using `bitrevorder` is equivalent to using `digitrevorder` with radix base 2.

`y = bitrevorder(x)` returns the input data in bit-reversed order in vector or matrix `y`. The length of `x` must be an integer power of 2. If `x` is a matrix, the bit-reversal occurs on the first dimension of `x` with size greater than 1. `y` is the same size as `x`.

`[y,i] = bitrevorder(x)` returns the bit-reversed vector or matrix `y` and the bit-reversed indices `i`, such that `y = x(i)`. Recall that MATLAB matrices use 1-based indexing, so the first index of `y` will be 1, not 0.

The following table shows the numbers 0 through 7, the corresponding bits and the bit-reversed numbers.

Linear Index	Bits	Bit- Reversed	Bit-Reversed Index
0	000	000	0
1	001	100	4
2	010	010	2
3	011	110	6
4	100	001	1

Linear Index	Bits	Bit- Reversed	Bit-Reversed Index
5	101	101	5
6	110	011	3
7	111	111	7

Examples

Obtain the bit-reversed ordered output of a vector:

```
x=[0:7]';                 % Create a column vector
[x,bitrevorder(x)]
% ans =
%     0     0
%     1     4
%     2     2
%     3     6
%     4     1
%     5     5
%     6     3
%     7     7
```

See Also

`fft` | `digitrevorder` | `ifft`

blackman

Purpose

Blackman window

Syntax

```
w = blackman(N)
w = blackman(N,SFLAG)
```

Description

`w = blackman(N)` returns the `N`-point symmetric Blackman window in the column vector `w`, where `N` is a positive integer.

`w = blackman(N,SFLAG)` returns an `N`-point Blackman window using the window sampling specified by `'sflag'`, which can be either `'periodic'` or `'symmetric'` (the default). The `'periodic'` flag is useful for DFT/FFT purposes, such as in spectral analysis. The DFT/FFT contains an implicit periodic extension and the periodic flag enables a signal windowed with a periodic window to have perfect periodic extension. When `'periodic'` is specified, `blackman` computes a length `N+1` window and returns the first `N` points. When using windows for filter design, the `'symmetric'` flag should be used.

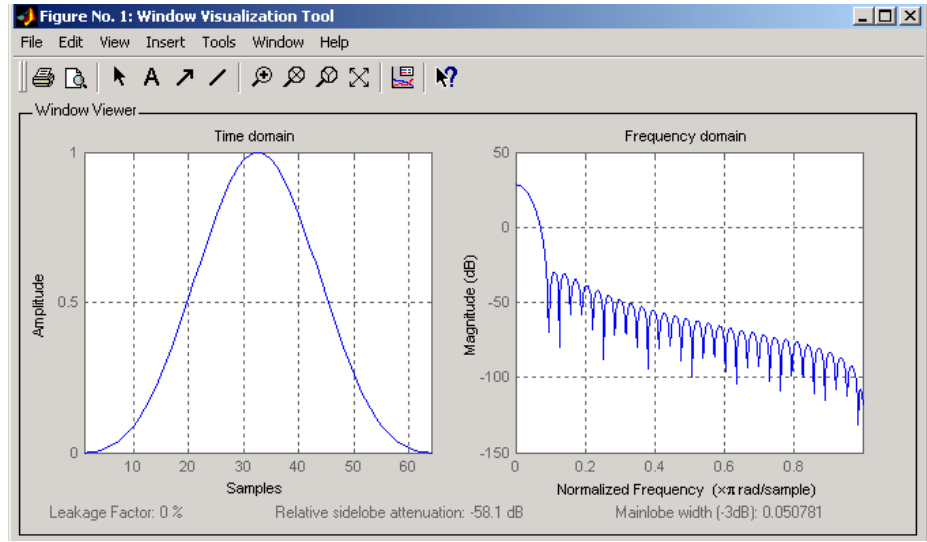
See "Definitions" on page 1-37 for a description of the difference between the symmetric and periodic windows.

Note If you specify a one-point window (set `N=1`), the value `1` is returned.

Examples

Create a 64-point Blackman window and display the result using WVTool:

```
L=64;
wvtool(blackman(L))
```



Definitions

The following equation defines the Blackman window of length N:

$$w(n) = 0.42 - 0.5\cos(2\pi n/(N-1)) + 0.08\cos(4\pi n/(N-1)) \quad 0 \le n \le M-1$$

where M is $N/2$ for N even and $(N+1)/2$ for N odd.

In the **symmetric** case, the second half of the Blackman window $M \le n \le N-1$ is obtained by flipping the first half around the midpoint. The symmetric option is the preferred method when using a Blackman window in FIR filter design.

The **periodic** Blackman window is constructed by extending the desired window length by one sample to $N+1$, constructing a symmetric window, and removing the last sample. The periodic version is the preferred method when using a Blackman window in spectral analysis because the discrete Fourier transform assumes periodic extension of the input vector.

References

[1] Oppenheim, A.V., and R.W. Schafer. *Discrete-Time Signal Processing*. Upper Saddle River, NJ: Prentice-Hall, 1999, pp. 468-471.

See Also `flattopwin` | `hamming` | `hann` | `window` | `wintool` | `wvtool`

Purpose

Minimum 4-term Blackman-Harris window

Syntax

```
w = blackmanharris(N)
w = blackmanharris(N,SFLAG)
```

Description

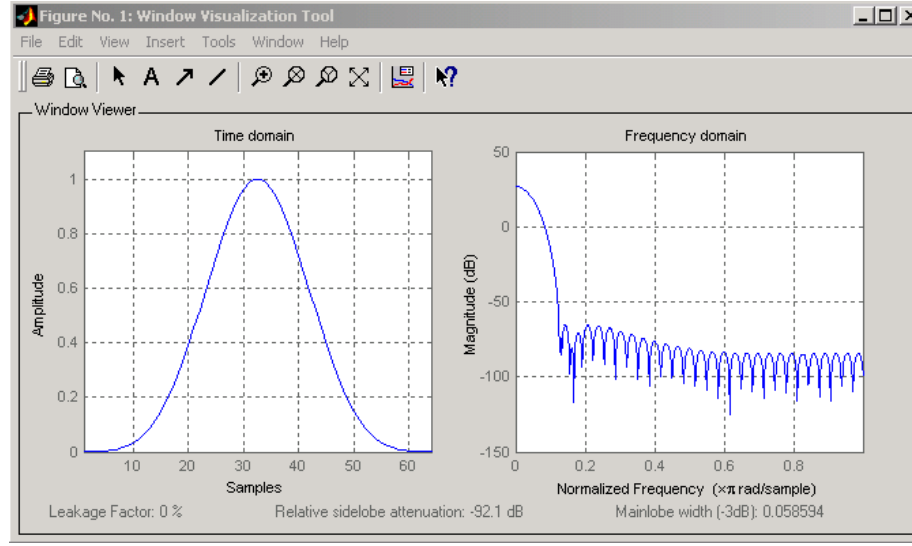
`w = blackmanharris(N)` returns an `N`-point symmetric 4-term Blackman-Harris window in the column vector `w`. The window is minimum in the sense that its maximum sidelobes are minimized.

`w = blackmanharris(N,SFLAG)` uses `SFLAG` window sampling. `SFLAG` can be `'symmetric'` or `'periodic'`. The default is `'symmetric'`. You can find the equations defining the symmetric and periodic windows in "Definitions" on page 1-40.

Examples

Create a 32-point symmetric Blackman-Harris window and display the result using WVTool:

```
N = 32;
wvtool(blackmanharris(N))
```



Definitions

The equation for the **symmetric** 4-term Blackman-harris window of length N is

$$w(n) = a_0 - a_1 \cos(\frac{2\pi n}{N-1}) + a_2 \cos(\frac{4\pi n}{N-1}) - a_3 \cos(\frac{6\pi n}{N-1}) \quad 0 \le n \le N-1$$

The equation for the **periodic** 4-term Blackman-harris window of length N is

$$w(n) = a_0 - a_1 \cos(\frac{2\pi n}{N}) + a_2 \cos(\frac{4\pi n}{N}) - a_3 \cos(\frac{6\pi n}{N}) \quad 0 \le n \le N-1$$

The periodic window is N-periodic.

The following table lists the coefficients:

Coefficient	Value
a0	0.35875
a1	0.48829
a2	0.14128
a3	0.01168

References

[1] Harris, F. J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform.” *Proceedings of the IEEE*. Vol. 66 (January 1978). pp. 51-84.

See Also

`barthannwin` | `bartlett` | `bohmanwin` | `nuttallwin` | `parzenwin` | `rectwin` | `triang` | `window` | `wintool` | `wvtool`

Purpose

Bohman window

Syntax

```
w = bohmanwin(L)
```

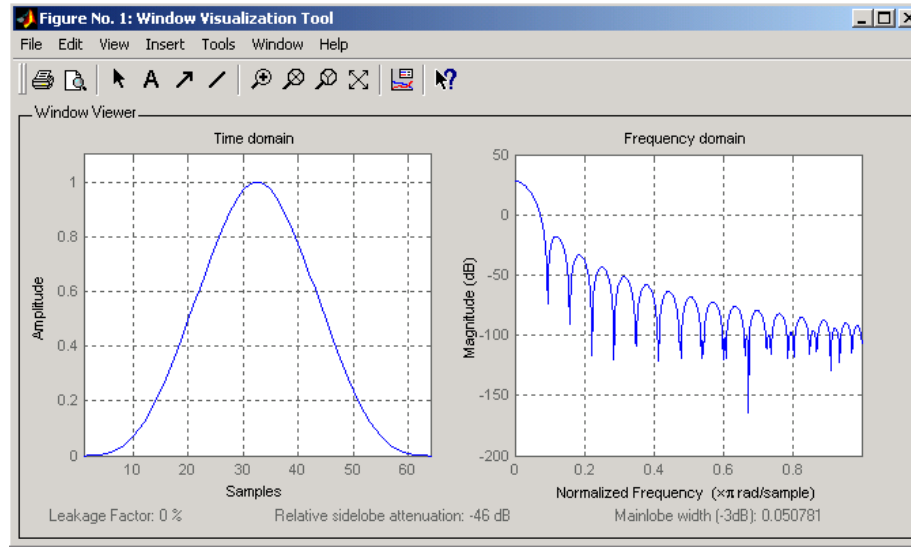
Description

`w = bohmanwin(L)` returns an `L`-point Bohman window in column vector `w`. A Bohman window is the convolution of two half-duration cosine lobes. In the time domain, it is the product of a triangular window and a single cycle of a cosine with a term added to set the first derivative to zero at the boundary. Bohman windows fall off as $1/w^4$.

Examples

Compute a 64-point Bohman window and display the result using WVTool:

```
L=64;
wvtool(bohmanwin(L))
```



Algorithms

The equation for computing the coefficients of a Bohman window is

$$w(x) = (1 - |x|)\cos(\pi |x|) + \frac{1}{\pi}\sin(\pi |x|) \quad -1 \le x \le 1$$

where x is a length L vector of linearly spaced values generated using `linspace`. The first and last elements of the Bohman window are forced to be identically zero.

References

[1] Harris, F. J. "On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform." *Proceedings of the IEEE*. Vol. 66 (January 1978). p. 67.

See Also

`barthannwin` | `bartlett` | `blackmanharris` | `nuttallwin` | `parzenwin` | `rectwin` | `triang` | `window` | `wintool` | `wvtool`

Purpose

Buffer signal vector into matrix of data frames

Syntax

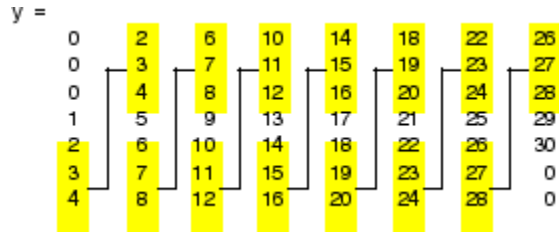
```
y = buffer(x,n)
y = buffer(x,n,p)
y = buffer(x,n,p,opt)
[y,z] = buffer(...)
[y,z,opt] = buffer(...)
```

Description

`y = buffer(x,n)` partitions a length-`L` signal vector `x` into nonoverlapping data segments (frames) of length `n`. Each data frame occupies one column of matrix output `y`, which has `n` rows and `ceil(L/n)` columns. If `L` is not evenly divisible by `n`, the last column is zero-padded to length `n`.

`y = buffer(x,n,p)` overlaps or underlaps successive frames in the output matrix by `p` samples:

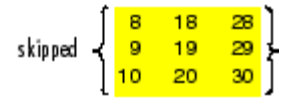
- For `0 < p < n` (overlap), `buffer` repeats the final `p` samples of each frame at the beginning of the following frame. For example, if `x = 1:30` and `n = 7`, an overlap of `p = 3` looks like this.



  The first frame starts with `p` zeros (the default initial condition), and the number of columns in `y` is `ceil(L/(n-p))`.

- For `p < 0` (underlap), `buffer` skips `p` samples between consecutive frames. For example, if `x = 1:30` and `n = 7`, a buffer with underlap of `p = -3` looks like this.

```
y =
     1    11    21
     2    12    22
     3    13    23
     4    14    24
     5    15    25
     6    16    26
     7    17    27
```



The number of columns in `y` is `ceil(L/(n-p))`.

`y = buffer(x,n,p,opt)` specifies a vector of samples to precede `x(1)` in an overlapping buffer, or the number of initial samples to skip in an underlapping buffer:

- For `0 < p < n` (overlap), `opt` specifies a length-`p` vector to insert before `x(1)` in the buffer. This vector can be considered an *initial condition*, which is needed when the current buffering operation is one in a sequence of consecutive buffering operations. To maintain the desired frame overlap from one buffer to the next, `opt` should contain the final `p` samples of the previous buffer in the sequence. See "Continuous Buffering" on page 1-47 below.

  By default, `opt` is `zeros(p,1)` for an overlapping buffer. Set `opt` to `'nodelay'` to skip the initial condition and begin filling the buffer immediately with `x(1)`. In this case, `L` must be `length(p)` or longer. For example, if `x = 1:30` and `n = 7`, a buffer with overlap of `p = 3` looks like this.

```
y =
     1     5     9    13    17    21    25
     2     6    10    14    18    22    26
     3     7    11    15    19    23    27
     4     8    12    16    20    24    28
     5     9    13    17    21    25    29
     6    10    14    18    22    26    30
     7    11    15    19    23    27     0
```

- For `p < 0` (underlap), `opt` is an integer value in the range `[0,-p]` specifying the number of initial input samples, `x(1:opt)`, to skip before adding samples to the buffer. The first value in the buffer

is therefore `x(opt+1)`. By default, `opt` is zero for an underlapping buffer.

This option is especially useful when the current buffering operation is one in a sequence of consecutive buffering operations. To maintain the desired frame underlap from one buffer to the next, `opt` should equal the difference between the total number of points to skip between frames (`p`) and the number of points that were *available* to be skipped in the previous input to `buffer`. If the previous input had fewer than `p` points that could be skipped after filling the final frame of that buffer, the remaining `opt` points need to be removed from the first frame of the current buffer. See "Continuous Buffering" on page 1-47 for an example of how this works in practice.

`[y,z] = buffer(...)` partitions the length-`L` signal vector `x` into frames of length `n`, and outputs only the *full* frames in `y`. If `y` is an overlapping buffer, it has `n` rows and `m` columns, where

```
m = floor(L/(n-p))                % When length(opt) = p
```

or

```
m = floor((L-n)/(n-p))+1           % When opt = 'nodelay'
```

If `y` is an underlapping buffer, it has `n` rows and `m` columns, where

```
m = floor((L-opt)/(n-p)) + (rem((L-opt),(n-p)) >= n)
```

If the number of samples in the input vector (after the appropriate overlapping or underlapping operations) exceeds the number of places available in the `n`-by-`m` buffer, the remaining samples in `x` are output in vector `z`, which for an overlapping buffer has length

```
length(z) = L - m*(n-p)           % When length(opt) = p
```

or

```
length(z) = L - ((m-1)*(n-p)+n)   % When opt = 'nodelay'
```

and for an underlapping buffer has length

```
length(z) = (L-opt) - m*(n-p)
```

Output `z` shares the same orientation (row or column) as `x`. If there are no remaining samples in the input after the buffer with the specified overlap or underlap is filled, `z` is an empty vector.

`[y,z,opt] = buffer(...)` returns the last `p` samples of a overlapping buffer in output `opt`. In an underlapping buffer, `opt` is the difference between the total number of points to skip between frames (`-p`) and the number of points in `x` that were *available* to be skipped after filling the last frame:

- For `0 < p < n` (overlap), `opt` (as an output) contains the final `p` samples in the last frame of the buffer. This vector can be used as the *initial condition* for a subsequent buffering operation in a sequence of consecutive buffering operations. This allows the desired frame overlap to be maintained from one buffer to the next. See "Continuous Buffering" on page 1-47 below.
- For `p < 0` (underlap), `opt` (as an output) is the difference between the total number of points to skip between frames (`-p`) and the number of points in `x` that were *available* to be skipped after filling the last frame.

  ```
  opt = m*(n-p) + opt - L        % z is the empty vector.
  ```

  where `opt` on the right-hand side is the input argument to `buffer`, and `opt` on the left-hand side is the output argument. Here `m` is the number of columns in the buffer, which is

  ```
  m = floor((L-opt)/(n-p)) + (rem((L-opt),(n-p))>=n)
  ```

  Note that for an underlapping buffer output `opt` is always zero when output `z` contains data.

  The `opt` output for an underlapping buffer is especially useful when the current buffering operation is one in a sequence of consecutive

buffering operations. The `opt` output from each buffering operation specifies the number of samples that need to be skipped at the start of the next buffering operation to maintain the desired frame underlap from one buffer to the next. If fewer than `p` points were available to be skipped after filling the final frame of the current buffer, the remaining `opt` points need to be removed from the first frame of the next buffer.

In a sequence of buffering operations, the `opt` output from each operation should be used as the `opt` input to the subsequent buffering operation. This ensures that the desired frame overlap or underlap is maintained from buffer to buffer, as well as from frame to frame within the same buffer. See "Continuous Buffering" on page 1-47 below for an example of how this works in practice.

Continuous Buffering

In a continuous buffering operation, the vector input to the `buffer` function represents one frame in a sequence of frames that make up a discrete signal. These signal frames can originate in a frame-based data acquisition process, or within a frame-based algorithm like the FFT.

As an example, you might acquire data from an A/D card in frames of 64 samples. In the simplest case, you could rebuffer the data into frames of 16 samples; `buffer` with `n = 16` creates a buffer of four frames from each 64-element input frame. The result is that the signal of frame size 64 has been converted to a signal of frame size 16; no samples were added or removed.

In the general case where the original signal frame size, `L`, is not equally divisible by the new frame size, `n`, the overflow from the last frame needs to be captured and recycled into the following buffer. You can do this by iteratively calling `buffer` on input `x` with the two-output-argument syntax:

```
[y,z] = buffer([z;x],n)     % x is a column vector.
[y,z] = buffer([z,x],n)     % x is a row vector.
```

This simply captures any buffer overflow in `z`, and prepends the data to the subsequent input in the next call to `buffer`. Again, the input signal,

x, of frame size L, has been converted to a signal of frame size n without any insertion or deletion of samples.

Note that continuous buffering cannot be done with the single-output syntax `y = buffer(...)`, because the last frame of y in this case is zero padded, which adds new samples to the signal.

Continuous buffering in the presence of overlap and underlap is handled with the opt parameter, which is used as both an input and output to buffer. The following two examples demonstrate how the opt parameter should be used.

Examples

Example 1: Continuous Overlapping Buffers

First create a buffer containing 100 frames, each with 11 samples:

```
data = buffer(1:1100,11);      % 11 samples per frame
```

Imagine that the frames (columns) in the matrix called data are the sequential outputs of a data acquisition board sampling a physical signal: `data(:,1)` is the first D/A output, containing the first 11 signal samples; `data(:,2)` is the second output, containing the next 11 signal samples, and so on.

You want to rebuffer this signal from the acquired frame size of 11 to a frame size of 4 with an overlap of 1. To do this, you will repeatedly call buffer to operate on each successive input frame, using the opt parameter to maintain consistency in the overlap from one buffer to the next.

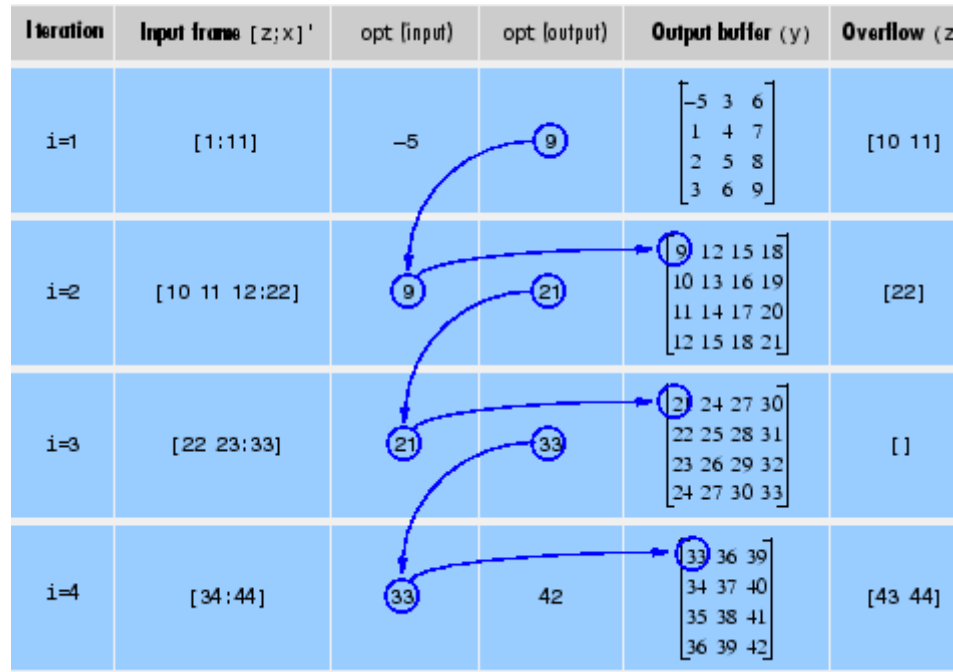
Set the buffer parameters:

```
n = 4;          % New frame size
p = 1;          % Overlap
opt = -5;       % Value of y(1)
z = [];         % Initialize the carry-over vector.
```

Now repeatedly call buffer, each time passing in a new signal frame from data. Note that overflow samples (returned in z) are carried over and prepended to the input in the subsequent call to buffer:

```
for i=1:size(data,2),       % Loop over each source
                            %    frame (column)
   x = data(:,i);           % Single frame of D/A output
   [y,z,opt] = buffer([z;x],n,p,opt);
   disp(y);                 % Display the buffer of data.
   pause
end
```

Here's what happens during the first four iterations.



Note that the size of the output matrix, `y`, can vary by a single column from one iteration to the next. This is typical for buffering operations with overlap or underlap.

Example 2: Continuous Underlapping Buffers

Again create a buffer containing 100 frames, each with 11 samples:

```
data = buffer(1:1100,11);  % 11 samples per frame
```

Again, imagine that `data(:,1)` is the first D/A output, containing the first 11 signal samples; `data(:,2)` is the second output, containing the next 11 signal samples, and so on.

You want to rebuffer this signal from the acquired frame size of 11 to a frame size of 4 with an underlap of 2. To do this, you will repeatedly call `buffer` to operate on each successive input frame, using the `opt` parameter to maintain consistency in the underlap from one buffer to the next.

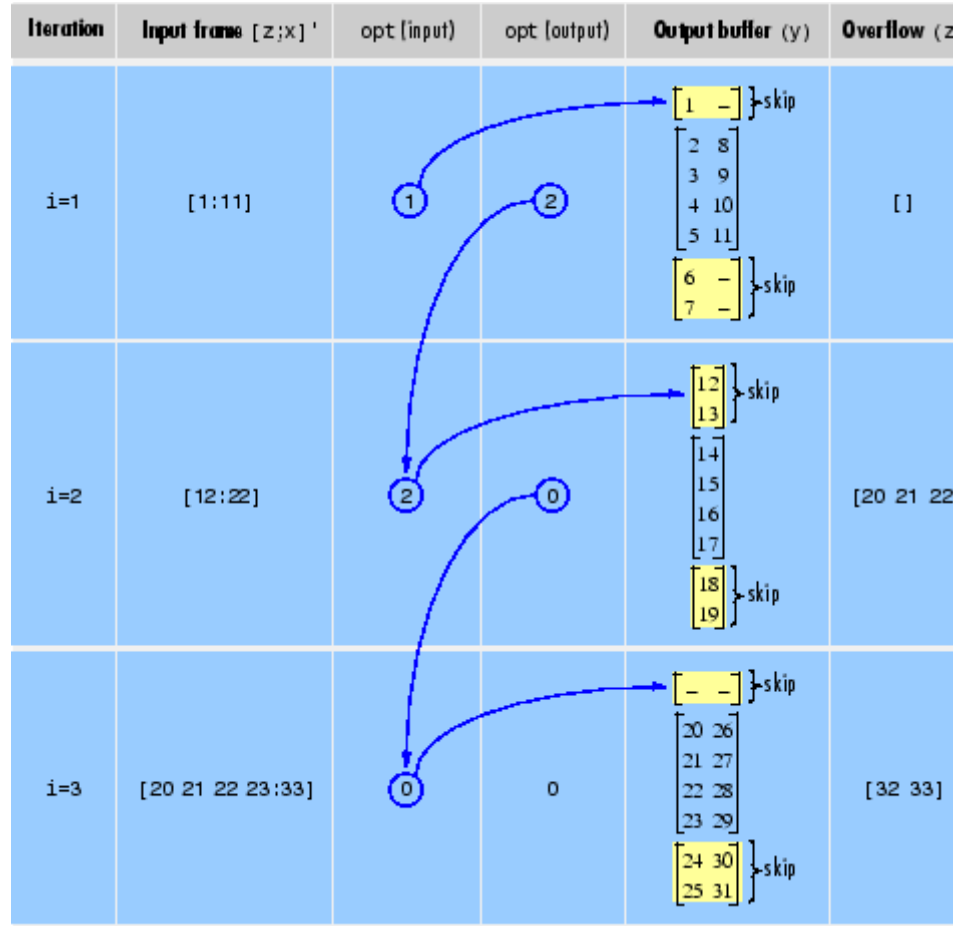
Set the buffer parameters:

```
n = 4;     % New frame size
p = -2;    % Underlap
opt = 1;   % Skip the first input element, x(1).
z = [];    % Initialize the carry-over vector.
```

Now repeatedly call `buffer`, each time passing in a new signal frame from `data`. Note that overflow samples (returned in `z`) are carried over and prepended to the input in the subsequent call to `buffer`:

```
for i=1:size(data,2),      % Loop over each source
                           %    frame (column)
   x = data(:,i);          % Single frame of D/A output
   [y,z,opt] = buffer([z;x],n,p,opt);
   disp(y);                % Display the buffer of data
   pause
end
```

Here's what happens during the first three iterations.



Diagnostics

Error messages are displayed when p ≥n or `length(opt)`≠`length(p)` in an overlapping buffer case:

```
Frame overlap P must be less than the buffer size N.
Initial conditions must be specified as a length-P vector.
```

See Also

`reshape`

buttap

Purpose

Butterworth filter prototype

Syntax

```
[z,p,k] = buttap(n)
```

Description

`[z,p,k] = buttap(n)` returns the poles and gain of an order `n` Butterworth analog lowpass filter prototype. The function returns the poles in the length `n` column vector `p` and the gain in scalar `k`. `z` is an empty matrix because there are no zeros. The transfer function is

$$H(s) = \frac{z(s)}{p(s)} = \frac{k}{(s - p(1))(s - p(2))\cdots(s - p(n))}$$

Butterworth filters are characterized by a magnitude response that is maximally flat in the passband and monotonic overall. In the lowpass case, the first 2`n`-1 derivatives of the squared magnitude response are zero at ω = 0. The squared magnitude response function is

$$|H(\omega)|^2 = \frac{1}{1 + (\omega / \omega_0)^{2n}}$$

corresponding to a transfer function with poles equally spaced around a circle in the left half plane. The magnitude response at the cutoff angular frequency ω_0 is always $1/\sqrt{2}$ regardless of the filter order. `buttap` sets ω_0 to 1 for a normalized result.

Algorithms

```
z = [];
p = exp(sqrt(-1)*(pi*(1:2:2*n-1)/(2*n)+pi/2)).';
k = real(prod(-p));
```

References

[1] Parks, T.W., and C.S. Burrus. *Digital Filter Design.* New York: John Wiley & Sons, 1987. Chapter 7.

See Also

`besselap` | `butter` | `cheb1ap` | `cheb2ap` | `ellipap`

Purpose

Butterworth filter design

Syntax

```
[z,p,k] = butter(n,Wn)
[z,p,k] = butter(n,Wn,'ftype')
[b,a] = butter(n,Wn)
[b,a] = butter(n,Wn,'ftype')
[A,B,C,D] = butter(n,Wn)
[A,B,C,D] = butter(n,Wn,'ftype')
[z,p,k] = butter(n,Wn,'s')
[z,p,k] = butter(n,Wn,'ftype','s')
[b,a] = butter(n,Wn,'s')
[b,a] = butter(n,Wn,'ftype','s')
[A,B,C,D] = butter(n,Wn,'s')
[A,B,C,D] = butter(n,Wn,'ftype','s')
```

Description

`butter` designs lowpass, bandpass, highpass, and bandstop digital and analog Butterworth filters. Butterworth filters are characterized by a magnitude response that is maximally flat in the passband and monotonic overall.

Butterworth filters sacrifice rolloff steepness for monotonicity in the pass- and stopbands. Unless the smoothness of the Butterworth filter is needed, an elliptic or Chebyshev filter can generally provide steeper rolloff characteristics with a lower filter order.

Digital Domain

`[z,p,k] = butter(n,Wn)` designs an order `n` lowpass digital Butterworth filter with normalized cutoff frequency `Wn`. It returns the zeros and poles in length `n` column vectors `z` and `p`, and the gain in the scalar `k`.

`[z,p,k] = butter(n,Wn,'ftype')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is one of the following:

- `'high'` for a highpass digital filter with normalized cutoff frequency `Wn`
- `'low'` for a lowpass digital filter with normalized cutoff frequency `Wn`

- `'stop'` for an order `2*n` bandstop digital filter if `Wn` is a two-element vector, `Wn = [w1 w2]`. The stopband is `w1` < ω < `w2`.

Cutoff frequency is that frequency where the magnitude response of the filter is $\sqrt{1/2}$. For `butter`, the normalized cutoff frequency `Wn` must be a number between 0 and 1, where 1 corresponds to the Nyquist frequency, π radians per sample.

If `Wn` is a two-element vector, `Wn = [w1 w2]`, `butter` returns an order `2*n` digital bandpass filter with passband `w1` < ω < `w2`.

With different numbers of output arguments, `butter` directly obtains other realizations of the filter. To obtain the transfer function form, use two output arguments as shown below.

Note See "Limitations" on page 1-57 below for information about numerical issues that affect forming the transfer function.

`[b,a] = butter(n,Wn)` designs an order `n` lowpass digital Butterworth filter with normalized cutoff frequency `Wn`. It returns the filter coefficients in length `n+1` row vectors `b` and `a`, with coefficients in descending powers of z.

$$H(z) = \frac{b(1) + b(2)z^{-1} + \ldots + b(n+1)z^{-n}}{1 + a(2)z^{-1} + \ldots + a(n+1)z^{-n}}$$

`[b,a] = butter(n,Wn,'ftype')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is `'high'`, `'low'`, or `'stop'`, as described above.

To obtain state-space form, use four output arguments as shown below:

`[A,B,C,D] = butter(n,Wn)` or

`[A,B,C,D] = butter(n,Wn,'ftype')` where `A`, `B`, `C`, and `D` are

$$x[n+1] = Ax[n] + Bu[n]$$
$$y[n] = Cx[n] + Du[n]$$

and u is the input, x is the state vector, and y is the output.

Analog Domain

`[z,p,k] = butter(n,Wn,'s')` designs an order `n` lowpass analog Butterworth filter with angular cutoff frequency `Wn` rad/s. It returns the zeros and poles in length `n` or `2*n` column vectors `z` and `p` and the gain in the scalar `k`. `butter`'s angular cutoff frequency `Wn` must be greater than 0 rad/s.

If `Wn` is a two-element vector with `w1 < w2`, `butter(n,Wn,'s')` returns an order `2*n` bandpass analog filter with passband `w1 < ω < w2`.

`[z,p,k] = butter(n,Wn,'ftype','s')` designs a highpass, lowpass, or bandstop filter using the *`ftype`* values described above.

With different numbers of output arguments, `butter` directly obtains other realizations of the analog filter. To obtain the transfer function form, use two output arguments as shown below:

`[b,a] = butter(n,Wn,'s')` designs an order `n` lowpass analog Butterworth filter with angular cutoff frequency `Wn` rad/s. It returns the filter coefficients in the length `n+1` row vectors `b` and `a`, in descending powers of *s,* derived from this transfer function:

$$H(s) = \frac{B(s)}{A(s)} = \frac{b(1)s^n + b(2)s^{n-1} + \ldots + b(n+1)}{s^n + a(2)s^{n-1} + \ldots + a(n+1)}$$

`[b,a] = butter(n,Wn,'ftype','s')` designs a highpass, lowpass, or bandstop filter using the *`ftype`* values described above.

To obtain state-space form, use four output arguments as shown below:

`[A,B,C,D] = butter(n,Wn,'s')` or

`[A,B,C,D] = butter(n,Wn,'ftype','s')` where `A`, `B`, `C`, and `D` are

$$x = Ax + Bu$$
$$y = Cx + Du$$

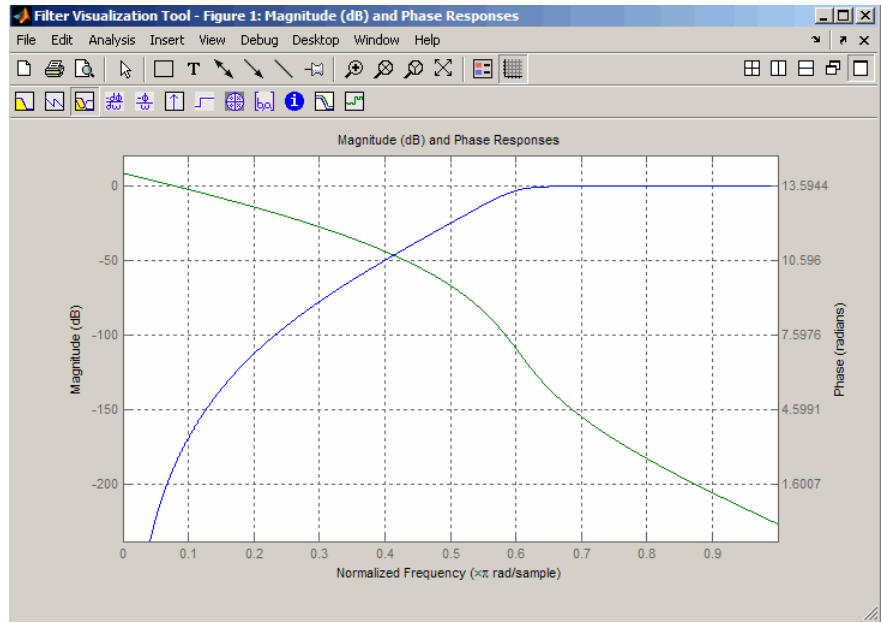
and u is the input, x is the state vector, and y is the output.

Examples

Highpass Filter

For data sampled at 1000 Hz, design a 9th-order highpass Butterworth filter with cutoff frequency of 300 Hz, which corresponds to a normalized value of 0.6:

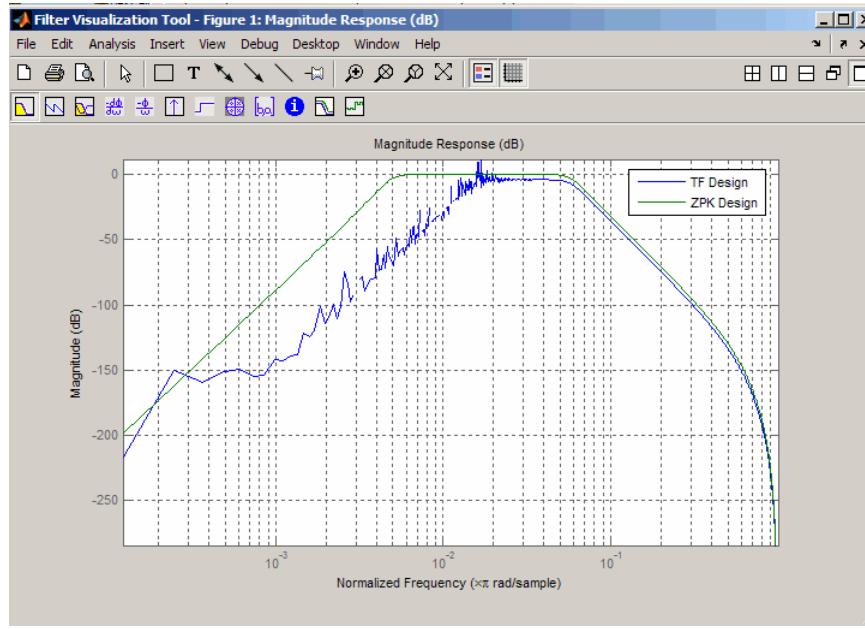
```
[z,p,k] = butter(9,300/500,'high');
[sos,g] = zp2sos(z,p,k);      % Convert to SOS form
Hd = dfilt.df2tsos(sos,g);   % Create a dfilt object
h = fvtool(Hd);              % Plot magnitude response
set(h,'Analysis','freq')      % Display frequency response
```



Limitations

In general, you should use the `[z,p,k]` syntax to design IIR filters. To analyze or implement your filter, you can then use the `[z,p,k]` output with `zp2sos` and an sos `dfilt` structure. For higher order filters (possibly starting as low as order 8), numerical problems due to roundoff errors may occur when forming the transfer function using the `[b,a]` syntax. The following example illustrates this limitation:

```
n = 6; Wn = [2.5e6 29e6]/500e6;
ftype = 'bandpass';

% Transfer Function design
[b,a] = butter(n,Wn,ftype);
h1=dfilt.df2(b,a);      % This is an unstable filter.

% Zero-Pole-Gain design
[z, p, k] = butter(n,Wn,ftype);
[sos,g]=zp2sos(z,p,k);
h2=dfilt.df2sos(sos,g);

% Plot and compare the results
hfvt=fvtool(h1,h2,'FrequencyScale','log');
legend(hfvt,'TF Design','ZPK Design')
```



Algorithms

`butter` uses a five-step algorithm:

1. It finds the lowpass analog prototype poles, zeros, and gain using the `buttap` function.
2. It converts the poles, zeros, and gain into state-space form.
3. It transforms the lowpass filter into a bandpass, highpass, or bandstop filter with desired cutoff frequencies, using a state-space transformation.
4. For digital filter design, `butter` uses `bilinear` to convert the analog filter into a digital filter through a bilinear transformation with frequency prewarping. Careful frequency adjustment guarantees that the analog filters and the digital filters will have the same frequency response magnitude at `Wn` or `w1` and `w2`.

5 It converts the state-space filter back to transfer function or zero-pole-gain form, as required.

See Also

`besself` | `buttap` | `buttord` | `cheby1` | `cheby2` | `ellip` | `maxflat`

buttord

Purpose

Butterworth filter order and cutoff frequency

Syntax

```
[n,Wn] = buttord(Wp,Ws,Rp,Rs)
[n,Wn] = buttord(Wp,Ws,Rp,Rs,'s')
```

Description

`buttord` calculates the minimum order of a digital or analog Butterworth filter required to meet a set of filter design specifications.

Digital Domain

`[n,Wn] = buttord(Wp,Ws,Rp,Rs)` returns the lowest order, `n`, of the digital Butterworth filter with no more than `Rp` dB of passband ripple and at least `Rs` dB of attenuation in the stopband. The scalar (or vector) of corresponding cutoff frequencies, `Wn`, is also returned. Use the output arguments `n` and `Wn` in `butter`.

Choose the input arguments to specify the stopband and passband according to the following table.

Description of Stopband and Passband Filter Parameters

Parameter	Description
`Wp`	Passband corner frequency `Wp`, the cutoff frequency, is a scalar or a two-element vector with values between 0 and 1, with 1 corresponding to the normalized Nyquist frequency, π radians per sample.
`Ws`	Stopband corner frequency `Ws`, is a scalar or a two-element vector with values between 0 and 1, with 1 corresponding to the normalized Nyquist frequency.
`Rp`	Passband ripple in decibels.
`Rs`	Stopband attenuation in decibels. This value is the number of decibels the stopband is down from the passband.

Use the following guide to specify filters of different types.

Filter Type Stopband and Passband Specifications

Filter Type	Stopband and Passband Conditions	Stopband	Passband
Lowpass	`Wp` < `Ws`, both scalars	`(Ws,1)`	`(0,Wp)`
Highpass	`Wp` > `Ws`, both scalars	`(0,Ws)`	`(Wp,1)`
Bandpass	The interval specified by `Ws` contains the one specified by `Wp` (`Ws(1) < Wp(1) < Wp(2) < Ws(2)`).	`(0,Ws(1))` and `(Ws(2),1)`	`(Wp(1),Wp(2))`
Bandstop	The interval specified by `Wp` contains the one specified by `Ws` (`Wp(1) < Ws(1) < Ws(2) < Wp(2)`).	`(Ws(1),Ws(2))`	`(0,Wp(1))` and `(Wp(2),1)`

If your filter specifications call for a bandpass or bandstop filter with unequal ripple in each of the passbands or stopbands, design separate lowpass and highpass filters according to the specifications in this table, and cascade the two filters together.

Analog Domain

`[n,Wn] = buttord(Wp,Ws,Rp,Rs,'s')` finds the minimum order `n` and cutoff frequencies `Wn` for an analog Butterworth filter. You specify the frequencies `Wp` and `Ws` similar those described in the Description of Stopband and Passband Filter Parameters on page 1-60 table above, only in this case you specify the frequency in radians per second, and the passband or the stopband can be infinite.

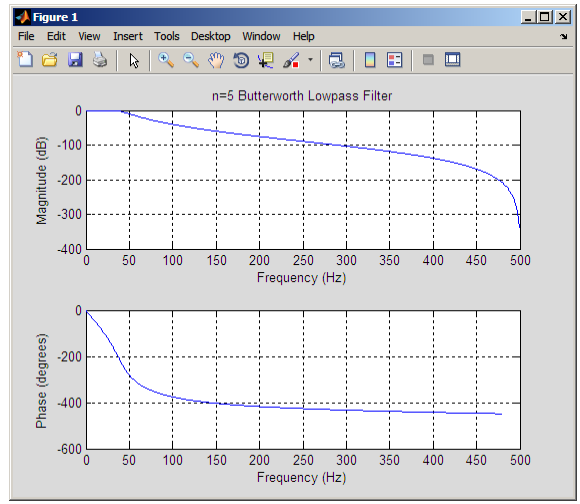
Use `buttord` for lowpass, highpass, bandpass, and bandstop filters as described in the Filter Type Stopband and Passband Specifications on page 1-61 table above.

Examples

Example 1

For data sampled at 1000 Hz, design a lowpass filter with no more than 3 dB of ripple in the passband from 0 to 40 Hz, and at least 60 dB of attenuation in the stopband. Plot the filter's frequency response.

```
Wp = 40/500; Ws = 150/500;
[n,Wn] = buttord(Wp,Ws,3,60);
% Returns n = 5; Wn=0.0810;
[b,a] = butter(n,Wn);
freqz(b,a,512,1000);
title('n=5 Butterworth Lowpass Filter')
```

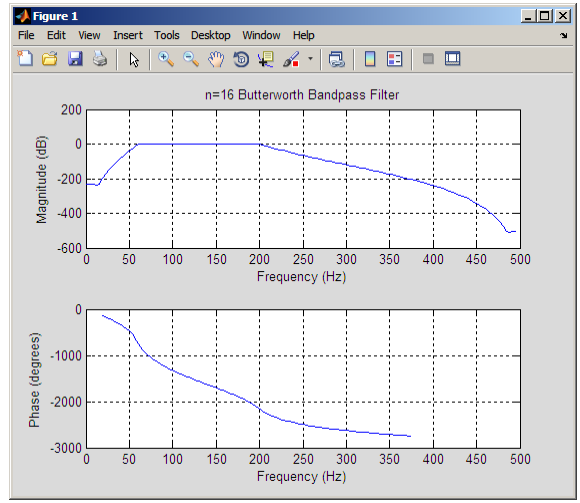


Example 2

Design a bandpass filter with a passband from 60 to 200 Hz with at most 3 dB of passband ripple and at least 40 dB attenuation in the stopbands that are 50 Hz wide on both sides of the passband:

```
Wp = [60 200]/500; Ws = [50 250]/500;
Rp = 3; Rs = 40;
[n,Wn] = buttord(Wp,Ws,Rp,Rs);
```

```
% Returns n =16; Wn =[0.1198 0.4005];
[b,a] = butter(n,Wn);
freqz(b,a,128,1000)
title('n=16 Butterworth Bandpass Filter')
```



Algorithms

`buttord`'s order prediction formula is described in [1]. It operates in the analog domain for both analog and digital cases. For the digital case, it converts the frequency parameters to the *s*-domain before estimating the order and natural frequency, and then converts back to the *z*-domain.

`buttord` initially develops a lowpass filter prototype by transforming the passband frequencies of the desired filter to 1 rad/s (for lowpass and highpass filters) and to -1 and 1 rad/s (for bandpass and bandstop filters). It then computes the minimum order required for a lowpass filter to meet the stopband specification.

References

[1] Rabiner, L.R., and B. Gold. *Theory and Application of Digital Signal Processing*. Englewood Cliffs, NJ: Prentice-Hall, 1975. Pg. 227.

See Also

`butter` | `cheb1ord` | `cheb2ord` | `ellipord` | `kaiserord`

cceps

Purpose

Complex cepstral analysis

Syntax

```
xhat = cceps(x)
[xhat,nd] = cceps(x)
[xhat,nd,xhat1] = cceps(x)
[...] = cceps(x,n)
```

Description

Cepstral analysis is a nonlinear signal processing technique that is applied most commonly in speech processing and homomorphic filtering [1].

Note `cceps` only works on real data.

`xhat = cceps(x)` returns the complex cepstrum of the real data sequence `x` using the Fourier transform. The input is altered, by the application of a linear phase term, to have no phase discontinuity at $\pm\pi$ radians. That is, it is circularly shifted (after zero padding) by some samples, if necessary, to have zero phase at π radians.

`[xhat,nd] = cceps(x)` returns the number of samples `nd` of (circular) delay added to `x` prior to finding the complex cepstrum.

`[xhat,nd,xhat1] = cceps(x)` returns a second complex cepstrum `xhat1` computed using an alternative factorization algorithm[1][2]. This method can be applied only to finite duration signals. See the Algorithm section below for a comparison of the Fourier and factorization methods of computing the complex cepstrum.

`[...] = cceps(x,n)` zero pads `x` to length `n` and returns the length `n` complex cepstrum of `x`.

Algorithms

`cceps` is an implementation of algorithm 7.1 in [3]. A lengthy Fortran program reduces to these three lines of MATLAB code, which compose the core of `cceps`:

```
h = fft(x);
```

```
logh = log(abs(h)) + sqrt(-1)*rcunwrap(angle(h));
y = real(ifft(logh));
```

Note `rcunwrap` in the above code segment is a special version of `unwrap` that subtracts a straight line from the phase. `rcunwrap` is a local function within `cceps` and is not available for use from the MATLAB command line.

The following table lists the pros and cons of the Fourier and factorization algorithms.

Algorithm	Pros	Cons
Fourier	Can be used for any signal.	Requires phase unwrapping. Output is aliased.
Factorization	Does not require phase unwrapping. No aliasing	Can be used only for short duration signals. Input signal must have an all-zero Z-transform with no zeros on the unit circle.

In general, you cannot use the results of these two algorithms to verify each other. You can use them to verify each other only when the first element of the input data is positive, the Z-transform of the data sequence has only zeros, all of these zeros are inside the unit circle, and the input data sequence is long (or padded with zeros).

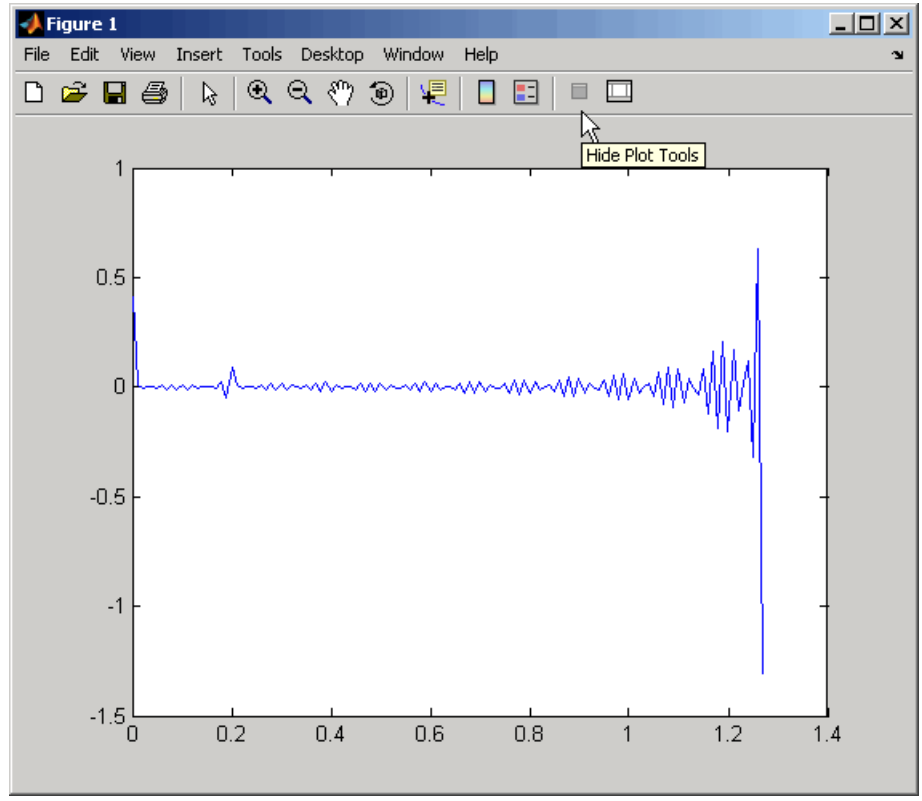
Examples

The following example uses `cceps` to show an echo.

```
Fs = 100;
t = 0:1/Fs:1.27;

% 45Hz sine sampled at 100Hz
s1 = sin(2*pi*45*t);
```

```
% Add an echo with half the amplitude and 0.2 second later
s2 = s1 + 0.5*[zeros(1,20) s1(1:108)];

c = cceps(s2);
plot(t,c)
```

Notice the echo at 0.2 second.



References

[1] Oppenheim, A.V., and R.W. Schafer. *Discrete-Time Signal Processing*. Upper Saddle River, NJ: Prentice-Hall, 1999, pp. 788-789.

[2] Steiglitz, K., and B. Dickinson. “Computation of the complex cepstrum by factorization of the Z-transform” in *Proc. Int. Conf. ASSP*. 1977, pp. 723–726.

[3] *IEEE Programs for Digital Signal Processing*. IEEE Press. New York: John Wiley & Sons, 1979.

See Also

`icceps` | `hilbert` | `rceps` | `unwrap`

Purpose

Modulo-N circular convolution

Syntax

```
c = cconv(a,b,n)
c = cconv(gpuArrayA,gpuArrayB,n)
```

Description

Circular convolution is used to convolve two discrete Fourier transform (DFT) sequences. For very long sequences, circular convolution may be faster than linear convolution.

`c = cconv(a,b,n)` circularly convolves vectors `a` and `b`. `n` is the length of the resulting vector. If you omit `n`, it defaults to `length(a)+length(B)-1`. When `n = length(a)+length(B)-1`, the circular convolution is equivalent to the linear convolution computed with `conv`. You can also use `cconv` to compute the circular cross-correlation of two sequences (see the example below).

`c = cconv(gpuArrayA,gpuArrayB,n)` returns the circular convolution of the input vectors of class `gpuArray`. See "Use gpuArray Data" for details on gpuArray objects. Using `cconv` with `gpuArray` objects requires Parallel Computing Toolbox™ software and a CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details. The output vector, `c`, is a gpuArray object. See "Circular Convolution using the GPU" on page 1-69 for an example of using the GPU to compute the circular convolution.

Examples

The following example calculates a modulo-4 circular convolution.

```
a = [2 1 2 1];
b = [1 2 3 4];
c = cconv(a,b,4)
% Returns
% c =
%    14    16    14    16
```

The following example compares a circular correlation, where `n` uses the default value, and a linear convolution. The resulting norm is a value

that is virtually zero, which shows that the two convolutions produce virtually the same result.

```
a = [1 2 -1 1];
b = [1 1 2 1 2 2 1 1];
c = cconv(a,b)      % Circular convolution
cref = conv(a,b)    % Linear convolution
norm(c-cref)
```

The following example uses `cconv` to compute the circular cross-correlation of two sequences. The result is compared to the cross-correlation computed using `xcorr`.

```
a = [1 2 2 1]+1i;
b = [1 3 4 1]-2*1i;
c = cconv(a,conj(fliplr(b)),7); % Compute using cconv
cref = xcorr(a,b);      % Compute using xcorr
norm(c-cref)
```

Circular Convolution using the GPU

The following example requires Parallel Computing Toolbox software and a CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details.

Create two signals consisting of a 1 kHz sine wave in additive white Gaussian noise. The sampling rate is 10 kHz

```
Fs = 1e4;
t = 0:1/Fs:10-(1/Fs);
x = cos(2*pi*1e3*t)+randn(size(t));
y = sin(2*pi*1e3*t)+randn(size(t));
```

Put `x` and `y` on the GPU using `gpuArray`. Obtain the circular convolution using the GPU.

```
x = gpuArray(x);
y = gpuArray(y);
cirC = cconv(x,y,length(x)+length(y)-1);
```

Compare the result to the linear convolution of `x` and `y`.

```
linC = conv(x,y);
norm(linC-cirC,2)
```

Return the circular convolution, `cirC`, to the MATLAB workspace using `gather`.

```
cirC = gather(cirC);
```

References

[1] Orfanidis, S.J., *Introduction to Signal Processing*, Englewood Cliffs, NJ: Prentice-Hall, Inc., 1996. pp. 524–529.

See Also

`conv`

Purpose

Convert second-order sections cell array to matrix

Syntax

```
m = cell2sos(c)
```

Description

`m = cell2sos(c)` changes a 1-by-*L* cell array `c` consisting of 1-by-2 cell arrays into an *L*-by-6 second-order section matrix m. Matrix `m` takes the same form as the matrix generated by `tf2sos`. You can use m = cell2sos(c) to invert the results of `c = sos2cell(m)`.

`c` must be a cell array of the form

```
c = { {b1 a1} {b2 a2} ... {bL aL} }
```

where both `bi` and `ai` are row vectors of at most length 3, and *i* = 1, 2, ..., *L*. The resulting matrix `m` is given by

```
m = [b1 a1;b2 a2; ... ;bL aL]
```

See Also

`sos2cell` | `tf2sos`

cfirpm

Purpose

Complex and nonlinear-phase equiripple FIR filter design

Syntax

```
b = cfirpm(n,f,@fresp)
b = cfirpm(n,f,@fresp,w)
b = cfirpm(n,f,a)
b = cfirpm(n,f,a,w)
b = cfirpm(...,'sym')
b = cfirpm(...,'skip_stage2')
b = cfirpm(..., 'debug')
b = cfirpm(...,{lgrid})
[b,delta] = cfirpm(...)
[b,delta,opt] = cfirpm(...)
```

Description

`cfirpm` allows arbitrary frequency-domain constraints to be specified for the design of a possibly complex FIR filter. The Chebyshev (or minimax) filter error is optimized, producing equiripple FIR filter designs.

`b = cfirpm(n,f,@fresp)` returns a length `n+1` FIR filter with the best approximation to the desired frequency response as returned by function *`fresp`*, which is called by its function handle (`@fresp`). `f` is a vector of frequency band edge pairs, specified in the range -1 and 1, where 1 corresponds to the normalized Nyquist frequency. The frequencies must be in increasing order, and `f` must have even length. 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.

Predefined `fresp` frequency response functions are included for a number of common filter designs, as described below. For all of the predefined frequency response functions, the symmetry option `'sym'` defaults to `'even'` if no negative frequencies are contained in `f` and `d = 0`; otherwise `'sym'` defaults to `'none'`. (See the `'sym'` option below for details.) For all of the predefined frequency response functions, `d` specifies a group-delay offset such that the filter response has a group delay of `n/2+d` in units of the sample interval. Negative values create less delay; positive values create more delay. By default `d = 0`:

- `@lowpass`, `@highpass`, `@allpass`, `@bandpass`, `@bandstop`

These functions share a common syntax, exemplified below by the string `'lowpass'`.

`b = cfirpm(n,f,@lowpass,...)` and

`b = cfirpm(n,f,{@lowpass,d},...)` design a linear-phase (`n/2+d` delay) filter.

Note For `@bandpass` filters, the first element in the frequency vector must be less than or equal to zero and the last element must be greater than or equal to zero.

- `@multiband` designs a linear-phase frequency response filter with arbitrary band amplitudes.

  `b = cfirpm(n,f,{@multiband,a},...)` and

  `b = cfirpm(n,f,{@multiband,a,d},...)` specify vector `a` containing the desired amplitudes at the band edges in `f`. The desired amplitude at frequencies between pairs of points `f(k)` and `f(k+1)` for `k` odd is the line segment connecting the points `(f(k),a(k))` and `(f(k+1),a(k+1))`.

- `@differentiator` designs a linear-phase differentiator. For these designs, zero-frequency must be in a transition band, and band weighting is set to be inversely proportional to frequency.

  `b = cfirpm(n,f,{@differentiator,fs},...)` and

  `b = cfirpm(n,f,{@differentiator,fs,d},...)` specify the sample rate `fs` used to determine the slope of the differentiator response. If omitted, `fs` defaults to 1.

- `@hilbfilt` designs a linear-phase Hilbert transform filter response. For Hilbert designs, zero-frequency must be in a transition band.

  `b = cfirpm(n,f,@hilbfilt,...)` and

  `b = cfirpm(N,F,{@hilbfilt,d},...)` design a linear-phase (`n/2+d` delay) Hilbert transform filter.

- `@invsinc` designs a linear-phase inverse-sinc filter response.

  `b = cfirpm(n,f,{@invsinc,a},...)` and

  `b = cfirpm(n,f,{@invsinc,a,d},...)` specify gain `a` for the sinc-function, computed as sinc(a*g), where g contains the optimization grid frequencies normalized to the range [-1,1]. By default, `a`=1. The group-delay offset is `d`, such that the filter response will have a group delay of N/2 + `d` in units of the sample interval, where N is the filter order. Negative values create less delay and positive values create more delay. By default, `d`=0.

`b = cfirpm(n,f,@fresp,w)` uses the real, non-negative weights in vector `w` to weight the fit in each frequency band. The length of `w` is half the length of `f`, so there is exactly one weight per band.

`b = cfirpm(n,f,a)` is a synonym for
`b = cfirpm(n,f,{@multiband,a})`.

`b = cfirpm(n,f,a,w)` applies an optional set of positive weights, one per band, for use during optimization. If `w` is not specified, the weights are set to unity.

`b = cfirpm(...,'sym')` imposes a symmetry constraint on the impulse response of the design, where `'sym'` may be one of the following:

- `'none'` indicates no symmetry constraint. This is the default if any negative band edge frequencies are passed, or if *`fresp`* does not supply a default.
- `'even'` indicates a real and even impulse response. This is the default for highpass, lowpass, allpass, bandpass, bandstop, invsinc, and multiband designs.
- `'odd'` indicates a real and odd impulse response. This is the default for Hilbert and differentiator designs.
- `'real'` indicates conjugate symmetry for the frequency response

If any `'sym'` option other than `'none'` is specified, the band edges should be specified only over positive frequencies; the negative frequency region is filled in from symmetry. If a `'sym'` option is not

specified, the *fresp* function is queried for a default setting. Any user-supplied *fresp* function should return a valid '*sym*' string when it is passed the string 'defaults' as the filter order N.

b = cfirpm(...,'skip_stage2') disables the second-stage optimization algorithm, which executes only when cfirpm determines that an optimal solution has not been reached by the standard firpm error-exchange. Disabling this algorithm may increase the speed of computation, but may incur a reduction in accuracy. By default, the second-stage optimization is enabled.

b = cfirpm(..., '*debug*') enables the display of intermediate results during the filter design, where '*debug*' may be one of 'trace', 'plots', 'both', or 'off'. By default it is set to 'off'.

b = cfirpm(...,{lgrid}) uses the integer lgrid to control the density of the frequency grid, which has roughly 2^nextpow2(lgrid*n) frequency points. The default value for lgrid is 25. Note that the {lgrid} argument must be a 1-by-1 cell array.

Any combination of the '*sym*', 'skip_stage2', '*debug*', and {lgrid} options may be specified.

[b,delta] = cfirpm(...) returns the maximum ripple height delta.

[b,delta,opt] = cfirpm(...) returns a structure opt of optional results computed by cfirpm and contains the following fields.

Field	Description
opt.fgrid	Frequency grid vector used for the filter design optimization
opt.des	Desired frequency response for each point in opt.fgrid
opt.wt	Weighting for each point in opt.fgrid
opt.H	Actual frequency response for each point in opt.fgrid
opt.error	Error at each point in opt.fgrid

Field	Description
`opt.iextr`	Vector of indices into `opt.fgrid` for extremal frequencies
`opt.fextr`	Vector of extremal frequencies

User-definable functions may be used, instead of the predefined frequency response functions for @*`fresp`*. The function is called from within `cfirpm` using the following syntax

```
[dh,dw] = fresp(n,f,gf,w,p1,p2,...)
```

where:

- `n` is the filter order.
- `f` is the vector of frequency band edges that appear monotonically between -1 and 1, where 1 corresponds to the Nyquist frequency.
- `gf` is a vector of grid points that have been linearly interpolated over each specified frequency band by `cfirpm`. `gf` determines the frequency grid at which the response function must be evaluated. This is the same data returned by `cfirpm` in the `fgrid` field of the `opt` structure.
- `w` is a vector of real, positive weights, one per band, used during optimization. `w` is optional in the call to `cfirpm`; if not specified, it is set to unity weighting before being passed to *`fresp`*.
- `dh` and `dw` are the desired complex frequency response and band weight vectors, respectively, evaluated at each frequency in grid `gf`.
- `p1`, `p2`, `...`, are optional parameters that may be passed to *`fresp`*.

Additionally, a preliminary call is made to *`fresp`* to determine the default symmetry property '*`sym`*'. This call is made using the syntax:

```
sym = fresp('defaults',{n,f,[],w,p1,p2,...})
```

The arguments may be used in determining an appropriate symmetry default as necessary. The function `private/lowpass.m` may be useful as a template for generating new frequency response functions.

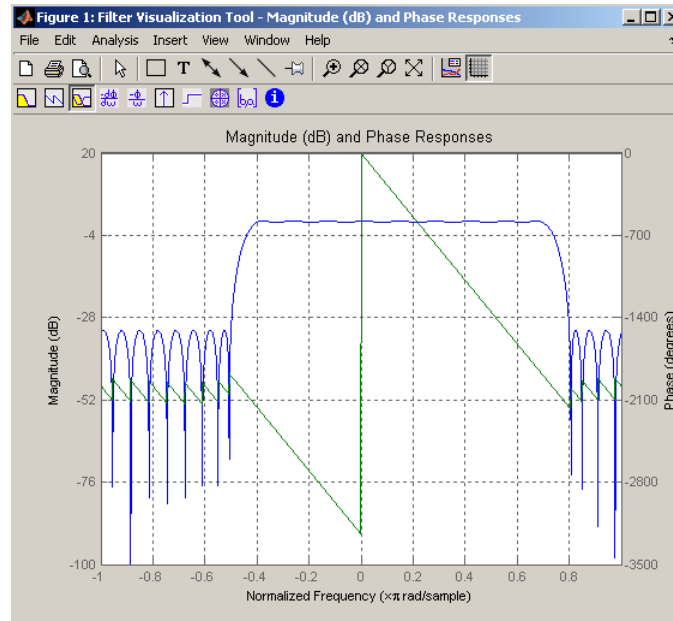
Examples

Example 1

Design a 31-tap, linear-phase, lowpass filter:

```
b = cfirpm(30,[-1 -0.5 -0.4 0.7 0.8 1],@lowpass);
fvtool(b,1)
```

Click the **Magnitude and Phase Response** button.



Example 2

Design a nonlinear-phase allpass FIR filter:

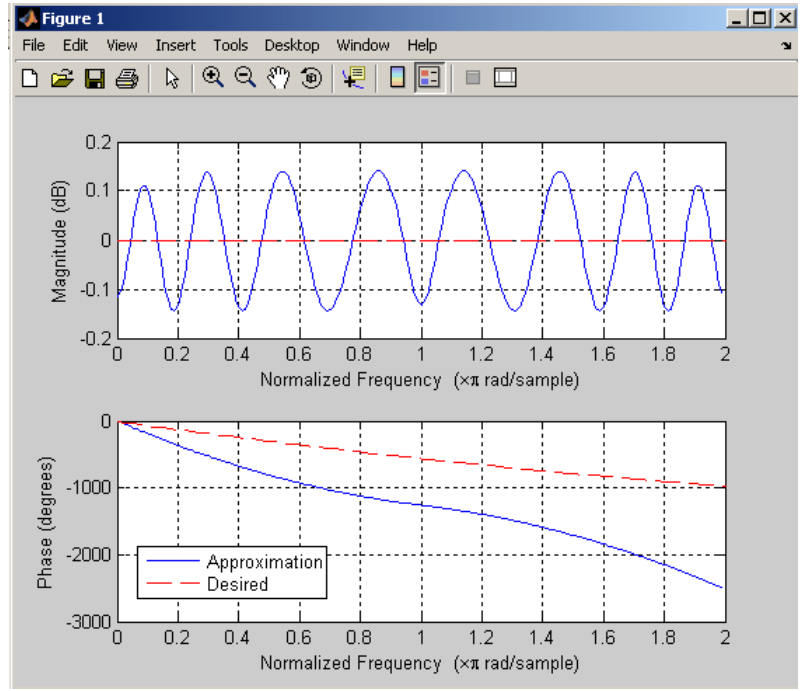
```
n = 22;                          % Filter order
```

```
f = [-1 1];                % Frequency band edges
w = [1 1];                 % Weights for optimization
gf = linspace(-1,1,256);    % Grid of frequency points
d = exp(-1i*pi*gf*n/2 + 1i*pi*pi*sign(gf).*gf.*gf*(4/pi));
               % Desired frequency response
```

Vector `d` now contains the complex frequency response that we desire for the FIR filter computed by `cfirpm`.

Now compute the FIR filter that best approximates this response:

```
b = cfirpm(n,f,'allpass',w,'real');  % Approximation
freqz(b,1,256,'whole');
subplot(2,1,1); hold on              % Overlay response
plot(pi*(gf+1),20*log10(abs(fftshift(d))),'r--')
subplot(2,1,2); hold on
plot(pi*(gf+1),unwrap(angle(fftshift(d)))*180/pi,'r--')
legend('Approximation','Desired')
```



Algorithms

An extended version of the Remez exchange method is implemented for the complex case. This exchange method obtains the optimal filter when the equiripple nature of the filter is restricted to have `n+2` extremals. When it does not converge, the algorithm switches to an ascent-descent algorithm that takes over to finish the convergence to the optimal solution. See the references for further details.

References

[1] Karam, L.J., and J.H. McClellan. "Complex Chebyshev Approximation for FIR Filter Design." *IEEE Trans. on Circuits and Systems II,*March 1995. Pgs. 207-216.

[2] Karam, L.J. *Design of Complex Digital FIR Filters in the Chebyshev Sense,* Ph.D. Thesis, Georgia Institute of Technology, March 1995.

[3] Demjanjov, V.F., and V.N. Malozemov. *Introduction to Minimax,* New York: John Wiley & Sons, 1974.

See Also

`fir1` | `fir2` | `firls` | `firpm` | `function_handle`

Purpose

Chebyshev Type I analog lowpass filter prototype

Syntax

```
[z,p,k] = cheb1ap(n,Rp)
```

Description

`[z,p,k] = cheb1ap(n,Rp)` returns the poles and gain of an order `n` Chebyshev Type I analog lowpass filter prototype with `Rp` dB of ripple in the passband. The function returns the poles in the length `n` column vector `p` and the gain in scalar `k`. `z` is an empty matrix, because there are no zeros. The transfer function is

$$H(s) = \frac{z(s)}{p(s)} = \frac{k}{(s - p(1))(s - p(2))\ldots(s - p(n))}$$

Chebyshev Type I filters are equiripple in the passband and monotonic in the stopband. The poles are evenly spaced about an ellipse in the left half plane. The Chebyshev Type I passband edge angular frequency ω_0 is set to 1.0 for a normalized result. This is the frequency at which the passband ends and the filter has magnitude response of $10^{-Rp/20}$.

References

[1] Parks, T.W., and C.S. Burrus. *Digital Filter Design,*New York: John Wiley & Sons, 1987. Chapter 7.

See Also

`besselap` | `buttap` | `cheby1` | `cheb2ap` | `ellipap`

Purpose

Chebyshev Type I filter order

Syntax

```
[n,Wp] = cheb1ord(Wp,Ws,Rp,Rs)
[n,Wp] = cheb1ord(Wp,Ws,Rp,Rs,'s')
```

Description

`cheb1ord` calculates the minimum order of a digital or analog Chebyshev Type I filter required to meet a set of filter design specifications.

Digital Domain

`[n,Wp] = cheb1ord(Wp,Ws,Rp,Rs)` returns the lowest order `n` of the Chebyshev Type I filter that loses no more than `Rp` dB in the passband and has at least `Rs` dB of attenuation in the stopband. The scalar (or vector) of corresponding cutoff frequencies `Wp`, is also returned. Use the output arguments `n` and `Wp` with the `cheby1` function.

Choose the input arguments to specify the stopband and passband according to the following table.

Description of Stopband and Passband Filter Parameters

Parameter	Description
`Wp`	Passband corner frequency `Wp`, the cutoff frequency, is a scalar or a two-element vector with values between 0 and 1, with 1 corresponding to the normalized Nyquist frequency, π radians per sample.
`Ws`	Stopband corner frequency `Ws`, is a scalar or a two-element vector with values between 0 and 1, with 1 corresponding to the normalized Nyquist frequency.
`Rp`	Passband ripple, in decibels. This value is the maximum permissible passband loss in decibels.
`Rs`	Stopband attenuation, in decibels. This value is the number of decibels the stopband is down from the passband.

Use the following guide to specify filters of different types.

Filter Type Stopband and Passband Specifications

Filter Type	Stopband and Passband Conditions	Stopband	Passband
Lowpass	`Wp` < `Ws`, both scalars	`(Ws,1)`	`(0,Wp)`
Highpass	`Wp` > `Ws`, both scalars	`(0,Ws)`	`(Wp,1)`
Bandpass	The interval specified by `Ws` contains the one specified by `Wp` (`Ws(1) < Wp(1) < Wp(2) < Ws(2)`).	`(0,Ws(1))` and `(Ws(2),1)`	`(Wp(1),Wp(2))`
Bandstop	The interval specified by `Wp` contains the one specified by `Ws` (`Wp(1) < Ws(1) < Ws(2) < Wp(2)`).	`(0,Wp(1))` and `(Wp(2),1)`	`(Ws(1),Ws(2))`

If your filter specifications call for a bandpass or bandstop filter with unequal ripple in each of the passbands or stopbands, design separate lowpass and highpass filters according to the specifications in this table, and cascade the two filters together.

Analog Domain

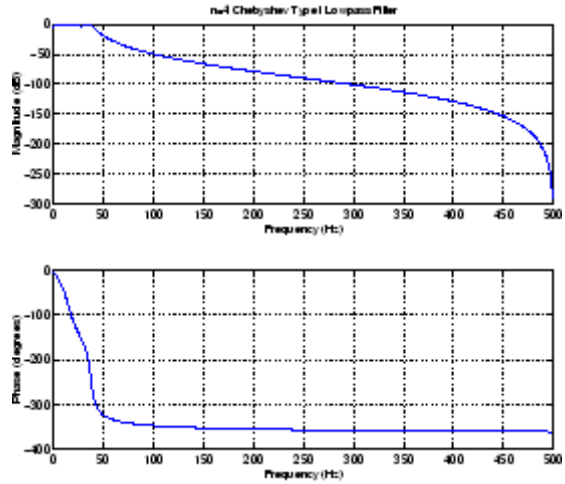
`[n,Wp] = cheb1ord(Wp,Ws,Rp,Rs,'s')` finds the minimum order `n` and cutoff frequencies `Wp` for an analog Chebyshev Type I filter. You specify the frequencies `Wp` and `Ws` similar to those described in the Description of Stopband and Passband Filter Parameters on page 1-82 table above, only in this case you specify the frequency in radians per second, and the passband or the stopband can be infinite.

Use `cheb1ord` for lowpass, highpass, bandpass, and bandstop filters as described in the Filter Type Stopband and Passband Specifications on page 1-83 table above.

Examples

For data sampled at 1000 Hz, design a lowpass filter with less than 3 dB of ripple in the passband defined from 0 to 40 Hz and at least 60 dB of ripple in the stopband defined from 150 Hz to the Nyquist frequency (500 Hz):

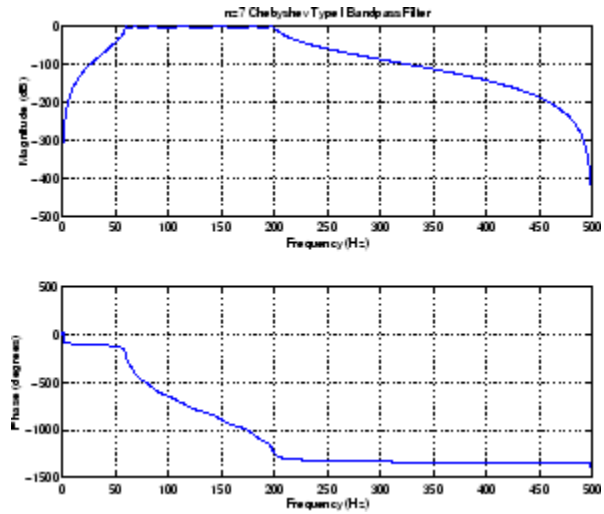
```
Wp = 40/500; Ws = 150/500;
Rp = 3; Rs = 60;
[n,Wp] = cheb1ord(Wp,Ws,Rp,Rs)
% Returns n = 4 Wp =0.0800
[b,a] = cheby1(n,Rp,Wp);
freqz(b,a,512,1000);
title('n=4 Chebyshev Type I Lowpass Filter')
```



Next design a bandpass filter with a passband of 60 Hz to 200 Hz, with less than 3 dB of ripple in the passband, and 40 dB attenuation in the stopbands that are 50 Hz wide on both sides of the passband:

```
Wp = [60 200]/500; Ws = [50 250]/500;
Rp = 3; Rs = 40;
[n,Wp] = cheb1ord(Wp,Ws,Rp,Rs)
% Returns n =7 Wp =[0.1200    0.4000]
[b,a] = cheby1(n,Rp,Wp);
```

```
freqz(b,a,512,1000);
title('n=7 Chebyshev Type I Bandpass Filter')
```



Algorithms

`cheb1ord` uses the Chebyshev lowpass filter order prediction formula described in [1]. The function performs its calculations in the analog domain for both analog and digital cases. For the digital case, it converts the frequency parameters to the *s*-domain before the order and natural frequency estimation process, and then converts them back to the *z*-domain.

`cheb1ord` initially develops a lowpass filter prototype by transforming the passband frequencies of the desired filter to 1 rad/s (for low- or highpass filters) or to -1 and 1 rad/s (for bandpass or bandstop filters). It then computes the minimum order required for a lowpass filter to meet the stopband specification.

References

[1] Rabiner, L.R., and B. Gold. *Theory and Application of Digital Signal Processing,* Englewood Cliffs, NJ: Prentice-Hall, 1975. Pg. 241.

See Also

`buttord` | `cheby1` | `cheb2ord` | `ellipord` | `kaiserord`

cheb2ap

Purpose

Chebyshev Type II analog lowpass filter prototype

Syntax

```
[z,p,k] = cheb2ap(n,Rs)
```

Description

`[z,p,k] = cheb2ap(n,Rs)` finds the zeros, poles, and gain of an order `n` Chebyshev Type II analog lowpass filter prototype with stopband ripple `Rs` dB down from the passband peak value. `cheb2ap` returns the zeros and poles in length `n` column vectors `z` and `p` and the gain in scalar `k`. If `n` is odd, `z` is length `n-1`. The transfer function is

$$H(s) = \frac{z(s)}{p(s)} = k\frac{(s-z(1))(s-z(2))\cdots(s-z(n))}{(s-p(1))(s-p(2))\cdots(s-p(n))}$$

Chebyshev Type II filters are monotonic in the passband and equiripple in the stopband. The pole locations are the inverse of the pole locations of `cheb1ap`, whose poles are evenly spaced about an ellipse in the left half plane. The Chebyshev Type II stopband edge angular frequency ω_0 is set to 1 for a normalized result. This is the frequency at which the stopband begins and the filter has magnitude response of $10^{-Rs/20}$.

Algorithms

Chebyshev Type II filters are sometimes called *inverse Chebyshev* filters because of their relationship to Chebyshev Type I filters. The `cheb2ap` function is a modification of the Chebyshev Type I prototype algorithm:

1. `cheb2ap` replaces the frequency variable ω with $1/\omega$, turning the lowpass filter into a highpass filter while preserving the performance at $\omega = 1$.
2. `cheb2ap` subtracts the filter transfer function from unity.

References

[1] Parks, T.W., and C.S. Burrus. *Digital Filter Design,* New York: John Wiley & Sons, 1987. Chapter 7.

See Also

`besselap` | `buttap` | `cheb1ap` | `cheby2` | `ellipap`

Purpose

Chebyshev Type II filter order

Syntax

```
[n,Ws] = cheb2ord(Wp,Ws,Rp,Rs)
[n,Ws] = cheb2ord(Wp,Ws,Rp,Rs,'s')
```

Description

`cheb2ord` calculates the minimum order of a digital or analog Chebyshev Type II filter required to meet a set of filter design specifications.

Digital Domain

`[n,Ws] = cheb2ord(Wp,Ws,Rp,Rs)` returns the lowest order `n` of the Chebyshev Type II filter that loses no more than `Rp` dB in the passband and has at least `Rs` dB of attenuation in the stopband. The scalar (or vector) of corresponding cutoff frequencies `Ws`, is also returned. Use the output arguments `n` and `Ws` in `cheby2`.

Choose the input arguments to specify the stopband and passband according to the following table.

Description of Stopband and Passband Filter Parameters

Parameter	Description
`Wp`	Passband corner frequency `Wp`, the cutoff frequency, is a scalar or a two-element vector with values between 0 and 1, with 1 corresponding to the normalized Nyquist frequency, π radians per sample.
`Ws`	Stopband corner frequency `Ws`, is a scalar or a two-element vector with values between 0 and 1, with 1 corresponding to the normalized Nyquist frequency.
`Rp`	Passband ripple, in decibels. This value is the maximum permissible passband loss in decibels.
`Rs`	Stopband attenuation, in decibels. This value is the number of decibels the stopband is down from the passband.

Use the following guide to specify filters of different types.

Filter Type Stopband and Passband Specifications

Filter Type	Stopband and Passband Conditions	Stopband	Passband
Lowpass	`Wp` < `Ws`, both scalars	`(Ws,1)`	`(0,Wp)`
Highpass	`Wp` > `Ws`, both scalars	`(0,Ws)`	`(Wp,1)`
Bandpass	The interval specified by `Ws` contains the one specified by `Wp` (`Ws(1) < Wp(1) < Wp(2) < Ws(2)`).	`(0,Ws(1))` and `(Ws(2),1)`	`(Wp(1),Wp(2))`
Bandstop	The interval specified by `Wp` contains the one specified by `Ws` (`Wp(1) < Ws(1) < Ws(2) < Wp(2)`).	`(0,Wp(1))` and `(Wp(2),1)`	`(Ws(1),Ws(2))`

If your filter specifications call for a bandpass or bandstop filter with unequal ripple in each of the passbands or stopbands, design separate lowpass and highpass filters according to the specifications in this table, and cascade the two filters together.

Analog Domain

`[n,Ws] = cheb2ord(Wp,Ws,Rp,Rs,'s')` finds the minimum order `n` and cutoff frequencies `Ws` for an analog Chebyshev Type II filter. You specify the frequencies `Wp` and `Ws` similar to those described in the Description of Stopband and Passband Filter Parameters on page 1-87 table above, only in this case you specify the frequency in radians per second, and the passband or the stopband can be infinite.

Use `cheb2ord` for lowpass, highpass, bandpass, and bandstop filters as described in the Filter Type Stopband and Passband Specifications on page 1-88 table above.

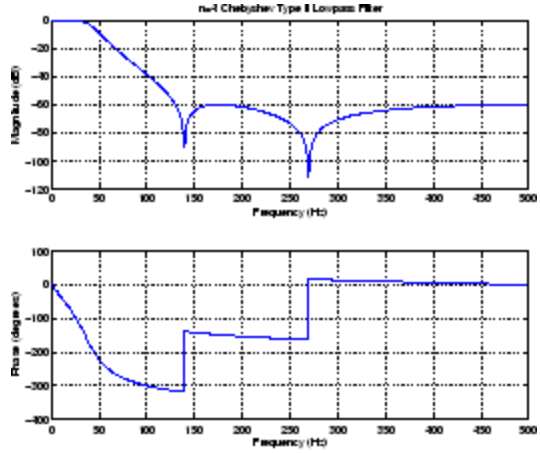
Examples

Example 1

For data sampled at 1000 Hz, design a lowpass filter with less than 3 dB of ripple in the passband defined from 0 to 40 Hz, and at least 60 dB

of attenuation in the stopband defined from 150 Hz to the Nyquist frequency (500 Hz):

```
Wp = 40/500; Ws = 150/500;
Rp = 3; Rs = 60;
[n,Ws] = cheb2ord(Wp,Ws,Rp,Rs)
% Returns n =4 Ws =0.3000
[b,a] = cheby2(n,Rs,Ws);
freqz(b,a,512,1000);
title('n=4 Chebyshev Type II Lowpass Filter')
```

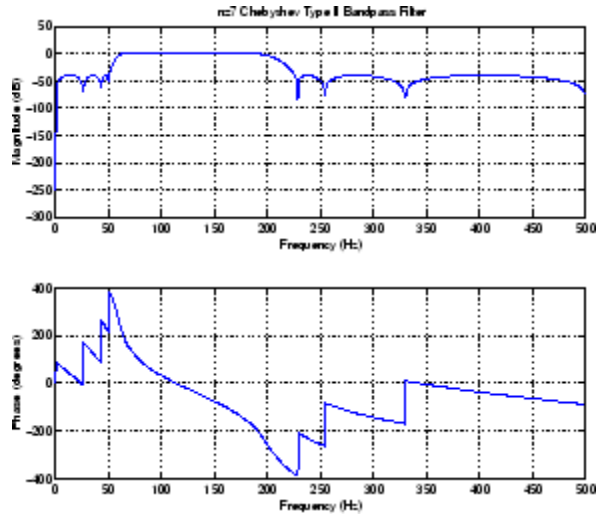


Example 2

Next design a bandpass filter with a passband of 60 Hz to 200 Hz, with less than 3 dB of ripple in the passband, and 40 dB attenuation in the stopbands that are 50 Hz wide on both sides of the passband:

```
Wp = [60 200]/500; Ws = [50 250]/500;
Rp = 3; Rs = 40;
[n,Ws] = cheb2ord(Wp,Ws,Rp,Rs)
% Returns n =7 Ws =[0.1000 0.5000]
[b,a] = cheby2(n,Rs,Ws);
freqz(b,a,512,1000)
```

```
title('n=7 Chebyshev Type II Bandpass Filter')
```



Algorithms

`cheb2ord` uses the Chebyshev lowpass filter order prediction formula described in [1]. The function performs its calculations in the analog domain for both analog and digital cases. For the digital case, it converts the frequency parameters to the *s*-domain before the order and natural frequency estimation process, and then converts them back to the *z*-domain.

`cheb2ord` initially develops a lowpass filter prototype by transforming the stopband frequencies of the desired filter to 1 rad/s (for low- and highpass filters) and to -1 and 1 rad/s (for bandpass and bandstop filters). It then computes the minimum order required for a lowpass filter to meet the passband specification.

References

[1] Rabiner, L.R., and B. Gold. *Theory and Application of Digital Signal Processing,* Englewood Cliffs, NJ: Prentice-Hall, 1975. Pg. 241.

See Also

`buttord` | `cheb1ord` | `cheby2` | `ellipord` | `kaiserord`

Purpose

Chebyshev window

Syntax

```
w = chebwin(L,r)
```

Description

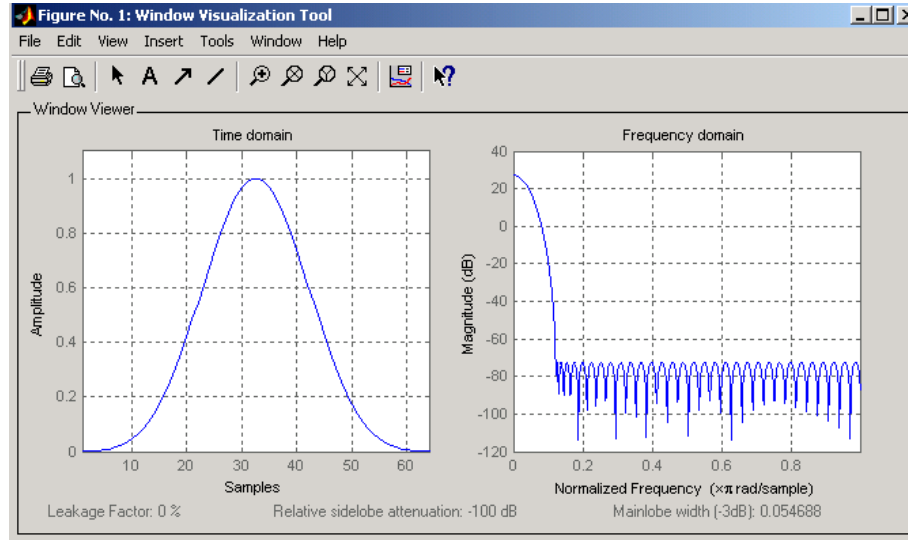
`w = chebwin(L,r)` returns the column vector `w` containing the length `L` Chebyshev window whose Fourier transform sidelobe magnitude is `r` dB below the mainlobe magnitude. The default value for `r` is 100.0 dB.

Note If you specify a one-point window (set `L=1`), the value `1` is returned.

Examples

Create a 64-point Chebyshev window with 100 dB of sidelobe attenuation and display the result using WVTool:

```
L=64;
wvtool(chebwin(L))
```



Algorithms

An artifact of the equiripple design method used in `chebwin` is the presence of impulses at the endpoints of the time-domain response. This is due to the constant-level sidelobes in the frequency domain. The magnitude of the impulses are on the order of the size of the spectral sidelobes. If the sidelobes are large, the effect at the endpoints may be significant. For more information on this effect, see [2].

References

[1] *IEEE Programs for Digital Signal Processing.* IEEE Press. New York: John Wiley & Sons, 1979. Program 5.2.

[2] Harris, Fredric J. *Multirate Signal Processing for Communication Systems,* New Jersey: Prentice Hall PTR, 2004, pp. 60-64.

See Also

`gausswin` | `kaiser` | `tukeywin` | `window` | `wintool` | `wvtool`

Purpose

Chebyshev Type I filter design (passband ripple)

Syntax

```
[z,p,k] = cheby1(n,R,Wp)
[z,p,k] = cheby1(n,R,Wp,'ftype')
[b,a] = cheby1(n,R,Wp)
[b,a] = cheby1(n,R,Wp,'ftype')
[A,B,C,D] = cheby1(n,R,Wp)
[A,B,C,D] = cheby1(n,R,Wp,'ftype')
[z,p,k] = cheby1(n,R,Wp,'s')
[z,p,k] = cheby1(n,R,Wp,'ftype','s')
[b,a] = cheby1(n,R,Wp,'s')
[b,a] = cheby1(n,R,Wp, 'ftype','s')
[A,B,C,D] = cheby1(n,R,Wp,'s')
[A,B,C,D] = cheby1(n,R,Wp,'ftype','s')
```

Description

`cheby1` designs lowpass, bandpass, highpass, and bandstop digital and analog Chebyshev Type I filters. Chebyshev Type I filters are equiripple in the passband and monotonic in the stopband. Type I filters roll off faster than type II filters, but at the expense of greater deviation from unity in the passband.

Digital Domain

`[z,p,k] = cheby1(n,R,Wp)` designs an order `n` `Chebyshev` lowpass digital Chebyshev filter with normalized passband edge frequency `Wp` and `R` dB of peak-to-peak ripple in the passband. It returns the zeros and poles in length `n` column vectors `z` and `p` and the gain in the scalar `k`.

`[z,p,k] = cheby1(n,R,Wp,'ftype')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is one of the following:

- `'high'` for a highpass digital filter with normalized passband edge frequency `Wp`
- `'low'` for a lowpass digital filter with normalized passband edge frequency `Wp`
- `'stop'` for an order `2*n` bandstop digital filter if `Wp` is a two-element vector, `Wp = [w1 w2]`. The stopband is `w1` < ω < `w2`.

Normalized passband edge frequency is the frequency at which the magnitude response of the filter is equal to `-R` dB. For `cheby1`, the normalized passband edge frequency `Wp` is a number between 0 and 1, where 1 corresponds to half the sample rate, π radians per sample. Smaller values of passband ripple `R` lead to wider transition widths (shallower rolloff characteristics).

If `Wp` is a two-element vector, `Wp = [w1 w2]`, `cheby1` returns an order `2*n` bandpass filter with passband `w1` < ω < `w2`.

With different numbers of output arguments, `cheby1` directly obtains other realizations of the filter. To obtain the transfer function form, use two output arguments as shown below.

Note See "Limitations" on page 1-97 for information about numerical issues that affect forming the transfer function.

`[b,a] = cheby1(n,R,Wp)` designs an order `n` `Chebyshev` lowpass digital Chebyshev filter with normalized passband edge frequency `Wp` and `R` dB of peak-to-peak ripple in the passband. It returns the filter coefficients in the length `n+1` row vectors `b` and `a`, with coefficients in descending powers of z.

$$H(z) = \frac{b(1) + b(2)z^{-1} + \ldots + b(n+1)z^{-n}}{1 + a(2)z^{-1} + \ldots + a(n+1)z^{-n}}$$

`[b,a] = cheby1(n,R,Wp,'ftype')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is `'high'`, `'low'`, or `'stop'`, as described above.

To obtain state-space form, use four output arguments as shown below:

`[A,B,C,D] = cheby1(n,R,Wp)` or

`[A,B,C,D] = cheby1(n,R,Wp,'ftype')` where `A`, `B`, `C`, and `D` are

$$x[n+1] = Ax[n] + Bu[n]$$
$$y[n] = Cx[n] + Du[n]$$

and u is the input, x is the state vector, and y is the output.

Analog Domain

`[z,p,k] = cheby1(n,R,Wp,'s')` designs an order `n` lowpass analog Chebyshev Type I filter with angular passband edge frequency `Wp` rad/s. It returns the zeros and poles in length `n` or `2*n` column vectors `z` and `p` and the gain in the scalar `k`.

Angular passband edge frequency is the frequency at which the magnitude response of the filter is `-R` dB. For `cheby1`, the angular passband edge frequency `Wp` must be greater than 0 rad/s.

If `Wp` is a two-element vector `Wp = [w1 w2]` with `w1 < w2`, then `cheby1(n,R,Wp,'s')` returns an order `2*n` bandpass analog filter with passband `w1` < ω< `w2`.

`[z,p,k] = cheby1(n,R,Wp,'ftype','s')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is `'high'`, `'low'`, or `'stop'`, as described above.

You can supply different numbers of output arguments for `cheby1` to directly obtain other realizations of the analog filter. To obtain the transfer function form, use two output arguments as shown below.

`[b,a] = cheby1(n,R,Wp,'s')` designs an order `n` lowpass analog Chebyshev Type I filter with angular passband edge frequency `Wp` rad/s. It returns the filter coefficients in length `n+1` row vectors `b` and `a`, in descending powers of s, derived from the transfer function

$$H(s) = \frac{B(s)}{A(s)} = \frac{b(1)s^n + b(2)s^{n-1} + \ldots + b(n+1)}{s^n + a(2)s^{n-1} + \ldots + a(n+1)}$$

`[b,a] = cheby1(n,R,Wp, 'ftype','s')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is `'high'`, `'low'`, or `'stop'`, as described above.

To obtain state-space form, use four output arguments as shown below:

`[A,B,C,D] = cheby1(n,R,Wp,'s')` or

`[A,B,C,D] = cheby1(n,R,Wp,'ftype','s')` where `A`, `B`, `C`, and `D` are defined as

$$x = Ax + Bu$$
$$y = Cx + Du$$

and u is the input, x is the state vector, and y is the output.

Examples

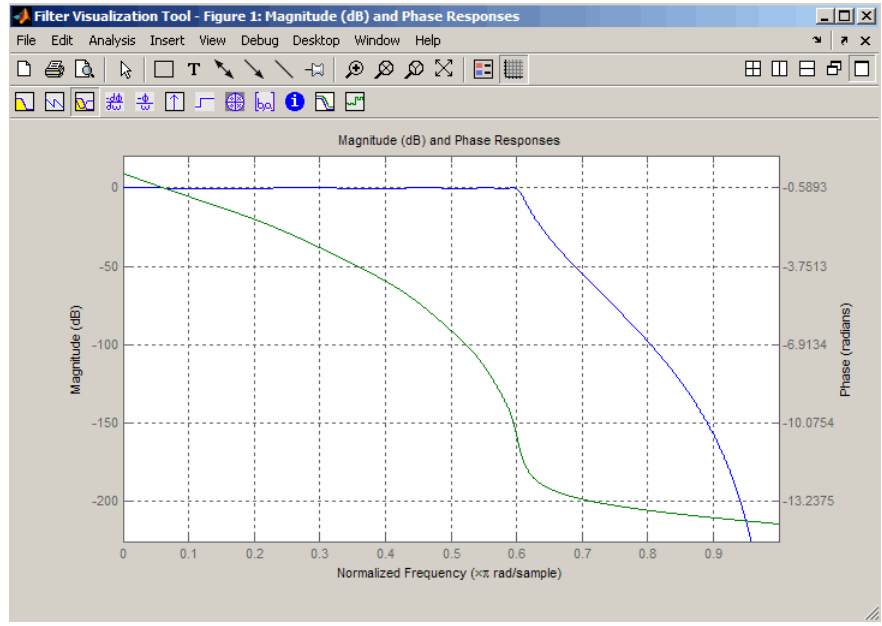
Lowpass Filter

For data sampled at 1000 Hz, design a 9th-order lowpass Chebyshev Type I filter with 0.5 dB of ripple in the passband and a passband edge frequency of 300 Hz, which corresponds to a normalized value of 0.6:

```
[z,p,k] = cheby1(9,0.5,300/500);
[sos,g] = zp2sos(z,p,k);   % Convert to SOS form
Hd = dfilt.df2tsos(sos,g);  % Create a dfilt object
h = fvtool(Hd)      % Plot magnitude response
set(h,'Analysis','freq')   % Display frequency response
```

The frequency response of the filter is

```
freqz(b,a,512,1000)
```

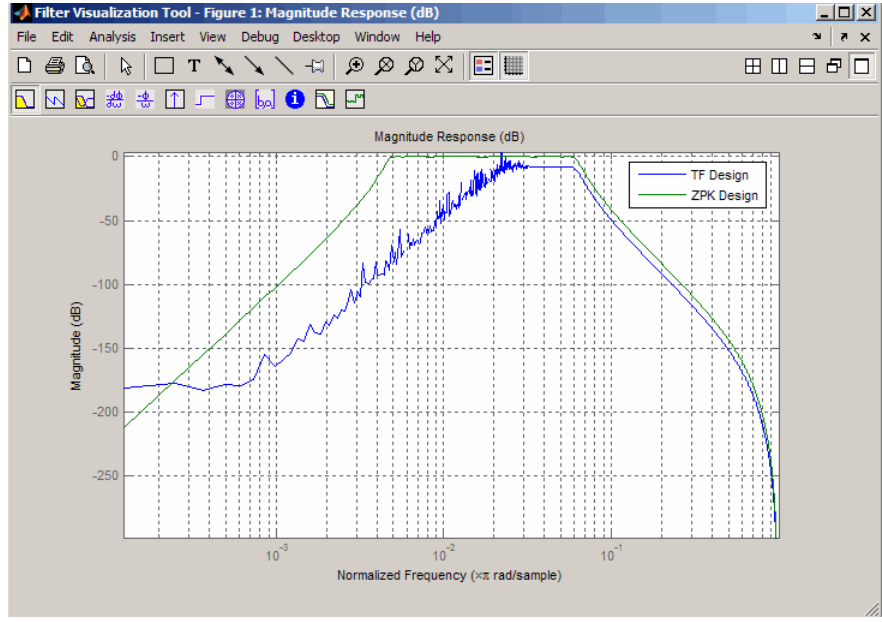


Limitations

In general, you should use the `[z,p,k]` syntax to design IIR filters. To analyze or implement your filter, you can then use the `[z,p,k]` output with `zp2sos` and an sos `dfilt` structure. For higher order filters (possibly starting as low as order 8), numerical problems due to roundoff errors may occur when forming the transfer function using the `[b,a]` syntax. The following example illustrates this limitation:

```
n = 6;
r = 0.1;
Wn = ([2.5e6 29e6]/500e6);
ftype = 'bandpass';

% Transfer Function design
[b,a] = cheby1(n,r,Wn,ftype);
h1=dfilt.df2(b,a);      % This is an unstable filter.
```

```
% Zero-Pole-Gain design
[z, p, k] = cheby1(n,r, Wn,ftype);
[sos,g]=zp2sos(z,p,k);
h2=dfilt.df2sos(sos,g);

% Plot and compare the results
hfvt=fvtool(h1,h2,'FrequencyScale','log');
legend(hfvt,'TF Design','ZPK Design')
```



Algorithms

`cheby1` uses a five-step algorithm:

1. It finds the lowpass analog prototype poles, zeros, and gain using the `cheb1ap` function.

2. It converts the poles, zeros, and gain into state-space form.

3 It transforms the lowpass filter into a bandpass, highpass, or bandstop filter with desired cutoff frequencies, using a state-space transformation.

4 For digital filter design, `cheby1` uses `bilinear` to convert the analog filter into a digital filter through a bilinear transformation with frequency prewarping. Careful frequency adjustment guarantees that the analog filters and the digital filters will have the same frequency response magnitude at `Wp` or `w1` and `w2`.

5 It converts the state-space filter back to transfer function or zero-pole-gain form, as required.

See Also

`besself` | `butter` | `cheb1ap` | `cheb1ord` | `cheby2` | `ellip`

Purpose

Chebyshev Type II filter design (stopband ripple)

Syntax

```
[z,p,k] = cheby2(n,R,Wst)
[z,p,k] = cheby2(n,R,Wst,'ftype')
[b,a] = cheby2(n,R,Wst)
[b,a] = cheby2(n,R,Wst,'ftype')
[A,B,C,D] = cheby2(n,R,Wst)
[A,B,C,D] = cheby2(n,R,Wst,'ftype')
[z,p,k] = cheby2(n,R,Wst,'s')
[z,p,k] = cheby2(n,R,Wst,'ftype','s')
[b,a] = cheby2(n,R,Wst,'s')
[b,a] = cheby2(n,R,Wst,'ftype','s')
[A,B,C,D] = cheby2(n,R,Wst,'s')
[A,B,C,D] = cheby2(n,R,Wst,'ftype','s')
```

Description

`cheby2` designs lowpass, highpass, bandpass, and bandstop digital and analog Chebyshev Type II filters. Chebyshev Type II filters are monotonic in the passband and equiripple in the stopband. Type II filters do not roll off as fast as type I filters, but are free of passband ripple.

Digital Domain

`[z,p,k] = cheby2(n,R,Wst)` designs an order `n` lowpass digital Chebyshev Type II filter with normalized stopband edge frequency `Wst` and stopband ripple `R` dB down from the peak passband value. It returns the zeros and poles in length `n` column vectors `z` and `p` and the gain in the scalar `k`.

Normalized stopband edge frequency is the beginning of the stopband, where the magnitude response of the filter is equal to `-R` dB. For `cheby2`, the normalized stopband edge frequency `Wst` is a number between 0 and 1, where 1 corresponds to half the sample rate. Larger values of stopband attenuation R lead to wider transition widths (shallower rolloff characteristics).

If `Wst` is a two-element vector, `Wst = [w1 w2]`, `cheby2` returns an order `2*n` bandpass filter with passband `w1` < ω < `w2`.

`[z,p,k] = cheby2(n,R,Wst,'ftype')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is one of the following:

- `'high'` for a highpass digital filter with normalized stopband edge frequency `Wst`
- `'low'` for a lowpass digital filter with normalized stopband edge frequency `Wst`
- `'stop'` for an order `2*n` bandstop digital filter if `Wst` is a two-element vector, `Wst = [w1 w2]`. The stopband is `w1` < ω < `w2`.

With different numbers of output arguments, `cheby2` directly obtains other realizations of the filter. To obtain the transfer function form, use two output arguments as shown below.

Note See "Limitations" on page 1-104 below for information about numerical issues that affect forming the transfer function.

`[b,a] = cheby2(n,R,Wst)` designs an order `n` lowpass digital Chebyshev Type II filter with normalized stopband edge frequency `Wst` and stopband ripple `R` dB down from the peak passband value. It returns the filter coefficients in the length `n+1` row vectors `b` and `a`, with coefficients in descending powers of z.

$$H(z) = \frac{B(z)}{A(z)} = \frac{b(1) + b(2)z^{-1} + \cdots + b(n+1)z^{-n}}{1 + a(2)z^{-1} + \cdots + a(n+1)z^{-n}}$$

`[b,a] = cheby2(n,R,Wst,'ftype')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is `'high'`, `'low'`, or `'stop'`, as described above.

To obtain state-space form, use four output arguments as shown below.

`[A,B,C,D] = cheby2(n,R,Wst)` or

`[A,B,C,D] = cheby2(n,R,Wst,'ftype')` where `A`, `B`, `C`, and `D` are

$$x[n+1] = Ax[n] + Bu[n]$$
$$y[n] \quad = Cx[n] + Du[n]$$

and u is the input, x is the state vector, and y is the output.

Analog Domain

`[z,p,k] = cheby2(n,R,Wst,'s')` designs an order `n` lowpass analog Chebyshev Type II filter with angular stopband edge frequency `Wst` rad/s.. It returns the zeros and poles in length `n` or `2*n` column vectors `z` and `p` and the gain in the scalar `k`.

Angular stopband edge frequency is the frequency at which the magnitude response of the filter is equal to `-R` dB. For `cheby2`, the angular stopband edge frequency `Wst` must be greater than 0 rad/s.

If `Wst` is a two-element vector `Wst = [w1 w2]` with `w1 < w2`, then `cheby2(n,R,Wst,'s')` returns an order `2*n` bandpass analog filter with passband `w1 < ω< w2`.

`[z,p,k] = cheby2(n,R,Wst,'ftype','s')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is `'high'`, `'low'`, or `'stop'`, as described above.

With different numbers of output arguments, `cheby2` directly obtains other realizations of the analog filter. To obtain the transfer function form, use two output arguments as shown below:

`[b,a] = cheby2(n,R,Wst,'s')` designs an order `n` lowpass analog Chebyshev Type II filter with angular stopband edge frequency `Wst` rad/s.. It returns the filter coefficients in the length `n+1` row vectors `b` and `a`, with coefficients in descending powers of s, derived from the transfer function.

$$H(s) = \frac{B(s)}{A(s)} = \frac{b(1)s^n + b(2)s^{n-1} + \cdots + b(n+1)}{s^n + a(2)s^{n-1} + \cdots + a(n+1)}$$

`[b,a] = cheby2(n,R,Wst,'ftype','s')` designs a highpass , lowpass, or bandstop filter, where the string `'ftype'` is `'high'`, `'low'`, or `'stop'`, as described above.

To obtain state-space form, use four output arguments as shown below:

`[A,B,C,D] = cheby2(n,R,Wst,'s')` or

`[A,B,C,D] = cheby2(n,R,Wst,'ftype','s')` where `A`, `B`, `C`, and `D` are

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

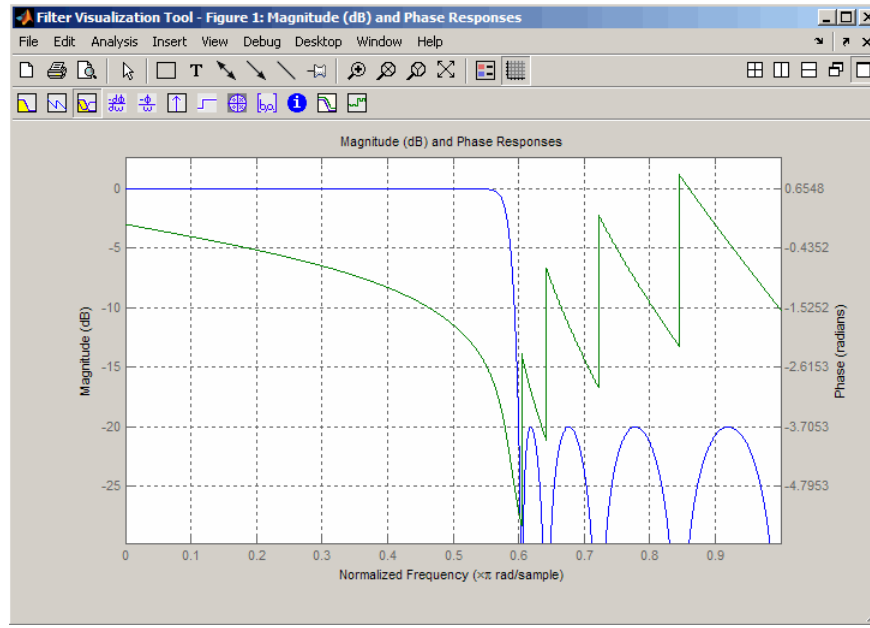
and u is the input, x is the state vector, and y is the output.

Examples

Lowpass Filter

For data sampled at 1000 Hz, design a ninth-order lowpass Chebyshev Type II filter with stopband attenuation 20 dB down from the passband and a stopband edge frequency of 300 Hz, which corresponds to a normalized value of 0.6:

```
[z,p,k] = cheby2(9,20,300/500);
[sos,g] = zp2sos(z,p,k);   % Convert to SOS form
Hd = dfilt.df2tsos(sos,g);  % Create a dfilt object
h = fvtool(Hd);      % Plot magnitude response
set(h,'Analysis','freq')   % Display frequency response
```

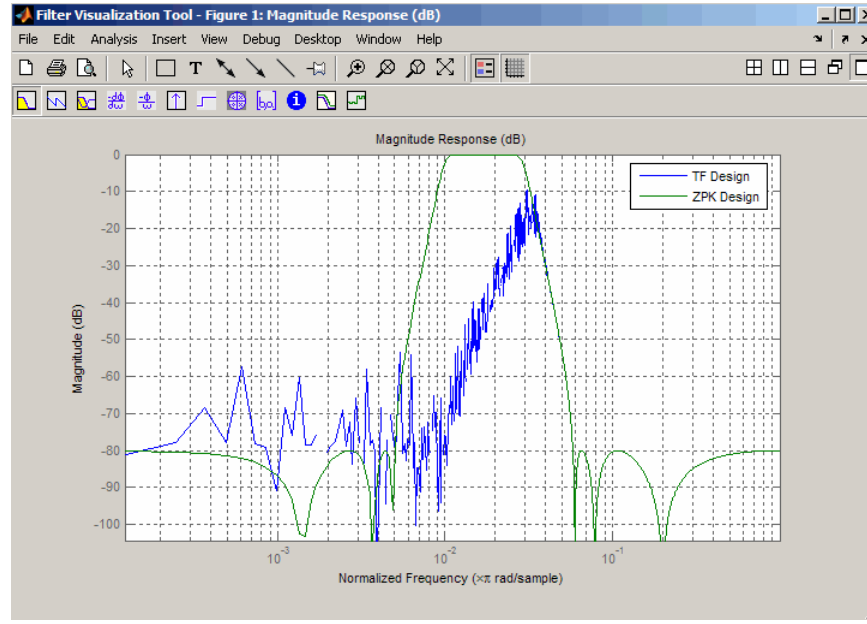


Limitations

In general, you should use the `[z,p,k]` syntax to design IIR filters. To analyze or implement your filter, you can then use the `[z,p,k]` output with `zp2sos` and an sos `dfilt` structure. For higher order filters (possibly starting as low as order 8), numerical problems due to roundoff errors may occur when forming the transfer function using the `[b,a]` syntax. The following example illustrates this limitation:

```
n = 6;
r = 80;
Wn = [2.5e6 29e6]/500e6;
ftype = 'bandpass';

% Transfer Function design
[b,a] = cheby2(n,r,Wn,ftype);
h1=dfilt.df2(b,a);      % This is an unstable filter.
```

```
% Zero-Pole-Gain design
[z, p, k] = cheby2(n,r,Wn,ftype);
[sos,g]=zp2sos(z,p,k);
h2=dfilt.df2sos(sos,g);

% Plot and compare the results
hfvt=fvtool(h1,h2,'FrequencyScale','log');
legend(hfvt,'TF Design','ZPK Design')
```



Algorithms

`cheby2` uses a five-step algorithm:

1. It finds the lowpass analog prototype poles, zeros, and gain using the `cheb2ap` function.

2. It converts poles, zeros, and gain into state-space form.

3. It transforms the lowpass filter into a bandpass, highpass, or bandstop filter with desired cutoff frequencies, using a state-space transformation.

4. For digital filter design, `cheby2` uses `bilinear` to convert the analog filter into a digital filter through a bilinear transformation with frequency prewarping. Careful frequency adjustment guarantees that the analog filters and the digital filters will have the same frequency response magnitude at `Wst` or `w1` and `w2`.

5. It converts the state-space filter back to transfer function or zero-pole-gain form, as required.

See Also

`besself` | `butter` | `cheb2ap` | `cheb1ord` | `cheby1` | `ellip`

Purpose

Swept-frequency cosine

Syntax

```
y = chirp(t,f0,t1,f1)
y = chirp(t,f0,t1,f1,'method')
y = chirp(t,f0,t1,f1,'method',phi)
y = chirp(t,f0,t1,f1,'quadratic',phi,'shape')
```

Description

`y = chirp(t,f0,t1,f1)` generates samples of a linear swept-frequency cosine signal at the time instances defined in array `t`, where `f0` is the instantaneous frequency at time 0, and `f1` is the instantaneous frequency at time `t1`. `f0` and `f1` are both in hertz. If unspecified, `f0` is e^{-6} for logarithmic chirp and 0 for all other methods, `t1` is `1`, and `f1` is `100`.

`y = chirp(t,f0,t1,f1,'method')` specifies alternative sweep method options, where *`method`* can be:

- `linear`, which specifies an instantaneous frequency sweep $f_i(t)$given by

  $$f_i(t) = f_0 + \beta t$$

  where

  $$\beta = (f_1 - f_0) / t_1$$

  and the default value for f_0 is 0. ß ensures that the desired frequency breakpoint f_1 at time t_1 is maintained.

- `quadratic`, which specifies an instantaneous frequency sweep $f_i(t)$ given by

  $$f_i(t) = f_0 + \beta t^2$$

  where

  $$\beta = (f_1 - f_0) / t_1^2$$

and the default value for f_0 is 0. If $f_0 > f_1$ (downsweep), the default shape is convex. If $f_0 < f_1$ (upsweep), the default shape is concave.

- `logarithmic` specifies an instantaneous frequency sweep $f_i(t)$ given by

  $$f_i(t) = f_0 \times \beta^t$$

  where

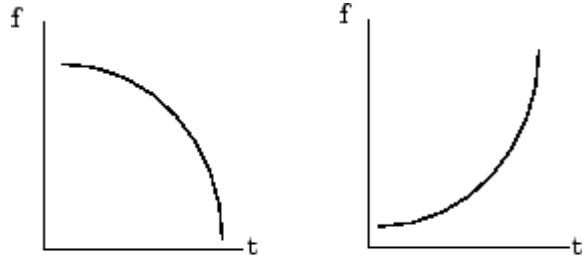
  $$\beta = \left(\frac{f_1}{f_0} \right)^{\frac{1}{t_1}}$$

  and the default value for f_0 is $1e^{-6}$. Both an upsweep ($f_1 > f_0$) and a downsweep ($f_0 > f_1$) of frequency is possible.

Each of the above methods can be entered as `'li'`, `'q'`, and `'lo'`, respectively.

`y = chirp(t,f0,t1,f1,'method',phi)` allows an initial phase `phi` to be specified in degrees. If unspecified, `phi` is `0`. Default values are substituted for empty or omitted trailing input arguments.

`y = chirp(t,f0,t1,f1,'quadratic',phi,'shape')` specifies the `shape` of the quadratic swept-frequency signal's spectrogram. `shape` is either `concave` or `convex`, which describes the shape of the parabola in the positive frequency axis. If `shape` is omitted, the default is convex for downsweep ($f_0 > f_1$) and is concave for upsweep ($f_0 < f_1$).

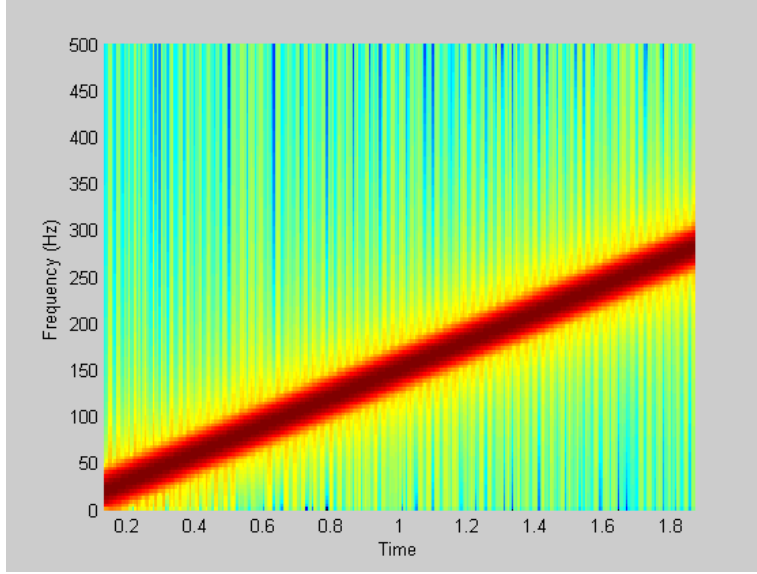


Examples

Example 1

Compute the spectrogram of a chirp with linear instantaneous frequency deviation:

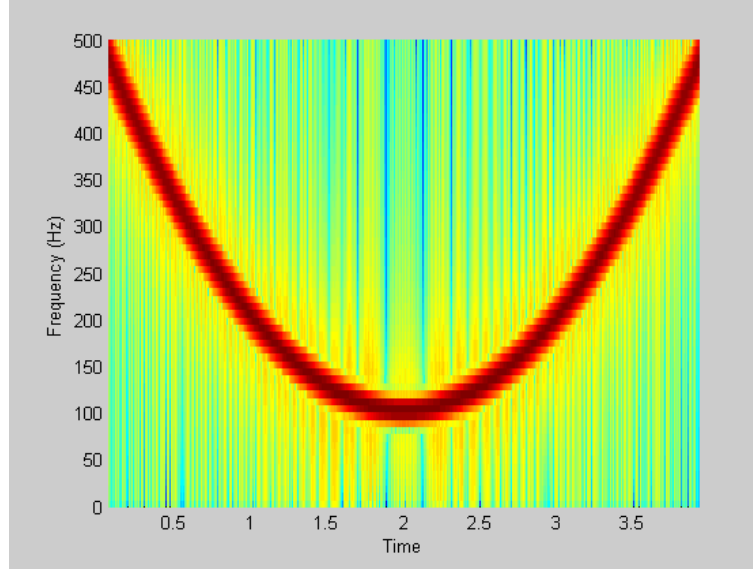
```
t = 0:0.001:2;            % 2 secs @ 1kHz sample rate
y = chirp(t,0,1,150);     % Start @ DC,
                          %   cross 150Hz at t=1 sec
spectrogram(y,256,250,256,1E3,'yaxis')
```



Example 2

Compute the spectrogram of a chirp with quadratic instantaneous frequency deviation:

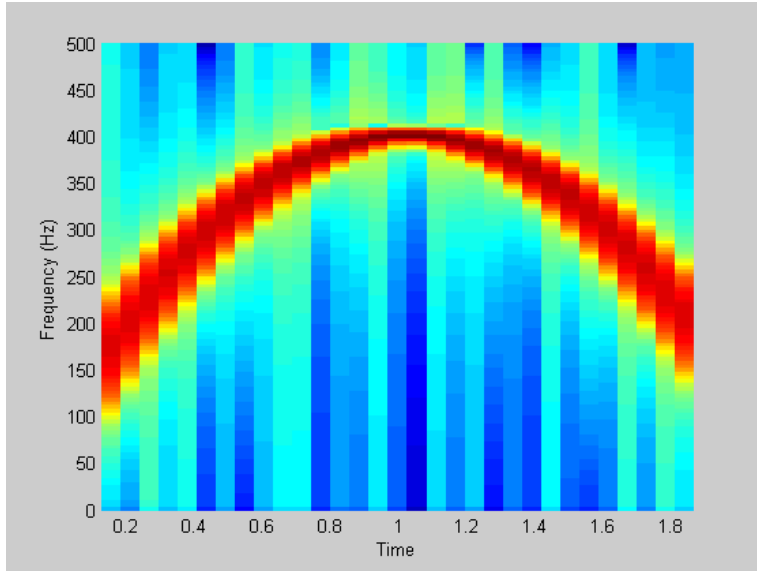
```
% –2 secs @ 1kHz sample rate
t = -2:0.001:2;

% Start @ 100Hz, cross 200Hz at t=1 sec
y = chirp(t,100,1,200,'quadratic');

spectrogram(y,128,120,128,1E3,'yaxis')
```



Example 3

Compute the spectrogram of a convex quadratic chirp:

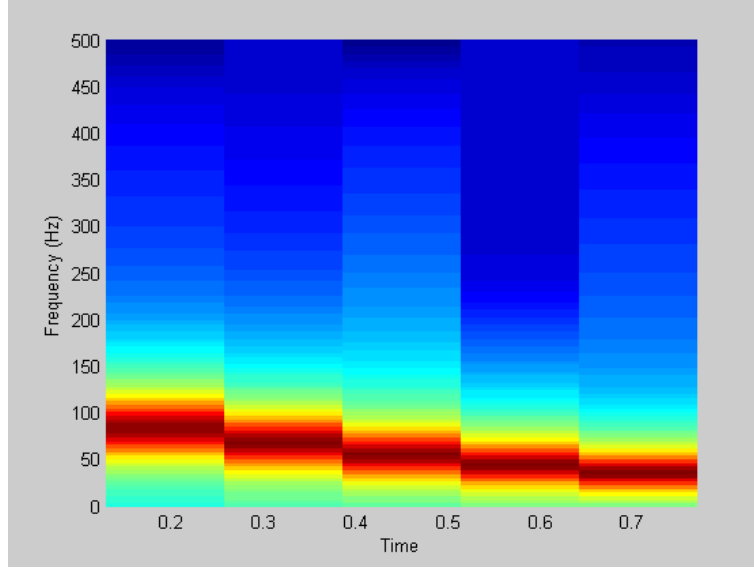
```
t = -1:0.001:1;          % +/-1 second @ 1kHz sample rate
fo = 100; f1 = 400;      % Start at 100Hz, go up to 400Hz
y = chirp(t,fo,1,f1,'q',[],'convex');
spectrogram(y,256,200,256,1000,'yaxis')
```



Example 4

Compute the spectrogram of a concave quadratic chirp:

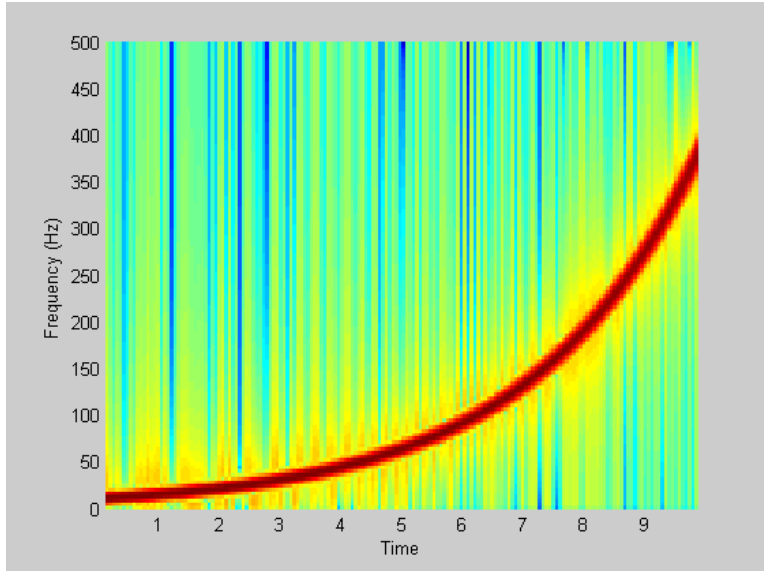
```
t = 0:0.001:1;       % 1 second @ 1kHz sample rate
fo = 100; f1 = 25;   % Start at 100Hz, go down to 25Hz
y = chirp(t,fo,1,f1,'q',[],'concave');
spectrogram(y,hanning(256),128,256,1000,'yaxis')
```



Example 5

Compute the spectrogram of a logarithmic chirp:

```
t = 0:0.001:10;       % 10 seconds @ 1kHz sample rate
fo = 10; f1 = 400;    % Start at 10Hz, go up to 400Hz
y = chirp(t,fo,10,f1,'logarithmic');
spectrogram(y,256,200,256,1000,'yaxis')
```



See Also

`cos` | `diric` | `gauspuls` | `pulstran` | `rectpuls` | `sawtooth` | `sin` | `sinc` | `square` | `tripuls`

Purpose

Convolution matrix

Syntax

```
A = convmtx(h,n)
```

Description

A *convolution matrix* is a matrix, formed from a vector, whose product with another vector is the convolution of the two vectors.

`A = convmtx(h,n)` returns the convolution matrix, `A`, such that the product of `A` and a vector, `x`, is the convolution of `h` and `x`. If `h` is a column vector of length `m`, `A` is `(m+n-1)`-by-`n` and the product of `A` and a column vector, `x`, of length `n` is the convolution of `h` and `x`. If `h` is a row vector of length `m`, `A` is `n`-by-`(m+n-1)` and the product of a row vector, `x`, of length `n` with `A` is the convolution of `h` and `x`.

Examples

Generate a simple convolution matrix:

```
h = [1 2 3 2 1];
convmtx(h,7);
```

Note that `convmtx` handles edge conditions by zero padding.

In practice, it is more efficient to compute convolution using

```
y = conv(c,x);
```

than by using a convolution matrix.

```
n = length(x);
y = convmtx(c,n)*x;
```

Algorithms

`convmtx` uses the function `toeplitz` to generate the convolution matrix.

See Also

`conv` | `convn` | `conv2` | `dftmtx`

Purpose

Data matrix for autocorrelation matrix estimation

Syntax

```
X = corrmtx(x,m)
X = corrmtx(x,m,'method')
[X,R] = corrmtx(...)
```

Description

`X = corrmtx(x,m)` returns an (*n*+`m`)-by-(`m`+1) rectangular Toeplitz matrix `X`, such that `X'X` is a (biased) estimate of the autocorrelation matrix for the length *n* data vector `x`.

`X = corrmtx(x,m,'method')` computes the matrix `X` according to the method specified by the string '`method`':

- `'autocorrelation'`: (default) `X` is the (*n*+`m`)-by-(`m`+1) rectangular Toeplitz matrix that generates an autocorrelation estimate for the length *n* data vector `x`, derived using *prewindowed* and *postwindowed* data, based on an `m`th order prediction error model.
- `'prewindowed'`: `X` is the *n*-by-(`m`+1) rectangular Toeplitz matrix that generates an autocorrelation estimate for the length *n* data vector `x`, derived using *prewindowed* data, based on an `m`th order prediction error model.
- `'postwindowed'`: `X` is the *n*-by-(`m`+1) rectangular Toeplitz matrix that generates an autocorrelation estimate for the length *n* data vector `x`, derived using *postwindowed* data, based on an `m`th order prediction error model.
- `'covariance'`: `X` is the (*n*-`m`)-by-(`m`+1) rectangular Toeplitz matrix that generates an autocorrelation estimate for the length *n* data vector `x`, derived using *nonwindowed* data, based on an `m`th order prediction error model.
- `'modified'`: `X` is the 2(*n*-`m`)-by-(`m`+1) modified rectangular Toeplitz matrix that generates an autocorrelation estimate for the length *n* data vector `x`, derived using forward and backward prediction error estimates, based on an `m`th order prediction error model.

`[X,R] = corrmtx(...)` also returns the (`m`+1)-by-(`m`+1) autocorrelation matrix estimate `R`, calculated as `X'*X`.

Examples

```
n=0:99;
s=exp(i*pi/2*n)+2*exp(i*pi/4*n)+exp(i*pi/3*n)+randn(1,100);
X=corrmtx(s,12,'mod');
```

Algorithms

The Toeplitz data matrix computed by `corrmtx` depends on the method you select. The matrix determined by the autocorrelation (default) method is given by the following matrix.

$$X = \left[\begin{array}{ccc} x(1) & \cdots & 0 \\ \vdots & \ddots & \vdots \\ \hline x(m+1) & \cdots & x(1) \\ \vdots & \ddots & \vdots \\ x(n-m) & \cdots & x(m+1) \\ \vdots & \ddots & \vdots \\ x(n) & \cdots & x(n-m) \\ \hline \vdots & \ddots & \vdots \\ 0 & \cdots & x(n) \end{array}\right]$$

In this matrix, m is the same as the input argument `m` to `corrmtx`, and n is `length(x)`. Variations of this matrix are used to return the output `X` of `corrmtx` for each method:

- `'autocorrelation'`: (default) `X` = X, above.
- `'prewindowed'`: `X` is the n-by-(m+1) submatrix of X that is given by the portion of X above the lower gray line.
- `'postwindowed'`: `X` is the n-by-(m+1) submatrix of X that is given by the portion of X below the upper gray line.
- 'covariance': `X` is the (n-m)-by-(m+1) submatrix of X that is given by the portion of X between the two gray lines.
- `'modified'`: `X` is the 2(n-m)-by-(m+1) matrix X_{mod} shown below.

$$X_{\text{mod}} = \begin{bmatrix} x(m+1) & \cdots & x(1) \\ \vdots & \ddots & \vdots \\ x(n-m) & \cdots & x(m+1) \\ \vdots & \ddots & \vdots \\ x(n) & \cdots & x(n-m) \\ x*(1) & \cdots & x*(m+1) \\ \vdots & \iddots & \vdots \\ x*(m+1) & \cdots & x*(n-m) \\ \vdots & \iddots & \vdots \\ x*(n-m) & \cdots & x*(n) \end{bmatrix}$$

References

[1] Marple, S.L. *Digital Spectral Analysis,*Englewood Cliffs, NJ, Prentice-Hall, 1987, pp. 216-223.

See Also

`peig` | `pmusic` | `rooteig` | `rootmusic` | `xcorr`

Purpose

Cross power spectral density

Syntax

```
Pxy = cpsd(x,y)
Pxy = cpsd(x,y,window)
Pxy = cpsd(x,y,window,noverlap)
[Pxy,W] = cpsd(x,y,window,noverlap,nfft)
[Pxy,F] = cpsd(x,y,window,noverlap,nfft,fs)
[...] = cpsd(...,'twosided')
cpsd(...)
```

Description

`Pxy = cpsd(x,y)` estimates the cross power spectral density `Pxy` of the discrete-time signals `x` and `y` using the Welch's averaged, modified periodogram method of spectral estimation. The cross power spectral density is the distribution of power per unit frequency and is defined as

$$P_{xy}(\omega) = \sum_{m=-\infty}^{\infty} R_{xy}(m) e^{-j\omega m}$$

The cross-correlation sequence is defined as

$$R_{xy}(m) = E\{x_{n+m} y^*_n\} = E\{x_n y^*_{n-m}\}$$

where x_n and y_n are jointly stationary random processes, $-\infty < n < \infty$, and $E\{\cdot\}$ is the expected value operator.

For real `x` and `y`, `cpsd` returns a one-sided CPSD and for complex `x` or `y`, it returns a two-sided CPSD.

`cpsd` uses the following default values:

Parameter	Description	Default Value
`nfft`	FFT length which determines the frequencies at which the PSD is estimated For real `x` and `y`, the length of `Pxy` is (`nfft`/2+1) if `nfft` is even or (`nfft`+1)/2 if `nfft` is odd. For complex `x` or `y`, the length of `Pxy` is `nfft`. If `nfft` is greater than the signal length, the data is zero-padded. If `nfft` is less than the signal length, the segment is wrapped using datawrap so that the length is equal to `nfft`.	Maximum of 256 or the next power of 2 greater than the length of each section of `x` or `y`
`fs`	Sampling frequency	1
`window`	Windowing function and number of samples to use for each section	Periodic Hamming window of length to obtain eight equal sections of `x` and `y`
`noverlap`	Number of samples by which the sections overlap	Value to obtain 50% overlap

Note You can use the empty matrix `[]` to specify the default value for any input argument except `x` or `y`. For example, `Pxy = cpsd(x,y,[],[],128)` uses a Hamming window, default `noverlap` to obtain 50% overlap, and the specified 128 `nfft`.

`Pxy = cpsd(x,y,window)` specifies a windowing function, divides `x` and `y` into overlapping sections of the specified window length, and windows each section using the specified window function. If you supply a scalar for `window`, `Pxy` uses a Hamming window of that length. The x and y vectors are divided into eight equal sections of that length. If the signal cannot be sectioned evenly with 50% overlap, it is truncated.

`Pxy = cpsd(x,y,window,noverlap)` overlaps the sections of `x` by `noverlap` samples. `noverlap` must be an integer smaller than the length of `window`.

`[Pxy,W] = cpsd(x,y,window,noverlap,nfft)` uses the specified FFT length `nfft` in estimating the CPSD. It also returns `W`, which is the vector of normalized frequencies (in rad/sample) at which the CPSD is estimated. For real signals, the range of `W` is [0, pi] when `nfft` is even and [0, pi) when `nfft` is odd. For complex signals, the range of `W` is [0, 2*pi).

`[Pxy,F] = cpsd(x,y,window,noverlap,nfft,fs)` returns `Pxy` as a function of frequency and a vector `F` of frequencies at which the CPSD is estimated. `fs` is the sampling frequency in Hz. For real signals, the range of `F` is [0, `fs`/2] when `nfft` is even and [0, `fs`/2) when `nfft` is odd. For complex signals, the range of `F` is [0, `fs`).

`[...] = cpsd(...,'twosided')` returns the two-sided CPSD of real signals `x` and `y`. The length of the resulting Pxy is nfft and its range is [0, 2*pi) if you do not specify fs. If you specify fs, the range is [0,fs). Entering'onesided'for a real signal produces the default. You can place the 'onesided' or 'twosided' string in any position after the `noverlap` parameter.

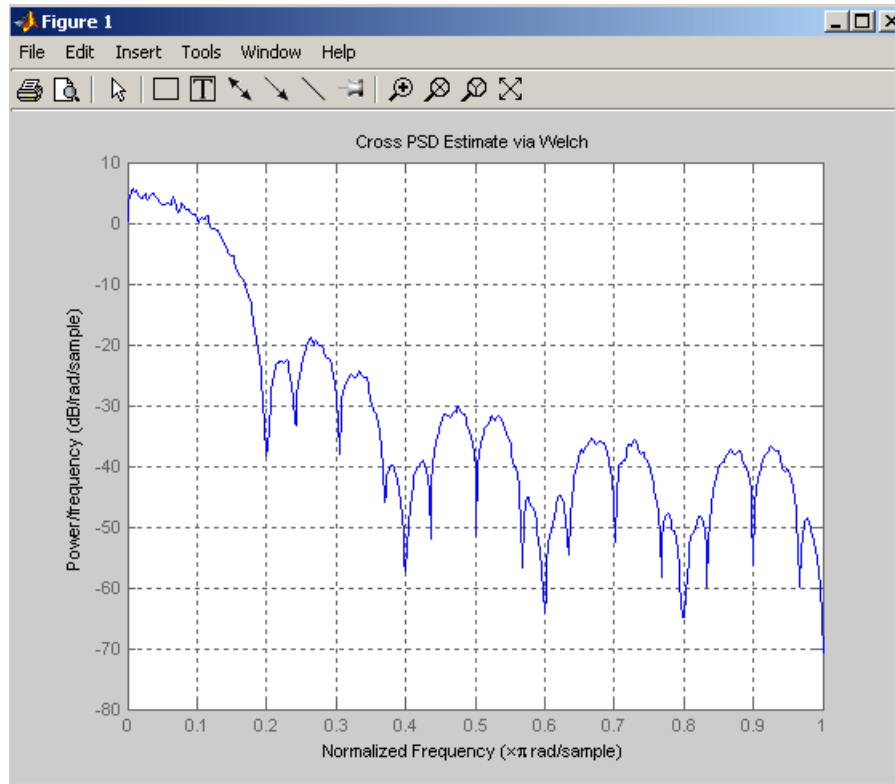
`cpsd(...)` plots the CPSD versus frequency in the current figure window.

Examples

Generate two colored noise signals and plot their CPSD. Specify a length 1024 FFT and a 500 point triangular window with no overlap.

```
rng default;
h = fir1(30,0.2,rectwin(31));
```

```
h1 = ones(1,10)/sqrt(10);
r = randn(16384,1);
x = filter(h1,1,r);
y = filter(h,1,x);
cpsd(x,y,triang(500),250,1024)
```



Algorithms

`cpsd` uses Welch's averaged periodogram method. See the references listed below.

References

[1] Rabiner, L.R., and B. Gold. *Theory and Application of Digital Signal Processing,* Englewood Cliffs, NJ: Prentice-Hall, 1975. Pgs. 414-419.

[2] Welch, P.D. "The Use of Fast Fourier Transform for the Estimation of Power Spectra: A Method Based on Time Averaging Over Short, Modified Periodograms." *IEEE Trans. Audio Electroacoust,* Vol. AU-15 (June 1967). Pgs. 70-73.

[3] Oppenheim, A.V., and R.W. Schafer. *Discrete-Time Signal Processing,* Upper Saddle River, NJ: Prentice-Hall, 1999, pp. 737.

See Also

`dspdata` | `mscohere` | `pburg` | `pcov` | `peig` | `periodogram` | `pmcov` | `pmtm` | `pmusic` | `pwelch` | `pyulear` | `spectrum` | `tfestimate`

Purpose

Chirp *z*-transform

Syntax

```
y = czt(x,m,w,a)
y = czt(x)
```

Description

`y = czt(x,m,w,a)` returns the chirp *z*-transform of signal `x`. The chirp *z*-transform is the *z*-transform of `x` along a spiral contour defined by `w` and `a`. `m` is a scalar that specifies the length of the transform, `w` is the ratio between points along the *z*-plane spiral contour of interest, and scalar `a` is the complex starting point on that contour. The contour, a spiral or "chirp" in the *z*-plane, is given by

```
z = a*(w.^-(0:m-1))
```

`y = czt(x)` uses the following default values:

- `m = length(x)`
- `w = exp(-j*2*pi/m)`
- `a = 1`

With these defaults, `czt` returns the *z*-transform of `x` at `m` equally spaced points around the unit circle. This is equivalent to the discrete Fourier transform of `x`, or `fft(x)`. The empty matrix `[]` specifies the default value for a parameter.

If `x` is a matrix, `czt(x,m,w,a)` transforms the columns of `x`.

Algorithms

`czt` uses the next power-of-2 length FFT to perform a fast convolution when computing the *z*-transform on a specified chirp contour [1].

Examples

Create a random vector `x` of length 1013 and compute its DFT using `czt`:

```
rng default;
x = randn(1013,1);
y = czt(x);
```

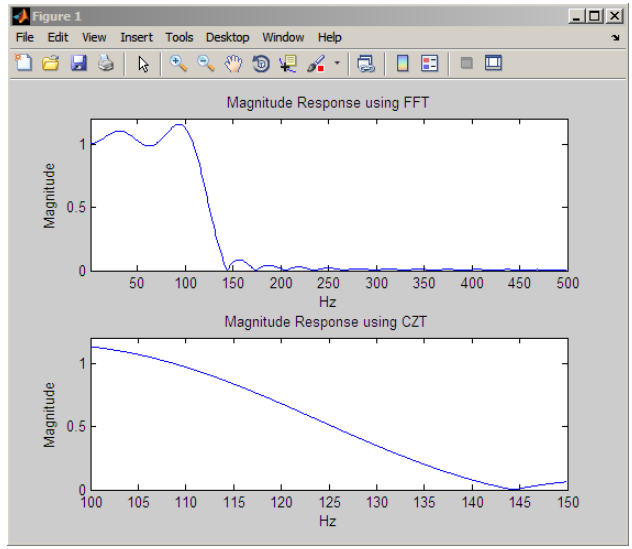
Use `czt` to zoom in on a narrow-band section (100 to 150 Hz) of a filter's frequency response. First design the filter:

```
h = fir1(30,125/500,rectwin(31));  % filter

fs = 1000; f1 = 100; f2 = 150;     % in hertz
m = 1024;
w = exp(-j*2*pi*(f2-f1)/(m*fs));
a = exp(j*2*pi*f1/fs);
```

Establish frequency and CZT parameters:

Compute the frequency response of the filter using `fft` and `czt`:

```
y = fft(h,1000);
z = czt(h,m,w,a);
fy = (0:length(y)-1)'*1000/length(y);
fz = ((0:length(z)-1)'*(f2-f1)/length(z)) + f1;
subplot(211);
plot(fy(1:500),abs(y(1:500))); axis([1 500 0 1.2])
xlabel('Hz'); ylabel('Magnitude');
title('Magnitude Response using FFT')
subplot(212);
plot(fz,abs(z)); axis([f1 f2 0 1.2])
xlabel('Hz'); ylabel('Magnitude');
title('Magnitude Response using CZT ')
```



Diagnostics

If `m`, `w`, or `a` is not a scalar, `czt` gives the following error message:

```
Inputs M, W, and A must be scalars.
```

References

[1] Rabiner, L.R., and B. Gold. *Theory and Application of Digital Signal Processing,* Englewood Cliffs, NJ: Prentice-Hall, 1975. Pgs. 393-399.

See Also

`fft` | `freqz`

Purpose

Convert energy or power measurements to decibels

Syntax

```
dboutput = db(X)
dboutput = db(X,SignalType)
dboutput = db(X,R)
dboutput = db(X,'voltage',R)
```

Description

`dboutput = db(X)` converts the elements of the vector or matrix `X` to decibels (dB). The elements of `X` are voltage measurements across a resistance of 1 ohm.

`dboutput = db(X,SignalType)` specifies the signal type represented by the elements of `X` as `'voltage'` or `'power'`. The entries are not case sensitive. The default value is `'voltage'`. For voltage measurements, the resistance defaults to 1 ohm. If you specify `SignalType` as `'power'`, the elements of `X` must be nonnegative.

`dboutput = db(X,R)` specifies the resistance `R` for voltage measurements. You can specify a resistance only when the signal measurements are voltages.

`dboutput = db(X,'voltage',R)` specifies the resistance `R` for voltage measurements. This syntax is equivalent to `db(X,R)`.

Input Arguments

X

Signal measurements. `X` must be a vector or matrix. If the elements of `X` are power measurements, all elements must be nonnegative.

SignalType

Type of signal measurements. Valid entries for `SignalType` are `'voltage'` or `'power'`. The entries are not case sensitive. If you specify `SignalType` as `'power'`, the elements of `X` must be nonnegative.

Default: `'voltage'`

R

Resistive load in ohms. You can specify resistance only when the `SignalType` is `'voltage'`.

Default: 1

Output Arguments

dboutput

The energy or power measurements in the input `X` in decibels. `dboutput` has the same dimensions as the input `X`.

If the input `X` contains voltage (energy) measurements, `dboutput` is:

$$dB = 10\log_{10}(|X|^2/R)$$

If the input `X` contains power measurements, `dboutput` is:

$$dB = 10\log_{10}(X)$$

Examples

Convert voltage to decibels. Assume that the resistance is 2 ohms.

```
V = 1;
R = 2;
dboutput = db(V,2)
% equivalent to 10*log10(1/2)
```

Convert a vector of power measurements to decibels.

```
rng default
X = abs(randn(10,1));
dboutput = db(X,'power')
```

Alternatives

- `mag2db` — Converts magnitude measurements to decibels.
- `pow2db` — Converts power measurements to decibels.

See Also

`db2mag` | `db2pow` | `mag2db` | `pow2db`

Purpose

Convert decibels (dB) to magnitude

Syntax

```
y = db2mag(ydb)
```

Description

`y = db2mag(ydb)` returns the corresponding magnitude `y` for a given decibel (dB) value `ydb`. The relationship between magnitude and decibels is `ydb` = $20*\log_{10}(y)$.

See Also

`mag2db`

db2pow

Purpose

Convert decibels (dB) to power

Syntax

```
y = db2pow(ydb)
```

Description

`y = db2pow(ydb)` returns the corresponding power value `y` for a given decibel (dB) value `ydb`. The relationship between power and decibels is `ydb` = $10*\log_{10}(y)$.

See Also

`pow2db`

Purpose

Discrete cosine transform (DCT)

Syntax

```
y = dct(x)
y = dct(x,n)
```

Description

`y = dct(x)` returns the unitary discrete cosine transform of `x`

$$y(k) = w(k)\sum_{n=1}^{N} x(n)\cos(\tfrac{\pi(2n-1)(k-1)}{2N}) \quad k = 1,2,\ldots N$$

where

$$w(k) = \begin{cases} \dfrac{1}{\sqrt{N}} & k = 1 \\ \sqrt{\dfrac{2}{N}} & 2 \le k \le N \end{cases}$$

N is the length of `x`, and `x` and `y` are the same size. If `x` is a matrix, `dct` transforms its columns. The series is indexed from n = 1 and k = 1 instead of the usual n = 0 and k = 0 because MATLAB vectors run from 1 to N instead of from 0 to N- 1.

`y = dct(x,n)` pads or truncates `x` to length `n` before transforming.

The DCT is closely related to the discrete Fourier transform. You can often reconstruct a sequence very accurately from only a few DCT coefficients, a useful property for applications requiring data reduction.

Examples

Find how many DCT coefficients represent 99% of the energy in a sequence:

```
x = (1:100) + 50*cos((1:100)*2*pi/40);
X = dct(x);
[XX,ind] = sort(abs(X)); ind = fliplr(ind);
i = 1;
while (norm([X(ind(1:i)) zeros(1,100-i)])/norm(X)<.99)
```

```
   i = i + 1;
end
% i = 3
```

References

[1] Jain, A.K. *Fundamentals of Digital Image Processing,* Englewood Cliffs, NJ: Prentice-Hall, 1989.

[2] Pennebaker, W.B., and J.L. Mitchell. *JPEG Still Image Data Compression Standard,* New York, NY: Van Nostrand Reinhold, 1993. Chapter 4.

See Also

`fft` | `idct` | `dct2` | `idct2`

Purpose

Decimation — decrease sampling rate

Syntax

```
y = decimate(x,r)
y = decimate(x,r,n)
y = decimate(x,r,'fir')
y = decimate(x,r,n,'fir')
```

Description

Decimation reduces the original sampling rate for a sequence to a lower rate, the opposite of interpolation. The decimation process filters the input data with a lowpass filter and then resamples the resulting smoothed signal at a lower rate.

`y = decimate(x,r)` reduces the sample rate of `x` by a factor `r`. The decimated vector `y` is `r` times shorter in length than the input vector `x`. By default, `decimate` employs an eighth-order lowpass Chebyshev Type I filter with a cutoff frequency of 0.8*(Fs/2)/`r`. It filters the input sequence in both the forward and reverse directions to remove all phase distortion, effectively doubling the filter order.

`y = decimate(x,r,n)` uses an order `n` Chebyshev filter. Orders above 13 are not recommended because of numerical instability. In this case, a warning is displayed.

Note For better results when `r` is greater than 13, you should break `r` into its factors and call `decimate` several times.

`y = decimate(x,r,'fir')` uses an order 30 FIR filter, instead of the Chebyshev IIR filter. Here `decimate` filters the input sequence in only one direction. This technique conserves memory and is useful for working with long sequences.

`y = decimate(x,r,n,'fir')` uses an order `n` FIR filter.

Examples

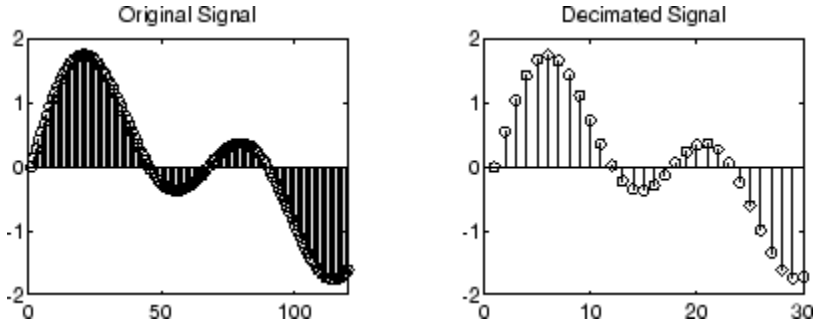
Decimate a signal by a factor of four:

```
t = 0:.00025:1;                       % Time vector
```

```
x = sin(2*pi*30*t) + sin(2*pi*60*t);
y = decimate(x,4);
```

View the original and decimated signals:

```
stem(x(1:120)), axis([0 120 -2 2])   % Original signal
title('Original Signal')
figure
stem(y(1:30))                        % Decimated signal
title('Decimated Signal')
```



Algorithms

`decimate` uses decimation algorithms 8.2 and 8.3 from [1]:

1. It designs a lowpass filter. By default, `decimate` uses a Chebyshev Type I filter with normalized cutoff frequency `0.8/r` and 0.05 dB of passband ripple. For the `fir` option, `decimate` designs a lowpass FIR filter with cutoff frequency `1/r` using `fir1`.

2. For the FIR filter, `decimate` applies the filter to the input vector in one direction. In the IIR case, `decimate` applies the filter in forward and reverse directions with `filtfilt`.

3. `decimate` resamples the filtered data by selecting every `r`th point.

Note Depending on the CPU and operating system of your computer, the decimate function may use a lower filter order. If the specified filter order will produce passband distortion, caused by roundoff errors accumulated from the convolutions needed to create the transfer function, the filter order is automatically reduced.

Diagnostics

If `r` is not an integer, `decimate` gives the following error message:

```
Resampling rate R must be an integer.
```

If `n` specifies an IIR filter with order greater than 13, `decimate` gives the following warning:

```
Warning: IIR filters above order 13 may be unreliable.
```

References

[1] *IEEE Programs for Digital Signal Processing,* IEEE Press. New York: John Wiley & Sons, 1979. Chapter 8.

See Also

`cheby1` | `downsample` | `filtfilt` | `fir1` | `mfilt` | `interp` | `resample`

demod

Purpose

Demodulation for communications simulation

Syntax

```
x = demod(y,fc,fs,'method')
x = demod(y,fc,fs,'method',opt)
x = demod(y,fc,fs,'pwm','centered')
```

Description

`demod` performs demodulation, that is, it obtains the original signal from a modulated version of the signal. `demod` undoes the operation performed by `modulate`.

`x = demod(y,fc,fs,'method')` and

`x = demod(y,fc,fs,'method',opt)` demodulate the real carrier signal `y` with a carrier frequency `fc` and sampling frequency `fs`, using one of the options listed below for *`method`*. (Note that some methods accept an option, `opt`.)

Note Use `demod` and `modulate` in the Signal Processing Toolbox™ with real-valued signals to obtain real-valued outputs. `demod` and `modulate` are not intended to accept complex-valued inputs or produce complex-valued outputs.

Method	Description

`amdsb-sc`
or
`am`
	Amplitude demodulation, double sideband, suppressed carrier. Multiplies `y` by a sinusoid of frequency `fc` and applies a fifth-order Butterworth lowpass filter using `filtfilt`.
	`x = y.*cos(2*pi*fc*t);`
	`[b,a] = butter(5,fc*2/fs);`
	`x = filtfilt(b,a,x);`

`amdsb-tc`	Amplitude demodulation, double sideband, transmitted carrier. Multiplies `y` by a sinusoid of

Method	Description

	frequency `fc`, and applies a fifth-order Butterworth lowpass filter using `filtfilt`.
	`x = y.*cos(2*pi*fc*t);`
	`[b,a] = butter(5,fc*2/fs);`
	`x = filtfilt(b,a,x);`
	If you specify `opt`, `demod` subtracts scalar `opt` from `x`. The default value for `opt` is 0.

`amssb`	Amplitude demodulation, single sideband. Multiplies `y` by a sinusoid of frequency `fc` and applies a fifth-order Butterworth lowpass filter using `filtfilt`.
	`x = y.*cos(2*pi*fc*t);`
	`[b,a] = butter(5,fc*2/fs);`
	`x = filtfilt(b,a,x);`

`fm`	Frequency demodulation. Demodulates the FM waveform by modulating the Hilbert transform of `y` by a complex exponential of frequency `-fc` Hz and obtains the instantaneous frequency of the result.
`pm`	Phase demodulation. Demodulates the PM waveform by modulating the Hilbert transform of `y` by a complex exponential of frequency `-fc` Hz and obtains the instantaneous phase of the result.
`ppm`	Pulse-position demodulation. Finds the pulse positions of a pulse-position modulated signal `y`. For correct demodulation, the pulses cannot overlap. `x` is length `length(t)*fc/fs`.

Method	Description
pwm	Pulse-width demodulation. Finds the pulse widths of a pulse-width modulated signal `y`. `demod` returns in `x` a vector whose elements specify the width of each pulse in fractions of a period. The pulses in `y` should start at the beginning of each carrier period, that is, they should be left justified.

qam	Quadrature amplitude demodulation.
	`[x1,x2] = demod(y,fc,fs,'qam')` multiplies `y` by a cosine and a sine of frequency `fc` and applies a fifth-order Butterworth lowpass filter using `filtfilt`.
	`x1 = y.*cos(2*pi*fc*t);`
	`x2 = y.*sin(2*pi*fc*t);`
	`[b,a] = butter(5,fc*2/fs);`
	`x1 = filtfilt(b,a,x1);`
	`x2 = filtfilt(b,a,x2);`

The default method is `'am'`. In all cases except `'ppm'` and `'pwm'`, `x` is the same size as `y`.

If `y` is a matrix, `demod` demodulates its columns.

`x = demod(y,fc,fs,'pwm','centered')` finds the pulse widths assuming they are centered at the beginning of each period. `x` is length `length(y)*fc/fs`.

See Also

`modulate` | `vco` | `fskdemod` | `genqamdemod` | `mskdemod` | `pamdemod` | `pmdemod` | `qamdemod`

Purpose

Apply design method to filter specification object

Syntax

```
H = design(D)
H = design(D,METHOD)
H = design(D,METHOD,PARAM1,VALUE1,PARAM2,VALUE2,...)
H = design(D,METHOD,OPTS)
Hs = design(D,...,'SystemObject',sysobjflag)
```

Description

`H = design(D)` uses the filter specifications object `D` to generate a filter `H`. When you do not provide a design method as an input argument, `design` uses a default design method. Use `designmethods(D,'default')` to see the default design method for your filter specifications object.

`H = design(D,METHOD)` forces the design method specified by the string `METHOD`. `METHOD` must be one of the strings returned by `designmethods`. Use `designmethods(D,'default')` to determine which algorithm is used by default.

The design method you provide as the `designmethod` input argument must be one of the methods returned by

```
designmethods(d)
```

To help you design filters more quickly, the input argument `METHOD` accepts a variety of special keywords that force `design` to behave in different ways. The following table presents the keywords you can use for `METHOD` and how `design` responds to the keyword.

Designmethod Keyword	Description of the design Response
`'FIR'`	Forces `design` to produce an FIR filter. When no FIR design method exists for object `D`, `design` returns an error.
`'IIR'`	Forces `design` to produce an IIR filter. When no IIR design method exists for object `D`, `design` returns an error.

Designmethod Keyword	Description of the design Response
'ALLFIR'	Produces filters from every applicable FIR design method for the specifications in D, one filter for each design method. As a result, design returns multiple filters in the output object.
'ALLIIR'	Produces filters from every applicable IIR design method for the specifications in D, one filter for each design method. As a result, design returns multiple filters in the output object.
'ALL'	Designs filters using all applicable design methods for the specifications object D. As a result, design returns multiple filters, one for each design method. design uses the design methods in the order that designmethods(D) returns them.

Keywords are not case sensitive

When design returns multiple filters in the output object, use indexing to see the individual filters. For example, to see the third filter in H, enter

```
H(3)
```

H = design(D,METHOD,PARAM1,VALUE1,PARAM2,VALUE2,...) specifies design-method options. Use help(D,METHOD) for complete information on which design-method-specific options are available. You can also use designopts(D,METHOD) for a less-detailed listing of the design-method-specific options.

H = design(D,METHOD,OPTS) specifies design-method options using the structure OPTS. OPTS is usually obtained from designopts and then specified as an input to design. Use help(D,METHOD) for more information on optional inputs.

Hs = design(D,...,'SystemObject',*sysobjflag*) uses the filter specifications object D to generate a filter System object Hs when

sysobjflag is `true`. To generate System objects, you must have the DSP System Toolbox™ product installed. When *sysobjflag* is `false`, the function generates a `dfilt` or `mfilt` object `H`, as described previously. Design methods and design options for filter System objects are not necessarily the same as those for `dfilt` and `mfilt` objects. To check design methods for System objects, use `designmethods` with the `'SystemObject',`*sysobjflag* syntax.

If you are specifying design-method-specific options using `OPTS`, you can also set `OPTS.SystemObject` to `true` instead of calling `design` with the `'SystemObject',`*sysobjflag* syntax.

Examples

Design an FIR equiripple lowpass filter. The passband edge frequency is 0.2π radians/sample, and the stopband edge frequency is 0.25π radians/sample. The passband ripple is 0.5 dB, and the stopband attenuation is 40 dB.

```
D = fdesign.lowpass('Fp,Fst,Ap,Ast',0.2,0.25,0.5,40);
H = design(D); % Uses the default equiripple method.
```

If you have the DSP System Toolbox software installed, you can design a minimum-phase FIR equripple filter. Design a minimum–phase filter and compare the pole-zero plots of the original and minimum-phase designs.

```
Hmin = design(D,'equiripple','MinPhase',true);
hfvt = fvtool([H Hmin],'analysis','polezero');
legend(hfvt,'Original Design','Minimum Phase Design');
```

Design a Butterworth lowpass filter. The passband edge frequency is 0.2π radians/sample, and the stopband edge frequency is 0.25π radians/sample. The passband ripple is 0.5 dB, and the stopband attenuation is 40 dB. Obtain help on the design options specific to the Butterworth design method. Design the filter with the `"MatchExactly"` option set to `'Passband'`.

```
D = fdesign.lowpass('Fp,Fst,Ap,Ast',0.2,0.25,0.5,40);
% Query design-method-specific options
```

```
help(D,'butter')
% Match passband exactly
H = design(D,'butter','MatchExactly','passband');
```

If you have the DSP System Toolbox software, you can specify the *P*-th norm scaling on the second-order sections. Use L-infinity norm scaling in the time domain.

```
H = design(D,'butter','MatchExactly','passband','SOSScaleNorm','linf');
```

If you have the DSP System Toolbox software, you can create a filter System object.

```
Hs = design(D,'SystemObject',true);
```

See Also

`designmethods` | `designopts`

Purpose

Methods available for designing filter from specification object

Syntax

```
M = designmethods(D)
M = designmethods(D,'default')
M = designmethods(D,TYPE)
M = designmethods(D,'full')
Ms = designmethods(D,...,'SystemObject',sysobjflag)
```

Description

`M = designmethods(D)` returns the available design methods for the filter specification object, `D`, and the current value of the `Specification` property.

`M = designmethods(D,'default')` returns the default design method for the filter specification object `D` and the current value of the `Specification` property.

`M = designmethods(D,TYPE)` returns either the `TYPE` design methods that apply to D. `TYPE` can be either `'FIR'` or `'IIR'`.

`M = designmethods(D,'full')` returns the full name for each of the available design methods. For example, `designmethods` with the `'full'` argument returns `Butterworth` for the `butter` method.

`Ms = designmethods(D,...,'SystemObject',sysobjflag)` returns the available design methods for designing filter System objects when *`sysobjflag`* is `true`. To use System objects, you must have the DSP System Toolbox product installed. When *`sysobjflag`* is `false`, the function checks methods for creating `dfilt` and `mfilt` objects, as described previously. Design methods and design options for filter System objects are not necessarily the same as those for `dfilt` and `mfilt` objects.

Examples

Construct a lowpass filter specification object and determine the valid design methods. Obtain detailed command line help on the Chebyshev type I design method.

```
D =fdesign.lowpass('Fp,Fst,Ap,Ast',500,600,0.5,60,1e4);
M = designmethods(D)
help(D,M{2})
```

The last line of the example is equivalent to `help(D,'cheby1')`.

If you have DSP System Toolbox software installed, use the `'SystemObject',sysobjflag` syntax to return design methods for a filter System object:

```
Ms = designmethods(D,'SystemObject',true);
```

See Also

`design` | `designopts` | `fdesign`

Purpose

Valid input arguments and values for specification object and method

Syntax

```
OPTS = designopts(D,METHOD)
```

Description

`OPTS = designopts(D,METHOD)` returns a structure array with the default design parameters used by the design method `METHOD`. `METHOD` must be one of the strings returned by `designmethods`.

Use `help(D,METHOD)` to get a description of the design parameters.

If you have DSP System Toolbox software installed, `OPTS` has the `SystemObject` property if at least one of the structures available for that design method is supported by System objects. However, not all structures for that design method are supported by System objects.

Examples

Create a lowpass filter with a numerator and denominator order of 10 and a 3-dB frequency of 0.2π radians/sample. Obtain the default design parameters for a Butterworth design, and test whether the filter structure is a direct-form II biquad.

```
D = fdesign.lowpass('Nb,Na,F3dB',10,10,0.2);
OPTS = designopts(D,'butter');
if (OPTS.FilterStructure == 'df2sos')
    fprintf('The default filter structure is Direct-Form II\n');
    fprintf('with second-order sections.\n');
end
```

If you have DSP System Toolbox software installed, `OPTS` has the `SystemObject` property.

See Also

`design` | `designmethods` | `fdesign` | `validstructures`

Purpose

Discrete-time filter

Syntax

```
Hd = dfilt.structure(input1,...)
Hd = [dfilt.structure(input1,...),dfilt.structure(input1,...),
   ...]
```

Description

`Hd = dfilt.structure(input1,...)` returns a discrete-time filter, `Hd`, of type *`structure`*. Each structure takes one or more inputs. If you specify a `dfilt.structure` with no inputs, a default filter is created.

Note You must use a *`structure`* with `dfilt`.

`Hd = [dfilt.structure(input1,...),dfilt.structure(input1,...),...]` returns a vector containing `dfilt` filters.

Structures

Available structures for the `dfilt` object are shown below. The target block for the `block` method depends on the filter structure. Depending on the target block, the DSP System Toolbox software may be required.

dfilt.structure	Description	Coefficient Mapping Support in realizemdl	Target Filter Block for block Method
`dfilt.delay`	Delay	Not supported	`Delay`Requires DSP System Toolbox
`dfilt.df1`	Direct-form I	Supported	`Discrete Filter`
`dfilt.df1sos`	Direct-form I, second-order sections	Supported	`Discrete Filter`Requires DSP System Toolbox
`dfilt.df1t`	Direct-form I transposed	Supported	`Discrete Filter`

dfilt.structure	Description	Coefficient Mapping Support in realizemdl	Target Filter Block for block Method
`dfilt.df1tsos`	Direct-form I transposed, second-order sections	Supported	`Biquad Filter` Requires DSP System Toolbox
`dfilt.df2`	Direct-form II	Supported	`Discrete Filter`
`dfilt.df2sos`	Direct-form II, second-order sections	Supported	`Discrete Filter`
`dfilt.df2t`	Direct-form II transposed	Supported	`Discrete Filter`
`dfilt.df2tsos`	Direct-form II transposed, second-order sections	Supported	`Biquad Filter` Requires DSP System Toolbox
`dfilt.dffir`	Direct-form FIR	Supported	`Discrete FIR Filter`
`dfilt.dffirt`	Direct-form FIR transposed	Supported	`Discrete FIR Filter`
`dfilt.dfsymfir`	Direct-form symmetric FIR	Supported	`Discrete FIR Filter`
`dfilt.dfasymfir`	Direct-form antisymmetric FIR	Supported	`Discrete FIR Filter`
`dfilt.fftfir`	Overlap-add FIR	Not supported	`Overlap-Add FFT FilterRequires` DSP System Toolbox
`dfilt.latticeallpass`	Lattice allpass	Supported	Not supported
`dfilt.latticear`	Lattice autoregressive (AR)	Supported	`Allpole FilterRequires` DSP System Toolbox

dfilt.structure	Description	Coefficient Mapping Support in realizemdl	Target Filter Block for block Method
`dfilt.latticearma`	Lattice autoregressive moving- average (ARMA)	Supported	Not supported
`dfilt.latticemamax`	Lattice moving-average (MA) for maximum phase	Supported	Not supported
`dfilt.latticemamin`	Lattice moving-average (MA) for minimum phase	Supported	`Discrete FIR Filter`
`dfilt.statespace`	State-space	Supported.	Not supported
`dfilt.scalar`	Scalar gain object	Supported	`Gain`Requires DSP System Toolbox
`dfilt.cascade`	Filters arranged in series	Supported	Target blocks depend on filter structures in the series
`dfilt.parallel`	Filters arranged in parallel	Supported	Target blocks depend on filter structures in the parallel system

For more information on each structure, use the syntax `help diflt.`*`structure`* at the MATLAB prompt or refer to its reference page.

Methods

Methods provide ways of performing functions directly on your `dfilt` object without having to specify the filter parameters again. You can apply these methods directly on the variable you assigned to your `dfilt` object.

For example, if you create a `dfilt` object, `Hd`, you can check whether it has linear phase with `islinphase(Hd)`, view its frequency response

plot with `fvtool(Hd)`, or obtain its frequency response values with `h=freqz(Hd)`. You can use all of the methods below in this way.

Note If your variable is a 1-D array of `dfilt` filters, the method is applied to each object in the array. Only `freqz`, `grpdelay`, `impz`, `is*`, `order`, and `stepz` methods can be applied to arrays. The `zplane` method can be applied to an array only if it is used without outputs.

Some of the methods listed below have the same name as Signal Processing Toolbox functions and they behave similarly. This is called *overloading* of functions.

Available methods are:

Method	Description
`addstage`	Adds a stage to a `cascade` or `parallel` object, where a stage is a separate, modular filter. See `dfilt.cascade` and `dfilt.parallel`.
`block`	`block(Hd)` creates a Simulink filter block of the `dfilt` object. The target filter block depends on the filter structure. You must have Simulink to use this method. Additionally, the DSP System Toolbox may be required depending on the filter structure. See “Structures” on page 1-146 for a mapping between the target blocks and filter structures. The `block` method can specify these properties/values: `'MapCoeffstoPorts'` indicates whether to map the filter coefficients to constant blocks connected to the generated block. Default value is `'off'`. Setting `'MapCoeffstoPorts'` to `'on'` turns on the mapping and enables

Method	Description
	the 'CoeffNames' property, which defines the constant block parameter names. 'CoeffNames' is a cell array of strings. Default values are {'Num'} for Direct form FIR filters, {'K'} for lattice filters, {'Num','Den'} for IIR filters, and {Num','Den','g'} for biquad filters. Variables, defined by 'CoeffNames', are created in the MATLAB workspace and have the same data type as the filter's 'Arithmetic' property. Any existing variable with the same name is overwritten. Note that you can use either 'Link2Obj' or 'MapCoeffstoPorts', but not both simultaneously. 'InputProcessing' specifies sample-based, 'elementsaschannels', frame-based, 'columnsaschannels', processing, or 'inherited'. The default is frame-based processing. If you do not have the DSP System Toolbox software, explicitly set the 'InputProcessing' property to 'elementsaschannels' to avoid a runtime error. Setting 'InputProcessing' to 'inherited' targets the Digital Filter block regardless of structure.
cascade	Returns the series combination of two dfilt objects. See dfilt.cascade.
coeffs	Returns the filter coefficients in a structure containing fields that use the same property names as those in the original dfilt.
convert	Converts a dfilt object from one filter structure to another filter structure.

Method	Description
`fcfwrite`	Writes a filter coefficient ASCII file. The file can contain a single filter or a vector of objects. If the DSP System Toolbox product is installed, the file can contain multirate filters (`mfilt`) or adaptive filters (`adaptfilt`). Default filename is `untitled.fcf`. `fcfwrite(Hd,filename)` writes to a disk file named `filename` in the current working directory. The `.fcf` extension is added automatically. `fcfwrite(...,fmt)` writes the coefficients in the format `fmt`, where valid `fmt` strings are: `'hex'` for hexadecimal `'dec'` for decimal `'bin'` for binary representation.
`fftcoeffs`	Returns the frequency-domain coefficients used when filtering with a `dfilt.fftfir`.
`filter`	Performs filtering using the `dfilt` object. `y = filter(Hd,x)` filters `x` using the `Hd` filter and returns the filtered data in `y`. See "Using Filter States" on page 1-158 for information on using initial conditions. If `x` is a matrix, each column is filtered as an independent channel. If `x` is a multidimensional array, `filter` operates on the first nonsingleton dimension. `y = filter(Hd,x,dim)` operates along the dimension `dim`. If `x` is a vector or matrix and `dim` is 1, every column of `x` is a channel. If `dim` is 2, every row is a channel.

Method	Description
`firtype`	Returns the type (1-4) of a linear phase FIR filter.
`freqz`	Plots the frequency response in `fvtool`. Note that unlike the `freqz` function, this `dfilt freqz` method has a default length of 8192.
`grpdelay`	Plots the group delay in `fvtool`.
`impz`	Plots the impulse response in `fvtool`.
`impzlength`	Returns the length of the impulse response.
`info`	Displays brief `dfilt` information, such as filter structure, length, stability, linear phase, and, when appropriate, lattice and ladder length. To display detailed information about the design method, options, etc, use `info(Hd, 'long')`. The default display is `'short'`. For multistage filters (`cascade` and `parallel`), use `info(Hd.Stage(x))`, where `x` is the stage number, to see information about that stage.
`isallpass`	Returns a logical `1` (i.e., true) if the `dfilt` object in an allpass filter or a logical `0` (i.e., false) if it is not.
`iscascade`	Returns a logical `1` if the `dfilt` object is cascaded or a logical `0` if it is not.
`isfir`	Returns a logical `1` if the `dfilt` object has finite impulse response (FIR) or a logical `0` if it does not.
`islinphase`	Returns a logical `1` if the `dfilt` object is linear phase or a logical `0` if it is not.
`ismaxphase`	Returns a logical `1` if the `dfilt` object is maximum-phase or a logical `0` if it is not.

Method	Description
`isminphase`	Returns a logical `1` if the `dfilt` object is minimum-phase or a logical `0` if it is not.
`isparallel`	Returns a logical `1` if the `dfilt` object has parallel stages or a logical `0` if it does not.
`isreal`	Returns a logical `1` if the `dfilt` object has real-valued coefficients or a logical 0 if it does not.
`isscalar`	Returns a logical `1` if the `dfilt` object is a scalar or a logical `0` if it is not scalar.
`issos`	Returns a logical `1` if the `dfilt` object has second-order sections or a logical `0` if it does not.
`isstable`	Returns a logical `1` if the `dfilt` object is stable or a logical `0` if it are not.
`nsections`	Returns the number of sections in a second-order sections filter. If a multistage filter contains stages with multiple sections, using `nsections` returns the total number of sections in all the stages (a stage with a single section returns 1).
`nstages`	Returns the number of stages of the filter, where a stage is a separate, modular filter.
`nstates`	Returns the number of states for an object.
`order`	Returns the filter order. If `Hd` is a single-stage filter, the order is given by the number of delays needed for a minimum realization of the filter. If `Hd` has multiple stages, the order is given by the number of delays needed for a minimum realization of the overall filter.
`parallel`	Returns the parallel combination of two `dfilt` filters. See `dfilt.parallel`.
`phasez`	Plots the phase response in `fvtool`.

Method	Description
`realizemdl`	(Available only with Simulink software.) `realizemdl(Hd)` creates a Simulink model containing a subsystem block realization of your `dfilt`. `realizemdl(Hd,p1,v1,p2,v2,...)` creates the block using the properties `p1`, `p2`,... and values `v1`, `v2`,.. specified. The following properties are available: `'Blockname'` specifies the name of the block. The default value is `'Filter'`. `'Destination'` specifies whether to add the block to a current Simulink model, create a new model, or place the block in an existing subsystem in your model. Valid values are `'current'`, `'new'`, or the name of an existing subsystem in your model. Default value is `'current'`. `'OverwriteBlock'` specifies whether to overwrite an existing block that was created by `realizemdl` or create a new block. Valid values are `'on'` and `'off'` and the default is `'off'`. Note that only blocks created by `realizemdl` are overwritten. The following properties optimize the block structure. Specifying `'on'` turns the optimization on and `'off'` creates the block without optimization. The default for each of the following is `'on'`. `'OptimizeZeros'` removes zero-gain blocks. `'OptimizeOnes'` replaces unity-gain blocks with a direct connection.

Method	Description
	`'OptimizeNegOnes'` replaces negative unity-gain blocks with a sign change at the nearest summation block. `'OptimizeDelayChains'` replaces cascaded chains of delay block with a single integer delay block set to the appropriate delay.
`removestage`	Removes a stage from a cascade or parallel `dfilt`. See `dfilt.cascade` and `dfilt.parallel`.
`setstage`	Overwrites a stage of a cascade or parallel `dfilt`. See `dfilt.cascade` and `dfilt.parallel`.
`sos`	Converts the `dfilt` to a second-order sections `dfilt`. If `Hd` has a single section, the returned filter has the same class. `sos(Hd,flag)` specifies the ordering of the second-order sections. If `flag='UP'`, the first row contains the poles closest to the origin, and the last row contains the poles closest to the unit circle. If `flag='down'`, the sections are ordered in the opposite direction. The zeros are always paired with the poles closest to them. `sos(Hd,flag,scale)` specifies the scaling of the gain and the numerator coefficients of all second-order sections. `scale` can be `'none'`, `'inf'` (infinity-norm) or `'two'` (2-norm). Using infinity-norm scaling with `up` ordering minimizes the probability of overflow in the realization. Using 2-norm scaling with `down` ordering minimizes the peak roundoff noise.

Method	Description
ss	Converts the dfilt to state-space. To see the separate A,B,C,D matrices for the state-space model, use [A,B,C,D]=ss(Hd).
stepz	Plots the step response in fvtool. stepz(Hd,n) computes the first n samples of the step response. stepz(Hd,n,Fs) separates the time samples by T = 1/Fs, where Fs is assumed to be in Hz.
sysobj	Converts the dfilt to a filter System object. See the reference page for a list of supported objects. To use this method, you must have DSP System Toolbox software installed.
tf	Converts the dfilt to a transfer function.
zerophase	Plots the zero-phase response in fvtool.
zpk	Converts the dfilt to zeros-pole-gain form.
zplane	Plots a pole-zero plot in fvtool.

For more information on each method, use the syntax `help diflt/`*`method`* at the MATLAB prompt.

Viewing Properties

As with any object, you can use `get` to view a `dfilt` properties. To see a specific property, use

```
get(Hd,'property')
```

To see all properties for an object, use

```
get(Hd)
```

Changing Properties

To set specific properties, use

```
set(Hd,'property1',value,'property2',value,...)
```

Note that you must use single quotation marks around the property name.

Alternatively, you can get or set a property value with `Object.property`:

```
b = [0.05 0.9 0.05];
Hd = dfilt.dffir(b);
% Lowpass direct-form I FIR filter
Hd.arithmetic % get arithmetic property
% returns double
Hd.arithmetic = 'single';
% Set arithmetic property to single precision
```

Copying an Object

To create a copy of an object, use the `copy` method.

```
H2 = copy(Hd)
```

Note Using the syntax `H2 = Hd` copies only the object handle and does not create a new object.

Converting Between Filter Structures

To change the filter structure of a `dfilt` object `Hd`, use

```
Hd2=convert(Hd,'structure_string');
```

where `structure_string` is any valid structure name in single quotation marks. If `Hd` is a `cascade` or `parallel` structure, each of its stages is converted to the new structure.

Using Filter States

Two properties control the 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. For `df1`, `df1t`, `df1sos` and `df1tsos` structures, `states` returns a `filtstate` object.
- `PersistentMemory` — controls whether filter `states` are saved. The default value is `'false'`, which causes the initial conditions to be reset to zero before filtering and turns off the display of `states` information. Setting `PersistentMemory` to `'true'` allows the filter to use your initial conditions or to reuse the final conditions of a previous filtering operation as the initial conditions of the next filtering operation. It also displays information about the filter `states`.

Note If you set `states` and want to use them for filtering, you must set `PersistentMemory` to `'true'` before you use the filter.

Examples

Create a direct-form I filter and use a method to see if it is stable.

```
[b,a] = butter(8,0.25);
Hd = dfilt.df1(b,a)
```

If a `dfilt`'s numerator values do not fit on a single line, a description of the vector is displayed. To see the specific numerator values for this example, use

```
get(Hd,'numerator')
```

or alternatively

```
Hd.numerator
```

Refer to the reference pages for each structure for more examples.

See Also

dfilt.cascade | dfilt.df1 | dfilt.df1t | dfilt.df2 | dfilt.df2t | dfilt.dfasymfir | dfilt.dffir | dfilt.dffirt | dfilt.dfsymfir | dfilt.latticeallpass | dfilt.latticear | dfilt.latticearma | dfilt.latticemamax | dfilt.latticemamin | dfilt.parallel | dfilt.statespace | filter | freqz | grpdelay | impz | step | tf | zpk | zplane

dfilt.cascade

Purpose

Cascade of discrete-time filters

Syntax

```
Hd = dfilt.cascade(Hd1,Hd2,...)
```

Description

`Hd = dfilt.cascade(Hd1,Hd2,...)` returns a discrete-time filter, `Hd`, of type `cascade`, which is a serial interconnection of two or more `dfilt` filters, `Hd1`, `Hd2`, etc. Each filter in a cascade is a separate stage.

To add a filter (`Hd1`) to the end of an existing cascade (`Hd`), use

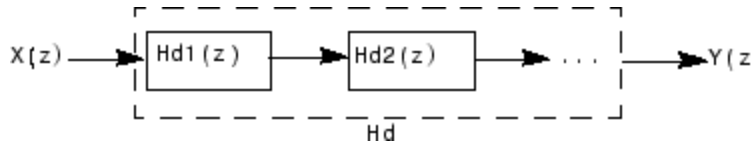
```
addstage(Hd,Hd1)
```

and to reorder the filters in a cascade, use the stage indices to indicate the desired ordering, such as.

```
Hd.stage = Hd.stage([1,3,2]);
```

You can also use the nondot notation format for calling a cascade:

```
cascade(Hd1,Hd2,...)
```



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-passband .4-.6
```

To view details of the first stage, use

```
info(Hcas.Stage(1))
```

To view the states of a stage, use

```
Hcas.stage(1).states
```

You can display states for individual stages only.

See Also

`dfilt` | `dfilt.parallel` | `dfilt.scalar`

dfilt.delay

Purpose

Delay filter

Syntax

```
Hd = dfilt.delay
Hd = dfilt.delay(latency)
```

Description

`Hd = dfilt.delay` returns a discrete-time filter, `Hd`, of type `delay`, which adds a single delay to any signal filtered with `Hd`. The filtered signal has its values shifted by one sample.

`Hd = dfilt.delay(latency)` returns a discrete-time filter, `Hd`, of type `delay`, which adds the number of delay units specified in `latency` to any signal filtered with `Hd`. The filtered signal has its values shifted by the `latency` number of samples. The values that appear before the shifted signal are the filter states.

Examples

Create a `delay` filter with a `latency` of 4 and filter a simple signal to view the impact of applying a delay.

```
h = dfilt.delay(4)
h =
     FilterStructure: 'Delay'
             Latency: 4
    PersistentMemory: false

sig = 1:7      % Create some simple signal data
sig =
     1     2     3     4     5     6     7

states = h.states    % Filter states before filtering
states =
     0
     0
     0
     0

filter(h,sig)       % Filter using the delay filter
ans =
```

```
     0     0     0     0     1     2     3

states=h.states     % Filter states after filtering
states =
     4
     5
     6
     7
```

See Also

dfilt

dfilt.df1

Purpose

Discrete-time, direct-form I filter

Syntax

```
Hd = dfilt.df1(b,a)
Hd = dfilt.df1
```

Description

`Hd = dfilt.df1(b,a)` returns a discrete-time, direct-form I filter, `Hd`, with numerator coefficients `b` and denominator coefficients `a`. The filter states for this object are stored in a `filtstates` object.

`Hd = dfilt.df1` returns a default, discrete-time, direct-form I filter, `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.

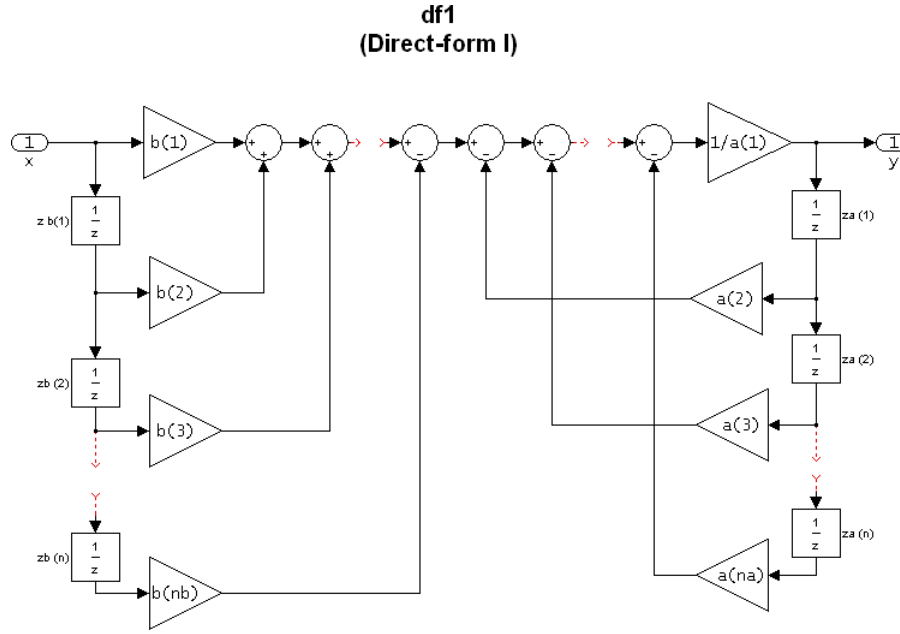


Image of direct form one filter diagram

To display the filter states, use this code to access the `filtstates` object.

```
Hs = Hd.states        % Where Hd is the dfilt.df1 object and
double (Hs)           % Hs is the filtstates object
```

The vector is

$$\begin{bmatrix} zb(1) \\ zb(2) \\ \cdots \\ zb(n) \\ za(1) \\ za(2) \\ \cdots \\ za(n) \end{bmatrix}$$

Examples

Create a direct-form I discrete-time filter with coefficients from a fourth-order lowpass Butterworth design

```
[b,a] = butter(4,.5);
Hd = dfilt.df1(b,a)
```

See Also

`dfilt` | `dfilt.df1t` | `dfilt.df2` | `dfilt.df2t`

Purpose

Discrete-time, second-order section, direct-form I filter

Syntax

```
Hd = dfilt.df1sos(s)
Hd = dfilt.df1sos(b1,a1,b2,a2,...)
Hd = dfilt.df1sos(...,g)
Hd = dfilt.df1sos
```

Description

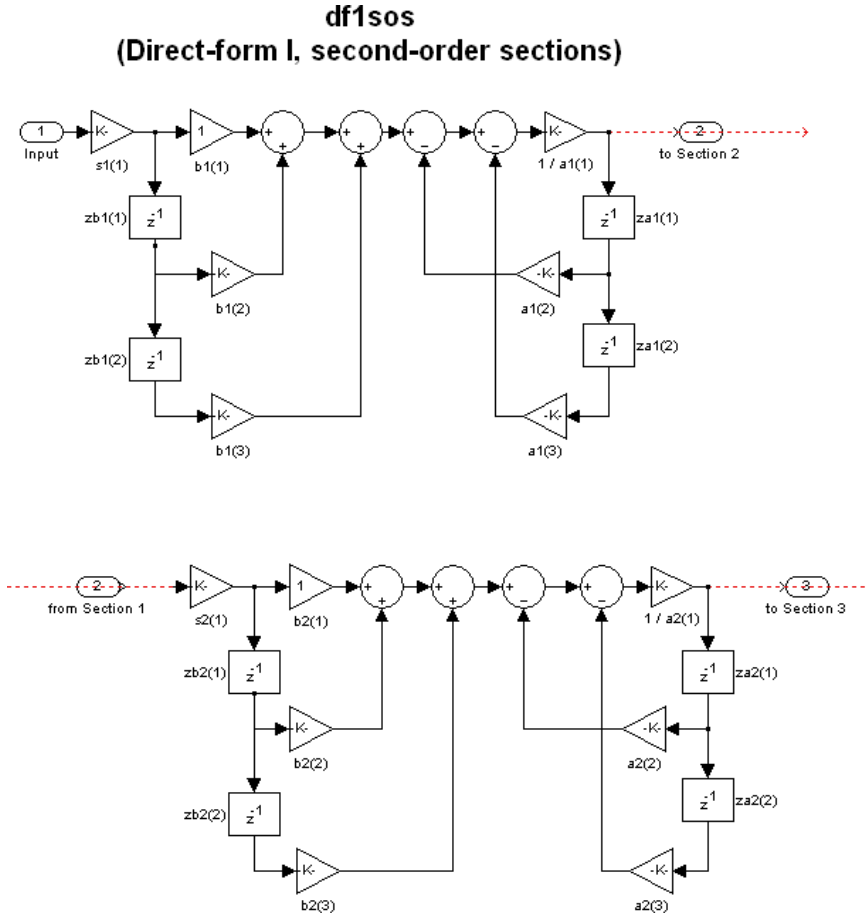
`Hd = dfilt.df1sos(s)` returns a discrete-time, second-order section, direct-form I filter, `Hd`, with coefficients given in the `s` matrix. The filter states for this object are stored in a `filtstates` object.

`Hd = dfilt.df1sos(b1,a1,b2,a2,...)` returns a discrete-time, second-order section, direct-form I filter, `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.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. If `g` is not specified, all gains default to one.

`Hd = dfilt.df1sos` returns a default, discrete-time, second-order section, direct-form I filter, `Hd`. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0.



To display the filter states, use this code to access the `filtstates` object.

```
Hs = Hd.states      % Where Hd is the dfilt.df1 object and
double (Hs)         % Hs is the filtstates object
```

The vector is

$$\begin{pmatrix} zb1(1) & zb2(1) \\ zb1(2) & zb2(2) \\ za1(1) & za2(1) \\ za1(2) & za2(2) \end{pmatrix}$$

For filters with more than one section, each section is a separate column in the matrix.

Examples

Specify a second-order sections, direct-form I discrete-time filter with coefficients from a sixth order, lowpass, elliptical filter using the following code. The resulting filter has three sections.

```
[z,p,k] = ellip(6,1,60,.4);  % Obtain filter coefficients
[s,g] = zp2sos(z,p,k);       % Convert to SOS
Hd = dfilt.df1sos(s,g)
```

See Also

`dfilt` | `dfilt.df1tsos` | `dfilt.df2sos` | `dfilt.df2tsos`

dfilt.df1t

Purpose

Discrete-time, direct-form I transposed filter

Syntax

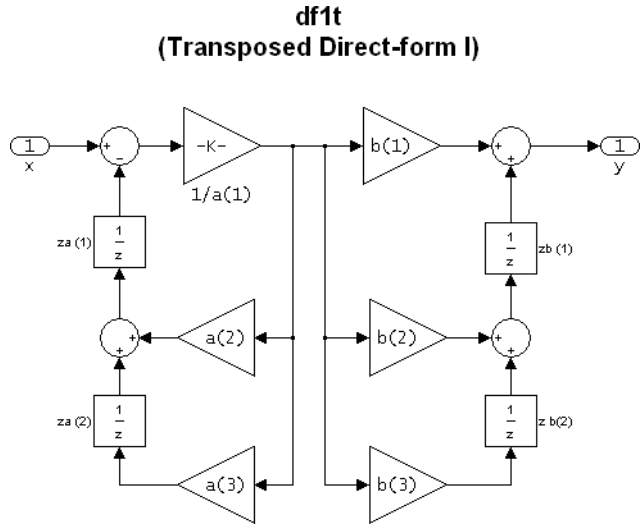
```
Hd = dfilt.df1t(b,a)
Hd = dfilt.df1t
```

Description

`Hd = dfilt.df1t(b,a)` returns a discrete-time, direct-form I transposed filter, `Hd`, with numerator coefficients `b` and denominator coefficients `a`. The filter states for this object are stored in a `filtstates` object.

`Hd = dfilt.df1t` returns a default, discrete-time, direct-form I transposed filter, `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 display the filter states, use this code to access the `filtstates` object.

```
Hs = Hd.states     % Where Hd is the dfilt.df1 object and
double (Hs)        % Hs is the filtstates object
```

The vector of states is:

$$\begin{pmatrix} zb(1) \\ zb(2) \\ \cdots \\ zb(M) \\ za(1) \\ za(2) \\ \cdots \\ za(N) \end{pmatrix}$$

Alternatively, you can access the states in the filtstates object:

```
b = [0.05 0.9 0.05];
Hd = dfilt.df1t(b,1);
Hd.States
% Returns
%  Numerator: [2x1 double]
%  Denominator: [0x1 double]
Hd.States.Numerator(1)=1; %Set zb(1) equal to 1.
```

Examples

Create a direct-form I transposed discrete-time filter with coefficients from a fourth-order lowpass Butterworth design:

```
[b,a] = butter(4,.5);
Hd = dfilt.df1t(b,a)
```

See Also

`dfilt` | `dfilt.df1` | `dfilt.df2` | `dfilt.df2t`

dfilt.df1tsos

Purpose

Discrete-time, second-order section, direct-form I transposed filter

Syntax

```
Hd = dfilt.df1tsos(s)
Hd = dfilt.df1tsos(b1,a1,b2,a2,...)
Hd = dfilt.df1tsos(...,g)
Hd = dfilt.df1tsos
```

Description

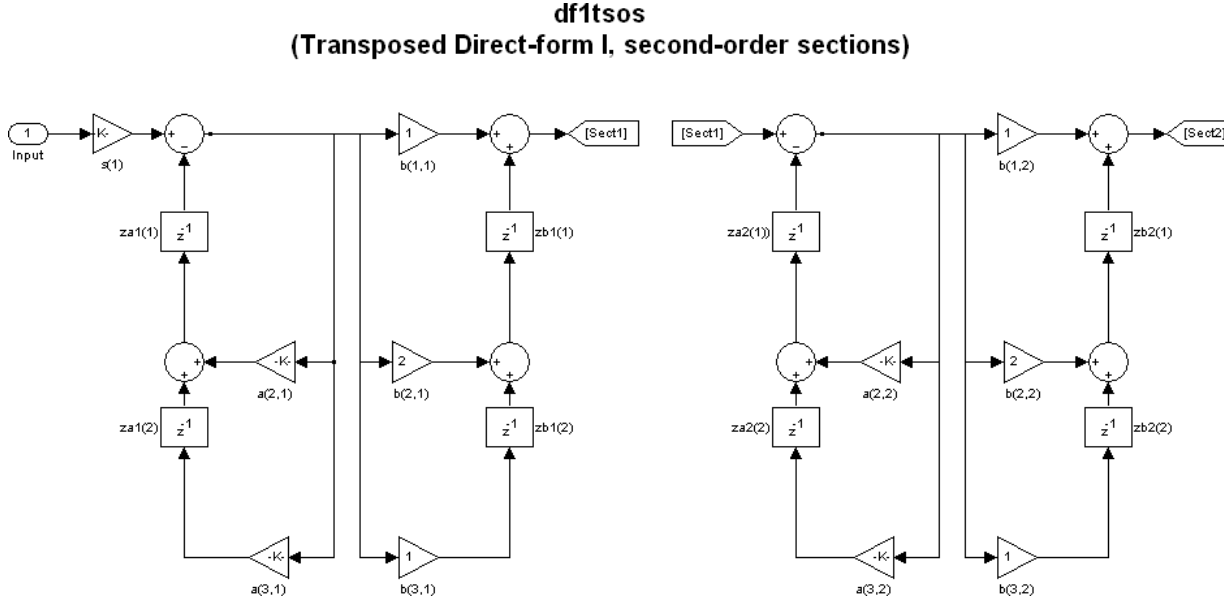
`Hd = dfilt.df1tsos(s)` returns a discrete-time, second-order section, direct-form I, transposed filter, `Hd`, with coefficients given in the `s` matrix. The filter states for this object are stored in a `filtstates` object.

`Hd = dfilt.df1tsos(b1,a1,b2,a2,...)` returns a discrete-time, second-order section, direct-form I, tranposed filter, `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, `Hd`. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0.



To display the filter states, use this code to access the `filtstates` object.

```
Hs = Hd.states      % Where Hd is the dfilt.df1 object and
double (Hs)         % Hs is the filtstates object
```

The matrix is

$$\begin{pmatrix} zb1(1) & zb2(1) \\ zb1(2) & zb2(2) \\ za1(1) & za2(1) \\ za1(2) & za2(2) \end{pmatrix}$$

Examples

Specify a second-order sections, direct-form I, transposed discrete-time filter with coefficients from a sixth order, lowpass, elliptical filter using the following code:

```
[z,p,k] = ellip(6,1,60,.4);    % Obtain filter coefficients
[s,g] = zp2sos(z,p,k);         % Convert to SOS
Hd = dfilt.df1tsos(s,g)
```

See Also

dfilt | dfilt.df1sos | dfilt.df2sos | dfilt.df2tsos

Purpose

Discrete-time, direct-form II filter

Syntax

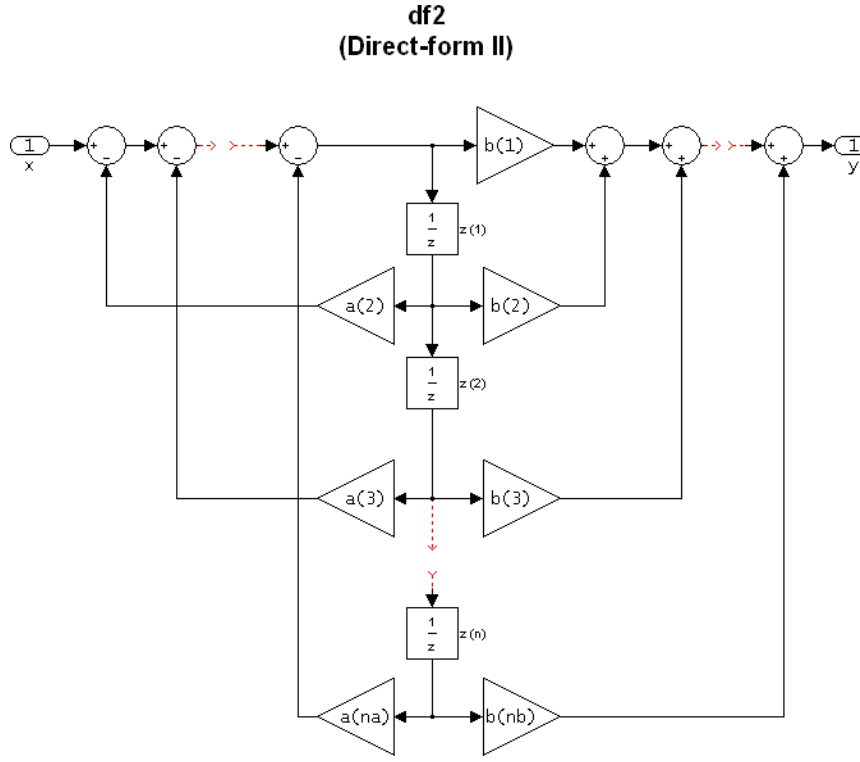
```
Hd = dfilt.df2(b,a)
Hd = dfilt.df2
```

Description

`Hd = dfilt.df2(b,a)` returns a discrete-time, direct-form II filter, `Hd`, with numerator coefficients `b` and denominator coefficients `a`.

`Hd = dfilt.df2` returns a default, discrete-time, direct-form II filter, `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.



The resulting filter states column vector is

$$\begin{bmatrix} z(1) \\ z(2) \\ \cdots \\ z(n) \end{bmatrix}$$

Examples

Create a direct-form II discrete-time filter with coefficients from a fourth-order lowpass Butterworth design:

```
[b,a] = butter(4,.5);
Hd = dfilt.df2(b,a)
```

See Also dfilt | dfilt.df1 | dfilt.df1t | dfilt.df2t

dfilt.df2sos

Purpose

Discrete-time, second-order section, direct-form II filter

Syntax

```
Hd = dfilt.df2sos(s)
Hd = dfilt.df2sos(b1,a1,b2,a2,...)
Hd = dfilt.df2sos(...,g)
Hd = dfilt.df2sos
```

Description

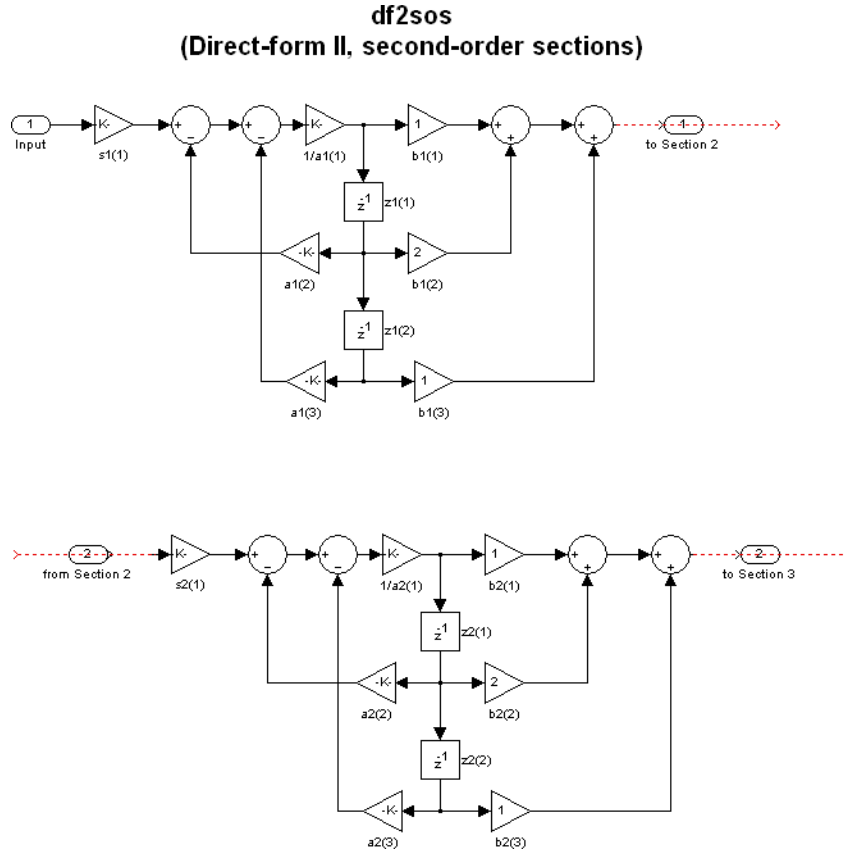
`Hd = dfilt.df2sos(s)` returns a discrete-time, second-order section, direct-form II filter, `Hd`, with coefficients given in the `s` matrix.

`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, `Hd`. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0.



The resulting filter states column vector is

$$\begin{pmatrix} z1(1) & z2(1) \\ z1(2) & z2(2) \end{pmatrix}$$

For filters with more than one section, each section is a separate column in the vector.

Examples

Specify a second-order sections, direct-form II discrete-time filter with coefficients from a sixth order, lowpass, elliptical filter using the following code:

```
[z,p,k] = ellip(6,1,60,.4);   % Obtain filter coefficients
[s,g] = zp2sos(z,p,k);        % Convert to SOS
Hd = dfilt.df2sos(s,g)
```

See Also

`dfilt` | `dfilt.df1sos` | `dfilt.df1tsos` | `dfilt.df2tsos`

Purpose

Discrete-time, direct-form II transposed filter

Syntax

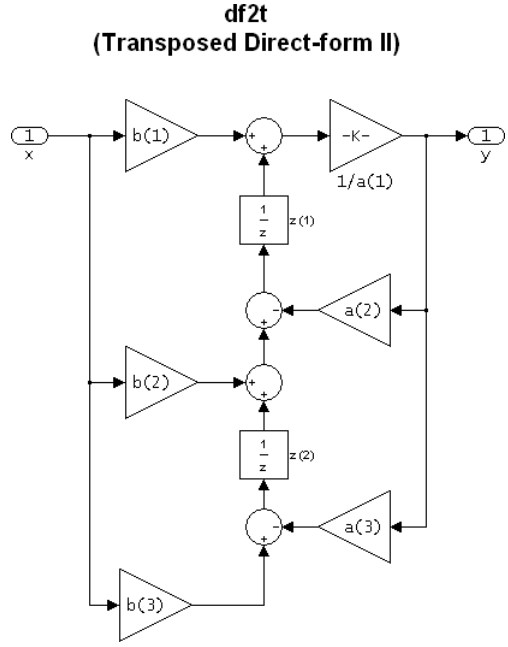
```
Hd = dfilt.df2t(b,a)
Hd = dfilt.df2t
```

Description

`Hd = dfilt.df2t(b,a)` returns a discrete-time, direct-form II transposed filter, `Hd`, with numerator coefficients `b` and denominator coefficients `a`.

`Hd = dfilt.df2t` returns a default, discrete-time, direct-form II transposed filter, `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.



The filter states of `dfilt.df2t` object can be extracted as a column vector with:

```
b =[1 2];
a =[1 -0.9];
Hd = dfilt.df2t(b,a);
FiltStates = double(Hd.States);
```

The resulting filter states column vector is

$$\begin{pmatrix} z(1) \\ z(2) \end{pmatrix}$$

Examples

Create a direct-form II transposed discrete-time filter with coefficients from a 4–th order lowpass Butterworth design:

```
[b,a] = butter(4,.5);
Hd = dfilt.df2t(b,a);
```

See Also

`dfilt` | `dfilt.df1` | `dfilt.df1t` | `dfilt.df2`

Purpose

Discrete-time, second-order section, direct-form II transposed filter

Syntax

```
Hd = dfilt.df2sos(s)
Hd = dfilt.df2tsos(b1,a1,b2,a2,...)
Hd = dfilt.df2tsos(...,g)
Hd = dfilt.df2tso
```

Description

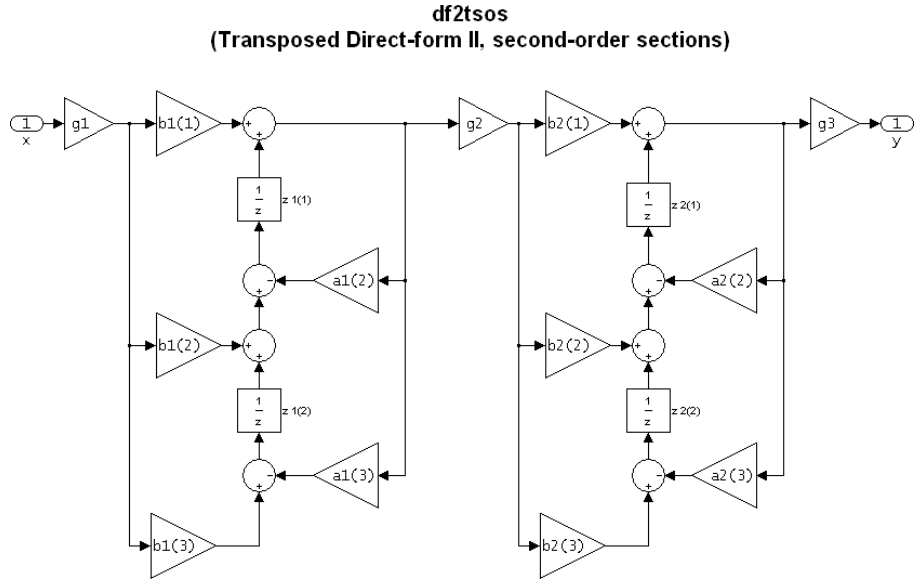
`Hd = dfilt.df2sos(s)` returns a discrete-time, second-order section, direct-form II, transposed filter, `Hd`, with coefficients given in the `s` matrix.

`Hd = dfilt.df2tsos(b1,a1,b2,a2,...)` returns a discrete-time, second-order section, direct-form II, tranposed filter, `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.df2tso` returns a default, discrete-time, second-order section, direct-form II, transposed filter, `Hd`. This filter passes the input through to the output unchanged.

Note The leading coefficient of the denominator `a(1)` cannot be 0.



The resulting filter states column vector is

$$\begin{pmatrix} z1(1) & z2(1) \\ z1(2) & z2(2) \end{pmatrix}$$

Examples

Specify a second-order sections, direct-form II, transposed discrete-time filter with coefficients from a sixth order, lowpass, elliptical filter using the following code:

```
[z,p,k] = ellip(6,1,60,.4);  % Obtain filter coefficients
[s,g] = zp2sos(z,p,k);       % Convert to SOS
Hd = dfilt.df2tsos(s,g)
```

See Also

`dfilt` | `dfilt.df1sos` | `dfilt.df1tsos` | `dfilt.df2sos`

Purpose

Discrete-time, direct-form antisymmetric FIR filter

Syntax

```
Hd = dfilt.dfasymfir(b)
Hd = dfilt.dfasymfir
```

Description

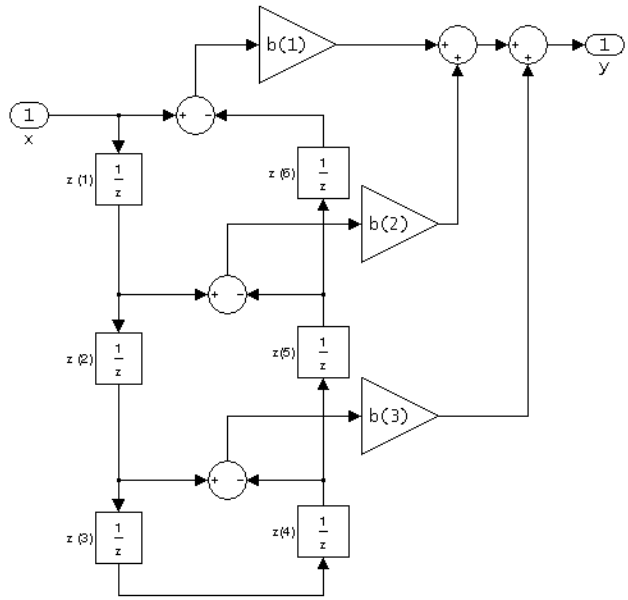
`Hd = dfilt.dfasymfir(b)` returns a discrete-time, direct-form, antisymmetric FIR filter, `Hd`, with numerator coefficients `b`.

`Hd = dfilt.dfasymfir` returns a default, discrete-time, direct-form, antisymmetric FIR filter, `Hd`, with `b`=1. This filter passes the input through to the output unchanged.

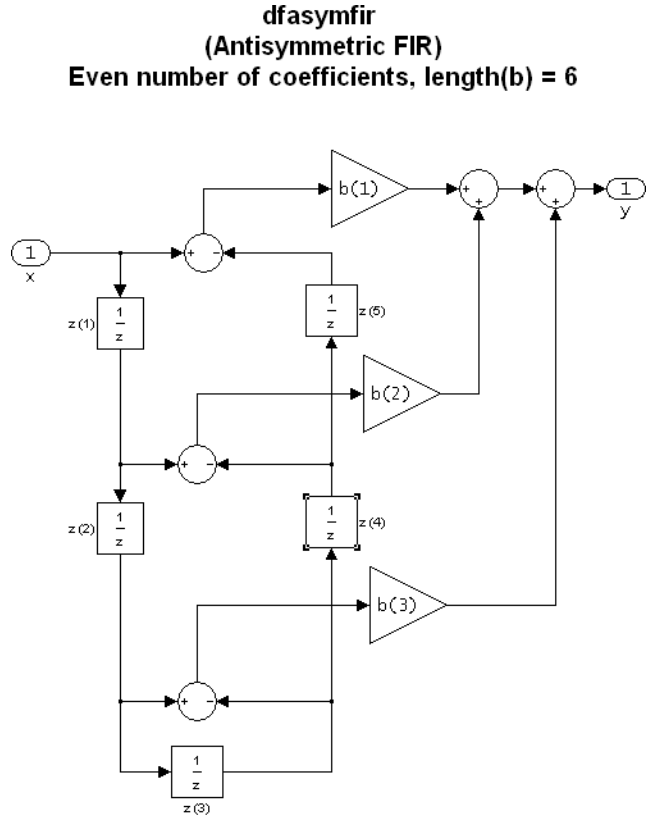
Note Only the first half of vector `b` is used because the second half is assumed to be antisymmetric. In the figure below for an odd number of coefficients, b(3) = 0, b(4) = –b(2) and b(5) = –b(1), and in the next figure for an even number of coefficients, b(4) = –b(3), b(5) = –b(2), and b(6) = –b(1).

---

dfasymfir
(Antisymmetric FIR)
Even order
Odd number of coefficients, length(b) = 7



Note that antisymmetry is defined as
b(i) == -b(end - i + 1)
so that the middle coefficient is zero for odd length
b((end+1)/2) = 0



The resulting filter states column vector for the odd number of coefficients example above is

$$\begin{bmatrix} z(1) \\ z(2) \\ z(3) \\ z(4) \\ z(5) \\ z(6) \end{bmatrix}$$

Examples

Odd Order

Create a Type 4 25^{th} order highpass direct-form antisymmetric FIR filter structure for a `dfilt` object, `Hd`, with the following code:

```
Num_coeffs = firpm(25,[0 .4 .5 1],[0 0 1 1],'h');
Hd = dfilt.dfasymfir(Num_coeffs);
```

Even Order

Create a 44^{th} order lowpass direct-form antisymmetric FIR differentiator filter structure for a `dfilt` object, `Hd`, with the following code:

```
Num_coeffs = firpm(44,[0 .3 .4 1],[0 .2 0 0],'differentiator');
Hd = dfilt.dfasymfir(Num_coeffs);
```

See Also

`dfilt` | `dfilt.dffir` | `dfilt.dffirt` | `dfilt.dfsymfir`

Purpose

Discrete-time, direct-form, FIR filter

Syntax

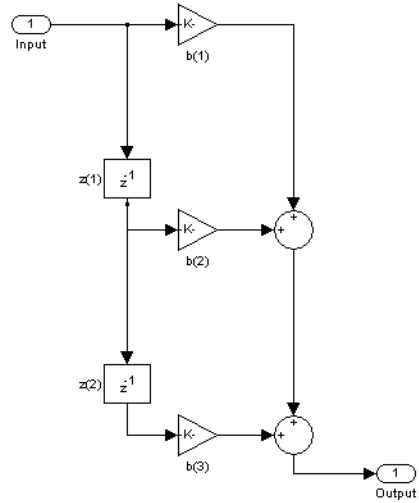
```
Hd = dfilt.dffir(b)
Hd = dfilt.dffir
```

Description

`Hd = dfilt.dffir(b)` returns a discrete-time, direct-form finite impulse response (FIR) filter, `Hd`, with numerator coefficients, `b`.

`Hd = dfilt.dffir` returns a default, discrete-time, direct-form FIR filter, `Hd`, with `b=1`. This filter passes the input through to the output unchanged.

dffir
(Direct-form FIR = Tapped delay line)



The resulting filter states column vector is

$$\begin{pmatrix} z(1) \\ z(2) \end{pmatrix}$$

Examples

Create a direct-form FIR discrete-time filter with coefficients from a 30^{th} order lowpass equiripple design:

```
b = firpm(30,[0 .1 .2 .5]*2,[1 1 0 0]);
Hd = dfilt.dffir(b)
```

See Also

dfilt | dfilt.dfasymfir | dfilt.dffirt | dfilt.dfsymfir

Purpose

Discrete-time, direct-form FIR transposed filter

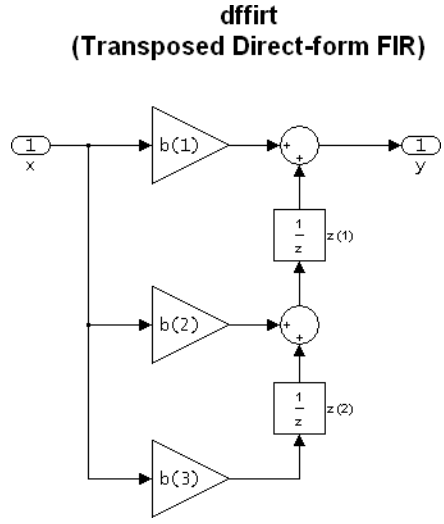
Syntax

```
Hd = dfilt.dffirt(b)
Hd = dfilt.dffirt
```

Description

`Hd = dfilt.dffirt(b)` returns a discrete-time, direct-form FIR transposed filter, `Hd`, with numerator coefficients `b`.

`Hd = dfilt.dffirt` returns a default, discrete-time, direct-form FIR transposed filter, `Hd`, with `b`=1. This filter passes the input through to the output unchanged.



The resulting filter states column vector is

$$\begin{pmatrix} z(1) \\ z(2) \end{pmatrix}$$

Examples

Create a direct-form FIR transposed discrete-time filter with coefficients from a 30^{th} order lowpass equiripple design:

```
b = firpm(30,[0 .1 .2 .5]*2,[1 1 0 0]);
Hd = dfilt.dffirt(b)
```

See Also

dfilt | dfilt.dffir | dfilt.dfasymfir | dfilt.dfsymfir

Purpose

Discrete-time, direct-form symmetric FIR filter

Syntax

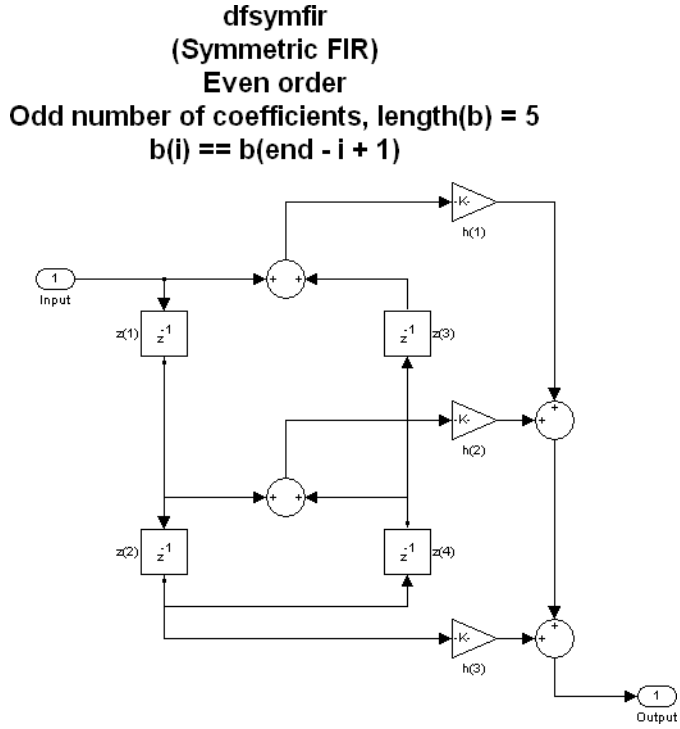
```
Hd = dfilt.dfsymfir(b)
Hd = dfilt.dfsymfir
```

Description

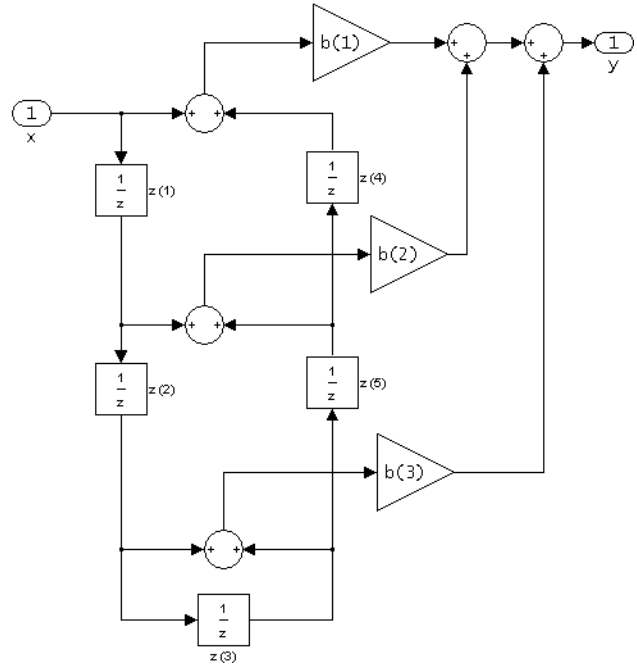
`Hd = dfilt.dfsymfir(b)` returns a discrete-time, direct-form symmetric FIR filter, `Hd`, with numerator coefficients `b`.

`Hd = dfilt.dfsymfir` returns a default, discrete-time, direct-form symmetric FIR filter, `Hd`, with `b=1`. This filter passes the input through to the output unchanged.

Note Only the first half of vector `b` is used because the second half is assumed to be symmetric. In the figure below for an odd number of coefficients, b(3) = 0, b(4) = b(2) and b(5) = b(1), and in the next figure for an even number of coefficients, b(4) = b(3), b(5) = b(2), and b(6) = b(1).


dfsymfir
(Symmetric FIR)
Even order
Odd number of coefficients, length(b) = 5
b(i) == b(end - i + 1)
1
Input
z(1)
z(2)
z(3)
z(4)
h(1)
h(2)
h(3)
1
Output

dfsymfir
(Symmetric FIR)
Odd order
Even number of coefficients, length(b) = 6
b(i) == b(end - i + 1)



The resulting filter states column vector for the odd number of coefficients example above is

$$\begin{bmatrix} z(1) \\ z(2) \\ z(3) \\ z(4) \end{bmatrix}$$

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)
```

Even Order

Specify a fourth-order 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)
```

See Also

`dfilt` | `dfilt.dfasymfir` | `dfilt.dffir` | `dfilt.dffirt`

Purpose

Discrete-time, overlap-add, FIR filter

Syntax

```
Hd = dfilt.fftfir(b,len)
Hd = dfilt.fftfir(b)
Hd = dfilt.fftfir
```

Description

This object uses the overlap-add method of block FIR filtering, which is very efficient for streaming data.

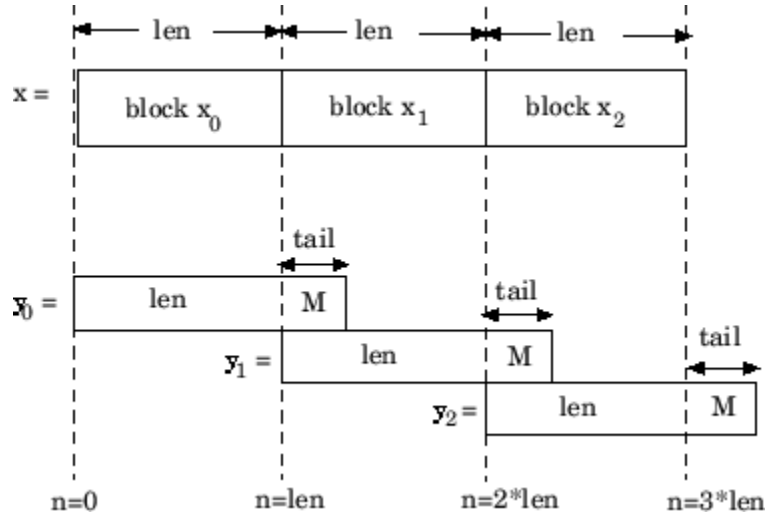
`Hd = dfilt.fftfir(b,len)` returns a discrete-time, FFT, FIR filter, `Hd`, with numerator coefficients, `b` and block length, `len`. The block length is the number of input points to use for each overlap-add computation.

`Hd = dfilt.fftfir(b)` returns a discrete-time, FFT, FIR filter, `Hd`, with numerator coefficients, `b` and block length, `len`=100.

`Hd = dfilt.fftfir` returns a default, discrete-time, FFT, FIR filter, `Hd`, with the numerator `b=1` and block length, `len`=100. This filter passes the input through to the output unchanged.

Note When you use a `dfilt.fftfir` object to filter data, the filter always operates on a segment of the signal equal in length to an integer multiple of the object's block length, `len`. If the input signal length is not equal to an integer multiple of the block length, the signal length is truncated to the nearest integer satisfying this requirement. If the `PersistentMemory` property is set to `true`, the next time you use the filter object the remaining signal samples are prepended to the subsequent input. The resulting number of FFT points = (filter length + the block length - 1). The filter is most efficient if the number of FFT points is a power of 2.

The `fftfir` uses an overlap-add block processing algorithm, which is represented as follows,



where `len` is the block length and `M` is the length of the numerator-1, (`length(b)-1`), which is also the number of states. The output of each convolution is a block that is longer than the input block by a tail of (`length(b)-1`) samples. These tails overlap the next block and are added to it. The states reported by `dfilt.fftfir` are the tails of the final convolution.

Examples

Create an FFT FIR discrete-time filter with coefficients from a 30th order lowpass equiripple design:

```
b = firpm(30,[0 .1 .2 .5]*2,[1 1 0 0]);
Hd = dfilt.fftfir(b)
```

To view the frequency domain coefficients used in the filtering, use the following command.

```
freq_coeffs = fftcoeffs(Hd);
```

See Also

`dfilt | dfilt.dffir | dfilt.dfasymfir | dfilt.dffirt | dfilt.dfsymfir`

Purpose

Discrete-time, lattice allpass filter

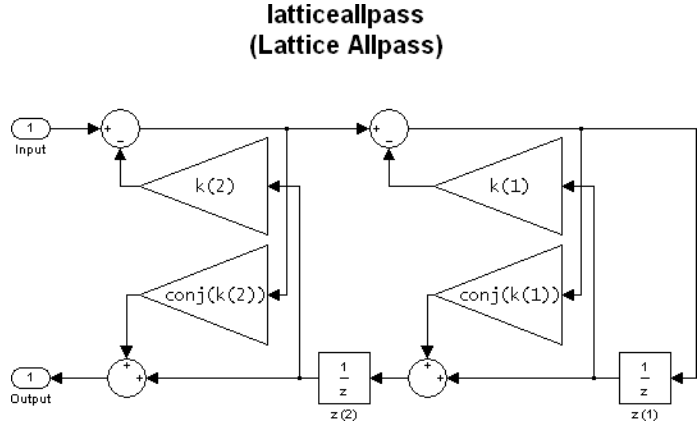
Syntax

```
Hd = dfilt.latticeallpass(k)
Hd = dfilt.latticeallpass
```

Description

`Hd = dfilt.latticeallpass(k)` returns a discrete-time, lattice allpass filter, `Hd`, with lattice coefficients, `k`.

`Hd = dfilt.latticeallpass` returns a default, discrete-time, lattice allpass filter, `Hd`, with `k=[ ]`. This filter passes the input through to the output unchanged.



The resulting filter states column vector `Hd.States` is

$$\begin{pmatrix} z(1) \\ z(2) \end{pmatrix}$$

Examples

Form a third-order lattice allpass filter structure for a `dfilt` object, `Hd`, using the following lattice coefficients:

```
k = [.66 .7 .44];
Hd = dfilt.latticeallpass(k)
```

See Also

dfilt | dfilt.latticear | dfilt.latticearma | dfilt.latticemamax | dfilt.latticemamin

Purpose

Discrete-time, lattice, autoregressive filter

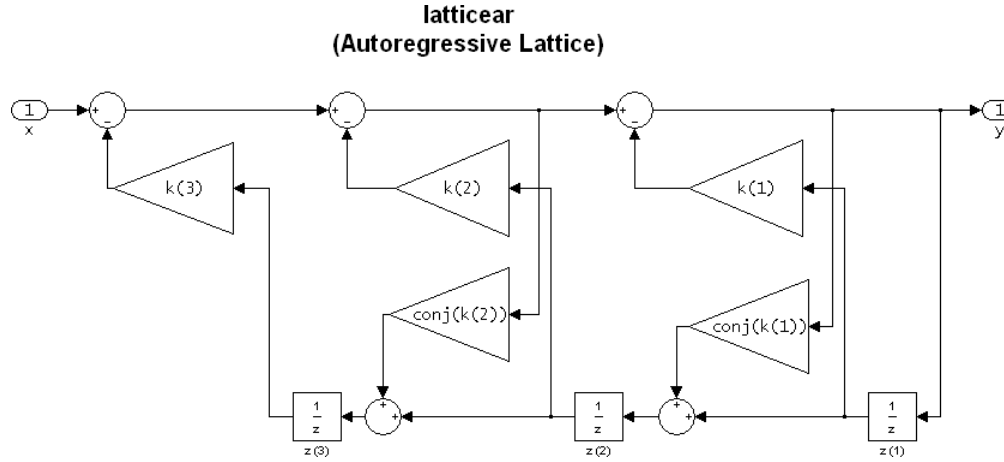
Syntax

```
Hd = dfilt.latticear(k)
Hd = dfilt.latticear
```

Description

`Hd = dfilt.latticear(k)` returns a discrete-time, lattice autoregressive filter, `Hd`, with lattice coefficients, `k`.

`Hd = dfilt.latticear` returns a default, discrete-time, lattice autoregressive filter, `Hd`, with `k=[ ]`. This filter passes the input through to the output unchanged.



The resulting filter states column vector is

$$\begin{bmatrix} z(1) \\ z(2) \\ z(3) \end{bmatrix}$$

Examples

Form a third-order lattice autoregressive filter structure for a `dfilt` object, `Hd`, using the following lattice coefficients:

```
k = [.66 .7 .44];
Hd = dfilt.latticear(k)
```

See Also

dfilt | dfilt.latticeallpass | dfilt.latticearma | dfilt.latticemamax | dfilt.latticemamin

Purpose

Discrete-time, lattice, autoregressive, moving-average filter

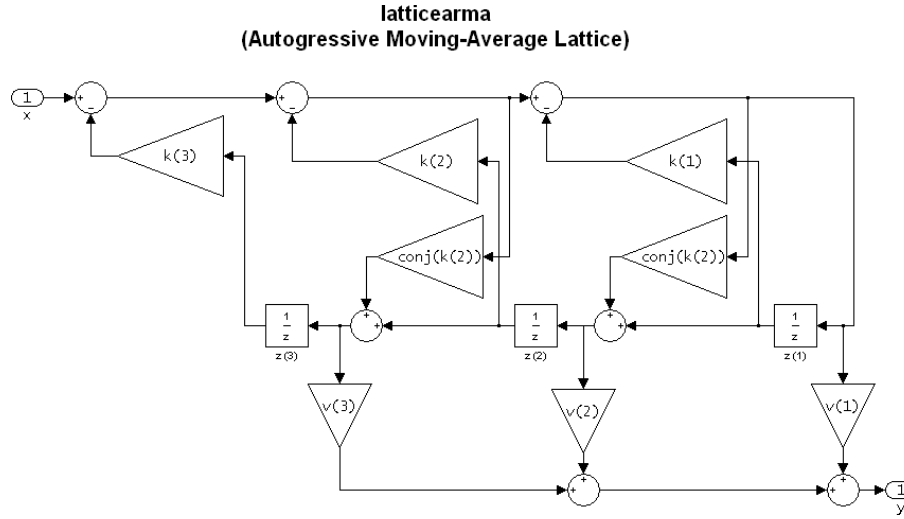
Syntax

```
Hd = dfilt.latticearma(k,v)
Hd = dfilt.latticearma
```

Description

`Hd = dfilt.latticearma(k,v)` returns a discrete-time, lattice autoregressive, moving-average filter, `Hd`, with lattice coefficients, `k` and ladder coefficients `v`.

`Hd = dfilt.latticearma` returns a default, discrete-time, lattice autoregressive, moving-average filter, `Hd`, with `k=[ ]` and `v=1`. This filter passes the input through to the output unchanged.



The resulting filter states column vector is

$$\begin{bmatrix} z(1) \\ z(2) \\ z(3) \end{bmatrix}$$

Examples

Form a third-order lattice autoregressive, moving-average filter structure for a `dfilt` object, `Hd`, using the following lattice coefficients:

```
k = [.66 .7 .44];
Hd = dfilt.latticearma(k)
```

See Also

`dfilt` | `dfilt.latticeallpass` | `dfilt.latticear` | `dfilt.latticemamax` | `dfilt.latticemamin`

Purpose

Discrete-time, lattice, moving-average filter

Syntax

```
Hd = dfilt.latticemamax(k)
Hd = dfilt.latticemamax
```

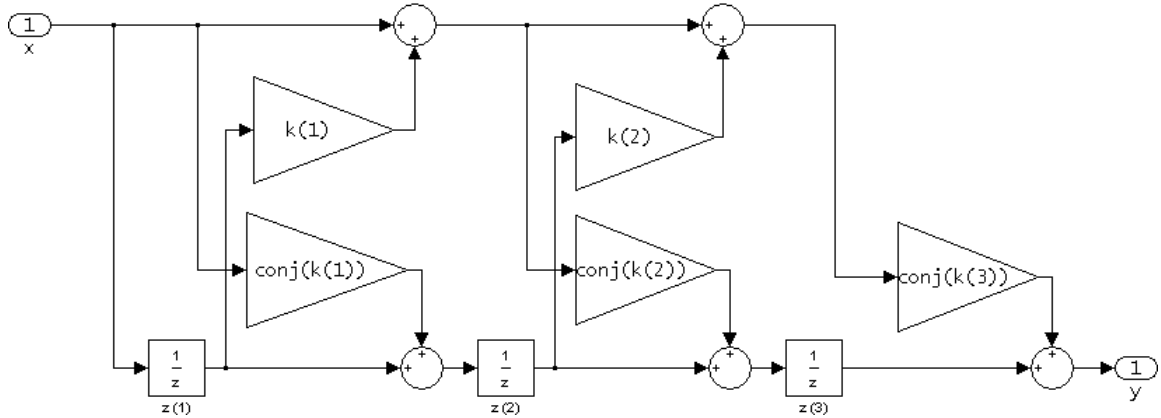
Description

`Hd = dfilt.latticemamax(k)` returns a discrete-time, lattice, moving-average filter, `Hd`, with lattice coefficients `k`.

Note If the `k` coefficients define a maximum phase filter, the resulting filter in this structure is maximum phase. If 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, `Hd`, with `k=[ ]`. This filter passes the input through to the output unchanged.

latticemamax
(Moving-Average, Maximum Phase Lattice)



The resulting filter states column vector is

$$\begin{bmatrix} z(1) \\ z(2) \\ z(3) \end{bmatrix}$$

Examples

Form a fourth-order lattice, moving-average, maximum phase filter structure for a `dfilt` object, `Hd`, using the following lattice coefficients:

```
k = [.66 .7 .44 .33];
Hd = dfilt.latticemamax(k)
```

See Also

`dfilt` | `dfilt.latticeallpass` | `dfilt.latticear` | `dfilt.latticearma` | `dfilt.latticemamin`

Purpose

Discrete-time, lattice, moving-average filter

Syntax

```
Hd = dfilt.latticemamin(k)
Hd = dfilt.latticemamin
```

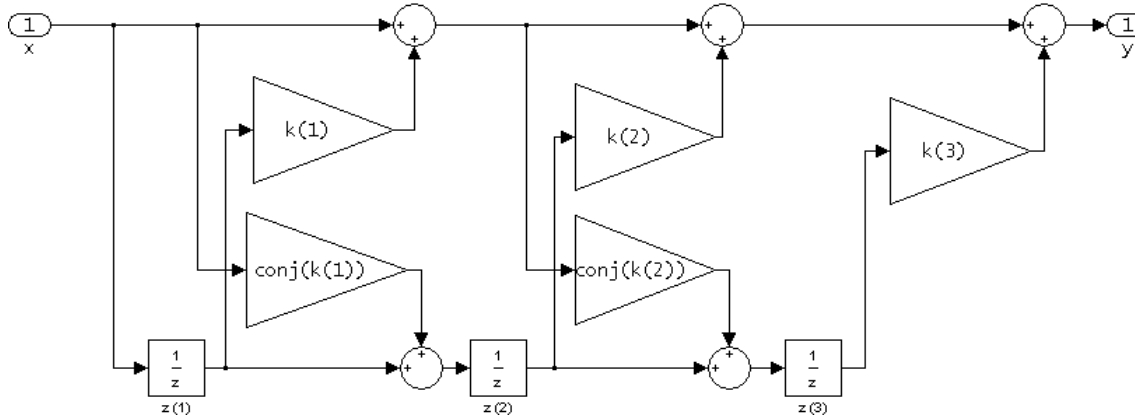
Description

`Hd = dfilt.latticemamin(k)` returns a discrete-time, lattice, moving-average, minimum phase, filter, `Hd`, with lattice coefficients `k`.

Note If the `k` coefficients define a minimum phase filter, the resulting filter in this structure is minimum phase. If 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, `Hd`, with `k=[ ]`. This filter passes the input through to the output unchanged.

latticemamin
(Moving-Average, Minimum Phase Lattice)



The resulting filter states column vector is

$$\begin{bmatrix} z(1) \\ z(2) \\ z(3) \end{bmatrix}$$

Examples

Form a third-order lattice, moving-average, minimum phase, filter structure for a `dfilt` object, `Hd`, using the following lattice coefficients.

```
k = [.66 .7 .44];
Hd = dfilt.latticemamin(k)
```

See Also

`dfilt` | `dfilt.latticeallpass` | `dfilt.latticear` | `dfilt.latticearma` | `dfilt.latticemamax`

Purpose

Discrete-time, parallel structure filter

Syntax

```
Hd = dfilt.parallel(Hd1,Hd2,...)
```

Description

`Hd = dfilt.parallel(Hd1,Hd2,...)` returns a discrete-time filter, `Hd`, which is a structure of two or more `dfilt` filters, `Hd1`, `Hd2`, etc. arranged in parallel. Each filter in a parallel structure is a separate stage. You can display states for individual stages only. To view the states of a stage use

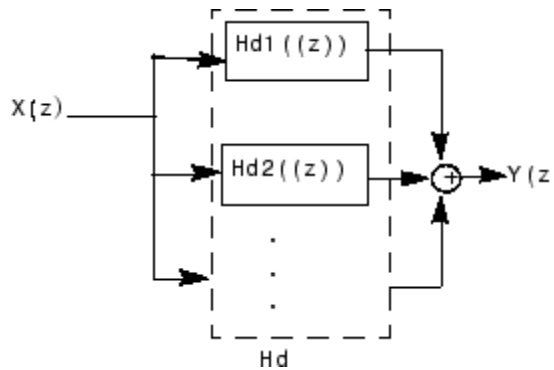
```
Hd.stage(1).states
```

To append a filter (`Hd1`) onto an existing parallel filter (`Hd`), use

```
addstage(Hd,Hd1)
```

You can also use the nondot notation format for calling a parallel structure.

```
parallel(Hd1,Hd2,...)
```



Examples

Using a parallel structure, create a coupled-allpass decomposition of a 7th order lowpass digital, elliptic filter with a normalized cutoff frequency of 0.5, 1 decibel of peak-to-peak ripple and a minimum stopband attenuation of 40 decibels.

```
k1 = [-0.0154    0.9846   -0.3048    0.5601];
Hd1 = dfilt.latticeallpass(k1);
k2 = [-0.1294    0.8341   -0.4165];
Hd2 = dfilt.latticeallpass(k2);
Hpar = parallel(Hd1 ,Hd2);
gain = dfilt.scalar(0.5);    % Normalize passband gain
Hcas = cascade(gain,Hpar);
```

For details on the stages of this filter, use

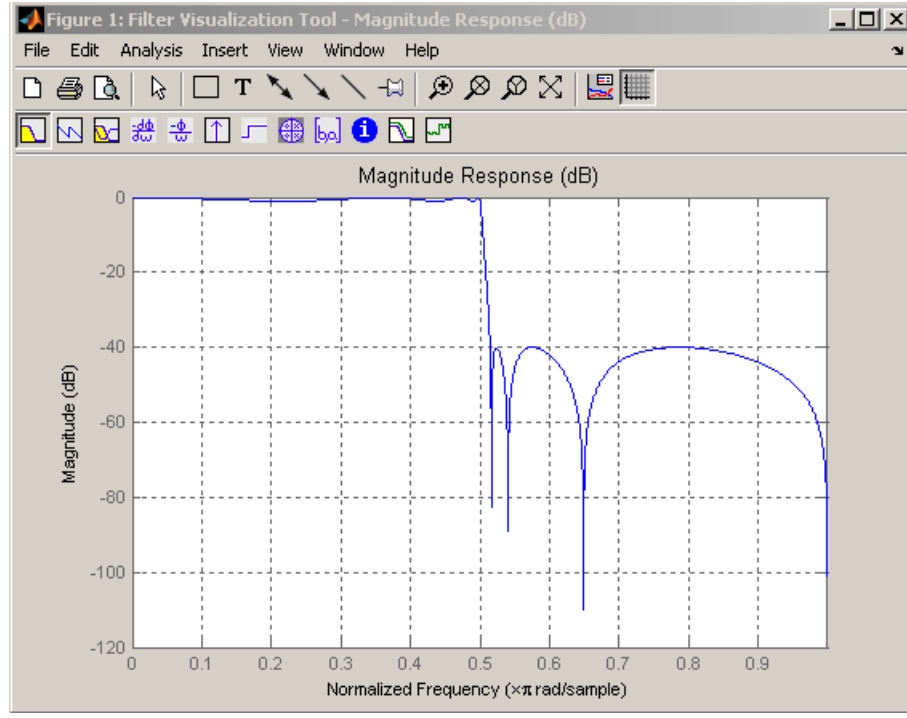
```
info(Hcas.Stage(1))
```

and

```
info(Hcas.Stage(2))
```

To view this filter, use

```
fvtool(Hcas)
```



See Also

`dfilt` | `dfilt.cascade`

dfilt.scalar

Purpose

Discrete-time, scalar filter

Syntax

```
Hd = dfilt.scalar(g)
Hd = dfilt.scalar
```

Description

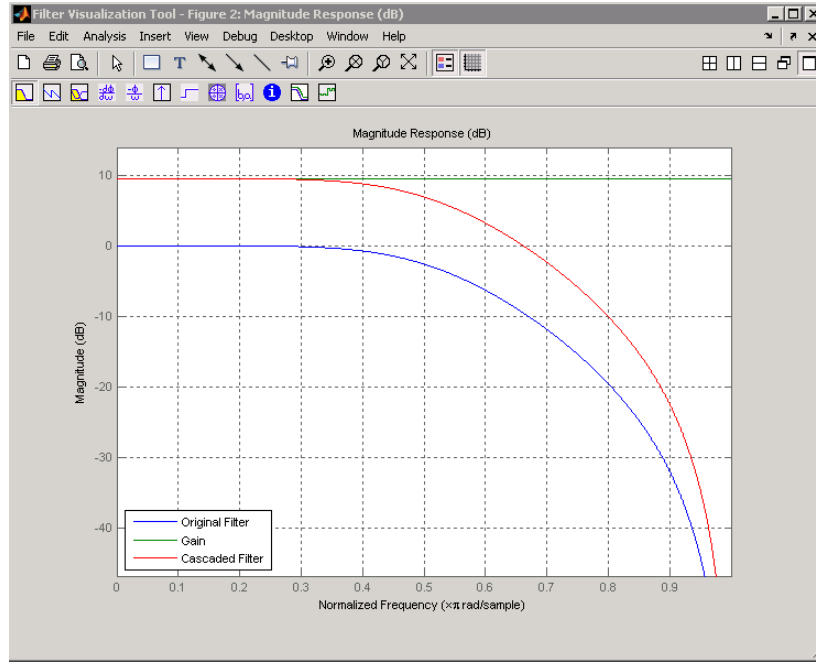
`Hd = dfilt.scalar(g)` returns a discrete-time, scalar filter, `Hd`, with gain `g`, where `g` is a scalar.

`Hd = dfilt.scalar` returns a default, discrete-time scalar gain filter, `Hd`, with gain `1`.

Examples

Create a direct-form I filter and a scalar object with a gain of 3 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 = cascade(Hd_gain,Hd_filt);
hfvt = fvtool(Hd_filt,Hd_gain,Hd_cascade);
legend(hfvt,'Original Filter','Gain','Cascaded Filter',...
'location','southwest');
```



To view the stages of the cascaded filter, use

```
Hd.stage(1)
```

and

```
Hd.stage(2)
```

See Also

`dfilt` | `dfilt.cascade`

dfilt.statespace

Purpose

Discrete-time, state-space filter

Syntax

```
Hd = dfilt.statespace(A,B,C,D)
Hd = dfilt.statespace
```

Description

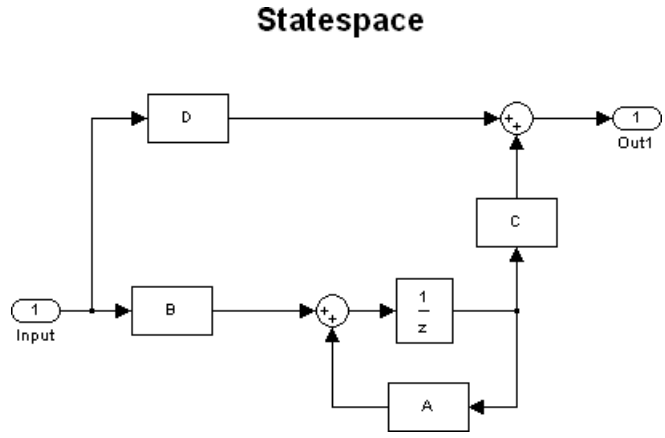
`Hd = dfilt.statespace(A,B,C,D)` returns a discrete-time state-space filter, `Hd`, with rectangular arrays `A`, `B`, `C`, and `D`.

`A`, `B`, `C`, and `D` are from the matrix or state-space form of a filter's difference equations

$$x(n+1) = Ax(n) + Bu(n)$$
$$y(n) = Cx(n) + Du(n)$$

where $x(n)$ is the vector states at time n, $u(n)$ is the input at time n, `y` is the output at time n, `A` is the state-transition matrix, `B` is the input-to-state transmission matrix, `C` is the state-to-output transmission matrix, and `D` is the input-to-ouput transmission matrix. For single-channel systems, `A` is an `m`-by-`m` matrix where `m` is the order of the filter, `B` is a column vector, `C` is a row vector, and `D` is a scalar.

`Hd = dfilt.statespace` returns a default, discrete-time state-space filter, `Hd`, with `A=[ ]`, `B=[ ]`, `C=[ ]`, and `D=1`. This filter passes the input through to the output unchanged.



The resulting filter states column vector has the same number of rows as the number of rows of `A` or `B`.

Examples

Create a second-order, state-space filter structure from a second-order, lowpass Butterworth design.

```
[A,B,C,D] = butter(2,0.5);
Hd = dfilt.statespace(A,B,C,D)
```

See Also

`dfilt`

dftmtx

Purpose

Discrete Fourier transform matrix

Syntax

```
A = dftmtx(n)
```

Description

A *discrete Fourier transform matrix* is a complex matrix of values around the unit circle, whose matrix product with a vector computes the discrete Fourier transform of the vector.

`A = dftmtx(n)` returns the `n`-by-`n` complex matrix `A` that, when multiplied into a length `n` column vector `x`.

```
y = A*x
```

computes the discrete Fourier transform of `x`.

The inverse discrete Fourier transform matrix is

```
Ai = conj(dftmtx(n))/n
```

Examples

In practice, the discrete Fourier transform is computed more efficiently and uses less memory with an FFT algorithm

```
x = 1:256;
y1 = fft(x);
```

than by using the Fourier transform matrix.

```
n = length(x);
y2 = x*dftmtx(n);
norm(y1-y2)
```

Algorithms

`dftmtx` takes the FFT of the identity matrix to generate the transform matrix.

See Also

`convmtx` | `fft`

Purpose

Permute input into digit-reversed order

Syntax

```
y = digitrevorder(x,r)
[y,i] = digitrevorder(x,r)
```

Description

`digitrevorder` is useful for pre-ordering a vector of filter coefficients for use in frequency-domain filtering algorithms, in which the `fft` and `ifft` transforms are computed without digit-reversed ordering for improved run-time efficiency.

`y = digitrevorder(x,r)` returns the input data in digit-reversed order in vector or matrix `y`. The digit-reversal is computed using the number system base (radix base) `r`, which can be any integer from 2 to 36. The length of `x` must be an integer power of `r`. If `x` is a matrix, the digit reversal occurs on the first dimension of `x` with size greater than 1. `y` is the same size as `x`.

`[y,i] = digitrevorder(x,r)` returns the digit-reversed vector or matrix `y` and the digit-reversed indices `i`, such that `y = x(i)`. Recall that MATLAB matrices use 1-based indexing, so the first index of `y` will be 1, not 0.

The following table shows the numbers 0 through 15, the corresponding digits and the digit-reversed numbers using radix base-4. The corresponding radix base-2 bits and bit-reversed indices are also shown.

Linear Index	Base-4 Digits	Digit-Reversed	Digit-Reversed Index	Base-2 Bits	Base-2 Reversed (bitrevorder)	Bit-Reversed Index
0	00	00	0	0000	0000	0
1	01	10	4	0001	1000	8
2	02	20	8	0010	0100	4
3	03	30	12	0011	1100	12
4	10	01	1	0100	0010	2
5	11	11	5	0101	1010	10

Linear Index	Base-4 Digits	Digit-Reversed	Digit-Reversed Index	Base-2 Bits	Base-2 Reversed (bitrevorder)	Bit-Reversed Index
6	12	21	9	0110	0110	6
7	13	31	13	0111	1110	14
8	20	02	2	1000	0001	1
9	21	12	6	1001	1001	9
10	22	22	10	1010	0101	5
11	23	32	14	1011	1101	13
12	30	03	3	1100	0011	3
13	31	13	7	1101	1011	11
14	32	23	11	1110	0111	7
15	33	33	15	1111	1111	15

Examples

Obtain the digit-reversed, radix base-3 ordered output of a vector containing 9 values:

```
x=[0:8]';                    % Create a column vector
[x,digitrevorder(x,3)]
% ans =
%
%     0     0
%     1     3
%     2     6
%     3     1
%     4     4
%     5     7
%     6     2
%     7     5
%     8     8
```

See Also

`bitrevorder` | `fft` | `ifft`

Purpose

Dirichlet or periodic sinc function

Syntax

```
y = diric(x,n)
```

Description

`y = diric(x,n)` returns a vector or array `y` the same size as `x`. The elements of `y` are the Dirichlet function of the elements of `x`. `n` must be a positive integer.

The Dirichlet function, or periodic sinc function, is

$$D(x) = \begin{cases} \dfrac{\sin(Nx/2)}{N\sin(x/2)} & x \neq 2\pi k, \quad k = 0, \pm 1, \pm 2, \pm 3, \ldots \\ (-1)^{k(N-1)} & x = 2\pi k, \quad k = 0, \pm 1, \pm 2, \pm 3, \ldots \end{cases}$$

for any nonzero integer `n`. This function has period 2π for `n` odd and period 4π for `n` even. Its peak value is 1, and its minimum value is -1 for `n` even. The magnitude of this function is `(1/n)` times the magnitude of the discrete-time Fourier transform of the `n`-point rectangular window.

Diagnostics

If `n` is not a positive integer, `diric` gives the following error message:

```
Requires n to be a positive integer.
```

See Also

`cos` | `gauspuls` | `pulstran` | `rectpuls` | `sawtooth` | `sin` | `sinc` | `square` | `tripuls`

downsample

Purpose

Decrease sampling rate by integer factor

Syntax

```
y = downsample(x,n)
y = downsample(x,n,phase)
```

Description

`y = downsample(x,n)` decreases the sampling rate of `x` by keeping every n-th sample starting with the first sample. `x` can be a vector or a matrix. If `x` is a matrix, each column is considered a separate sequence.

`y = downsample(x,n,phase)` specifies the number of samples by which to offset the downsampled sequence. `phase` must be an integer from 0 to `n`-1.

Examples

Decrease the sampling rate of a sequence by 3:

```
x = [1 2 3 4 5 6 7 8 9 10];
y = downsample(x,3)
% y =  1    4    7    10
```

Decrease the sampling rate of the sequence by 3 and add a phase offset of 2:

```
y = downsample(x,3,2)
% y = 3    6    9
```

Decrease the sampling rate of a matrix by 3:

```
x = [1 2 3; 4 5 6; 7 8 9; 10 11 12];
y = downsample(x,3);
```

See Also

`decimate` | `interp` | `interp1` | `resample` | `spline` | `upfirdn` | `upsample`

Purpose

Discrete prolate spheroidal (Slepian) sequences

Syntax

```
dps_seq = dpss(seq_length,time_halfbandwidth)
[dps_seq,lambda] = dpss(seq_length,time_halfbandwidth)
[...] = dpss(seq_length,time_halfbandwidth,num_seq)
[...] = dpss(seq_length,time_halfbandwidth,'interp_method')
[...] = dpss(...,Ni)
[...] = dpss(...,'trace')
```

Description

`dps_seq = dpss(seq_length,time_halfbandwidth)` returns the first `round(2*time_halfbandwidth)` discrete prolate spheroidal (DPSS), or Slepian sequences of length `seq_length`. `dps_seq` is a matrix with `seq_length` rows and `round(2*time_halfbandwidth)` columns. `time_halfbandwidth` must be strictly less than `seq_length/2`.

`[dps_seq,lambda] = dpss(seq_length,time_halfbandwidth)` returns the frequency-domain energy concentration ratios of the column vectors in `dps_seq`. The ratios represent the amount of energy in the passband *[-W,W]* to the total energy from *[-Fs/2, Fs/2]* where *Fs* is the sampling frequency. `lambda` is a column vector equal in length to the number of Slepian sequences.

`[...] = dpss(seq_length,time_halfbandwidth,num_seq)` returns the first `num_seq` Slepian sequences with time half bandwidth product `time_halfbandwidth` ordered by their energy concentration ratios. If `num_seq` is a two-element vector, the returned Slepian sequences range from `num_seq(1)` to `num_seq(2)`.

`[...] = dpss(seq_length,time_halfbandwidth,'interp_method')` uses interpolation to compute the DPSSs from a user-created database of DPSSs. Create the database of DPSSs with `dpsssave` and ensure that the resulting dpss.mat file is in the MATLAB search path. Valid options for `'interp_method'` are `'spline'` and `'linear'`. The interpolation method uses the Slepian sequences in the database with time half bandwidth product `time_halfbandwidth` and length closest to `seq_length`.

`[...] = dpss(...,Ni)` interpolates from DPSSs of length `Ni` in the database dpss.mat.

`[...] = dpss(...,'trace')` prints the method used to compute the DPSSs in the command window. Possible methods include: direct, spline interpolation, and linear interpolation.

Definitions

Discrete Prolate Spheroidal Sequences

The discrete prolate spheroidal or Slepian sequences derive from the following time-frequency concentration problem. For all finite-energy sequences $x[n]$ index limited to some set $[N_1, N_1 + N_2]$, which sequence maximizes the following ratio:

$$\lambda = \frac{\int_{-W}^{W} |X(f)|^2 df}{\int_{-Fs/2}^{Fs/2} |X(f)|^2 \, df}$$

where *Fs* is the sampling frequency and $|W| < Fs/2$. Accordingly, this ratio determines which index-limited sequence has the largest proportion of its energy in the band *[-W,W]*. For index-limited sequences, the ratio must satisfy the inequality $0 < \lambda < 1$. The sequence maximizing the ratio is the first discrete prolate spheroidal or Slepian sequence. The second Slepian sequence maximizes the ratio and is orthogonal to the first Slepian sequence. The third Slepian sequence maximizes the ratio of integrals and is orthogonal to both the first and second Slepian sequences. Continuing in this way, the Slepian sequences form an orthogonal set of band limited sequences.

Time Half Bandwidth Product

The time half bandwidth product is *NW* where *N* is the length of the sequence and *[-W,W]* is the effective bandwidth of the sequence. In constructing Slepian sequences, you choose the desired sequence length and bandwidth *2W*. Both the sequence length and bandwidth affect how many Slepian sequences have concentration ratios near one. As a rule,

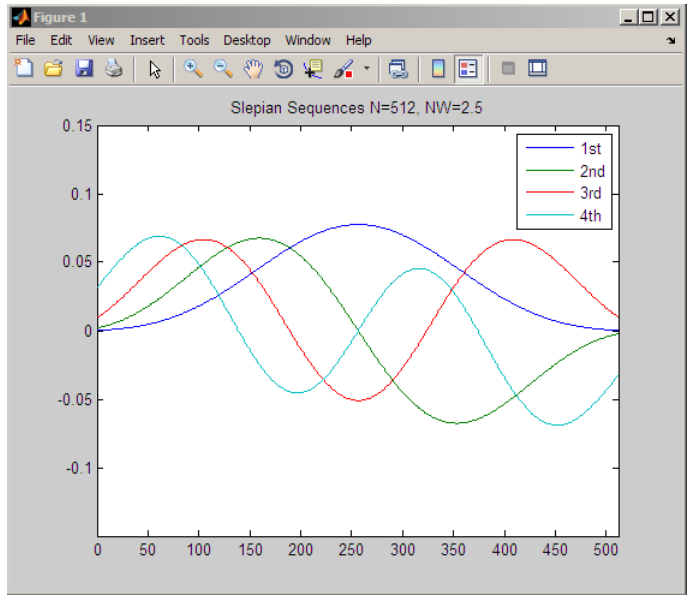
there are *2NW-1* Slepian sequences with energy concentration ratios approximately equal to one. Beyond *2NW-1* Slepian sequences, the concentration ratios begin to approach zero. Common choices for the time half bandwidth product are: 2.5, 3, 3.5, and 4.

You can specify the bandwidth of the Slepian sequences in Hz by defining the time half bandwidth product as *NW/Fs* where *Fs* is the sampling frequency.

Examples

Construct a set of Slepian sequences:

```
seq_length = 512;
time_halfbandwidth = 2.5;
num_seq = 2*(2.5)-1;
%Obtain DPSSs
[dps_seq,lambda] = dpss(seq_length,time_halfbandwidth,num_seq);
% Plot the Slepian sequences
plot(dps_seq);
title('Slepian Sequences N=512, NW=2.5');
axis([0 512 -0.15 0.15]);
legend('1st','2nd','3rd','4th');
%Concentration ratios in lambda:
%1.0000    0.9998    0.9962    0.9521
```



References

Percival, D.B., and A.T. Walden. *Spectral Analysis for Physical Applications*. Cambridge: Cambridge University Press, 1993.

See Also

`dpssclear` | `dpssload` | `dpsssave` | `spectrum.mtm`

How To

- "Nonparametric Methods"

Purpose

Remove discrete prolate spheroidal sequences from database

Syntax

```
dpssclear(n,nw)
```

Description

`dpssclear(n,nw)` removes sequences with length `n` and time-bandwidth product `nw` from the DPSS MAT-file database `dpss.mat`.

See Also

`dpss` | `dpssdir` | `dpssload` | `dpsssave`

dpssdir

Purpose

Discrete prolate spheroidal sequences database directory

Syntax

```
dpssdir
dpssdir(n)
dpssdir(nw,'nw')
dpssdir(n,nw)
index = dpssdir
```

Description

`dpssdir` manages the database directory that contains the generated DPSS samples in the DPSS MAT-file database `dpss.mat`. Create the DPSS MAT-file database with `dpsssave`.

`dpssdir` lists the directory of saved sequences in `dpss.mat`.

`dpssdir(n)` lists the sequences saved with length `n`.

`dpssdir(nw,'nw')` lists the sequences saved with time-bandwidth product `nw`.

`dpssdir(n,nw)` lists the sequences saved with length `n` and time-bandwidth product `nw`.

`index = dpssdir` is a structure array describing the DPSS database. Pass `n` and `nw` options as for the no output case to get a filtered `index`.

See Also

`dpss` | `dpssclear` | `dpssload` | `dpsssave`

Purpose

Load discrete prolate spheroidal sequences from database

Syntax

```
[e,v] = dpssload(n,nw)
```

Description

`[e,v] = dpssload(n,nw)` loads all sequences with length `n` and time-bandwidth product `nw` in the columns of `e` and their corresponding concentrations in vector `v` from the DPSS MAT-file database `dpss.mat`. Create the `dpss.mat` file using `dpssave`.

See Also

`dpss` | `dpssclear` | `dpssdir` | `dpsssave`

dpsssave

Purpose

Discrete prolate spheroidal or Slepian sequence database

Syntax

```
dpsssave(time_halfbandwith,dps_seq,lambda)
status = dpsssave(time_halfbandwith,dps_seq,lambda)
```

Description

`dpsssave(time_halfbandwith,dps_seq,lambda)` creates a database of discrete prolate spheroidal (DPSS) or Slepian sequences and saves the results in `dpss.mat`. The time half bandwidth product`time_halfbandwith` is a real-valued scalar determining the frequency concentration of the Slepian sequences in `dps_seq`. `dps_seq` is a *NxK* matrix of Slepian sequences where *N* is the length of the sequences. `lambda` is a *1xK* vector containing the frequency concentration ratios of the Slepian sequences in `dps_seq`.

If the database `dpss.mat` exists, subsequent calls to `dpsssave` append the Slepian sequences to the existing file.

`status = dpsssave(time_halfbandwith,dps_seq,lambda)` returns a 0 if the database operation was successful or a 1 if unsuccessful.

Definitions

Discrete Prolate Spheroidal Sequences

The discrete prolate spheroidal or Slepian sequences derive from the following time-frequency concentration problem. For all finite-energy sequences $x[n]$ index limited to some set $[N_1, N_1 + N_2]$, which sequence maximizes the following ratio:

$$\lambda = \frac{\int_{-W}^{W} |X(f)|^2 df}{\int_{-Fs/2}^{Fs/2} |X(f)|^2 \, df}$$

where *Fs* is the sampling frequency $|W| < Fs/2$. In other words, which index-limited sequence has the largest proportion of its energy in the band *[-W,W]*. For index-limited sequences, the ratio must satisfy the inequality $0 < \lambda < 1$. The sequence maximizing the ratio is the first

discrete prolate spheroidal or Slepian sequence. The second Slepian sequence maximizes the ratio and is orthogonal to the first Slepian sequence. The third Slepian sequence maximizes the ratio of integrals and is orthogonal to both the first and second Slepian sequences. Continuing in this way, the Slepian sequences form an orthogonal set of band limited sequences.

Time Half Bandwidth Product

The time half bandwidth product is *NW* where *N* is the length of the sequence and *[-W,W]* is the effective bandwidth of the sequence. In constructing Slepian sequences, you choose the desired sequence length and bandwidth *2W*. Both the sequence length and bandwidth affect how many Slepian sequences have concentration ratios near one. As a rule, there are *2NW-1* Slepian sequences with energy concentration ratios approximately equal to one. Beyond *2NW-1* Slepian sequences, the concentration ratios begin to approach zero. Common choices for the time half bandwidth product are: 2.5, 3, 3.5, and 4.

You can specify the bandwidth of the Slepian sequences in Hz by defining the time half bandwidth product as *NW/Fs* where *Fs* is the sampling frequency.

Examples

Create Slepian sequence database in current directory:

```
seq_length=512;
time_halfbandwidth=2.5;
num_seq=4;
[dps_seq,lambda]=dpss(seq_length,time_halfbandwidth);
% Create databased dpss.mat in current working directory
status=dpsssave(time_halfbandwidth,dps_seq,lambda);
% status should equal 1
```

References

Percival, D.B., and A.T. Walden. *Spectral Analysis for Physical Applications*. Cambridge: Cambridge University Press, 1993.

See Also

`dpss` | `dpssclear` | `dpssdir` | `dpssload`

dspdata

Purpose

DSP data parameter information

Syntax

```
Hs = dspdata.dataobj(input1,...)
```

Description

`Hs = dspdata.dataobj(input1,...)` returns a `dspdata` object `Hs` of type *`dataobj`*. This object contains all the parameter information needed for the specified type of *`dataobj`*. Each *`dataobj`* takes one or more inputs, which are described on the individual reference pages. If you do not specify any input values, the returned object has default property values appropriate for the particular *`dataobj`* type.

Note You must use a *`dataobj`* with `dspdata`.

Data Objects

A data object (*`dataobj`*) for `dspdata` specifies the type of data stored in the object. Available *`dataobj`* types for `dspdata` are shown below.

dspdata.dataobj	Description
`dspdata.msspectrum`	Mean-square spectrum data (power)
`dspdata.psd`	Power spectral density data (power/frequency)
`dspdata.pseudospectrum`	Pseudospectrum data (power)

For more information on each *`dataobj`* type, use the syntax `help dspdata.dataobj` at the MATLAB prompt or refer to its reference page.

Methods

Methods provide ways of performing functions directly on your `dspdata` object. You can apply these methods directly on the variable you assigned to your `dspdata` object.

Method	Description
`avgpower`	Note that this method applies only to `dspdata.psd` objects. `avgpower(Hs)` computes the average power in a given frequency band. The technique uses a rectangle approximation of the integral of the `Hs` signal's power spectral density (PSD). If the signal is a matrix, the computation is done on each column. The average power is the total signal power and the `SpectrumType` property determines whether the total average power is contained in the one-sided or two-sided spectrum. For aa one-sided spectrum, the range is [0,pi] for even number of frequency points and [0,pi) for odd. For a two-sided spectrum the range is [0,2pi). `avgpower(Hs,freqrange)` specifies the frequency range over which to calculate the average power. `freqrange` is a two-element vector of the frequencies between which to calculate. If a frequency value does not match exactly the frequency in `Hs`, the next closest value is used. Note that the first frequency value in `freqrange` is included in the calculation and the second value is excluded.
`centerdc`	`centerdc(Hs)` or `centerdc(Hs,true)` shifts the data and frequency values so that the DC component is at the center of the spectrum. If the `SpectrumType` property is `'onesided'`, it is changed to `'twosided'` and then the DC component is centered. `centerdc(Hs,'false')` shifts the data and frequency values so that the DC component is at the left edge of the spectrum.

Method	Description
`findpeaks`	`findpeaks(Hs)` finds local maxima or peaks. If no peaks are found, `findpeaks` returns an empty vector. `[pks,frqs] = findpeaks(x)` returns peaks values (`pks`) and the frequencies (`frqs`) at which the peaks occur. `findpeaks(x,'minpeakheight',mph)` returns only peaks greater than the minimum peak height `mph`, where `mph` is a real scalar. Default is `-Inf`. `findpeaks(x,'minpeakdistance',mpd)` returns only peaks separated by the minimum frequency units distance `mpd`, which is a positive integer. Setting the minimum peak distance ignores smaller peaks that may occur close to larger local peaks. Default is 1. `findpeaks(x,'threshold',th)` returns only peaks greater than their neighbors by at least the threshold `th`, which is a real, scalar value greater than or equal to 0. Default is 0. `findpeaks(x,'npeaks',np)` returns a maximum of `np` number of peaks. When `np` peaks are found, the search stops. Default is to return all peaks. `findpeaks(x,'sortstr',str)` specifies the sorting order, where `str` is `'ascend'`, `'descend'` or `'none'`. For `'ascend'` , the peaks are returned in order from smallest to largest, and vice versa for `'descend'`. For `'none'`, the peaks are returned in the order in which they occur.

Method	Description
`halfrange`	`halfrange(Hs)` converts the `Hs` spectrum to a spectrum calculated over half the Nyquist interval. All associated properties affected by the new frequency range are adjusted automatically. This method is used for `dspdata.pseudospectrum` objects. Note that the spectrum is assumed to be from a real signal (that is, `halfrange` uses half the data points regardless of whether the data is symmetric).
`normalizefreq`	`normalizefreq(Hs)` or `normalizefreq(Hs,true)` normalizes the frequency specifications in the `Hs` object to `Fs` so the frequencies are between 0 and 1. It also sets the `NormalizedFrequency` property to `true`. `normalizefreq(Hs,false)` converts the frequencies to linear frequencies. `normalizefreq(Hs,false,Fs)` sets a new sampling frequency `Fs`. This can be used only with `false`.
`onesided`	`onesided(Hs)` converts the `Hs` spectrum to a spectrum calculated over half the Nyquist interval and containing the total signal power. All associated properties affected by the new frequency range are adjusted automatically. This method is used for `dspdata.psd` and `dspdata.msspectrum` objects. Note that the spectrum is assumed to be from a real signal (that is, `onesided` uses half the data points regardless of whether the data is symmetric).

Method	Description
`plot`	Displays the data graphically in the current figure window. For a `dspdata.psd` object, it displays the power spectral density in dB/Hz. For a `dspdata.msspectrum` object, it displays the mean–square in dB. For a `dspdata.pseudospectrum` object, it displays the pseudospectrum in dB.
`sfdr`	This method applies only to `dspdata.msspectrum` objects. `sfdr(Hs)` computes the spurious-free dynamic range (SFDR) in dB of a mean square spectrum object `Hs`. SFDR is the usable range before spurious noise interferes with the signal. `[sfd,spur,frq] = sfdr(Hs)` returns the magnitude of the highest `spur` and the frequency `frq` at which it occurs. `sfdr(Hs,'minspurlevel',msl)` ignores spurs below the minimum spur level `msl`, which is a real scalar in dB. `sfdr(Hs,'minspurdistance',msd)` includes spurs only if they are separated by at least the minimum spur distance `msd`, which is a real, positive scalar in frequency units.

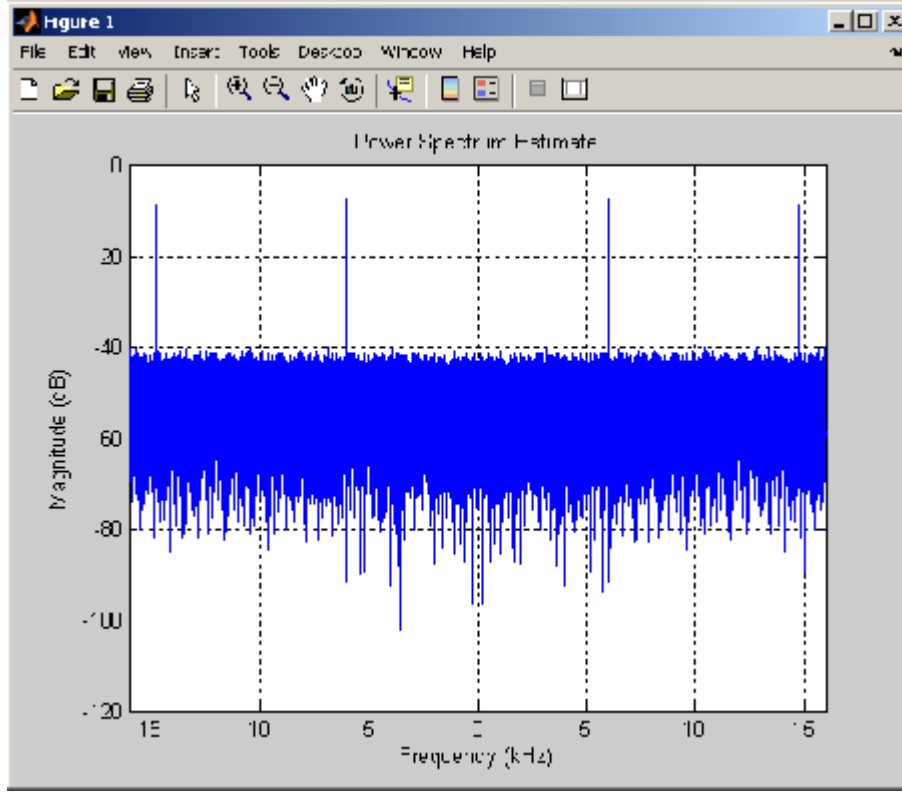
Method	Description
`twosided`	`twosided(Hs)` converts the `Hs` spectrum to a spectrum calculated over the whole Nyquist interval. All associated properties affected by the new frequency range are adjusted automatically. This method is used for `dspdata.psd` and `dspdata.msspectrum` objects. Note that if your data is nonuniformly sampled, converting from `onesided` to `twosided` may produce incorrect results.
`wholerange`	`wholerange(Hs)` converts the `Hs` spectrum to a spectrum calculated over the whole Nyquist interval. All associated properties affected by the new frequency range are adjusted automatically. This method is used for `dspdata.pseudospectrum` objects. Note that if your data is nonuniformly sampled, converting from `half` to `wholerange` may produce incorrect results.

For more information on each method, use the syntax `help dspdata/`*`method`* at the MATLAB prompt.

Plotting a dspdata Object

The plot method displays the `dspdata` object spectrum in a separate figure window.

```
plot(Hs)              % Plots an existing Hs object
```



Modifying a dspdata Object

After you create a `dspdata` object, you can use any of the methods in the table above to modify the object properties.

For example, to change the object from two-sided to one-sided, use

```
onesided(Hs)
```

The `Hs` object is modified.

Examples

See the `msspectrum`, `psd`, or `pseudospectrum` reference pages for specific examples.

See Also

`dspdata.msspectrum` | `dspdata.psd` | `dspdata.pseudospectrum`

dspdata.msspectrum

Purpose

Mean-square (power) spectrum

Syntax

```
Hmss = dspdata.msspectrum(Data)
Hmss = dspdata.msspectrum(Data,Frequencies)
Hmss = dspdata.msspectrum(...,'Fs',Fs)
Hmss = dspdata.msspectrum(...,'SpectrumType',SpectrumType)
Hmss = dspdata.msspectrum(...,'CenterDC',flag)
```

Description

The mean-squared spectrum (MSS) is intended for discrete spectra. Unlike the power spectral density (PSD), the peaks in the MSS reflect the power in the signal at a given frequency. The MSS of a signal is the Fourier transform of that signal's autocorrelation.

`Hmss = dspdata.msspectrum(Data)` uses the mean-square (power) spectrum data contained in `Data`, which can be in the form of a vector or a matrix, where each column is a separate set of data. Default values for other properties of the object are as follows:

Property	Default Value	Description
Name	`'Mean-square Spectrum'`	Read-only string
`Frequencies`	`[]` type `double`	Vector of frequencies at which the spectrum is evaluated. The range of this vector depends on the `SpectrumType` value. For a one-sided spectrum, the default range is [0, pi) or [0, `Fs`/2) for odd length, and [0, pi] or [0, `Fs`/2] for even length, if `Fs` is specified. For a two-sided spectrum, it is [0, 2pi) or [0, `Fs`). The length of the `Frequencies` vector must match the length of the columns of `Data`. If you do not specify `Frequencies`, a default vector is created. If one-sided is selected, then the whole number of FFT points (nFFT) for this vector is assumed to be even. If `onesided` is selected and you specify `Frequencies`, the last frequency point is compared to the next-to-last point and to pi (or `Fs`/2, if `Fs` is specified). If the last point is closer to pi (or `Fs`/2) than it is to the previous point, nFFT is assumed to be even. If it is closer to the previous point, nFFT is assumed to be odd.
`Fs`	`'Normalized'`	Sampling frequency, which is `'Normalized'` if `NormalizedFrequency` is `true`. If `NormalizedFrequency` is `false` `Fs` defaults to `1` Hz.

Property	Default Value	Description
SpectrumType	'Onesided'	Nyquist interval over which the spectral density is calculated. Valid values are 'Onesided' and 'Twosided'. See the onesided and twosided methods in dspdata for information on changing this property. The interval for Onesided is [0 pi) or [0 pi] depending on the number of FFT points, and for Twosided the interval is [0 2pi).
NormalizedFrequency	true	Whether the frequency is normalized (true) or not (false). This property is set automatically at construction time based on Fs. If Fs is specified, NormalizedFrequency is set to false. See the normalizefreq method in dspdata for information on changing this property.

`Hmss = dspdata.msspectrum(Data,Frequencies)` uses the mean–square spectrum data contained in `Data` and `Frequencies` vectors.

`Hmss = dspdata.msspectrum(...,'Fs',Fs)` uses the sampling frequency `Fs`. Specifying `Fs` uses a default set of linear frequencies (in `Hz`) based on `Fs` and sets `NormalizedFrequency` to `false`.

`Hmss = dspdata.msspectrum(...,'SpectrumType',SpectrumType)` uses the `SpectrumType` string to specify the interval over which the mean–square spectrum was calculated. For data that ranges from [0 pi) or [0 pi], set the `SpectrumType` to `onesided`; for data that ranges from [0 2pi), set the the `SpectrumType` to `twosided`.

`Hmss = dspdata.msspectrum(...,'CenterDC',flag)` uses the value of `flag` to indicate whether the zero-frequency (DC) component is centered. If `flag` is `true`, it indicates that the DC component is in

the center of the two-sided spectrum. Set the `flag` to `false` if the DC component is on the left edge of the spectrum.

Methods

Methods provide ways of performing functions directly on your `dspdata` object without having to specify the parameters again. You can apply a method directly on the variable you assigned to your `dspdata.msspectrum` object. You can use the following methods with a `dspdata.msspectrum` object.

- `centerdc`
- `normalizefreq`
- `onesided`
- `plot`
- `sfdr`
- `twosided`

For example, to normalize the frequency and set the `NormalizedFrequency` parameter to true, use

```
Hmss = normalizefreq(Hs)
```

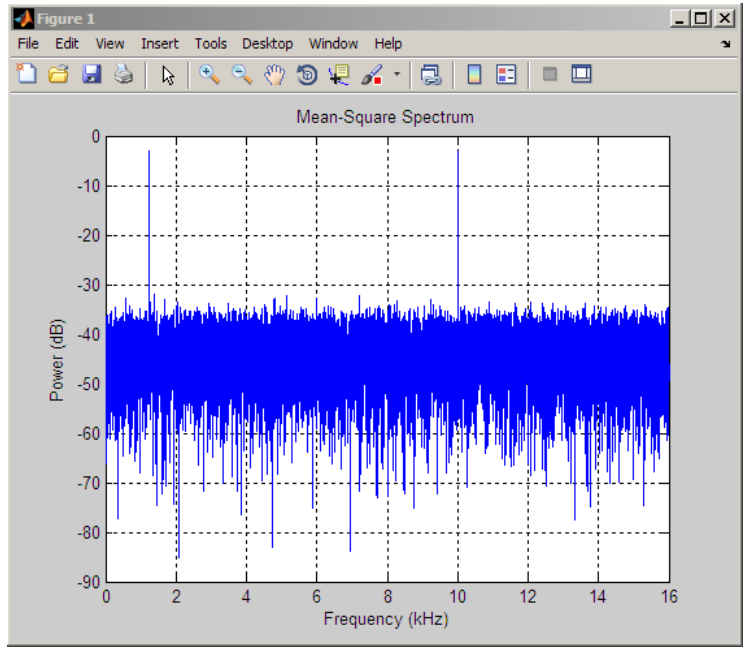
For detailed information on using the methods and plotting the spectrum, see the `dspdata` reference page.

Examples

In this example, we construct a mean-square spectrum data object from the one-sided PSD estimate of a signal. The signal consists of two sinusoids in additive noise.

```
Fs = 32e3;
t = 0:1/Fs:1-(1/Fs);
x = cos(2*pi*t*1.24e3)+cos(2*pi*t*10e3)+randn(size(t));
X = fft(x);
X=X(1:length(X)/2+1); %one-sided DFT
P = (abs(X)/length(x)).^2;     % Compute the mean-square power
P(2:end-1)=2*P(2:end-1); % Factor of two for one-sided estimate
```

```
% at all frequencies except zero and the Nyquist
Hmss=dspdata.msspectrum(P,'Fs',Fs,'spectrumtype','onesided');
plot(Hmss);          % Plot the mean-square spectrum.
```



See Also

dspdata.psd | dspdata.pseudospectrum | spectrum

Purpose

Power spectral density

Syntax

```
Hpsd = dspdata.psd(Data)
Hpsd = dspdata.psd(Data,Frequencies)
Hpsd = dspdata.psd(...,'Fs',Fs)
Hpsd = dspdata.psd(...,'SpectrumType',SpectrumType)
Hpsd = dspdata.psd(...,'CenterDC',flag)
```

Description

The power spectral density (PSD) is intended for continuous spectra. The integral of the PSD over a given frequency band computes the average power in the signal over that frequency band. In contrast to the mean-squared spectrum, the peaks in this spectra do not reflect the power at a given frequency. See the `avgpower` method of `dspdata` for more information.

A one-sided PSD contains the total power of the signal in the frequency interval from DC to half of the Nyquist rate. A two-sided PSD contains the total power in the frequency interval from DC to the Nyquist rate.

`Hpsd = dspdata.psd(Data)` uses the power spectral density data contained in `Data`, which can be in the form of a vector or a matrix, where each column is a separate set of data. Default values for other properties of the object are shown below:

Property	Default Value	Description
Name	`'Power Spectral Density'`	Read-only string
`Frequencies`	`[]` type `double`	Vector of frequencies at which the power spectral density is evaluated. The range of this vector depends on the `SpectrumType` value. For one-sided, the default range is [0, pi) or [0, `Fs`/2) for odd length, and [0, pi] or [0, `Fs`/2] for even length, if `Fs` is specified. For two-sided, it is [0, 2pi) or [0, `Fs`).

Property	Default Value	Description
		If you do not specify `Frequencies`, a default vector is created. If one-sided is selected, then the whole number of FFT points (nFFT) for this vector is assumed to be even. If `onesided` is selected and you specify `Frequencies`, the last frequency point is compared to the next-to-last point and to pi (or `Fs`/2, if `Fs` is specified). If the last point is closer to pi (or `Fs`/2) than it is to the previous point, nFFT is assumed to be even. If it is closer to the previous point, nFFT is assumed to be odd. The length of the `Frequencies` vector must match the length of the columns of `Data`.
`Fs`	`'Normalized'`	Sampling frequency, which is `'Normalized'` if `NormalizedFrequency` is `true`. If `NormalizedFrequency` is `false` `Fs` defaults to `1`.

Property	Default Value	Description
`SpectrumType`	`'Onesided'`	Nyquist interval over which the power spectral density is calculated. Valid values are `'Onesided'` and `'Twosided'`. A one-sided PSD contains the total signal power in half the Nyquist interval. See the `onesided` and `twosided` methods in `dspdata` for information on changing this property. The range for half the Nyquist interval is [0 pi) or [0 pi] depending on the number of FFT points. For the whole Nyquist interval, the range is [0 2pi).
`NormalizedFrequency`	`true`	Whether the frequency is normalized (`true`) or not (`false`). This property is set automatically at construction time based on `Fs`. If `Fs` is specified, `NormalizedFrequency` is set to `false`. See the `normalizefreq` method in `dspdata` for information on changing this property.

`Hpsd = dspdata.psd(Data,Frequencies)` uses the power spectral density estimation data contained in `Data` and `Frequencies` vectors.

`Hpsd = dspdata.psd(...,'Fs',Fs)` uses the sampling frequency `Fs`. Specifying `Fs` uses a default set of linear frequencies (in `Hz`) based on `Fs` and sets `NormalizedFrequency` to `false`.

`Hpsd = dspdata.psd(...,'SpectrumType',SpectrumType)` uses the `SpectrumType` string to specify the interval over which the power spectral density was calculated. For data that ranges from [0 pi) or [0 pi], set the `SpectrumType` to `onesided`; for data that ranges from [0 2pi), set the `SpectrumType` to `twosided`.

`Hpsd = dspdata.psd(...,'CenterDC',flag)` uses the value of `flag` to indicate whether the zero-frequency (DC) component is centered. If

`flag` is `true`, it indicates that the DC component is in the center of the two-sided spectrum. Set the `flag` to `false` if the DC component is on the left edge of the spectrum.

Methods

Methods provide ways of performing functions directly on your `dspdata` object. You can apply a method directly on the variable you assigned to your `dspdata.psd` object. You can use the following methods with a `dspdata.psd` object.

- `avgpower`
- `centerdc`
- `normalizefreq`
- `onesided`
- `plot`
- `twosided`

For example, to normalize the frequency and set the `NormalizedFrequency` parameter to true, use

```
Hpsd = normalizefreq(Hpsd)
```

For detailed information on using the methods and plotting the spectrum, see the `dspdata` reference page.

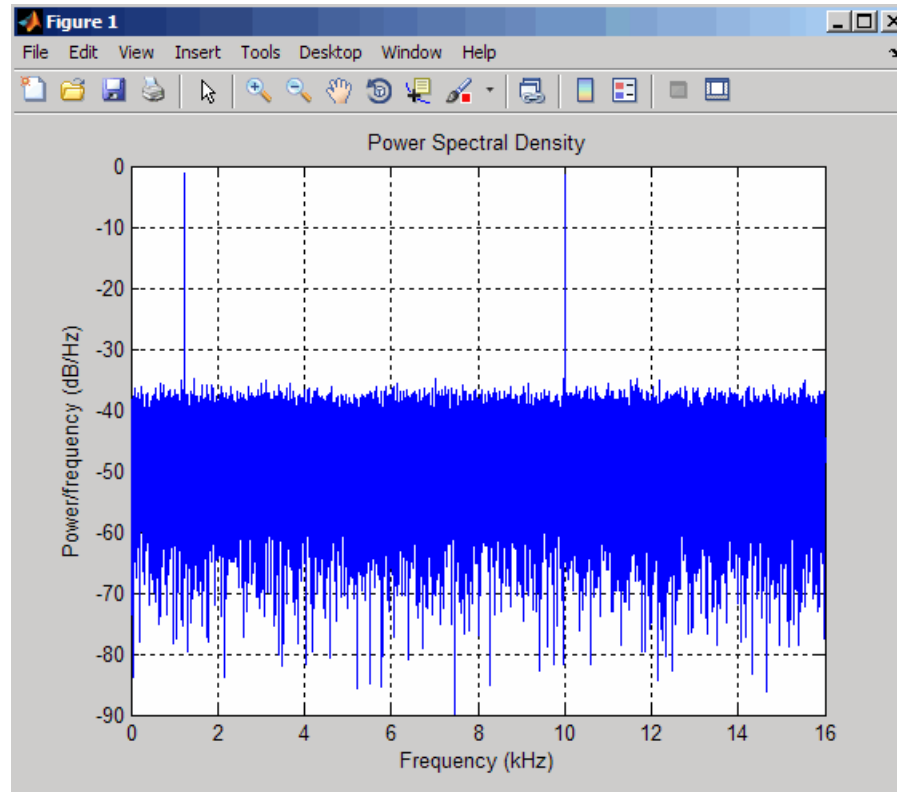
Examples

Resolving Signal Components

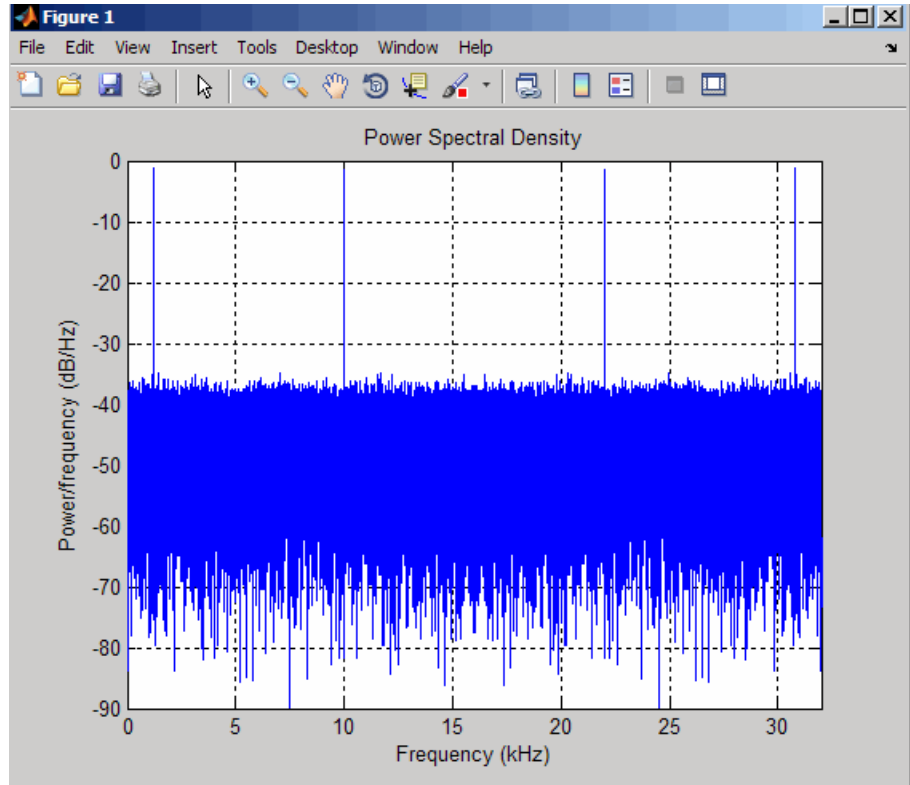
Estimate the power spectral density of a noisy sinusoidal signal with two frequency components and then store the results in a PSD data object and plot it.

```
Fs = 32e3;
t = 0:1/Fs:2.96;
x = cos(2*pi*t*1.24e3)+ cos(2*pi*t*10e3)+ randn(size(t));
nfft = 2^nextpow2(length(x));
Pxx = abs(fft(x,nfft)).^2/length(x)/Fs;
```

```
% Create a single-sided spectrum
Hpsd = dspdata.psd(Pxx(1:length(Pxx)/2),'Fs',Fs);
plot(Hpsd);
```




```
% Create a double-sided spectrum
Hpsd = dspdata.psd(Pxx,'Fs',Fs,'SpectrumType','twosided');
plot(Hpsd)
```



See Also `dspdata.msspectrum` | `dspdata.pseudospectrum` | `spectrum`

Purpose

Pseudospectrum `dspdata` object

Syntax

```
Hps = dspdata.pseudospectrum(Data)
Hps = dspdata.pseudospectrum(Data,Frequencies)
Hps = dspdata.pseudospectrum(...,'Fs',Fs)
Hps = dspdata.pseudospectrum(...,'SpectrumRange',SpectrumRange)
Hps = dspdata.pseudospectrum(...,'CenterDC',flag)
```

Description

A pseudospectrum is an indicator of the presence of sinusoidal components in a signal.

`Hps = dspdata.pseudospectrum(Data)` uses the pseudospsectrum data contained in `Data`, which can be in the form of a vector or a matrix, where each column is a separate set of data. Default values for other properties of the object are:

Property	Default Value	Description
Name	`'Pseudospectrum'`	Read-only string
`Frequencies`	`[]` type `double`	Vector of frequencies at which the power spectral density is evaluated. The range of this vector depends on the `SpectrumRange` value. For half, the default range is [0, pi) or [0, `Fs`/2) for odd length, and [0, pi] or [0, `Fs`/2] for even length, if `Fs` is specified. For whole, it is [0, 2pi) or [0, `Fs`). If you do not specify `Frequencies`, a default vector is created. If half the Nyquist range is selected, then the whole number of FFT points (nFFT) for this vector is assumed to be even. If `half` the Nyquist range is selected and you specify `Frequencies`, the last frequency point is compared to the next-to-last point and to pi (or `Fs`/2, if `Fs` is specified). If the last point is closer to pi (or `Fs`/2) than it is to the previous point, nFFT is assumed to be even. If it is closer to the previous point, nFFT is assumed to be odd. The length of the `Frequencies` vector must match the length of the columns of `Data`.
`Fs`	`'Normalized'`	Sampling frequency, which is `'Normalized'` if `NormalizedFrequency` is `true`. If `NormalizedFrequency` is `false` `Fs` defaults to `1`.

Property	Default Value	Description
SpectrumRange	'Half'	Nyquist interval over which the pseudospectrum is calculated. Valid values are 'Half' and 'Whole'. See the half and whole methods in dspdata for information on changing this property. The interval for Half is [0 pi) or [0 pi] depending on the number of FFT points, and for Whole the interval is [0 2pi).
NormalizedFrequency	true	Whether the frequency is normalized (true) or not (false). This property is set automatically at construction time based on Fs. If Fs is specified, NormalizedFrequency is set to false. See the normalizefreq method in dspdata for information on changing this property.

`Hps = dspdata.pseudospectrum(Data,Frequencies)` uses the pseudospectrum estimation data contained in the `Data` and `Frequencies` vectors.

`Hps = dspdata.pseudospectrum(...,'Fs',Fs)` uses the sampling frequency `Fs`. Specifying `Fs` uses a default set of linear frequencies (in `Hz`) based on `Fs` and sets `NormalizedFrequency` to `false`.

`Hps = dspdata.pseudospectrum(...,'SpectrumRange',SpectrumRange)` uses the `SpectrumRange` string to specify the interval over which the pseudospectrum was calculated. For data that ranges from [0 pi) or [0 pi], set the `SpectrumRange` to `half`; for data that ranges from [0 2pi), set the `SpectrumRange` to `whole`.

`Hps = dspdata.pseudospectrum(...,'CenterDC',flag)` uses the value of `flag` to indicate whether the zero-frequency (DC) component is centered. If `flag` is `true`, it indicates that the DC component is in the center of the whole Nyquist range spectrum. Set the `flag` to `false` if the DC component is on the left edge of the spectrum.

Methods

Methods provide ways of performing functions directly on your `dspdata` object. You can apply a method directly on the variable you assigned to your `dspdata.pseudospectrum` object. You can use the following methods with a `dspdata.pseudospectrum` object.

- `centerdc`
- `halfrange`
- `normalizefreq`
- `plot`
- `wholerange`

For example, to normalize the frequency and set the `NormalizedFrequency` parameter to true, use

```
Hps = normalizefreq(Hps)
```

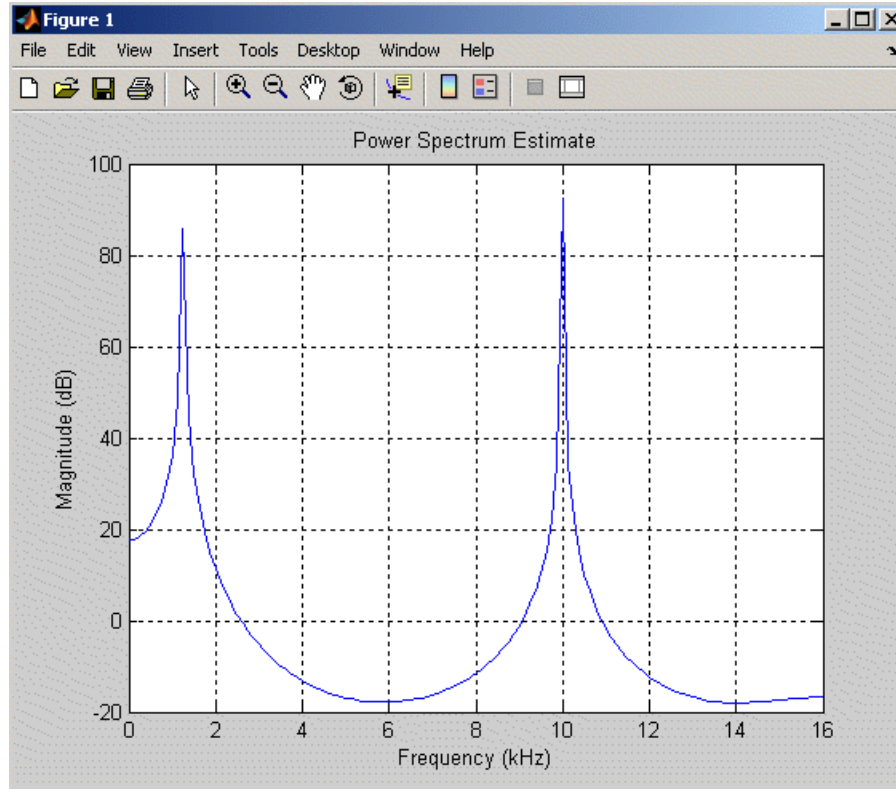
For detailed information on using the methods and plotting the pseudospectrum, see the `dspdata` reference page.

Examples

Storing and Plotting Pseudospectrum Data

Use eigenanalysis to estimate the pseudospectrum of a noisy sinusoidal signal with two frequency components. Then store the results in a `pseudospectrum` data object and plot it.

```
Fs = 32e3;
t  = 0:1/Fs:2.96;
x = cos(2*pi*t*1.24e3) + cos(2*pi*t*10e3) + randn(size(t));
P = pmusic(x,4);
% Create data object
hps = dspdata.pseudospectrum(P,'Fs',Fs);
% Plot the pseudospectrum
plot(hps);
```



See Also

`dspdata.msspectrum` | `dspdata.psd` | `spectrum`

dspfwiz

Purpose

Open FDATool Realize Model panel to create Simulink filter block

Syntax

```
dspfwiz
```

Description

Note You must have the Simulink product installed to use this function.

`dspfwiz` opens FDATool with the Realize Model panel displayed.

Use other panels in FDATool to design your filter and then use the Realize Model panel to create your filter as a subsystem block, which is a combination of Sum, Gain, and Delay blocks, in a Simulink model.

If you also have the DSP System Toolbox software installed, you can create a Digital Filter block instead of a subsystem block, by deselecting the **Build model using basic elements** check box.

See Also

`fdatool` | `dfilt`

Purpose

Duty cycle of pulse waveform

Syntax

```
D = dutycycle(X)
D = dutycycle(X,FS)
D = dutycycle(X,T)
D = dutycycle(TAU,PRF)
[D,INITCROSS] = dutycycle(X,...)
[D,INITCROSS,FINALCROSS] = dutycycle(X,...)
[D,INITCROSS,FINALCROSS,NEXTCROSS] = dutycycle(X,...)
[D,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] =
dutycycle(X,...)
[D,INITCROSS,FINALCROSS,NEXTCROSS] = dutycycle(X,...,Name,
   Value)
dutycycle(X,...)
```

Description

`D = dutycycle(X)` returns the ratio of pulse width to pulse period for each positive-polarity pulse. `D` has length equal to the number of pulse periods in `X`. The sample instants of `X` correspond to the indices of `X`. To determine the transitions that define each pulse, `dutycycle` estimates the state levels of the input waveform by a histogram method. `dutycycle` identifies all regions, which cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-261.

`D = dutycycle(X,FS)` specifies the sampling frequency, `FS`, in hertz as a positive scalar. The first sample instant of `X` corresponds to t=0.

`D = dutycycle(X,T)` specifies the sample instants, `T`, as a vector with the same number of elements as `X`.

`D = dutycycle(TAU,PRF)` returns the ratio of pulse width to pulse period for a pulse width of `TAU` seconds and a pulse repetition frequency of `PRF`. The product of `TAU` and `PRF` must be less than or equal to 1.

`[D,INITCROSS] = dutycycle(X,...)` returns a vector, `INITCROSS`, whose elements correspond to the mid-crossings (mid-reference level

instants) of the initial transition of each pulse with a corresponding NEXTCROSS.

`[D,INITCROSS,FINALCROSS] = dutycycle(X,...)` returns a vector, FINALCROSS, whose elements correspond to the mid-crossings (mid-reference level instants) of the final transition of each pulse with a corresponding NEXTCROSS.

`[D,INITCROSS,FINALCROSS,NEXTCROSS] = dutycycle(X,...)` returns a vector, NEXTCROSS, whose elements correspond to the mid-crossings (mid-reference level instants) of the next detected transition for each pulse.

`[D,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] = dutycycle(X,...)` returns the mid-reference level, MIDLEV. Because in a bilevel pulse waveform the state levels are constant, MIDLEV is a scalar.

`[D,INITCROSS,FINALCROSS,NEXTCROSS] = dutycycle(X,...,Name,Value)` returns the ratio of pulse width to pulse period with additional options specified by one or more `Name,Value` pair arguments.

`dutycycle(X,...)` plots the waveform, X, and marks the location of the mid-reference level instants and the associated reference levels. The state levels and associated lower and upper state boundaries are also plotted.

Input Arguments

X

Bilevel waveform. X is a real-valued row or column vector.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of T must equal the length of the bilevel waveform, X.

TAU

Pulse width in seconds. The product of `TAU` and `PRF` must be less than or equal to 1.

PRF

Pulse repetition frequency in pulses/second. The product of `TAU` and `PRF` must be less than or equal to 1.

Name-Value Pair Arguments

'MidPct'

Mid-reference level as a percentage of the waveform amplitude.

Default: `50`

'Polarity'

Pulse polarity. Specify the polarity as `'positive'` or `'negative'`. If you specify `'positive'`, `dutycycle` looks for pulses with positive-going (positive polarity) initial transitions. If you specify `'negative'`, `dutycycle` looks for pulses with negative-going (negative polarity) initial transitions. See "Pulse Polarity" on page 1-259 for examples of positive and negative-polarity pulses.

Default: `'positive'`

'StateLevels'

Low- and high-state levels. `StateLevels` is a 1-by-2 real-valued vector. The first element is the low-state level. The second element is the high-state level. If you do not specify low- and high-state levels, `dutycycle` estimates the state levels from the input waveform using the histogram method.

'Tolerance'

Tolerance levels (lower- and upper-state boundaries) expressed as a percentage. See “State-Level Tolerances” on page 1-261.

Default: `2`

Output Arguments

D

Duty cycle. Duty cycle is the ratio of the pulse width to the pulse period. Because the pulse width cannot exceed the pulse period, $0 \le D \le 1$.

INITCROSS

Mid-reference level instant of initial transition. Because the duty cycle is defined as the ratio of pulse width to pulse period, initial transitions are only reported when `dutycycle` finds a corresponding `NEXTCROSS`.

FINALCROSS

Mid-reference level instant of final transition. The duty cycle is defined as the ratio of pulse width to pulse period. Thus, final transitions are only reported when `dutycycle` finds a corresponding `NEXTCROSS`.

NEXTCROSS

Mid-reference level instant of the first initial transition after the final transition of the preceding pulse.

MIDLEV

Mid-reference level. The waveform value that corresponds to the mid-reference level.

Definitions

Duty Cycle

The energy in a bilevel, or rectangular, pulse is equal to the product of the peak power, *Pt*, and the pulse width, τ. Devices to measure energy in a waveform operate on time scales longer than the duration of a single pulse. Therefore, it is common to measure the average power

$$P_{\text{av}} = \frac{P_t \tau}{T},$$

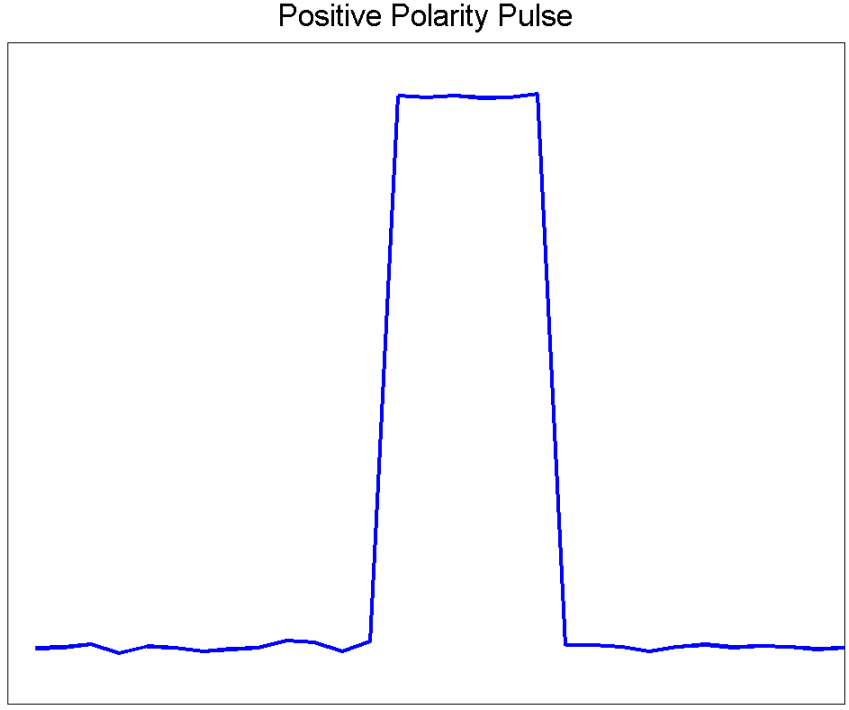
where T is the pulse period.

The ratio of average power to peak power is the duty cycle:

$$D = \frac{P_t \tau / T}{P_t}$$

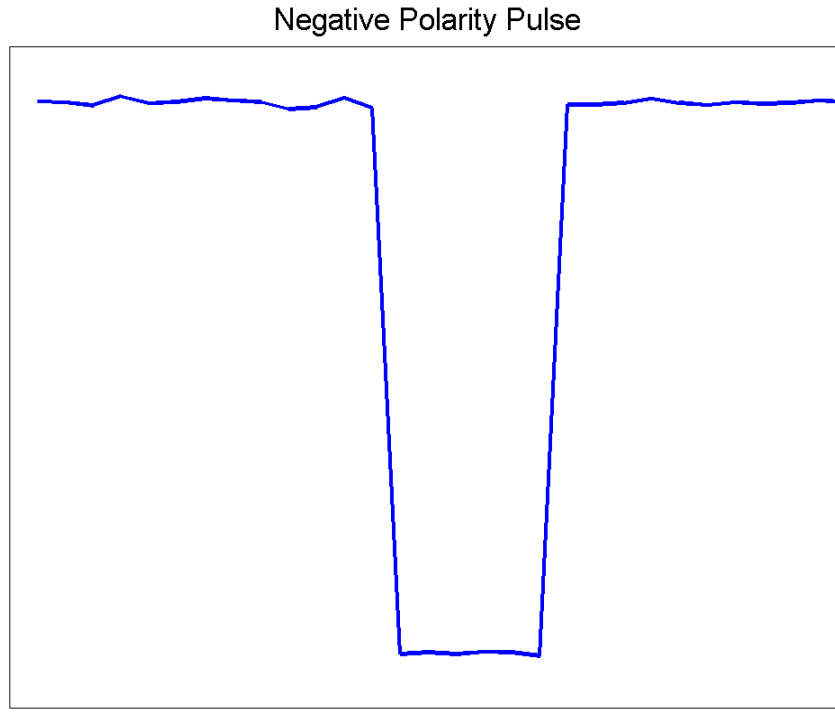
Pulse Polarity

If the pulse has a positive-going initial transition, the pulse has positive polarity. The following figure shows a positive polarity pulse.



Equivalently, a positive-polarity (positive-going) pulse has a terminating state more positive than the originating state.

If the pulse has a negative-going initial transition, the pulse has negative polarity. The following figure shows a negative-polarity pulse.



Equivalently, a negative-polarity (negative-going) pulse has a originating state more positive than the terminating state.

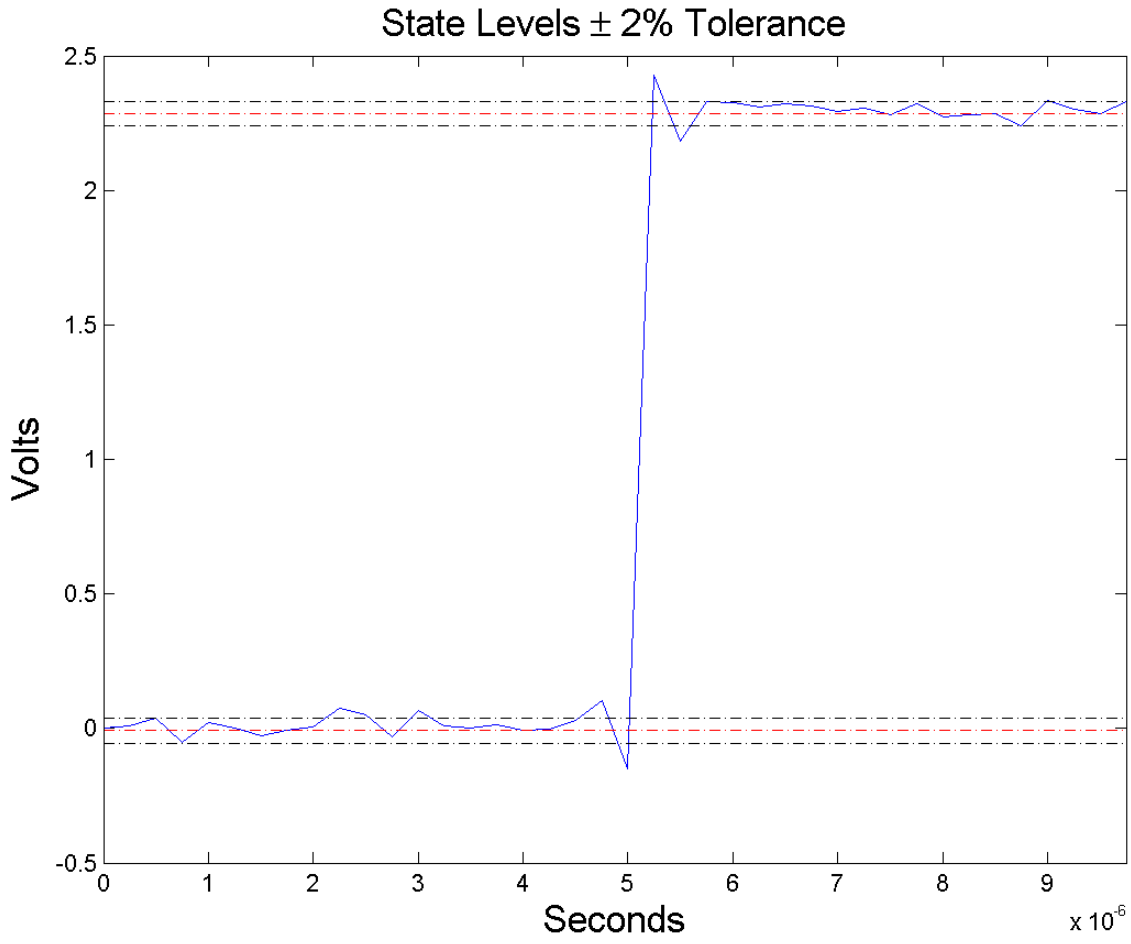
State-Level Tolerances

Each state level can have an associated lower- and upper-state boundary. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The estimated state levels are indicated by a dashed red line.



Examples

Duty Cycle of Bilevel Waveform

Determine the duty cycle of a bilevel waveform. Use the vector indices as the sample instants.

```
load('pulseex.mat', 'x');
d = dutycycle(x);
```

Duty Cycle of Bilevel Waveform with Sampling Frequency

Determine the duty cycle of a bilevel waveform. The sampling frequency is 4 MHz.

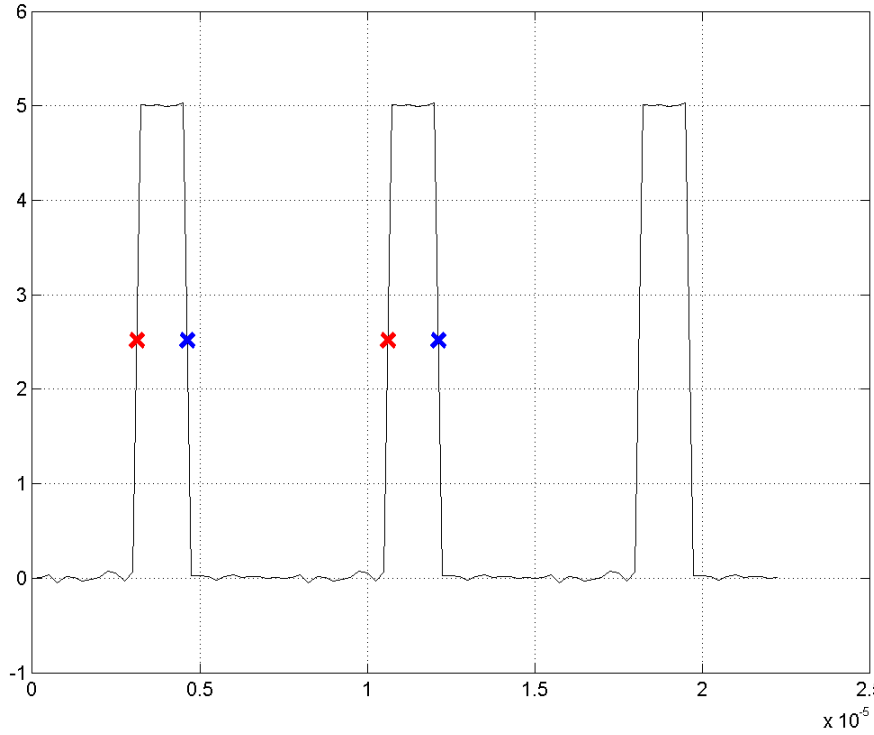
```
load('pulseex.mat', 'x','t');
fs = 1/(t(2)-t(1));
d = dutycycle(x,fs);
```

Duty Cycle of Bilevel Waveform with Three Pulses

Create a pulse waveform with three pulses. The sampling frequency is 4 MHz. Determine the initial and final mid-reference level instants. Plot the result.

Even though there are three pulses, only two pulses have corresponding subsequent transitions.

```
load('pulseex.mat','x');
dt = 1/4e6;
ts = reshape(repmat(x(1:30),1,3),90,1);
t = 0:dt:(length(ts)*dt)-dt;
[d,initcross,finalcross,~,midlev] = dutycycle(ts,t);
plot(t,ts,'k'); hold on; grid on;
h0 = plot(initcross, midlev*ones(length(initcross)),'rx');
set(h0,'markersize',10,'linewidth',2.5);
h1 = plot(finalcross,midlev*ones(length(finalcross)),'bx');
set(h1,'markersize',10,'linewidth',2.5);
```



References

[1] Skolnik, M.I. *Introduction to Radar Systems*. New York, NY: McGraw-Hill, 1980.

[2] *IEEE Standard on Transitions, Pulses, and Related Waveforms*. IEEE Standard 181, 2003.

See Also

`midcross` | `pulseperiod` | `pulsesep` | `pulsewidth`

Purpose

Elliptic filter design

Syntax

```
[z,p,k] = ellip(n,Rp,Rs,Wp)
[z,p,k] = ellip(n,Rp,Rs,Wp,'ftype')
[b,a] = ellip(n,Rp,Rs,Wp)
[b,a] = ellip(n,Rp,Rs,Wp,'ftype')
[A,B,C,D] = ellip(n,Rp,Rs,Wp)
[A,B,C,D] = ellip(n,Rp,Rs,Wp,'ftype')
[z,p,k] = ellip(n,Rp,Rs,Wp,'s')
[z,p,k] = ellip(n,Rp,Rs,Wp,'ftype','s')
[b,a] = ellip(n,Rp,Rs,Wp,'s')
[b,a] = ellip(n,Rp,Rs,Wp,'ftype','s')
[A,B,C,D] = ellip(n,Rp,Rs,Wp,'s')
[A,B,C,D] = ellip(n,Rp,Rs,Wp,'ftype','s')
```

Description

`ellip` designs lowpass, bandpass, highpass, and bandstop digital and analog elliptic filters. Elliptic filters offer steeper rolloff characteristics than Butterworth or Chebyshev filters, but are equiripple in both the pass- and stopbands. In general, elliptic filters meet given performance specifications with the lowest order of any filter type.

Digital Domain

`[z,p,k] = ellip(n,Rp,Rs,Wp)` designs an order `n` lowpass digital elliptic filter with normalized passband edge frequency `Wp`, `Rp` dB of ripple in the passband, and a stopband `Rs` dB down from the peak value in the passband. It returns the zeros and poles in length `n` column vectors `z` and `p` and the gain in the scalar `k`.

The *normalized passband edge frequency* is the edge of the passband, at which the magnitude response of the filter is `-Rp` dB. For `ellip`, the normalized cutoff frequency `Wp` is a number between 0 and 1, where 1 corresponds to half the sampling frequency (Nyquist frequency). Smaller values of passband ripple `Rp` and larger values of stopband attenuation `Rs` both lead to wider transition widths (shallower rolloff characteristics).

If `Wp` is a two-element vector, `Wp = [w1 w2]`, `ellip` returns an order `2*n` bandpass filter with passband `w1` < ω < `w2`.

`[z,p,k] = ellip(n,Rp,Rs,Wp,'ftype')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is one of the following:

- `'high'` for a highpass digital filter with normalized passband edge frequency `Wp`
- `'low'` for a lowpass digital filter with normalized passband edge frequency `Wp`
- `'stop'` for an order `2*n` bandstop digital filter if `Wp` is a two-element vector, `Wp = [w1 w2]`. The stopband is `w1` < ω < `w2`.

With different numbers of output arguments, `ellip` directly obtains other realizations of the filter. To obtain the transfer function form, use two output arguments as shown below.

Note See "Limitations" on page 1-270 for information about numerical issues that affect forming the transfer function.

`[b,a] = ellip(n,Rp,Rs,Wp)` designs an order `n` lowpass digital elliptic filter with normalized passband edge frequency `Wp`, `Rp` dB of ripple in the passband, and a stopband `Rs` dB down from the peak value in the passband. It returns the filter coefficients in the length `n+1` row vectors `b` and `a`, with coefficients in descending powers of z.

$$H(z) = \frac{b(1) + b(2)z^{-1} + \ldots + b(n+1)z^{-n}}{1 + a(2)z^{-1} + \ldots + a(n+1)z^{-n}}$$

`[b,a] = ellip(n,Rp,Rs,Wp,'ftype')` designs a highpass, lowpass, or bandstop filter, where the string `'ftype'` is `'high'`, `'low'`, or `'stop'`, as described above.

To obtain state-space form, use four output arguments as shown below:

`[A,B,C,D] = ellip(n,Rp,Rs,Wp)` or

`[A,B,C,D] = ellip(n,Rp,Rs,Wp,'ftype')` where `A`, `B`, `C`, and `D` are

$$x[n+1] = Ax[n] + Bu[n]$$
$$y[n] = Cx[n] + Du[n]$$

and u is the input, x is the state vector, and y is the output.

Analog Domain

`[z,p,k] = ellip(n,Rp,Rs,Wp,'s')` designs an order `n` lowpass analog elliptic filter with angular passband edge frequency `Wp` rad/s and returns the zeros and poles in length `n` or `2*n` column vectors `z` and `p` and the gain in the scalar `k`.

The *angular passband edge frequency* is the edge of the passband, at which the magnitude response of the filter is `-Rp` dB. For `ellip`, the angular passband edge frequency `Wp` must be greater than 0 rad/s.

If `Wp` is a two-element vector with `w1` < `w2`, then ellip(n,Rp,`Rs,Wp,`'s') returns an order `2*n` bandpass analog filter with passband `w1` < ω< `w2`.

`[z,p,k] = ellip(n,Rp,Rs,Wp,'`*`ftype`*`','s')` designs a highpass, lowpass, or bandstop filter, where the string '*`ftype`*' is `'high'`, `'low'`, or `'stop'`, as described above.

With different numbers of output arguments, ellip directly obtains other realizations of the analog filter. To obtain the transfer function form, use two output arguments as shown below:

`[b,a] = ellip(n,Rp,Rs,Wp,'s')` designs an order `n` lowpass analog elliptic filter with angular passband edge frequency `Wp` rad/s and returns the filter coefficients in the length `n+1` row vectors `b` and `a`, in descending powers of s, derived from this transfer function:

$$H(s) = \frac{B(s)}{A(s)} = \frac{b(1)s^n + b(2)s^{n-1} + \ldots + b(n+1)}{s^n + a(2)s^{n-1} + \ldots + a(n+1)}$$

`[b,a] = ellip(n,Rp,Rs,Wp,'`*`ftype`*`','s')` designs a highpass, lowpass, or bandstop filter, where the string '*`ftype`*' is `'high'`, `'low'`, or `'stop'`, as described above.

To obtain state-space form, use four output arguments as shown below:

`[A,B,C,D] = ellip(n,Rp,Rs,Wp,'s')` or

`[A,B,C,D] = ellip(n,Rp,Rs,Wp,'ftype','s')` where `A`, `B`, `C`, and `D` are

$$x = Ax + Bu$$
$$y = Cx + Du$$

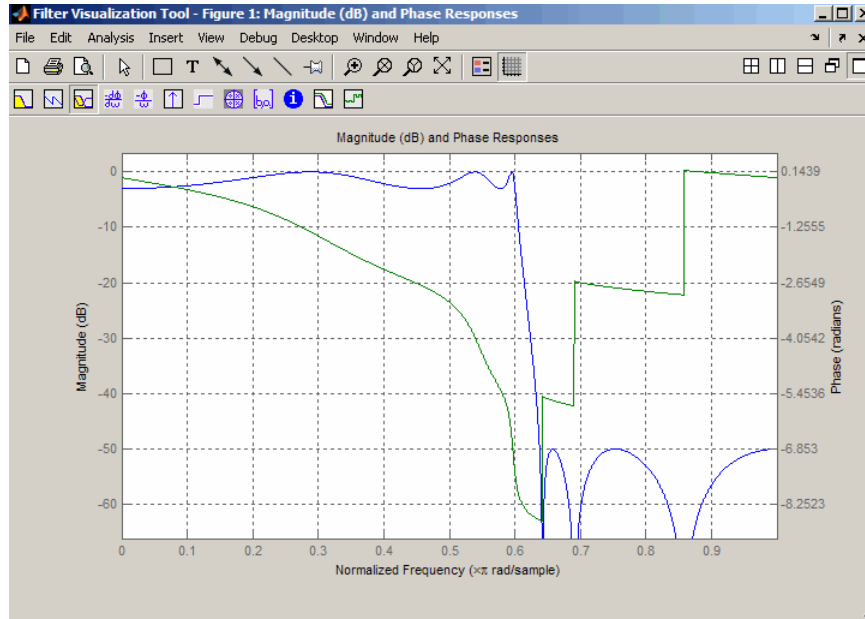
and u is the input, x is the state vector, and y is the output.

Examples

Lowpass Filter

For data sampled at 1000 Hz, design a sixth-order lowpass elliptic filter with a passband edge frequency of 300 Hz, which corresponds to a normalized value of 0.6, 3 dB of ripple in the passband, and 50 dB of attenuation in the stopband:

```
[z,p,k] = ellip(6,3,50,300/500);
[sos,g] = zp2sos(z,p,k);       % Convert to SOS form
Hd = dfilt.df2tsos(sos,g);    % Create a dfilt object
h = fvtool(Hd)                % Plot magnitude response
set(h,'Analysis','freq')      % Display frequency response
```

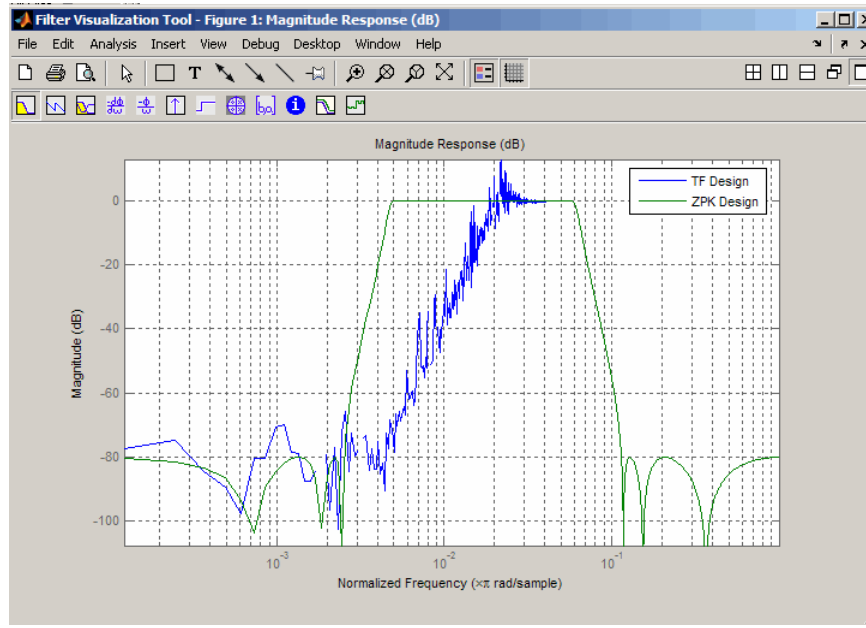


Limitations

In general, you should use the `[z,p,k]` syntax to design IIR filters. To analyze or implement your filter, you can then use the `[z,p,k]` output with `zp2sos` and an sos `dfilt` structure. For higher order filters (possibly starting as low as order 8), numerical problems due to roundoff errors may occur when forming the transfer function using the `[b,a]` syntax. The following example illustrates this limitation:

```
n = 6;
Rp = .1; Rs = 80;
Wn = [2.5e6 29e6]/500e6;
ftype = 'bandpass';

% Transfer Function design
[b,a] = ellip(n,Rp,Rs,Wn,ftype);
h1=dfilt.df2(b,a);    % This is an unstable filter.

% Zero-Pole-Gain design
```

```
[z, p, k] = ellip(n,Rp,Rs,Wn,ftype);
[sos,g]=zp2sos(z,p,k);
h2=dfilt.df2sos(sos,g);

% Plot and compare the results
hfvt=fvtool(h1,h2,'FrequencyScale','log');
legend(hfvt,'TF Design','ZPK Design')
```



Algorithms

The design of elliptic filters is the most difficult and computationally intensive of the Butterworth, Chebyshev Type I and II, and elliptic designs. `ellip` uses a five-step algorithm:

1. It finds the lowpass analog prototype poles, zeros, and gain using the `ellipap` function.

2. It converts the poles, zeros, and gain into state-space form.

3. It transforms the lowpass filter to a bandpass, highpass, or bandstop filter with the desired cutoff frequencies using a state-space transformation.

4. For digital filter design, `ellip` uses `bilinear` to convert the analog filter into a digital filter through a bilinear transformation with frequency prewarping. Careful frequency adjustment guarantees that the analog filters and the digital filters will have the same frequency response magnitude at `Wp` or `w1` and `w2`.

5. It converts the state-space filter back to transfer function or zero-pole-gain form, as required.

See Also

`besself` | `butter` | `cheby1` | `cheby2` | `ellipap` | `ellipord`

Purpose

Elliptic analog lowpass filter prototype

Syntax

```
[z,p,k] = ellipap(n,Rp,Rs)
```

Description

`[z,p,k] = ellipap(n,Rp,Rs)` returns the zeros, poles, and gain of an order `n` elliptic analog lowpass filter prototype, with `Rp` dB of ripple in the passband, and a stopband `Rs` dB down from the peak value in the passband. The zeros and poles are returned in length `n` column vectors `z` and `p` and the gain in scalar `k`. If `n` is odd, `z` is length `n - 1`. The transfer function in factored zero-pole form is

$$H(s) = \frac{z(s)}{p(s)} = k\frac{(s-z_1)(s-z_2)\ldots(s-z_N)}{(s-p_1)(s-p_2)\ldots(s-p_M)}$$

Elliptic filters offer steeper rolloff characteristics than Butterworth and Chebyshev filters, but they are equiripple in both the passband and the stopband. Of the four classical filter types, elliptic filters usually meet a given set of filter performance specifications with the lowest filter order.

`ellipap` sets the passband edge angular frequency ω_0 of the elliptic filter to 1 for a normalized result. The *passband edge angular frequency* is the frequency at which the passband ends and the filter has a magnitude response of $10^{-Rp/20}$.

Algorithms

`ellipap` uses the algorithm outlined in [1]. It employs `ellipk` to calculate the complete elliptic integral of the first kind and `ellipj` to calculate Jacobi elliptic functions.

References

[1] Parks, T.W., and C.S. Burrus. *Digital Filter Design,* New York: John Wiley & Sons, 1987. Chapter 7.

See Also

`besselap` | `buttap` | `cheb1ap` | `cheb2ap` | `ellip`

Purpose

Minimum order for elliptic filters

Syntax

```
[n,Wp] = ellipord(Wp,Ws,Rp,Rs)
[n,Wp] = ellipord(Wp,Ws,Rp,Rs,'s')
```

Description

`ellipord` calculates the minimum order of a digital or analog elliptic filter required to meet a set of filter design specifications.

Digital Domain

`[n,Wp] = ellipord(Wp,Ws,Rp,Rs)` returns the lowest order `n` of the elliptic filter that loses no more than `Rp` dB in the passband and has at least `Rs` dB of attenuation in the stopband. The scalar (or vector) of corresponding cutoff frequencies `Wp`, is also returned. Use the output arguments `n` and `Wp` in `ellip`.

Choose the input arguments to specify the stopband and passband according to the following table.

Description of Stopband and Passband Filter Parameters

Parameter	Description
`Wp`	Passband corner frequency `Wp`, the cutoff frequency, is a scalar or a two-element vector with values between 0 and 1, with 1 corresponding to the normalized Nyquist frequency, π radians per sample.
`Ws`	Stopband corner frequency `Ws`, is a scalar or a two-element vector with values between 0 and 1, with 1 corresponding to the normalized Nyquist frequency.
`Rp`	Passband ripple, in decibels. This value is the maximum permissible passband loss in decibels.
`Rs`	Stopband attenuation, in decibels. This value is the number of decibels the stopband is attenuated with respect to the passband response.

Use the following guide to specify filters of different types.

Filter Type Stopband and Passband Specifications

Filter Type	Stopband and Passband Conditions	Stopband	Passband
Lowpass	`Wp` < `Ws`, both scalars	`(Ws,1)`	`(0,Wp)`
Highpass	`Wp` > `Ws`, both scalars	`(0,Ws)`	`(Wp,1)`
Bandpass	The interval specified by `Ws` contains the one specified by `Wp` (`Ws(1) < Wp(1) < Wp(2) < Ws(2)`).	`(0,Ws(1))` and `(Ws(2),1)`	`(Wp(1),Wp(2))`
Bandstop	The interval specified by `Wp` contains the one specified by `Ws` (`Wp(1) < Ws(1) < Ws(2) < Wp(2)`).	`(0,Wp(1))` and `(Wp(2),1)`	`(Ws(1),Ws(2))`

If your filter specifications call for a bandpass or bandstop filter with unequal ripple in each of the passbands or stopbands, design separate lowpass and highpass filters according to the specifications in this table, and cascade the two filters together.

Analog Domain

`[n,Wp] = ellipord(Wp,Ws,Rp,Rs,'s')` finds the minimum order `n` and cutoff frequencies `Wp` for an analog filter. You specify the frequencies `Wp` and `Ws` similar to those described in the Description of Stopband and Passband Filter Parameters on page 1-274 table above, only in this case you specify the frequency in radians per second, and the passband or the stopband can be infinite.

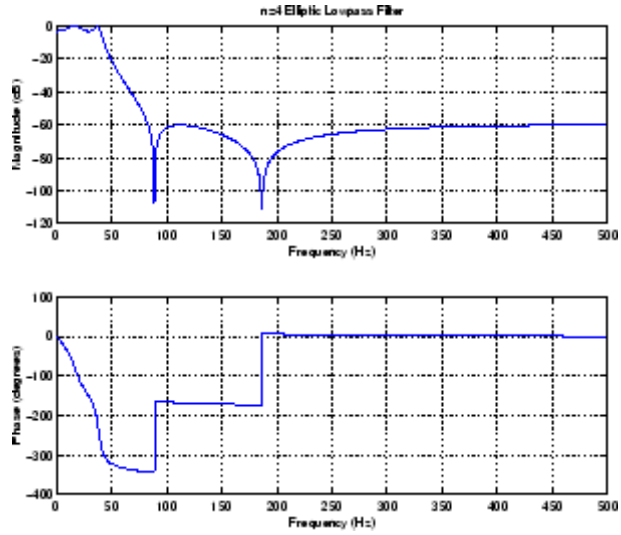
Use `ellipord` for lowpass, highpass, bandpass, and bandstop filters as described in the Filter Type Stopband and Passband Specifications on page 1-275 table above.

Examples

Example 1

For 1000 Hz data, design a lowpass filter with less than 3 dB of ripple in the passband defined from 0 to 40 Hz and at least 60 dB of ripple in the stopband defined from 150 Hz to the Nyquist frequency (500 Hz):

```
Wp = 40/500; Ws = 150/500;
Rp = 3; Rs = 60;
[n,Wp] = ellipord(Wp,Ws,Rp,Rs)
% Returns n =4 Wp =0.0800
[b,a] = ellip(n,Rp,Rs,Wp);
freqz(b,a,512,1000);
title('n=4 Elliptic Lowpass Filter')
```

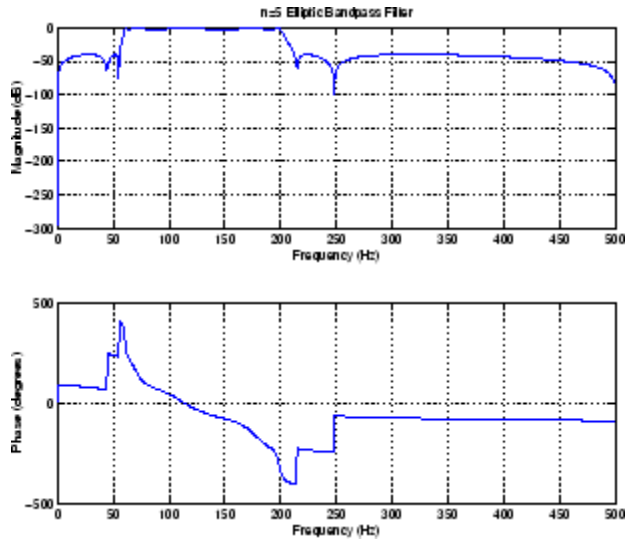


Example 2

Now design a bandpass filter with a passband from 60 Hz to 200 Hz, with less than 3 dB of ripple in the passband, and 40 dB attenuation in the stopbands that are 50 Hz wide on both sides of the passband:

```
Wp = [60 200]/500; Ws = [50 250]/500;
```

```
Rp = 3; Rs = 40;
[n,Wp] = ellipord(Wp,Ws,Rp,Rs)
% Returns n =5 Wp =[0.1200    0.4000]
[b,a] = ellip(n,Rp,Rs,Wp);
freqz(b,a,512,1000);
title('n=5 Elliptic Bandpass Filter')
```



Algorithms

`ellipord` uses the elliptic lowpass filter order prediction formula described in [1]. The function performs its calculations in the analog domain for both the analog and digital cases. For the digital case, it converts the frequency parameters to the *s*-domain before estimating the order and natural frequencies, and then converts them back to the *z*-domain.

`ellipord` initially develops a lowpass filter prototype by transforming the passband frequencies of the desired filter to 1 rad/s (for low- and highpass filters) and to -1 and 1 rad/s (for bandpass and bandstop filters). It then computes the minimum order required for a lowpass filter to meet the stopband specification.

References

[1] Rabiner, L.R., and B. Gold. *Theory and Application of Digital Signal Processing,* Englewood Cliffs, NJ: Prentice-Hall, 1975. Pg. 241.

See Also

`buttord` | `cheb1ord` | `cheb2ord` | `ellip`

Purpose

Equivalent noise bandwidth

Syntax

```
bw = enbw(window)
bw = enbw(window,fs)
```

Description

`bw = enbw(window)` returns the two-sided equivalent noise bandwidth, `bw`, for a uniformly sampled window, `window`. The equivalent noise bandwidth is normalized by the noise power per frequency bin.

`bw = enbw(window,fs)` returns the two-sided equivalent noise bandwidth, `bw`, in Hz.

Input Arguments

window - Window vector

real-valued row or column vector

Uniformly sampled window vector, specified as a row or column vector with real-valued elements.

Example: `hamming(1000)`

Data Types
`double | single`

fs - Sampling frequency

positive scalar

Sampling frequency, specified as a positive scalar.

Output Arguments

bw - Equivalent noise bandwidth

positive scalar

Equivalent noise bandwidth, specified as a positive scalar.

Data Types
`double | single`

Examples

Equivalent Noise Bandwidth of Hamming Window

Determine the equivalent noise bandwidth of a Hamming window 1,000 samples in length.

```
bw = enbw(hamming(1000));
```

Equivalent Noise Bandwidth of Flat Top Window

Determine the equivalent noise bandwidth in Hz of a flat top window 10,000 samples in length. The sampling frequency is 44.1 kHz.

```
bw = enbw(flattopwin(10000), 44.1e3);
```

Equivalent Rectangular Noise Bandwidth

Obtain the equivalent rectangular noise bandwidth of a Von Hann window and overlay the equivalent rectangular bandwidth on the window's magnitude spectrum. The window is 1000 samples in length and the sampling frequency is 10 kHz.

Set the sampling frequency, create the window, and obtain the discrete Fourier transform of the window with 0 frequency in the center of the spectrum.

```
Fs = 10000;
win = hann(1000);
windft = fftshift(fft(win));
```

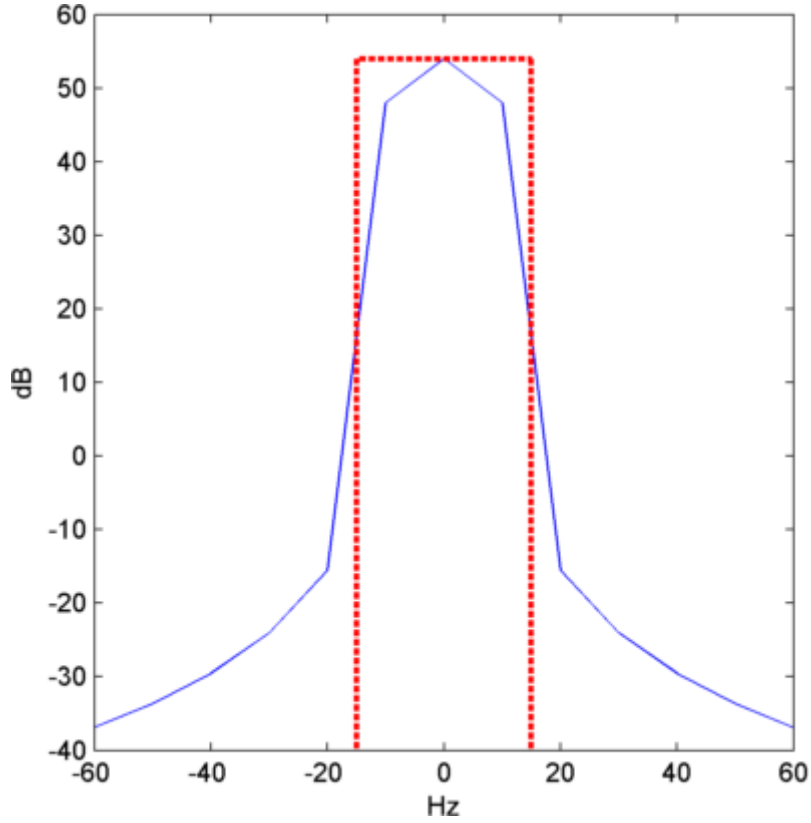
Obtain the equivalent (rectangular) noise bandwidth of the Von Hann window.

```
bw = enbw(hann(1000),Fs);
```

Plot the squared-magnitude DFT of the window and use the equivalent noise bandwidth to overlay the equivalent rectangle.

```
freq = -(Fs/2):Fs/length(win):Fs/2-(Fs/length(win));
plot(freq,20*log10(abs(windft))); xlabel('Hz'); ylabel('dB');
axis([-60 60 -40 60])
maxgain = 20*log10(abs(windft(length(win)/2+1)));
```

```
hold on;
plot([-bw -bw],[-40 maxgain],'r--',...
    [bw bw],[-40 maxgain],'r--','linewidth',2);
plot([-bw bw],[maxgain maxgain],'r--','linewidth',2);
```



Definitions

Equivalent Noise Bandwidth

The equivalent noise bandwidth of a window is the width of a rectangle whose area contains the same total power as the window. The height of the rectangle is the peak squared magnitude of the window's Fourier transform.

Assuming a sampling interval of 1, the total energy for the window, *w(n)*, can be expressed in the frequency or time-domain as

$$\int_{-1/2}^{1/2} |W(f)|^2 \, df = \sum_n |w(n)|^2$$

The peak magnitude of the window's spectrum occurs at *f*=0. This is given by

$$|W(0)|^2 = |\sum_n w(n)|^2$$

To find the width of the equivalent rectangular bandwidth, divide the area by the height.

$$\frac{\int_{-1/2}^{1/2} |W(f)|^2 \, df}{|W(0)|^2} = \frac{\sum_n |w(n)|^2}{|\sum_n w(n)|^2}$$

See "Equivalent Rectangular Noise Bandwidth" on page 1-280 for an example that plots the equivalent rectangular bandwidth over the magnitude spectrum of a Von Hann window.

See Also

`bandpower` | `sfdr`

Purpose

Equiripple single-rate FIR filter from specification object

Syntax

```
hd = design(d,'equiripple')
hd = design(d,'equiripple',designoption,value,designoption,
...value,...)
```

Description

`hd = design(d,'equiripple')` designs an equiripple FIR digital filter using the specifications supplied in the object `d`. Equiripple filter designs minimize the maximum ripple in the passbands and stopbands. `hd` is a `dfilt` object

`hd = design(d,'equiripple',designoption,value,designoption, ...value,...)` returns an equiripple FIR filter where you specify design options as input arguments.

To determine the available design options, use `designopts` with the specification object and the design method as input arguments as shown.

```
designopts(d,'method')
```

For complete help about using `equiripple`, refer to the command line help system. For example, to get specific information about using `equiripple` with `d`, the specification object, enter the following at the MATLAB prompt.

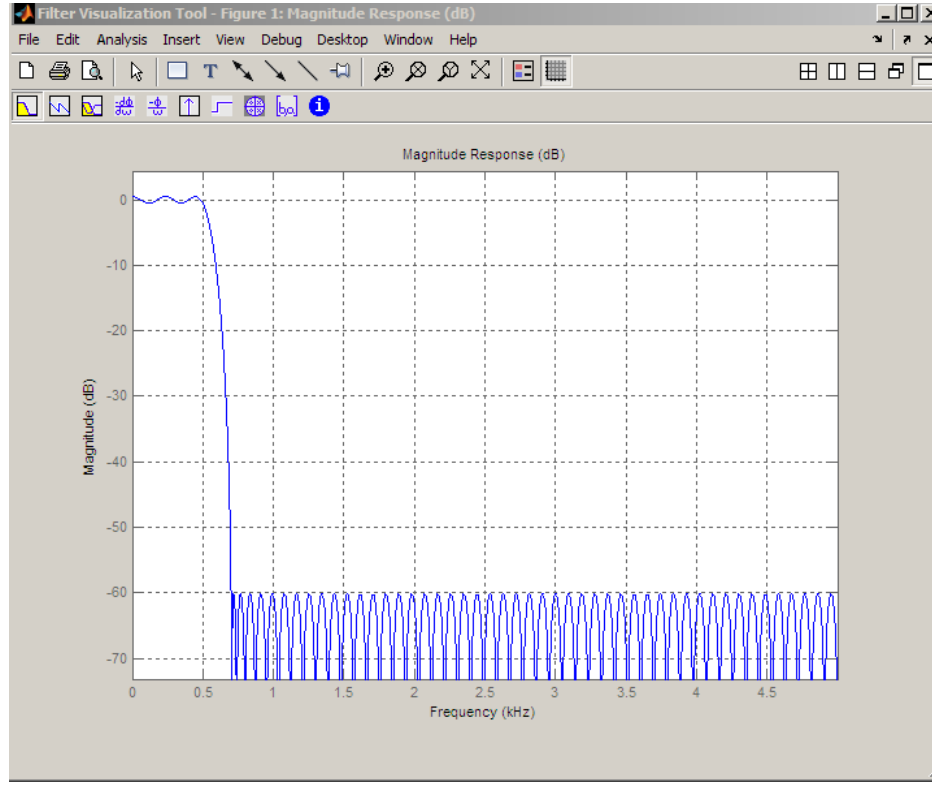
```
help(d,'equiripple')
```

Examples

First create a lowpass equiripple filter. Assume the data is sampled at 10,000 Hertz. The passband frequency is 500 Hertz with a stopband frequency of 700 Hz. The desired passband ripple is 1 dB with 60 dB of stopband attenuation.

```
Fs=10000;
Hd=fdesign.lowpass('Fp,Fst,Ap,Ast',500,700,1,60,10000);
d=design(Hd,'equiripple');
fvtool(d);
```

Displaying the filter in FVTool shows the equiripple nature of the filter.



The next example designs a lowpass equiripple filter with a direct-form transposed structure and density factor of 20 by specifying the `FilterStructure` and `DensityFactor` properties.

To set the design options for the filter, use the `designopts` method to obtain a structure array containing the current design options.

Change the fields of the structure array to specify your design options and invoke the `design` method with the structure array as an input argument.

```
% Use the same filter design as the previous example
Fs = 10000;
```

```
Hd = fdesign.lowpass('Fp,Fst,Ap,Ast',500,700,1,60,10000);
% Return the design options for the filter as a struct array
opts = designopts(Hd,'equiripple');
fieldnames(opts)
% Print out the filter structure- - direct-form FIR
opts.FilterStructure
% Change the filter structure to direct-form FIR transposed
opts.FilterStructure = 'dffirt';
% Change the filter density factor to 20
opts.DensityFactor = 20;
% Design the filter
d = design(Hd,'equiripple',opts)
```

An alternate way to design the preceding filter without using the structure array is:

```
Fs = 10000;
Hd = fdesign.lowpass('Fp,Fst,Ap,Ast',500,700,1,60,10000);
d = design(Hd,'equiripple','FilterStructure','dffirt','DensityFactor',
```

See Also

`design` | `designmethods`

eqtflength

Purpose

Equalize lengths of transfer function's numerator and denominator

Syntax

```
[b,a] = eqtflength(num,den)
[b,a,n,m] = eqtflength(num,den)
```

Description

`[b,a] = eqtflength(num,den)` modifies the vector `num` and/or the vector `den`, so that the resulting output vectors `b` and `a` have the same length. The input vectors `num` and `den` may have different lengths. The vector `num` represents the numerator polynomial of a given discrete-time transfer function, and the vector `den` represents its denominator. The resulting numerator `b` and denominator `a` represent the same discrete-time transfer function, but these vectors have the same length.

`[b,a,n,m] = eqtflength(num,den)` modifies the vectors as above and also returns the numerator order `n` and the denominator `m`, not including any trailing zeros.

Use `eqtflength` to obtain a numerator and denominator of equal length before applying transfer function conversion functions such as `tf2ss` and `tf2zp` to discrete-time models.

Examples

```
num = [1 0.5];
den = [1 0.75 0.6 0];
[b,a,n,m] = eqtflength(num,den);
```

Algorithms

`eqtflength(num,den)` appends zeros to either `num` or `den` as necessary. If both `num` and `den` have trailing zeros in common, these are removed.

See Also

`tf2ss` | `tf2zp`

Purpose

Fall time of negative-going bilevel waveform transitions

Syntax

```
F = falltime(X)
F = falltime(X,FS)
F = falltime(X,T)
[F,LT,UT] = falltime(...)
[F,LT,UT,LL,UL] = falltime(...)
[...] = falltime(...,Name,Value)
falltime(...)
```

Description

`F = falltime(X)` returns a vector, `F`, containing the time each transition of the bilevel waveform, `X`, takes to cross from the 90% to 10% reference levels. See "Percent Reference Levels" on page 1-291. To determine the transitions, `falltime` estimates the state levels of the input waveform by a histogram method. `falltime` identifies all regions, which cross the lower-state boundary of the high state and the upper-state boundary of the low state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-291. Because `falltime` uses interpolation, `F` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`F = falltime(X,FS)` specifies the sampling frequency in hertz. The sampling frequency determines the sample instants corresponding to the elements in `X`. The first sample instant in `X` corresponds to t=0. Because `falltime` uses interpolation, `F` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`F = falltime(X,T)` specifies the sample instants, `T`, as a vector with the same number of elements as `X`.

`[F,LT,UT] = falltime(...)` returns vectors, `LT` and `UT`, whose elements correspond to the time instants where `X` crosses the lower and upper percent reference levels.

`[F,LT,UT,LL,UL] = falltime(...)` returns the levels, `LL` and `UL`, corresponding to the lower- and upper-percent reference levels.

`[...] = falltime(...,Name,Value)` returns the fall times with additional options specified by one or more `Name,Value` pair arguments.

`falltime(...)` plots the signal and darkens the regions of each transition where fall time is computed. The plot marks the lower and upper crossings and the associated reference levels. The state levels and the associated lower- and upper-state boundaries are also displayed.

Input Arguments

X

Bilevel waveform. `X` is a real-valued row or column vector.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

'PctRefLevels'

Reference levels as a percentage of the waveform amplitude. The low-state level is defined to be 0 percent. The high-state level is defined to be 100 percent. See "Percent Reference Levels" on page 1-291. `'PCTREFLEVELS'` is a 2-element real row vector whose elements correspond to the lower- and upper-percent reference levels.

Default: `[10 90]`

'StateLevels'

Low and high-state levels. Specifies the levels to use for the low- and high-state levels as a 2-element real-valued row vector whose first and second elements correspond to the low- and high-state levels.

'Tolerance'

Tolerance levels (lower- and upper-state boundaries) expressed as a percentage. See "State-Level Tolerances" on page 1-291.

Default: 2

Output Arguments

F

Fall times. F is a vector containing the duration of each negative-going transition. If you specify the sampling rate, FS, or the sampling instants, T, fall times are in seconds. If you do not specify a sampling rate, or sampling instants, fall times are in samples.

LT

Instants when negative-going transition crosses the lower-reference level. By default, the lower-reference level is the 10% reference level. You can change the default reference levels by specifying the 'PctRefLevels' name-value pair.

UT

Instants when negative-going transition crosses the upper-reference level. By default, the upper reference level is the 90% reference level. You can change the default reference levels by specifying the 'PctRefLevels' name-value pair.

LL

Lower-reference level in waveform amplitude units. LL is a vector containing the waveform values corresponding to the lower-reference level in each negative-going transition. By default, the lower-reference level is the 10% reference level. You can change the default reference levels by specifying the 'PctRefLevels' name-value pair.

UL

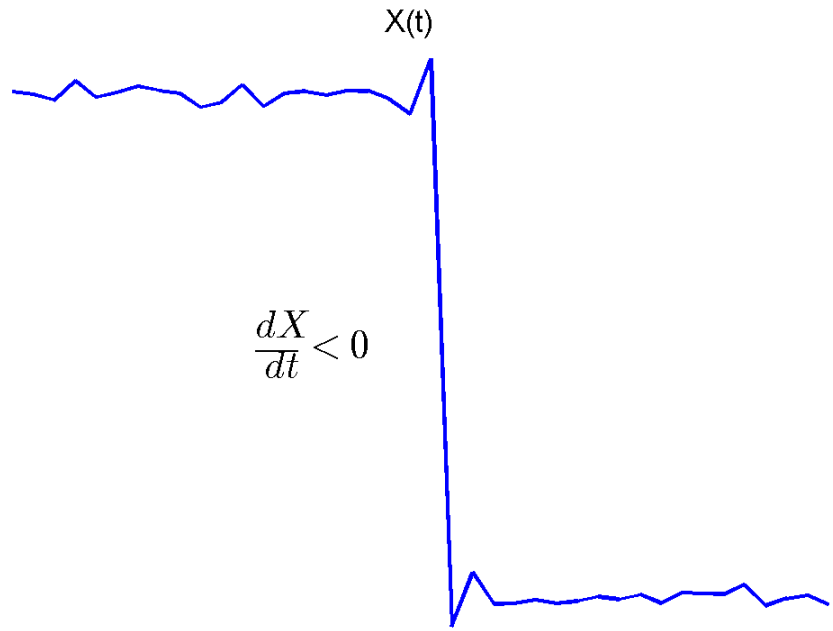
Upper-reference level in waveform amplitude units. LL is a vector containing the waveform values corresponding to the upper-reference

level in each negative-going transition. By default, the upper-reference level is the 90% reference level. You can change the default reference levels by specifying the `'PctRefLevels'` name-value pair.

Definitions

Negative-Going Transition

A negative-going transition in a bilevel waveform is a transition from the high–state level to the low-state level. If the waveform is differentiable in the neighborhood of the transition, an equivalent definition is a transition with a negative first derivative. The following figure shows a negative-going transition.



In the preceding figure, the amplitude values of the waveform are not displayed because a negative-going transition does not depend on the actual waveform values. A negative-going transition is defined by the direction of the transition.

Percent Reference Levels

If S_1 is the low state, S_2 is the high state, and U is the *upper*-percent reference level. The waveform value corresponding to the upper percent reference level is

$$S_1 + \frac{U}{100}(S_2 - S_1)$$

If L is the *lower* percent reference level, the waveform value corresponding to the lower percent reference level is

$$S_1 + \frac{L}{100}(S_2 - S_1)$$

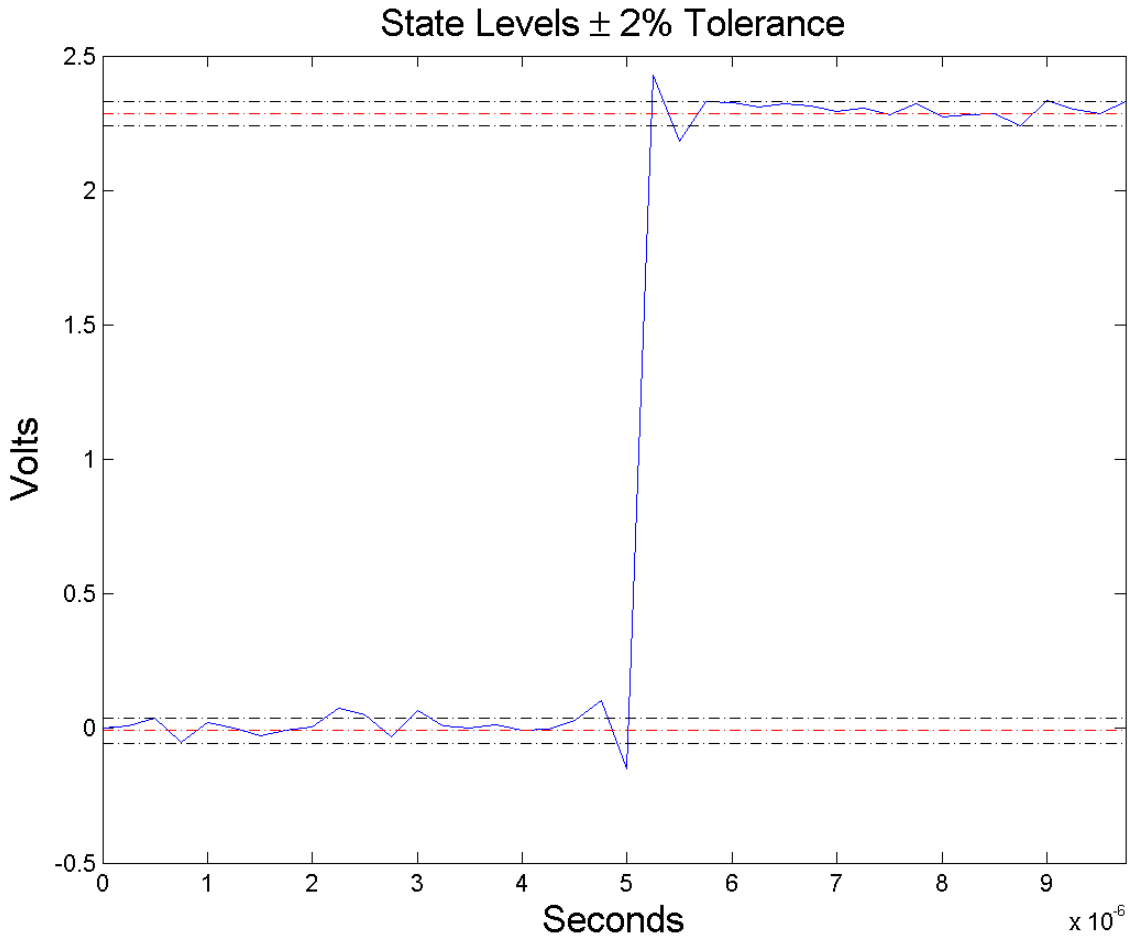
State-Level Tolerances

Each state level can have associated lower- and upper-state boundaries. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity (positive-going) bilevel waveform. The estimated state levels are indicated by a dashed red line.



Examples

Falltime in a Bilevel Waveform

Determine the fall time in samples for a 2.3 V clock waveform.

Load the 2.3 V clock data. Determine the fall time in samples. Use the default [10 90] percent reference levels.

```
load('negtransitionex.mat', 'x');
F = falltime(x);
```

The fall time is less than 1, indicating that the transition occurred in a fraction of a sample.

Falltime with 20% and 80% Reference Levels

Determine the fall time in a 2.3 V clock waveform sampled at 4 MHz. Compute the fall time using the 20% and 80% reference levels.

Load the 2.3 V clock data with sampling instants. Plot the waveform.

```
load('negtransitionex.mat','x','t');
plot(t,x);
```

Determine the fall time using the 20% and 80% reference levels..

```
F = falltime(x,'PctRefLevels',[20 80]);
```

Falltime, Reference-Level Instants, and Reference Levels

Determine the fall time, reference-level instants, and reference levels in a 2.3 V clock waveform sampled at 4 MHz.

Load the 2.3 V clock waveform along with the sampling instants.

```
load('negtransitionex.mat','x','t');
```

Determine the falltime, reference-level instants, and reference levels.

```
[F,LT,UT,LL,UL] = falltime(x,t);
```

Plot the waveform in microseconds with the upper and lower reference levels and reference level instants. Show that the fall time is the difference between the lower- and upper-reference level instants.

```
plot(t.*1e6,x);
```

```
xlabel('microseconds'); ylabel('Volts');
hold on; grid on;
plot(LT.*1e6,LL,'ro','markerfacecolor',[1 0 0]);
plot(UT.*1e6,UL,'ro','markerfacecolor',[1 0 0]);
fprintf('Rise time is %1.4f microseconds.\n',(LT-UT)*1e6)
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003, pp. 15–17.

See Also

`risetime` | `slewrate` | `statelevels`

Purpose

Open Filter Design and Analysis Tool

Syntax

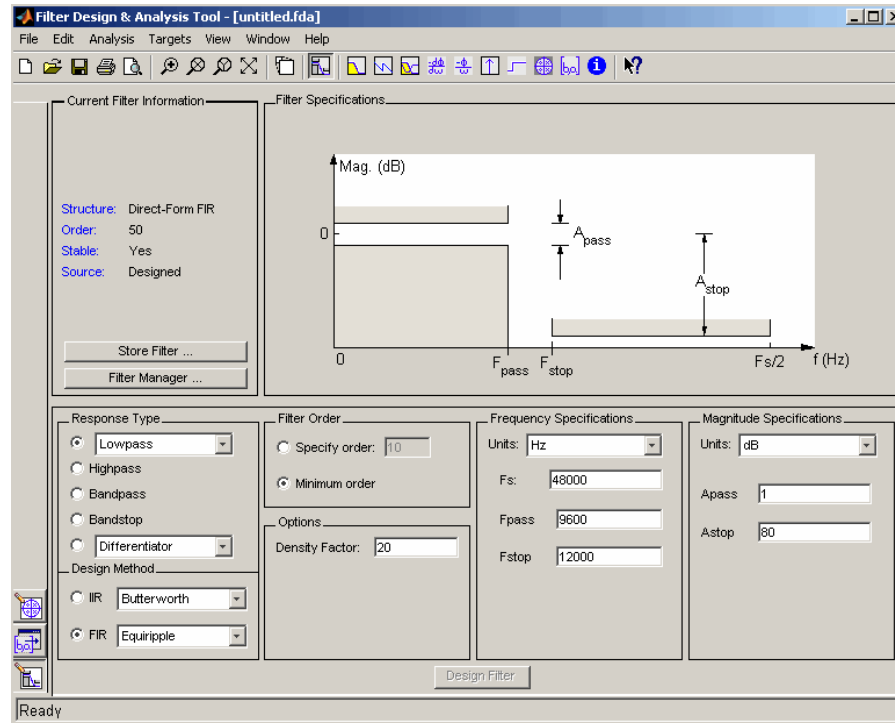
```
fdatool
```

Description

`fdatool` opens the Filter Design and Analysis Tool (FDATool). Use this tool to

- Design filters
- Analyze filters
- Modify existing filter designs

See “FDATool” and “Using FDATool” for detailed information about the Filter Design and Analysis Tool.



Tips

The Filter Design and Analysis Tool provides more design methods than the SPTool Filter Designer, which will be removed in a future release. The Filter Design and Analysis Tool also integrates advanced filter design methods from the DSP System Toolbox software.

Note The Filter Design and Analysis Tool requires a screen resolution greater than 640 x 480.

See Also

`fvtool` | `sptool` | `wvtool`

Purpose

Filter specification object

Syntax

```
d = fdesign.response
d = fdesign.response(spec)
d = fdesign.response(...,Fs)
d = fdesign.response(...,magunits)
```

Description

Filter Specification Objects

`d = fdesign.response` returns a filter specification object `d`, of filter response *`response`*. To create filters from `d`, use one of the design methods listed in "Using Filter Design Methods with Specification Objects" on page 1-304

Note Several of the filter response types described below are only available if your installation includes the DSP System Toolbox. The DSP System Toolbox significantly expands the functionality available for the specification, design, and analysis of filters.

Here is how you design filters using `fdesign`.

1. Use `fdesign.response` to construct a filter specification object.
2. Use `designmethods` to determine which filter design methods work for your new filter specification object.
3. Use `design` to apply your filter design method from step 2 to your filter specification object to construct a filter object.
4. Use FVTool to inspect and analyze your filter object.

Note `fdesign` does not create filters. `fdesign` returns a filter specification object that contains the specifications for a filter, such as the passband cutoff or attenuation in the stopband. To design a filter `hd` from a filter specification object `d`, use `d` with a filter design method such as `butter` —`hd = design(d,'butter')`.

`response` 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 Response String	Description
`arbgrpdelay`	`fdesign.arbgrpdelay` creates an object to specify allpass arbitrary group delay filters. Requires the DSP System Toolbox
`arbmag`	`fdesign.arbmag` creates an object to specify IIR filters that have arbitrary magnitude responses defined by the input arguments.
`arbmagnphase`	`fdesign.arbmagnphase` creates an object to specify IIR filters that have arbitrary magnitude and phase responses defined by the input arguments. Requires the DSP System Toolbox.
`audioweighting`	`fdesign.audioweighting` creates a filter specification object for audio weighting filters. The supported audio weighting types are: A, C, C-message, ITU-T 0.41, and ITU-R 468-4 weighting. Requires the DSP System Toolbox
`bandpass`	`fdesign.bandpass` creates an object to specify bandpass filters.
`bandstop`	`fdesign.bandstop` creates an object to specify bandstop filters.

fdesign Response String	Description
`ciccomp`	`fdesign.ciccomp` creates an object to specify filters that compensate for the CIC decimator or interpolator response curves. Requires the DSP System Toolbox.
`comb`	`fdesign.comb` creates an object to specify a notching or peaking comb filter. Requires the DSP System Toolbox.
`decimator`	`fdesign.decimator` creates an object to specify decimators. Requires the DSP System Toolbox
`differentiator`	`fdesign.differentiator` creates an object to specify an FIR differentiator filter.
`fracdelay`	`fdesign.fracdelay` creates an object to specify fractional delay filters. Requires the DSP System Toolbox.
`halfband`	`fdesign.halfband` creates an object to specify halfband filters. Requires the DSP System Toolbox.
`highpass`	`fdesign.highpass` creates an object to specify highpass filters.
`hilbert`	`fdesign.hilbert` creates an object to specify an FIR Hilbert transformer.
`interpolator`	`fdesign.interpolator` creates an object to specify interpolators. Requires the DSP System Toolbox.
`isinchp`	`fdesign.isinchp` creates an object to specify an inverse sinc highpass filter. Requires the DSP System Toolbox.

fdesign Response String	Description
`isinclp`	`fdesign.isinclp` creates an object to specify an inverse sinc lowpass filters. Requires the DSP System Toolbox.
`lowpass`	`fdesign.lowpass` creates an object to specify lowpass filters.
`notch`	`fdesign.notch` creates an object to specify notch filters. Requires the DSP System Toolbox.
`nyquist`	`fdesign.nyquist` creates an object to specify nyquist filters. Requires the DSP System Toolbox.
`octave`	`fdesign.octave` creates an object to specify octave and fractional octave filters. Requires the DSP System Toolbox.
`parameq`	`fdesign.parameq` creates an object to specify parametric equalizer filters. Requires the DSP System Toolbox.
`peak`	`fdesign.peak` creates an object to specify peak filters. Requires the DSP System Toolbox.
`polysrc`	`fdesign.polysrc` creates an object to specify polynomial sample-rate converter filters. Requires the DSP System Toolbox.
`pulseshaping`	`fdesign.pulseshaping` creates an object to specify pulse-shaping filters.
`rsrc`	`fdesign.rsrc` creates an object to specify rational-factor sample-rate convertors. Requires the DSP System Toolbox.

Use the `doc fdesign.`*`response`* 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 `response` has a property `Specification` that defines the specifications to use to design your filter. You can use defaults or specify the `Specification` property when you construct the specifications object.

With the strings for the `Specification` property, you provide filter constraints such as the filter order or the passband attenuation to use when you construct your filter from the specification object.

Properties

`fdesign` returns a filter specification object. Every filter specification object has the following properties.

Property Name	Default Value	Description
`Response`	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.
`Specification`	Depends on the chosen type	Defines the filter characteristics used to define the desired filter performance, such as the cutoff frequency `Fc` or the filter order `N`.

Property Name	Default Value	Description
`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. Audio weighting filters do not support normalized frequency.

In addition to these properties, filter specification 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 rate-change

Added Properties for mfilt Objects	Description
	factor filters. Total filter length is the product of `pl` and the rate change factors. `pl` must be an even integer.

`d = fdesign.`*`response`*`(spec)`. In `spec`, you specify the variables to use that define your filter design, such as the passband frequency or the stopband attenuation. The specifications are applied to the filter design method you choose to design your filter.

For example, when you create a default lowpass filter specification object, `fdesign.lowpass` sets the passband frequency `Fp`, the stopband frequency `Fst`, the stopband attenuation `Ast`, and the passband ripple `Ap` :

```
H = fdesign.lowpass
% Without a terminating semicolon
% the filter specifications are displayed
```

The default specification `'Fp,Fst,Ap,Ast'` is only one of the possible specifications for `fdesign.lowpass`. To see all available specifications:

```
H = fdesign.lowpass;
set(H,'specification')
```

The DSP System Toolbox software supports all available specification strings. The Signal Processing Toolbox supports a subset of the specification strings. See the reference pages for the filter specification object to determine which specification strings your installation supports.

One important note is that the specification string you choose determines which design methods apply to the filter specifications object.

Specifications that do not contain the filter order result in minimum order designs when you invoke the `design` method:

```
d = fdesign.lowpass;
% Specification is Fp,Fst,Ap,Ast
Hd = design(d,'equiripple');
length(Hd.Numerator) % returns 43
% Filter order is 42
fvtool(Hd) %view magnitude
```

`d = fdesign.`*`response`*`(...,Fs)` specifies the sampling frequency in Hz to use in the filter specifications. The sampling frequency is a scalar trailing all other input arguments. If you specify a sampling frequency, all frequency specifications are in Hz.

`d = fdesign.`*`response`*`(...,magunits)` specifies the units for any magnitude specification you provide in the input arguments. `magunits` can be one of the following strings:

- `'linear'` — specify the magnitude in linear units
- `'dB'` — specify the magnitude in decibels
- `'squared'` — specify the magnitude in power units

When you omit the `magunits` argument, `fdesign` assumes that all magnitudes are in decibels. Note that `fdesign` stores all magnitude specifications in decibels (converting to decibels when necessary) regardless of how you specify the magnitudes.

Using Filter Design Methods with Specification Objects

After you create a filter specification object, you use a filter design method to implement your filter with a selected algorithm. Use `designmethods` to determine valid design methods for your filter specification object.

```
d = fdesign.lowpass('N,Fc,Ap,Ast',10,0.2,0.5,40);
designmethods(d)
% Design FIR equiripple filter
hd = design(d,'equiripple');
```

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 specification objects or determine which design methods work for a given specifications object.

Supporting Function	Description
`setspecs`	Set all of the specifications simultaneously.
`designmethods`	Return the design methods.
`designopts`	Return the input arguments and default values that apply to a specifications object and method

You can set filter specification values by passing them after the `Specification` argument, or by passing the values without the `Specification` string.

Filter object constructors take the input arguments in the same order as `setspecs` and the order in the strings for `Specification`. Enter `doc setspecs` at the prompt for more information about using `setspecs`.

When the first input to `fdesign` is not a valid `Specification` string like '`n,fc`', `fdesign` assumes that the input argument is a filter specification and applies it using the default `Specification` string —`fp,fst,ap,ast` for a lowpass object, for example.

Examples

The following examples require only the Signal Processing Toolbox.

Example 1–Bandstop Filter

A bandstop filter specification object for data sampled at 8 kHz. The stopband between 2 and 2.4 kHz is attenuated at least 80 dB:

```
H = fdesign.bandstop('Fp1,Fst1,Fst2,Fp2,Ap1,Ast,Ap2',...
           1600,2000,2400,2800,1,80,1,8000);
```

Example 2–Lowpass Filter

A lowpass filter specification object for data sampled at 10 kHz. The passband frequency is 500 Hz and the stopband frequency is 750 Hz.

The passband ripple is set to 1 dB and the required attenuation in the stopband is 80 dB.

```
H = fdesign.lowpass('Fp,Fst,Ap,Ast',500,750,1,80,10000);
```

Example 3–Highpass Filter

A default highpass filter specification object.

```
H = fdesign.highpass % Creates specifications object.
H.Description
```

Notice the correspondence between the property values in `Specification` and `Description` — in `Description` you see in words the definitions of the variables shown in `Specification`.

Example 4–Filter Specification and Design

Lowpass Butterworth filter specification object

Use a filter specification object to construct a lowpass Butterworth filter with default `Specification` `'Fp,Fst,Ap,Ast'`. Set the passband edge frequency to 0.4π radians/sample, a stopband frequency of 0.5π radians/sample, a passband ripple of 1 dB, and 80 dB of stopband attenuation.

```
d = fdesign.lowpass(0.4,0.5,1,80);
```

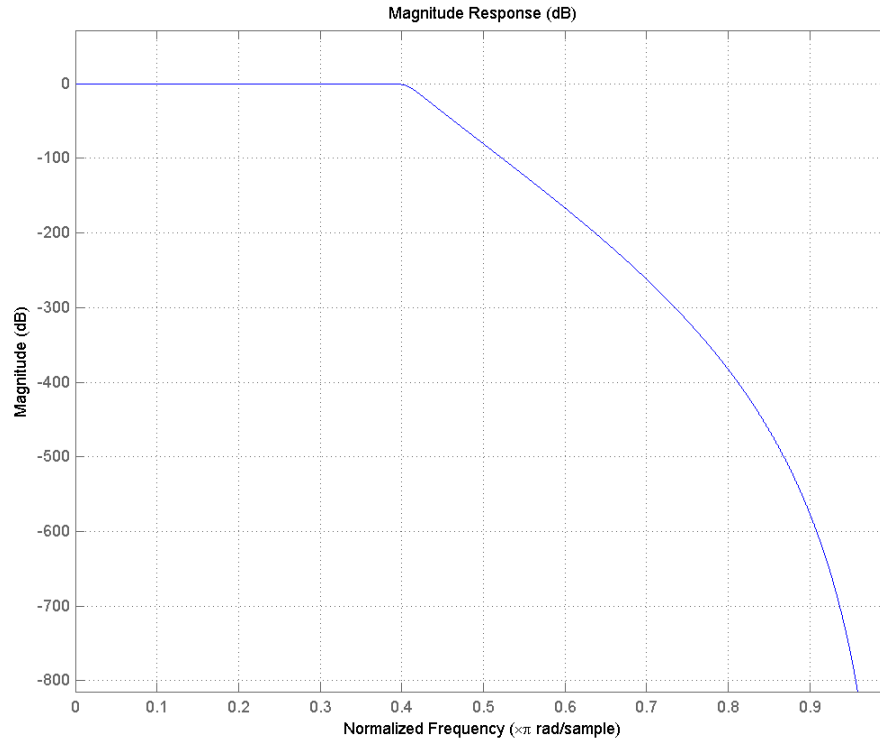
Determine which design methods apply to `d`.

```
designmethods(d)
```

You can use `d` and the `butter` design method to design a Butterworth filter.

```
hd = design(d,'butter','matchexactly','passband');
fvtool(hd);
```

The resulting filter magnitude response shown by FVTool appears in the following figure.



If you have the DSP System Toolbox software installed, the preceding figure appears with the filter specification mask.

See Also

`designmethods` | `designopts` | `fdatool` | `filterbuilder` | `fvtool`

fdesign.arbmag

Purpose

Arbitrary response magnitude filter specification object

Syntax

```
D= fdesign.arbmag
D= fdesign.arbmag(SPEC)
D = fdesign.arbmag(SPEC,specvalue1,specvalue2,...)
D = fdesign.arbmag(specvalue1,specvalue2,specvalue3)
D = fdesign.arbmag(...,Fs)
```

Description

`D= fdesign.arbmag` constructs an arbitrary magnitude filter specification object `D`.

`D= fdesign.arbmag(SPEC)` initializes the `Specification` property to `SPEC`. The input argument `SPEC` must be one of the strings shown in the following table. Specification strings are not case sensitive.

Note Specifications strings marked with an asterisk require the DSP System Toolbox software.

- `'N,F,A'` — Single band design (default)
- `'F,A,R'` — Single band minimum order design *
- `'N,B,F,A'` — Multiband design
- `'N,B,F,A,C'` — Constrained multiband design *
- `'B,F,A,R'` — Multiband minimum order design *
- `'Nb,Na,F,A'` — Single band design *
- `'Nb,Na,B,F,A'` — Multiband design *

The string entries are defined as follows:

- `A` — Amplitude vector. Values in `A` define the filter amplitude at frequency points you specify in `f`, the frequency vector. If you use `A`, you must use `F` as well. Amplitude values must be real. For complex values designs, use `fdesign.arbmagnphase`.
- `B` — Number of bands in the multiband filter

- `C` — Constrained band flag. This enables you to constrain the passband ripple in your multiband design. You cannot constrain the passband ripple in all bands simultaneously.
- `F` — Frequency vector. Frequency values in specified in `F` indicate locations where you provide specific filter response amplitudes. When you provide `F`, you must also provide `A`.
- `N` — Filter order for FIR filters and the numerator and denominator orders for IIR filters.
- `Nb` — Numerator order for IIR filters
- `Na` — Denominator order for IIR filter designs
- `R` — Ripple

By default, this method assumes that all frequency specifications are supplied in normalized frequency.

Specifying Frequency and Amplitude Vectors

`F` and `A` are the input arguments you use to define the filter response desired. Each frequency value you specify in `F` must have a corresponding response value in `A`. The following table shows how `F` and `A` are related.

Define the frequency vector `F` as `[0 0.25 0.3 0.4 0.5 0.6 0.7 0.75 1.0]`

Define the response vector `A` as `[1 1 0 0 0 0 0 1 1]`

These specifications connect `F` and `A` as shown here:

F (Normalized Frequency)	A (Response Desired at F)
0	1
0.25	1
0.3	0
0.4	0

F (Normalized Frequency)	A (Response Desired at F)
0.5	0
0.6	0
0.7	0
0.75	1
1.0	1

Different specifications can have different design methods available. Use `designmethods` to get a list of design methods available for a given specification string and filter specification object.

Use `designopts` to get a list of design options available for a filter specification object and a given design method. Enter `help(D,METHOD)` to get detailed help on the available design options for a given design method.

`D = fdesign.arbmag(SPEC,specvalue1,specvalue2,...)` initializes the specifications with `specvalue1`, `specvalue2`. Use `get(D,'Description')` for descriptions of the various specifications `specvalue1`, `specvalue2`, ... `specvalueN`.

`D = fdesign.arbmag(specvalue1,specvalue2,specvalue3)` uses the default specification string `'N,F,A'`, setting the filter order, filter frequency vector, and the amplitude vector to the values `specvalue1`, `specvalue2`, and `specvalue3`.

`D = fdesign.arbmag(...,Fs)` specifies the sampling frequency in Hz. All other frequency specifications are also assumed to be in Hz when you specify `Fs`.

Examples

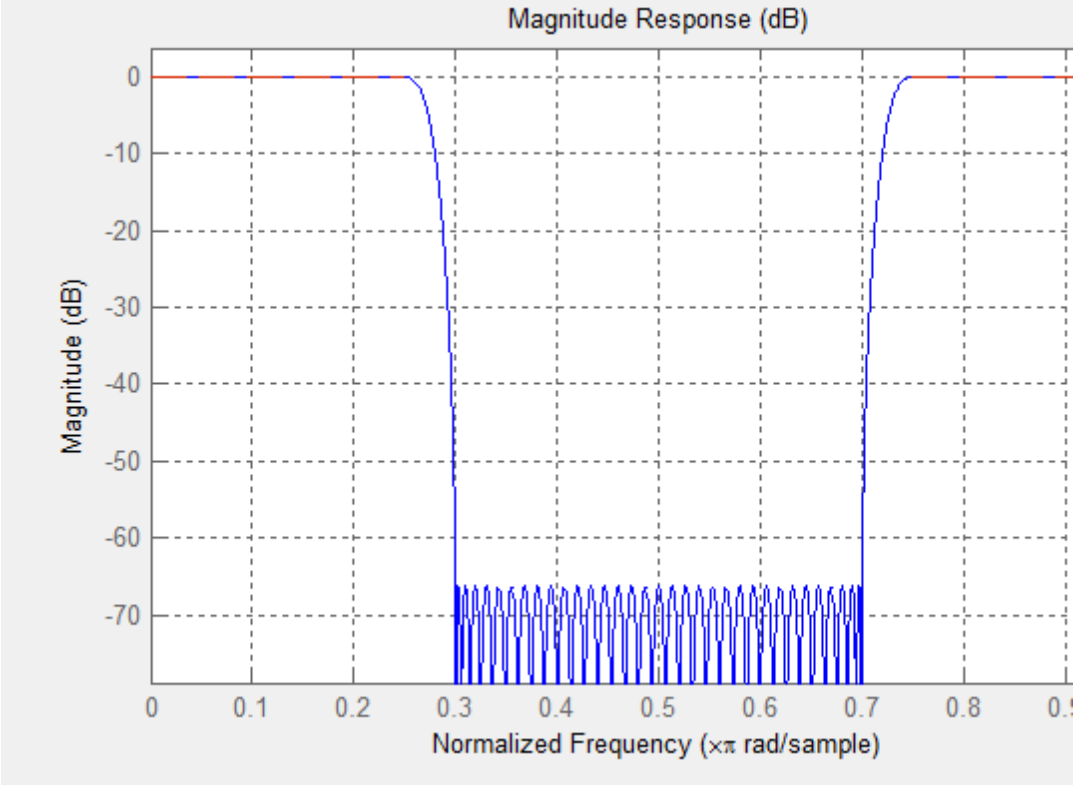
Design of a multiband arbitrary-magnitude filter

Use fdesign.arbmag to design a 3–band filter.

Use the given frequency and amplitude vectors in “Specifying Frequency and Amplitude Vectors” on page 1-309.

```
N = 150;
B = 3;
F = [0 .25 .3 .4 .5 .6 .7 .75 1];
A = [1 1 0 0 0 0 0 1 1];
A1 = A(1:2);
A2 = A(3:7);
A3 = A(8:end);
F1 = F(1:2);
F2 = F(3:7);
F3 = F(8:end);
d = fdesign.arbmag('N,B,F,A',N,B,F1,A1,F2,A2,F3,A3);
Hd = design(d);
fvtool(Hd)
```

A response with two passbands — one roughly between 0 and 0.25 and the second between 0.75 and 1 — results from the mapping between `F` and `A`.



Design of a single band arbitrary-magnitude filter

Use fdesign.arbmag to design a single band equiripple filter.

```
n = 120;
f = linspace(0,1,100);  % 100 frequency points.
as = ones(1,100)-f*0.2;
absorb = [ones(1,30),(1-0.6*bohmanwin(10))',...
ones(1,5), (1-0.5*bohmanwin(8))',ones(1,47)];
a = as.*absorb;
d = fdesign.arbmag('N,F,A',n,f,a);
hd1 = design(d,'equiripple');
```

If you have the DSP System Toolbox, you can design a minimum-phase equiripple filter.

```
hd2 = design(d,'equiripple','MinPhase',true);
hfvt = fvtool([hd1 hd2],'analysis','polezero');
legend(hfvt,'Equiripple Filter','Minimum-phase Equiripple Filter');
```

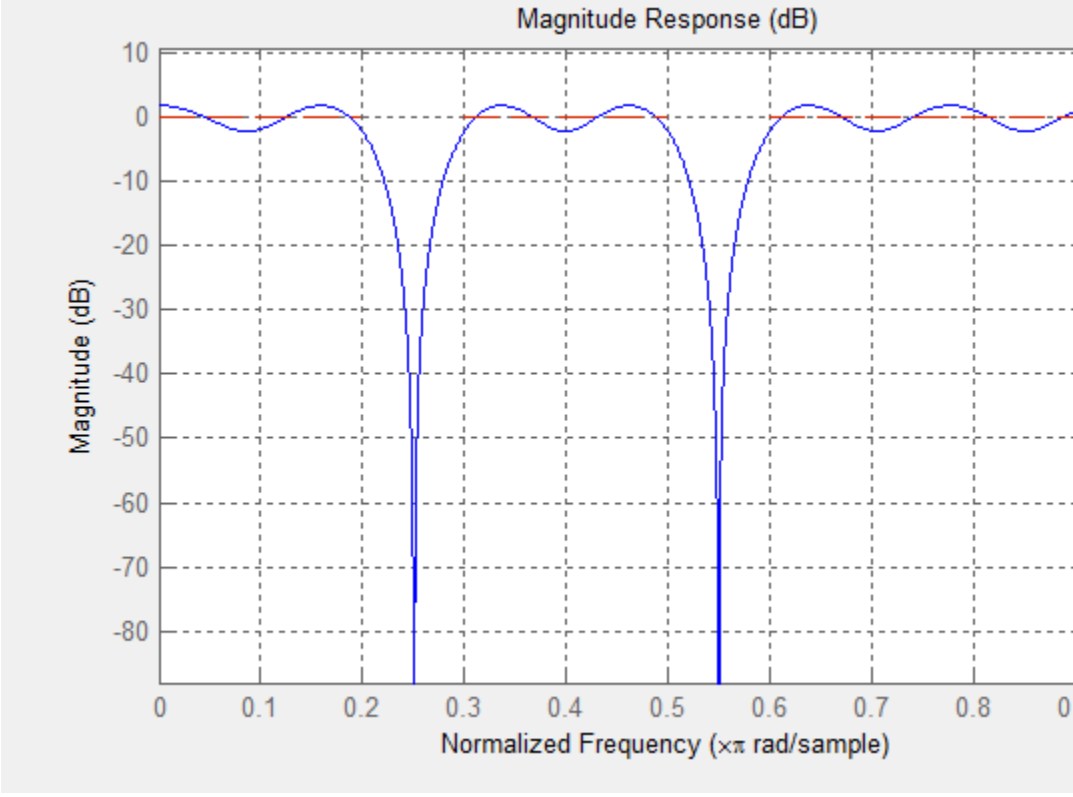
Design of a multiband minimum order arbitrary-magnitude filter

Use fdesign.arbmag to design a multiband minimum order filter.

This example requires the DSP System Toolbox.

Place the notches at 0.25π and 0.55π radians/sample

```
d = fdesign.arbmag('B,F,A,R');
d.NBands = 5;
d.B1Frequencies = [0 0.2];
d.B1Amplitudes = [1 1];
d.B1Ripple = 0.25;
d.B2Frequencies = 0.25;
d.B2Amplitudes = 0;
d.B3Frequencies = [0.3 0.5];
d.B3Amplitudes = [1 1];
d.B3Ripple = 0.25;
d.B4Frequencies = 0.55;
d.B4Amplitudes = 0;
d.B5Frequencies = [0.6 1];
d.B5Amplitudes = [1 1];
d.B5Ripple = 0.25;
Hd = design(d,'equiripple');
fvtool(Hd)
```



Design of a multiband constrained arbitrary-magnitude filter

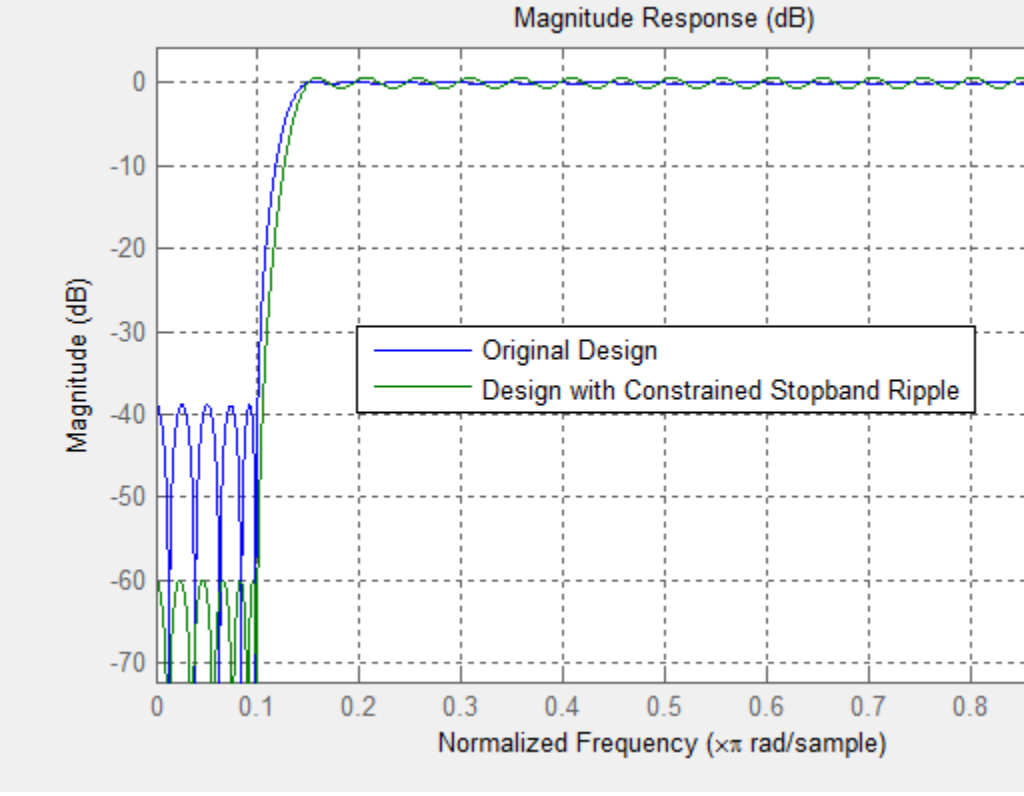
Use fdesign.arbmag to design a multiband constrained FIR filter.

This example requires the DSP System Toolbox.

Force the frequency response at 0.15π radians/sample to 0 dB.

```
d = fdesign.arbmag('N,B,F,A,C',82,2);
d.B1Frequencies = [0 0.06 .1];
d.B1Amplitudes = [0 0 0];
d.B2Frequencies = [.15 1];
d.B2Amplitudes = [1 1];
```

```
% Design a filter with no constraints
Hd1 = design(d,'equiripple','B2ForcedFrequencyPoints',0.15);
% Add a constraint to the first band to increase attenuation
d.B1Constrained = true;
d.B1Ripple = .001;
Hd2 =  design(d,'equiripple','B2ForcedFrequencyPoints',0.15);
hfvt = fvtool(Hd1,Hd2);
legend(hfvt,'Original Design','Design with Constrained Stopband Ripple
```



See Also

`design` | `designmethods` | `fdesign`

fdesign.bandpass

Purpose

Bandpass filter specification object

Syntax

```
D = fdesign.bandpass
D = fdesign.bandpass(SPEC)
D = fdesign.bandpass(spec,specvalue1,specvalue2,...)
D = fdesign.bandpass(specvalue1,specvalue2,specvalue3,
specvalue4,...specvalue4,specvalue5,specvalue6)
D = fdesign.bandpass(...,Fs)
D = fdesign.bandpass(...,MAGUNITS)
```

Description

`D = fdesign.bandpass` constructs a bandpass filter specification 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.

`D = fdesign.bandpass(SPEC)` constructs object `D` and sets its `Specification` property to `SPEC`. Entries in the `SPEC` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for `SPEC` are shown below and used to define the bandpass filter. The strings are not case sensitive.

Note Specifications strings marked with an asterisk require the DSP System Toolbox software.

- `'Fst1,Fp1,Fp2,Fst2,Ast1,Ap,Ast2'` (default `spec`)
- `'N,F3dB1,F3dB2'`
- `"N,F3dB1,F3dB2,Ap'` *
- `'N,F3dB1,F3dB2,Ast'` *
- `'N,F3dB1,F3dB2,Ast1,Ap,Ast2'` *
- `'N,F3dB1,F3dB2,BWp` *
- `'N,F3dB1,F3dB2,BWst'` *
- `'N,Fc1,Fc2'`

- `'N,Fc1,Fc2,Ast1,Ap,Ast2'`
- `'N,Fp1,Fp2,Ap'`
- `'N,Fp1,Fp2,Ast1,Ap,Ast2'`
- `'N,Fst1,Fp1,Fp2,Fst2'`
- `'N,Fst1,Fp1,Fp2,Fst2,C'` *
- `'N,Fst1,Fp1,Fp2,Fst2,Ap'` *
- `'N,Fst1,Fst2,Ast'`
- `'Nb,Na,Fst1,Fp1,Fp2,Fst2'` *

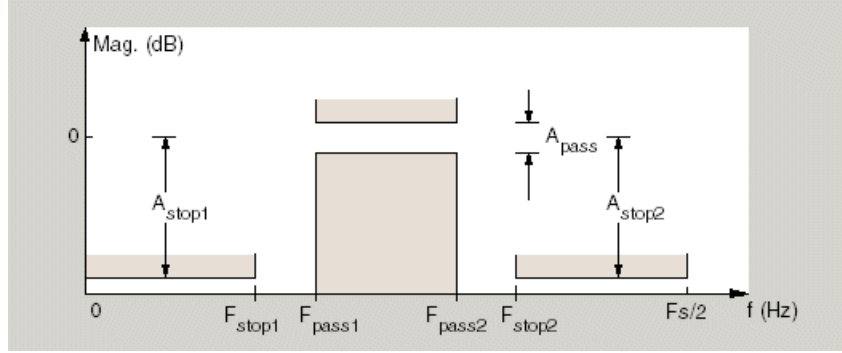
The string entries are defined as follows:

- `Ap` — amount of ripple allowed in the pass band. Also called Apass.
- `Ast1` — attenuation in the first stop band in decibels (the default units). Also called Astop1.
- `Ast2` — attenuation in the second stop band in decibels (the default units). Also called Astop2.
- `BWp` — bandwidth of the filter passband. Specified in normalized frequency units.
- `BWst` — bandwidth of the filter stopband. Specified in normalized frequency units.
- `C` — Constrained band flag. This enables you to specify passband ripple or stopband attenuation for fixed-order designs in one or two of the three bands.

  In the specification string `'N,Fst1,Fp1,Fp2,Fst2,C'`, you cannot specify constraints in both stopbands and the passband simultaneously. You can specify constraints in any one or two bands.
- `F3dB1` — cutoff frequency for the point 3 dB point below the passband value for the first cutoff. Specified in normalized frequency units. (IIR filters)

- `F3dB2` — cutoff frequency for the point 3 dB point below the passband value for the second cutoff. Specified in normalized frequency units. (IIR filters)
- `Fc1` — cutoff frequency for the point 6 dB point below the passband value for the first cutoff. Specified in normalized frequency units. (FIR filters)
- `Fc2` — cutoff frequency for the point 6 dB point below the passband value for the second cutoff. Specified in normalized frequency units. (FIR filters)
- `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 for FIR filters. Or both the numerator and denominator orders for IIR filters when na and nb are not provided.
- `Na` — denominator order for IIR filters
- `Nb` — numerator order for IIR filters

Graphically, the filter specifications look similar to those shown in the following figure.



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 specification object change depending on the `Specification` string. Use `designmethods` to determine which design methods apply to an object and the `Specification` property value.

Use `designopts` to determine the design options for a given design method. Enter `help(D,METHOD)` at the MATLAB command line to obtain detailed help on the design options for a given design method, `METHOD`.

`D = fdesign.bandpass(spec,specvalue1,specvalue2,...)` constructs an object `D` and sets its specifications at construction time.

`D = fdesign.bandpass(specvalue1,specvalue2,specvalue3, specvalue4,...specvalue4,specvalue5,specvalue6)` constructs `D`with the default `Specification` property string, using the values you provide as input arguments for `specvalue1,specvalue2,specvalue3,specvalue4,specvalue4,specvalue5, specvalue6` and `specvalue7`.

`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 decibels. Note that `fdesign` stores all magnitude specifications in decibels (converting to decibels when necessary) regardless of how you specify the magnitudes.

Examples

Filter a discrete-time signal with a bandpass filter. The signal is a sum of three discrete-time sinusoids, π/8, π/2, and 3π/4 radians/sample.

```
n = 0:159;
x = cos(pi/8*n)+cos(pi/2*n)+sin(3*pi/4*n);
```

Design an FIR equiripple bandpass filter to remove the lowest and highest discrete-time sinusoids.

```
d = fdesign.bandpass('Fst1,Fp1,Fp2,Fst2,Ast1,Ap,Ast2',1/4,3/8,5/8,6/8,60,
Hd = design(d,'equiripple');
```

Apply the filter to the discrete-time signal.

```
y = filter(Hd,x);
freq = 0:(2*pi)/length(x):pi;
xdft = fft(x);
ydft = fft(y);
plot(freq,abs(xdft(1:length(x)/2+1)));
hold on;
plot(freq,abs(ydft(1:length(x)/2+1)),'r','linewidth',2);
legend('Original Signal','Bandpass Signal');
```

Design an IIR Butterworth filter of order 10 with 3–dB frequencies of 1 and 1.2 kHz. The sampling frequency is 10 kHz

```
d = fdesign.bandpass('N,F3dB1,F3dB2',10,1e3,1.2e3,1e4);
Hd = design(d,'butter');
fvtool(Hd)
```

This example requires the DSP System Toolbox software.

Design a constrained-band FIR equiripple filter of order 100 with a passband of [1, 1.4] kHz. Both stopband attenuation values are constrained to 60 dB. The sampling frequency is 10 kHz.

```
d = fdesign.bandpass('N,Fst1,Fp1,Fp2,Fst2,C',100,800,1e3,1.4e3,1.6e3,1
d.Stopband1Constrained = true; d.Astop1 = 60;
d.Stopband2Constrained = true; d.Astop2 = 60;
Hd = design(d,'equiripple');
fvtool(Hd);
measure(Hd)
```

The passband ripple is slightly over 2 dB. Because the design constrains both stopbands, you cannot constrain the passband ripple.

See Also

`fdesign`, `fdesign.bandstop`, `fdesign.highpass`, `fdesign.lowpass`

fdesign.bandstop

Purpose

Bandstop filter specification object

Syntax

```
D = fdesign.bandstop
D = fdesign.bandstop(SPEC)
D = fdesign.bandstop(SPEC,specvalue1,specvalue2,...)
D = fdesign.bandstop(specvalue1,specvalue2,specvalue3,specvalue4,...
specvalue5,specvalue6,specvalue7)
D = fdesign.bandstop(...,Fs)
D = fdesign.bandstop(...,MAGUNITS)
```

Description

`D = fdesign.bandstop` constructs a bandstop filter specification object `D`, applying default values for the properties `Fpass1`, `Fstop1`, `Fstop2`, `Fpass2`, `Apass1`, `Astop1` and `Apass2`.

`D = fdesign.bandstop(SPEC)` constructs object `D` and sets the `Specification` property to `SPEC`. Entries in the `SPEC` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for `SPEC` are shown below. The strings are not case sensitive.

Note Specifications strings marked with an asterisk require the DSP System Toolbox software.

- `'Fp1,Fst1,Fst2,Fp2,Ap1,Ast,Ap2'` (default spec)
- `'N,F3dB1,F3dB2'`
- `'N,F3dB1,F3dB2,Ap'` *
- `'N,F3dB1,F3dB2,Ap,Ast'` *
- `'N,F3dB1,F3dB2,Ast'` *
- `'N,F3dB1,F3dB2,BWp'` *
- `'N,F3dB1,F3dB2,BWst'` *
- `'N,Fc1,Fc2'`

- `'N,Fc1,Fc2,Ap1,Ast,Ap2'`
- `'N,Fp1,Fp2,Ap'`
- `'N,Fp1,Fp2,Ap,Ast'`
- `'N,Fp1,Fst1,Fst2,Fp2'`
- `'N,Fp1,Fst1,Fst2,Fp2,C'` *
- `'N,Fp1,Fst1,Fst2,Fp2,Ap'` *
- `'N,Fst1,Fst2,Ast'`
- `'Nb,Na,Fp1,Fst1,Fst2,Fp2'` *

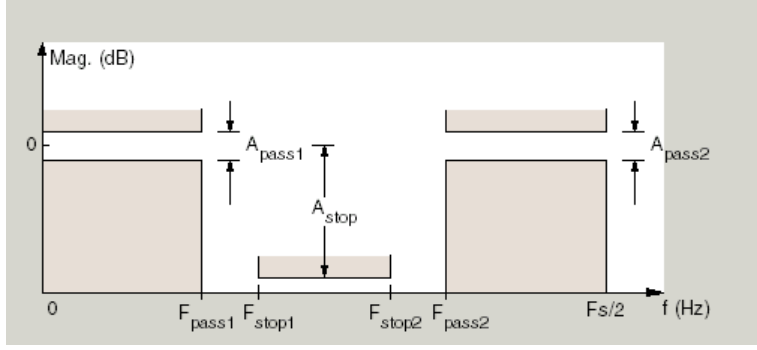
The string entries are defined as follows:

- `Ap` — amount of ripple allowed in the passband in decibels (the default units). Also called Apass.
- `Ap1` — amount of ripple allowed in the pass band in decibels (the default units). Also called Apass1.
- `Ap2` — amount of ripple allowed in the pass band in decibels (the default units). Also called Apass2.
- `Ast` — attenuation in the first stopband in decibels (the default units). Also called Astop1.
- `BWp` — bandwidth of the filter passband. Specified in normalized frequency units.
- `BWst` — bandwidth of the filter stopband. Specified in normalized frequency units.
- `C` — Constrained band flag. This enables you to specify passband ripple or stopband attenuation for fixed-order designs in one or two of the three bands.

  In the specification string `'N,Fp1,Fst1,Fst2,Fp2,C'`, you cannot specify constraints simultaneously in both passbands and the stopband. You can specify constraints in any one or two bands.

- `F3dB1` — cutoff frequency for the point 3 dB point below the passband value for the first cutoff.
- `F3dB2` — cutoff frequency for the point 3 dB point below the passband value for the second cutoff.
- `Fc1` — cutoff frequency for the point 6 dB point below the passband value for the first cutoff. (FIR filters)
- `Fc2` — cutoff frequency for the point 6 dB point below the passband value for the second cutoff. (FIR filters)
- `Fp1` — frequency at the start of the pass band. Also called Fpass1.
- `Fp2` — frequency at the end of the pass band. Also called Fpass2.
- `Fst1` — frequency at the end of the first stop band. Also called Fstop1.
- `Fst2` — frequency at the start of the second stop band. Also called Fstop2.
- `N` — filter order.
- `Na` — denominator order for IIR filters.
- `Nb` — numerator order for IIR filters.

Graphically, the filter specifications look similar to those shown in the following figure.



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 specification object change depending on the `Specification` string. Use `designmethods` to determine which design methods apply to an object and the `Specification` property value.

Use `designopts` to determine the design options for a given design method. Enter `help(D,METHOD)` at the MATLAB command line to obtain detailed help on the design options for a given design method, `METHOD`.

`D = fdesign.bandstop(SPEC,specvalue1,specvalue2,...)` constructs an object `D` and sets its specifications at construction time.

`D = fdesign.bandstop(specvalue1,specvalue2,specvalue3,specvalue4,... specvalue5,specvalue6,specvalue7)` constructs an object `D` with the default `Specification` property string , using the values you provide in `specvalue1,specvalue2,specvalue3,specvalue4,specvalue5, specvalue6` and `specvalue7`.

`D = fdesign.bandstop(...,Fs)` adds the argument `Fs`, specified in Hz to define the sampling frequency. If you specify the sampling frequency as a trailing scalar, 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 decibels. Note that `fdesign` stores all magnitude specifications in decibels (converting to decibels when necessary) regardless of how you specify the magnitudes.

Examples

Construct a bandstop filter to reject the discrete frequency band between 3π/8 and 5π/8 radians/sample. Apply the filter to a discrete-time signal consisting of the superposition of three discrete-time sinusoids.

Design an FIR equiripple filter and view the magnitude response.

```
d = fdesign.bandstop('Fp1,Fst1,Fst2,Fp2,Ap1,Ast,Ap2',2/8,3/8,5/8,6/8,1,60
Hd = design(d,'equiripple');
fvtool(Hd)
```

Construct the discrete-time signal to filter.

```
n = 0:99;
x = cos(pi/5*n)+sin(pi/2*n)+cos(4*pi/5*n);
y = filter(Hd,x);
xdft = fft(x);
ydft = fft(y);
freq = 0:(2*pi)/length(x):pi;
plot(freq,abs(xdft(1:length(x)/2+1)));
hold on;
plot(freq,abs(ydft(1:length(y)/2+1)),'r','linewidth',2);
xlabel('Radians/Sample'); ylabel('Magnitude');
legend('Original Signal','Bandstop Signal');
```

Create a Butterworth bandstop filter for data sampled at 10 kHz. The stopband is [1,1.5] kHz. The order of the filter is 20.

```
d = fdesign.bandstop('N,F3dB1,F3dB2',20,1e3,1.5e3,1e4);
Hd = design(d,'butter');
fvtool(Hd);
```

Zoom in on the magnitude response plot to verify that the 3-dB down points are located at 1 and 1.5 kHz.

The following example requires the DSP System Toolbox license.

Design a constrained-band FIR equiripple filter of order 100 for data sampled at 10 kHz. You can specify constraints on at most two of the three bands: two passbands and one stopband. In this example, you choose to constrain the passband ripple to be 0.5 dB in each passband. Design the filter, visualize the magnitude response and measure the filter's design.

```
d = fdesign.bandstop('N,Fp1,Fst1,Fst2,Fp2,C',100,800,1e3,1.5e3,1.7e3,1
d.Passband1Constrained = true; d.Apass1 = 0.5;
d.Passband2Constrained = true; d.Apass2 = 0.5;
Hd = design(d,'equiripple');
fvtool(Hd);
measure(Hd)
```

With this order filter and passband ripple constraints, you achieve approximately 50 dB of stopband attentuation.

See Also

`fdesign`, `fdesign.bandpass`, `fdesign.highpass`, `fdesign.lowpass`

fdesign.differentiator

Purpose

Differentiator filter specification object

Syntax

```
D = fdesign.differentiator
D = fdesign.differentiator(SPEC)
D = fdesign.differentiator(SPEC,specvalue1,specvalue2, ...)
D = fdesign.differentiator(specvalue1)
D = fdesign.differentiator(...,Fs)
D = fdesign.differentiator(...,MAGUNITS)
```

Description

`D = fdesign.differentiator` constructs a default differentiator filter designer `D` with the filter order set to 31.

`D = fdesign.differentiator(SPEC)` initializes the filter designer `Specification` property to `SPEC`. You provide one of the following strings as input to replace `SPEC`. The string you provide is not case sensitive.

Note Specifications strings marked with an asterisk require the DSP System Toolbox software.

- `'N'` — Full band differentiator (default)
- `'N,Fp,Fst'` — Partial band differentiator
- `'N,Fp,Fst,Ap'` — Partial band differentiator *
- `'N,Fp,Fst,Ast'` — Partial band differentiator *
- `'Ap'` — Minimum order full band differentiator *
- `'Fp,Fst,Ap,Ast'` — Minimum order partial band differentiator *

The string entries are defined as follows:

- `Ap` — amount of ripple allowed in the pass band in decibels (the default units). Also called Apass.
- `Ast` — attenuation in the stop band in decibels (the default units). Also called Astop.

- `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.

By default, `fdesign.differentiator` assumes that all frequency specifications are provided in normalized frequency units. Also, decibels is the default for all magnitude specifications.

Use `designopts` to determine the design options for a given design method. Enter `help(D,METHOD)` at the MATLAB command line to obtain detailed help on the design options for a given design method, `METHOD`.

`D = fdesign.differentiator(SPEC,specvalue1,specvalue2, ...)` initializes the filter designer specifications in `SPEC` with `specvalue1`, `specvalue2`, and so on. To get a description of the specifications `specvalue1`, `specvalue2`, and more, enter

```
get(d,'description')
```

at the Command prompt.

`D = fdesign.differentiator(specvalue1)` assumes the default specification string `N`, setting the filter order to the value you provide.

`D = fdesign.differentiator(...,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.differentiator(...,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 decibels. Note that `fdesign` stores all magnitude specifications in decibels (converting to decibels when necessary) regardless of how you specify the magnitudes.

Examples

Use an FIR equiripple differentiator to transform frequency modulation into amplitude modulation, which can be detected using an envelope detector.

Modulate a message signal consisting of a 20-Hz sine wave with a 1 kHz carrier frequency. The sampling frequency is 10 kHz .

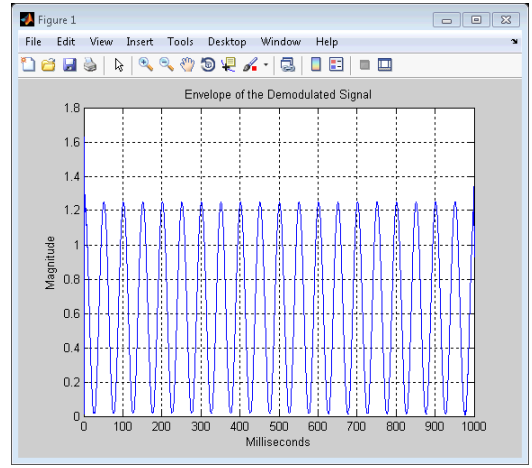
```
t = linspace(0,1,1e4);
x = cos(2*pi*20*t);
Fc = 1e3;
Fs = 1e4;
y = modulate(x,Fc,Fs,'fm');
```

Design the equiripple FIR differentiator of order 31.

```
d = fdesign.differentiator(31,1e4);
Hd = design(d,'equiripple');
```

Filter the modulated signal and take the Hilbert transform to obtain the envelope.

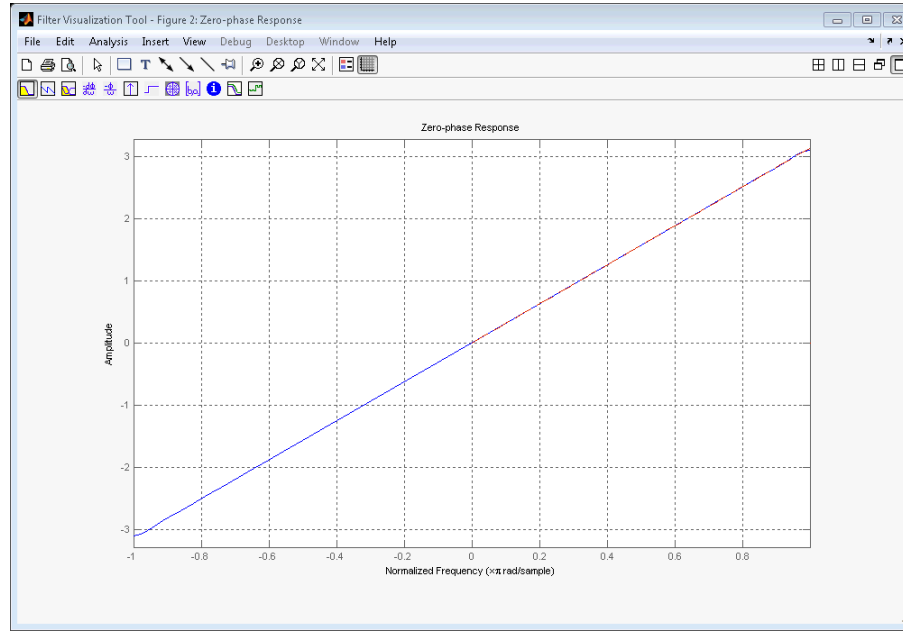
```
y1 = filter(Hd,y);
y1 = hilbert(y1);
% Plot the envelope
plot(t.*1000,abs(y1));
xlabel('Milliseconds'); ylabel('Magnitude');
grid on;
title('Envelope of the Demodulated Signal');
```



From the preceding figure, you see that the envelope completes two cycles every 100 milliseconds. The envelope is oscillating at 20 Hz, which corresponds to the frequency of the message signal.

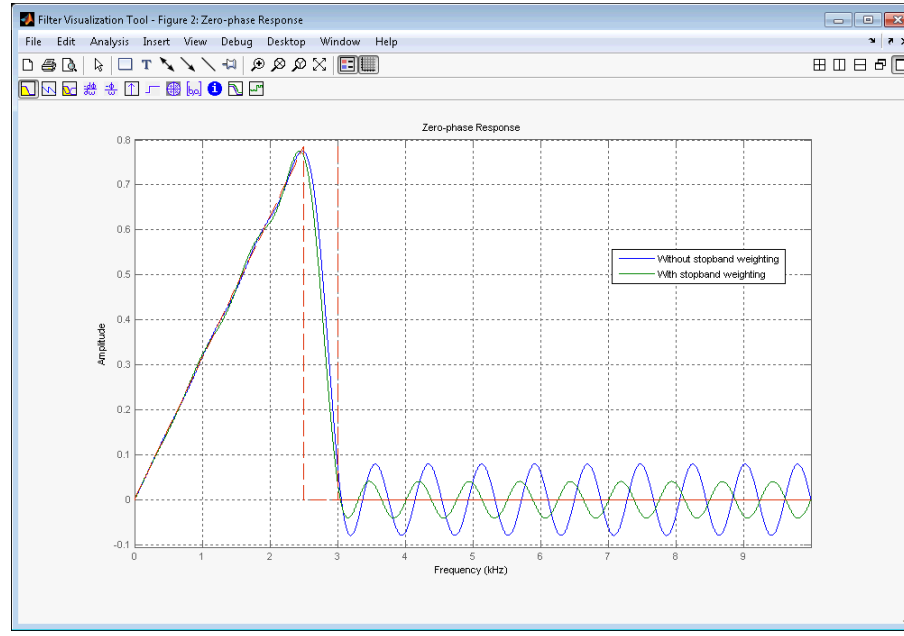
Design an FIR differentiator using least squares and plot the zero phase response.

```
d = fdesign.differentiator(33); % Filter order is 33.
hd = design(d,'firls');
fvtool(hd,'magnitudedisplay','zero-phase',...
'frequencyrange','[-pi, pi)')
```



Design a narrow band differentiator. Differentiate the first 25 percent of the frequencies in the Nyquist range and filter the higher frequencies.

```
Fs=20000; %sampling frequency
d = fdesign.differentiator('N,Fp,Fst',54,2500,3000,Fs);
Hd= design(d,'equiripple');
% Weight the stopband to increase attenuation
Hd1 = design(d,'equiripple','Wstop',4);
hfvt = fvtool(Hd,Hd1,'magnitudedisplay','zero-phase',...
'frequencyrange','[0, Fs/2)');
legend(hfvt,'Without stopband weighting',...
'With stopband weighting');
```



See Also

`design` | `fdesign`

fdesign.highpass

Purpose

Highpass filter specification object

Syntax

```
D = fdesign.highpass
D = fdesign.highpass(SPEC)
D = fdesign.highpass(SPEC,specvalue1,specvalue2,...)
D = fdesign.highpass(specvalue1,specvalue2,specvalue3,
specvalue4)
D = fdesign.highpass(...,Fs)
D = fdesign.highpass(...,MAGUNITS)
```

Description

`D = fdesign.highpass` constructs a highpass filter specification object `D`, applying default values for the specification string, `'Fst,Fp,Ast,Ap'`.

`D = fdesign.highpass(SPEC)` constructs object `D` and sets the `Specification` property to `SPEC`. Entries in the `SPEC` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for `SPEC` are shown below. The strings are not case sensitive.

Note Specifications strings marked with an asterisk require the DSP System Toolbox software.

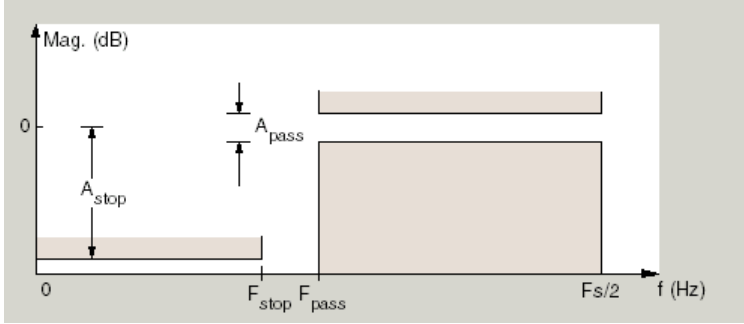
- `'Fst,Fp,Ast,Ap'` (default `spec`)
- `'N,F3db'`
- `'N,F3db,Ap'` *
- `'N,F3db,Ast'` *
- `'N,F3db,Ast,Ap'` *
- `'N,F3db,Fp` *
- `'N,Fc'`
- `'N,Fc,Ast,Ap'`
- `'N,Fp,Ap'`

- `'N,Fp,Ast,Ap'`
- `'N,Fst,Ast'`
- `'N,Fst,Ast,Ap'`
- `'N,Fst,F3db'` *
- `'N,Fst,Fp'`
- `'N,Fst,Fp,Ap'` *
- `'N,Fst,Fp,Ast'` *
- `'Nb,Na,Fst,Fp'` *

The string entries are defined as follows:

- `Ap` — amount of ripple allowed in the pass band in decibels (the default units). Also called Apass.
- `Ast` — attenuation in the stop band in decibels (the default units). Also called Astop.
- `F3db` — cutoff frequency for the point 3 dB point below the passband value. Specified in normalized frequency units.
- `Fc` — cutoff frequency for the point 6 dB 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.
- `Na` and `Nb` are the order of the denominator and numerator.

Graphically, the filter specifications look similar to those shown in the following figure.



Regions between specification values like `Fst` and `Fp` are transition regions where the filter response is not explicitly defined.

The filter design methods that apply to a highpass filter specification object change depending on the `Specification` string. Use `designmethods` to determine which design method applies to an object and its specification string.

Use `designopts` to determine which design options are valid for a given design method. For detailed information on design options for a given design method, `METHOD`, enter `help(D,METHOD)` at the MATLAB command line.

`D = fdesign.highpass(SPEC,specvalue1,specvalue2,...)` constructs an object `d` and sets its specification values at construction time.

`D = fdesign.highpass(specvalue1,specvalue2,specvalue3, specvalue4)` constructs an object `D` with the default `Specification` property and the values you enter for `specvalue1,specvalue2,...`.

`D = fdesign.highpass(...,Fs)` provides the sampling frequency for the filter specification object. `Fs` is in Hz and must be specified as a scalar trailing the other numerical values provided. If you specify a sampling frequency, all other frequency specifications are in Hz.

`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 decibels. Note that `fdesign` stores all magnitude specifications in decibels (converting to decibels when necessary) regardless of how you specify the magnitudes.

Examples

Higpass filter a discrete-time signal consisting of two sine waves.

Create a highpass filter specification object. Specify the passband frequency to be 0.25π radians/sample and the stopband frequency to be 0.15π radians/sample. Specify 1 dB of allowable passband ripple and a stopband attenuation of 60 dB.

```
d = fdesign.highpass('Fst,Fp,Ast,Ap',0.15,0.25,60,1);
```

Query the valid design methods for your filter specification object, `d`.

```
designmethods(d)
```

Create an FIR equiripple filter and view the filter magnitude response with `fvtool`.

```
Hd = design(d,'equiripple');
fvtool(Hd);
```

Create a signal consisting of the sum of two discrete-time sinusoids with frequencies of π/8 and π/4 radians/sample and amplitudes of 1 and 0.25 respectively. Filter the discrete-time signal with the FIR equiripple filter object, `Hd`

```
n = 0:159;
x = cos((pi/8)*n)+0.25*sin((pi/4)*n);
y = filter(Hd,x);
Domega = (2*pi)/160;
freq = 0:(2*pi)/160:pi;
```

```
xdft = fft(x);
ydft = fft(y);
plot(freq,abs(xdft(1:length(x)/2+1)));
hold on;
plot(freq,abs(ydft(1:length(y)/2+1)),'r','linewidth',2);
legend('Original Signal','Lowpass Signal', ...
'Location','NorthEast');
ylabel('Magnitude'); xlabel('Radians/Sample');
```

Create a filter of order 10 with a 6-dB frequency of 9.6 kHz and a sampling frequency of 48 kHz.

```
d=fdesign.highpass('N,Fc',10,9600,48000);
designmethods(d)
% only valid design method is FIR window method
Hd = design(d);
% Display filter magnitude response
fvtool(Hd);
```

If you have the DSP System Toolbox software, you can specify the shape of the stopband and the rate at which the stopband decays.

Create two FIR equiripple filters with different linear stopband slopes. Specify the passband frequency to be 0.3π radians/sample and the stopband frequency to be 0.35π radians/sample. Specify 1 dB of allowable passband ripple and a stopband attenuation of 60 dB. Design one filter with a 20 dB/rad/sample stopband slope and another filter with 40 dB/rad/sample.

```
D = fdesign.highpass('Fst,Fp,Ast,Ap',0.3,0.35,60,1);
Hd1 = design(D,'equiripple','StopBandShape','linear','StopBandDecay',20);
Hd2 = design(D,'equiripple','StopBandShape','linear','StopBandDecay',40);
hfvt = fvtool([Hd1 Hd2]);
legend(hfvt,'20 dB/rad/sample','40 dB/rad/sample');
```

See Also

`design` | `designmethods` | `fdesign`

fdesign.hilbert

Purpose

Hilbert filter specification object

Syntax

```
d = fdesign.hilbert
d = fdesign.hilbert(specvalue1,specvalue2)
d = fdesign.hilbert(spec)
d = fdesign.hilbert(spec,specvalue1,specvalue2)
d = fdesign.hilbert(...,Fs)
d = fdesign.hilbert(...,MAGUNITS)
```

Description

`d = fdesign.hilbert` constructs a default Hilbert filter designer `d` with `N`, the filter order, set to 30 and `TW`, the transition width set to 0.1π radians/sample.

`d = fdesign.hilbert(specvalue1,specvalue2)` constructs a Hilbert filter designer `d` assuming the default specification string `'N,TW'`. You input `specvalue1` and `specvalue2` for `N` and `TW`.

`d = fdesign.hilbert(spec)` initializes the filter designer `Specification` property to `spec`. You provide one of the following strings as input to replace `spec`. The specification strings are not case sensitive.

Note Specifications strings marked with an asterisk require the DSP System Toolbox software.

- `'N,TW'` default spec string.
- `'TW,Ap'` *

The string entries are defined as follows:

- `Ap` — amount of ripple allowed in the pass band in decibels (the default units). Also called Apass.
- `N` — filter order.
- `TW` — width of the transition region between the pass and stop bands.

By default, `fdesign.hilbert` assumes that all frequency specifications are provided in normalized frequency units. Also, decibels is the default for all magnitude specifications.

Different specification strings may have different design methods available. Use `designmethods(d)` to get a list of the design methods available for a given specification string.

`d = fdesign.hilbert(`*`spec`*`,specvalue1,specvalue2)` initializes the filter designer specifications in `spec` with `specvalue1`, `specvalue2`, and so on. To get a description of the specifications `specvalue1` and `specvalue2`, enter

```
get(d,'description')
```

at the Command prompt.

`d = fdesign.hilbert(...,Fs)` adds the argument `Fs`, specified in Hz to define the sampling frequency. In this case, all frequencies in the specifications are in Hz as well.

`d = fdesign.hilbert(...,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

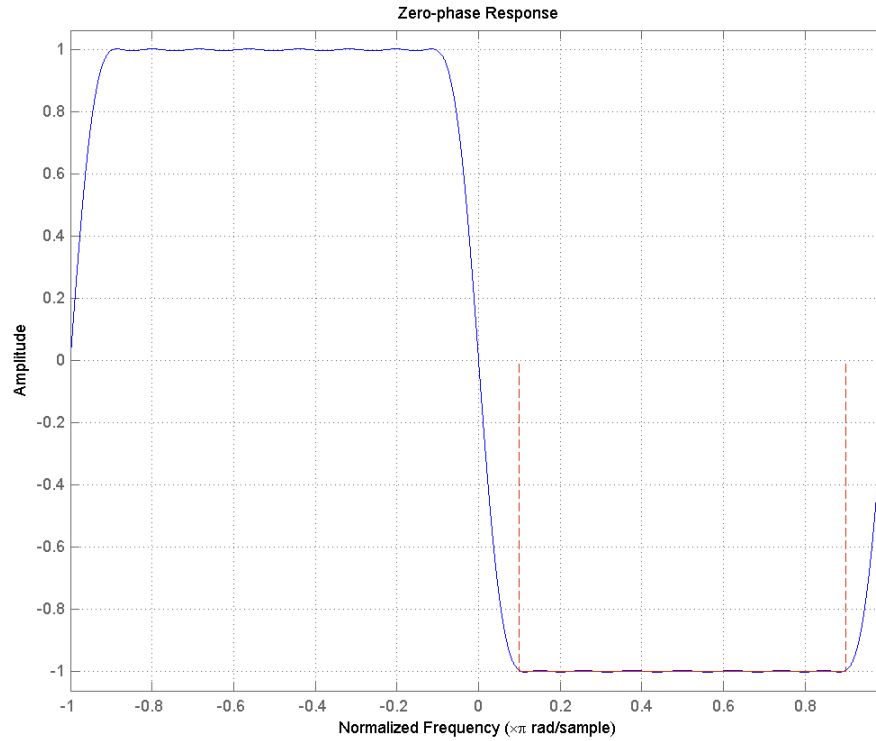
When you omit the `MAGUNITS` argument, `fdesign` assumes that all magnitudes are in decibels. Note that `fdesign` stores all magnitude specifications in decibels (converting to decibels when necessary) regardless of how you specify the magnitudes.

Examples

Design a Hilbert transformer of order 30 with a transition width of 0.2π radians/sample. Plot the zero phase response from [-π,π) radians/sample and the impulse response.

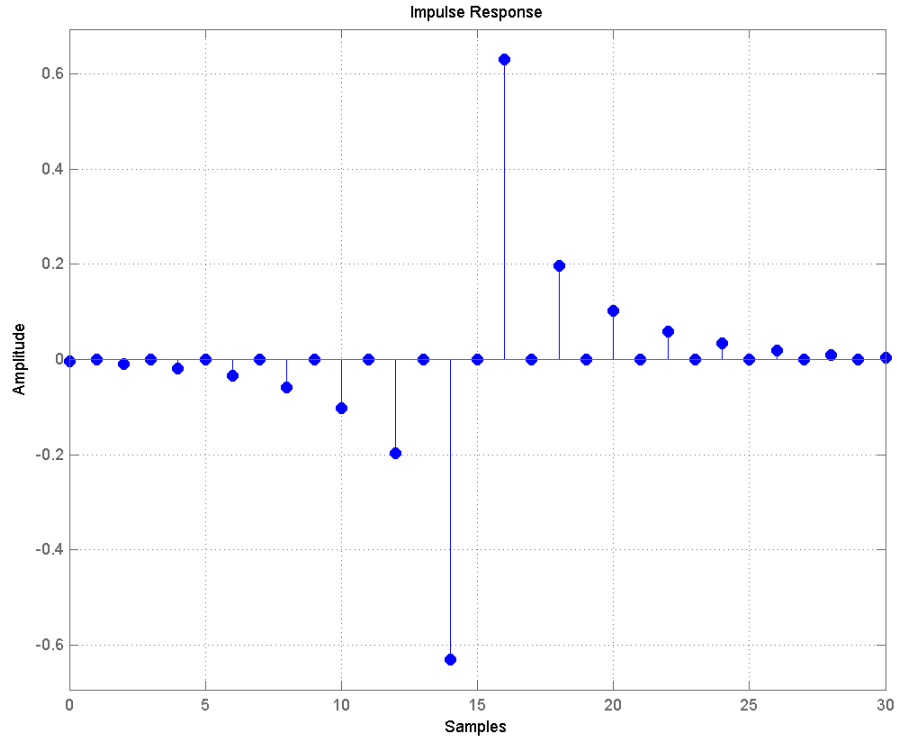
```
d = fdesign.hilbert('N,TW',30,0.2);
```

```
% Show available design methods
designmethods(d)
% Use least square minimization to obtain linear-phase FIR filter
Hd = design(d,'equiripple');
% Display zero phase response from [-pi,pi)
fvtool(Hd,'magnitudedisplay','zero-phase',...
'frequencyrange','[-pi, pi)')
```



The impulse response of this even order filter is antisymmetric (type III).

```
fvtool(Hd,'analysis','impulse')
```

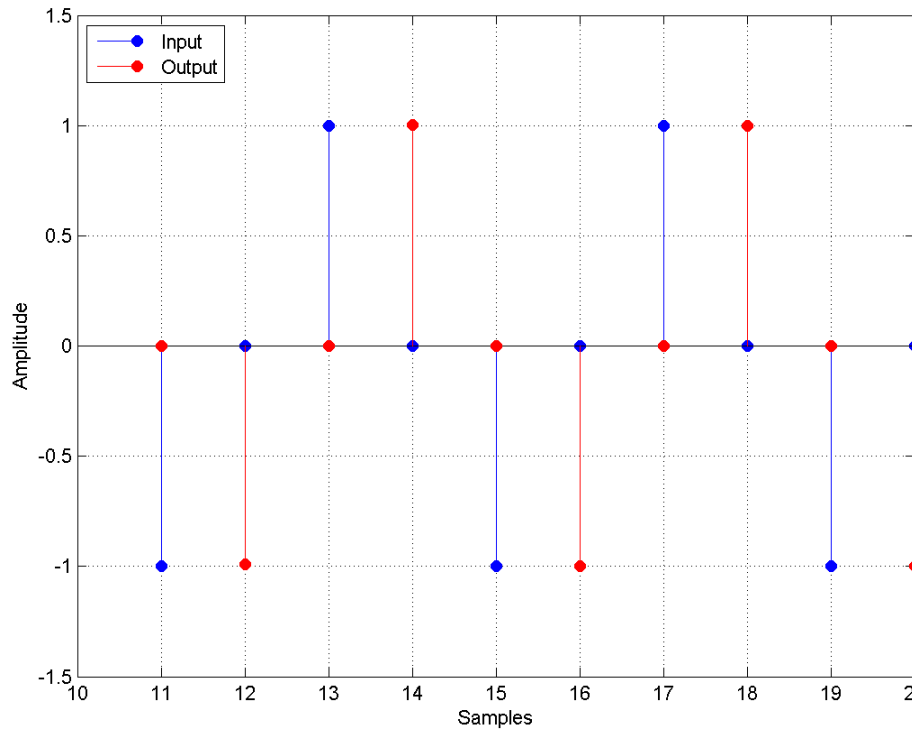


Apply the filter to a discrete-time sinusoid with a frequency of π/2 radians/sample.

```
n = 0:99;
x = cos(pi/2*n);
y = filter(Hd,x);
% Correct for the filter delay
Delay = floor(length(Hd.Numerator)/2);
y = y(Delay+1:end);
```

Plot a the filter input and output and validate the approximate π/2 phase shift obtained with the Hilbert transformer.

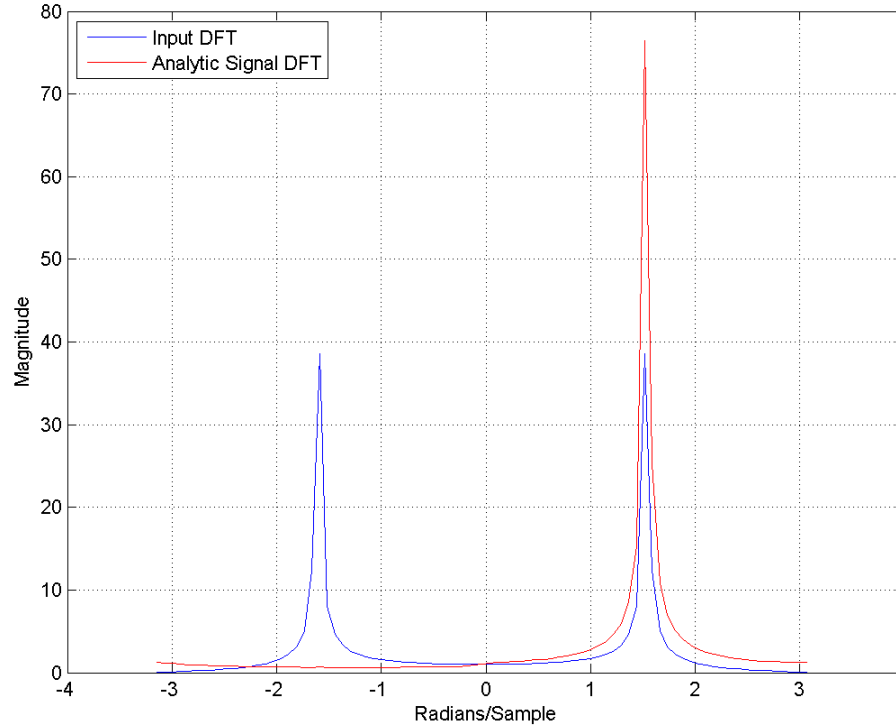
```
stem(x(1:end-Delay),'markerfacecolor',[0 0 1]);
hold on;
stem(y,'Color',[1 0 0],'markerfacecolor',[1 0 0]);
axis([10 20 -1.5 1.5]); grid on;
xlabel('Samples'); ylabel('Amplitude');
legend('Input','Output','Location','NorthWest')
```



Because the frequency of the discrete-time sinusoid is π/2 radians/sample, a one sample shift corresponds to a phase shift of π/2.

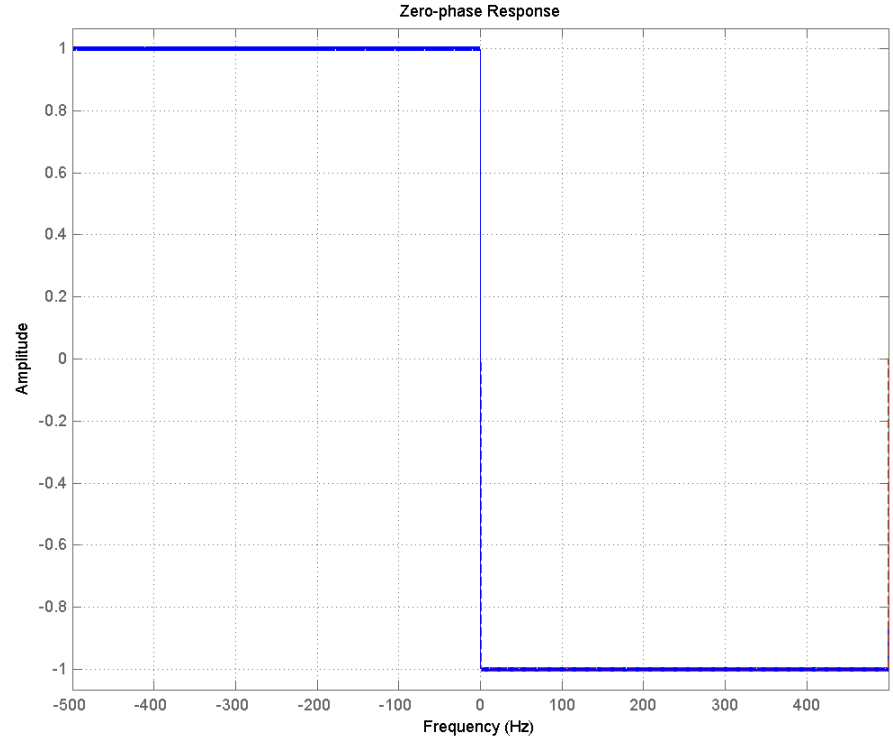
Form the analytic signal and demonstrate that the frequency content of the analytic signal is zero for negative frequencies and approximately twice the spectrum of the input for positive frequencies.

```
x1 = x(1:end-Delay);
% Form the analytic signal
xa = x1+1j*y;
freq = -pi:(2*pi)/length(x1):pi-(2*pi)/length(x);
plot(freq,abs(fftshift(fft(x1))));
hold on;
plot(freq,abs(fftshift(fft(xa))),'r'); grid on;
xlabel('Radians/Sample'); ylabel('Magnitude');
legend('Input DFT','Analytic Signal DFT','Location','NorthWest');
```



Design a minimum-order Hilbert transformer that has a sampling frequency of 1 kHz. Specify the passband ripple to be 1 dB.

```
d = fdesign.hilbert('TW,Ap',1,0.1,1e3);
hd = design(d,'equiripple');
fvtool(hd,'magnitudedisplay','zero-phase', ...
'frequencyrange','[-Fs/2, Fs/2)');
```



See Also

`design` | `fdesign` | `setspecs`

fdesign.lowpass

Purpose

Lowpass filter specification

Syntax

```
D = fdesign.lowpass
D = fdesign.lowpass(SPEC)
D = fdesign.lowpass(SPEC,specvalue1,specvalue2,...)
D = fdesign.lowpass(specvalue1,specvalue2,specvalue3,
   specvalue4)
D = fdesign.lowpass(...,Fs)
D = fdesign.lowpass(...,MAGUNITS)
```

Description

`D = fdesign.lowpass` constructs a lowpass filter specification object `D`, applying default values for the default specification string `'Fp,Fst,Ap,Ast'`.

`D = fdesign.lowpass(SPEC)` constructs object `D` and sets the `Specification` property to the string in `SPEC`. Entries in the `SPEC` string represent various filter response features, such as the filter order, that govern the filter design. Valid entries for `SPEC` are shown below. The strings are not case sensitive.

Note Specifications strings marked with an asterisk require the DSP System Toolbox software.

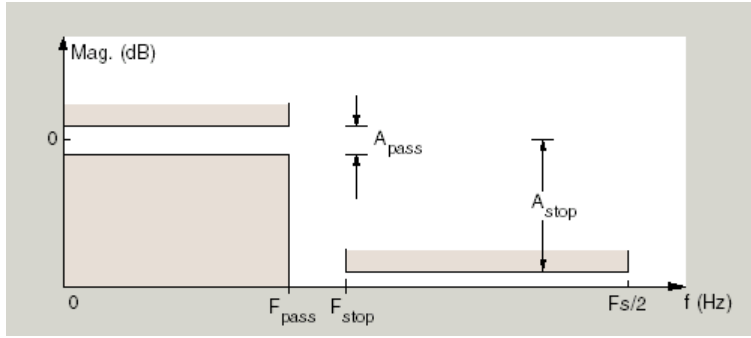
- `'Fp,Fst,Ap,Ast'` (default `spec`)
- `'N,F3db'`
- `'N,F3db,Ap'` *
- `'N,F3db,Ap,Ast'` *
- `'N,F3db,Ast'` *
- `'N,F3db,Fst'` *
- `'N,Fc'`
- `"N,Fc,Ap,Ast'`

- `'N,Fp,Ap'`
- `'N,Fp,Ap,Ast'`
- `'N,Fp,Fst,Ap'` *
- `'N,Fp,F3db'` *
- `'N,Fp,Fst'`
- `'N,Fp,Fst,Ast'` *
- `'N,Fst,Ap,Ast'` *
- `'N,Fst,Ast'`
- `'Nb,Na,Fp,Fst'` *

The string entries are defined as follows:

- `Ap` — amount of ripple allowed in the pass band in decibels (the default units). Also called Apass.
- `Ast` — attenuation in the stop band in decibels (the default units). Also called Astop.
- `F3db` — cutoff frequency for the point 3 dB point below the passband value. Specified in normalized frequency units.
- `Fc` — cutoff frequency for the point 6 dB 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.
- `Na` and `Nb` are the order of the denominator and numerator.

Graphically, the filter specifications look similar to those shown in the following figure.



Regions between specification values like `Fp` and `Fst` are transition regions where the filter response is not explicitly defined.

`D = fdesign.lowpass(SPEC,specvalue1,specvalue2,...)` constructs an object `D` and sets the specification values at construction time using `specvalue1`, `specvalue2`, and so on for all of the specification variables in `SPEC`.

`D = fdesign.lowpass(specvalue1,specvalue2,specvalue3,specvalue4)` constructs an object `D` with values for the default `Specification` property string `'Fp,Fst,Ap,Ast'` using the specifications you provide as input arguments `specvalue1,specvalue2,specvalue3,specvalue4`.

`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 MAGNUNITS argument, fdesign assumes that all magnitudes are in decibels. Note that fdesign stores all magnitude specifications in decibels (converting to decibels when necessary) regardless of how you specify the magnitudes.

Examples

Lowpass filter a discrete-time signal consisting of two sine waves.

Create a lowpass filter specification object. Specify the passband frequency to be 0.15π radians/sample and the stopband frequency to be 0.25π radians/sample. Specify 1 dB of allowable passband ripple and a stopband attenuation of 60 dB.

```
d=fdesign.lowpass('Fp,Fst,Ap,Ast',0.15,0.25,1,60);
```

Query the valid design methods for your filter specification object, d.

```
designmethods(d)
```

Create an FIR equiripple filter and view the filter magnitude response with fvtool.

```
Hd = design(d,'equiripple');
fvtool(Hd);
```

Create a signal consisting of the sum of two discrete-time sinusoids with frequencies of π/8 and π/4 radians/sample and amplitudes of 1 and 0.25 respectively. Filter the discrete-time signal with the FIR equiripple filter object, Hd.

```
n = 0:159;
x = 0.25*cos((pi/8)*n)+sin((pi/4)*n);
y = filter(Hd,x);
Domega = (2*pi)/160;
freq = 0:(2*pi)/160:pi;
xdft = fft(x);
ydft = fft(y);
plot(freq,abs(xdft(1:length(x)/2+1)));
hold on;
plot(freq,abs(ydft(1:length(y)/2+1)),'r','linewidth',2);
```

```
legend('Original Signal','Highpass Signal', ...
'Location','NorthEast');
ylabel('Magnitude'); xlabel('Radians/Sample');
```

Create a filter of order 10 with a 6-dB frequency of 9.6 kHz and a sampling frequency of 48 kHz.

```
d=fdesign.lowpass('N,Fc',10,9600,48000);
designmethods(d)
% only valid design method is FIR window method
Hd = design(d);
% Display filter magnitude response
fvtool(Hd);
```

Zoom in on the magnitude response to verify that the -6 dB point is at 9.6 kHz.

If you have the DSP System Toolbox software, you can specify the shape of the stopband and the rate at which the stopband decays. The following example requires the DSP System Toolbox.

Create an FIR equiripple filter with a passband frequency of 0.2π radians/sample, a stopband frequency of 0.25π radians/sample, a passband ripple of 1 dB, and a stopband attenuation of 60 dB. Design the filter with a 20 dB/rad/sample linear stopband.

```
D = fdesign.lowpass('Fp,Fst,Ap,Ast',0.2,0.25,1,60);
Hd = design(D,'equiripple','StopBandShape','linear','StopBandDecay',20);
fvtool(Hd);
```

See Also

`design` | `designmethods` | `fdesign`

Purpose

Pulse-shaping filter specification object

Syntax

```
D = fdesign.pulseshaping
D = fdesign.pulseshaping(sps)
D = fdesign.pulseshaping(sps,shape)
d = fdesign.pulseshaping(sps,shape,spec,value1,value2,...)
d = fdesign.pulseshaping(...,fs)
d = fdesign.pulseshaping(...,magunits)
```

Description

`D = fdesign.pulseshaping` constructs a specification object `D`, which can be used to design a minimum-order raised cosine filter object with a default stop band attenuation of 60dB and a rolloff factor of 0.25.

`D = fdesign.pulseshaping(sps)` constructs a minimum-order raised cosine filter specification object `d` with a positive integer-valued oversampling factor, `SamplesPerSymbol` .

`D = fdesign.pulseshaping(sps,shape)` constructs `d` where `shape` specifies the `PulseShape` property. Valid entries for `shape` are:

- `'Raised Cosine'`
- `'Square Root Raised Cosine'`
- `'Gaussian'`

`d = fdesign.pulseshaping(sps,shape,spec,value1,value2,...)` constructs `d` where `spec` defines the `Specification` properties. The string entries for `spec` specify various properties of the filter, including the order and frequency response. Valid entries for `spec` depend upon the `shape` property. For `'Raised Cosine'` and `'Square Root Raised Cosine'` filters, the valid entries for `spec` are:

- `'Ast,Beta'` (minimum order; default)
- `'Nsym,Beta'`
- `'N,Beta'`

The string entries are defined as follows:

- `Ast` —stopband attenuation (in dB). The default stopband attenuation for a raised cosine filter is 60 dB. The default stopband attenuation for a square root raised cosine filter is 30 dB. If `Ast` is specified, the minimum-order filter is returned.
- `Beta` —rolloff factor expressed as a real-valued scalar ranging from 0 to 1. Smaller rolloff factors result in steeper transitions between the passband and stopband of the filter.
- `Nsym` —filter order in symbols. The length of the impulse response is given by `Nsym*SamplesPerSymbol+1`. The product `Nsym*SamplesPerSymbol` must be even.
- `N` —filter order (must be even). The length of the impulse response is `N+1`.

If the `shape` property is specified as `'Gaussian'`, the valid entries for `spec` are:

- `'Nsym,BT'` (default)

The string entries are defined as follows:

- `Nsym`—filter order in symbols. `Nsym` defaults to 6. The length of the filter impulse response is `Nsym*SamplesPerSymbol+1`. The product `Nsym*SamplesPerSymbol` must be even.
- `BT` —the 3–dB bandwidth-symbol time product. `BT` is a positive real-valued scalar, which defaults to 0.3. Larger values of `BT` produce a narrower pulse width in time with poorer concentration of energy in the frequency domain.

`d = fdesign.pulseshaping(...,fs)` specifies the sampling frequency of the signal to be filtered. `fs` must be specified as a scalar trailing the other numerical values provided. For this case, `fs` is assumed to be in Hz and is used for analysis and visualization.

`d = fdesign.pulseshaping(...,magunits)` specifies the units for any magnitude specification you provide in the input arguments. Valid entries for `magunits` are:

- `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 decibels. Note that `fdesign` stores all magnitude specifications in decibels (converting to decibels when necessary) regardless of how you specify the magnitudes.

After creating the specification object `d`, you can use the `design` function to create a filter object such as `h` in the following example:

```
d = fdesign.pulseshaping(8,'Raised Cosine','Nsym,Beta',6,0.25);
h = design(d);
```

Normally, the `Specification` property of the specification object also determines which design methods you can use when you create the filter object. Currently, regardless of the `Specification` property, the `design` function uses the `window` design method with all `fdesign.pulseshaping` specification objects. The `window` method creates an FIR filter with a windowed impulse response.

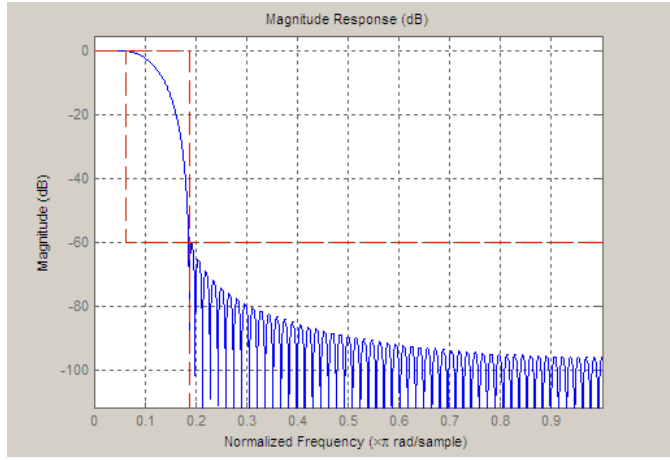
Examples

Pulse-shaping can be used to change the waveform of transmitted pulses so the signal bandwidth matches that of the communication channel. This helps to reduce distortion and intersymbol interference (ISI).

This example shows how to design a minimum-order raised cosine filter that provides a stop band attenuation of 60 dB, rolloff factor of 0.50, and 8 samples per symbol.

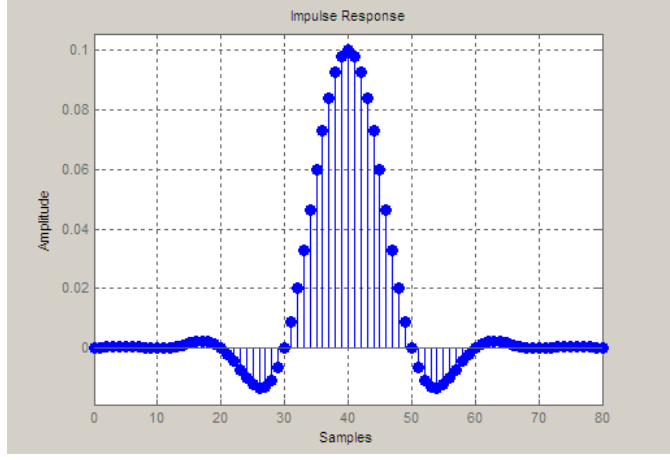
```
h  = fdesign.pulseshaping(8,'Raised Cosine','Ast,Beta',60,0.50);
    Hd = design(h);
    fvtool(Hd)
```

This code generates the following figure.



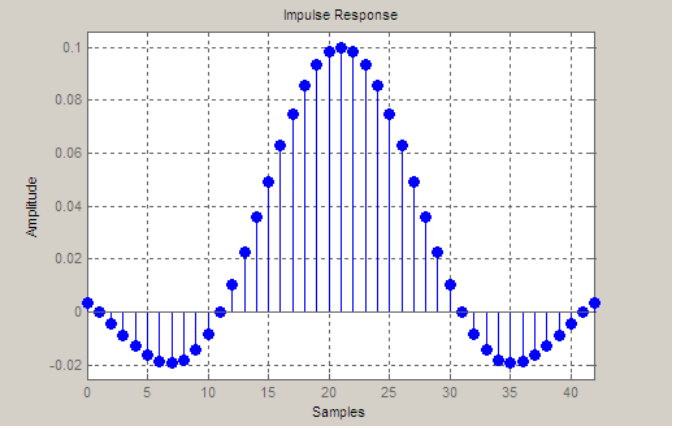
This example shows how to design a raised cosine filter that spans 8 symbol durations (i.e., of order 8 symbols), has a rolloff factor of 0.50, and oversampling factor of 10.

```
h  = fdesign.pulseshaping(10,'Raised Cosine','Nsym,Beta',8,0.50);
     Hd = design(h);
     fvtool(Hd, 'impulse')
```

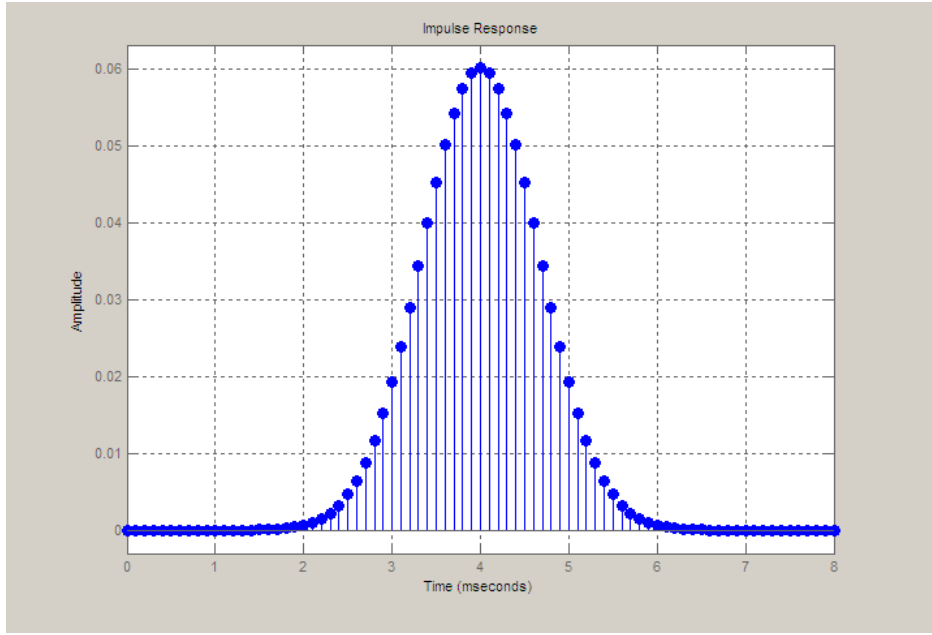


This example shows how to design a square root raised cosine filter of order 42, rolloff factor of 0.25, and 10 samples per symbol.

```
h  = fdesign.pulseshaping(10,'Square Root Raised Cosine','N,Beta',42);
    Hd = design(h);
    fvtool(Hd, 'impulse')
```



The following example demonstrates how to create a Gaussian pulse-shaping filter with an oversampling factor (sps) of 10, a bandwidth-time symbol product of 0.2, and 8 symbol periods. The sampling frequency is specified as 10 kHz.


Impulse Response
Amplitude
Time (mseconds)
0
0.01
0.02
0.03
0.04
0.05
0.06
0
1
2
3
4
5
6
7
8

Purpose

FFT-based FIR filtering using overlap-add method

Syntax

```
y = fftfilt(b,x)
y = fftfilt(b,x,n)
y = fftfilt(gpuArrayb,gpuArrayX,n)
```

Description

`fftfilt` filters data using the efficient FFT-based method of *overlap-add*, a frequency domain filtering technique that works only for FIR filters.

`y = fftfilt(b,x)` filters the data in vector `x` with the filter described by coefficient vector `b`. It returns the data vector `y`. The operation performed by `fftfilt` is described in the *time domain* by the difference equation:

$$y(n) = b(1)x(n) + b(2)x(n-1) + \cdots + b(nb+1)x(n-nb)$$

An equivalent representation is the *z*-transform or *frequency domain* description:

$$Y(z) = \left(b(1) + b(2)z^{-1} + \cdots + b(nb+1)z^{-nb}\right)X(z)$$

By default, `fftfilt` chooses an FFT length and data block length that guarantee efficient execution time.

If `x` is a matrix, `fftfilt` filters its columns. If `b` is a matrix, `fftfilt` applies the filter in each column of `b` to the signal vector `x`. If `b` and `x` are both matrices with the same number of columns, the i-th column of `b` is used to filter the i-th column of `x`.

`y = fftfilt(b,x,n)` uses n to determine the length of the FFT. See "Algorithms" on page 1-361 for information.

`y = fftfilt(gpuArrayb,gpuArrayX,n)` filters the data in the `gpuArray` object, `gpuArrayX`, with the FIR filter coefficients in the `gpuArray`, `gpuArrayb`. See "Use gpuArray Data" for details on gpuArray objects. Using `fftfilt` with `gpuArray` objects requires Parallel Computing Toolbox software and a

CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details. The filtered data, `y`, is a gpuArray object. See "Overlap-Add Filtering on the GPU" on page 1-361 for example of overlap-add filtering on the GPU.

`fftfilt` works for both real and complex inputs.

Comparison to filter function

When the input signal is relatively large, it is advantageous to use `fftfilt` instead of `filter`, which performs N multiplications for each sample in `x`, where N is the filter length. `fftfilt` performs 2 FFT operations — the FFT of the signal block of length L plus the inverse FT of the product of the FFTs — at the cost of

$$1/2*L*\log2(L)$$

where L is the block length. It then performs L pointwise multiplications for a total cost of

$$L+L*\log2(L) = L*(1+\log2(L))$$

multiplications. The cost ratio is therefore

$$L*(1+\log2(L))/(N*L) = (1+\log2(L))/N$$

which is approximately log2(L)/N.

Therefore, `fftfilt` becomes advantageous when log2(L) is less than N.

Examples

Show that the results from `fftfilt` and `filter` are identical:

```
b = [1 2 3 4];
x = [1 zeros(1,99)]';
norm(fftfilt(b,x) - filter(b,1,x))
```

Overlap-Add Filtering on the GPU

The following example requires Parallel Computing Toolbox software and a CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details.

Create a signal consisting of a sum of sine waves in white Gaussian additive noise. The sine wave frequencies are 2.5, 5, 10, and 15 kHz. The sampling frequency is 50 kHz.

```
Fs = 50e3;
t = 0:1/Fs:10-(1/Fs);
x = cos(2*pi*2500*t)+0.5*sin(2*pi*5000*t)+0.25*cos(2*pi*10000*t)+0.125
```

Design a lowpass FIR equiripple filter using `fdesign.lowpass`.

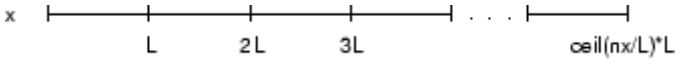
```
d = fdesign.lowpass('Fp,Fst,Ap,Ast',5500,6000,0.5,50,50e3);
Hd  = design(d);
B = Hd.Numerator;
```

Filter the data on the GPU using the overlap-add method. Put the data on the GPU using `gpuArray`. Return the output to the MATLAB workspace using `gather` and plot the power spectral density estimate of the filtered data.

```
y = fftfilt(gpuArray(B),gpuArray(x));
periodogram(gather(y),rectwin(length(y)),length(y),50e3);
```

Algorithms

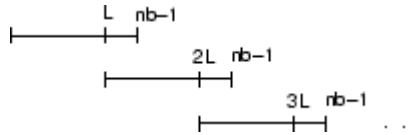
`fftfilt` uses `fft` to implement the *overlap-add method* [1], a technique that combines successive frequency domain filtered blocks of an input sequence. `fftfilt` breaks an input sequence `x` into length `L` data blocks, where L must be greater than the filter length N.



and convolves each block with the filter `b` by

```
y = ifft(fft(x(i:i+L-1),nfft).*fft(b,nfft));
```

where `nfft` is the FFT length. `fftfilt` overlaps successive output sections by `n-1` points, where `n` is the length of the filter, and sums them.



`fftfilt` chooses the key parameters `L` and `nfft` in different ways, depending on whether you supply an FFT length `n` and on the lengths of the filter and signal. If you do not specify a value for `n` (which determines FFT length), `fftfilt` chooses these key parameters automatically:

- If `length(x)` is greater than `length(b)`, `fftfilt` chooses values that minimize the number of blocks times the number of flops per FFT.
- If `length(b)` is greater than or equal to `length(x)`, `fftfilt` uses a single FFT of length

  ```
  2^nextpow2(length(b) + length(x) - 1)
  ```

  This essentially computes

  ```
  y = ifft(fft(B,nfft).*fft(X,nfft))
  ```

If you supply a value for `n`, `fftfilt` chooses an FFT length, `nfft`, of `2^nextpow2(n)` and a data block length of `nfft - length(b) + 1`. If `n` is less than `length(b)`, `fftfilt` sets `n` to `length(b)`.

References

[1] Oppenheim, A.V., and R.W. Schafer. *Discrete-Time Signal Processing,* Prentice-Hall, 1989.

See Also

`conv` | `dfilt.fftfir` | `filter` | `filtfilt`

Purpose

Filter data with recursive (IIR) or nonrecursive (FIR) filter

Description

`filter` is a MATLAB function.

Signal-Specific Information

Filter Method of DFILT

Filter is also an overloaded method of the discrete-time filter object (`dfilt`). You can pass an object handle, data, and optionally, the dimension into the filter method.

The MATLAB `filter` function describes a `zi` input for initial conditions. Note that the recommended way of passing initial conditions into a `dfilt` is by using the `states` property. For more information, see the `dfilt` reference page.

Filter Normalization

Using the `filter` function on `b` and `a` coefficients normalizes the filter by forcing the a_0 coefficient to be equal to 1.

Using the `filter` method on a `dfilt` object does not normalize the a_0 coefficient.

FIR Filters

The denominator of FIR filters is, by definition, equal to 1. To use the `filter` function with the `b` coefficients from an FIR function, use `y = filter(b,1,x)` .

filterbuilder

Purpose

GUI-based filter design

Syntax

```
filterbuilder(h)
filterbuilder('response')
```

Description

`filterbuilder` starts a GUI-based tool for building filters. It relies on the `fdesign` object-object oriented filter design paradigm, and is intended to reduce development time during the filter design process. `filterbuilder` uses a specification-centered approach to find the best algorithm for the desired response.

Note You must have the Signal Processing Toolbox installed to use `fdesign` and `filterbuilder`. Some of the features described below may be unavailable if your installation does not additionally include the DSP System Toolbox. You can verify the presence of both toolboxes by typing `ver` at the command prompt.

The `filterbuilder` GUI contains many features not available in FDATool. For more information on how to use `filterbuilder`, see "Filterbuilder Design Process""Filterbuilder Design Process".

To use `filterbuilder`, enter `filterbuilder` at the MATLAB command line using one of three approaches:

- Simply enter `filterbuilder`. MATLAB opens a dialog for you to select a filter response type. After you select a filter response type, `filterbuilder` launches the appropriate filter design dialog box.
- Enter `filterbuilder(h)`, where h is an existing filter object. For example, if `h` is a bandpass filter, `filterbuilder(h)` opens the bandpass filter design dialog box. (The `h` object must have been created using `filterbuilder` or must be a `dfilt` ,`mfilt`, or filter System object created using fdesign.)

Note You must have the DSP System Toolbox software to create and import filter System objects.

- Enter `filterbuilder('response')`, replacing *response* with a response string from the following table. MATLAB opens a filter design dialog that corresponds to the response string.

Note You must have the DSP System Toolbox software to implement a number of the filter designs listed in the following table. If you only have the Signal Processing Toolbox software, you can design a limited set of the following filter-response types.

Response String	Description of Resulting Filter Design	Filter Object
arbgrpdelay	Arbitrary group delay filter design	`fdesign.arbgrpdelay`
arbmag	Arbitrary magnitude filter design	`fdesign.arbmag`
arbmagnphase	Arbitrary response filter (magnitude and phase)	`fdesign.arbmagnphase`
audioweighting	Audio weighting filter	`fdesign.audioweighting`
bandpass or `bp`	Bandpass filter	`fdesign.bandpass`
bandstop or `bs`	Bandstop filter	`fdesign.bandstop`
cic	CIC filter	`fdesign.decimator(M,'cic',...` or
ciccomp	CIC compensator	`fdesign.ciccomp` `r(L,'cic',`
comb	Comb filter	`fdesign.comb` and `fdesign.interpolator`

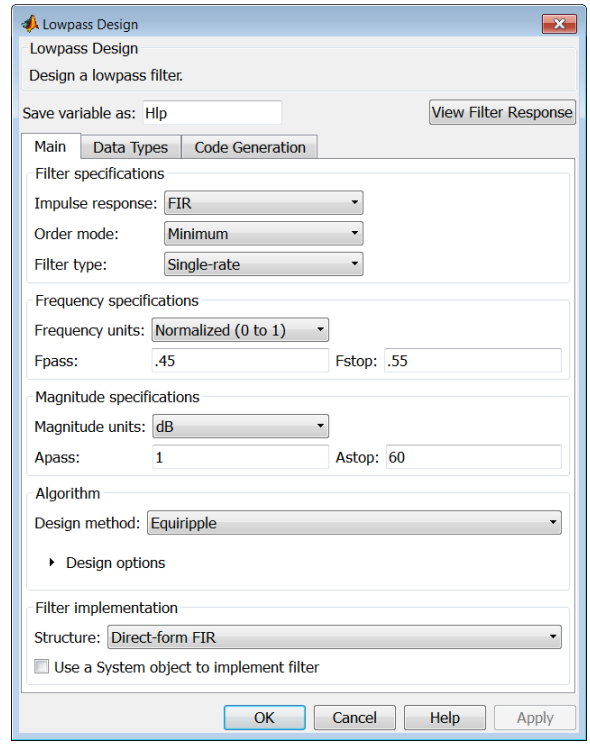
Response String	Description of Resulting Filter Design	Filter Object
diff	Differentiator filter	`fdesign.differentiator`
fracdelay	Fractional delay filter	`fdesign.fracdelay`
halfband or `hb`	Halfband filter	`fdesign.halfband`
highpass or `hp`	Highpass filter	`fdesign.highpass`
hilb	Hilbert filter	`fdesign.hilbert`
isinc, isinclp, or isinchp	Inverse sinc lowpass or highpass filter	`fdesign.isinclp` and `fdesign.isinchp`
lowpass or `lp`	Lowpass filter (default)	`fdesign.lowpass`
notch	Notch filter	`fdesign.notch`
nyquist	Nyquist filter	`fdesign.nyquist`
octave	Octave filter	`fdesign.octave`
parameq	Parametric equalizer filter	`fdesign.parameq`
peak	Peak filter	`fdesign.peak`
pulseshaping	Pulse-shaping filter	`fdesign.pulseshaping`

Note Because they do not change the filter structure, the magnitude specifications and design method are tunable when using `filterbuilder`.

Filterbuilder Design Panes

Main Design Pane

The main pane of filterbuilder varies depending on the filter response type, but the basic structure is the same. The following figure shows the basic layout of the dialog box.



As you choose the response for the filter, the available options and design parameters displayed in the dialog box change. This display allows you to focus only on parameters that make sense in the context of your filter design.

Every filter design dialog box includes the options displayed at the top of the dialog box, shown in the following figure.

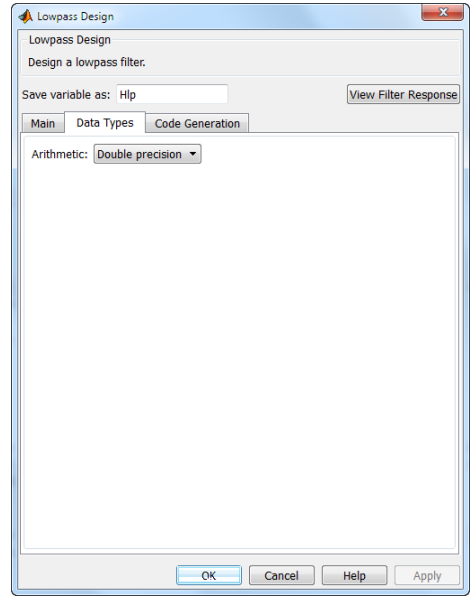
- **Save variable as** — When you click **Apply** to apply your changes or **OK** to close this dialog box, `filterbuilder` saves the current filter to your MATLAB workspace as a filter object with the name you enter.
- **View Filter Response** — Displays the magnitude response for the current filter specifications and design method by opening the Filter Visualization Tool (`fvtool`).

Note The `filterbuilder` dialog box includes an **Apply** option. Each time you click **Apply**, `filterbuilder` writes the modified filter to your MATLAB workspace. This modified filter has the variable name you assign in **Save variable as**. To apply changes without overwriting the variable in you workspace, change the variable name in **Save variable as** before you click **Apply**.

There are three tabs in the Filterbuilder dialog box, containing three panes: **Main**, **Data Types**, and **Code Generation**. The first pane changes according to the filter being designed. The last two panes are the same for all filters. These panes are discussed in the following sections.

Data Types Pane

The second tab in the Filterbuilder dialog box is shown in the following figure.

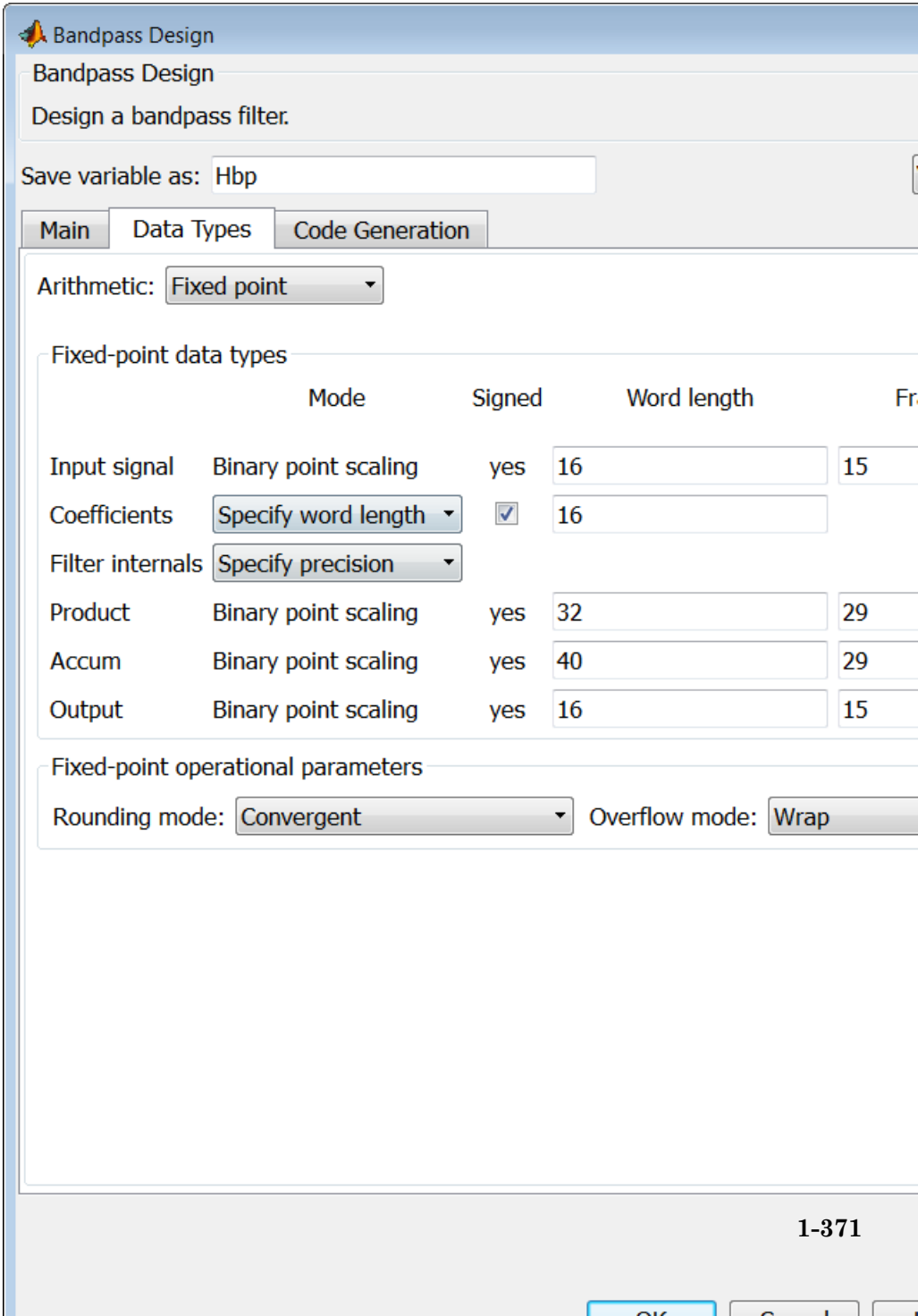


The **Arithmetic** drop down box allows the choice of `Double precision`, `Single precision`, or `Fixed point`. Some of these options may be unavailable depending on the filter parameters. The following table describes these options.

Arithmetic List Entry	Effect on the Filter
`Double precision`	All filtering operations and coefficients use double-precision, floating-point representations and math. When you use `filterbuilder` to

Arithmetic List Entry	Effect on the Filter
	create a filter, `double precision` is the default value for the Arithmetic property.
`Single precision`	All filtering operations and coefficients use single-precision floating-point representations and math.
`Fixed point`	This string applies selected default values, typically used on many digital processors, for the properties in the fixed-point filter. These properties include coefficient word lengths, fraction lengths, and various operating modes. This setting allows signed fixed data types only. Fixed-point filter design with `filterbuilder` is available only when you install Fixed-Point Designer™ software along with DSP System Toolbox software.

The following figure shows the **Data Types** pane after you select `Fixed point` for **Arithmetic** and set **Filter internals** to `Specify precision`. This figure shows the **Data Types** pane for the case where the **Use a System object to implement filter** check box is not selected in the **Main** pane.


Bandpass Design
Bandpass Design
Design a bandpass filter.
Save variable as: Hbp
Main
Data Types
Code Generation
Arithmetic: Fixed point
Fixed-point data types
Mode
Signed
Word length
Input signal
Binary point scaling
yes
16
15
Coefficients
Specify word length
16
Filter internals
Specify precision
Product
Binary point scaling
yes
32
29
Accum
Binary point scaling
yes
40
29
Output
Binary point scaling
yes
16
15
Fixed-point operational parameters
Rounding mode: Convergent
Overflow mode: Wrap

Input signal

Specify the format the filter applies to data to be filtered. For all cases, `filterbuilder` implements filters that use binary point scaling and signed input. You set the word length and fraction length as needed.

Coefficients

Choose how you specify the word length and the fraction length of the filter numerator and denominator coefficients:

- `Specify word length` enables you to enter the word length of the coefficients in bits. In this mode, `filterbuilder` automatically sets the fraction length of the coefficients to the binary-point only scaling that provides the best possible precision for the value and word length of the coefficients.
- `Binary point scaling` enables you to enter the word length and the fraction length of the coefficients in bits. If applicable, enter separate fraction lengths for the numerator and denominator coefficients.
- The filter coefficients do not obey the **Rounding mode** and **Overflow mode** parameters that are available when you select `Specify precision` from the Filter internals list. Coefficients are always saturated and rounded to `Nearest`.

Section Input

Choose how you specify the word length and the fraction length of the fixed-point data type going into each section of an SOS filter. This parameter is visible only when the selected filter structure is IIR and SOS.

- `Binary point scaling` enables you to enter the word and fraction lengths of the section input in bits.
- `Specify word length` enables you to enter the word lengths in bits.

Section Output

Choose how you specify the word length and the fraction length of the fixed-point data type coming out of each section of an SOS

filter. This parameter is visible only when the selected filter structure is IIR and SOS.

- `Binary point scaling` enables you to enter the word and fraction lengths of the section output in bits.
- `Specify word length` enables you to enter the output word lengths in bits.

State

Contains the filter states before, during, and after filter operations. States act as filter memory between filtering runs or sessions. Use this parameter to specify how to designate the state word and fraction lengths. This parameter is not visible for direct form and direct form I filter structures because `filterbuilder` deduces the state directly from the input format. States always use signed representation:

- `Binary point scaling` enables you to enter the word length and the fraction length of the accumulator in bits.
- `Specify precision` enables you to enter the word length and fraction length in bits (if available).

Product

Determines how the filter handles the output of product operations. Choose from the following options:

- `Full precision` — Maintain full precision in the result.
- `Keep LSB` — Keep the least significant bit in the result when you need to shorten the data words.
- `Specify Precision` — Enables you to set the precision (the fraction length) used by the output from the multiplies.

Filter internals

Specify how the fixed-point filter performs arithmetic operations within the filter. The affected filter portions are filter products, sums, states, and output. Select one of these options:

- `Full precision` — Specifies that the filter maintains full precision in all calculations for products, output, and in the accumulator.
- `Specify precision` — Set the word and fraction lengths applied to the results of product operations, the filter output, and the accumulator. Selecting this option enables the word and fraction length controls.

Signed

Selecting this option directs the filter to use signed representations for the filter coefficients.

Word length

Sets the word length for the associated filter parameter in bits.

Fraction length

Sets the fraction length for the associate filter parameter in bits.

Accum

Use this parameter to specify how you would like to designate the accumulator word and fraction lengths.

Determines how the accumulator outputs stored values. Choose from the following options:

- `Full precision` — Maintain full precision in the accumulator.
- `Keep MSB` — Keep the most significant bit in the accumulator.
- `Keep LSB` — Keep the least significant bit in the accumulator when you need to shorten the data words.
- `Specify Precision` — Enables you to set the precision (the fraction length) used by the accumulator.

Output

Sets the mode the filter uses to scale the output data after filtering. You have the following choices:

- `Avoid Overflow` — Set the output data fraction length to avoid causing the data to overflow. `Avoid overflow` is considered

the conservative setting because it is independent of the input data values and range.

- `Best Precision` — Set the output data fraction length to maximize the precision in the output data.
- `Specify Precision` — Set the fraction length used by the filtered data.

Fixed-point operational parameters

Parameters in this group control how the filter rounds fixed-point values and how it treats values that overflow.

Rounding mode

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).

- `ceil` - Round toward positive infinity.
- `convergent` - Round to the closest representable integer. Ties round to the nearest even stored integer. This is the least biased of the methods available in this software.
- `zero/fix` - Round toward zero.
- `floor` - Round toward negative infinity.
- `nearest` - Round toward nearest. Ties round toward positive infinity.
- `round` - Round toward nearest. Ties round toward negative infinity for negative numbers, and toward positive infinity for positive numbers.

The choice you make affects everything except coefficient values and input data which always round. In most cases, products do not overflow—they maintain full precision.

Overflow mode

Sets the mode the filter uses to respond to overflow conditions in fixed-point arithmetic. Choose from the following options:

- `Saturate` — Limit the output to the largest positive or negative representable value.
- `Wrap` — Set overflowing values to the nearest representable value using modular arithmetic.

The choice you make affects everything except coefficient values and input data which always round. In most cases, products do not overflow—they maintain full precision.

Cast before sum

Specifies whether to cast numeric data to the appropriate accumulator format before performing sum operations. Selecting **Cast before sum** ensures that 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.

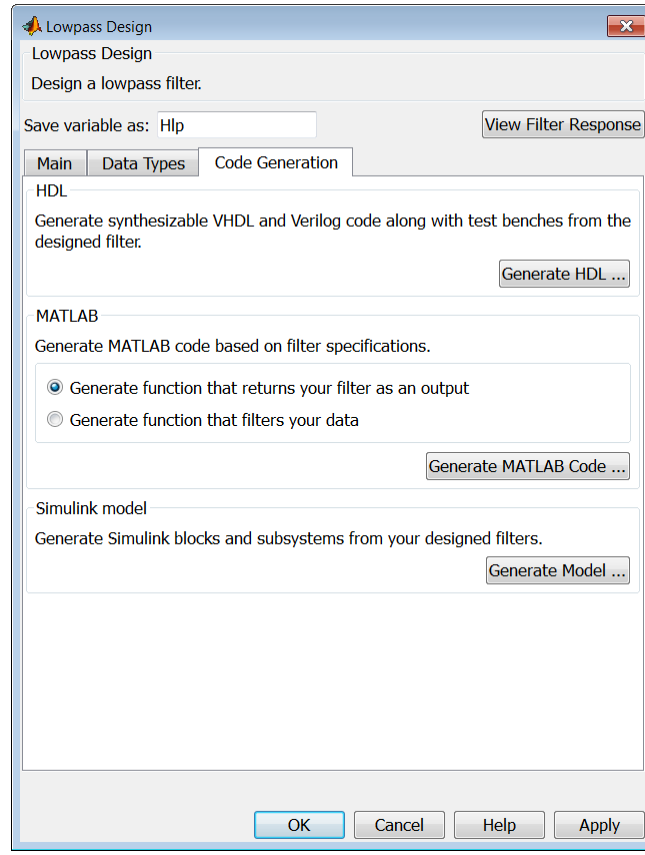
If you clear **Cast before sum**, the filter prevents the addends from being cast to the sum format before the addition operation. Choose this setting to get the most accurate results from summations without considering the hardware your filter might use. The input format referenced by **Cast before sum** depends on the filter structure you are using.

The effect of clearing or selecting **Cast before sum** is as follows:

- Cleared — Configures filter summation operations to retain the addends in the format carried from the previous operation.
- Selected — Configures filter summation operations to convert the input format of the addends to match the summation output format before performing the summation operation. Usually, selecting **Cast before sum** generates results from the summation that more closely match those found from digital signal processors.

Code Generation Pane

The code generation pane contains options for various implementations of the completed filter design. Depending on your installation, you can generate MATLAB, VHDL, and Verilog code from the designed filter. You can also choose to create or update a Simulink model from the designed filter. The following section explains these options.



HDL

For more information on this option, see "Opening the Filter Design HDL Coder™ GUI From the filterbuilder GUI".

MATLAB

Generate MATLAB code based on filter specifications

- **Generate function that returns your filter as an output**

  Selecting this option generates a function that designs either a DFILT/MFILT object or a system object (depending on whether you have selected the **Use a System object to implement the filter** check box) using fdesign. The function call returns a filter object.

- **Generate function that filters your data**

  Selecting this option generates a function that takes data as input, and outputs data filtered with the designed filter.

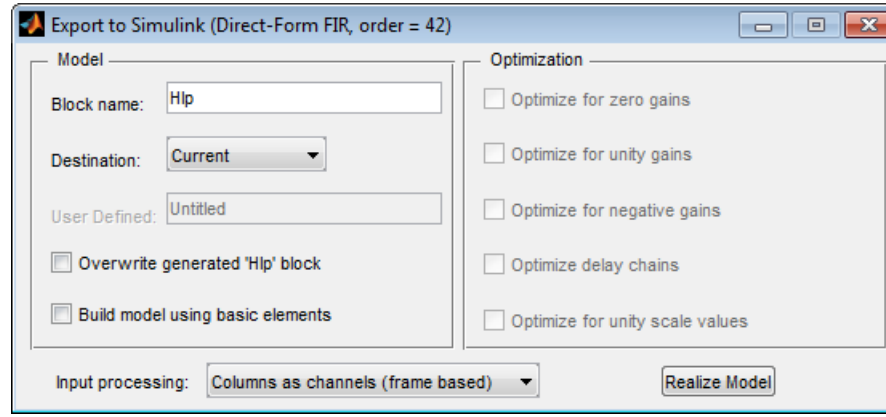
Clicking on the **Generate MATLAB code** button, brings up a Save File dialog. Specify the file name and location, and save. The filter is now contained in an editable file.

Simulink Model

Generate Simulink blocks and subsystems from your designed filters

When the **Use a System object to implement filter** check box is selected in the **Main** pane, you are able to generate Simulink models as long as the **Arithmetic** is not set to `Fixed point` in the **Data Types** pane. If the **Arithmetic** is set to `Fixed point`, the **Generate Model** button in the **Simulink model** panel will be disabled.

Clicking on the **Generate Model** button brings up the **Export to Simulink** dialog box, as shown in the following figure.



You can set the following parameters in this dialog box:

- **Block Name** — The name for the new subsystem block, set to **Filter** by default.
- **Destination** — **Current** saves the generated model to the current Simulink model; **New** creates a new model to contain the generated block; **User Defined** creates a new model or subsystem to the user-specified location enumerated in the **User Defined** text box.
- **Overwrite generated 'Filter' block** — When this check box is selected, DSP System Toolbox software overwrites an existing block with the name specified in **Block Name**; when cleared, creates a new block with the same name.
- **Build model using basic elements** — When this check box is selected, DSP System Toolbox software builds the model using only basic blocks.
- **Optimize for zero gains** — When this check box is selected, DSP System Toolbox software removes all zero gain blocks from the model.

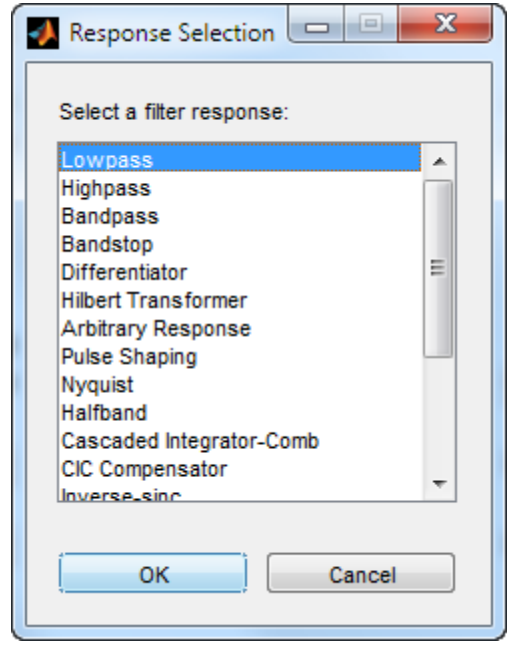
- **Optimize for unity gains** — When this check box is selected, DSP System Toolbox software replaces all unity gains with direct connections.
- **Optimize for negative gains** — When this check box is selected, DSP System Toolbox software removes all negative unity gain blocks, and changes sign at the nearest summation block.
- **Optimize delay chains** — When this check box is selected, DSP System Toolbox software replaces delay chains made up of n unit delays with a single delay by n.
- **Optimize for unity scale values** — When this check box is selected, DSP System Toolbox software removes all scale value multiplications by 1 from the filter structure.
- **Input processing** — Specify how the generated filter block or subsystem block processes the input. Depending on the type of filter you are designing, one or both of the following options may be available:
  - `Columns as channels (frame based)` — When you select this option, the block treats each column of the input as a separate channel.
  - `Elements as channels (sample based)` — When you select this option, the block treats each element of the input as a separate channel.

  For more information about sample- and frame-based processing, see "Sample- and Frame-Based Concepts".
- **Realize Model** — DSP System Toolbox software builds the model with the set parameters.

Filter Responses

Select your filter response from the `filterbuilder` **Response Selection** main menu.

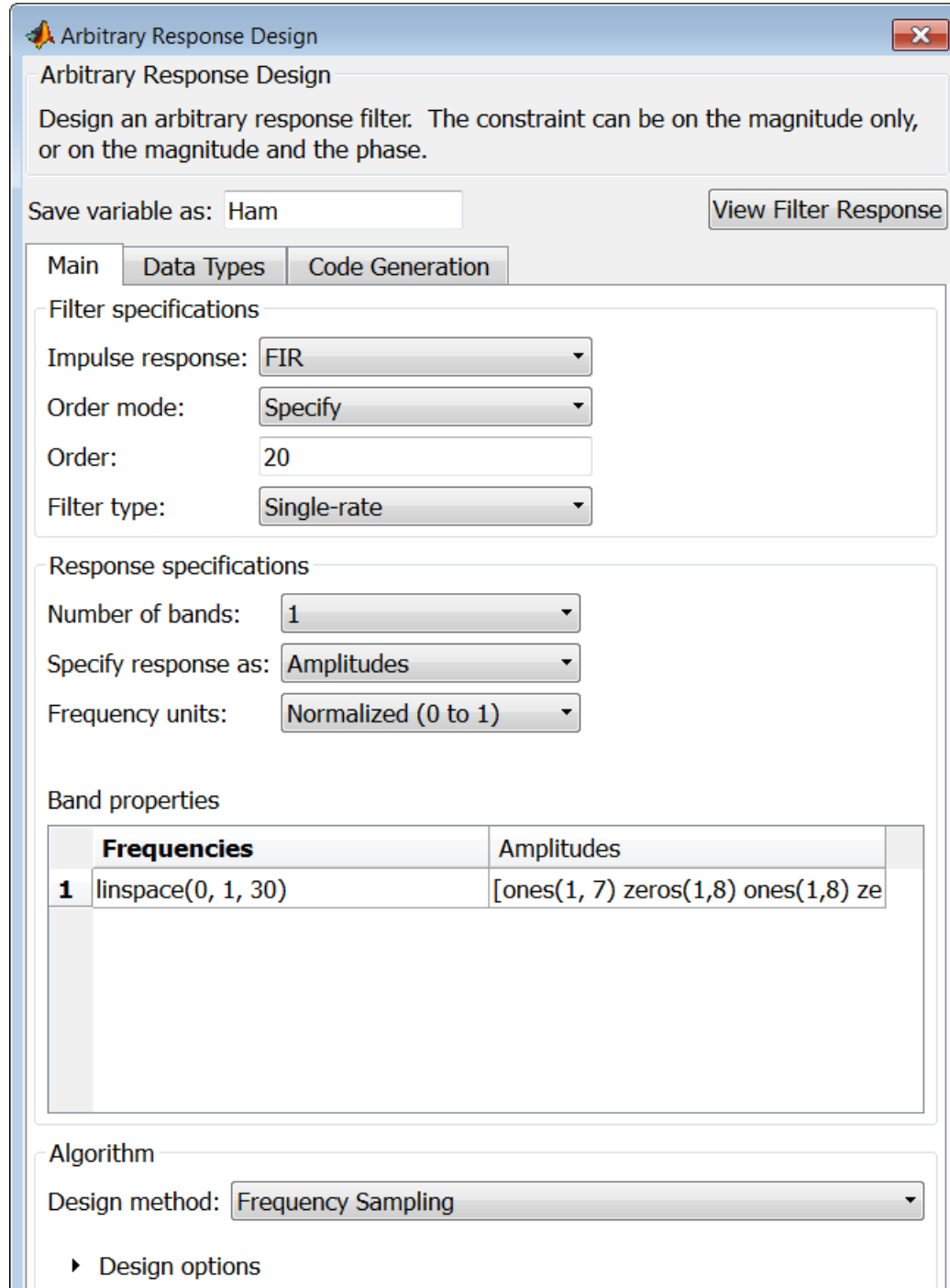
If you have the DSP System Toolbox software, the following **Response Selection** menu appears.



Select your desired filter response from the menu and design your filter.

The following sections describe the options available for each response type.

Arbitrary Response Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter format, such as the impulse response and the filter order.

Impulse response

This dialog only applies if you have the DSP System Toolbox software. Select either `FIR` or `IIR` from the drop down list, where `FIR` is the default impulse response. When you choose an impulse response, the design methods and structures you can use to implement your filter change accordingly. Arbitrary group delay designs are only available if **Impulse response** is `IIR`. Without the DSP System Toolbox, the only available arbitrary response filter design is FIR.

Order mode

This dialog only applies if you have the DSP System Toolbox software. Choose `Minimum` or `Specify`. Choosing `Specify` enables the **Order** dialog.

Order

This dialog only applies when **Order mode** is `Specify`. For an FIR design, specify the filter order. For an IIR design, you can specify an equal order for the numerator and denominator, or you can specify different numerator and denominator orders. The default is equal orders. To specify a different denominator order, check the **Denominator order** box. Because the Signal Processing Toolbox only supports FIR arbitrary-magnitude filters, you do not have the option to specify a denominator order.

Denominator order

Select the check box and enter the denominator order. This option is enabled only if `IIR` is selected for **Impulse response**.

Filter type

This dialog only applies if you have the DSP System Toolbox software and is only available for FIR filters. Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods

and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

When you design either a decimator or interpolator, the resulting filter is a bandpass filter that either decimates or interpolates your input signal.

Decimation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2 for `Decimator` and 3 for `Sample-rate converter`.

Interpolation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Response Specification

Number of Bands

Select the number of bands in the filter. Multiband design is available for both FIR and IIR filters.

Specify response as:

Specify the response as `Amplitudes`, `Magnitudes and phase`, `Frequency response`, or `Group delay`. `Amplitudes` is the only option if you do not have the DSP System Toolbox software. `Group delay` is only available for IIR designs.

Frequency units

Specify frequency units as either `Normalized`, `Hz`, `kHz`, `MHz`, or `GHz`.

Input Fs

Enter the input sampling frequency in the units specified in the **Frequency units** drop-down box. This option is enabled when **Frequency units** is set to an option in hertz.

Band Properties

These properties are modified automatically depending on the response chosen in the **Specify response as** drop-down box. Two or three columns are presented for input. The first column is always Frequencies. The other columns are either Amplitudes, Magnitudes, Phases, or Frequency Response. Enter the corresponding vectors of values for each column.

- **Frequencies** and **Amplitudes** — These columns are presented for input if you select `Amplitudes` in the **Specify response as** drop-down box.
- **Frequencies**, **Magnitudes**, and **Phases** — These columns are presented for input if the response chosen in the **Specify response as** drop-down box is `Magnitudes and phases`.
- **Frequencies** and **Frequency response** — These columns are presented for input if the response chosen in the **Specify response as** drop-down box is `Frequency response`.

Algorithm

The options for each design are specific for each design method. In the arbitrary response design, the available options also depend on the **Response specifications**. This section does not present all of the available options for all designs and design methods.

Design Method

Select the design method for the filter. Different methods are enabled depending on the defining parameters entered in the previous sections.

Design Options

- **Window** — Valid when the **Design method** is `Frequency Sampling`. Replace the square brackets with the name of a `window` function or function handle. For example, `'hamming'` or `@hamming`. If the window function takes parameters other than the length, use a cell array. For example, `{`kaiser',3.5}` or `{@chebwin,60}`.
- **Density factor** — Valid when the **Design method** is `equiripple`. Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

  Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

  The default changes to 20 for an IIR arbitrary group delay design.
- **Phase constraint** — Valid when the **Design method** is `equiripple`, you have the DSP System Toolbox installed, and **Specify response as** is set to `Amplitudes`. Choose one of `Linear`, `Minimum`, or `Maximum`.
- **Weights** — Uses the weights in **Weights** to weight the error for a single-band design. If you have multiple frequency bands, the **Weights** design option changes to **B1 Weights**, **B2 Weights** to designate the separate bands. Use **Bi Weights** to specify weights for the i-th band. The **Bi Weights** design option is only available when you specify the i-th band as an unconstrained.
- **Bi forced frequency point** — This option is only available in a multi-band constrained equiripple design when **Specify response as** is `Amplitudes`. **Bi forced frequency point** is

the frequency point in the i-th band at which the response is forced to be zero. The index **i** corresponds to the frequency bands in **Band properties**. For example, if you specify two bands in **Band properties**, you have **B1 forced frequency point** and **B2 forced frequency point**.

- **Norm** — Valid only for IIR arbitrary group delay designs. **Norm** is the norm used in the optimization. The default value is 128, which essentially equals the L-infinity norm. The norm must be even.
- **Max pole radius** — Valid only for IIR arbitrary group delay designs. Constrains the maximum pole radius. The default is 0.999999. Reducing the **Max pole radius** can produce a transfer function more resistant to quantization.
- **Init norm** — Valid only for IIR arbitrary group delay designs. The initial norm used in the optimization. The default initial norm is 2.
- **Init numerator** — Specifies an initial estimate of the filter numerator coefficients.
- **Init denominator** — Specifies an initial estimate of the filter denominator coefficients. This may be useful in difficult optimization problems. In allpass filters, you only have to specify either the denominator or numerator coefficients. If you specify the denominator coefficients, you can obtain the numerator coefficients.

Filter implementation

Structure

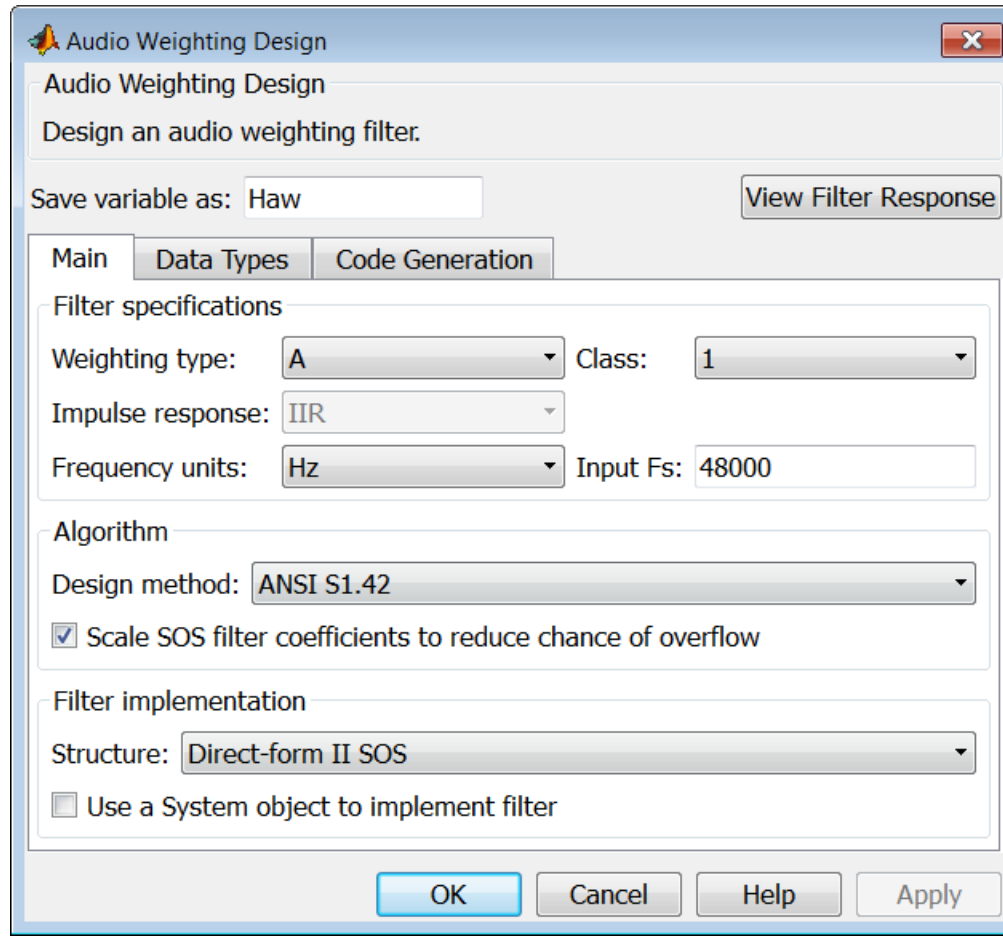
Select the structure for the filter. The available filter structures depend on the parameters you select for your filter.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned

off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Audio Weighting Filter Design Dialog Box — Main Pane



Filter specifications

- **Weighting type** — The weighting type defines the frequency response of the filter. The valid weighting types are: A, C , C-message, ITU-T 0.41, and ITU-R 468–4 weighting. See `fdesign.audioweighting` for definitions of the weighting types.
- **Class** — Filter class is only applicable for A weighting and C weighting filters. The filter class describes the frequency-dependent tolerances specified in the relevant standards. There are two possible class values: 1 and 2. Class 1 weighting filters have stricter tolerances than class 2 filters. The filter class value does not affect the design. The class value is only used to provide a specification mask in `fvtool` for the analysis of the filter design.
- **Impulse response** — Impulse response type as one of `IIR` or `FIR`. For A, C , C-message, and ITU-R 468–4 filter, IIR is the only option. For a ITU-T 0.41 weighting filter, FIR is the only option.
- **Frequency units** — Choose `Hz`, `kHz`, `MHz`, or `GHz`. Normalized frequency designs are not supported for audio weighting filters.
- **Input Fs** — The sampling frequency in **Frequency units**. For example, if **Frequency units** is set to `kHz`, setting **Input Fs** to 40 is equivalent to a 40 kHz sampling frequency.

Algorithm

- **Design method** — Valid design methods depend on the weighting type. For type A and C weighting filters, the only valid design type is `ANSI S1.42`. This is an IIR design method that follows ANSI standard S1.42–2001. For a C message filter, the only valid design method is `Bell 41009`, which is an IIR design method following the Bell System Technical Reference PUB 41009. For a ITU-R 468–4 weighting filter, you can design an IIR or FIR filter. If you choose an IIR design, the design method is `IIR least p-norm`. If you choose an FIR design, the design method choices are: `Equirriple` or `Frequency Sampling`. For an ITU-T 0.41 weighting filter, the available FIR design methods are `equirriple` or `Frequency Sampling`

- **Scale SOS filter coefficients to reduce chance of overflow** — Selecting this parameter directs the design to scale the filter coefficients to reduce the chances that the inputs or calculations in the filter overflow and exceed the representable range of the filter. Clearing this option removes the scaling. This parameter applies only to IIR filters.

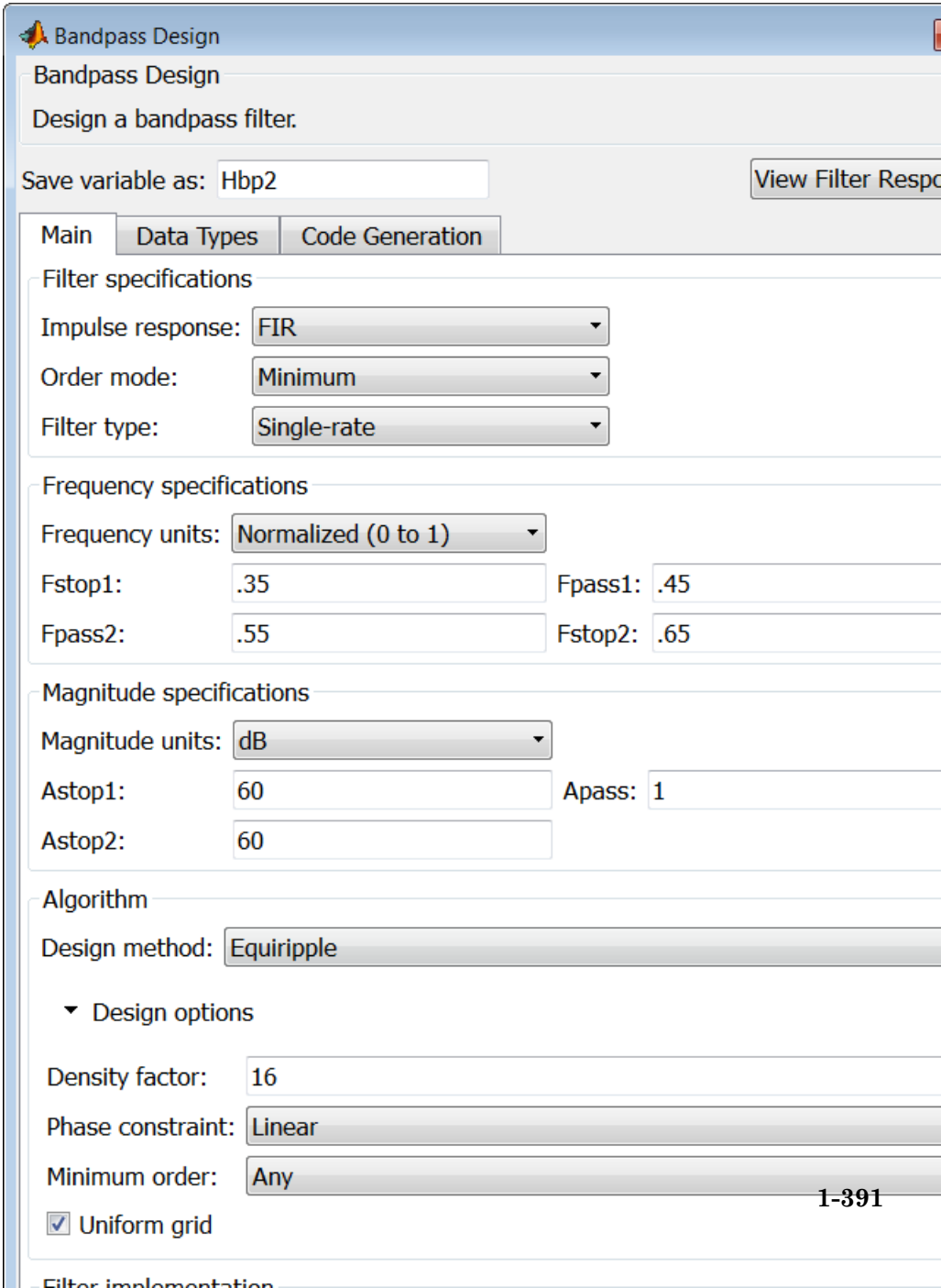
Filter implementation

- **Structure** — For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. For audio weighting IIR filter designs, you can choose direct form I or II biquad (SOS). You can also choose to implement these structures in transposed form.

  For FIR designs, you can choose direct form, direct-form transposed, direct-form symmetric, direct-form asymmetric structures, or an overlap and add structure.

- **Use a System object to implement filter** — Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Bandpass Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter format, such as the impulse response and the filter order.

Impulse response

Select `FIR` or `IIR` from the drop-down list, where `FIR` is the default impulse response. When you choose an impulse response, the design methods and structures you can use to implement your filter change accordingly.

Note The design methods and structures for FIR filters are not the same as the methods and structures for IIR filters.

Order mode

Select `Minimum` (the default) or `Specify` from the drop-down box. Selecting `Specify` enables the **Order** option so you can enter the filter order.

If you have the DSP System Toolbox software installed, you can specify IIR filters with different numerator and denominator orders. The default is equal orders. To specify a different denominator order, check the **Denominator order** box.

Filter type — This dialog only applies if you have the DSP System Toolbox software.

Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

Order

Enter the filter order. This option is enabled only if you select `Specify` for **Order mode**.

Decimation Factor

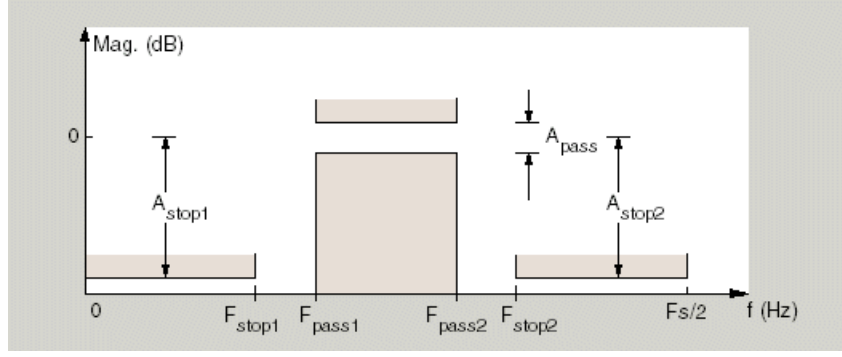
Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2.

Interpolation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Frequency specifications

The parameters in this group allow you to specify your filter response curve. Graphically, the filter specifications look similar to those shown in the following figure.



In the figure, regions between specification values such as Fstop1 and Fpass1 represent transition regions where the filter response is not explicitly defined.

Frequency constraints

Select the filter features to use to define the frequency response characteristics. This dialog applies only when **Order mode** is **Specify**.

- `Passband and stopband edges` — Define the filter by specifying the frequencies for the edges for the stop- and passbands.
- `Passband edges` — Define the filter by specifying frequencies for the edges of the passband.
- `Stopband edges` — Define the filter by specifying frequencies for the edges of the stopbands.
- `3dB points` — Define the filter response by specifying the locations of the 3 dB points (IIR filters). The 3-dB point is the frequency for the point 3 dB below the passband value.
- `3dB points and passband width` — Define the filter by specifying frequencies for the 3-dB points in the filter response and the width of the passband. (IIR filters)
- `3dB points and stopband widths` — Define the filter by specifying frequencies for the 3-dB points in the filter response and the width of the stopband. (IIR filters)
- `6dB points` — Define the filter response by specifying the locations of the 6-dB points. The 6-dB point is the frequency for the point 6dB below the passband value. (FIR filters)

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in hertz, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Fstop1

Enter the frequency at the edge of the end of the first stopband. Specify the value in either normalized frequency units or the absolute units you select in **Frequency units**.

Fpass1

Enter the frequency at the edge of the start of the passband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Fpass2

Enter the frequency at the edge of the end of the passband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Fstop2

Enter the frequency at the edge of the start of the second stopband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude constraints

Specify as `Unconstrained` or `Constrained bands`. You must have the DSP System Toolbox software to select `Constrained bands`. Selecting `Constrained bands` enables dialogs for both stopbands and the passband: **Astop1**, **Astop2**, and **Apass**. You cannot specify constraints for all three bands simultaneously.

Setting **Magnitude constraints** to `Constrained bands` enables the **Wstop** and **Wpass** options under **Design options**.

Magnitude units
Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in dB (decibels). This is the default setting.
- `Squared` — Specify the magnitude in squared units.

Astop1
Enter the filter attenuation in the first stopband in the units you choose for **Magnitude units**, either linear or decibels.

Apass
Enter the filter ripple allowed in the passband in the units you choose for **Magnitude units**, either linear or decibels.

Astop2
Enter the filter attenuation in the second stopband in the units you choose for **Magnitude units**, either linear or decibels.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method
Lists the design methods available for the frequency and magnitude specifications you entered. When you change the specifications for a filter, such as changing the impulse response, the methods available to design filters changes as well. The default IIR design method is usually Butterworth, and the default FIR method is equiripple.

Scale SOS filter coefficients to reduce chance of overflow
Selecting this parameter directs the design to scale the filter coefficients to reduce the chances that the inputs or calculations in the filter overflow and exceed the representable range of the filter. Clearing this option removes the scaling. This parameter applies only to IIR filters.

Design Options

The options for each design are specific for each design method. This section does not present all of the available options for all designs and design methods. There are many more that you encounter as you select different design methods and filter specifications. The following options represent some of the most common ones available.

Density factor
Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

Phase constraint
Valid when the **Design method** is `equiripple` and you have the DSP System Toolbox installed. Choose one of `Linear`, `Minimum`, or `Maximum`.

Minimum order
This option only applies when you have the DSP System Toolbox software and **Order mode** is `Minimum`.

Select `Any` (default), `Even`, or `Odd`. Selecting `Even` or `Odd` forces the minimum-order design to be an even or odd order.

Wstop1
Weight for the first stopband.

Wpass
Passband weight.

Wstop2
Weight for the second stopband.

Max pole radius
Valid only for IIR designs. Constrains the maximum pole radius. The default is 1. Reducing the max pole radius can produce a transfer function more resistant to quantization.

Init norm
Valid only for IIR designs. The initial norm used in the optimization. The default initial norm is 2.

Init numerator
Specifies an initial estimate of the filter numerator coefficients. This may be useful in difficult optimization problems.

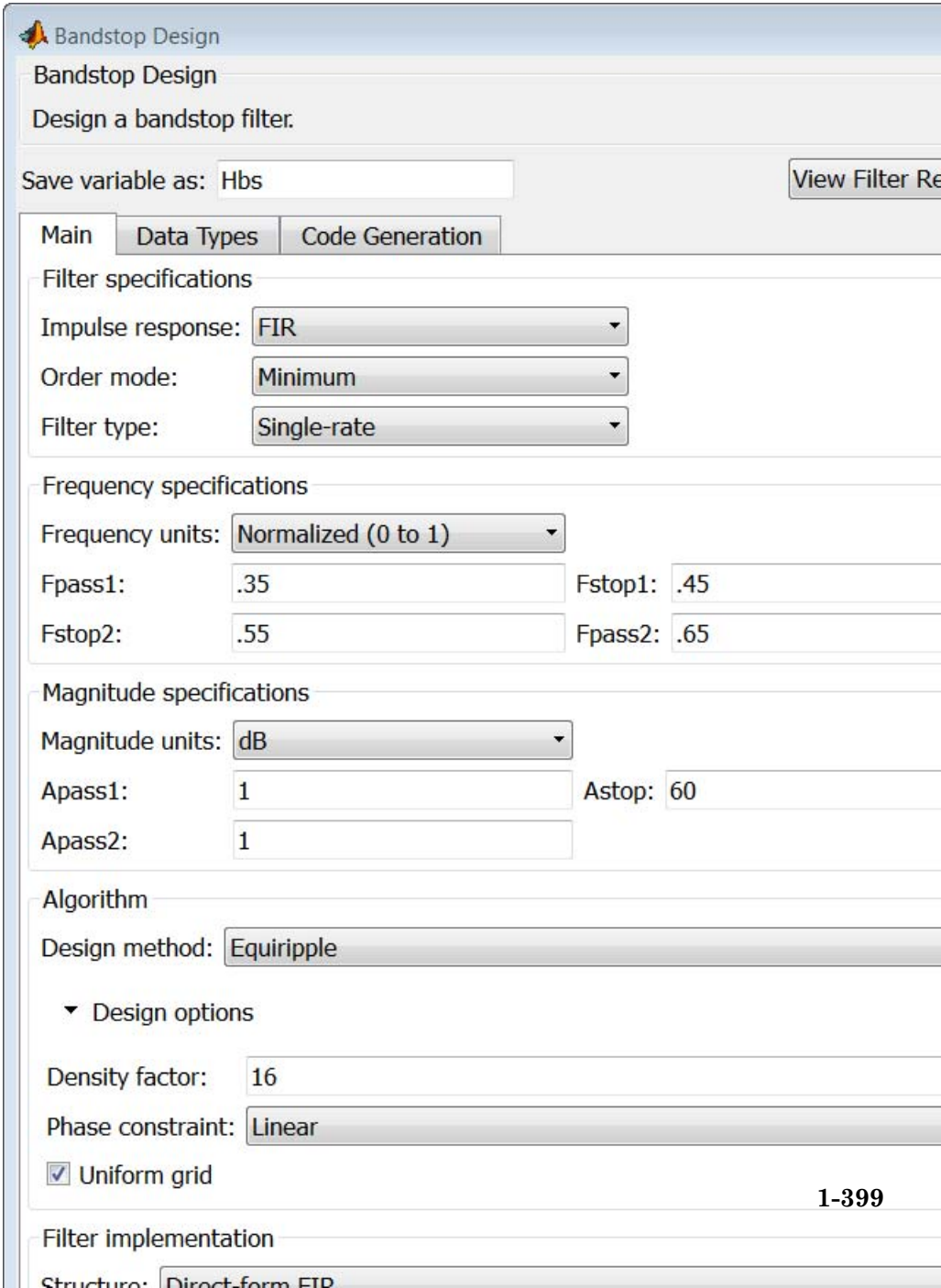
Init denominator
Specifies an initial estimate of the filter denominator coefficients. This may be useful in difficult optimization problems.

Filter implementation

Structure
For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure, and IIR filters use direct-form II filters with SOS.

Use a System object to implement filter
Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Bandstop Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter format, such as the impulse response and the filter order.

Impulse response

Select `FIR` or `IIR` from the drop-down list, where `FIR` is the default impulse response. When you choose an impulse response, the design methods and structures you can use to implement your filter change accordingly.

Note The design methods and structures for FIR filters are not the same as the methods and structures for IIR filters.

Order mode

Select `Minimum` (the default) or `Specify` from the drop-down list. Selecting `Specify` enables the **Order** option so you can enter the filter order.

If you have the DSP System Toolbox software installed, you can specify IIR filters with different numerator and denominator orders. The default is equal orders. To specify a different denominator order, check the **Denominator order** box.

Filter type

Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

When you design either a decimator or an interpolator, the resulting filter is a bandpass filter that either decimates or interpolates your input signal.

Order

Enter the filter order. This option is enabled only if `Specify` was selected for **Order mode**.

Decimation Factor

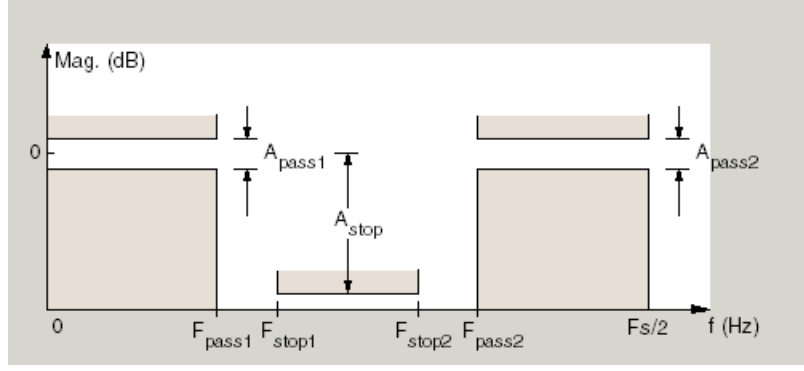
Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2.

Interpolation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Frequency specifications

The parameters in this group allow you to specify your filter response curve. Graphically, the filter specifications look similar to those shown in the following figure.



Frequency constraints

Select the filter features to use to define the frequency response characteristics. This dialog applies only when **Order mode** is **Specify**.

- `Passband and stopband edges` — Define the filter by specifying the frequencies for the edges for the stop- and passbands.
- `Passband edges` — Define the filter by specifying frequencies for the edges of the passband.
- `Stopband edges` — Define the filter by specifying frequencies for the edges of the stopbands.
- `3dB points` — Define the filter response by specifying the locations of the 3 dB points (IIR filters). The 3 dB point is the frequency for the point 3 dB point below the passband value.
- `3dB points and passband width` — Define the filter by specifying frequencies for the 3 dB points in the filter response and the width of the passband (IIR filters).
- `3dB points and stopband widths` — Define the filter by specifying frequencies for the 3 dB points in the filter response and the width of the stopband (IIR filters).
- `6dB points` — Define the filter response by specifying the locations of the 6-dB points (FIR filters). The 6-dB point is the frequency for the point 6 dB point below the passband value.

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Output Fs

When you design an interpolator, Fs represents the sampling frequency at the filter output rather than the filter input. This option is available only when you set **Filter type** is `interpolator`.

Fpass1

Enter the frequency at the edge of the end of the first passband. Specify the value in either normalized frequency units or the absolute units you select in **Frequency units**.

Fstop1

Enter the frequency at the edge of the start of the stopband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Fstop2

Enter the frequency at the edge of the end of the stopband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Fpass2

Enter the frequency at the edge of the start of the second passband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude constraints

Specify as `Unconstrained` or `Constrained bands`. You must have the DSP System Toolbox software to select `Constrained bands`. Selecting `Constrained bands` enables dialogs for both passbands and the stopband: **Apass1**, **Apass2**, and **Astop**. You cannot specify constraints for all three bands simultaneously.

Setting **Magnitude constraints** to `Constrained bands` enables the **Wstop** and **Wpass** options under **Design options**.

Magnitude units

Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in decibels (default).
- `Squared` — Specify the magnitude in squared units.

Apass1

Enter the filter ripple allowed in the first passband in the units you choose for **Magnitude units**, either linear or decibels.

Astop

Enter the filter attenuation in the stopband in the units you choose for **Magnitude units**, either linear or decibels

Apass2

Enter the filter ripple allowed in the second passband in the units you choose for **Magnitude units**, either linear or decibels

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

Lists the design methods available for the frequency and magnitude specifications you entered. When you change the

specifications for a filter, such as changing the impulse response, the methods available to design filters changes as well. The default IIR design method is usually Butterworth, and the default FIR method is equiripple.

Scale SOS filter coefficients to reduce chance of overflow
Selecting this parameter directs the design to scale the filter coefficients to reduce the chances that the inputs or calculations in the filter overflow and exceed the representable range of the filter. Clearing this option removes the scaling. This parameter applies only to IIR filters.

Design Options

The options for each design are specific for each design method. This section does not present all of the available options for all designs and design methods. There are many more that you encounter as you select different design methods and filter specifications. The following options represent some of the most common ones available.

Density factor
Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

Phase constraint
Valid when the **Design method** is `equiripple` and you have the DSP System Toolbox installed. Choose one of `Linear`, `Minimum`, or `Maximum`.

Minimum order
This option only applies when you have the DSP System Toolbox software and **Order mode** is `Minimum`.

Select `Any` (default), `Even`, or `Odd`. Selecting `Even` or `Odd` forces the minimum-order design to be an even or odd order.

Wpass1
Weight for the first passband.

Wstop
Stopband weight.

Wpass2
Weight for the second passband.

Match exactly
Specifies that the resulting filter design matches either the passband or stopband or both bands when you select `passband` or `stopband` .

Max pole radius
Valid only for IIR designs. Constrains the maximum pole radius. The default is 1. Reducing the max pole radius can produce a transfer function more resistant to quantization.

Init norm
Valid only for IIR designs. The initial norm used in the optimization. The default initial norm is 2.

Init numerator
Specifies an initial estimate of the filter numerator coefficients. This may be useful in difficult optimization problems.

Init denominator
Specifies an initial estimate of the filter denominator coefficients. This may be useful in difficult optimization problems.

Filter implementation

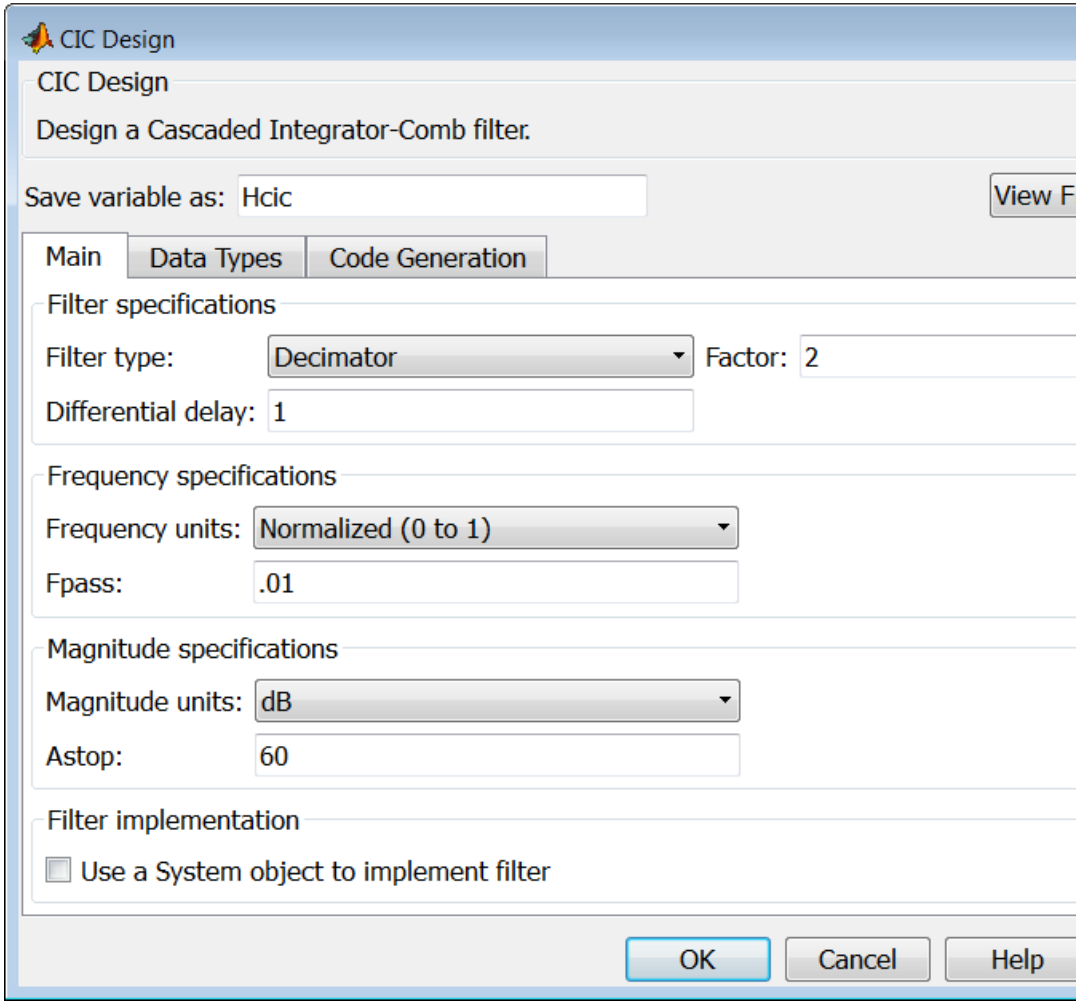
Structure

For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure, and IIR filters use direct-form II filters with SOS.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

CIC Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your CIC filter format, such as the filter type and the differential delay.

Filter type

Select whether your filter will be a `decimator` or an `interpolator`. Your choice determines the type of filter and the design methods and structures that are available to implement your filter. Selecting `decimator` or `interpolator` activates the **Factor** option. When you design an interpolator, you enable the **Output Fs** parameter.

When you design either a decimator or interpolator, the resulting filter is a CIC filter that decimates or interpolates your input signal.

Differential Delay

Specify the differential delay of your CIC filter as an integer value greater than or equal to 1. The default value is 1. The differential delay changes the shape, number, and location of nulls in the filter response. Increasing the differential delay increases the sharpness of the nulls and the response between the nulls. In practice, differential delay values of 1 or 2 are the most common.

Factor

Specify the decimation or interpolation factor for your filter as an integer value greater than or equal to 1. The default value is 2.

Frequency specifications

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Output Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter output. When you provide an output sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available only when you design interpolators.

Fpass

Enter the frequency at the end of the passband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Magnitude specifications

Magnitude units

Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in decibels (default).
- `Squared` — Specify the magnitude in squared units.

Astop

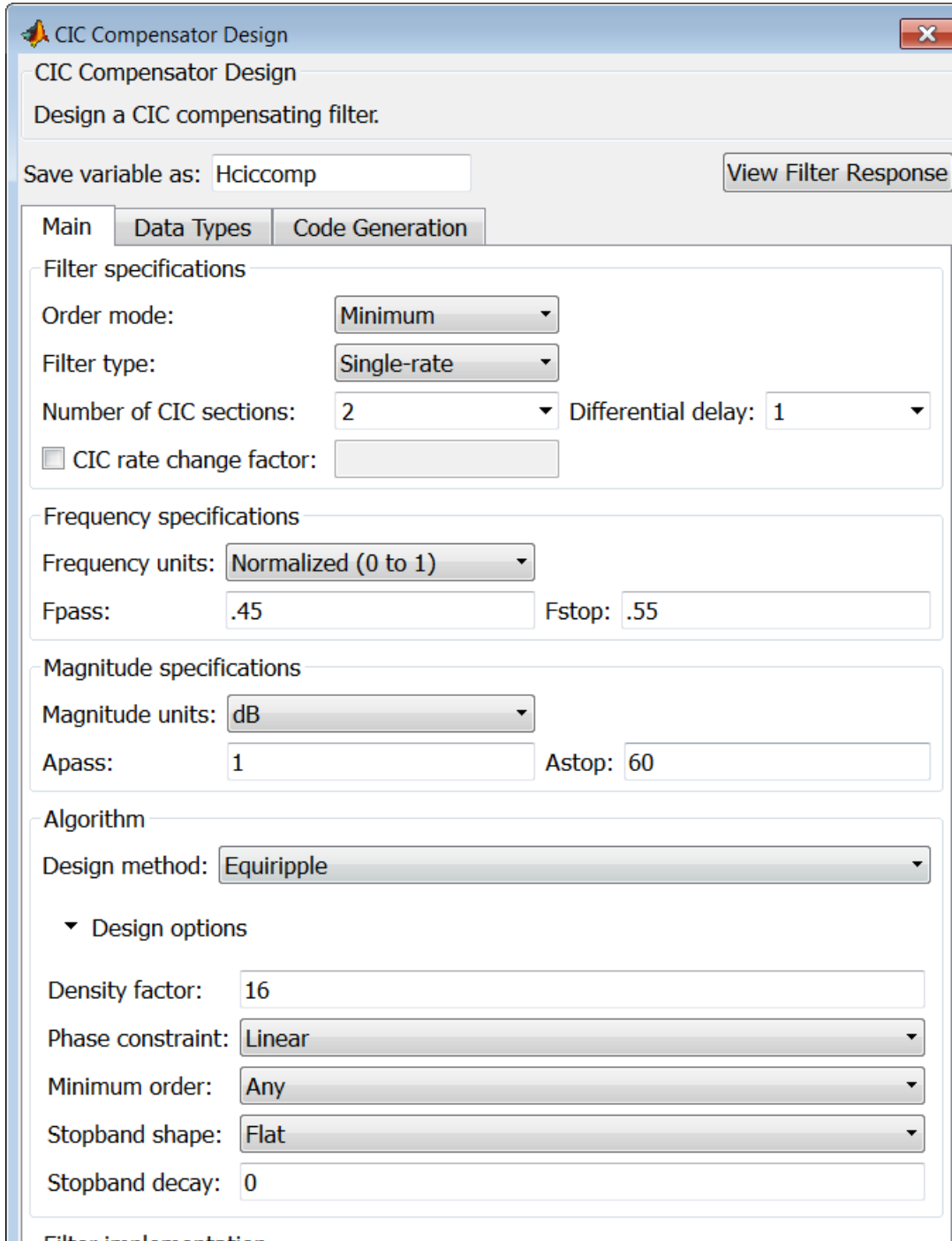
Enter the filter attenuation in the stopband in the units you choose for **Magnitude units**, either linear or decibels.

Filter implementation

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

CIC Compensator Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter format, such as the filter order mode and the filter type.

Order mode

Select `Minimum` (the default) or `Specify` from the drop-down list. Selecting `Specify` enables the **Order** option (see the following sections) so you can enter the filter order.

Filter type

Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

When you design either a decimator or an interpolator, the resulting filter is a bandpass filter that either decimates or interpolates your input signal.

Order

Enter the filter order. This option is enabled only if `Specify` was selected for **Order mode**.

Decimation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2.

Interpolation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Number of CIC sections

Specify the number of sections in the CIC filter for which you are designing this compensator. Select the number of sections from the drop-down list or enter the number.

Differential Delay

Specify the differential delay of your target CIC filter. The default value is `1`. Most CIC filters use 1 or 2.

Frequency specifications

The parameters in this group allow you to specify your filter response curve.

Frequency specifications

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Output Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter output. When you provide an output sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available only when you design interpolators.

Fpass
Enter the frequency at the end of the passband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Fstop
Enter the frequency at the start of the stopband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude units
Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in decibels (default).
- `Squared` — Specify the magnitude in squared units.

Apass
Enter the filter ripple allowed in the passband in the units you choose for **Magnitude units**, either linear or decibels

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method
Lists the design methods available for the frequency and magnitude specifications you entered. When you change the specifications for a filter, such as changing the impulse response, the methods available to design filters changes as well. The

default IIR design method is usually Butterworth, and the default FIR method is equiripple.

Design Options

The options for each design are specific for each design method. This section does not present all of the available options for all designs and design methods. There are many more that you encounter as you select different design methods and filter specifications. The following options represent some of the most common ones available.

Density factor

Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

Minimum phase

To design a filter that is minimum phase, select **Minimum phase**. Clearing the **Minimum phase** option removes the phase constraint—the resulting design is not minimum phase.

Minimum order

When you select this parameter, the design method determines and design the minimum order filter to meet your specifications. Some filters do not provide this parameter. Select `Any`, `Even`, or `Odd` from the drop-down list to direct the design to be any minimum order, or minimum even order, or minimum odd order.

Note Generally, **Minimum order** designs are not available for IIR filters.

Match exactly

Specifies that the resulting filter design matches either the passband or stopband or both bands when you select `passband` or `stopband` or `both` from the drop-down list.

Stopband Shape

Stopband shape lets you specify how the stopband changes with increasing frequency. Choose one of the following options:

- `Flat` — Specifies that the stopband is flat. The attenuation does not change as the frequency increases.
- `Linear` — Specifies that the stopband attenuation changes linearly as the frequency increases. Change the slope of the stopband by setting **Stopband decay**.
- `1/f` — Specifies that the stopband attenuation changes exponentially as the frequency increases, where `f` is the frequency. Set the power (exponent) for the decay in **Stopband decay**.

Stopband Decay

When you set Stopband shape, Stopband decay specifies the amount of decay applied to the stopband. the following conditions apply to Stopband decay based on the value of Stopband Shape:

- When you set **Stopband shape** to `Flat`, **Stopband decay** has no affect on the stopband.
- When you set **Stopband shape** to `Linear`, enter the slope of the stopband in units of dB/rad/s. `filterbuilder` applies that slope to the stopband.
- When you set **Stopband shape** to `1/f`, enter a value for the exponent n in the relation $(1/f)^n$ to define the stopband decay.

`filterbuilder` applies the $(1/f)^n$ relation to the stopband to result in an exponentially decreasing stopband attenuation.

Filter implementation

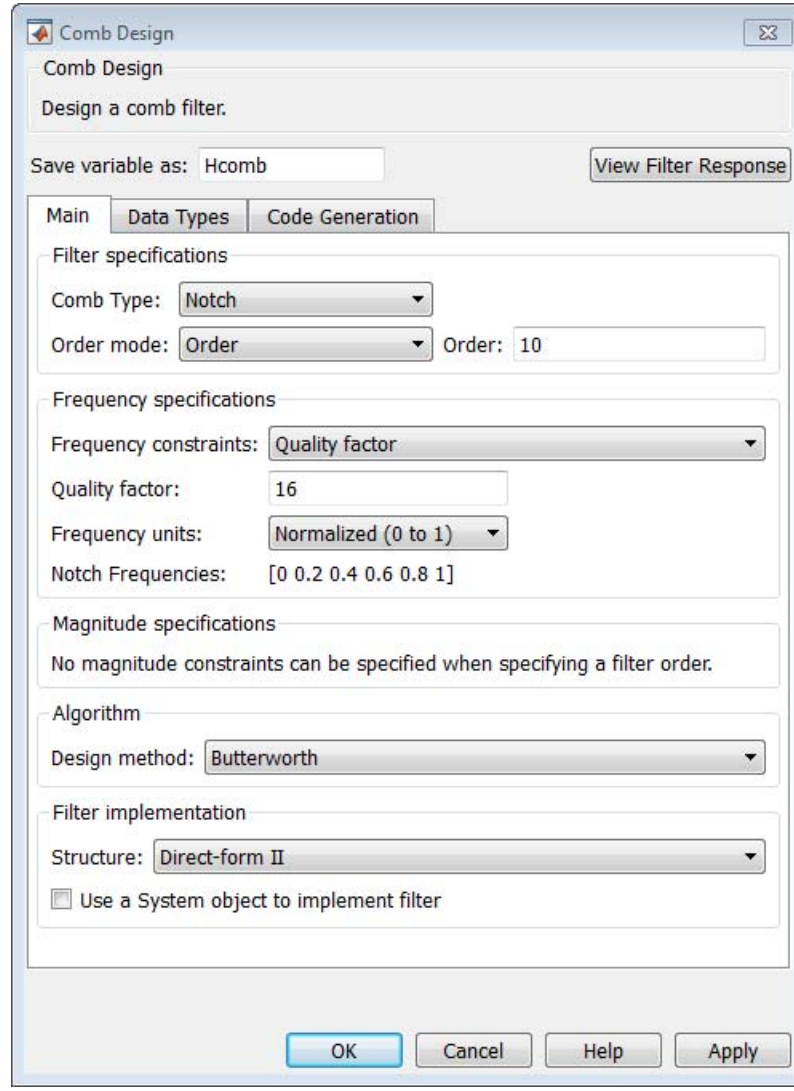
Structure

For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure, and IIR filters use direct-form II filters with SOS.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Comb Filter Design Dialog Box—Main Pane



Filter specifications

Parameters in this group enable you to specify the type of comb filter and the number of peaks or notches.

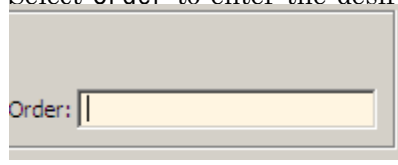
Comb Type

Select `Notch` or `Peak` from the drop-down list. `Notch` creates a comb filter that attenuates a set of harmonically related frequencies. `Peak` creates a comb filter that amplifies a set of harmonically related frequencies.

Order mode

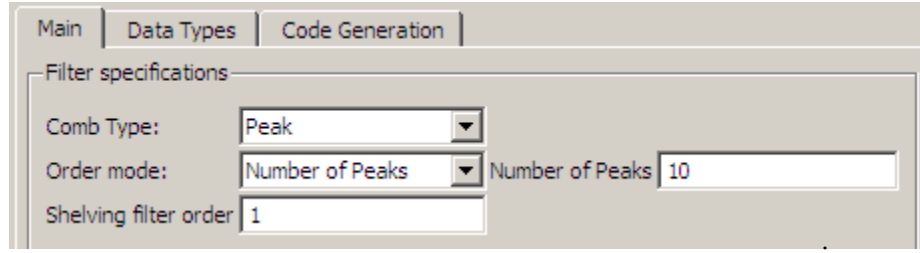
Select `Order` or `Number of Peaks/Notches` from the drop-down menu.

Select `Order` to enter the desired filter order in the



dialog box. The comb filter has notches or peaks at increments of 2/`Order` in normalized frequency units.

Select `Number of Peaks` or `Number of Notches` to specify the number of peaks or notches and the `Shelving filter order`



Shelving filter order

The `Shelving filter order` is a positive integer that determines the sharpness of the peaks or notches. Larger values result in sharper peaks or notches.

Frequency specifications

Parameters in this group enable you to specify the frequency constraints and frequency units.

Frequency specifications

Select `Quality factor` or `Bandwidth`.

`Quality factor` is the ratio of the center frequency of the peak or notch to the bandwidth calculated at the –3 dB point.

`Bandwidth` specifies the bandwidth of the peak or notch. By default the bandwidth is measured at the –3 dB point. For example, setting the bandwidth equal to 0.1 results in 3 dB frequencies at normalized frequencies 0.05 above and below the center frequency of the peak or notch.

Frequency Units

Specify the frequency units. The default is normalized frequency. Choosing an option in Hz enables the **Input Fs** dialog box.

Magnitude specifications

Specify the units for the magnitude specification and the gain at which the bandwidth is measured. This menu is disabled if you specify a filter order. Select one of the following magnitude units from the drop down list:

- `dB` — Specify the magnitude in decibels (default).
- `Squared` — Specify the magnitude in squared units.

Bandwidth gain — Specify the gain at which the bandwidth is measured. The default is –3 dB.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

The IIR Butterworth design is the only option for peaking or notching comb filters.

Filter implementation

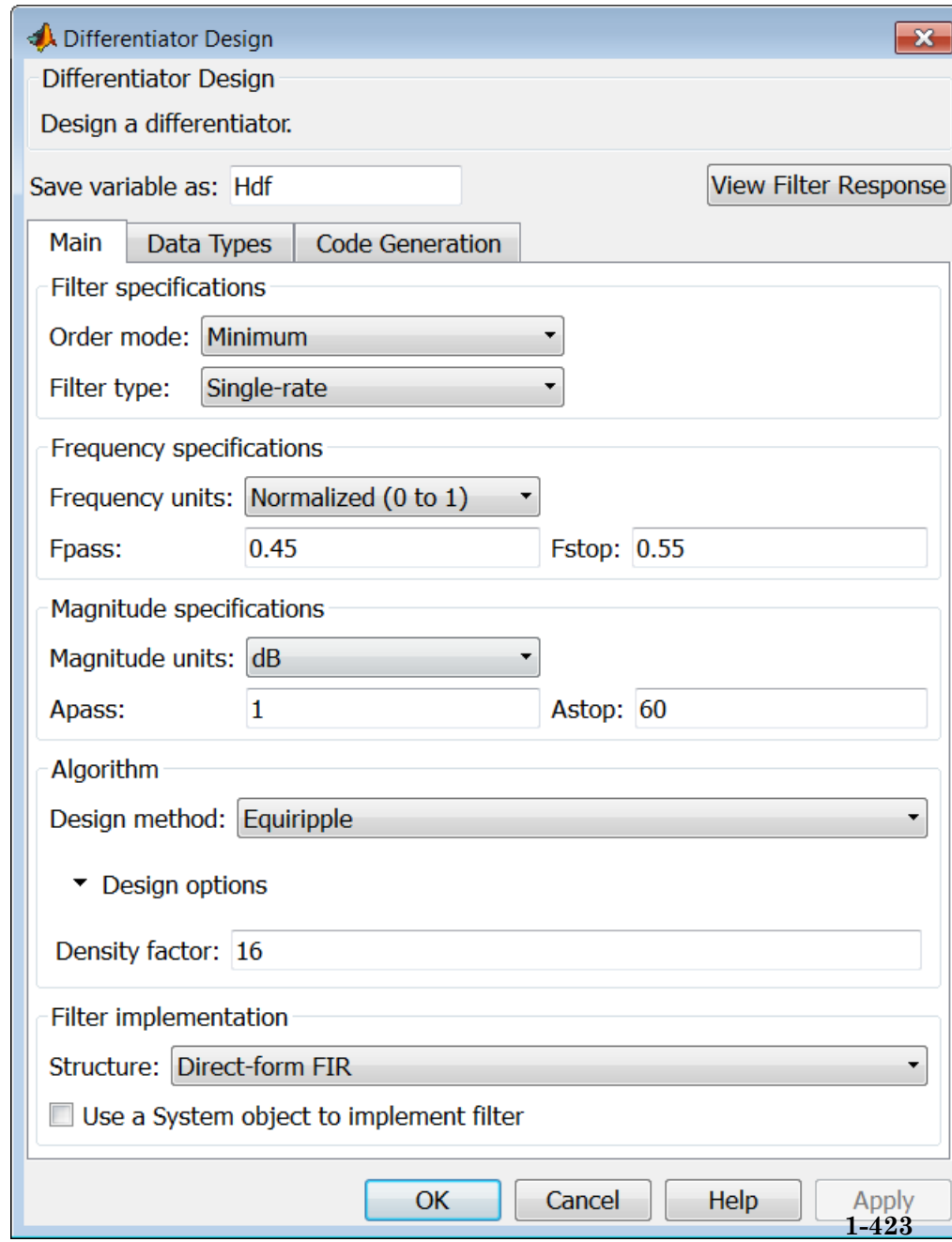
Structure

For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter.

Use a System object to implement filter

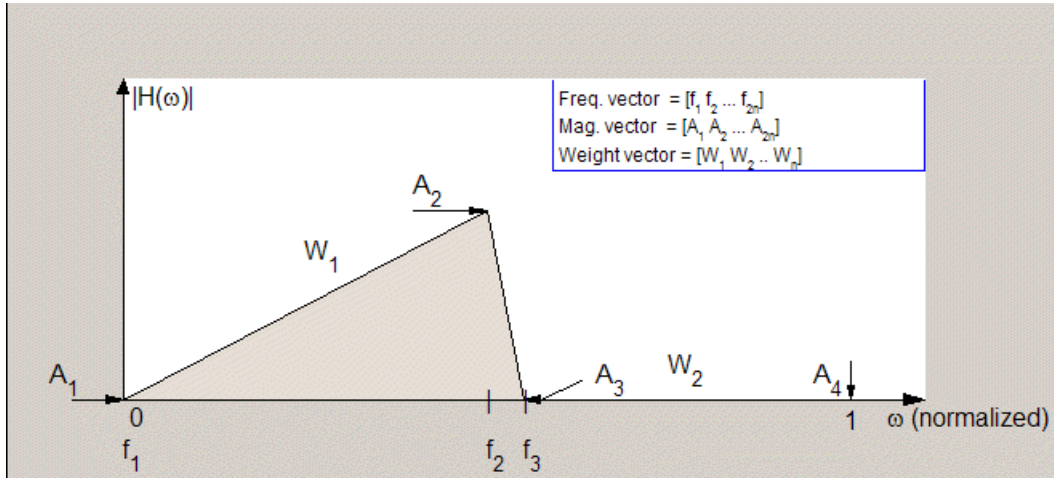
Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off.

Differentiator Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter format, such as the impulse response and the filter order. Graphically, the filter specifications look similar to those shown in the following figure.



In the figure, regions between specification values such as **Fpass** (f_1) and **Fstop** (f_3) represent transition regions where the filter response is not explicitly defined.

Order mode

Select `Minimum` (the default) or `Specify` from the drop-down list. Selecting `Specify` enables the **Order** option (see the following sections) so you can enter the filter order.

Filter type

Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

When you design either a decimator or an interpolator, the resulting filter is a bandpass filter that either decimates or interpolates your input signal.

Order

Enter the filter order. This option is enabled only if `Specify` was selected for **Order mode**.

Decimation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2.

Interpolation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Frequency specifications

The parameters in this group allow you to specify your filter response curve.

Frequency constraints

This option is only available when you specify the order of the filter design. Supported options are `Unconstrained` and `Passband edge and stopband edge`.

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the

frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Fpass

Enter the frequency at the end of the passband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Fstop

Enter the frequency at the start of the stopband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude constraints

This option is only available when you specify the order of your filter design. The options for **Magnitude constraints** depend on the value of the **Frequency constraints**. If the value of **Frequency constraints** is `Unconstrained`, **Magnitude constraints** must be `Unconstrained`. If the value of **Frequency constraints** is `Passband edge and stopband edge`, **Magnitude constraints** can be `Unconstrained`, `Passband ripple`, or `Stopband attenuation`.

Magnitude units

Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in decibels (default).
- `Squared` — Specify the magnitude in squared units.

Apass

Enter the filter ripple allowed in the passband in the units you choose for **Magnitude units**, either linear or decibels.

Astop2

Enter the filter attenuation in the second stopband in the units you choose for **Magnitude units**, either linear or decibels.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

Lists the design methods available for the frequency and magnitude specifications you entered. When you change the specifications for a filter, such as changing the impulse response, the methods available to design filters changes as well. The default IIR design method is usually Butterworth, and the default FIR method is equiripple.

Scale SOS filter coefficients to reduce chance of overflow

Selecting this parameter directs the design to scale the filter coefficients to reduce the chances that the inputs or calculations in the filter overflow and exceed the representable range of the filter. Clearing this option removes the scaling. This parameter applies only to IIR filters.

Design Options

The options for each design are specific for each design method. This section does not present all of the available options for all designs and design methods. There are many more that you encounter as you select different design methods and filter specifications.

Density factor

Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

Wpass

Passband weight. This option is only available for a specified-order design when **Frequency constraints** is equal to `Passband edge and stopband edge` and the **Design method** is `Equiripple`.

Wstop

Stopband weight. This option is only available for a specified-order design when **Frequency constraints** is equal to `Passband edge and stopband edge` and the **Design method** is `Equiripple`.

Filter implementation

Structure

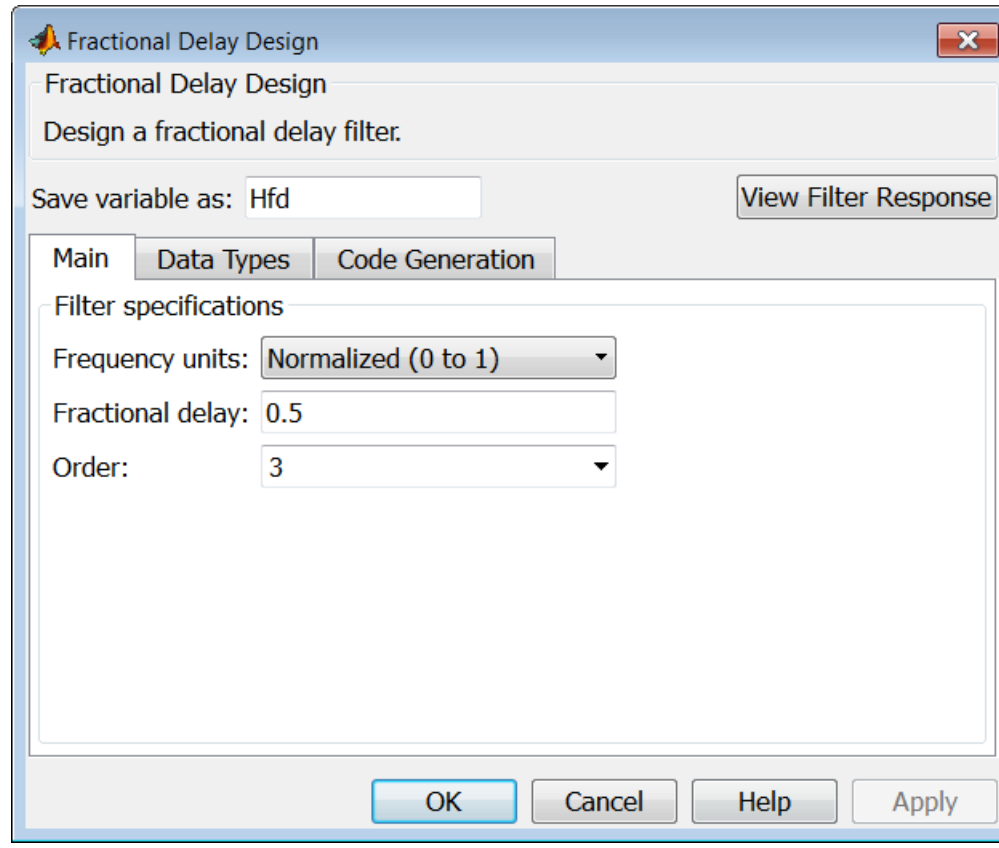
For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure, and IIR filters use direct-form II filters with SOS.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned

off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Fractional Delay Filter Design Dialog Box — Main Pane



Frequency specifications

Parameters in this group enable you to specify your filter format, such as the fractional delay and the filter order.

Order

If you choose `Specify` for **Order mode**, enter your filter order in this field, or select the order from the drop-down list.`filterbuilder` designs a filter with the order you specify.

Fractional delay

Specify a value between 0 and 1 samples for the filter fractional delay. The default value is `0.5` samples.

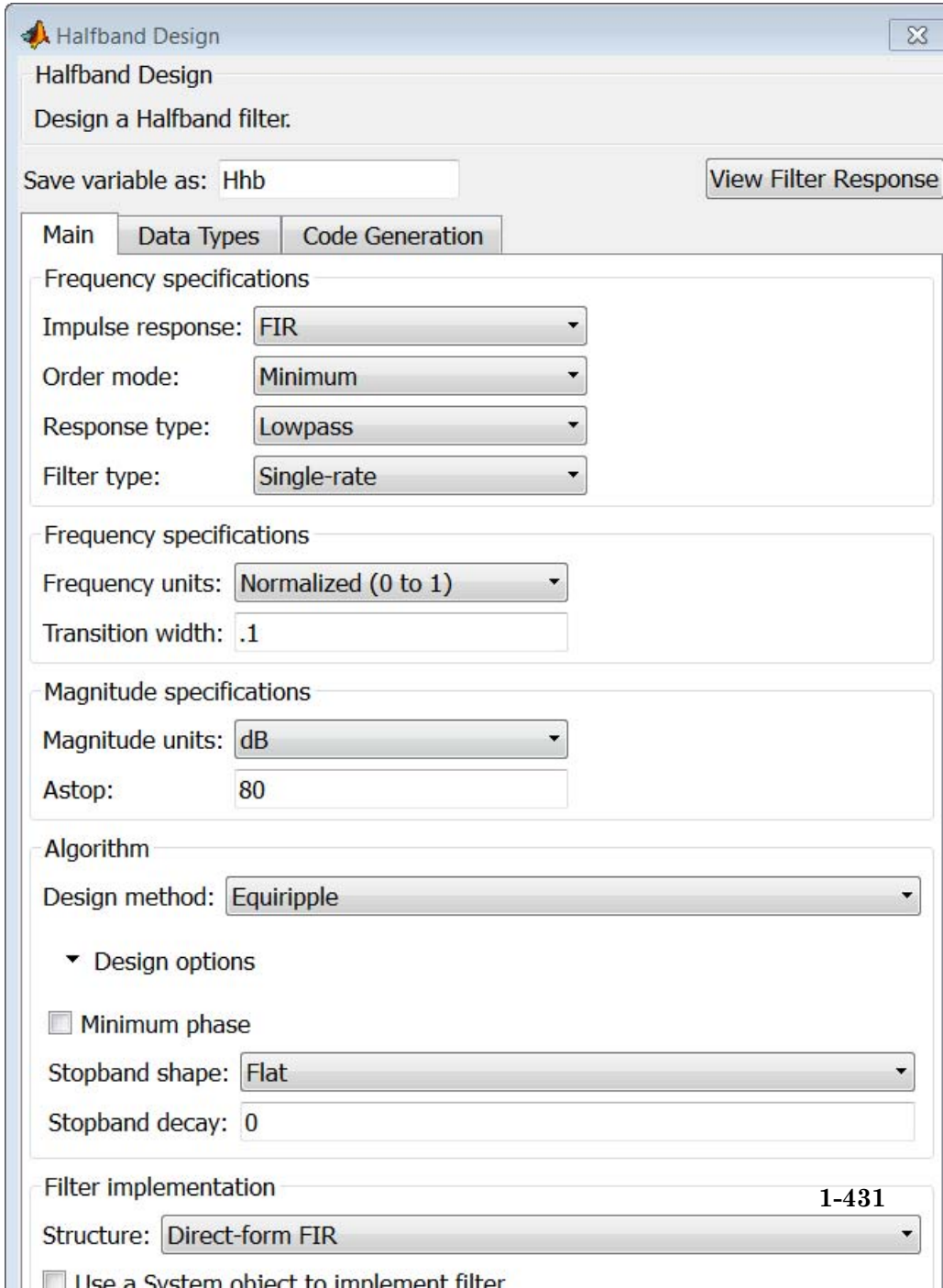
Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Halfband Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter type and order.

Impulse response
Select `FIR` or `IIR` from the drop-down list, where `FIR` is the default impulse response. When you choose an impulse response, the design methods and structures you can use to implement your filter change accordingly.

Note The design methods and structures for FIR filters are not the same as the methods and structures for IIR filters.

Order mode
Select `Minimum` (the default) or `Specify` from the drop-down list. Selecting `Specify` enables the **Order** option (see the following sections) so you can enter the filter order.

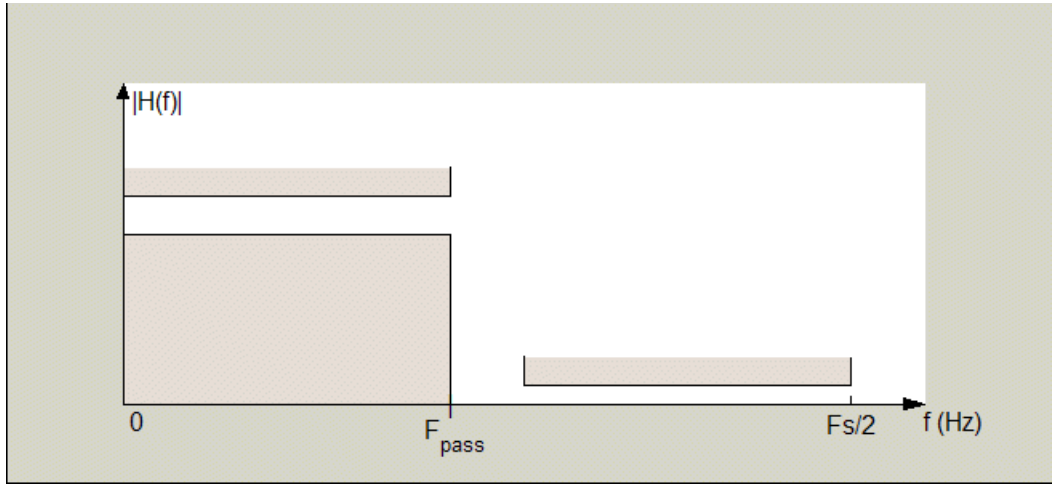
Filter type
Select `Single-rate`, `Decimator`, or `Interpolator`. By default, `filterbuilder` specifies single-rate filters.

When you design either a decimator or an interpolator, the resulting filter is a bandpass filter that decimates or interpolates your input by a factor of two.

Order
Enter the filter order. This option is enabled only if `Specify` was selected for **Order mode**.

Frequency specifications

The parameters in this group allow you to specify your filter response curve. Graphically, the filter specifications for a halfband lowpass filter look similar to those shown in the following figure.



In the figure, the transition region lies between the end of the passband and the start of the stopband. The width is defined explicitly by the value of **Transition width**.

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Transition width

Specify the width of the transition between the end of the passband and the edge of the stopband. Specify the value in normalized frequency units or the absolute units you select in **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude units

Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in decibels (default).

Astop

Enter the filter attenuation in the stopband in the units you choose for **Magnitude units**, either linear or decibels.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

Lists the design methods available for the frequency and magnitude specifications you entered. For FIR halfband filters, the available design options are `Equiripple` and `Kaiser window`. For IIR halfband filters, the available design options are `Butterworth`, `Elliptic`, and `IIR quasi-linear phase`.

Design Options

The following design options are available for FIR halfband filters when the user specifies an equiripple design:

Minimum phase

To design a filter that is minimum phase, select **Minimum phase**. Clearing the **Minimum phase** option removes the phase constraint—the resulting design is not minimum phase.

Stopband Shape

Stopband shape lets you specify how the stopband changes with increasing frequency. Choose one of the following options:

- `Flat` — Specifies that the stopband is flat. The attenuation does not change as the frequency increases.
- `Linear` — Specifies that the stopband attenuation changes linearly as the frequency increases. Change the slope of the stopband by setting **Stopband decay**.
- `1/f` — Specifies that the stopband attenuation changes exponentially as the frequency increases, where `f` is the frequency. Set the power (exponent) for the decay in **Stopband decay**.

Stopband Decay

When you set Stopband shape, Stopband decay specifies the amount of decay applied to the stopband. the following conditions apply to Stopband decay based on the value of Stopband Shape:

- When you set **Stopband shape** to `Flat`, **Stopband decay** has no affect on the stopband.
- When you set **Stopband shape** to `Linear`, enter the slope of the stopband in units of dB/rad/s. `filterbuilder` applies that slope to the stopband.
- When you set **Stopband shape** to `1/f`, enter a value for the exponent n in the relation $(1/f)^n$ to define the stopband decay. `filterbuilder` applies the $(1/f)^n$ relation to the stopband to result in an exponentially decreasing stopband attenuation.

Filter implementation

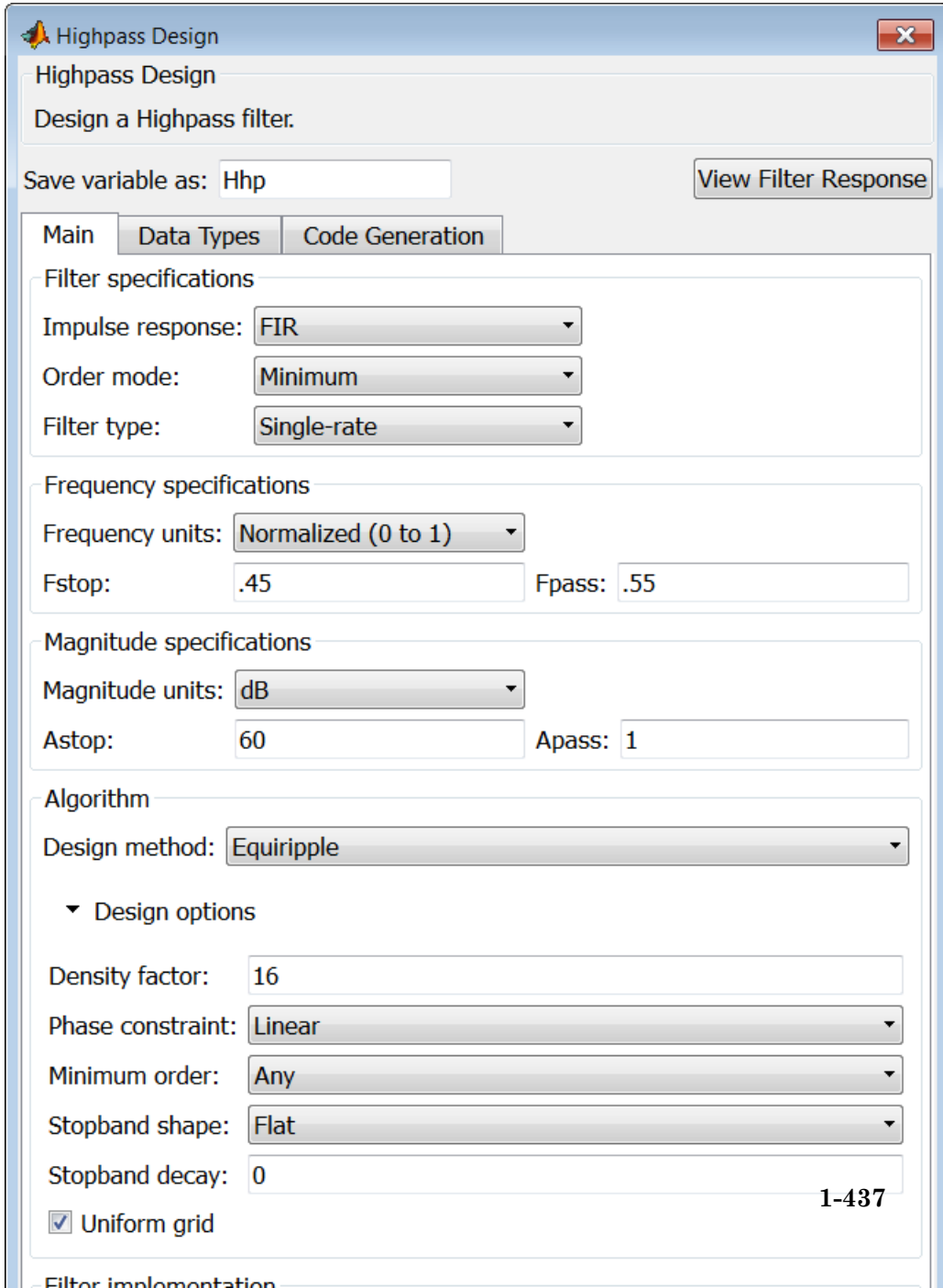
Structure

For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Highpass Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter format, such as the impulse response and the filter order.

Impulse response

Select `FIR` or `IIR` from the drop-down list, where `FIR` is the default impulse response. When you choose an impulse response, the design methods and structures you can use to implement your filter change accordingly.

Note The design methods and structures for FIR filters are not the same as the methods and structures for IIR filters.

Order mode

Select `Minimum` (the default) or `Specify` from the drop-down list. Selecting `Specify` enables the **Order** option so you can enter the filter order.

If your **Impulse response** is `IIR`, you can specify an equal order for the numerator and denominator, or different numerator and denominator orders. The default is equal orders. To specify a different denominator order, check the **Denominator order** box.

Filter type

This option is only available if you have the DSP System Toolbox software. Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

When you design either a decimator or an interpolator, the resulting filter is a highpass filter that either decimates or interpolates your input signal.

Order

Enter the filter order. This option is enabled only if `Specify` was selected for **Order mode**.

Decimation Factor

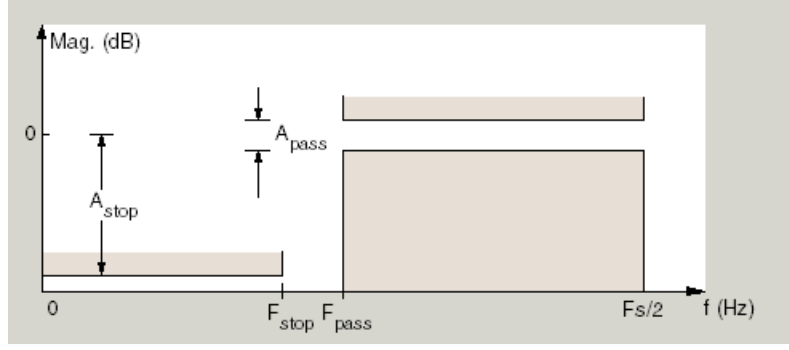
Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2.

Interpolation Factor

Enter the interpolation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Frequency specifications

The parameters in this group allow you to specify your filter response curve. Graphically, the filter specifications look similar to those shown in the following figure.



In the figure, the region between specification values Fstop and Fpass represents the transition region where the filter response is not explicitly defined.

Frequency constraints

Select the filter features to use to define the frequency response characteristics. The list contains the following options, when available for the filter specifications.

- `Stopband edge and passband edge` — Define the filter by specifying the frequencies for the edges for the stopband and passband.
- `Passband edge` — Define the filter by specifying the frequency for the edge of the passband.
- `Stopband edge` — Define the filter by specifying the frequency for the edges of the stopband.
- `Stopband edge and 3dB point` — Define the filter by specifying the stopband edge frequency and the 3-dB down point (IIR designs).
- `3dB point and passband edge` — Define the filter by specifying the 3-dB down point and passband edge frequency (IIR designs).
- `3dB point` — Define the filter by specifying the frequency for the 3-dB point (IIR designs or maxflat FIR).
- `6dB point` — Define the filter by specifying the frequency for the 6-dB point in the filter response (FIR designs).

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (O 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the

specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Fpass
Enter the frequency at the of the passband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Fstop
Enter the frequency at the start of the stopband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude units
Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in decibels (default).
- `Squared` — Specify the magnitude in squared units.

Astop
Enter the filter attenuation in the stopband in the units you choose for **Magnitude units**, either linear or decibels.

Apass
Enter the filter ripple allowed in the passband in the units you choose for **Magnitude units**, either linear or decibels.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

Lists the design methods available for the frequency and magnitude specifications you entered. When you change the specifications for a filter, such as changing the impulse response, the methods available to design filters changes as well. The default IIR design method is usually Butterworth, and the default FIR method is equiripple.

Scale SOS filter coefficients to reduce chance of overflow

Selecting this parameter directs the design to scale the filter coefficients to reduce the chances that the inputs or calculations in the filter overflow and exceed the representable range of the filter. Clearing this option removes the scaling. This parameter applies only to IIR filters.

Design Options

The options for each design are specific for each design method. This section does not present all of the available options for all designs and design methods. There are many more that you encounter as you select different design methods and filter specifications. The following options represent some of the most common ones available.

Density factor

Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a

reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

Phase constraint
This option only applies when you have the DSP System Toolbox software and when the **Design method** is `equiripple`. Select one of `Linear`, `Minimum`, or `Maximum`.

Minimum order — This option only applies when you have the DSP System Toolbox software and the **Order mode** is `Minimum`.
Select `Any` (default), `Even`, or `Odd`. Selecting `Even` or `Odd` forces the minimum-order design to be an even or odd order.

Match Exactly
Specifies that the resulting filter design matches either the passband or stopband when you select `Passband` or `Stopband`.

Stopband Shape
Stopband shape lets you specify how the stopband changes with increasing frequency. Choose one of the following options:

- `Flat` — Specifies that the stopband is flat. The attenuation does not change as the frequency increases.
- `Linear` — Specifies that the stopband attenuation changes linearly as the frequency increases. Change the slope of the stopband by setting **Stopband decay**.
- `1/f` — Specifies that the stopband attenuation changes exponentially as the frequency increases, where `f` is the frequency. Set the power (exponent) for the decay in **Stopband decay**.

Stopband Decay
When you set Stopband shape, Stopband decay specifies the amount of decay applied to the stopband. the following conditions apply to Stopband decay based on the value of Stopband Shape:

- When you set **Stopband shape** to `Flat`, **Stopband decay** has no affect on the stopband.

- When you set **Stopband shape** to `Linear`, enter the slope of the stopband in units of dB/rad/s. `filterbuilder` applies that slope to the stopband.
- When you set **Stopband shape** to `1/f`, enter a value for the exponent n in the relation $(1/f)^n$ to define the stopband decay. `filterbuilder` applies the $(1/f)^n$ relation to the stopband to result in an exponentially decreasing stopband attenuation.

Wpass

Passband weight. This option only applies when **Impulse response** is `FIR` and **Order mode** is `Specify`.

Wstop

Stopband weight. This option only applies when **Impulse response** is `FIR` and **Order mode** is `Specify`.

Filter implementation

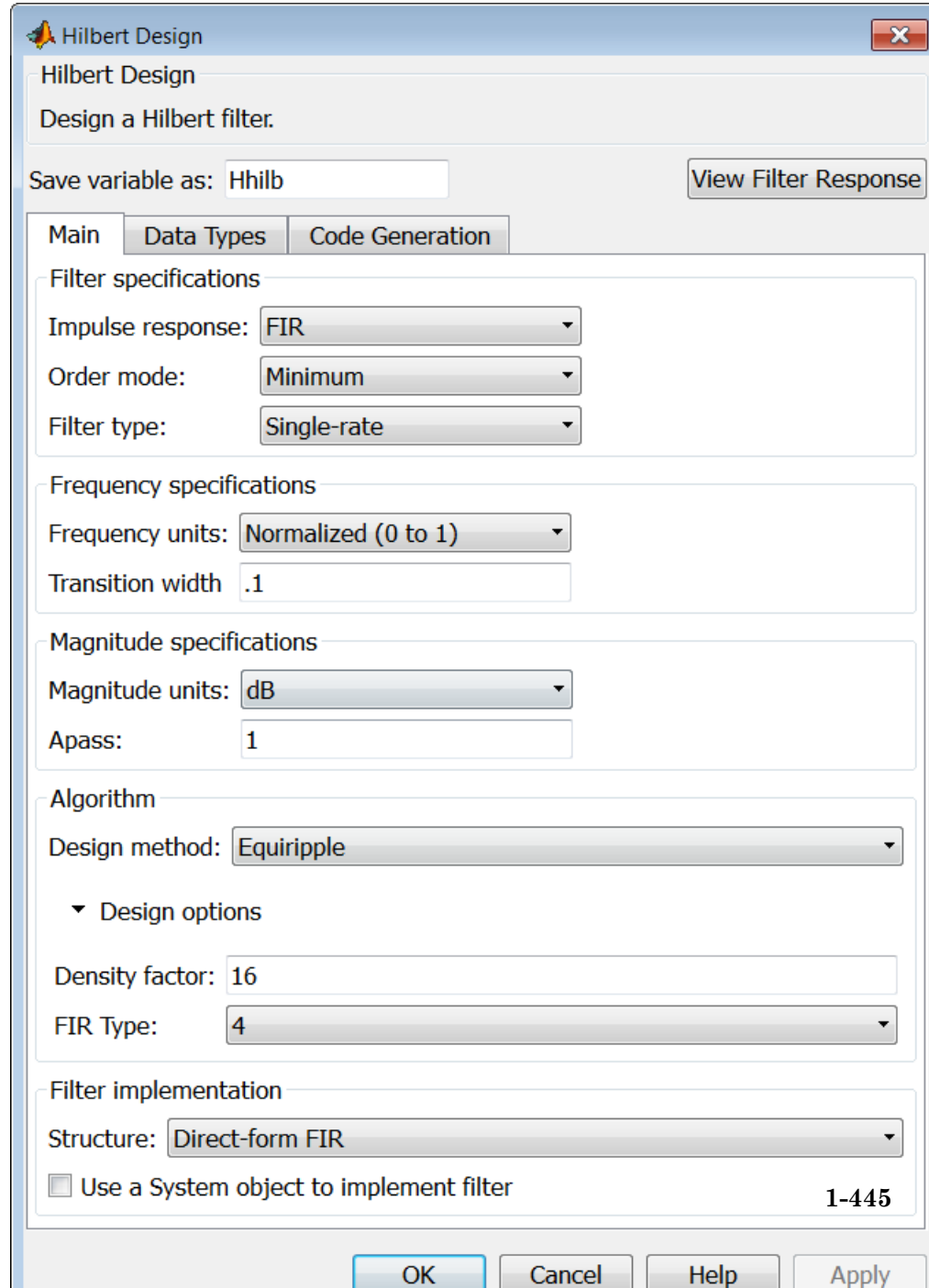
Structure

For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure, and IIR filters use direct-form II filters with SOS.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Hilbert Filter Design Dialog Box — Main Pane



Impulse response

Select `FIR` or `IIR` from the drop-down list, where `FIR` is the default impulse response. When you choose an impulse response, the design methods and structures you can use to implement your filter change accordingly.

Note The design methods and structures for FIR filters are not the same as the methods and structures for IIR filters.

Order mode

This option is only available if you have the DSP System Toolbox software. Select either `Minimum` (the default) or `Specify` from the drop-down list. Selecting `Specify` enables the **Order** option (see the following sections) so you can enter the filter order.

Filter type

This option is only available if you have the DSP System Toolbox software. Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

When you design either a decimator or an interpolator, the resulting filter is a bandpass filter that either decimates or interpolates your input signal.

Order

Enter the filter order. This option is enabled only if `Specify` was selected for **Order mode**.

Decimation Factor

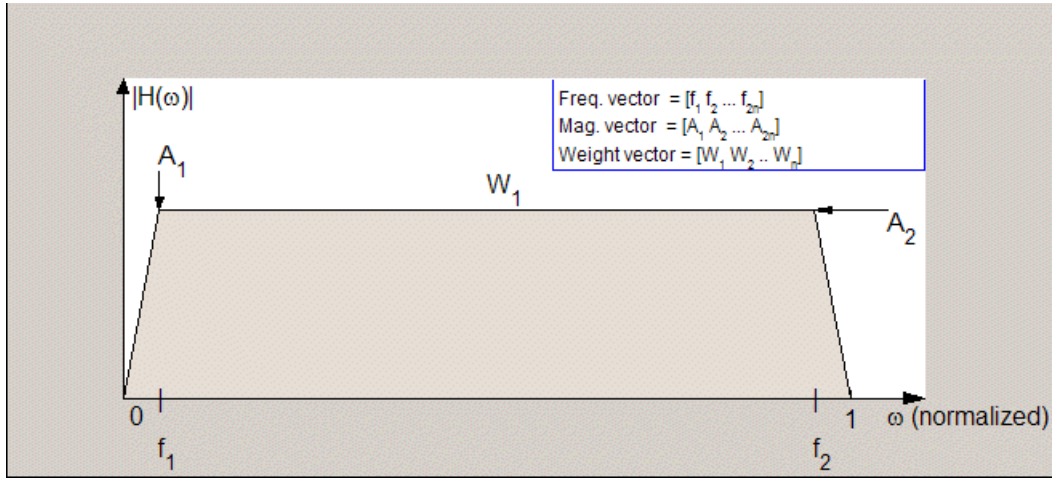
Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2.

Interpolation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Frequency specifications

The parameters in this group allow you to specify your filter response curve. Graphically, the filter specifications look similar to those shown in the following figure.



In the figure, the regions between 0 and f_1 and between f_2 and 1 represent the transition regions where the filter response is explicitly defined by the transition width.

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Transition width

Specify the width of the transitions at the ends of the passband. Specify the value in normalized frequency units or the absolute units you select in **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude units

Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in decibels (default)
- `Squared` — Specify the magnitude in squared units.

Apass

Enter the filter ripple allowed in the passband in the units you choose for **Magnitude units**, either linear or decibels.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

Lists the design methods available for the frequency and magnitude specifications you entered. When you change the specifications for a filter, such as changing the impulse response, the methods available to design filters changes as well. The default IIR design method is usually Butterworth, and the default FIR method is equiripple.

Scale SOS filter coefficients to reduce chance of overflow

Selecting this parameter directs the design to scale the filter coefficients to reduce the chances that the inputs or calculations in the filter overflow and exceed the representable range of the filter. Clearing this option removes the scaling. This parameter applies only to IIR filters.

Design Options

The options for each design are specific for each design method. This section does not present all of the available options for all designs and design methods. There are many more that you encounter as you select different design methods and filter specifications. The following options represent some of the most common ones available.

Density factor

Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a

reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

FIR Type

This option is only available in a minimum-order design. Specify whether to design a type 3 or a type 4 FIR filter. The filter type is defined as follows:

- Type 3 — FIR filter with even order antisymmetric coefficients
- Type 4 — FIR filter with odd order antisymmetric coefficients

Select `3` or `4` from the drop-down list.

Filter implementation

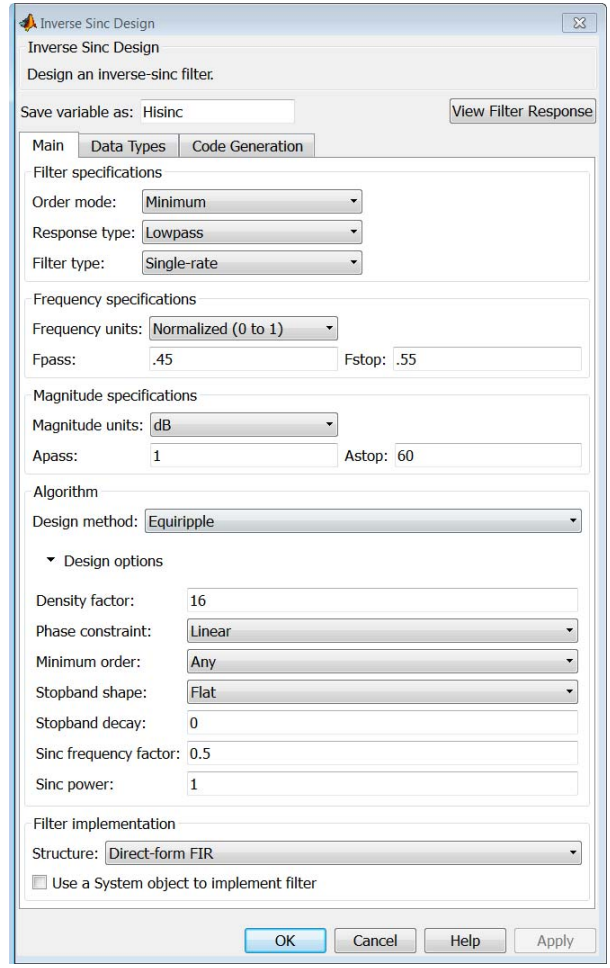
Structure

For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure, and IIR filters use direct-form II filters with SOS.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Inverse Sinc Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter format, such as the impulse response and the filter order.

Order mode

Select `Minimum` (the default) or `Specify` from the drop-down list. Selecting `Specify` enables the **Order** option (see the following sections) so you can enter the filter order.

Response type

Select `Lowpass` or `Highpass` to design an inverse sinc lowpass or highpass filter.

Filter type

Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

When you design either a decimator or an interpolator, the resulting filter is a bandpass filter that either decimates or interpolates your input signal.

Order

Enter the filter order. This option is enabled only if `Specify` was selected for **Order mode**.

Decimation Factor

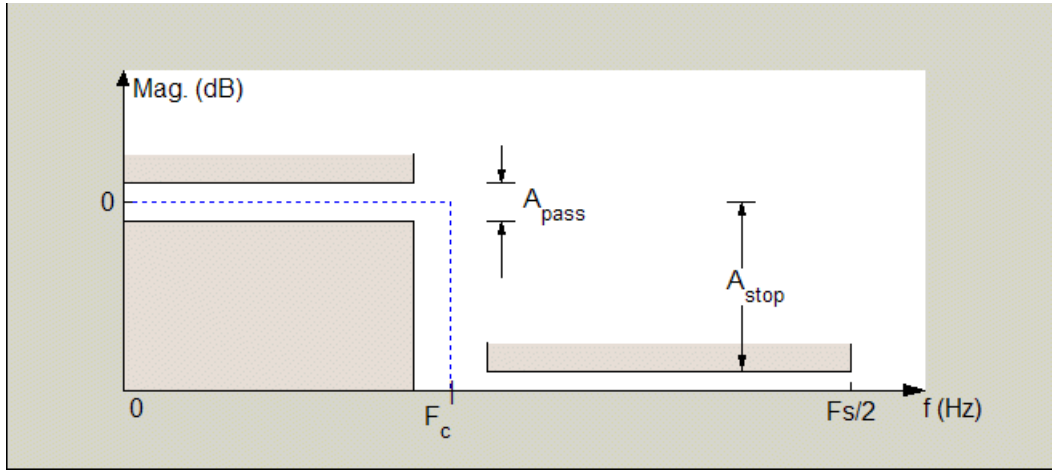
Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2.

Interpolation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Frequency specifications

The parameters in this group allow you to specify your filter response curve. Graphically, the filter specifications look similar to those shown in the following figure.



Regions between specification values such as Fpass and Fstop represent transition regions where the filter response is not explicitly defined.

Frequency constraints

This option is only available when you specify the filter order. The following options are available:

- `Passband and stopband edges` — Define the filter by specifying the frequencies for the edges for the stop- and passbands.
- `Passband edge` — Define the filter by specifying frequencies for the edges of the passband.
- `Stopband edge` — Define the filter by specifying frequencies for the edges of the stopbands.

- `6dB point` — The 6-dB point is the frequency for the point 6 dB point below the passband value.

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Fpass

Enter the frequency at the end of the passband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Fstop

Enter the frequency at the start of the stopband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude units

Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.

- `dB` — Specify the magnitude in decibels (default)
- `Squared` — Specify the magnitude in squared units.

Apass

Enter the filter ripple allowed in the passband in the units you choose for **Magnitude units**, either linear or decibels.

Astop

Enter the filter attenuation in the stopband in the units you choose for **Magnitude units**, either linear or decibels.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

Lists the design methods available for the frequency and magnitude specifications you entered. When you change the specifications for a filter, such as changing the impulse response, the methods available to design filters changes as well. The default IIR design method is usually Butterworth, and the default FIR method is equiripple.

Design Options

The options for each design are specific for each design method. This section does not present all of the available options for all designs and design methods. There are many more that you encounter as you select different design methods and filter specifications. The following options represent some of the most common ones available.

Density factor

Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

Phase constraint

Available options are `Linear`, `Minimum`, and `Maximum`.

Stopband Shape

Stopband shape lets you specify how the stopband changes with increasing frequency. Choose one of the following options;

- `Flat` — Specifies that the stopband is flat. The attenuation does not change as the frequency increases.
- `Linear` — Specifies that the stopband attenuation changes linearly as the frequency increases. Change the slope of the stopband by setting **Stopband decay**.
- `1/f` — Specifies that the stopband attenuation changes exponentially as the frequency increases, where `f` is the frequency. Set the power (exponent) for the decay in **Stopband decay**.

Stopband Decay

When you set Stopband shape, Stopband decay specifies the amount of decay applied to the stopband. the following conditions apply to Stopband decay based on the value of Stopband Shape:

- When you set **Stopband shape** to `Flat`, **Stopband decay** has no affect on the stopband.
- When you set **Stopband shape** to `Linear`, enter the slope of the stopband in units of dB/rad/s. `filterbuilder` applies that slope to the stopband.
- When you set **Stopband shape** to `1/f`, enter a value for the exponent n in the relation $(1/f)^n$ to define the stopband decay. `filterbuilder` applies the $(1/f)^n$ relation to the stopband to result in an exponentially decreasing stopband attenuation.

Sinc frequency factor

A frequency dilation factor. The sinc frequency factor, *C* , parameterizes the passband magnitude response for a lowpass design through H(ω) = sinc(Cω)^(-P) and for a highpass design through H(ω) = sinc(C(1-ω))^(-P).

Sinc power

Negative power of passband magnitude response. The sinc power, *P*, parameterizes the passband magnitude response for a lowpass design through H(ω) = sinc(Cω)^(-P) and for a highpass design through H(ω) = sinc(C(1-ω))^(-P).

Filter implementation

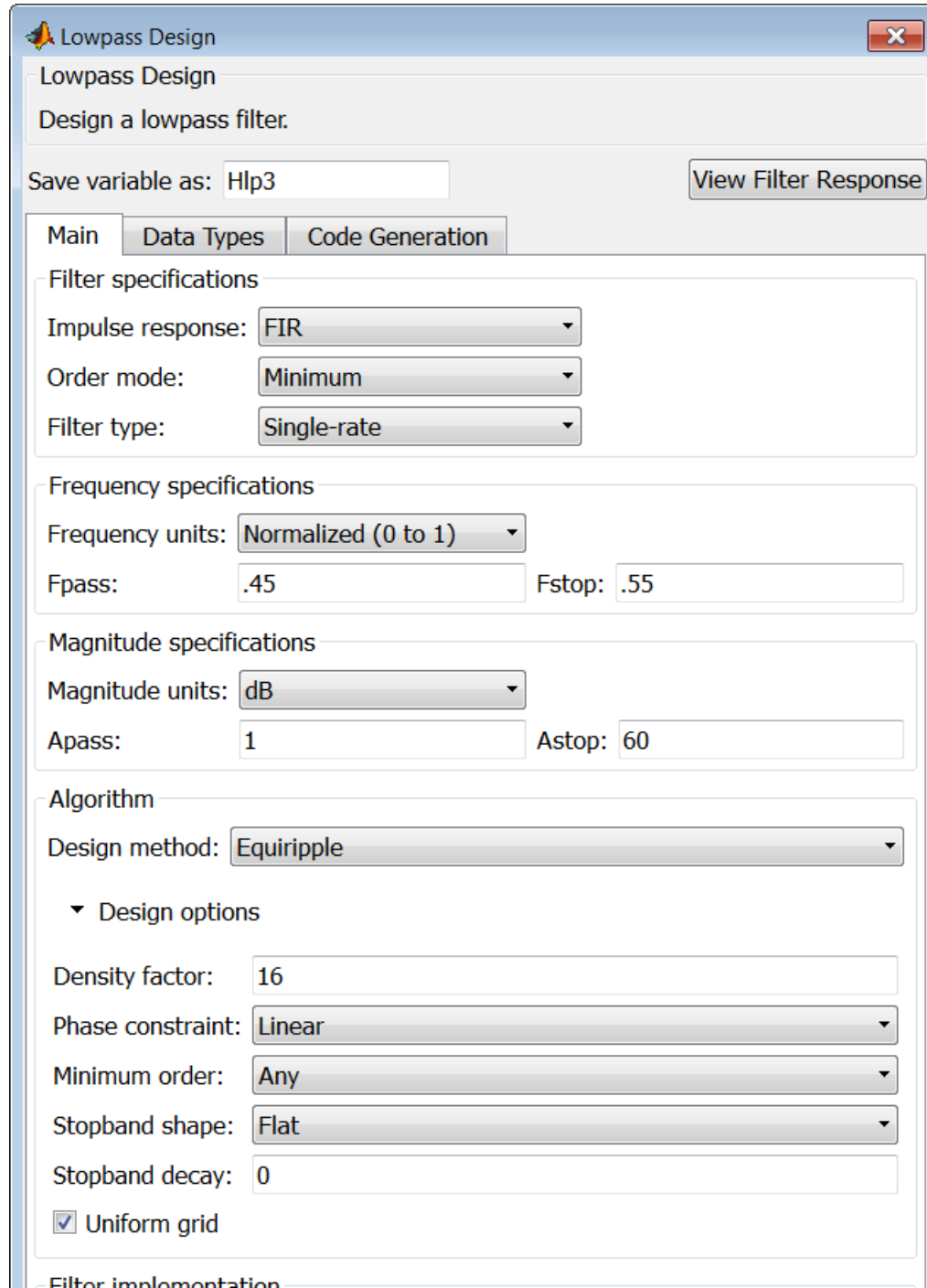
Structure

For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Lowpass Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter format, such as the impulse response and the filter order.

Impulse response

Select `FIR` or `IIR` from the drop-down list, where `FIR` is the default impulse response. When you choose an impulse response, the design methods and structures you can use to implement your filter change accordingly.

Note The design methods and structures for FIR filters are not the same as the methods and structures for IIR filters.

Order mode

Select `Minimum` (the default) or `Specify` from the drop-down list. Selecting `Specify` enables the **Order** option (see the following sections) so you can enter the filter order.

If your **Impulse response** is `IIR`, you can specify an equal order for the numerator and denominator, or different numerator and denominator orders. The default is equal orders. To specify a different denominator order, check the **Denominator order** box.

Filter type

This option is only available if you have the DSP System Toolbox. Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

When you design either a decimator or an interpolator, the resulting filter is a bandpass filter that either decimates or interpolates your input signal.

Order

Enter the filter order. This option is enabled only if `Specify` was selected for **Order mode**.

Decimation Factor

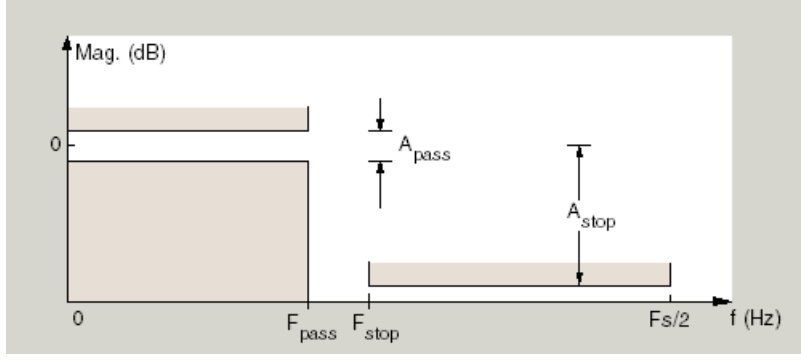
Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2.

Interpolation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Frequency specifications

The parameters in this group allow you to specify your filter response curve. Graphically, the filter specifications look similar to the one shown in the following figure.



In the figure, regions between specification values such as F_{pass} and F_{stop} represent transition regions where the filter response is not explicitly defined.

Frequency constraints

Select the filter features to use to define the frequency response characteristics. The list contains the following options, when available for the filter specifications.

- `Passband and stopband edge` — Define the filter by specifying the frequencies for the edge of the stopband and passband.
- `Passband edge` — Define the filter by specifying the frequency for the edge of the passband.
- `Stopband edge` — Define the filter by specifying the frequency for the edges of the stopband.
- `Passband edge and 3dB point` — Define the filter by specifying the passband edge frequency and the 3-dB down point (IIR designs).
- `3dB point and stopband edge` — Define the filter by specifying the 3-dB down point and stopband edge frequency (IIR designs).
- `3dB point` — Define the filter by specifying the frequency for the 3-dB point (IIR designs or maxflat FIR).
- `6dB point` — Define the filter by specifying the frequency for the 6-dB point in the filter response (FIR designs).

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is

available when you select one of the frequency options from the **Frequency units** list.

Fpass
Enter the frequency at the of the passband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Fstop
Enter the frequency at the start of the stopband. Specify the value in either normalized frequency units or the absolute units you select **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude units
Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in decibels (default)
- `Squared` — Specify the magnitude in squared units.

Apass
Enter the filter ripple allowed in the passband in the units you choose for **Magnitude units**, either linear or decibels.

Astop
Enter the filter attenuation in the stopband in the units you choose for **Magnitude units**, either linear or decibels.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

Lists the design methods available for the frequency and magnitude specifications you entered. When you change the specifications for a filter, such as changing the impulse response, the methods available to design filters changes as well. The default IIR design method is usually Butterworth, and the default FIR method is equiripple.

Scale SOS filter coefficients to reduce chance of overflow

Selecting this parameter directs the design to scale the filter coefficients to reduce the chances that the inputs or calculations in the filter overflow and exceed the representable range of the filter. Clearing this option removes the scaling. This parameter applies only to IIR filters.

Design Options

The options for each design are specific for each design method. This section does not present all of the available options for all designs and design methods. There are many more that you encounter as you select different design methods and filter specifications. The following options represent some of the most common ones available.

Density factor

Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

Phase constraint

This option only applies when you have the DSP System Toolbox software and when the **Design method** is `equiripple`. Select one of `Linear`, `Minimum`, or `Maximum`.

Minimum order — This option only applies when you have the DSP System Toolbox software and the **Order mode** is `Minimum`.

Select `Any` (default), `Even`, or `Odd`. Selecting `Even` or `Odd` forces the minimum-order design to be an even or odd order.

Match Exactly

Specifies that the resulting filter design matches either the passband or stopband when you select `Passband` or `Stopband`.

Stopband Shape

Stopband shape lets you specify how the stopband changes with increasing frequency. Choose one of the following options:

- `Flat` — Specifies that the stopband is flat. The attenuation does not change as the frequency increases.
- `Linear` — Specifies that the stopband attenuation changes linearly as the frequency increases. Change the slope of the stopband by setting **Stopband decay**.
- `1/f` — Specifies that the stopband attenuation changes exponentially as the frequency increases, where `f` is the frequency. Set the power (exponent) for the decay in **Stopband decay**.

Stopband Decay

When you set Stopband shape, Stopband decay specifies the amount of decay applied to the stopband. the following conditions apply to Stopband decay based on the value of Stopband Shape:

- When you set **Stopband shape** to `Flat`, **Stopband decay** has no affect on the stopband.
- When you set **Stopband shape** to `Linear`, enter the slope of the stopband in units of dB/rad/s. `filterbuilder` applies that slope to the stopband.

- When you set **Stopband shape** to `1/f`, enter a value for the exponent n in the relation $(1/f)^n$ to define the stopband decay. `filterbuilder` applies the $(1/f)^n$ relation to the stopband to result in an exponentially decreasing stopband attenuation.

Wpass
Passband weight. This option only applies when **Impulse response** is `FIR` and **Order mode** is `Specify`.

Wstop
Stopband weight. This option only applies when **Impulse response** is `FIR` and **Order mode** is `Specify`.

Filter implementation

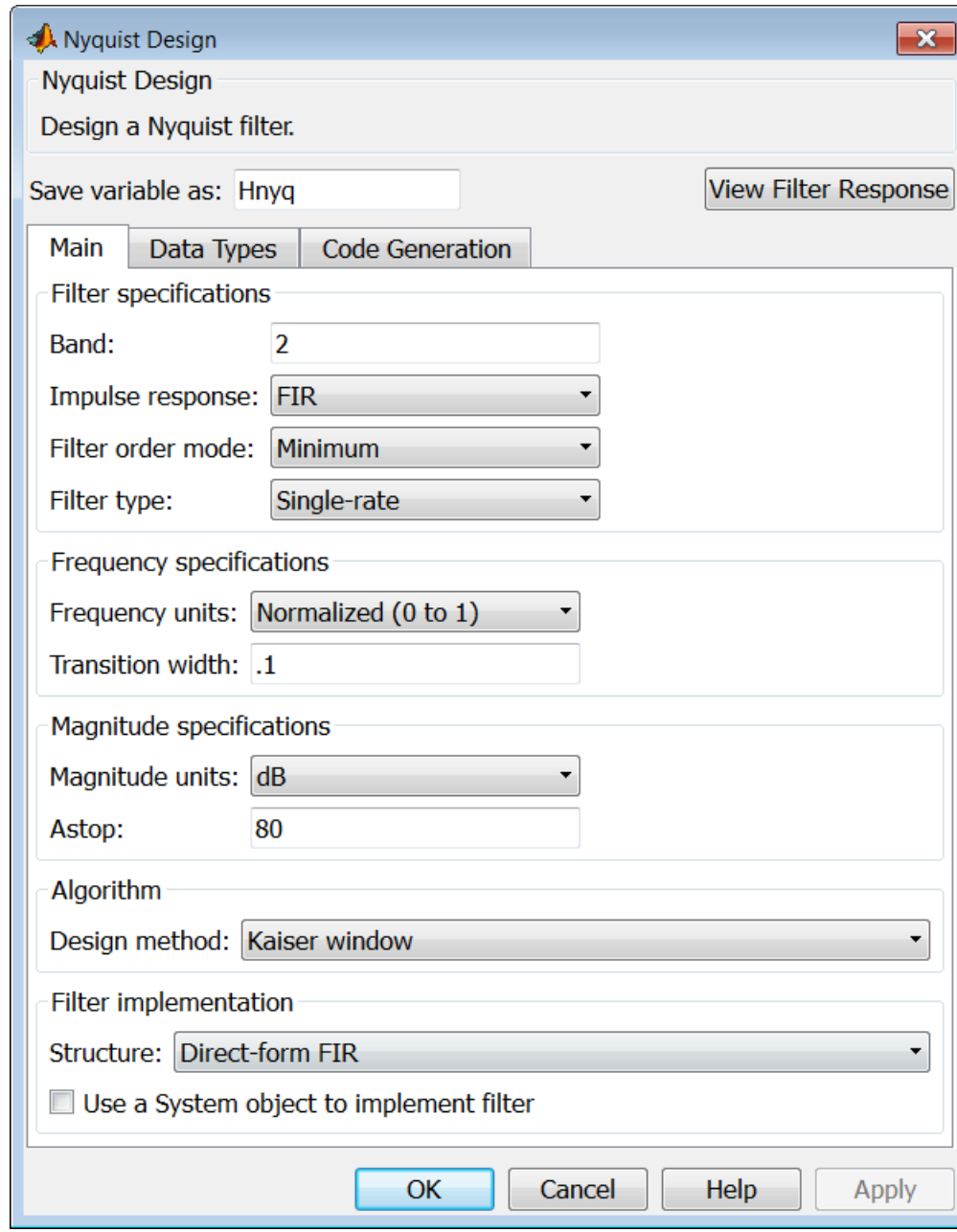
Structure
For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure, and IIR filters use direct-form II filters with SOS.

Use a System object to implement filter
Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Notch

See “Peak/Notch Filter Design Dialog Box — Main Pane” on page 1-483.

Nyquist Filter Design Dialog Box — Main Pane



Filter specifications

Parameters in this group enable you to specify your filter format, such as the impulse response and the filter order.

Band

Specifies the location of the center of the transition region between the passband and the stopband. The center of the transition region, bw, is calculated using the value for Band:

bw = Fs/(2*Band).

Impulse response

Select `FIR` or `IIR` from the drop-down list, where `FIR` is the default impulse response. When you choose an impulse response, the design methods and structures you can use to implement your filter change accordingly.

Note The design methods and structures for FIR filters are not the same as the methods and structures for IIR filters.

Order mode

Select `Minimum` (the default) or `Specify` from the drop-down list. Selecting `Specify` enables the **Order** option (see the following sections) so you can enter the filter order.

Filter type

Select `Single-rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. Your choice determines the type of filter as well as the design methods and structures that are available to implement your filter. By default, `filterbuilder` specifies single-rate filters.

- Selecting `Decimator` or `Interpolator` activates the **Decimation Factor** or the **Interpolation Factor** options respectively.
- Selecting `Sample-rate converter` activates both factors.

When you design either a decimator or an interpolator, the resulting filter is a bandpass filter that either decimates or interpolates your input signal.

Order

Enter the filter order. This option is enabled only if `Specify` was selected for **Order mode**.

Decimation Factor

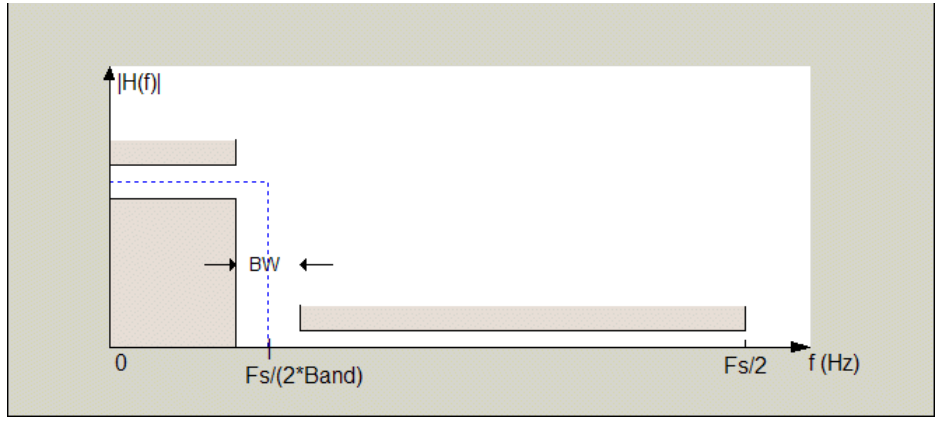
Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Decimator` or `Sample-rate converter`. The default factor value is 2.

Interpolation Factor

Enter the decimation factor. This option is enabled only if the **Filter type** is set to `Interpolator` or `Sample-rate converter`. The default factor value is 2.

Frequency specifications

The parameters in this group allow you to specify your filter response curve. Graphically, the filter specifications look similar to those shown in the following figure.



In the figure, BW is the width of the transition region and **Band** determines the location of the center of the region.

Frequency constraints

Select the filter features to use to define the frequency response characteristics. The list contains the following options, when available for the filter specifications.

- `Passband and stopband edges` — Define the filter by specifying the frequencies for the edges for the stopbands and passbands.
- `Passband edges` — Define the filter by specifying frequencies for the edges of the passband.
- `Stopband edges` — Define the filter by specifying frequencies for the edges of the stopbands.
- `3 dB points` — Define the filter response by specifying the locations of the 3 dB points. The 3 dB point is the frequency for the point 3 dB point below the passband value.
- `3 dB points and passband width` — Define the filter by specifying frequencies for the 3 dB points in the filter response and the width of the passband.
- `3 dB points and stopband widths` — Define the filter by specifying frequencies for the 3 dB points in the filter response and the width of the stopband.

Frequency units

Use this parameter to specify whether your frequency settings are normalized or in absolute frequency. Select `Normalized (0 1)` to enter frequencies in normalized form. This behavior is the default. To enter frequencies in absolute values, select one of the frequency units from the drop-down list—`Hz`, `kHz`, `MHz`, or `GHz`. Selecting one of the unit options enables the **Input Fs** parameter.

Input Fs

Fs, specified in the units you selected for **Frequency units**, defines the sampling frequency at the filter input. When you provide an input sampling frequency, all frequencies in the specifications are in the selected units as well. This parameter is available when you select one of the frequency options from the **Frequency units** list.

Transition width

Specify the width of the transition between the end of the passband and the edge of the stopband. Specify the value in normalized frequency units or the absolute units you select in **Frequency units**.

Magnitude specifications

The parameters in this group let you specify the filter response in the passbands and stopbands.

Magnitude units

Specify the units for any parameter you provide in magnitude specifications. Select one of the following options from the drop-down list.

- `Linear` — Specify the magnitude in linear units.
- `dB` — Specify the magnitude in decibels (default)
- `Squared` — Specify the magnitude in squared units.

Astop

Enter the filter attenuation in the stopband in the units you choose for **Magnitude units**, either linear or decibels.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

Lists the design methods available for the frequency and magnitude specifications you entered. When you change the specifications for a filter, such as changing the impulse response, the methods available to design filters changes as well. The default IIR design method is usually Butterworth, and the default FIR method is equiripple.

Scale SOS filter coefficients to reduce chance of overflow

Selecting this parameter directs the design to scale the filter coefficients to reduce the chances that the inputs or calculations in the filter overflow and exceed the representable range of the filter. Clearing this option removes the scaling. This parameter applies only to IIR filters.

Design Options

The options for each design are specific for each design method. This section does not present all of the available options for all designs and design methods. There are many more that you encounter as you select different design methods and filter specifications. The following options represent some of the most common ones available.

Density factor

Density factor controls the density of the frequency grid over which the design method optimization evaluates your filter response function. The number of equally spaced points in the grid is the value you enter for **Density factor** times (filter order + 1).

Increasing the value creates a filter that more closely approximates an ideal equiripple filter but increases the time required to design the filter. The default value of 16 represents a reasonable trade between the accurate approximation to the ideal filter and the time to design the filter.

Minimum phase

To design a filter that is minimum phase, select **Minimum phase**. Clearing the **Minimum phase** option removes the phase constraint—the resulting design is not minimum phase.

Minimum order

When you select this parameter, the design method determines and designs the minimum order filter to meet your specifications. Some filters do not provide this parameter. Select `Any`, `Even`, or `Odd` from the drop-down list to direct the design to be any minimum order, or minimum even order, or minimum odd order.

Note Generally, **Minimum order** designs are not available for IIR filters.

Match Exactly

Specifies that the resulting filter design matches either the passband or stopband or both bands when you select `passband` or `stopband` or `both` from the drop-down list.

Stopband Shape

Stopband shape lets you specify how the stopband changes with increasing frequency. Choose one of the following options:

- `Flat` — Specifies that the stopband is flat. The attenuation does not change as the frequency increases.
- `Linear` — Specifies that the stopband attenuation changes linearly as the frequency increases. Change the slope of the stopband by setting **Stopband decay**.
- `1/f` — Specifies that the stopband attenuation changes exponentially as the frequency increases, where `f` is the frequency. Set the power (exponent) for the decay in **Stopband decay**.

Stopband Decay

When you set Stopband shape, Stopband decay specifies the amount of decay applied to the stopband. the following conditions apply to Stopband decay based on the value of Stopband Shape:

- When you set **Stopband shape** to `Flat`, **Stopband decay** has no affect on the stopband.
- When you set **Stopband shape** to `Linear`, enter the slope of the stopband in units of dB/rad/s. `filterbuilder` applies that slope to the stopband.
- When you set **Stopband shape** to `1/f`, enter a value for the exponent n in the relation $(1/f)^n$ to define the stopband decay. `filterbuilder` applies the $(1/f)^n$ relation to the stopband to result in an exponentially decreasing stopband attenuation.

Filter implementation

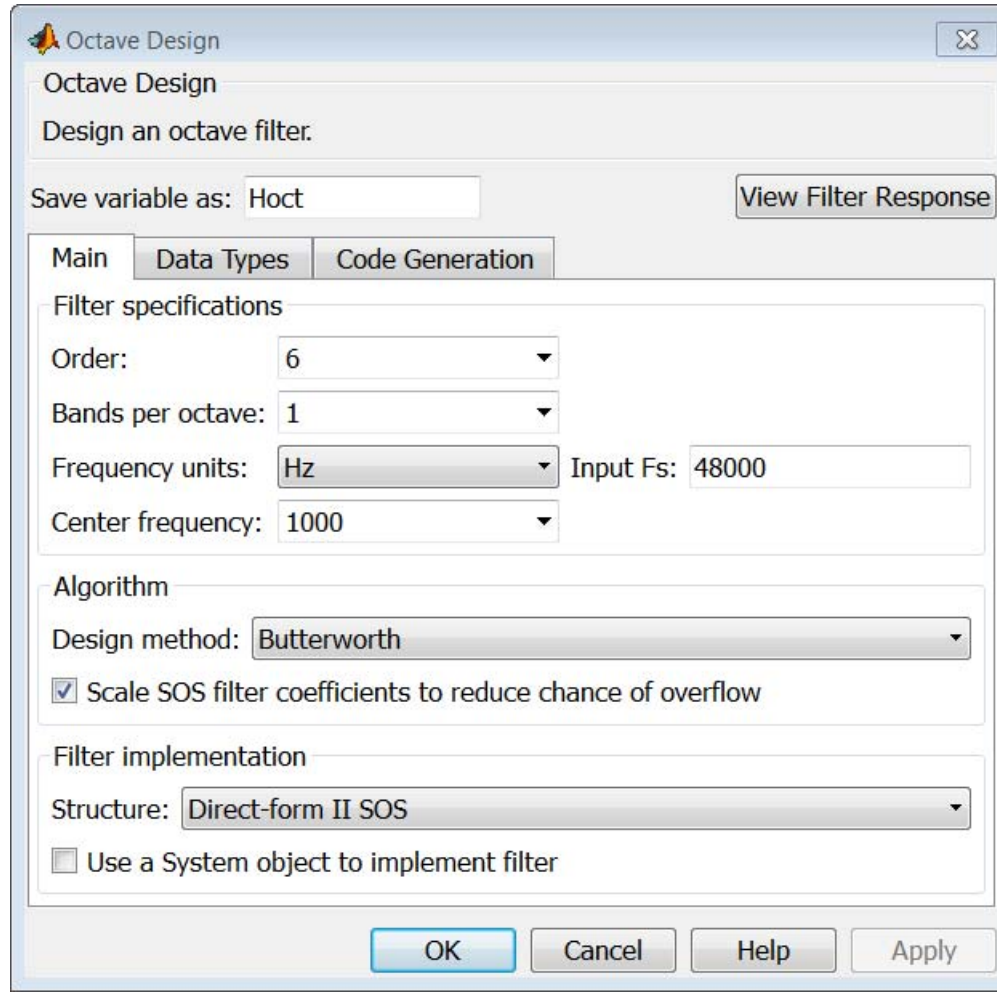
Structure

For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure, and IIR filters use direct-form II filters with SOS.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Octave Filter Design Dialog Box — Main Pane



Filter specifications

Order
Specify filter order. Possible values are: `4, 6, 8, 10`.

Bands per octave
Specify the number of bands per octave. Possible values are: `1, 3, 6, 12, 24`.

Frequency units
Specify frequency units as `Hz` or `kHz`.

Input Fs
Specify the input sampling frequency in the frequency units specified previously.

Center Frequency
Select from the drop-down list of available center frequency values.

Algorithm

Design Method
Butterworth is the design method used for this type of filter.

Scale SOS filter coefficients to reduce chance of overflow
Select the check box to scale the filter coefficients.

Filter implementation

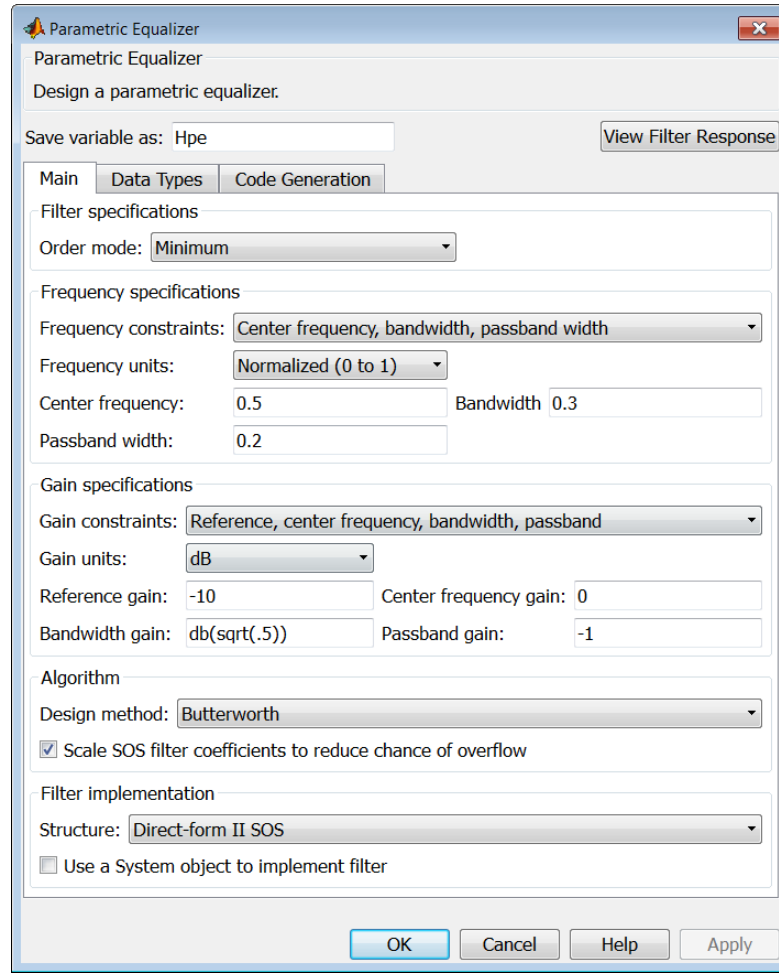
Structure
Specify filter structure. Choose from:

- Direct-form I SOS
- Direct-form II SOS
- Direct-form I transposed SOS
- Direct-form II transposed SOS

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default, the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Parametric Equalizer Filter Design Dialog Box — Main Pane



Filter specifications

Order mode

Select `Minimum` to design a minimum order filter that meets the design specifications, or `Specify` to enter a specific filter order. The order mode also affects the possible frequency constraints, which in turn limit the gain specifications. For example, if you specify a `Minimum` order filter, the available frequency constraints are:

- `Center frequency, bandwidth, passband width`
- `Center frequency, bandwidth, stopband width`

If you select `Specify`, the available frequency constraints are:

- `Center frequency, bandwidth`
- `Center frequency, quality factor`
- `Shelf type, cutoff frequency, quality factor`
- `Shelf type, cutoff frequency, shelf slope parameter`
- `Low frequency, high frequency`

Order

This parameter is enabled only if the **Order mode** is set to `Specify`. Enter the filter order in this text box.

Frequency specifications

Depending on the filter order, the possible frequency constraints change. Once you choose the frequency constraints, the input boxes in this area change to reflect the selection.

Frequency constraints

Select the specification to represent the frequency constraints. The following options are available:

- `Center frequency, bandwidth, passband width` (available for minimum order only)

- `Center frequency, bandwidth, stopband width` (available for minimum order only)
- `Center frequency, bandwidth` (available for a specified order only)
- `Center frequency, quality factor` (available for a specified order only)
- `Shelf type, cutoff frequency, quality factor` (available for a specified order only)
- `Shelf type, cutoff frequency, shelf slope parameter` (available for a specified order only)
- `Low frequency, high frequency` (available for a specified order only)

Frequency units

Select the frequency units from the available drop down list (`Normalized, Hz, kHz, MHz, GHz`). If `Normalized` is selected, then the **Input Fs** box is disabled for input.

Input Fs

Enter the input sampling frequency. This input box is disabled for input if `Normalized` is selected in the **Frequency units** input box.

Center frequency

Enter the center frequency in the units specified by the value in **Frequency units**.

Bandwidth

The bandwidth determines the frequency points at which the filter magnitude is attenuated by the value specified as the **Bandwidth gain** in the **Gain specifications** section. By default, the **Bandwidth gain** defaults to `db(sqrt(.5))`, or –3 dB relative to the center frequency. The **Bandwidth** property only applies when the **Frequency constraints** are: `Center frequency, bandwidth, passband width`, `Center frequency, bandwidth, stopband width`, or `Center frequency, bandwidth`.

Passband width

The passband width determines the frequency points at which the filter magnitude is attenuated by the value specified as the **Passband gain** in the **Gain specifications** section. This option is enabled only if the filter is of minimum order, and the frequency constraint selected is `Center frequency, bandwidth, passband width`.

Stopband width

The stopband width determines the frequency points at which the filter magnitude is attenuated by the value specified as the **Stopband gain** in the **Gain specifications** section. This option is enabled only if the filter is of minimum order, and the frequency constraint selected is `Center frequency, bandwidth, stopband width`.

Low frequency

Enter the low frequency cutoff. This option is enabled only if the filter order is user specified and the frequency constraint selected is `Low frequency, high frequency`. The filter magnitude is attenuated by the amount specified in **Bandwidth gain**.

High frequency

Enter the high frequency cutoff. This option is enabled only if the filter order is user specified and the frequency constraint selected is `Low frequency, high frequency`. The filter magnitude is attenuated by the amount specified in **Bandwidth gain**.

Gain specifications

Depending on the filter order and frequency constraints, the possible gain constraints change. Also, once you choose the gain constraints the input boxes in this area change to reflect the selection.

Gain constraints

Select the specification array to represent gain constraints, and remember that not all of these options are available for all configurations. The following is a list of all available options:

- `Reference, center frequency, bandwidth, passband`
- `Reference, center frequency, bandwidth, stopband`
- `Reference, center frequency, bandwidth,passband, stopband`
- `Reference, center frequency, bandwidth`

Gain units

Specify the gain units either `dB` or `squared`. These units are used for all gain specifications in the dialog box.

Reference gain

The reference gain determines the level to which the filter magnitude attenuates in **Gain units**. The reference gain is a *floor* gain for the filter magnitude response. For example, you may use the reference gain together with the **Center frequency gain** to leave certain frequencies unattenuated (reference gain of 0 dB) while boosting other frequencies.

Bandwidth gain

Specifies the gain in **Gain units** at which the bandwidth is defined. This property applies only when the **Frequency constraints** specification contains a `bandwidth` parameter, or is `Low frequency, high frequency`.

Center frequency gain

Specify the center frequency in **Gain units**

Passband gain

The passband gain determines the level in **Gain units** at which the passband is defined. The passband is determined either by the **Passband width** value, or the **Low frequency** and **High frequency** values in the **Frequency specifications** section.

Stopband gain

The stopband gain is the level in **Gain units** at which the stopband is defined. This property applies only when the **Order mode** is minimum and the **Frequency constaints** are `Center frequency, bandwidth, stopband width`.

Boost/cut gain
The boost/cut gain applies only when the designing a shelving filter. Shelving filters include the `Shelf type` parameter in the **Frequency constraints** specification. The gain in the passband of the shelving filter is increased by **Boost/cut gain** dB from a *floor* gain of 0 dB.

Algorithm

Design method
Select the design method from the drop-down list. Different IIR design methods are available depending on the filter constraints you specify.

Scale SOS filter coefficients to reduce chance of overflow
Select the check box to scale the filter coefficients.

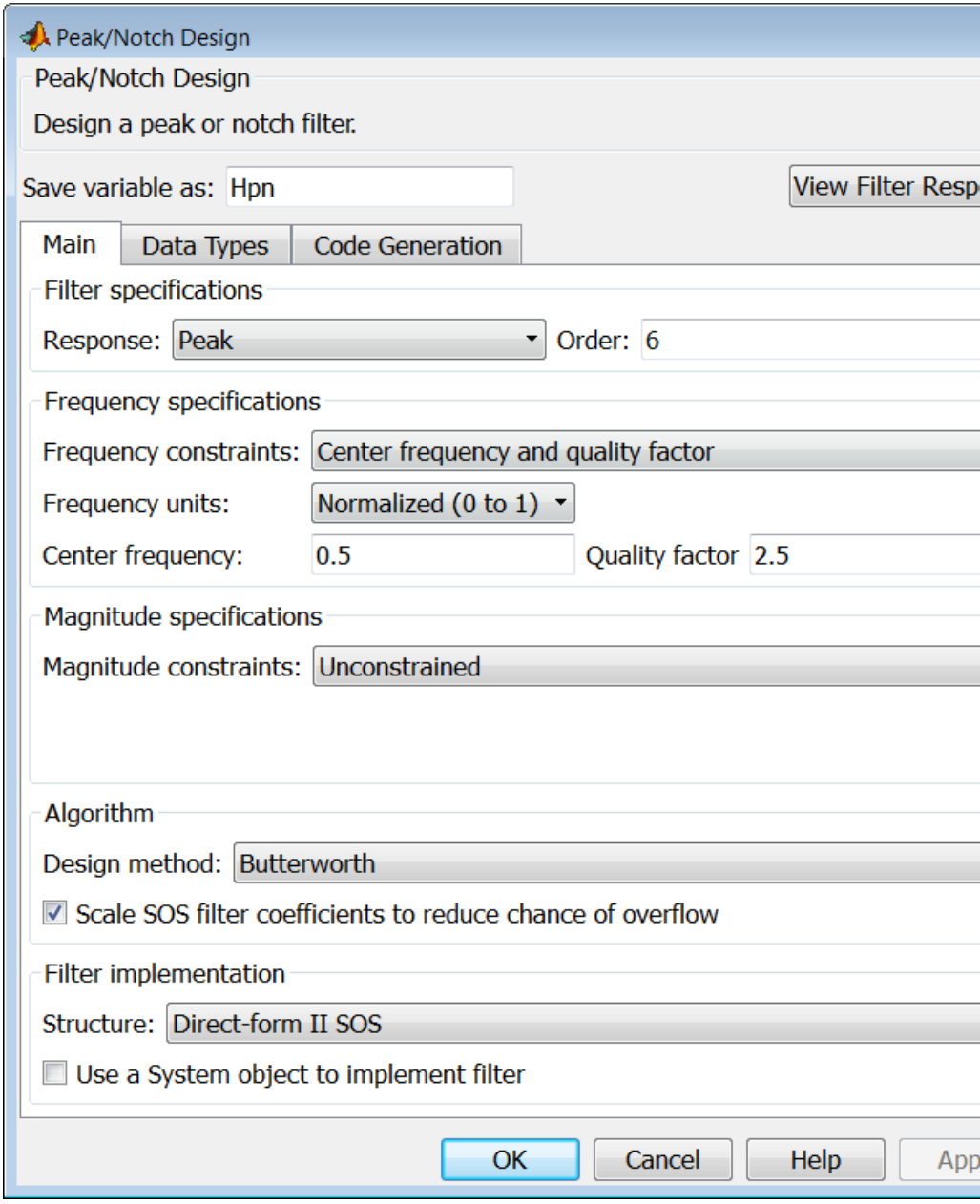
Filter implementation

Structure
Select filter structure. The possible choices are:

- Direct-form I SOS
- Direct-form II SOS
- Direct-form I transposed SOS
- Direct-form II transposed SOS

Use a System object to implement filter
Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Peak/Notch Filter Design Dialog Box — Main Pane



Filter specifications

In this area you can specify whether you want to design a peaking filter or a notching filter, as well as the order of the filter.

Response
Select `Peak` or `Notch` from the drop-down box.

Order
Enter the filter order. The order must be even.

Frequency specifications

This group of parameters allows you to specify frequency constraints and units.

Frequency Constraints
Select the frequency constraints for filter specification. There are two choices as follows:

- `Center frequency and quality factor`
- `Center frequency and bandwidth`

Frequency units
The frequency units are normalized by default. If you specify units other than normalized, `filterbuilder` assumes that you wish to specify an input sampling frequency, and enables this input box. The choice of frequency units are: `Normalized (0 to 1), Hz, kHz, MHz, GHz`.

Input Fs
This input box is enabled if **Frequency units** other than `Normalized (0 to 1)` are specified. Enter the input sampling frequency.

Center frequency
Enter the center frequency in the units you specified in **Frequency units**.

Quality Factor

This input box is enabled only when `Center frequency and quality factor` is chosen for the **Frequency Constraints**. Enter the quality factor.

Bandwidth

This input box is enabled only when `Center frequency and bandwidth` is chosen for the **Frequency Constraints**. Enter the bandwidth.

Magnitude specifications

This group of parameters allows you to specify the magnitude constraints, as well as their values and units.

Magnitude Constraints

Depending on the choice of constraints, the other input boxes are enabled or disabled. Select from four magnitude constraints available:

- `Unconstrained`
- `Passband ripple`
- `Stopband attenuation`
- `Passband ripple and stopband attenuation`

Magnitude units

Select the magnitude units: either `dB` or `squared`.

Apass

This input box is enabled if the magnitude constraints selected are `Passband ripple` or `Passband ripple and stopband attenuation`. Enter the passband ripple.

Astop

This input box is enabled if the magnitude constraints selected are `Stopband attenuation` or `Passband ripple and stopband attenuation`. Enter the stopband attenuation.

Algorithm

The parameters in this group allow you to specify the design method and structure that `filterbuilder` uses to implement your filter.

Design Method

Lists all design methods available for the frequency and magnitude specifications you entered. When you change the specifications for a filter the methods available to design filters changes as well.

Scale SOS filter coefficients to reduce chance of overflow

Selecting this parameter directs the design to scale the filter coefficients to reduce the chances that the inputs or calculations in the filter overflow and exceed the representable range of the filter. Clearing this option removes the scaling. This parameter applies only to IIR filters.

Filter implementation

Structure

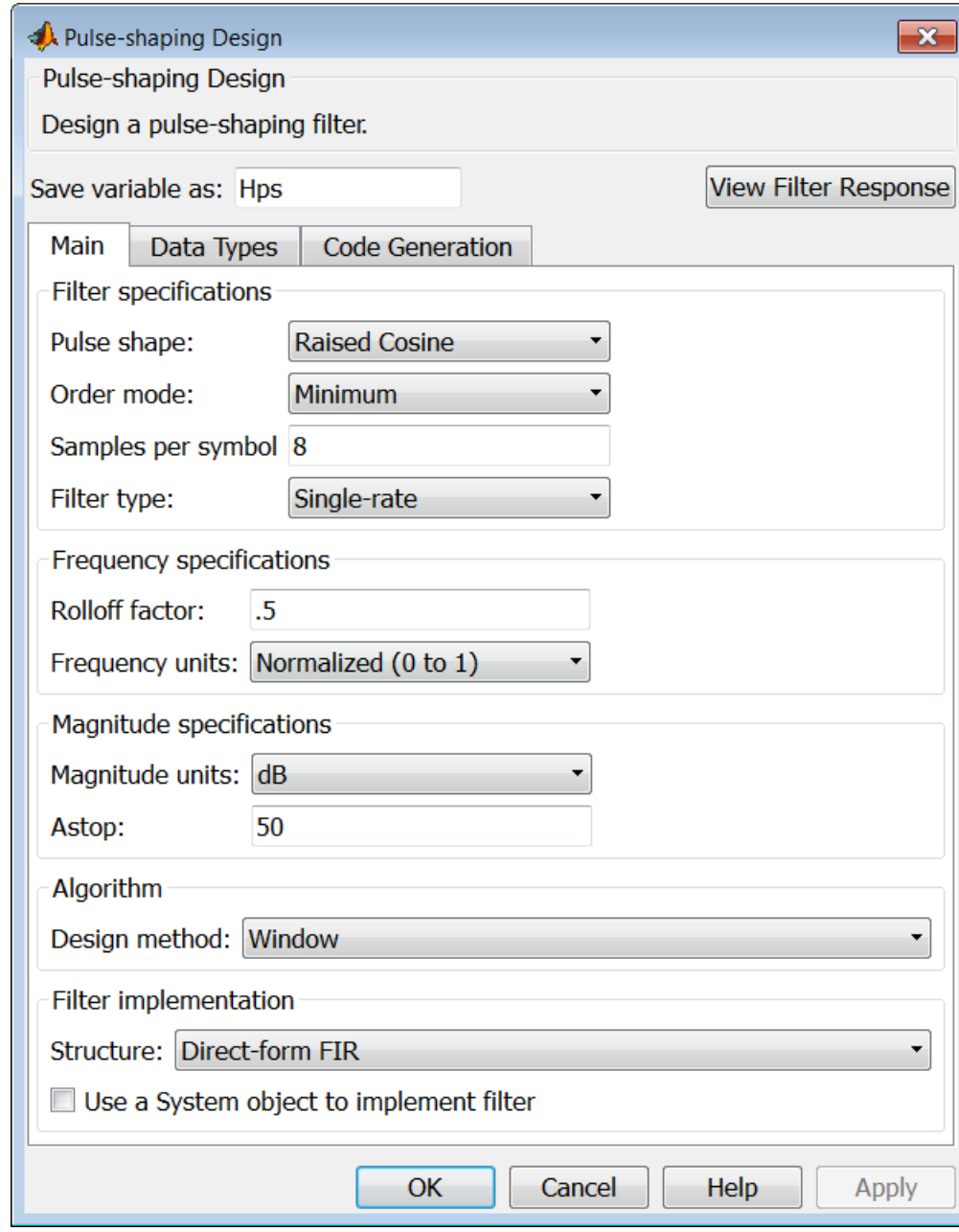
Lists all available filter structures for the filter specifications and design method you select. The typical options are:

- Direct-form I SOS
- Direct-form II SOS
- Direct-form I transposed SOS
- Direct-form II transposed SOS

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Pulse-shaping Filter Design Dialog Box—Main Pane



Filter specifications

Parameters in this group enable you to specify the shape and length of the filter.

Pulse shape

Select the shape of the impulse response from the following options:

- `Raised Cosine`
- `Square Root Raised Cosine`
- `Gaussian`

Order mode

This specification is only available for raised cosine and square root raised cosine filters. For these filters, select one of the following options:

- `Minimum`— This option will result in the minimum-length filter satisfying the user-specified **Frequency specifications**.
- `Specify order`—This option allows the user to construct a raised cosine or square root cosine filter of a specified order by entering an even number in the **Order** input box. The length of the impulse response will be `Order+1` .
- `Specify symbols`—This option enables the user to specify the length of the impulse response in an alternative manner. If `Specify symbols` is chosen, the **Order** input box changes to the **Number of symbols** input box.

Samples per symbol

Specify the oversampling factor. Increasing the oversampling factor guards against aliasing and improves the FIR filter approximation to the ideal frequency response. If **Order** is specified in **Number of symbols**, the filter length will be **Number of symbols*Samples per symbol**+1. The product **Number of symbols*Samples per symbol** must be an even number.

If a Gaussian filter is specified, the filter length must be specified in **Number of symbols** and **Samples per symbol**. The product **Number of symbols*Samples per symbol** must be an even number. The filter length will be **Number of symbols*Samples per symbol**+1.

Filter Type

This option is only available if you have the DSP System Toolbox software. Choose `Single rate`, `Decimator`, `Interpolator`, or `Sample-rate converter`. If you select `Decimator` or `Interpolator`, the decimation and interpolation factors default to the value of the **Samples per symbol**. If you select `Sample-rate converter`, the interpolation factor defaults to **Samples per symbol** and the decimation factor defaults to 3.

Frequency specifications

Parameters in this group enable you to specify the frequency response of the filter. For raised cosine and square root raised cosine filters, the frequency specifications include:

Rolloff factor

The rolloff factor takes values in the range [0,1]. The smaller the rolloff factor, the steeper the transition in the stopband.

Frequency units

The frequency units are normalized by default. If you specify units other than normalized, `filterbuilder` assumes that you wish to specify an input sampling frequency, and enables this input box. The choice of frequency units are: `Normalized (0 to 1), Hz, kHz, MHz, GHz`

For a Gaussian pulse shape, the available frequency specifications are:

Bandwidth-time product

This option allows the user to specify the width of the Gaussian filter. Note that this is independent of the length of the filter. The bandwidth-time product (BT) must be a positive real number.

Smaller values of the bandwidth-time product result in larger pulse widths in time and steeper stopband transitions in the frequency response.

Frequency units

The frequency units are normalized by default. If you specify units other than normalized, `filterbuilder` assumes that you wish to specify an input sampling frequency, and enables this input box. The choice of frequency units are: `Normalized (0 to 1), Hz, kHz, MHz, GHz`

Magnitude specifications

If the **Order mode** is specified as `Minimum`, the **Magnitude units** may be selected from:

- `dB`—Specify the magnitude in decibels (default).
- `Linear`—Specify the magnitude in linear units.

Algorithm

The only **Design method** available for FIR pulse-shaping filters is the `Window` method.

Filter implementation

Structure

For the filter specifications and design method you select, this parameter lists the filter structures available to implement your filter. By default, FIR filters use direct-form structure.

Use a System object to implement filter

Selecting this check box gives you the choice of using a system object to implement the filter. By default the check box is turned off. When the current design method or structure is not supported by a system object filter, then this check box is disabled.

Purpose

2-norm or infinity-norm of digital filter

Syntax

```
filternorm(b,a)
filternorm(b,a,pnorm)
filternorm(b,a,2,tol)
```

Description

A typical use for filter norms is in digital filter scaling to reduce quantization effects. Scaling often improves the signal-to-noise ratio of the filter without resulting in data overflow. You, also, can use the 2-norm to compute the energy of the impulse response of a filter.

`filternorm(b,a)` computes the 2-norm of the digital filter defined by the numerator coefficients in `b` and denominator coefficients in `a`.

`filternorm(b,a,pnorm)` computes the 2- or infinity-norm (inf-norm) of the digital filter, where `pnorm` is either 2 or `inf`.

`filternorm(b,a,2,tol)` computes the 2-norm of an IIR filter with the specified tolerance, `tol`. The tolerance can be specified only for IIR 2-norm computations. `pnorm` in this case must be `2`. If `tol` is not specified, it defaults to `1e-8`.

Examples

Compute the 2-norm with a tolerance of 1e-10 of an IIR filter:

```
[b,a]=butter(5,.5);
L2=filternorm(b,a,2,1e-10)

L2 =

    0.7071
```

Compute the inf-norm of an FIR filter:

```
b=firpm(30,[.1 .9],[1 1],'Hilbert');
Linf=filternorm(b,1,inf)

Linf =
```

```
1.0028
```

Algorithms

Given a filter with frequency reponse $H(e^{j\omega})$, the L_p-norm for $1 \le p < \infty$ is given by

$$||H(e^{j\omega}||_p = \left(\frac{1}{2\pi}\int_{-\pi}^{\pi} |H(e^{j\omega})|^p \, d\omega\right)^{1/p}$$

For the case $p=\infty$, the L_∞ norm is

$$||H(e^{j\omega})||_\infty = \max_{-\pi \le \omega \le \pi} |H(e^{j\omega})|$$

For the case p=2, Parseval's theorem states that

$$||H(e^{j\omega})||_2 = \left(\frac{1}{2\pi}\int_{-\pi}^{\pi} |H(e^{j\omega})|^2 \, d\omega\right)^{1/2} = \left(\sum_n |h(n)|^2\right)^{1/2}$$

where h(n) is the impulse response of the filter. The energy of the impulse response is the squared L_2 norm.

References

[1] Jackson, L.B., *Digital Filters and Signal Processing, Third Edition,* Kluwer Academic Publishers, 1996, Chapter 11.

See Also

`zp2sos` | `norm`

Purpose

Zero-phase digital filtering

Syntax

```
y = filtfilt(b,a,x)
y = filtfilt(SOS,G,x)
```

Description

`y = filtfilt(b,a,x)` performs zero-phase digital filtering by processing the input data, `x`, in both the forward and reverse directions [1]. `filtfilt` operates along the first nonsingleton dimension of `x`. The vector `b` provides the numerator coefficients of the filter and the vector `a` provides the denominator coefficients. If you use an all-pole filter, enter `1` for `b`. If you use an all-zero filter (FIR), enter `1` for `a`. After filtering the data in the forward direction, `filtfilt` reverses the filtered sequence and runs it back through the filter. The result has the following characteristics:

- Zero-phase distortion
- A filter transfer function, which equals the squared magnitude of the original filter transfer function
- A filter order that is double the order of the filter specified by `b` and `a`

`filtfilt` minimizes start-up and ending transients by matching initial conditions, and you can use it for both real and complex inputs. Do not use `filtfilt` with differentiator and Hilbert FIR filters, because the operation of these filters depends heavily on their phase response.

Note The length of the input `x` must be more than three times the filter order defined as `max(length(b)-1,length(a)-1)`.

`y = filtfilt(SOS,G,x)` zero-phase filters the data `x` using the second-order section (biquad) filter represented by the matrix `SOS` and scale values `G`. The matrix `SOS` is an L-by-6 matrix containing the L second-order sections. The matrix `SOS` must be of the form:

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

where each row are the coefficients of a biquad filter. The vector of filter scale values, `G`, must have a length between 1 and L+1.

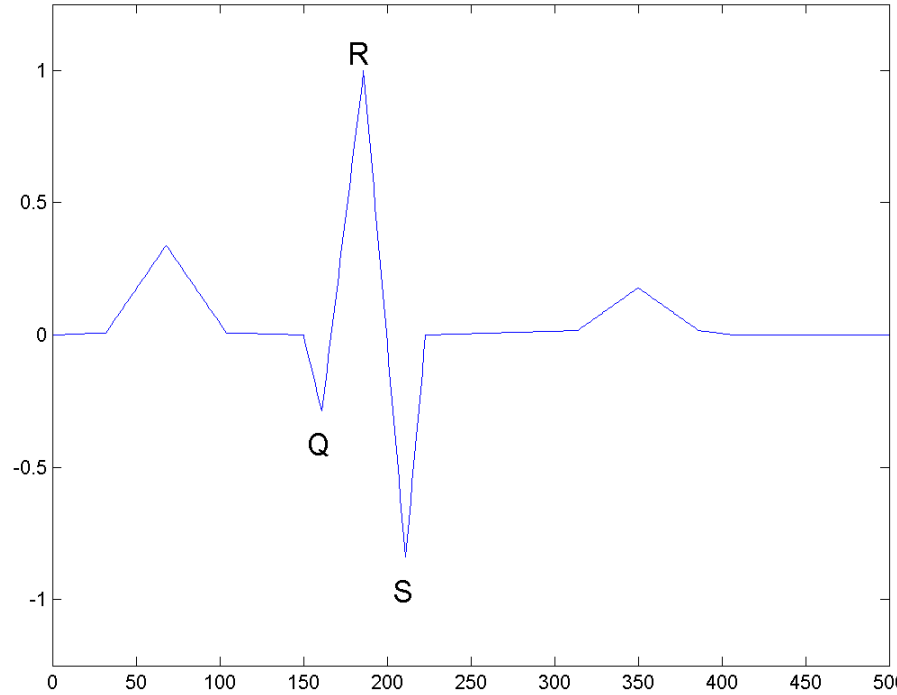
Note When implementing zero-phase filtering using a second-order section filter, the length of the input `x` must be more than 6 samples.

Examples

Zero-phase filtering helps preserve features in the filtered time waveform exactly where those features occur in the unfiltered waveform. To illustrate the use of `filtfilt` for zero-phase filtering, consider an electrocardiogram waveform as an example.

```
plot(ecg(500)); %plot ECG signal
axis([0 500 -1.25 1.25]);
```

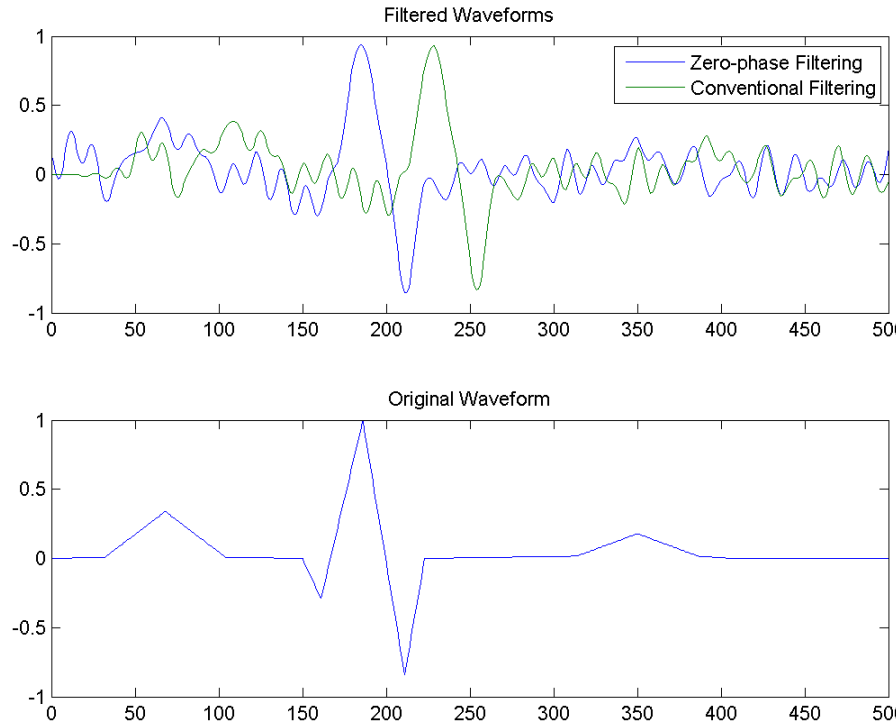
The QRS complex is an important feature in the ECG waveform beginning around time point 160 in this example.



The following example corrupts the ECG waveform with additive noise, constructs a lowpass FIR equiripple filter, and filters the noisy waveform using both zero-phase and conventional filtering. Because the filter is an all-zero (FIR) filter, the denominator equals 1. Seed the random number generator for reproducible results.

```
rng default;
x=ecg(500)'+0.25*randn(500,1); %noisy waveform
h=fdesign.lowpass('Fp,Fst,Ap,Ast',0.15,0.2,1,60);
d=design(h,'equiripple'); %Lowpass FIR filter
y=filtfilt(d.Numerator,1,x); %zero-phase filtering
```

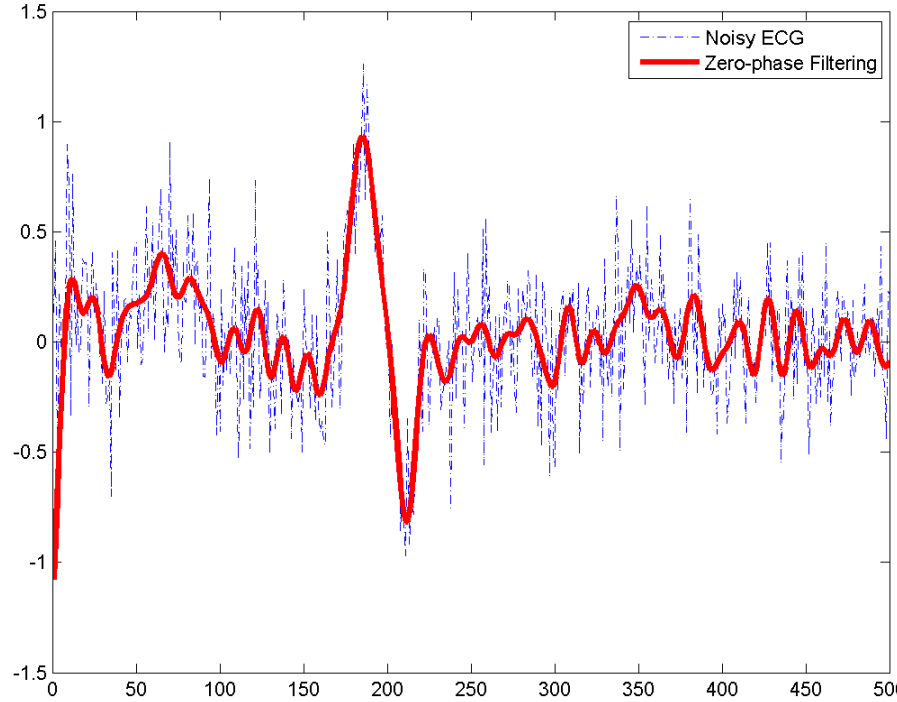
```
y1=filter(d.Numerator,1,x); %conventional filtering
subplot(211);
plot([y y1]);
title('Filtered Waveforms');
legend('Zero-phase Filtering','Conventional Filtering');
subplot(212);
plot(ecg(500));
title('Original Waveform');
```



Zero-phase filtering reduces noise in the signal and preserves the QRS complex at the same time it occurs in the original signal. Conventional filtering reduces noise in the signal, but delays the QRS complex.

Repeat the above using a Butterworth second-order section filter:

```
h=fdesign.lowpass('N,F3dB',12,0.15);
d1 = design(h,'butter');
y = filtfilt(d1.sosMatrix,d1.ScaleValues,x);
plot(x,'b-.'); hold on;
plot(y,'r','linewidth',3);
legend('Noisy ECG','Zero-phase Filtering','location','NorthEast');
```



References

[1] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing,* Prentice-Hall, 1989, pp. 284–285.

[2] Mitra, S.K., *Digital Signal Processing, 2nd ed.,* McGraw-Hill, 2001, Sections 4.4.2 and 8.2.5.

[3] Gustafsson, F., Determining the initial states in forward-backward filtering, *IEEE Transactions on Signal Processing*, April 1996, Volume 44, Issue 4, pp. 988–992.

See Also

`fftfilt` | `filter` | `filter2`

Purpose

Initial conditions for transposed direct-form II filter implementation

Syntax

```
z = filtic(b,a,y,x)
z = filtic(b,a,y)
```

Description

`z = filtic(b,a,y,x)` finds the initial conditions, `z`, for the delays in the *transposed direct-form II* filter implementation given past outputs `y` and inputs `x`. The vectors `b` and `a` represent the numerator and denominator coefficients, respectively, of the filter's transfer function.

The vectors `x` and `y` contain the most recent input or output first, and oldest input or output last.

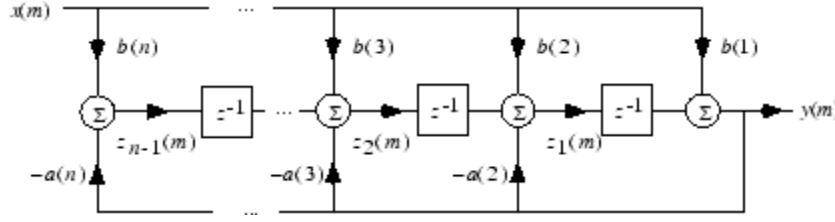
$$x = [x(-1), x(-2), x(-3), \ldots, x(-n)]$$
$$y = [y(-1), y(-2), y(-3), \ldots, y(-m)]$$

where `n` is `length(b)-1` (the numerator order) and `m` is `length(a)-1` (the denominator order). If `length(x)` is less than `n`, `filtic` pads it with zeros to length `n`; if `length(y)` is less than `m`, `filtic` pads it with zeros to length `m`. Elements of `x` beyond `x(n-1)` and elements of `y` beyond `y(m-1)` are unnecessary so `filtic` ignores them.

Output `z` is a column vector of length equal to the larger of n and m. `z` describes the state of the delays given past inputs `x` and past outputs `y`.

`z = filtic(b,a,y)` assumes that the input `x` is 0 in the past.

The transposed direct-form II structure is shown in the following illustration.



n-1 is the filter order.

`filtic` works for both real and complex inputs.

Algorithms

`filtic` performs a reverse difference equation to obtain the delay states `z`.

Diagnostics

If any of the input arguments `y`, `x`, `b`, or `a` is not a vector (that is, if any argument is a scalar or array), `filtic` gives the following error message:

```
Requires vector inputs.
```

References

[1] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, 1989, pp. 296, 301-302.

See Also

`filter` | `filtfilt`

filtord

Purpose

Filter order

Syntax

```
n = filtord(b,a)
n = filtord(sos)
```

Description

`n = filtord(b,a)` returns the filter order, `n`, for the causal rational system function specified by the numerator coefficients, `b`, and denominator coefficients, `a`.

`n = filtord(sos)` returns the filter order for the filter specified by the second order sections matrix, `sos`. `sos` is a K-by-6 matrix. The number of sections, K, must be greater than or equal to 2. Each row of `sos` corresponds to the coefficients of a second order filter. The i-th row of the second order section matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

Input Arguments

b - Numerator coefficients

vector | scalar

Numerator coefficients, specified as a scalar, or a vector. If the filter is an allpole filter, `b` is a scalar. Otherwise, `b` is a row or column vector.

Example: `b = fir1(20,0.25)`

Data Types
`single | double`
Complex Number Support: Yes

a - Denominator coefficients

vector | scalar

Denominator coefficients, specified as a scalar, or a vector. If the filter is an FIR filter, `a` is a scalar. Otherwise, `a` is a row or column vector.

Example: `[b,a] = butter(20,0.25)`

Data Types
`single | double`
Complex Number Support: Yes

sos - Matrix of second order sections

matrix

Matrix of second order sections, specified as a K-by-6 matrix. The system function of the K-th biquad filter has the rational z-transform

$$H_k(z) = \frac{B_k(1) + B_k(2)z^{-1} + B_k(3)z^{-2}}{A_k(1) + A_k(2)z^{-1} + A_k(3)z^{-2}}$$

The coefficients in the K-th row of the matrix, `sos`, are ordered as follows

$$[B_k(1)B_k(2)B_k(3)A_k(1)A_k(2)A_k(3)]$$

The frequency response of the filter is system function evaluated on the unit circle with

$$z = e^{i2\pi f}$$

Data Types
`single | double`
Complex Number Support: Yes

Output Arguments

n - Filter order

integer

Filter order, specifed as an integer.

Examples

Verify Order of FIR Filter

Design an order-20 FIR filter with a cutoff frequency of 0.5π radians/sample using the window method. Verify the filter order.

```
b = fir1(20,0.5);
n = filtord(b,1)
```

Determine Order Difference Between FIR and IIR Designs

Design FIR equiripple and IIR Butterworth filters from the same set of specifications. Determine the difference in filter order between the two designs.

```
d = fdesign.lowpass('Fp,Fst,Ap,Ast',100,120,0.5,60,1000);
Hd_FIR = design(d,'equiripple');
Hd_IIR = design(d,'butter');
filtord(Hd_FIR.Numerator,1)-filtord(Hd_IIR.sosMatrix)
```

See Also

isallpass | isminphase | ismaxphase | isstable

Purpose

Filter states

Syntax

```
Hs = filtstates.structure(input1,...)
```

Description

`Hs = filtstates.structure(input1,...)` returns a filter states object `Hs`, which contains the filter states.

You can extract a `filtstates` object from the states property of an object with

```
Hd = dfilt.df1
Hs = Hd.states
```

or, for an `mfilt` object in the DSP System Toolbox product, with

```
Hm = mfilt.cicdecim
Hs = Hm.states
```

Structures

Structures for `filtstates` specify the type of filter structure. Available types of structures for `filtstates` are shown below.

filtstates.structure	Description
`filtstates.dfiir`	filtstates for IIR direct-form I filters (`dfilt.df1`, `dfilt.df1t`, `dfilt.df1sos`, and `dfilt.df1tsos`)
`filtstates.cic`	filtstates for cascaded integrator comb filters. (Available only with DSP System Toolbox and Fixed-Point Designer products.)

Refer to the particular `filtstates.structure` reference page or use the syntax `help filtstates.structure` at the MATLAB prompt for more information.

See Also

`filtstates.dfiir` | `dfilt` | `dfilt.df1` | `dfilt.df1t` | `dfilt.df1sos` | `dfilt.df1tsos`

Purpose

IIR direct-form filter states

Syntax

```
Hs = filtstates.dfiir(numstates,denstates)
```

Description

`Hs = filtstates.dfiir(numstates,denstates)` returns an IIR direct-form filter states object `Hs` with two properties — `Numerator` and `Denominator`, which contain the filter states. These two properties are column vectors with each column representing a separate channel of filter states. The number of states is always one less than the number of filter numerator or denominator coefficients.

You can extract a filtstates object from the states property of an IIR direct-form I object with

```
Hd = dfilt.df1
Hs = Hd.states
```

Methods

You can use the following methods on a `filtstates.dfiir` object.

Method	Description
`double`	Converts a `filtstates` object to a double-precision vector containing the values of the numerator and denominator states. The numerator states are listed first in this vector, followed by the denominator states.
`single`	Converts a `filtstates` object to a single-precision vector containing the values of the numerator and denominator states. (This method is used with the DSP System Toolbox product.)

Examples

This example demonstrates the interaction of `filtstates` with a `dfilt.df1` object.

```
[b,a] = butter(4,0.5);    % Design butterworth filter
```

```
Hd = dfilt.df1(b,a);      % Create dfilt object
Hs = Hd.states            % Extract filter states object
                          %  from dfilt states property
Hs.Numerator = [1,1,1,1]  % Modify numerator states
Hd.states = Hs            % Set modified states back to
                          %  original object

Dbl = double(Hs)          % Create double vector from
                          %  states
```

See Also

filtstates | dfilt | dfilt.df1 | dfilt.df1t | dfilt.df1sos | dfilt.df1tsos

Purpose

Generate Simulink filter block

Syntax

```
filt2block(b)
filt2block(b,'subsystem')
filt2block(___,'FilterStructure',structure)

filt2block(b,a)
filt2block(b,a'subsystem')
filt2block(___,'FilterStructure',structure)

filt2block(sos)
filt2block(sos,'subsystem')
filt2block(___,'FilterStructure',structure)

filt2block(___,Name,Value)
```

Description

`filt2block(b)` generates a `Discrete FIR Filter block` with filter coefficients, `b`.

`filt2block(b,'subsystem')` generates a Simulink subsystem block that implements an FIR filter using sum, gain, and delay blocks.

`filt2block(___,'FilterStructure',structure)` specify the filter `structure` for the FIR filter.

`filt2block(b,a)` generates a `Discrete Filter block` with numerator coefficients, `b`, and denominator coefficients, `a`.

`filt2block(b,a'subsystem')` generates a Simulink subsystem block that implements an IIR filter using sum, gain, and delay blocks.

`filt2block(___,'FilterStructure',structure)` specify the filter `structure` for the IIR filter.

`filt2block(sos)` generates a `Biquad Filter block` with second order sections matrix, `sos`. `sos` is a K-by-6 matrix, where the number of sections, `K`, must be greater than or equal to 2. You must have the DSP System Toolbox software installed to use this syntax.

`filt2block(sos,'subsystem')` generates a Simulink subsystem block that implements a biquad filter using sum, gain, and delay blocks.

`filt2block( ___ ,'FilterStructure',structure)` specify the filter`structure` for the biquad filter.

`filt2block( ___ ,Name,Value))` uses additional options specified by one or more `Name,Value` pair arguments.

Input Arguments

b - Numerator filter coefficients

row or column vector

Numerator filter coefficients, specified as a row or column vector. The filter coefficients are ordered in descending powers of z^{-1} with the first element corresponding to the coefficient for z^0.

Example: `B = fir1(30,0.25);`

Data Types
`single | double`
Complex Number Support: Yes

structure - Filter structure

string

Filter structure, specified as a string. Valid options for `structure` depend on the input arguments. The following table lists the valid filter structures by input.

Input	Filter Structures
b	'directForm' (default), 'directFormTransposed', 'directFormSymmetric', 'directFormAntiSymmetric', 'overlapAdd'. The 'overlapAdd' structure is only available when you omit 'subsystem'
a	'directForm2' (default),'directForm1', 'directForm1Transposed', 'directForm2', 'directForm2Transposed'
sos	'directForm2Transposed' (default), 'directForm1', 'directForm1Transposed', 'directForm2'

a - Denominator filter coefficients

row or column vector

Denominator filter coefficients, specified as a row or column vector. The filter coefficients are ordered in descending powers of z^{-1} with the first element corresponding to the coefficient for z^0. The first filter coefficient must be 1.

Data Types
single | double
Complex Number Support: Yes

sos - Second order section matrix

K-by-2 matrix

Second order section matrix, specified as a K-by-2 matrix. Each row of the matrix contains the coefficients for a biquadratic rational function

in $z^{(-1)}$. The z-transform of the K-th rational biquadratic system impulse response is

$$H_k(z) = \frac{B_k(1) + B_k(2)z^{-1} + B_k(3)z^{-2}}{A_k(1) + A_k(2)z^{-1} + A_k(3)z^{-2}}$$

The coefficients in the K-th row of the matrix, `sos`, are ordered as follows

$$[B_k(1)B_k(2)B_k(3)A_k(1)A_k(2)A_k(3)]$$

The frequency response of the filter is system function evaluated on the unit circle with

$$z = e^{i2\pi f}$$

Data Types
`single | double`
Complex Number Support: Yes

Name-Value Pair Arguments

Specify optional comma-separated pairs of `Name,Value` arguments. `Name` is the argument name and `Value` is the corresponding value. `Name` must appear inside single quotes (`' '`). You can specify several name and value pair arguments in any order as `Name1,Value1,...,NameN,ValueN`.

'Destination' - Destination for Simulink filter block

`'current'` (default) | `'new'` | user-defined string

Destination for the Simulink filter block, specified as a string. You can add the filter block to your current model with `'current'`, add the filter block to a new model with `'new'`, or specify the name of a a target subsystem.

Example: `filt2block(b,'subsystem','MyFilterBlock')`

Data Types
`char`

'BlockName' - Block name

string

Block name, specified as a string.

'OverwriteBlock' - Overwrite block

`false` (default) | `true`

Overwrite block, specified as a logical `false` or `true`. If you use a value for `'BlockName'` that is the same as an existing block, the value of `'OverwriteBlock'` determines whether the block is overwritten. The default value is `false`.

Data Types

`logical`

'MapCoefficientsToPorts' - Map coefficients to ports

`false` (default) | `true`

Map coefficients to ports, specified as a logical `false` or `true`.

Data Types

`logical`

'CoefficientNames' - Coefficient variable names

cell array of strings

Coefficient variable names, specified as a cell array. This name-value pair is only applicable when `'MapCoefficientsToPorts'` is `true`. The default values are `{'Num'}`, `{'Num','Den'}`, and `{'Num','Den','g'}` for FIR, IIR, and biquad filters.

Data Types

`cell`

'FrameBasedProcessing' - Frame-based or sample-based processing

`true` (default) | `false`

Frame-based or sample-based processing, specified as a logical `true` or `false`. The default is `true` and frame-based processing is used.

Data Types
`logical`

'OptimizeZeros' - Remove zero-gain blocks

`true` (default) | `false`

Remove zero-gain blocks, specified as a logical `true` or `false`. By default zero-gain blocks are removed.

Data Types
`logical`

'OptimizeOnes' - Replace unity-gain blocks with direct connection

`true` (default) | `false`

Replace unity-gain blocks with direct connection, specified as a logical `true` or `false`. The default is `true`.

Data Types
`logical`

'OptimizeNegativeOnes' - Replace negative unity-gain blocks with sign change

`true` (default) | `false`

Replace negative unity-gain blocks with a sign change at the nearest block, specified as a logical `true` or `false`. The default is `true`.

Data Types
`logical`

'OptimizeDelayChains' - Replace cascaded delays with a single delay

`true` (default) | `false`

Replace cascaded delays with a single delay, specified as a logical `true` or `false`. The default is `true`.

Data Types
`logical`

Examples

Generate Block from FIR Filter

Design an order 30 FIR filter using the window method. Specify the cutoff frequency of π/4 radians/sample. Create a Simulink block.

```
b = fir1(30,0.25);
filt2block(b)
```

Generate Block from IIR Filter

Design an order 30 IIR Butterworth filter. Specify the cutoff frequency of π/4 radians/sample. Create a Simulink block.

```
[b,a] = butter(30,0.25);
filt2block(b,a)
```

Generate FIR Filter with Direct Form I Transposed Structure

Design an order 30 FIR filter using the window method. Specify the cutoff frequency of π/4 radians/sample. Create a Simulink block with a direct form I transposed structure

```
b = fir1(30,0.25);
filt2block(b,'FilterStructure','directFormTransposed')
```

Generate IIR Filter with Direct Form I Structure

Design an order 30 IIR Butterworth filter. Specify the cutoff frequency of π/4 radians/sample. Create a Simulink block with a direct form I structure.

```
[b,a] = butter(30,0.25);
filt2block(b,a,'FilterStructure','directForm1')
```

Generate Simulink Subsytem Block from Second Order Section Matrix

Design a 5-th order Butterworth filter with a cutoff frequency of 0.2π radians/sample. Obtain the filter in biquad form and generate a Simulink subsystem block from the second order sections.

```
[z,p,k] = butter(5,0.2);
sos = zp2sos(z,p,k);
filt2block(sos,'subsystem')
```

Lowpass FIR Filter Block with Sample-Based Processing

Generate a Simulink subsystem block that implements an FIR lowpass filter using sum, gain, and delay blocks. Specify the input processing to be elements as channels by specifying `'FrameBasedProcessing'` as `false`.

```
B = fir1(30,.25);
filt2block(B,'subsystem','BlockName','Lowpass FIR',...
              'FrameBasedProcessing',false)
```

See Also

`realizemdl`

Purpose

Find local maxima

Syntax

```
pks = findpeaks(data)
[pks,locs] = findpeaks(data)
[...] = findpeaks(data,'Name',value)
```

Description

`pks = findpeaks(data)` returns local maxima or peaks, `pks`, in the input `data`. `data` requires a row or column vector with real-valued elements with a minimum length of three. `findpeaks` compares each element of `data` to its neighboring values. If an element of `data` is larger than both of its neighbors or equals `Inf`, the element is a local peak. If there are no local maxima, `pks` is an empty vector.

`[pks,locs] = findpeaks(data)` returns the indices of the local peaks.

`[...] = findpeaks(data,'Name',value)` accepts one or more comma-separated name/value pairs. Specify `'Name'` inside single quotes. `'Name'` is not case sensitive.

Input Arguments

Name-Value Pair Arguments

'MINPEAKHEIGHT'

Minimum peak height

Specify the minimum peak height as a real-valued scalar. `findpeaks` only returns peaks that exceed the `MINPEAKHEIGHT`. Sometimes, specifying a minimum peak height reduces processing time.

Default: `-Inf`

'MINPEAKDISTANCE'

Minimum peak separation

Specify the minimum peak distance, or minimum separation between peaks as a positive integer. You can use the `'MINPEAKDISTANCE'` option to specify that the algorithm ignore small peaks that occur in the neighborhood of a larger peak. When you specify a value for

'MINPEAKDISTANCE, the algorithm initially identifies all the peaks in the input data and sorts those peaks in descending order. Beginning with the largest peak, the algorithm ignores all identified peaks not separated by more than the value of 'MINPEAKDISTANCE'.

Default: 1

'THRESHOLD'

Minimum height difference

Specify the threshold height difference between a peak and its neighboring values as a positive real number. `findpeaks` only returns peaks that exceed their neighbors by at least the value of 'THRESHOLD'.

Default: 0

'NPEAKS'

Number of peaks

Specify the maximum number of peaks to return as a positive integer. `findpeaks` operates from the first element of the input data and terminates when the number of peaks reaches the value of 'NPEAKS'.

Default: Returns all peaks that meet the specified criteria

'SORTSTR'

Peak sorting

Specify whether to return the peaks in order. Possible values for 'SORTSTR' are 'ascend', 'descend', and 'none'. 'ascend' returns peaks in ascending, or increasing order from the smallest to largest value. The option 'descend' specifies peaks in descending order, from the largest to smallest value. Using 'none' returns peaks in the order they occur in the input data. Specify the value string in lowercase only.

Default: 'none'

Examples

Find peaks in a vector:

```
data = [2 12 4 6 9 4 3 1 19 7];
pks=findpeaks(data);
% returns the 1x3 vector [12 9 19];
```

Find peaks separated by more than three elements and return their locations:

```
data = [2 12 4 6 9 4 3 1 19 7];
[pks,locs]=findpeaks(data,'minpeakdistance',3);
% returns pks=[12 19]
% locs=[2 9]
```

Create a signal with 11 peaks. Find each peak and mark the peaks in a plot:

```
x = linspace(0,1,1024);
Pos = [0.1 0.13 0.15 0.23 0.25 0.40 ...
 0.44 0.65  0.76 0.78 0.81];
Hgt = [ 4  5   3   4  5  4.2 2.1 4.3  3.1 5.1 4.2];
Wdt = [.005 .005 .006 .01 .01 .03 .01 .01  .005 .008 .005];
PeakSig = zeros(size(x));
  for n =1:length(Pos)
    PeakSig = ...
PeakSig + Hgt(n)./( 1 + abs((x - Pos(n))./Wdt(n))).^4;
  end
% find peaks with defaults
[pks,locs] = findpeaks(PeakSig);
plot(x,PeakSig); hold on;
% offset values of peak heights for plotting
plot(x(locs),pks+0.05,'k^','markerfacecolor',[1 0 0]);
```

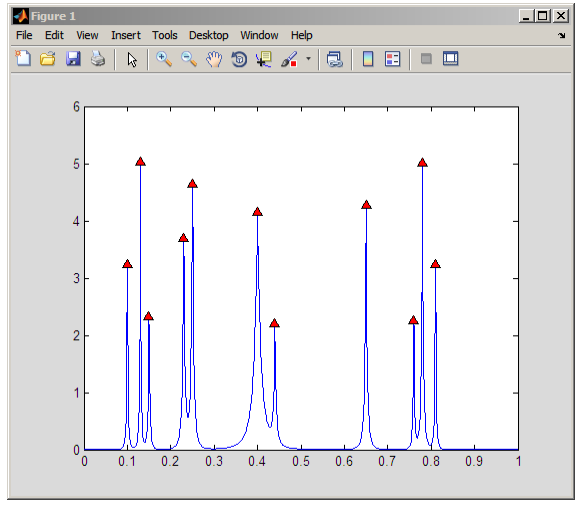


Figure 1
File Edit View Insert Tools Desktop Window Help
0 1 2 3 4 5 6
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1

Purpose

Window-based finite impulse response filter design

Syntax

```
b = fir1(n,Wn)
b = fir1(n,Wn,'ftype')
b = fir1(n,Wn,window)
b = fir1(n,Wn,'ftype',window)
b = fir1(...,'normalization')
```

Description

`fir1` implements the classical method of windowed linear-phase FIR digital filter design [1]. It designs filters in standard lowpass, highpass, bandpass, and bandstop configurations. By default the filter is normalized so that the magnitude response of the filter at the center frequency of the passband is 0 dB.

Note Use `fir2` for windowed filters with arbitrary frequency response.

`b = fir1(n,Wn)` returns row vector `b` containing the `n+1` coefficients of an order `n` lowpass FIR filter. This is a Hamming-window based, linear-phase filter with normalized cutoff frequency `Wn`. The output filter coefficients, `b`, are ordered in descending powers of z.

$$B(z) = b(1) + b(2)z^{-1} + \ldots + b(n+1)z^{-N}$$

`Wn` is a number between 0 and 1, where 1 corresponds to the Nyquist frequency.

If `Wn` is a two-element vector, `Wn = [w1 w2]`, `fir1` returns a bandpass filter with passband `w1` < ω< `w2`.

If `Wn` is a multi-element vector, `Wn = [w1 w2 w3 w4 w5 ... wn]`, `fir1` returns an order `n` multiband filter with bands `0` < ω< `w1`, `w1` < ω< `w2`, ..., `wn` < ω< `1`.

By default, the filter is scaled so that the center of the first passband has a magnitude of exactly 1 after windowing.

`b = fir1(n,Wn,'ftype')` specifies a filter type, where `'ftype'` is:

- `'high'` for a highpass filter with cutoff frequency `Wn`.
- `'stop'` for a bandstop filter, if `Wn = [w1 w2]`. The stopband frequency range is specified by this interval.
- `'DC-1'` to make the first band of a multiband filter a passband.
- `'DC-0'` to make the first band of a multiband filter a stopband.

`fir1` always uses an even filter order for the highpass and bandstop configurations. This is because for odd orders, the frequency response at the Nyquist frequency is 0, which is inappropriate for highpass and bandstop filters. If you specify an odd-valued `n`, `fir1` increments it by 1.

`b = fir1(n,Wn,window)` uses the window specified in column vector `window` for the design. The vector `window` must be `n+1` elements long. If no window is specified, `fir1` uses a Hamming window (see `hamming`) of length `n+1`.

`b = fir1(n,Wn,'ftype',window)` accepts both `'ftype'` and `window` parameters.

`b = fir1(...,'normalization')` specifies whether or not the filter magnitude is normalized. The string `'normalization'` can be the following:

- `'scale'` (default): Normalize the filter so that the magnitude response of the filter at the center frequency of the passband is 0 dB.
- `'noscale'`: Do not normalize the filter.

The group delay of the FIR filter designed by `fir1` is `n/2`.

Algorithms

`fir1` uses the window method of FIR filter design [1]. If $w(n)$ denotes a window, where $1 \le n \le N$, and the impulse response of the ideal filter is $h(n)$, then the windowed digital filter coefficients are given by

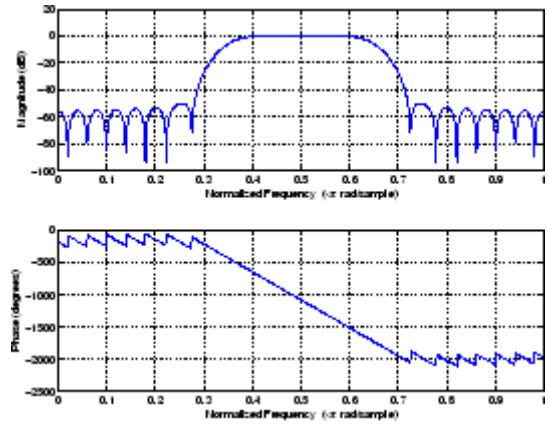
$$b(n) = w(n)h(n) \quad 1 \le n \le N$$

Examples

Example 1

Design a 48th-order FIR bandpass filter with passband $0.35 \le \omega \le 0.65$:

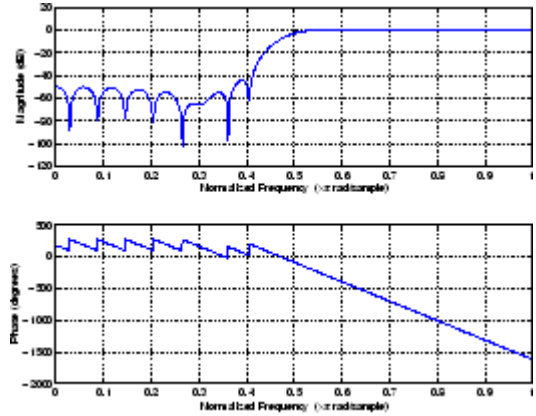
```
b = fir1(48,[0.35 0.65]);
freqz(b,1,512)
```



Example 2

The `chirp.mat` file contains a signal, `y`, that has most of its power above `fs/4`, or half the Nyquist frequency. Design a 34th-order FIR highpass filter to attenuate the components of the signal below `fs/4`. Use a cutoff frequency of 0.48 and a Chebyshev window with 30 dB of ripple:

```
load chirp          % Load y and fs.
b = fir1(34,0.48,'high',chebwin(35,30));
freqz(b,1,512)
```



References

[1] *Programs for Digital Signal Processing*, IEEE Press, New York, 1979. Algorithm 5.2.

See Also

`cfirpm` | `filter` | `fir2` | `fircls` | `fircls1` | `firls` | `freqz` | `kaiserord` | `firpm` | `window`

Purpose

Frequency sampling-based finite impulse response filter design

Syntax

```
b = fir2(n,f,m)
b = fir2(n,f,m,window)
b = fir2(n,f,m,npt)
b = fir2(n,f,m,npt,window)
b = fir2(n,f,m,npt,lap)
b = fir2(n,f,m,npt,lap,window)
```

Description

`fir2` designs frequency sampling-based digital FIR filters with arbitrarily shaped frequency response.

Note Use `fir1` for windows-based standard lowpass, bandpass, highpass, and bandstop configurations.

`b = fir2(n,f,m)` returns row vector `b` containing the `n+1` coefficients of an order `n` FIR filter. The frequency-magnitude characteristics of this filter match those given by vectors `f` and `m`:

- `f` is a vector of frequency points in the range from 0 to 1, where 1 corresponds to the Nyquist frequency. The first point of `f` must be `0` and the last point `1`. The frequency points must be in increasing order.
- `m` is a vector containing the desired magnitude response at the points specified in `f`.
- `f` and `m` must be the same length.
- Duplicate frequency points are allowed, corresponding to steps in the frequency response.

Use `plot(f,m)` to view the filter shape.

The output filter coefficients, `b`, are ordered in descending powers of z.

$$B(z) = b(1) + b(2)z^{-1} + \cdots + b(n+1)z^{-n}$$

`fir2` always uses an even filter order for configurations with a passband at the Nyquist frequency. This is because for odd orders, the frequency response at the Nyquist frequency is necessarily 0. If you specify an odd-valued `n`, `fir2` increments it by 1.

`b = fir2(n,f,m,window)` uses the window specified in the column vector `window`. The vector `window` must be `n+1` elements long. If no window is specified, `fir2` uses a Hamming window (see `hamming`) of length `n+1`.

`b = fir2(n,f,m,npt)` or

`b = fir2(n,f,m,npt,window)` specifies the number of points, `npt`, for the grid onto which `fir2` linearly interpolates the frequency response with or without the `window` specification. `npt` must be greater than 1/2 the filter order (`npt>n/2`). If desired, you can interpolate `f` and `m` before passing them to `fir2`.

`b = fir2(n,f,m,npt,lap)` and

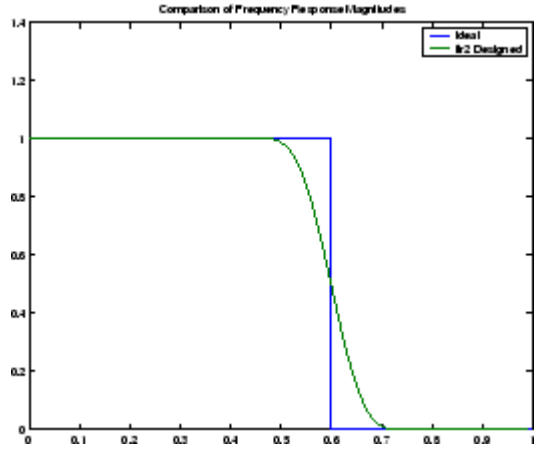
`b = fir2(n,f,m,npt,lap,window)` specify the size of the region, `lap`, that `fir2` inserts around duplicate frequency points, with or without a `window` specification.

See “Algorithms” on page 1-527 for more on `npt` and `lap`.

Examples

Design a 30th-order lowpass filter and overplot the desired frequency response with the actual frequency response:

```
f = [0 0.6 0.6 1]; m = [1 1 0 0];
b = fir2(30,f,m);
[h,w] = freqz(b,1,128);
plot(f,m,w/pi,abs(h))
legend('Ideal','fir2 Designed')
title('Comparison of Frequency Response Magnitudes')
```



Algorithms

The desired frequency response is linearly interpolated onto a dense, evenly spaced grid of length `npt`. `npt` is 512 by default. If two successive values of `f` are the same, a region of `lap` points is set up around this frequency to provide a smooth but steep transition in the requested frequency response. By default, `lap` is 25. The filter coefficients are obtained by applying an inverse fast Fourier transform to the grid and multiplying by a window; by default, this is a Hamming window.

References

[1] Mitra, S.K., *Digital Signal Processing A Computer Based Approach, First Edition*, McGraw-Hill, New York, 1998, pp. 462-468.

[2] Jackson, L.B., *Digital Filters and Signal Processing, Third Edition*, Kluwer Academic Publishers, Boston, 1996, pp. 301-307.

See Also

`butter` | `cheby1` | `cheby2` | `ellip` | `fir1` | `maxflat` | `firpm` | `yulewalk`

fircls

Purpose

Constrained least square, FIR multiband filter design

Syntax

```
b = fircls(n,f,amp,up,lo)
fircls(n,f,amp,up,lo,'design_flag')
```

Description

`b = fircls(n,f,amp,up,lo)` generates a length `n+1` linear phase FIR filter `b`. The frequency-magnitude characteristics of this filter match those given by vectors `f` and `amp`:

- `f` is a vector of transition frequencies in the range from 0 to 1, where 1 corresponds to the Nyquist frequency. The first point of `f` must be `0` and the last point `1`. The frequency points must be in increasing order.
- `amp` is a vector describing the piecewise constant desired amplitude of the frequency response. The length of `amp` is equal to the number of bands in the response and should be equal to `length(f)-1`.
- `up` and `lo` are vectors with the same length as `amp`. They define the upper and lower bounds for the frequency response in each band.

`fircls` always uses an even filter order for configurations with a passband at the Nyquist frequency (that is, highpass and bandstop filters). This is because for odd orders, the frequency response at the Nyquist frequency is necessarily 0. If you specify an odd-valued `n`, `fircls` increments it by 1.

`fircls(n,f,amp,up,lo,'design_flag')` enables you to monitor the filter design, where `'design_flag'` can be

- `'trace'`, for a textual display of the design error at each iteration step.
- `'plots'`, for a collection of plots showing the filter's full-band magnitude response and a zoomed view of the magnitude response in each sub-band. All plots are updated at each iteration step. The O's on the plot are the estimated extremals of the new iteration and the X's are the estimated extremals of the previous iteration, where the extremals are the peaks (maximum and minimum) of the filter

ripples. Only ripples that have a corresponding O and X are made equal.

- `'both'`, for both the textual display and plots.

Examples

Design an order 150 lowpass filter:

```
n=150;
f=[0 0.4 1];
a=[1 0];
up=[1.02 0.01];
lo =[0.98 -0.01];
b = fircls(n,f,a,up,lo,'both');   % Display plots of bands
%  The Bound Violations indicate iterations as
%  the design converges.
fvtool(b)                          % Display magnitude plot
```

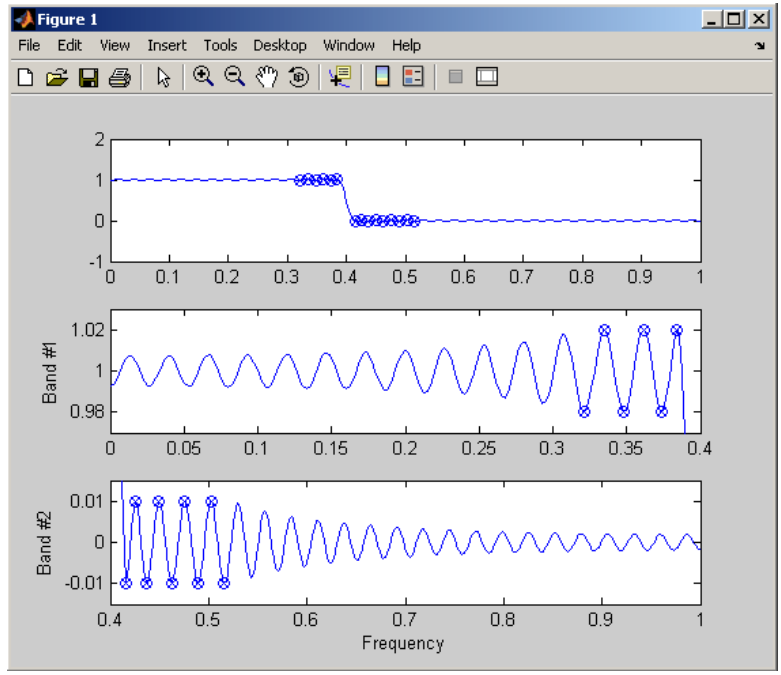
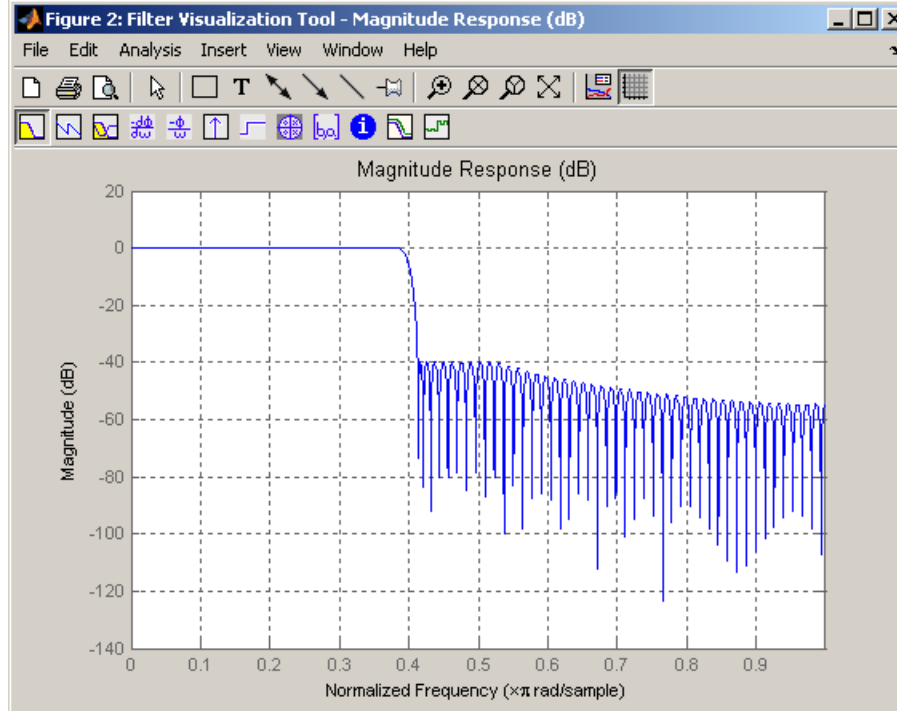


Figure 1
File Edit View Insert Tools Desktop Window Help
2
1
0
-1
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Band #1
1.02
1
0.98
0 0.05 0.1 0.15 0.2 0.25 0.3 0.35 0.4
Band #2
0.01
0
-0.01
0.4 0.5 0.6 0.7 0.8 0.9 1
Frequency



Note Normally, the lower value in the stopband will be specified as negative. By setting `lo` equal to `0` in the stopbands, a nonnegative frequency response amplitude can be obtained. Such filters can be spectrally factored to obtain minimum phase filters.

Algorithms

`fircls` uses an iterative least-squares algorithm to obtain an equiripple response. The algorithm is a multiple exchange algorithm that uses Lagrange multipliers and Kuhn-Tucker conditions on each iteration.

References

[1] Selesnick, I.W., M. Lang, and C.S. Burrus, “Constrained Least Square Design of FIR Filters without Specified Transition

Bands," *Proceedings of the IEEE Int. Conf. Acoust., Speech, Signal Processing, Vol. 2* (May 1995), pp. 1260-1263.

[2] Selesnick, I.W., M. Lang, and C.S. Burrus. "Constrained Least Square Design of FIR Filters without Specified Transition Bands." *IEEE Transactions on Signal Processing, Vol. 44*, No. 8 (August 1996).

See Also

`fircls1` | `firls` | `firpm`

Purpose

Constrained least square, lowpass and highpass, linear phase, FIR filter design

Syntax

```
b = fircls1(n,wo,dp,ds)
b = fircls1(n,wo,dp,ds,'high')
b = fircls1(n,wo,dp,ds,wt)
b = fircls1(n,wo,dp,ds,wt,'high')
b = fircls1(n,wo,dp,ds,wp,ws,k)
b = fircls1(n,wo,dp,ds,wp,ws,k,'high')
b = fircls1(n,wo,dp,ds,...,'design_flag')
```

Description

`b = fircls1(n,wo,dp,ds)` generates a lowpass FIR filter `b`, where `n+1` is the filter length, `wo` is the normalized cutoff frequency in the range between 0 and 1 (where 1 corresponds to the Nyquist frequency), `dp` is the maximum passband deviation from 1 (passband ripple), and `ds` is the maximum stopband deviation from 0 (stopband ripple).

`b = fircls1(n,wo,dp,ds,'high')` generates a highpass FIR filter `b`. `fircls1` always uses an even filter order for the highpass configuration. This is because for odd orders, the frequency response at the Nyquist frequency is necessarily 0. If you specify an odd-valued `n`, `fircls1` increments it by 1.

`b = fircls1(n,wo,dp,ds,wt)` and

`b = fircls1(n,wo,dp,ds,wt,'high')` specifies a frequency `wt` above which (for `wt > wo`) or below which (for `wt < wo`) the filter is guaranteed to meet the given band criterion. This will help you design a filter that meets a passband or stopband edge requirement. There are four cases:

- Lowpass:
  - `0 < wt < wo < 1`: the amplitude of the filter is within `dp` of 1 over the frequency range $0 < \omega < wt$.
  - `0 < wo < wt < 1`: the amplitude of the filter is within `ds` of 0 over the frequency range $wt < \omega < 1$.
- Highpass:

- `0 < wt < wo < 1`: the amplitude of the filter is within `ds` of 0 over the frequency range `0 < ω < wt`.
- `0 < wo < wt < 1`: the amplitude of the filter is within `dp` of 1 over the frequency range `wt < ω < 1`.

`b = fircls1(n,wo,dp,ds,wp,ws,k)` generates a lowpass FIR filter `b` with a weighted function, where `n+1` is the filter length, `wo` is the normalized cutoff frequency, `dp` is the maximum passband deviation from 1 (passband ripple), and `ds` is the maximum stopband deviation from 0 (stopband ripple). `wp` is the passband edge of the L2 weight function and `ws` is the stopband edge of the L2 weight function, where `wp < wo < ws`. `k` is the ratio (passband L2 error)/(stopband L2 error)

$$k = \frac{\int_0^{w_p} |A(\omega) - D(\omega)|^2 \, d\omega}{\int_{w_z}^{\pi} |A(\omega) - D(\omega)|^2 \, d\omega}$$

`b = fircls1(n,wo,dp,ds,wp,ws,k,'high')` generates a highpass FIR filter `b` with a weighted function, where `ws < wo < wp`.

`b = fircls1(n,wo,dp,ds,...,'design_flag')` enables you to monitor the filter design, where `'design_flag'` can be

- `'trace'`, for a textual display of the design table used in the design
- `'plots'`, for plots of the filter's magnitude, group delay, and zeros and poles. All plots are updated at each iteration step. The O's on the plot are the estimated extremals of the new iteration and the X's are the estimated extremals of the previous iteration, where the extremals are the peaks (maximum and minimum) of the filter ripples. Only ripples that have a corresponding O and X are made equal.
- `'both'`, for both the textual display and plots

Note In the design of very narrow band filters with small `dp` and `ds`, there may not exist a filter of the given length that meets the specifications.

Examples

Design an order 55 lowpass filter with a cutoff frequency at 0.3:

```
n = 55;     wo = 0.3;
dp = 0.02;  ds = 0.008;
b = fircls1(n,wo,dp,ds,'both');    % Display plots of bands
    Bound Violation = 0.0870385343920
    Bound Violation = 0.0149343456540
    Bound Violation = 0.0056513587932
    Bound Violation = 0.0001056264205
    Bound Violation = 0.0000967624352
    Bound Violation = 0.0000000226538
    Bound Violation = 0.0000000000038
%  The above Bound Violations indicate iterations as
%  the design converges.
fvtool(b)            % Display magnitude plot
```

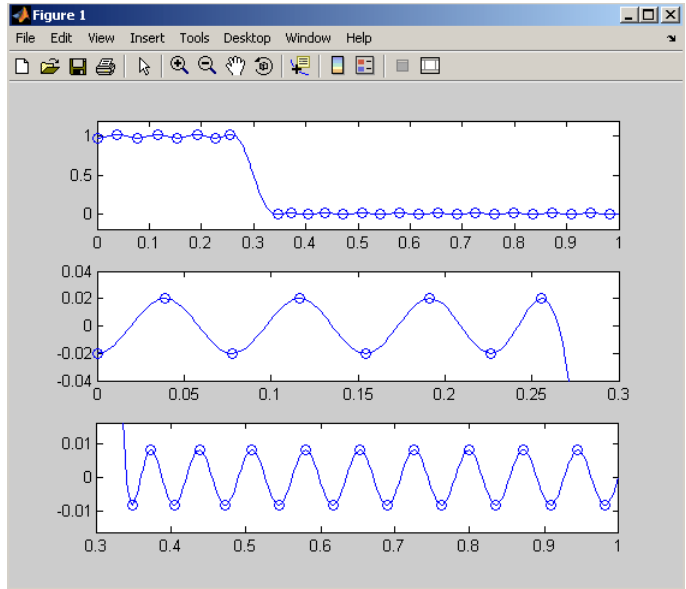


Figure 1
File Edit View Insert Tools Desktop Window Help
1
0.5
0
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
0.04
0.02
0
-0.02
-0.04
0 0.05 0.1 0.15 0.2 0.25 0.3
0.01
0
-0.01
0.3 0.4 0.5 0.6 0.7 0.8 0.9 1

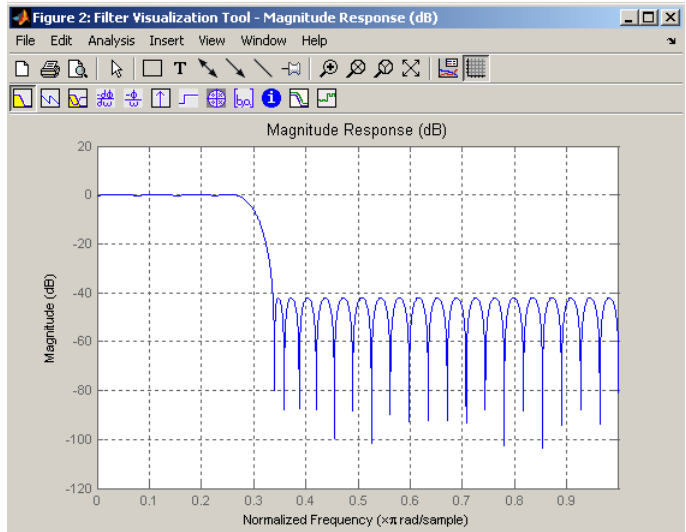


Figure 2: Filter Visualization Tool - Magnitude Response (dB)
File Edit Analysis Insert View Window Help
Magnitude Response (dB)
Magnitude (dB)
20
0
-20
-40
-60
-80
-100
-120
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9
Normalized Frequency (×π rad/sample)

Algorithms

`fircls1` uses an iterative least-squares algorithm to obtain an equiripple response. The algorithm is a multiple exchange algorithm that uses Lagrange multipliers and Kuhn-Tucker conditions on each iteration.

References

[1] Selesnick, I.W., M. Lang, and C.S. Burrus, "Constrained Least Square Design of FIR Filters without Specified Transition Bands," *Proceedings of the IEEE Int. Conf. Acoust., Speech, Signal Processing, Vol. 2* (May 1995), pp.1260-1263.

[2] Selesnick, I.W., M. Lang, and C.S. Burrus, "Constrained Least Square Design of FIR Filters without Specified Transition Bands," *IEEE Transactions on Signal Processing, Vol. 44*, No. 8 (August 1996).

See Also

`fircls` | `firls` | `firpm`

firls

Purpose

Least square linear-phase FIR filter design

Syntax

```
b = firls(n,f,a)
b = firls(n,f,a,w)
b = firls(n,f,a,'ftype')
b = firls(n,f,a,w,'ftype')
```

Description

`firls` designs a linear-phase FIR filter that minimizes the weighted, integrated squared error between an ideal piecewise linear function and the magnitude response of the filter over a set of desired frequency bands.

`b = firls(n,f,a)` returns row vector `b` containing the `n+1` coefficients of the order `n` FIR filter whose frequency-amplitude characteristics approximately match those given by vectors `f` and `a`. The output filter coefficients, or "taps," in `b` obey the symmetry relation.

$$b(k) = b(n+2-k), \qquad k = 1,\ldots,n+1$$

These are type I (`n` odd) and type II (`n` even) linear-phase filters. Vectors `f` and `a` specify the frequency-amplitude characteristics of the filter:

- `f` is a vector of pairs of frequency points, specified in the range between 0 and 1, where 1 corresponds to the Nyquist frequency. The frequencies must be in increasing order. Duplicate frequency points are allowed and, in fact, can be used to design a filter exactly the same as those returned by the `fir1` and `fir2` functions with a rectangular (`rectwin`) window.
- `a` is a vector containing the desired amplitude at the points specified in `f`.

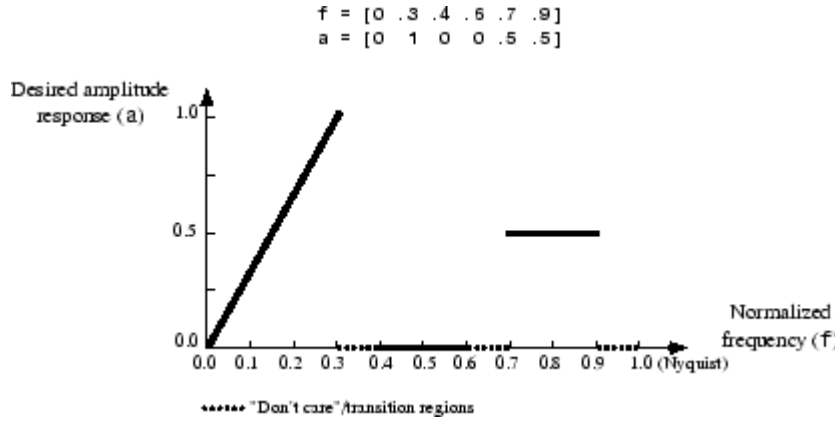
  The desired amplitude function at frequencies between pairs of points ($f(k)$, $f(k+1)$) for k odd is the line segment connecting the points ($f(k)$, $a(k)$) and ($f(k+1)$, $a(k+1)$).

  The desired amplitude function at frequencies between pairs of points ($f(k)$, $f(k+1)$) for k even is unspecified. These are transition or "don't care" regions.

- `f` and `a` are the same length. This length must be an even number.

`firls` always uses an even filter order for configurations with a passband at the Nyquist frequency. This is because for odd orders, the frequency response at the Nyquist frequency is necessarily 0. If you specify an odd-valued `n`, `firls` increments it by 1.

The figure below illustrates the relationship between the `f` and `a` vectors in defining a desired amplitude response.



`b = firls(n,f,a,w)` uses the weights in vector `w` to weight the fit in each frequency band. The length of `w` is half the length of `f` and `a`, so there is exactly one weight per band.

`b = firls(n,f,a,'ftype')` and

`b = firls(n,f,a,w,'ftype')` specify a filter type, where `'ftype'` is:

- `'hilbert'` for linear-phase filters with odd symmetry (type III and type IV). The output coefficients in `b` obey the relation

  $$b(k) = -b(n + 2 - k),\ k = 1, \ldots, n + 1.$$

  This class of filters includes the Hilbert transformer, which has a desired amplitude of 1 across the entire band.

- `'differentiator'` for type III and type IV filters, using a special weighting technique. For nonzero amplitude bands, the integrated squared error has a weight of $(1/f)^2$ so that the error at low frequencies is much smaller than at high frequencies. For FIR differentiators, which have an amplitude characteristic proportional to frequency, the filters minimize the relative integrated squared error (the integral of the square of the ratio of the error to the desired amplitude).

Examples

Example 1

Design an order 255 lowpass filter with transition band:

```
b = firls(255,[0 0.25 0.3 1],[1 1 0 0]);
```

Example 2

Design a 31 coefficient differentiator:

```
b = firls(30,[0 0.9],[0 0.9*pi],'differentiator');
```

An ideal differentiator has the response

$$D(w) = jw$$

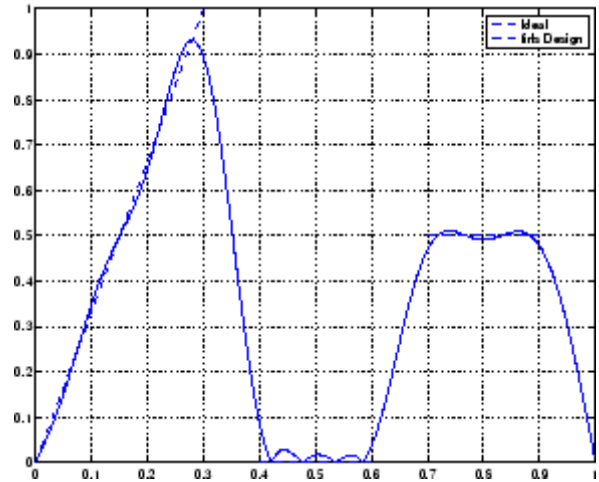
The amplitudes include a pi multiplier because the frequencies are normalized by pi.

Example 3

Design a 24th-order anti-symmetric filter with piecewise linear passbands and plot the desired and actual frequency response:

```
F = [0 0.3  0.4 0.6  0.7 0.9];
A = [0  1   0  0  0.5 0.5];
b = firls(24,F,A,'hilbert');
for i=1:2:6,
   plot([F(i) F(i+1)],[A(i) A(i+1)],'--'), hold on
end
[H,f] = freqz(b,1,512,2);
plot(f,abs(H)), grid on, hold off
```

```
legend('Ideal','firls Design')
```



Algorithms

Reference [1] describes the theoretical approach behind `firls`. The function solves a system of linear equations involving an inner product matrix of size roughly `n/2` using the MATLAB \ operator.

This function designs type I, II, III, and IV linear-phase filters. Type I and II are the defaults for n even and odd respectively, while the `'hilbert'` and `'differentiator'` flags produce type III (n even) and IV (n odd) filters. The various filter types have different symmetries and constraints on their frequency responses (see [2] for details).

Linear Phase Filter Type	Filter Order	Symmetry of Coefficients	Response H(f), f = 0	Response H(f), f = 1 (Nyquist)
Type I	Even	$b(k) = b(n+2-k), k = 1,\ldots,n+1$	No restriction	No restriction
Type II	Even	$b(k) = b(n+2-k), k = 1,\ldots,n+1$	No restriction	$H(1) = 0$

Linear Phase Filter Type	Filter Order	Symmetry of Coefficients	Response H(f), f = 0	Response H(f), f = 1 (Nyquist)
Type III	Odd	$b(k) = -b(n+2-k), k = 1,...,n+1$	$H(0) = 0$	$H(1) = 0$
Type IV	Odd	$b(k) = -b(n+2-k), k = 1,...,n+1$	$H(0) = 0$	No restriction

Diagnostics

One of the following diagnostic messages is displayed when an incorrect argument is used:

```
F must be even length.
F and A must be equal lengths.
Requires symmetry to be 'hilbert' or 'differentiator'.
Requires one weight per band.
Frequencies in F must be nondecreasing.
Frequencies in F must be in range [0,1].
```

A more serious warning message is

```
Warning: Matrix is close to singular or badly scaled.
```

This tends to happen when the product of the filter length and transition width grows large. In this case, the filter coefficients b might not represent the desired filter. You can check the filter by looking at its frequency response.

References

[1] Parks, T.W., and C.S. Burrus, *Digital Filter Design*, John Wiley & Sons, 1987, pp. 54-83.

[2] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, 1989, pp. 256-266.

See Also

`fir1` | `fir2` | `firrcos` | `firpm`

Purpose

Parks-McClellan optimal FIR filter design

Syntax

```
b = firpm(n,f,a)
b = firpm(n,f,a,w)
b = firpm(n,f,a, 'ftype')
b = firpm(n,f,a,w, 'ftype')
b = firpm(...,{lgrid})
[b,err] = firpm(...)
[b,err,res] = firpm(...)
b = firpm(n,f,@fresp,w)
b = firpm(n,f,@fresp,w,'ftype')
```

Description

`firpm` designs a linear-phase FIR filter using the Parks-McClellan algorithm [1]. The Parks-McClellan algorithm uses the Remez exchange algorithm and Chebyshev approximation theory to design filters with an optimal fit between the desired and actual frequency responses. The filters are optimal in the sense that the maximum error between the desired frequency response and the actual frequency response is minimized. Filters designed this way exhibit an equiripple behavior in their frequency responses and are sometimes called *equiripple* filters. `firpm` exhibits discontinuities at the head and tail of its impulse response due to this equiripple nature.

`b = firpm(n,f,a)` returns row vector `b` containing the `n+1` coefficients of the order `n` FIR filter whose frequency-amplitude characteristics match those given by vectors `f` and `a`.

The output filter coefficients (taps) in `b` obey the symmetry relation:

$$b(k) = b(n+2-k), \quad k = 1,\ldots,n+1$$

Vectors `f` and `a` specify the frequency-magnitude characteristics of the filter:

- `f` is a vector of pairs of normalized frequency points, specified in the range between 0 and 1, where 1 corresponds to the Nyquist frequency. The frequencies must be in increasing order.

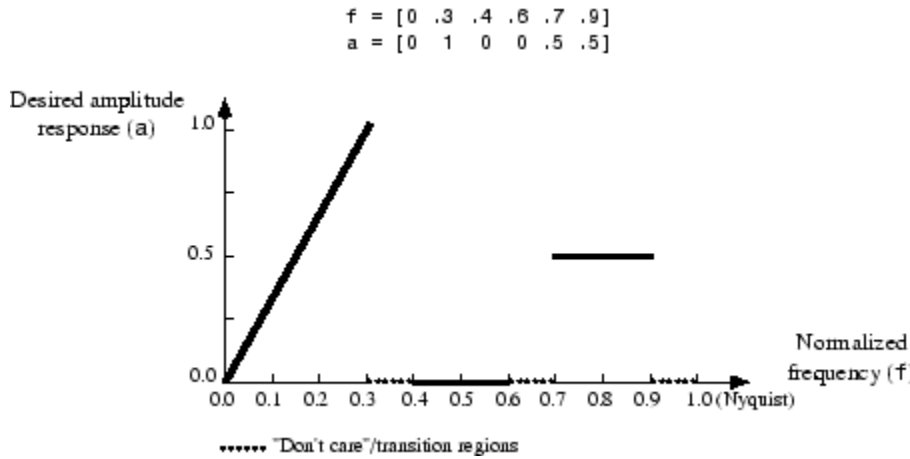
- `a` is a vector containing the desired amplitudes at the points specified in f.

  The desired amplitude at frequencies between pairs of points ($f(k)$, $f(k+1)$) for k odd is the line segment connecting the points ($f(k)$, $a(k)$) and ($f(k+1)$, $a(k+1)$).

  The desired amplitude at frequencies between pairs of points ($f(k)$, $f(k+1)$) for k even is unspecified. The areas between such points are transition or "don't care" regions.

- `f` and `a` must be the same length. The length must be an even number.

The relationship between the `f` and `a` vectors in defining a desired frequency response is shown in the illustration below.



`firpm` always uses an even filter order for configurations with even symmetry and a nonzero passband at the Nyquist frequency. This is because for impulse responses exhibiting even symmetry and odd orders, the frequency response at the Nyquist frequency is necessarily 0. If you specify an odd-valued `n`, `firpm` increments it by 1.

`b = firpm(n,f,a,w)` uses the weights in vector `w` to weight the fit in each frequency band. The length of `w` is half the length of `f` and `a`, so there is exactly one weight per band.

Note `b = firpm(n,f,a,w)` is a synonym for `b = firpm(n,f,{@firpmfrf,a},w)`, where, `@firpmfrf` is the predefined frequency response function handle for `firpm`. If desired, you can write your own response function. Use `help private/firpmfrf` for information.

`b = firpm(n,f,a, 'ftype')` and

`b = firpm(n,f,a,w, 'ftype')` specify a filter type, where `'ftype'` is

- `'hilbert'`, for linear-phase filters with odd symmetry (type III and type IV)

  The output coefficients in `b` obey the relation $b(k) = -b(n+2-k)$, $k = 1, \ldots, n+1$. This class of filters includes the Hilbert transformer, which has a desired amplitude of 1 across the entire band.

  For example,

  ```
  h = firpm(30,[0.1 0.9],[1 1],'hilbert');
  ```

  designs an approximate FIR Hilbert transformer of length 31.

- `'differentiator'`, for type III and type IV filters, using a special weighting technique

  For nonzero amplitude bands, it weights the error by a factor of $1/f$ so that the error at low frequencies is much smaller than at high frequencies. For FIR differentiators, which have an amplitude characteristic proportional to frequency, these filters minimize the maximum relative error (the maximum of the ratio of the error to the desired amplitude).

`b = firpm(...,{lgrid})` uses the integer `lgrid` to control the density of the frequency grid, which has roughly `(lgrid*n)/(2*bw)` frequency

points, where `bw` is the fraction of the total frequency band interval [0,1] covered by `f`. Increasing `lgrid` often results in filters that more exactly match an equiripple filter, but that take longer to compute. The default value of `16` is the minimum value that should be specified for `lgrid`. Note that the `{lgrid}` argument must be a 1-by-1 cell array.

`[b,err] = firpm(...)` returns the maximum ripple height in err.

`[b,err,res] = firpm(...)` returns a structure `res` with the following fields.

`res.fgrid`	Frequency grid vector used for the filter design optimization
`res.des`	Desired frequency response for each point in `res.fgrid`
`res.wt`	Weighting for each point in `opt.fgrid`
`res.H`	Actual frequency response for each point in `res.fgrid`
`res.error`	Error at each point in `res.fgrid` (`res.des-res.H`)
`res.iextr`	Vector of indices into `res.fgrid` for extremal frequencies
`res.fextr`	Vector of extremal frequencies

You can also use `firpm` to write a function that defines the desired frequency response. The predefined frequency response function handle for `firpm` is `@firpmfrf`, which designs a linear-phase FIR filter.

`b = firpm(n,f,@fresp,w)` returns row vector `b` containing the `n+1` coefficients of the order `n` FIR filter whose frequency-amplitude characteristics best approximate the response returned by function handle `@fresp`. The function is called from within `firpm` with the following syntax.

```
[dh,dw] = fresp(n,f,gf,w)
```

The arguments are similar to those for `firpm`:

- `n` is the filter order.
- `f` is the vector of normalized frequency band edges that appear monotonically between 0 and 1, where 1 is the Nyquist frequency.
- `gf` is a vector of grid points that have been linearly interpolated over each specified frequency band by `firpm`. `gf` determines the frequency grid at which the response function must be evaluated, and contains the same data returned by `cfirpm` in the `fgrid` field of the `opt` structure.
- `w` is a vector of real, positive weights, one per band, used during optimization. `w` is optional in the call to `firpm`; if not specified, it is set to unity weighting before being passed to `fresp`.
- `dh` and `dw` are the desired complex frequency response and band weight vectors, respectively, evaluated at each frequency in grid `gf`.

`b = firpm(n,f,@fresp,w,'ftype')` designs antisymmetric (odd) filters, where `'ftype'` is either `'d'` for a differentiator or `'h'` for a Hilbert transformer. If you do not specify an *`ftype`*, a call is made to *`fresp`* to determine the default symmetry property `sym`. This call is made using the syntax.

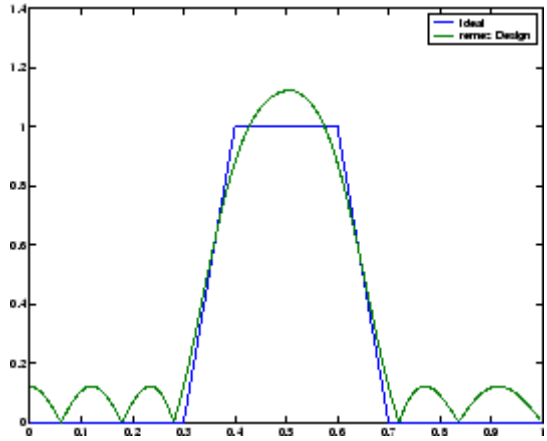
```
sym = fresp('defaults',{n,f,[],w,p1,p2,...})
```

The arguments `n`, `f`, `w`, etc., may be used as necessary in determining an appropriate value for `sym`, which `firpm` expects to be either `'even'` or `'odd'`. If *`fresp`* does not support this calling syntax, `firpm` defaults to even symmetry.

Examples

Graph the desired and actual frequency responses of a 17th-order Parks-McClellan bandpass filter:

```
f = [0 0.3 0.4 0.6 0.7 1]; a = [0 0 1 1 0 0];
b = firpm(17,f,a);
[h,w] = freqz(b,1,512);
plot(f,a,w/pi,abs(h))
legend('Ideal','firpm Design')
```



Algorithms

`firpm` is a MEX-file version of the original Fortran code from [1], altered to design arbitrarily long filters with arbitrarily many linear bands.

`firpm` designs type I, II, III, and IV linear-phase filters. Type I and type II are the defaults for `n` even and `n` odd, respectively, while type III (`n` even) and type IV (`n` odd) are obtained with the `'hilbert'` and `'differentiator'` flags. The different types of filters have different symmetries and certain constraints on their frequency responses (see [5] for more details).

Linear Phase Filter Type	Filter Order	Symmetry of Coefficients	Response H(f), f = 0	Response H(f), f = 1 (Nyquist)
Type I	Even	even: $b(k) = b(n+2-k), k = 1,\ldots,n+1$	No restriction	No restriction
Type II	Odd	even: $b(k) = b(n+2-k), k = 1,\ldots,n+1$	No restriction	*H(1)=0*`firpm` increments the filter order by 1 if you attempt

Linear Phase Filter Type	Filter Order	Symmetry of Coefficients	Response H(f), f = 0	Response H(f), f = 1 (Nyquist)
				to construct a type II filter with a nonzero passband at the Nyquist frequency.
Type III	Even	odd: $b(k) = -b(n+2-k), k = 1,...,n+1$	$H(0) = 0$	$H(1) = 0$
Type IV	Odd	odd: $b(k) = -b(n+2-k), k = 1,...,n+1$	$H(0) = 0$	No restriction

Diagnostics

If you get the following warning message,

```
-- Failure to Converge --
Probable cause is machine rounding error.
```

it is possible that the filter design may still be correct. Verify the design by checking its frequency response.

References

[1] *Programs for Digital Signal Processing*, IEEE Press, New York, 1979, Algorithm 5.1.

[2] *Selected Papers in Digital Signal Processing*, II, IEEE Press, New York, 1979.

[3] Parks, T.W., and C.S. Burrus, *Digital Filter Design*, John Wiley & Sons, New York:, 1987, p. 83.

[4] Rabiner, L.R., J.H. McClellan, and T.W. Parks, “FIR Digital Filter Design Techniques Using Weighted Chebyshev Approximations,” Proc. IEEE 63 (1975).

[5] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, Englewood Cliffs, NJ, 1989, pp. 256-266.

See Also

`butter` | `cheby1` | `cheby2` | `cfirpm` | `ellip` | `fir1` | `fir2` | `fircls` | `fircls1` | `firls` | `firrcos` | `firpmord` | `function_handle` | `yulewalk`

Purpose

Parks-McClellan optimal FIR filter order estimation

Syntax

```
[n,fo,ao,w] = firpmord(f,a,dev)
[n,fo,ao,w] = firpmord(f,a,dev,fs)
c = firpmord(f,a,dev,fs,'cell')
```

Description

`[n,fo,ao,w] = firpmord(f,a,dev)` finds the approximate order, normalized frequency band edges, frequency band amplitudes, and weights that meet input specifications `f`, `a`, and `dev`.

- `f` is a vector of frequency band edges (between 0 and $F_s/2$, where F_s is the sampling frequency), and `a` is a vector specifying the desired amplitude on the bands defined by `f`. The length of `f` is two less than twice the length of `a`. The desired function is piecewise constant.
- `dev` is a vector the same size as `a` that specifies the maximum allowable deviation or ripples between the frequency response and the desired amplitude of the output filter for each band.

Use `firpm` with the resulting order `n`, frequency vector `fo`, amplitude response vector `ao`, and weights `w` to design the filter `b` which approximately meets the specifications given by `firpmord` input parameters `f`, `a`, and `dev`.

```
b = firpm(n,fo,ao,w)
```

`[n,fo,ao,w] = firpmord(f,a,dev,fs)` specifies a sampling frequency `fs`. `fs` defaults to 2 Hz, implying a Nyquist frequency of 1 Hz. You can therefore specify band edges scaled to a particular application's sampling frequency.

In some cases `firpmord` underestimates the order `n`. If the filter does not meet the specifications, try a higher order such as `n+1` or `n+2`.

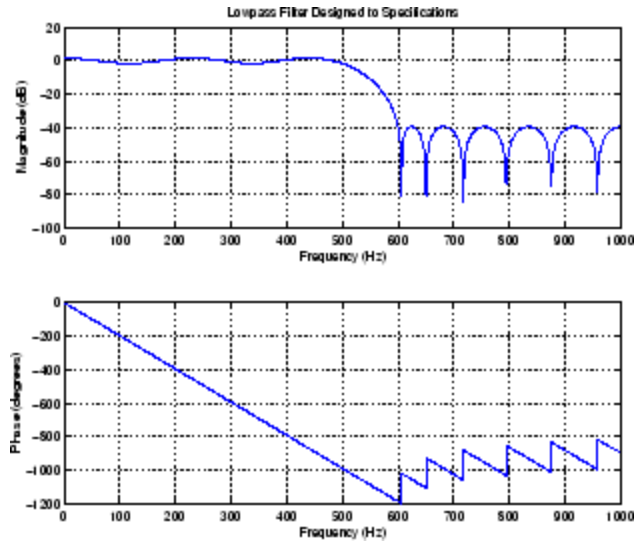
`c = firpmord(f,a,dev,fs,'cell')` generates a cell-array whose elements are the parameters to `firpm`.

Examples

Example 1

Design a minimum-order lowpass filter with a 500 Hz passband cutoff frequency and 600 Hz stopband cutoff frequency, with a sampling frequency of 2000 Hz, at least 40 dB attenuation in the stopband, and less than 3 dB of ripple in the passband:

```
rp = 3;           % Passband ripple
rs = 40;          % Stopband ripple
fs = 2000;        % Sampling frequency
f = [500 600];    % Cutoff frequencies
a = [1 0];        % Desired amplitudes
% Compute deviations
dev = [(10^(rp/20)-1)/(10^(rp/20)+1)  10^(-rs/20)];
[n,fo,ao,w] = firpmord(f,a,dev,fs);
b = firpm(n,fo,ao,w);
freqz(b,1,1024,fs);
title('Lowpass Filter Designed to Specifications');
```



Note that the filter falls slightly short of meeting the stopband attenuation and passband ripple specifications. Using `n+1` in the call to `firpm` instead of `n` achieves the desired amplitude characteristics.

Example 2

Design a lowpass filter with a 1500 Hz passband cutoff frequency and 2000 Hz stopband cutoff frequency, with a sampling frequency of 8000 Hz, a maximum stopband amplitude of 0.1, and a maximum passband error (ripple) of 0.01:

```
[n,fo,ao,w] = firpmord([1500 2000],[1 0],[0.01 0.1],8000 );
b = firpm(n,fo,ao,w);
```

This is equivalent to

```
c = firpmord( [1500 2000],[1 0],[0.01 0.1],8000,'cell');
b = firpm(c{:});
```

Note In some cases, `firpmord` underestimates or overestimates the order `n`. If the filter does not meet the specifications, try a higher order such as `n+1` or `n+2`.

Results are inaccurate if the cutoff frequencies are near 0 or the Nyquist frequency.

Algorithms

`firpmord` uses the algorithm suggested in [1]. This method is inaccurate for band edges close to either 0 or the Nyquist frequency (`fs/2`).

References

[1] Rabiner, L.R., and O. Herrmann, "The Predictability of Certain Optimum Finite Impulse Response Digital Filters," *IEEE Trans. on Circuit Theory, Vol. CT-20*, No. 4 (July 1973), pp. 401-408.

[2] Rabiner, L.R., and B. Gold. *Theory and Application of Digital Signal Processing. Englewood Cliffs*, NJ: Prentice-Hall, 1975, pp. 156-157.

See Also

buttord | cheb1ord | cheb2ord | ellipord | kaiserord | firpm

Purpose

Raised cosine FIR filter design

Syntax

```
b = firrcos(n,Fc,df)
b = firrcos(n,Fc,df,Fs)
b = firrcos(n,Fc,df,Fs,'bandwidth')
b = firrcos(n,Fc,df,Fs,'type')
b = firrcos(...,'type',delay)
b = firrcos(...,'type',delay,window)
b = firrcos(n,Fc,r,Fs,'rolloff')
b = firrcos(...,'rolloff','type')
[b,a] = firrcos(...)
```

Description

`b = firrcos(n,Fc,df)` uses a default sampling frequency of `Fs = 2`.

`b = firrcos(n,Fc,df,Fs)` or, equivalently,

`b = firrcos(n,Fc,df,Fs,'bandwidth')` returns an order `n` lowpass linear-phase FIR filter with a raised cosine transition band. The order `n` must be even. The filter has cutoff frequency `Fc`, transition bandwidth `df`, and sampling frequency `Fs`, all in hertz. `df` must be small enough so that `Fc ± df/2` is between `0` and `Fs/2`. The coefficients in `b` are normalized so that the nominal passband gain is always equal to 1.

`b = firrcos(n,Fc,df,Fs,'type')` designs either a normal raised cosine filter or a square root raised cosine filter according to how you specify the string `'type'`. Specify `'type'` as:

- `'normal'`, for a regular raised cosine filter. This is the default, and is also in effect when the `'type'` argument is left empty, `[]`, or unspecified.
- `'sqrt'`, for a square root raised cosine filter.

`b = firrcos(...,'type',delay)` specifies an integer delay in the range `[0,n+1]`. The default is `n/2` for all `n`.

`b = firrcos(...,'type',delay,window)` applies a length `n+1` window to the designed filter to reduce the ripple in the frequency response. `window` must be a length `n+1` column vector. If no window is specified, a

rectangular (`rectwin`) window is used. Care must be exercised when using a window with a delay other than the default.

`b = firrcos(n,Fc,r,Fs,'rolloff')` interprets the third argument, `r`, as the rolloff factor instead of the transition bandwidth, `df`. `r` must be in the range `[0,1]`.

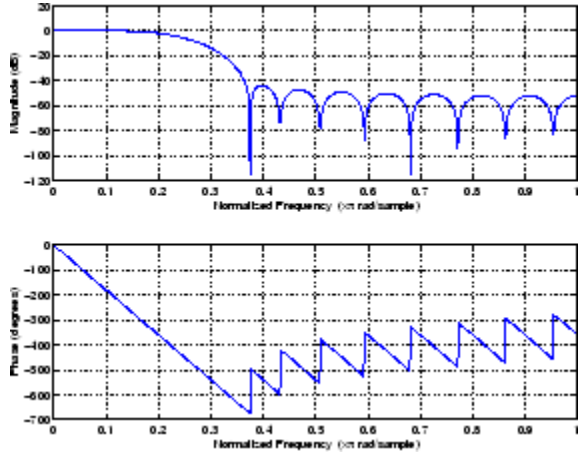
`b = firrcos(...,'rolloff','type')` specifies the type of raised cosine filter.

`[b,a] = firrcos(...)` always returns `a = 1`.

Examples

Design an order 20 raised cosine FIR filter with cutoff frequency 0.25 of the Nyquist frequency and a transition bandwidth of 0.25:

```
h = firrcos(20,0.25,0.25);
freqz(h,1)
```



See Also

`fir1` | `fir2` | `firls` | `firpm`

Purpose

Type of linear phase FIR filter

Syntax

```
t = firtype(b)
t = firtype(hd)
t = firtype(hm)
t = firtype(hs)
```

Description

`t = firtype(b)` determines the type, `t`, (1 through 4) of an FIR filter with coefficients, `b`. The filter must be real and have linear phase.

`t = firtype(hd)` determines the type of a discrete-time FIR filter object `hd`. The filter must be real and have linear phase.

`t = firtype(hm)` determines the type of the multirate filter object `hm`. The filter must be real and have linear phase. 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. You must have the DSP System Toolbox software to use this syntax.

`t = firtype(hs)` determines the type of the FIR filter System object™ `hs`. The filter must be real and have linear phase. You must have the DSP System Toolbox software to use this syntax.

Input Arguments

b

vector

Filter coefficients for the FIR filter, specified as a double- or single-precision real-valued row or column vector.

hd

`dfilt` filter object.

hm

Multirate `mfilt` filter object. Requires DSP System Toolbox.

hs

Filter System object. Requires DSP System Toolbox. The function supports the following filters .

- `dsp.FIRFilter`
- `dsp.BiquadFilter`
- `dsp.FIRInterpolator`
- `dsp.CICInterpolator`
- `dsp.FIRDecimator`
- `dsp.CICDecimator`
- `dsp.FIRRateConverter`

Output Arguments

t

Filter type. `t` is either 1, 2, 3, or 4. These types 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

Examples

Determine the filter type for an FIR filter designed using the window method. Plot the impulse response.

```
b1 = fir1(5,0.5);
t = firtype(b1)
stem(0:5,b1); set(gca,'xtick',0:5)
```

Determine the type of the default interpolator for L=4. Requires DSP System Toolbox .

```
l = 4;
hm = mfilt.firinterp(l);
t = firtype(hm)
```

See Also

`islinphase`

Purpose

Flat Top weighted window

Syntax

```
w = flattopwin(L)
w = flattopwin(L,sflag)
```

Description

Flat Top windows have very low passband ripple (< 0.01 dB) and are used primarily for calibration purposes. Their bandwidth is approximately 2.5 times wider than a Hann window.

`w = flattopwin(L)` returns the `L`-point symmetric flat top window in column vector `w`.

`w = flattopwin(L,sflag)` returns the `L`-point symmetric flat top window using `sflag` window sampling, where `sflag` is either `'symmetric'` or `'periodic'`. The `'periodic'` flag is useful for DFT/FFT purposes, such as in spectral analysis. The DFT/FFT contains an implicit periodic extension and the periodic flag enables a signal windowed with a periodic window to have perfect periodic extension. When `'periodic'` is specified, `flattopwin` computes a length `L+1` window and returns the first `L` points. When using windows for filter design, the `'symmetric'` flag should be used.

Algorithms

Flat top windows are summations of cosines. The coefficients of a flat top window are computed from the following equation

$$w(n) = a_0 - a_1\cos\left(\frac{2\pi n}{N}\right) + a_2\cos\left(\frac{4\pi n}{N}\right) - a_3\cos\left(\frac{6\pi n}{N}\right) + a_4\cos\left(\frac{8\pi n}{N}\right)$$

where $0 \le n \le N$ and $w(n) = 0$ elsewhere and the window length is L = N +1. The coefficient values are

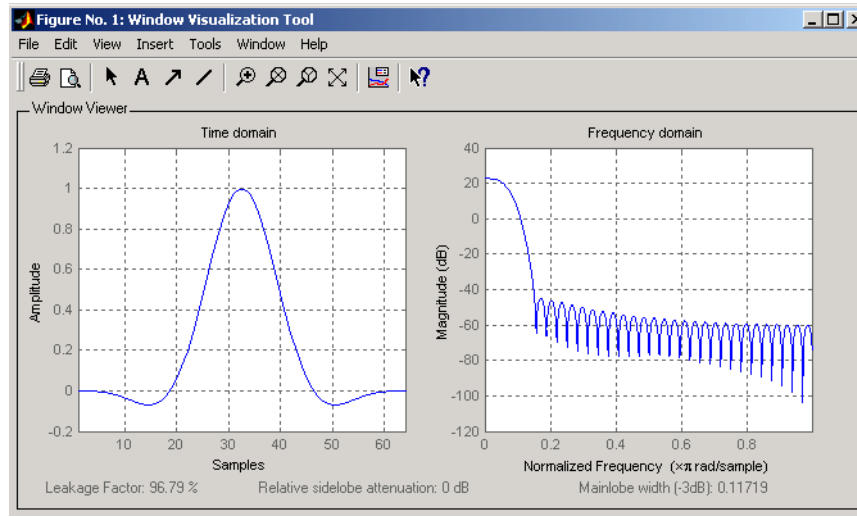
Coefficient	Value
a_0	0.21557895
a_1	0.41663158

Coefficient	Value
a_2	0.277263158
a_3	0.083578947
a_4	0.006947368

Examples

Create a 64-point, symmetric Flat Top window and view the window using WVTool:

```
w = flattopwin(64);
wvtool(w);
```



References

[1] D'Antona, Gabriele. and A. Ferrero, *Digital Signal Processing for Measurement Systems*, New York: Springer Media, Inc., 2006, pp. 70–72.

[2] Gade, Svend and H. Herlufsen, “Use of Weighting Functions in DFT/FFT Analysis (Part I),” Brüel & Kjær, *Windows to FFT Analysis (Part I) Technical Review, No. 3*, 1987, pp. 19-21.

See Also

`blackman` | `hamming` | `hann`

Purpose

Frequency response of analog filters

Syntax

```
h = freqs(b,a,w)
[h,w] = freqs(b,a,n)
freqs
```

Description

`freqs` returns the complex frequency response $H(j\omega)$ (Laplace transform) of an analog filter

$$H(s) = \frac{B(s)}{A(s)} = \frac{b(1)s^n + b(2)s^{n-1} + \cdots + b(n+1)}{a(1)s^m + a(2)s^{m-1} + \cdots + a(m+1)}$$

given the numerator and denominator coefficients in vectors `b` and `a`.

`h = freqs(b,a,w)` returns the complex frequency response of the analog filter specified by coefficient vectors `b` and `a`. `freqs` evaluates the frequency response along the imaginary axis in the complex plane at the angular frequencies in rad/sec specified in real vector `w`, where `w` is a vector containing more than one frequency.

`[h,w] = freqs(b,a,n)` uses `n` frequency points to compute the frequency response `h`, where `n` is a real, scalar value. The frequency vector `w` is auto-generated and has length `n`. If you omit `n` as an input, 200 frequency points are used. If you do not need the generated frequency vector returned, you can use the form `h = freqs(b,a,n)` to return only the frequency response `h`.

`freqs` with no output arguments plots the magnitude and phase response versus frequency in the current figure window.

`freqs` works only for real input systems and positive frequencies.

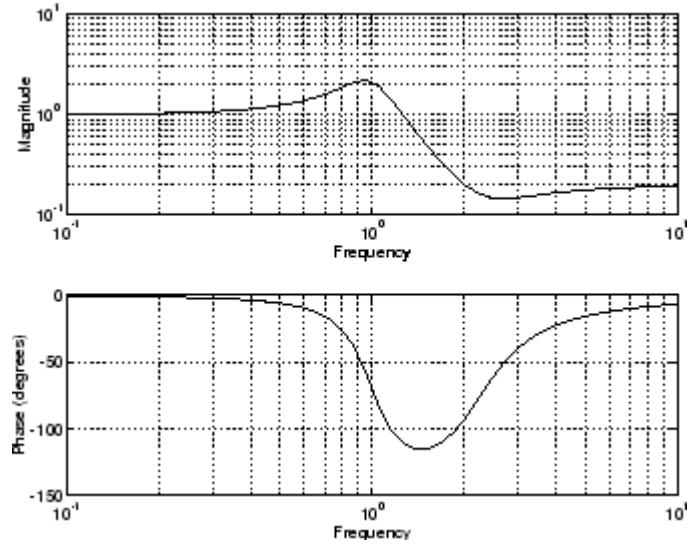
Examples

Find and graph the frequency response of the transfer function given by:

$$H(s) = \frac{0.2s^2 + 0.3s + 1}{s^2 + 0.4s + 1}$$

```
a = [1 0.4 1];
```

```
b = [0.2 0.3 1];
w = logspace(-1,1);
freqs(b,a,w)
```



You can also create the plot with

```
h = freqs(b,a,w);
mag = abs(h);
phase = angle(h);
subplot(2,1,1), loglog(w,mag)
subplot(2,1,2), semilogx(w,phase)
```

To convert to hertz, degrees, and decibels, use

```
f = w/(2*pi);
mag = 20*log10(mag);
phase = phase*180/pi;
```

Algorithms

`freqs` evaluates the polynomials at each frequency point, then divides the numerator response by the denominator response:

```
s = i*w;
h = polyval(b,s)./polyval(a,s);
```

See Also

abs | angle | freqz | invfreqs | logspace | polyval

freqsamp

Purpose

Real or complex frequency-sampled FIR filter from specification object

Syntax

```
hd = design(d,'freqsamp')
hd = design(...,'filterstructure',structure)
hd = design(...,'window',window)
```

Description

`hd = design(d,'freqsamp')` designs a frequency-sampled filter specified by the filter specifications object `d`.

`hd = design(...,'filterstructure',structure)` returns a filter with the filter structure you specify by the `structure` input argument. `structure` is `dffir` by default and can be any one of the following filter structures.

Structure String	Description of Resulting Filter Structure
`dffir`	Direct-form FIR filter
`dffirt`	Transposed direct-form FIR filter
`dfsymfir`	Symmetrical direct-form FIR filter
`dfasymfir`	Asymmetrical direct-form FIR filter
`fftfir`	Fast Fourier transform FIR filter

`hd = design(...,'window',window)` designs filters using the window specified by the string in `window`. Provide the input argument `window` as

- A string for the window type. For example, use `'bartlett'` , or `'hamming'`. See `window` for the full list of windows available in the *Signal Processing Toolbox User's Guide*.
- A function handle that references the `window` function. When the `window` function requires more than one input, use a cell array to hold the required arguments. The first example shows a cell array input argument.
- The window vector itself.

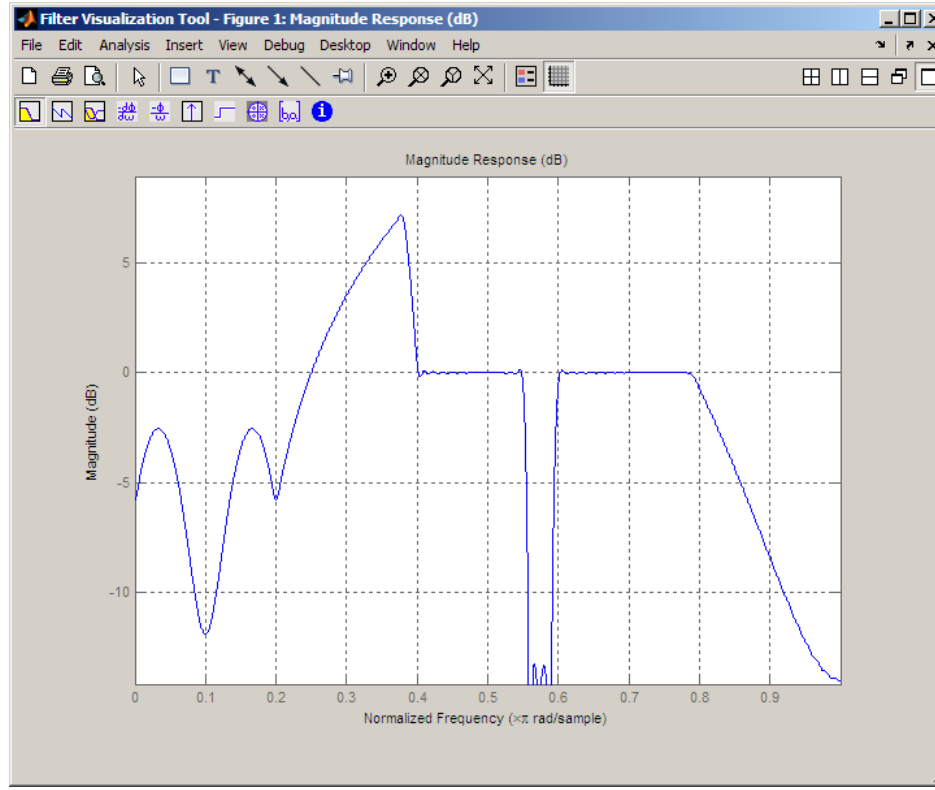
Examples

These examples design FIR filters that have arbitrary magnitude responses. In the first filter, the response has three distinct sections and the resulting filter is real.

The second example creates a complex filter.

```
b1 = 0:0.01:0.18;
b2 = [.2 .38 .4 .55 .562 .585 .6 .78];
b3 = [0.79:0.01:1];
a1 = .5+sin(2*pi*7.5*b1)/4;    % Sinusoidal response section.
a2 = [.5 2.3 1 1 -.2 -.2 1 1]; % Piecewise linear response section.
a3 = .2+18*(1-b3).^2;          % Quadratic response section.
f = [b1 b2 b3];
a = [a1 a2 a3];
n = 300;
d = fdesign.arbmag('n,f,a',n,f,a); % First specifications object.
hd = design(d,'freqsamp','window',{@kaiser,.5}); % Filter.
fvtool(hd)
```

The plot from FVTool shows the response for `hd`.

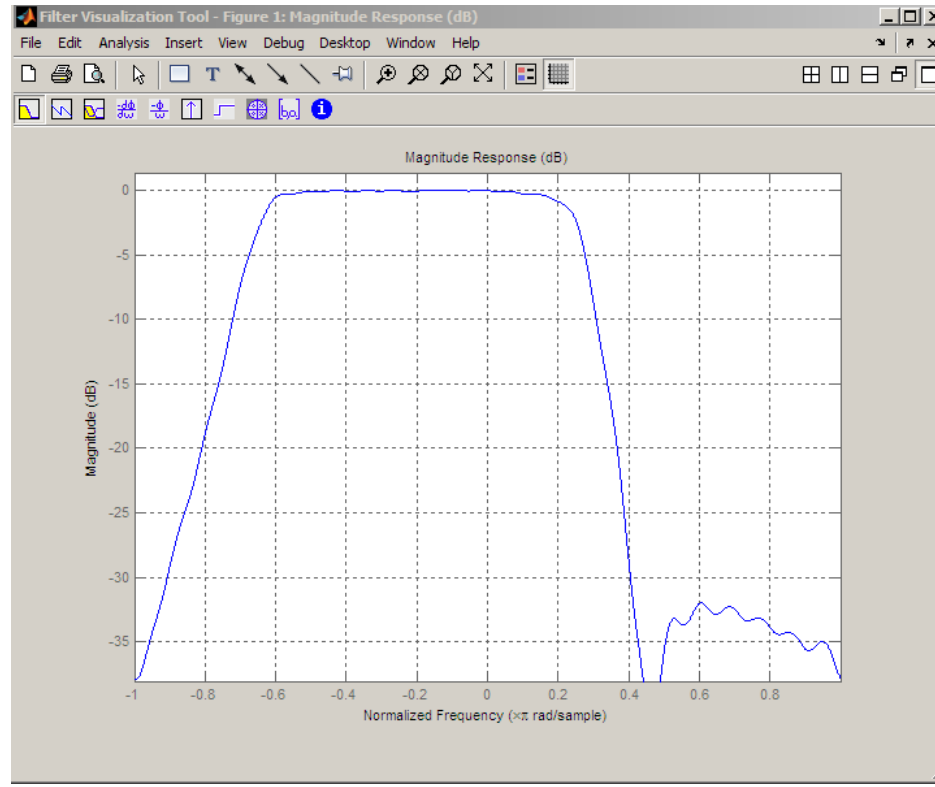


Now design the arbitrary-magnitude complex FIR filter. Recall that vector `f` contains frequency locations and vector `a` contains the desired filter response values at the locations specified in `f`.

```
f = [-1 -.93443 -.86885 -.80328 -.7377 -.67213 -.60656 -.54098 ...
-.47541,-.40984 -.34426 -.27869 -.21311 -.14754 -.081967 ...
-.016393 .04918 .11475,.18033 .2459 .31148 .37705 .44262 ...
 .5082 .57377 .63934 .70492 .77049,.83607 .90164 1];
a = [.0095848 .021972 .047249 .099869 .23119 .57569 .94032 ...
.98084 .99707,.99565 .9958 .99899 .99402 .99978 .99995 .99733 ...
.99731 .96979 .94936,.8196 .28502 .065469 .0044517 .018164 ...
.023305 .02397 .023141 .021341,.019364 .017379 .016061];
```

```
n = 48;
d = fdesign.arbmag('n,f,a',n,f,a); % Second spec. object.
hdc = design(d,'freqsamp','window','rectwin'); % Filter.
fvtool(hdc)
```

FVTool shows you the response for `hdc` from -1 to 1 in normalized frequency because the filter's transfer function is not symmetric around 0. Since the Fourier transform of the filter does not exhibit conjugate symmetry, `design(d,...)` returns a complex—valued filter for `hdc` .



See Also

`design` | `designmethods` | `fdesign.arbmag`

freqz

Purpose

Frequency response of digital filter

Syntax

```
[h,w] = freqz(b,a,n)
[h,w] = freqz(sos,n)
[h,w] = freqz(Hd,n)
[h,w] = freqz(...,n,'whole')
h = freqz(...,w)
[h,f] = freqz(...,fs)
h = freqz(...,f,fs)
[h,f] = freqz(...,n,'whole',fs)
freqz(...)
```

Description

`[h,w] = freqz(b,a,n)` returns the frequency response vector `h` and the corresponding angular frequency vector `w` for the digital filter whose transfer function is determined by the (real or complex) numerator and denominator polynomials represented in the vectors `b` and `a`, respectively. The vectors `h` and `w` are both of length `n`. `n` must be a positive integer greater than or equal to two. The angular 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 512 samples.

`[h,w] = freqz(sos,n)` returns the `n`-point complex frequency response corresponding to the second order sections matrix, `sos`. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. If the number of sections is less than 2, `freqz` considers the input to be the numerator vector, `b`. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`[h,w] = freqz(Hd,n)` returns the `n`-point complex frequency response for the `dfilt` filter object, `Hd`, or the array of `dfilt` filter objects. If `Hd` is an array of `dfilt` objects, each column of `h` is the complex-valued frequency response of the corresponding `dfilt` object.

`[h,w] = freqz(...,n,'whole')` uses `n` sample points around the entire unit circle to calculate the frequency response. The frequency vector `w` has length `n` and has values ranging from 0 to 2π radians per sample.

`h = freqz(...,w)` returns the frequency response vector `h` calculated at the frequencies (in radians per sample) supplied by the vector `w`. `w` must be a vector and have a minimum length of two.

`[h,f] = freqz(...,fs)` returns the frequency response vector `h` and the corresponding frequency vector `f` for the digital filter whose transfer function is determined by the (real or complex) numerator and denominator polynomials represented in the vectors `b` and `a`, respectively. The vectors `h` and f are both of length `n`. For this syntax, the frequency response is calculated using the sampling frequency specified by the scalar `fs` (in hertz). The frequency vector `f` is calculated in units of hertz (Hz). The frequency vector `f` has values ranging from `0` to `fs/2` Hz.

`h = freqz(...,f,fs)` returns the frequency response vector `h` calculated at the frequencies (in Hz) supplied in the vector `f`. The vector `f` must have at least two elements.

`[h,f] = freqz(...,n,'whole',fs)` uses `n` points around the entire unit circle to calculate the frequency response. The frequency vector `f` has length `n` and has values ranging from `0` to `fs` Hz.

`freqz(...)` plots the magnitude and unwrapped phase of the frequency response. The plot is displayed in the current figure window. If the input is the numerator and denominator coefficients, a second-order sections matrix, or a single `dfilt` object, the magnitude and phase response of the single filter is displayed. If the input is an array of `dfilt` objects, the magnitude and unwrapped phase responses of all filters in the array are displayed.

Note If the input to `freqz` is single precision, the frequency response is calculated using single-precision arithmetic. The output, `h`, is single precision.

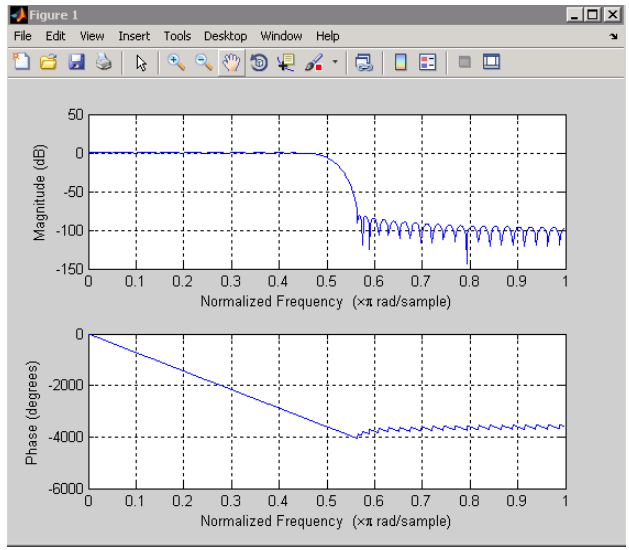
Tips

It is best to choose a power of 2 for the third input argument `n`, because `freqz` uses an FFT algorithm to calculate the frequency response. See the reference description of `fft` for more information.

Examples

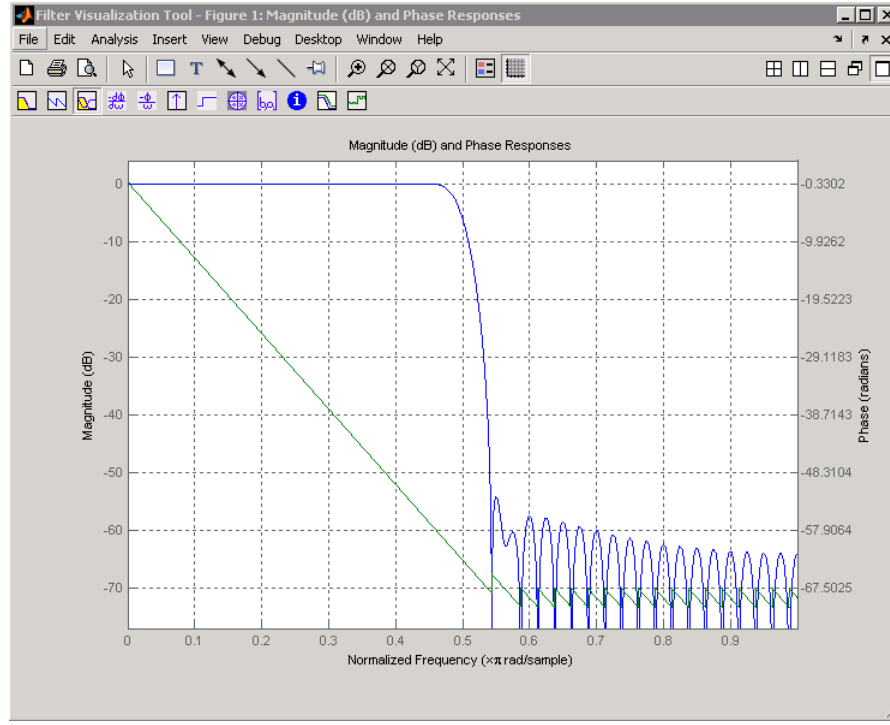
Plot the magnitude and phase response of an FIR filter:

```
b = fir1(80,0.5,kaiser(81,8));
freqz(b,1);
```



The same example using a `dfilt` object and displaying the result in the Filter Visualization Tool (`fvtool`) is

```
d=fdesign.lowpass('N,Fc',80,0.5);
Hd=design(d);
freqz(Hd);
```



Algorithms

The frequency response [1] of a digital filter can be interpreted as the transfer function evaluated at $z = e^{j\omega}$. You can always write a rational transfer function in the following form.

$$H(e^{j\omega}) = \frac{\sum_{k=0}^{M-1} b(k)e^{-j\omega k}}{\sum_{l=0}^{N-1} a(l)e^{-j\omega l}}.$$

`freqz` determines the transfer function from the (real or complex) numerator and denominator polynomials you specify, and returns the complex frequency response $H(e^{j\omega})$ of a digital filter. The frequency

response is evaluated at sample points determined by the syntax that you use.

`freqz` generally uses an FFT algorithm to compute the frequency response whenever you don't supply a vector of frequencies as an input argument. It computes the frequency response as the ratio of the transformed numerator and denominator coefficients, padded with zeros to the desired length.

When you do supply a vector of frequencies as an input argument, then `freqz` evaluates the polynomials at each frequency point using Horner's method of nested polynomial evaluation [1], dividing the numerator response by the denominator response.

References

[1] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing, Prentice-Hall*, 1989, pp. 203-205.

See Also

`abs` | `angle` | `fft` | `filter` | `freqs` | `impz` | `invfreqs` | `logspace`

Purpose

Open Filter Visualization Tool

Syntax

```
fvtool(b,a)
fvtool(sos)
fvtool(b1,a1,b2,a2,...bN,aN)
fvtool(sos1,sos2,...,sosN)
fvtool(Hd)
fvtool(Hd1,Hd2,...,HdN)
h = fvtool(...)
```

Description

`fvtool(b,a)` opens FVTool and displays the magnitude response of the digital filter defined with numerator, `b` and denominator, `a`. Using FVTool you can display the phase response, group delay, impulse response, step response, pole-zero plot, and coefficients of the filter. You can export the displayed response to a file with **File > Export**.

Note If the input to `fvtool` is single precision, the magnitude response is calculated using single-precision arithmetic.

`fvtool(sos)` opens FVTool and displays the magnitude response of the digital filter defined with the matrix of second order sections, `sos`. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. If the number of sections is less than 2, `fvtool` considers the input to be the numerator vector, `b`. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`fvtool(b1,a1,b2,a2,...bN,aN)` opens FVTool and displays the magnitude responses of multiple filters defined with numerators, `b1`...`b1N` and denominators, `a1`...`aN`.

`fvtool(sos1,sos2,...,sosN)` opens FVTool and displays the magnitude responses of multiple filters defined with second order section matrices, `sos1`, `sos2`, ...`sosN`.

`fvtool(Hd)` opens FVTool and displays the magnitude responses for the `dfilt` filter object, `Hd`, or the array of `dfilt` filter objects.

`fvtool(Hd1,Hd2,...,HdN)` opens FVTool and displays the magnitude responses of the filters in the `dfilt` objects `Hd1`, `Hd2`, ...`HdN`.

If you have the DSP System Toolbox product installed, you can also use `fvtool(H)` and `fvtool(`H_1`,`H_2`,...)` to analyze:

- Quantized filter objects (`dfilt` with arithmetic set to `'single'` or `'fixed'`)
- Multirate filter (`mfilt`) objects
- Any of the following filter System objects: `dsp.FIRFilter`, `dsp.IIRFilter`, `dsp.AllpoleFilter`, `dsp.BiquadFilter`, `dsp.FIRInterpolator`, `dsp.CICInterpolator`, `dsp.FIRDecimator`, `dsp.CICDecimator`, or `dsp.FIRRateConverter`
- Adaptive filter (`adaptfilt`) objects

When the input filter is a `dfilt` or `mfilt` object, FVTool performs fixed-point analysis if the arithmetic property of the filter objects is set to 'fixed'. However, for filter System objects, `fvtool(H,'Arithmetic',ARITH,...)` analyzes `H`, based on the arithmetic specified in the `ARITH` input.

`ARITH` can be one of `'double'`, `'single'`, or `'fixed'`. The `'Arithmetic'` input is only relevant for the analysis of filter System objects. The arithmetic setting `ARITH`, applies to all the filter System objects that you input to fvtool. When you specify `'double'` or `'single'`, the function performs double- or single-precision analysis. When you specify `'fixed'` , the arithmetic changes depending on the setting of the `CoefficientDataType` property and whether the System object is locked or unlocked.

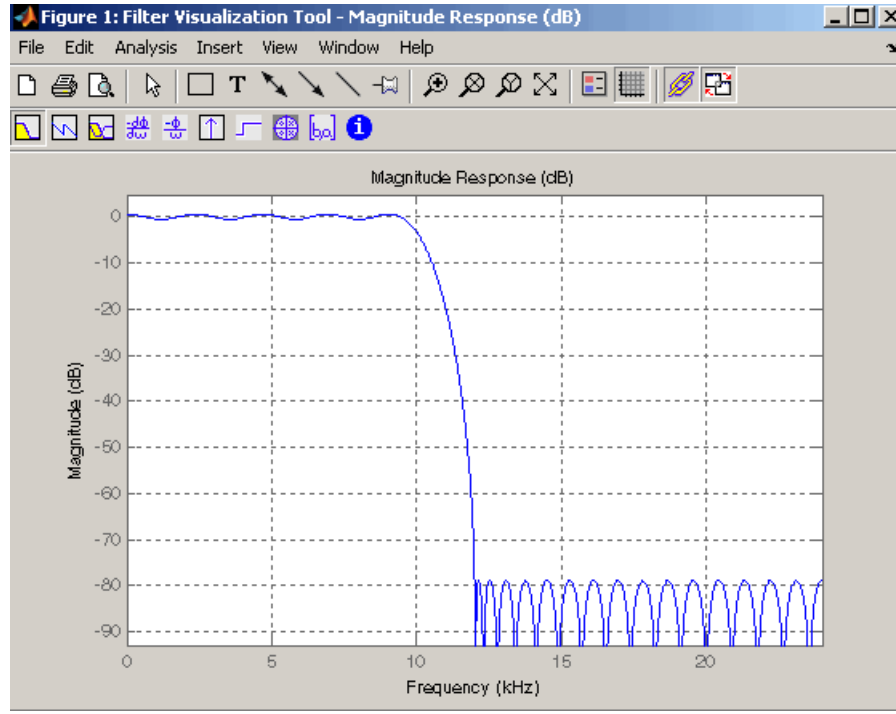
Details for Fixed-Point Arithmetic

System Object State	Coefficient Data Type	Rule
Unlocked	`'Same as input'`	The function assumes that the coefficient data type is signed, 16 bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Unlocked	`'Custom'`	The function performs fixed-point analysis based on the setting of the `CustomCoefficientsDataType` property.
Locked	`'Same as input'`	When the input data type is `'double'` or `'fixed'`, the function assumes that the coefficient data type is signed, 16-bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Locked	`'Custom'`	The function performs fixed-point analysis based on the setting of the `CustomCoefficientsDataType` property.

If you do not specify the arithmetic for non-CIC structures, and the System object is in an unlocked state, the function uses double-precision arithmetic. If the System object is locked, the function performs analysis based on the locked input data type. CIC structures only support fixed-point arithmetic.

Analysis methods `noisepsd` and `freqrespest` have behavior restrictions in `fvtool`. To see the rules, click the links to these methods.

`h = fvtool(...)` returns a figure handle `h`. You can use this handle to interact with FVTool from the command line. See "Controlling FVTool from the MATLAB Command Line" on page 1-584.



FVTool has two toolbars.

- An extended version of the MATLAB plot editing toolbar. The following table shows the toolbar icons specific to FVTool.

Icon	Description
	Restore default view. This view displays buffer regions around the data and shows only significant data. To see the response using standard MATLAB plotting, which shows all data values, use **View > Full View**.
	Toggle legend
	Toggle grid
	Link to FDATool (appears only if FVTool was started from FDATool)
	Toggle Add mode/Replace mode (appears only if FVTool was launched from FDATool)

- Analysis toolbar with the following icons

	Magnitude response of the current filter. See `freqz` and `zerophase` for more information. To see the zero-phase response, right-click the *y*-axis label of the Magnitude plot and select **Zero-phase** from the context menu.
	Phase response of the current filter. See `phasez` for more information.
	Superimposes the magnitude response and the phase response of the current filter. See `freqz` for more information.
	Shows the group delay of the current filter. Group delay is the average delay of the filter as a function of frequency. See `grpdelay` for more information.

	Shows the phase delay of the current filter. Phase delay is the time delay the filter imposes on each component of the input signal. See `phasedelay` for more information.
	Impulse response of the current filter. The impulse response is the response of the filter to a impulse input. See `impz` for more information.
	Step response of the current filter. The step response is the response of the filter to a step input. See `stepz` for more information.
	Pole-zero plot, which shows the pole and zero locations of the current filter on the *z*-plane. See `zplane` for more information.
	Filter coefficients of the current filter, which depend on the filter structure (e.g., direct-form, lattice, etc.) in a text box. For SOS filters, each section is displayed as a separate filter.
	Detailed filter information.

Linking to FDATool

In `fdatool`, selecting **View > Filter Visualization Tool** or the **Full View Analysis** toolbar button  when an analysis is displayed starts FVTool for the current filter. You can synchronize FDATool and FVTool with the **FDAToolLink** toolbar button  . Any changes made to the filter in FDATool are immediately reflected in FVTool.

Two FDATool link modes are provided via the **Set Link Mode** toolbar button:

- Replace  — removes the filter currently displayed in FVTool and inserts the new filter.
- Add  — retains the filter currently displayed in FVTool and adds the new filter to the display.

Modifying the Axes

You can change the *x*- or *y*-axis units by right-clicking the mouse on the axis label or by right-clicking on the plot and selecting **Analysis Parameters**. Available options for the axes units are as follows.

Plot	X-Axis Units	Y-Axis Units
Magnitude	Normalized Frequency Linear Frequency	Magnitude Magnitude(dB) Magnitude squared Zero-Phase
Phase	Normalized Frequency Linear Frequency	Phase Continuous Phase Degrees Radians
Magnitude and Phase	Normalized Frequency Linear Frequency	(*y*-axis on left side) Magnitude Magnitude(dB) Magnitude squared Zero-Phase (*y*-axis on right side) Phase Continuous Phase Degrees Radians
Group Delay	Normalized Frequency Linear Frequency	Samples Time

Plot	X-Axis Units	Y-Axis Units
Phase Delay	Normalized Frequency Linear Frequency	Degrees Radians
Impulse Response	Samples Time	Amplitude
Step Response	Samples Time	Amplitude
Pole-Zero	Real Part	Imaginary Part

Modifying the Plot

You can use any of the plot editing toolbar buttons to change the properties of your plot.

Analysis Parameters are parameters that apply to the displayed analyses. To display them, right-click in the plot area and select **Analysis Parameters** from the menu. (Note that you can access the menu only if the **Edit Plot** button is inactive.) The following analysis parameters are displayed. (If more than one response is displayed, parameters applicable to each plot are displayed.) Not all of these analysis fields are displayed for all types of plots:

- **Normalized Frequency** — if checked, frequency is normalized between 0 and 1, or if not checked, frequency is in Hz
- **Frequency Scale** — *y*-axis scale (`Linear` or `Log`)
- **Frequency Range** — range of the frequency axis or `Specify freq. vector`
- **Number of Points** — number of samples used to compute the response
- **Frequency Vector** — vector to use for plotting, if `Specify freq. vector` is selected in **Frequency Range**.

- **Magnitude Display** — *y*-axis units (`Magnitude`, `Magnitude (dB)`, `Magnitude squared`, or `Zero-Phase`)
- **Phase Units** — *y*-axis units (`Degrees` or `Radians`)
- **Phase Display** — type of phase plot (`Phase` or `Continuous Phase`)
- **Group Delay Units** — *y*-axis units (`Samples` or `Time`)
- **Specify Length** — length type of impulse or step response (`Default` or `Specified`)
- **Length** — number of points to use for the impulse or step response

In addition to the above analysis parameters, you can change the plot type for Impulse and Step Response plots by right-clicking and selecting **Line with Marker**, **Stem** or **Line** from the context menu. You can change the *x*-axis units by right-clicking the *x*-axis label and selecting `Samples` or `Time`.

To save the displayed parameters as the default values to use when FDATool or FVTool is opened, click **Save as default**.

To restore the default values, click **Restore original defaults**.

Data tips display information about a particular point in the plot. See "Data Cursor — Displaying Data Values Interactively" in the MATLAB documentation for information on data tips.

If you have the DSP System Toolbox software, FVTool displays a specification mask along with your designed filter on a magnitude plot.

Note To use **View > Passband zoom**, your filter must have been designed using `fdesign` or FDATool. Passband zoom is not provided for cascaded integrator-comb (CIC) filters because CICs do not have conventional passbands.

Overlaying a Response

You can overlay a second response on the plot by selecting **Analysis > Overlay Analysis** and selecting an available response.

A second *y*-axis is added to the right side of the response plot. The Analysis Parameters dialog box shows parameters for the *x*-axis and both *y*-axes. See "Example 2" on page 1-588 for a sample Analysis Parameters dialog box.

Controlling FVTool from the MATLAB Command Line

After you obtain the handle for FVTool, you can control some aspects of FVTool from the command line. In addition to the standard Handle Graphics® properties (see Handle Graphics in the MATLAB documentation), FVTool has the following properties:

- `'Filters'` — returns a cell array of the filters in FVTool.
- `'Analysis'` — displays the specified type of analysis plot. The following table lists the analyses and corresponding analysis strings. Note that the only analyses that use filter internals are magnitude response estimate and round-off noise power, which are available only with the DSP System Toolbox product.

Analysis Type	Analysis String
Magnitude plot	`'magnitude'`
Phase plot	`'phase'`
Magnitude and phase plot	`` `freq' ``
Group delay plot	`'grpdelay'`
Phase delay plot	`` `phasedelay' ``
Impulse response plot	`'impulse'`
Step response plot	`'step'`
Pole-zero plot	`'polezero'`
Filter coefficients	`'coefficients'`
Filter information	`'info'`

Analysis Type	Analysis String
Magnitude response estimate (available only with the DSP System Toolbox product, see `freqrespest` for more information)	`'magestimate'`
Round-off noise power (available only with the DSP System Toolbox product, see `noisepsd` for more information)	`'noisepower'`

- `'Grid'` — controls whether the grid is `'on'` or `'off'`
- `'Legend'` — controls whether the legend is `'on'` or `'off'`
- `'Fs'` — controls the sampling frequency of filters in FVTool. The sampling frequency vector must be of the same length as the number of filters or a scalar value. If it is a vector, each value is applied to its corresponding filter. If it is a scalar, the same value is applied to all filters.
- `SosViewSettings` — (This option is available only if you have the DSP System Toolbox product.) For second-order sections filters, this controls how the filter is displayed. The `SOSViewSettings` property contains an object so you must use this syntax to set it: `set(h.SOSViewSettings,'View',`*`viewtype`*`)`, where *`viewtype`* is one of the following:
  - `'Complete'` — Displays the complete response of the overall filter
  - `'Individual'` — Displays the response of each section separately
  - 'Cumulative' — Displays the response for each section accumulated with each prior section. If your filter has three sections, the first plot shows section one, the second plot shows the accumulation of sections one and two, and the third plot show the accumulation of all three sections.

You can also define whether to use `SecondaryScaling`, which determines where the sections should be split. The secondary scaling points are the scaling locations between the recursive and the nonrecursive parts of the section. The default value is `false`, which does not use secondary scaling. To turn on secondary scaling, use this syntax: `set(h.SOSViewSettings,'View','Cumulative',true)`

- `'UserDefined'` — Allows you to define which sections to display and the order in which to display them. Enter a cell array where each section is represented by its index. If you enter one index, only that section is plotted. If you enter a range of indices, the combined response of that range of sections is plotted. For example, if your filter has four sections, entering `{1:4}` plots the combined response for all four sections, and entering `{1,2,3,4}` plots the response for each section individually.

Note You can change other properties of FVTool from the command line using the `set` function. Use `get(h)` to view property tags and current property settings.

You can use the following methods with the FVTool handle.

`addfilter(h,filtobj)` adds a new filter to FVTool. The new filter, `filtobj`, must be a `dfilt` filter object. You can specify the sampling frequency of the new filter with `addfilter(h,filtobj,'Fs',10)`.

`setfilter(h,filtobj)` replaces the filter in FVTool with the filter specified in `filtobj`. You can set the sampling frequency as described above.

`deletefilter(h, index)` deletes the filter at the FVTool cell array `index` location.

`legend(h,str1,str2,...)` creates a legend in FVTool by associating `str1` with filter 1, `str2` with filter 2, etc. See `legend` in the MATLAB documentation for information.

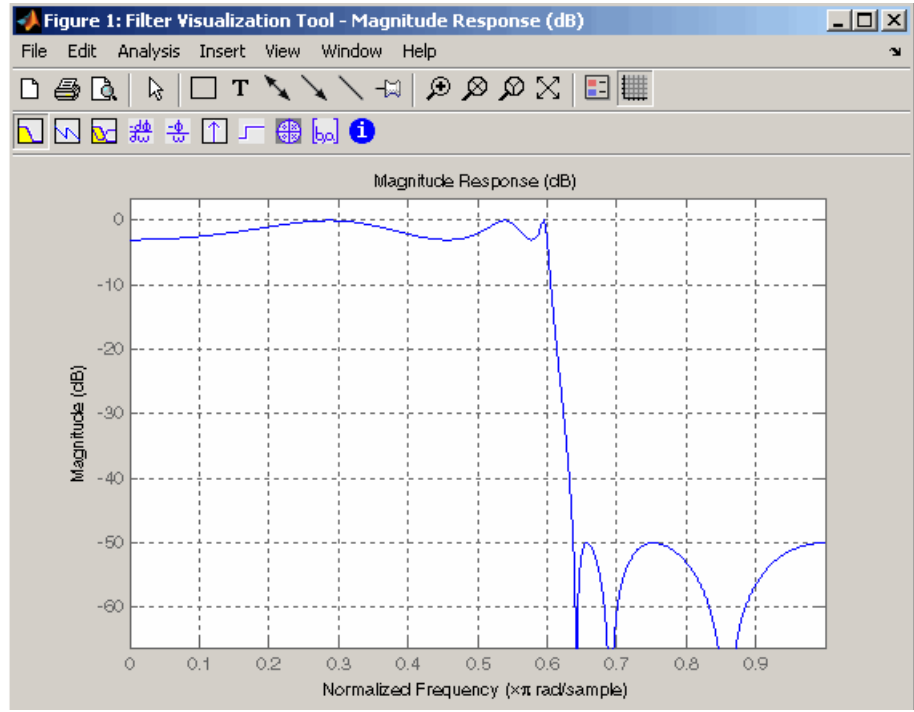
For more information on using FVTool from the command line, see the example `fvtooldemo`.

Examples

Example 1

Display the magnitude response of an elliptic filter, starting FVTool from the command line:

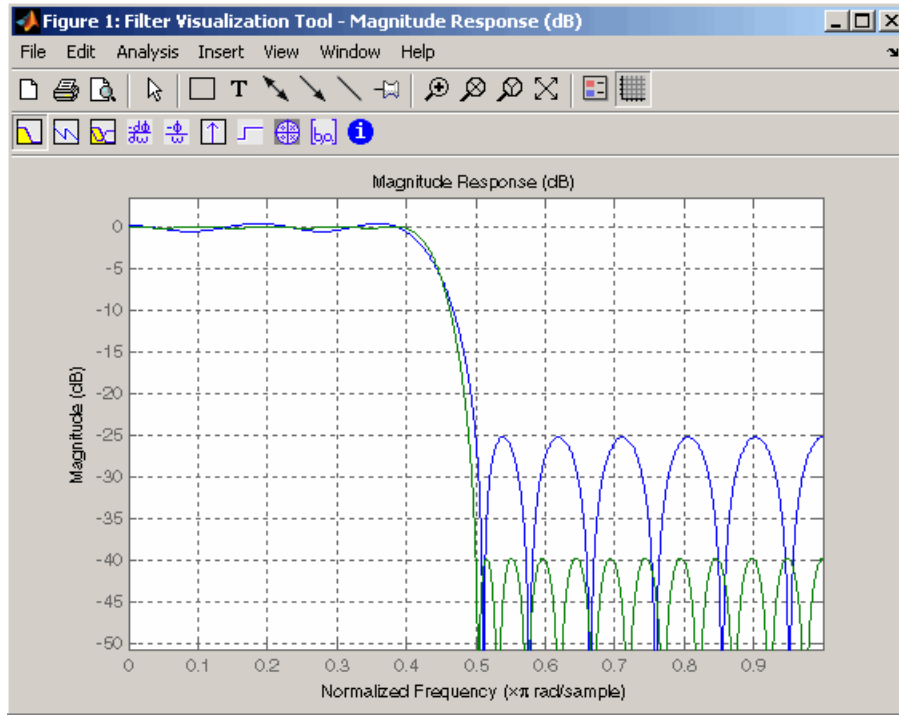
```
[b,a]=ellip(6,3,50,300/500);
fvtool(b,a);
```

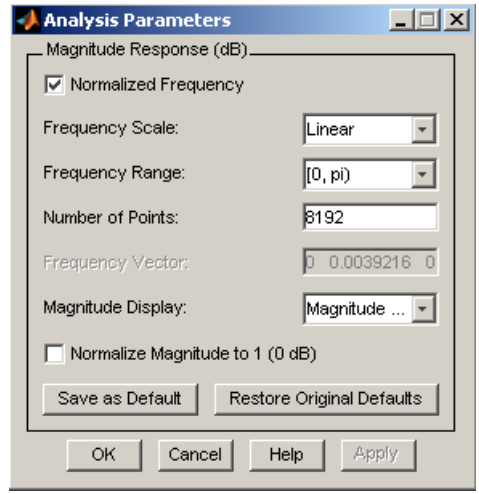


Example 2

Display and analyze multiple FIR filters, starting FVTool from the command line. Then, display the associated analysis parameters for the magnitude:

```
b1 = firpm(20,[0 0.4 0.5 1],[1 1 0 0]);
b2 = firpm(40,[0 0.4 0.5 1],[1 1 0 0]);
fvtool(b1,1,b2,1);
```





Example 3

Create a lowpass, equiripple filter of order 20 in FDATool and display it in FVTool.

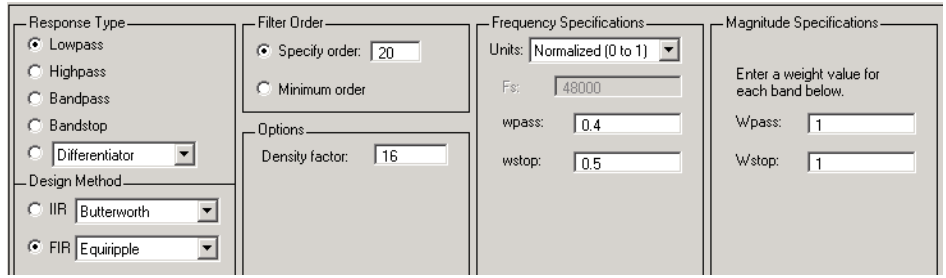
```
fdatool          % Start FDATool
```

Set these parameters in `fdatool`:

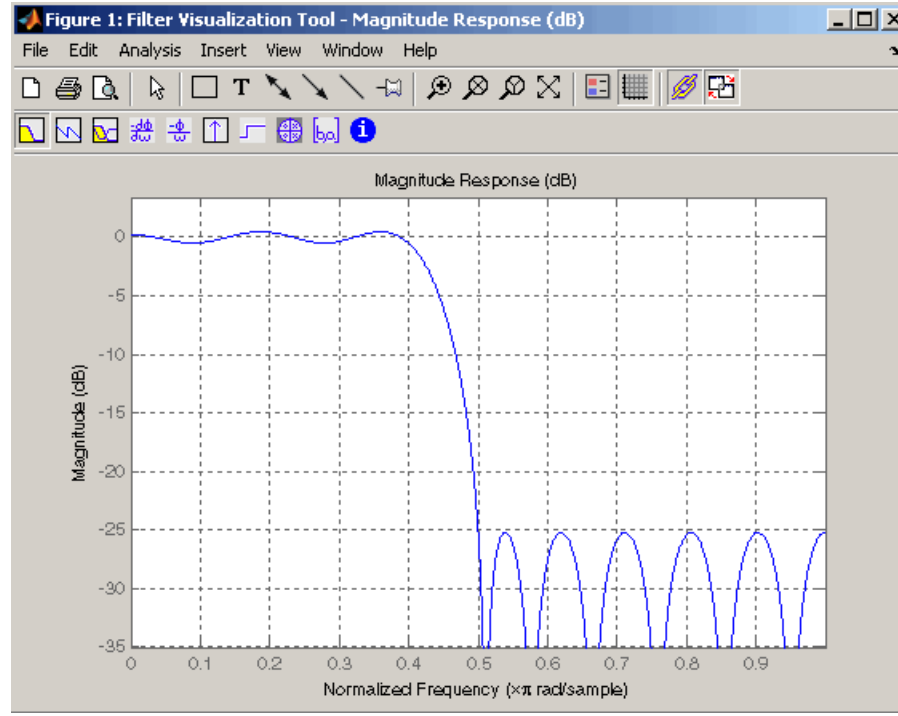
Parameter	Setting
Response Type	Lowpass
Design Method	FIR Equiripple
Filter Order	Specify order: 20
Density factor	16
Frequency specifications -- units	Normalized (0 to 1)
Wpass	0.4

Parameter	Setting
`Wstop`	`0.5`
`Magnitude specifications -- Wpass and Wstop`	`1`

and then click the **Design Filter** button.



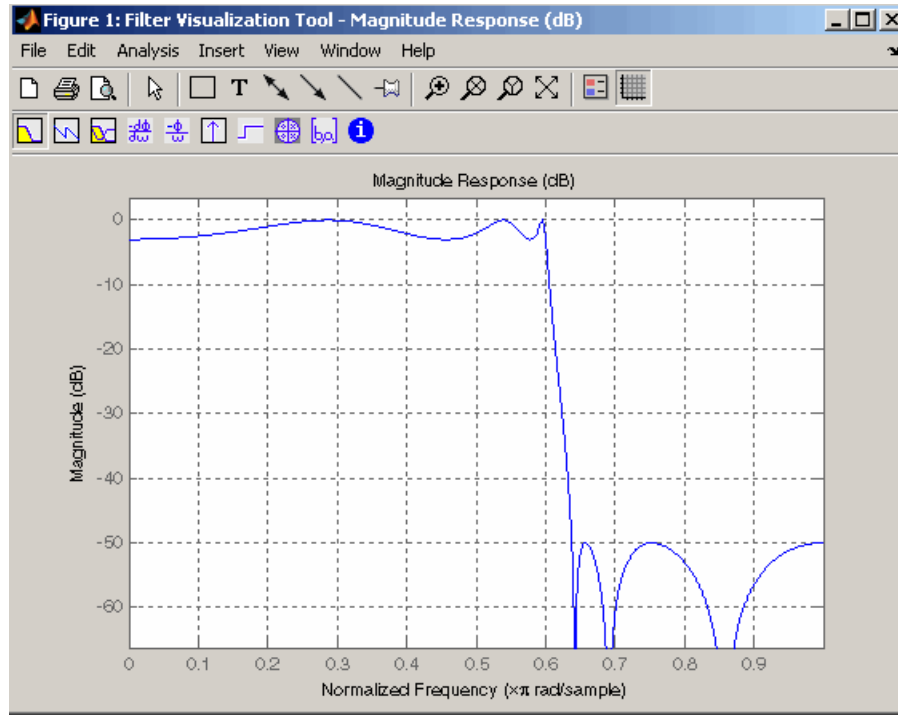
Click the **Full View Analysis** button to start FVTool.



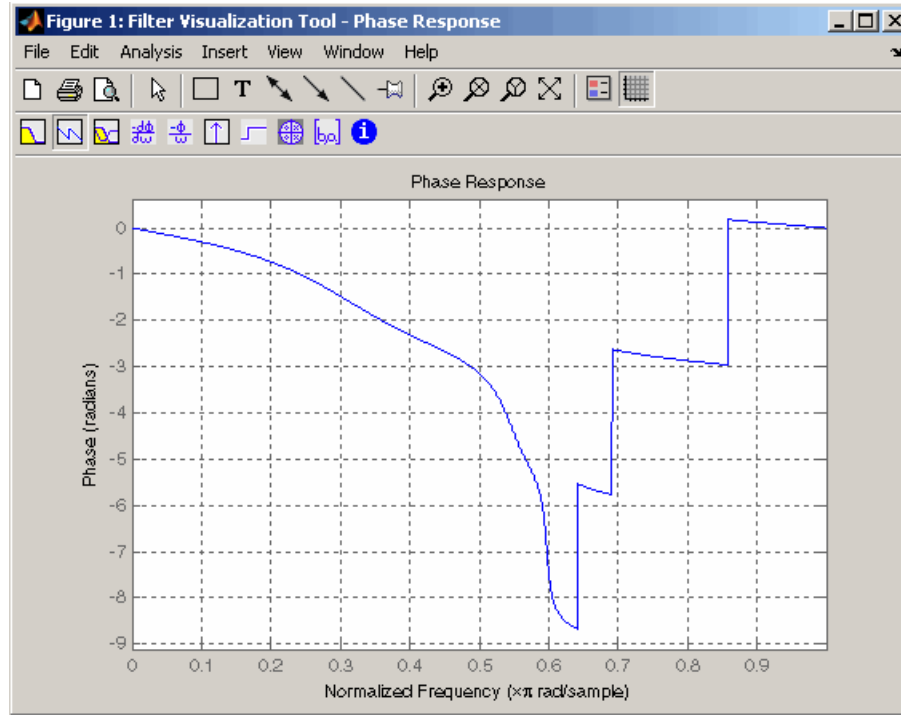
Example 4

Create an elliptic filter and use some of FVTool's figure handle commands:

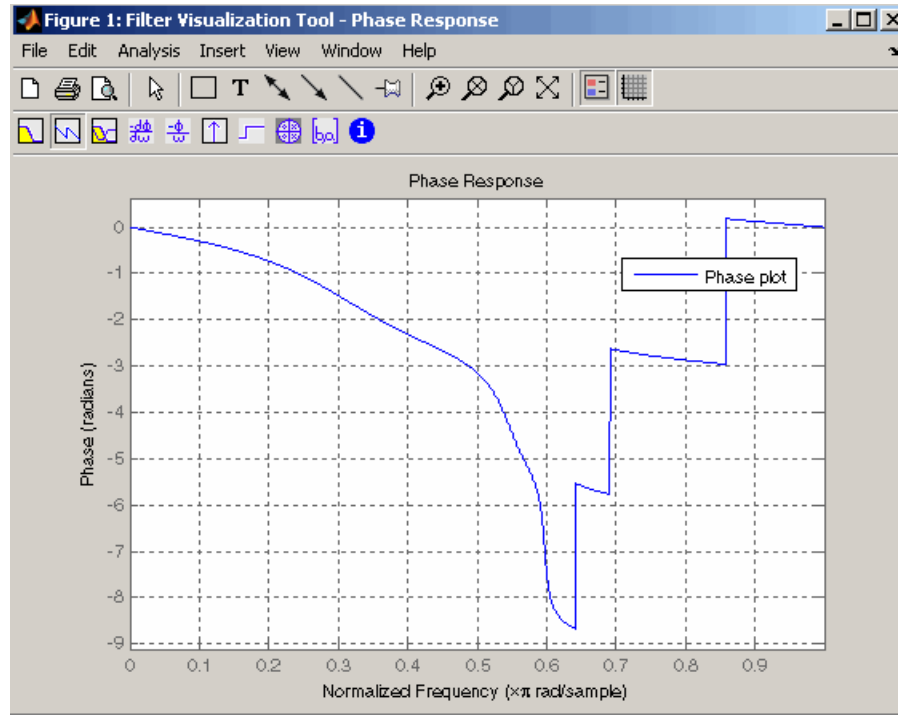
```
[b,a]=ellip(6,3,50,300/500);
h = fvtool(b,a);    % Create handle, h and start FVTool
                    % with magnitude plot
```




```
set(h,'Analysis','phase')   % Change display to phase plot
```




```
set(h,'Legend','on')          % Turn legend on
legend(h,'Phase plot')        % Add legend text
```




```
get(h)                % View all properties
                      % FVTool-specific properties are
                      %  at the end of this list.

                AlphaMap: [1x64 double]
          CloseRequestFcn: 'closereq'
                    Color: [0.8314 0.8157 0.7843]
                ColorMap: [64x3 double]
            CurrentAxes: 208.0084
         CurrentCharacter: ''
           CurrentObject: []
            CurrentPoint: [0 0]
            DockControls: 'on'
            DoubleBuffer: 'on'
```

```
                FileName: ''
              FixedColors: [11x3 double]
            IntegerHandle: 'on'
           InvertHardcopy: 'on'
              KeyPressFcn: ''
                  MenuBar: 'none'
              MinColormap: 64
                     Name: 'Filter Visualization Tool - Phase Response'
                 NextPlot: 'new'
              NumberTitle: 'on'
               PaperUnits: 'inches'
          PaperOrientation: 'portrait'
             PaperPosition: [0.2500 2.5000 8 6]
        PaperPositionMode: 'manual'
                PaperSize: [8.5000 11]
                PaperType: 'usletter'
                  Pointer: 'arrow'
        PointerShapeCData: [16x16 double]
     PointerShapeHotSpot: [1 1]
                Position: [360 292 560 345]
                Renderer: 'painters'
            RendererMode: 'auto'
                  Resize: 'on'
               ResizeFcn: ''
           SelectionType: 'normal'
                 Toolbar: 'auto'
                   Units: 'pixels'
   WindowButtonDownFcn: ''
 WindowButtonMotionFcn: ''
     WindowButtonUpFcn: ''
           WindowStyle: 'normal'
          BeingDeleted: 'off'
         ButtonDownFcn: ''
              Children: [15x1 double]
              Clipping: 'on'
             CreateFcn: ''
             DeleteFcn: ''
```

```
              BusyAction: 'queue'
        HandleVisibility: 'on'
                 HitTest: 'on'
            Interruptible: 'on'
                  Parent: 0
                Selected: 'off'
      SelectionHighlight: 'on'
                     Tag: 'filtervisualizationtool'
           UIContextMenu: []
                UserData: []
                 Visible: 'on'
         AnalysisToolbar: 'on'
           FigureToolbar: 'on'
                 Filters: {[1x1 dfilt.df2t]}
                    Grid: 'on'
                  Legend: 'on'
              DesignMask: 'off'
                      Fs: 1
         SOSViewSettings: [1x1 dspopts.sosview]
                Analysis: 'phase'
      OverlayedAnalysis: ''
          ShowReference: 'on'
           PolyphaseView: 'off'
    NormalizedFrequency: 'on'
         FrequencyScale: 'Linear'
         FrequencyRange: '[0, pi)'
         NumberofPoints: 8192
        FrequencyVector: [1x256 double]
             PhaseUnits: 'Radians'
           PhaseDisplay: 'Phase'
```

See Also

fdatool | sptool

Purpose

Fast Walsh–Hadamard transform

Syntax

```
y = fwht(x)
y = fwht(x,n)
y = fwht(x,n,ordering)
```

Description

`y = fwht(x)` returns the coefficients of the discrete Walsh–Hadamard transform of the input `x`. If `x` is a matrix, the FWHT is calculated on each column of `x`. The FWHT operates only on signals with length equal to a power of 2. If the length of `x` is less than a power of 2, its length is padded with zeros to the next greater power of two before processing.

`y = fwht(x,n)` returns the `n`-point discrete Walsh–Hadamard transform, where `n` must be a power of 2. `x` and `n` must be the same length. If `x` is longer than `n`, `x` is truncated; if `x` is shorter than `n`, `x` is padded with zeros.

`y = fwht(x,n,ordering)` specifies the ordering to use for the returned Walsh–Hadamard transform coefficients. To specify ordering, you must enter a value for the length `n` or, to use the default behavior, specify an empty vector `[]` for `n`. Valid values for ordering are the following strings:

Ordering	Description
`'sequency'`	Coefficients in order of increasing sequency value, where each row has an additional zero crossing. This is the default ordering.
`'hadamard'`	Coefficients in normal Hadamard order.
`'dyadic'`	Coefficients in Gray code order, where a single bit change occurs from one coefficient to the next.

For more information on the Walsh functions and ordering, see "Walsh–Hadamard Transform".

Examples

This example shows a simple input signal and the resulting transformed signal.

```
x = [19 -1 11 -9 -7 13 -15 5];
```

```
y = fwht(x);
```

y contains nonzero values at these locations: 0, 1, 3, and 6. By forming the Walsh functions with the sequency values of 0, 1, 3, and 6, we can recreate x, as follows.

```
w0 = [1 1 1 1 1 1 1 1];
w1 = [1 1 1 1 -1 -1 -1 -1];
w3 = [1 1 -1 -1 1 1 -1 -1];
w6 = [1 -1 1 -1 -1 1 -1 1];
w = 2*w0 + 3*w1 + 4*w3 + 10*w6;
y1=fwht(w);
x1 = ifwht(y);
```

Algorithms

The fast Walsh-Hadamard tranform algorithm is similar to the Cooley-Tukey algorithm used for the FFT. Both use a butterfly structure to determine the transform coefficients. See the references for details.

References

[1] Beauchamp, K.G., *Applications of Walsh and Related Functions*, Academic Press, 1984.

[2] Beer, T., *Walsh Transforms*, American Journal of Physics, Volume 49, Issue 5, May 1981.

See Also

`ifwht` | `dct` | `idct` | `fft` | `ifft`

Purpose

Gaussian-modulated sinusoidal pulse

Syntax

```
yi = gauspuls(t,fc,bw)
yi = gauspuls(t,fc,bw,bwr)
[yi,yq] = gauspuls(...)
[yi,yq,ye] = gauspuls(...)
tc = gauspuls('cutoff',fc,bw,bwr,tpe)
```

Description

`gauspuls` generates Gaussian-modulated sinusoidal pulses.

`yi = gauspuls(t,fc,bw)` returns a unity-amplitude Gaussian RF pulse at the times indicated in array `t`, with a center frequency `fc` in hertz and a fractional bandwidth `bw`, which must be greater than 0. The default value for `fc` is 1000 Hz and for `bw` is 0.5.

`yi = gauspuls(t,fc,bw,bwr)` returns a unity-amplitude Gaussian RF pulse with a fractional bandwidth of `bw` as measured at a level of `bwr` dB with respect to the normalized signal peak. The fractional bandwidth reference level `bwr` must be less than 0, because it indicates a reference level less than the peak (unity) envelope amplitude. The default value for `bwr` is -6 dB. Note that the fractional bandwidth is specified in terms of power ratios. This corresponds to the -3 dB point expressed in magnitude ratios.

`[yi,yq] = gauspuls(...)` returns both the in-phase and quadrature pulses.

`[yi,yq,ye] = gauspuls(...)` returns the RF signal envelope.

`tc = gauspuls('cutoff',fc,bw,bwr,tpe)` returns the cutoff time `tc` (greater than or equal to 0) at which the trailing pulse envelope falls below `tpe` dB with respect to the peak envelope amplitude. The trailing pulse envelope level `tpe` must be less than 0, because it indicates a reference level less than the peak (unity) envelope amplitude. The default value for `tpe` is -60 dB.

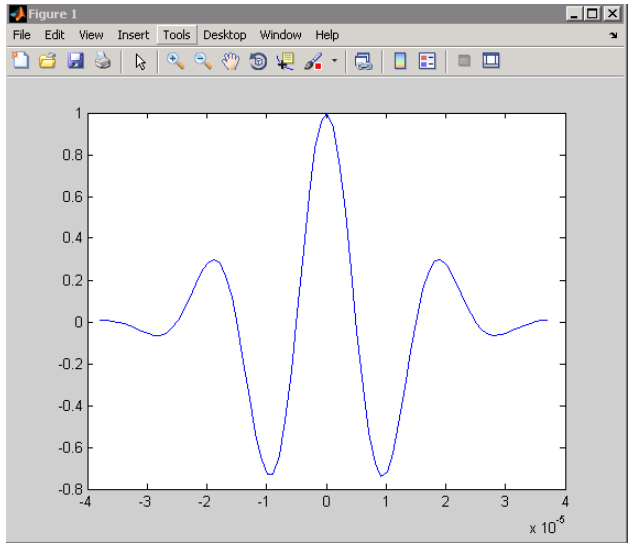
Tips

Default values are substituted for empty or omitted trailing input arguments.

Examples

Plot a 50 kHz Gaussian RF pulse with 60% bandwidth, sampled at a rate of 1 MHz. Truncate the pulse where the envelope falls 40 dB below the peak:

```
tc = gauspuls('cutoff',50e3,0.6,[],-40);
t = -tc : 1e-6 : tc;
yi = gauspuls(t,50e3,0.6);
plot(t,yi)
```



See Also

`chirp` | `cos` | `diric` | `pulstran` | `rectpuls` | `sawtooth` | `sin` | `sinc` | `square` | `tripuls`

Purpose

Gaussian FIR pulse-shaping filter

Syntax

```
h = gaussfir(bt)
h = gaussfir(bt,n)
h = gaussfir(bt,n,o)
```

Description

This filter is used primarily in Gaussian minimum shift keying (GMSK) communications applications.

`h = gaussfir(bt)` designs a low pass FIR Gaussian pulse-shaping filter and returns the filter coefficients in the `h` vector. `bt` is the 3-dB bandwidth-symbol time product where `b` is the one-sided bandwidth in hertz and `t` is in seconds. Smaller `bt` products produce larger pulse widths. The number of symbol periods (`n`) defaults to 3 and the oversampling factor (`o`) defaults to 2.

The length of the impulse response of the filter is given by `2*o*n+1`. The coefficients `h` are normalized so that the nominal passband gain is always equal to 1.

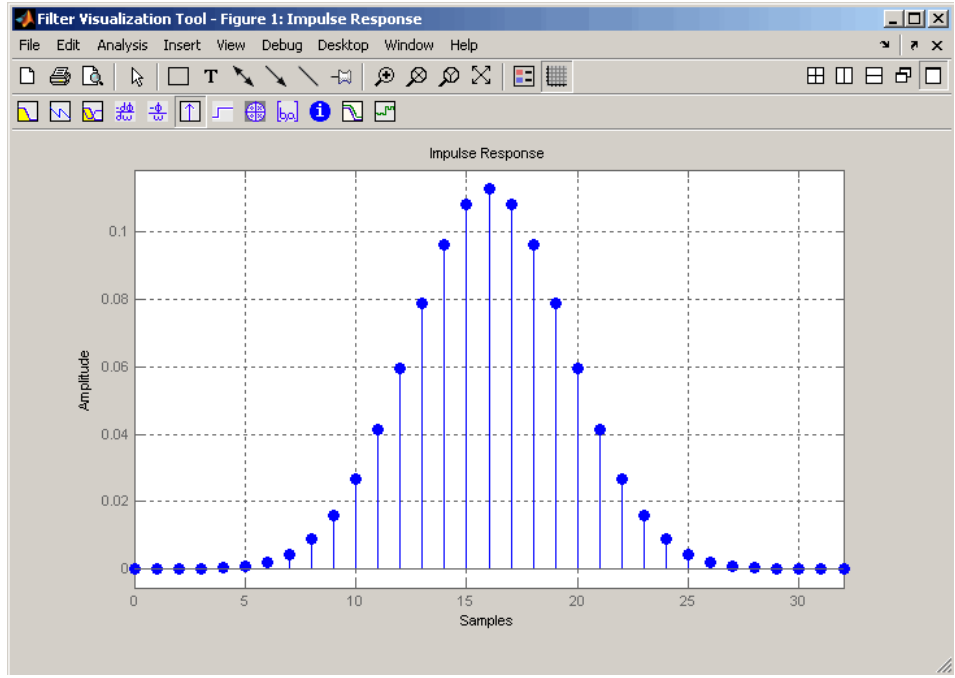
`h = gaussfir(bt,n)` uses `n` number of symbol periods between the start of the filter impulse response and its peak.

`h = gaussfir(bt,n,o)` uses an oversampling factor of `o`, which is the number of samples per symbol.

Examples

Design a Gaussian filter to be used in a Global System for Mobile (GSM) communications GMSK scheme.

```
bt = .3;          % 3-dB bandwidth-symbol time
o = 8;            % Oversampling factor
n = 2;            % 2 symbol periods to the filters peak
h = gaussfir(bt,n,o);
hfvt = fvtool(h,'impulse');
```



References

[1] Rappaport T.S., Wireless Communications Principles and Practice, 2nd Edition, Prentice Hall, 2001.

[2] Krishnapura N., Pavan S., Mathiazhagan C., Ramamurthi B., "A Baseband Pulse Shaping Filter for Gaussian Minimum Shift Keying," *Proceedings of the 1998 IEEE International Symposium on Circuits and Systems*, 1998.

See Also

`firrcos` | `rcosfir`

Purpose

Gaussian window

Syntax

```
w = gausswin(N)
w=gausswin(N,Alpha)
```

Description

`w = gausswin(N)` returns an `N`-point Gaussian window in the column vector `w`. `N` is a positive integer. The coefficients of a Gaussian window are computed from the following equation.

$$w(n) = e^{-\frac{1}{2}\left(\alpha\frac{n}{N/2}\right)^2}$$

where $-\frac{(N-1)}{2} \le n \le \frac{(N-1)}{2}$, and α is inversely proportional to the standard deviation of a Gaussian random variable. The exact correspondence with the standard deviation, σ, of a Gaussian probability density function is

$$\sigma = \frac{N}{2\alpha}$$

The value of α defaults to 2.5.

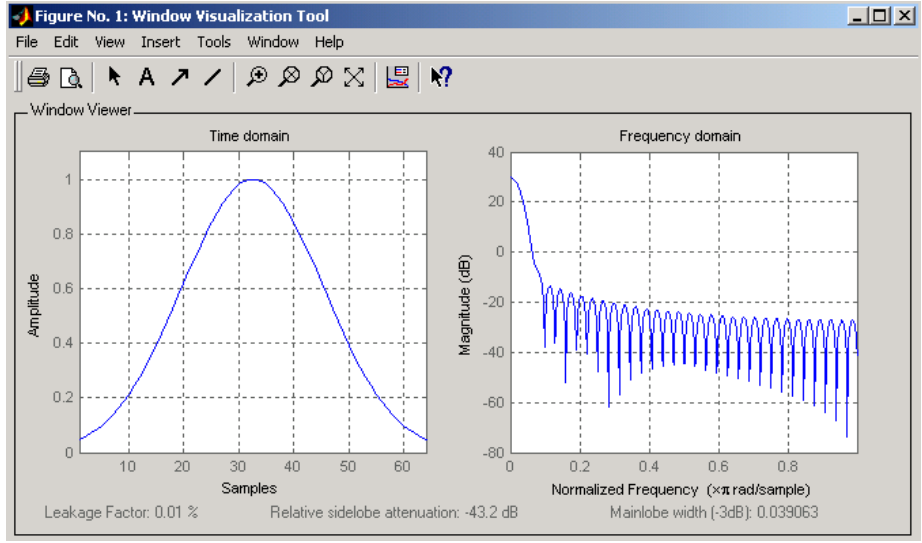
`w=gausswin(N,Alpha)` returns an `N`-point Gaussian window where `Alpha` is proportional to reciprocal of the standard deviation. The width of the window is inversely related to the value of α; a larger value of α produces a more narrow window.

Note If the window appears to be clipped, increase the number of points (`N`).

Examples

Create a 64-point Gaussian window and display the result in WVTool:

```
L=64;
wvtool(gausswin(L))
```



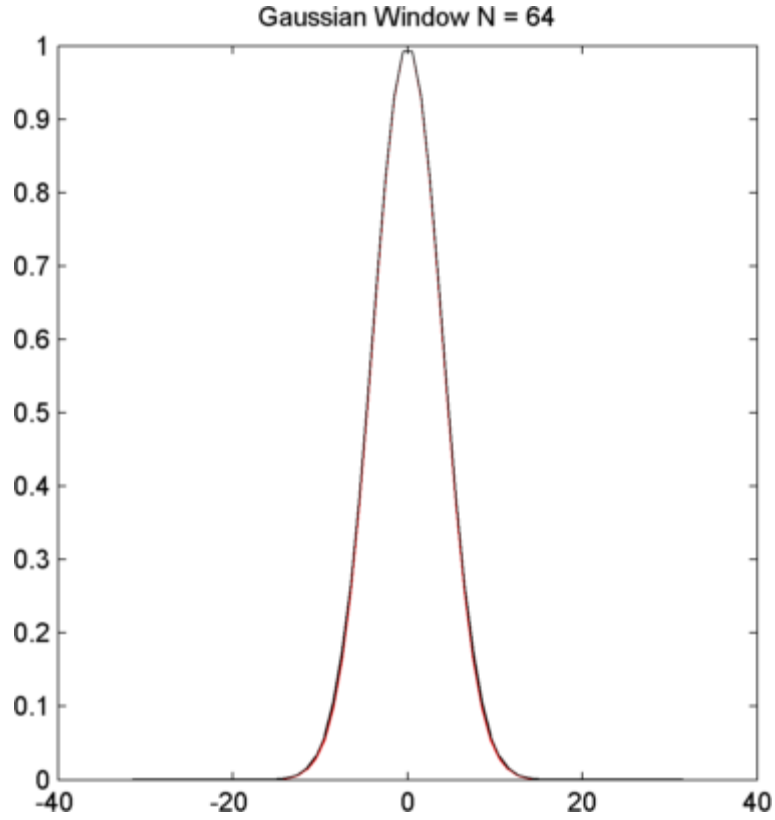
Gaussian Window and the Fourier Transform

This example demonstrates that the Fourier transform of the Gaussian window is also Gaussian with a reciprocal standard deviation. This is an illustration of the time-frequency uncertainty principle. Additionally, the example shows that the output of `gausswin` is equivalent to the equation given in the “Description” on page 1-603 section.

Create a Gaussian window of length 64 by using `gausswin` and the defining equation. Set α=8, which results in a standard deviation of 64/16=4. Accordingly, you expect that the Gaussian is essentially limited to the mean plus or minus 3 standard deviations, or an approximate support of [-12, 12].

```
N = 64;
n = -(N-1)/2:(N-1)/2;
alpha = 8;
y = exp(-1/2*(alpha*n/(N/2)).^2);
w = gausswin(N,alpha);
```

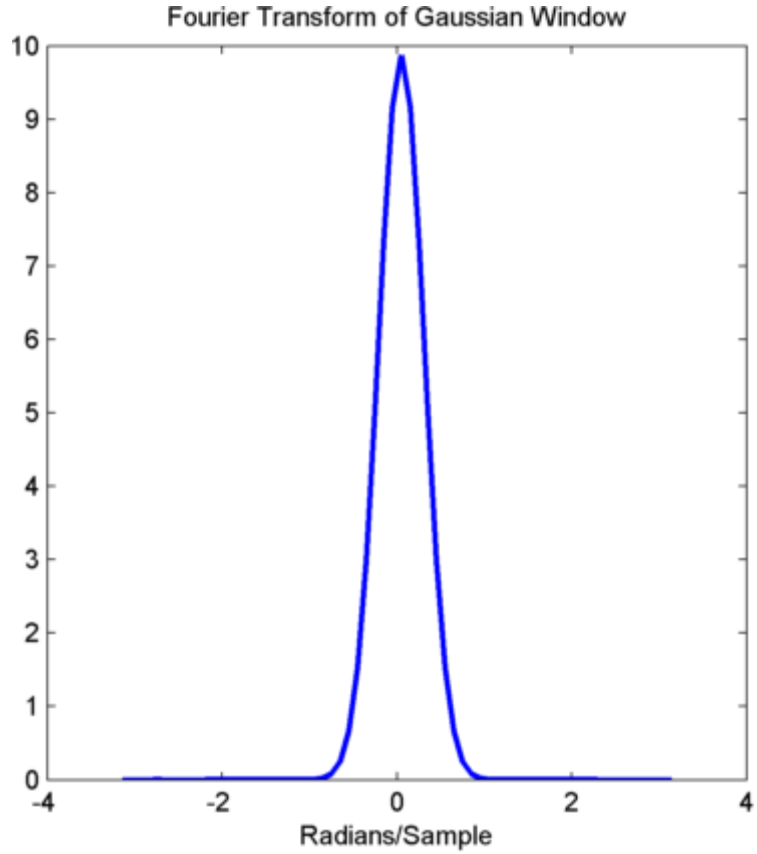
```
plot(n,w,'r')
hold on;
plot(n,y,'k')
title('Gaussian Window N = 64');
```



Obtain the Fourier transform of the Gaussian window and use `fftshift` to center the Fourier transform at zero frequency (DC).

```
figure
wdft = fftshift(fft(w));
freq = linspace(-pi,pi,length(wdft));
```

```
plot(freq,abs(wdft),'linewidth',2)
xlabel('Radians/Sample');
title('Fourier Transform of Gaussian Window');
```



The Fourier transform of the Gaussian window is also Gaussian with a standard deviation that is the reciprocal of the time-domain standard deviation.

References

[1] Harris, F.J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform.” *Proceedings of the IEEE. Vol. 66*, No. 1 (January 1978).

[2] Roberts, Richard A., and C.T. Mullis. *Digital Signal Processing*. Reading, MA: Addison-Wesley, 1987, pp. 135-136.

See Also

`chebwin` | `kaiser` | `tukeywin` | `window` | `wintool` | `wvtool`

gmonopuls

Purpose

Gaussian monopulse

Syntax

```
y = gmonopuls(t,fc)
tc = gmonopuls('cutoff',fc)
```

Description

`y = gmonopuls(t,fc)` returns samples of the unity-amplitude Gaussian monopulse with center frequency `fc` (in hertz) at the times indicated in array `t`. By default, `fc` = 1000 Hz.

`tc = gmonopuls('cutoff',fc)` returns the time duration between the maximum and minimum amplitudes of the pulse.

Tips

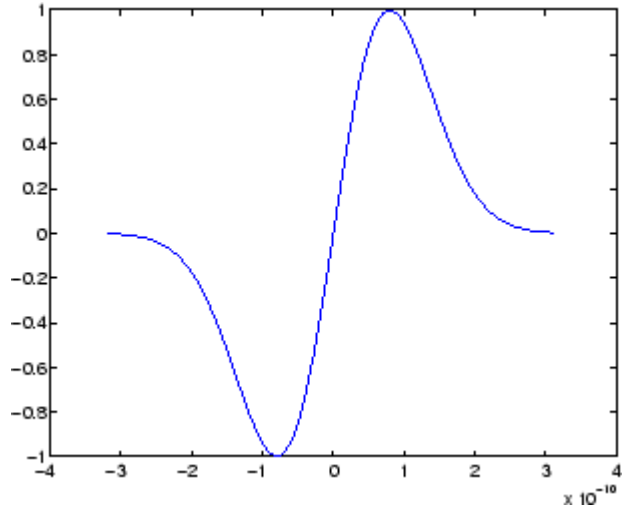
Default values are substituted for empty or omitted trailing input arguments.

Examples

Example 1

Plot a 2 GHz Gaussian monopulse sampled at a rate of 100 GHz:

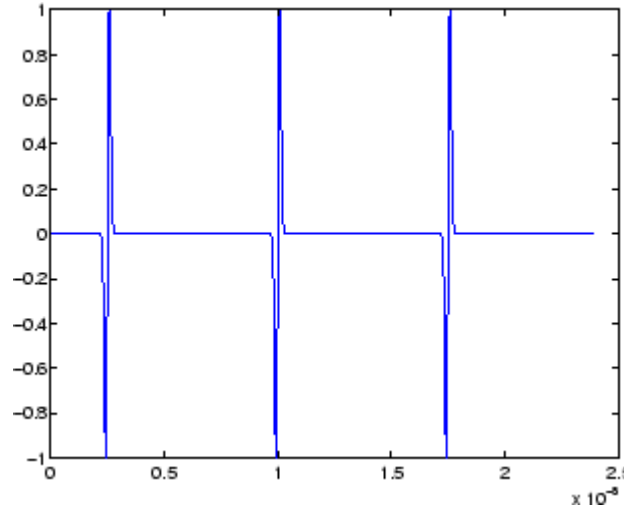
```
fc = 2E9; fs=100E9;
tc = gmonopuls('cutoff',fc);
t  = -2*tc : 1/fs : 2*tc;
y = gmonopuls(t,fc); plot(t,y)
```



Example 2

Construct a pulse train from the monopulse of Example 1 using a spacing of 7.5 ns:

```
fc = 2E9;  fs=100E9;              % Center freq, sample freq
D = [2.5 10 17.5]' * 1e-9;        % Pulse delay times
tc = gmonopuls('cutoff',fc);      % Width of each pulse
t  = 0 : 1/fs : 150*tc;           % Signal evaluation time
yp = pulstran(t,D,@gmonopuls,fc);
plot(t,yp)
```



See Also

`chirp` | `gauspuls` | `pulstran` | `rectpuls` | `tripuls`

Purpose

Discrete Fourier transform with second-order Goertzel algorithm

Syntax

```
dft_data = goertzel(data)
dft_data = goertzel(data,freq_indices)
dft_data = goertzel(data,freq_indices,dim)
```

Description

`dft_data = goertzel(data)` returns the discrete Fourier transform (DFT) of the input data `data` using a second-order Goertzel algorithm. If `data` is a matrix, `goertzel` computes the DFT of each column separately.

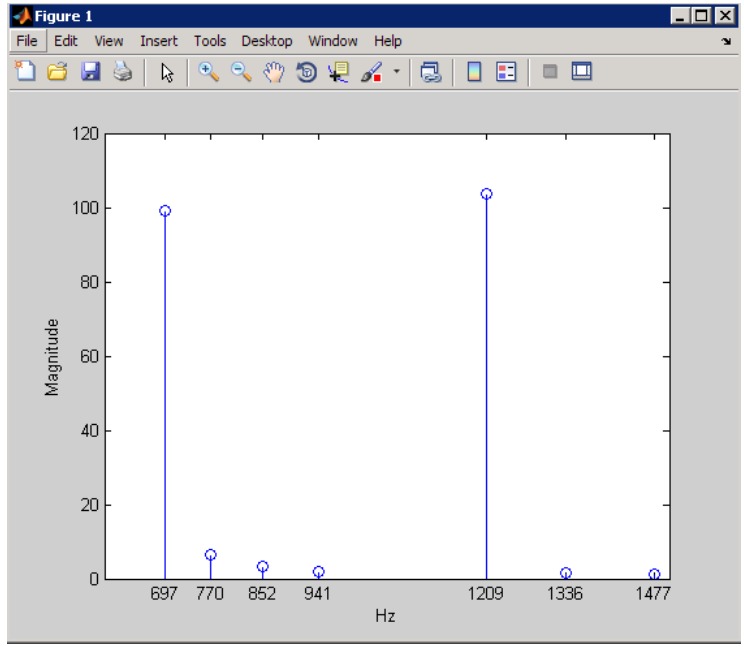
`dft_data = goertzel(data,freq_indices)` returns the DFT for the frequency indices `freq_indices`.

`dft_data = goertzel(data,freq_indices,dim)` computes the DFT of the matrix `data` along the dimension `dim`.

Examples

Estimate the frequency of the two tones generated by pressing the 1 button on a telephone keypad:

```
f=[697 770 852 941 1209 1336 1477];
% frequencies for numbers 0:9 on keypad
Fs = 8000; %sampling frequency
N = 205; %Number of points
% Tones generated by a "1": 697 and 1209 Hz
data = sum(sin(2*pi*[697;1209]*(0:N-1)/Fs));
% Indices of the DFT for the frequencies f
freq_indices = round(f/Fs*N)+1;
%Compute DFT using Goertzel algorithm
dft_data = goertzel(data,freq_indices);
%Plot the DFT magnitudes
stem(f,abs(dft_data));
set(gca,'xtick',f); xlabel('Hz');
ylabel('Magnitude');
```



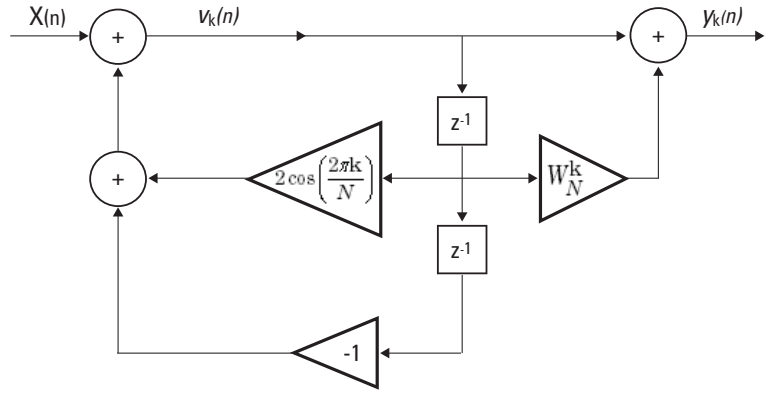
Algorithms

The Goertzel algorithm implements the DFT as a recursive difference equation. To establish this difference equation, express the DFT as the convolution of an N-point input *x(n)* with the impulse response $h(n) = W_N^{-kn}u(n)$, where $W_N^{-kn} = e^{-i2\pi k/N}$ and *u(n)* is the unit step sequence.

The z-transform of the impulse response is:

$$H(z) = \frac{1 - W_N^k z^{-1}}{1 - 2\cos(2\pi k/N)z^{-1} + z^{-2}}$$

The direct form II implementation is:



References

Proakis, J.G. and D.G. Manolakis. *Digital Signal Processing: Principles, Algorithms, and Applications*, Upper Saddle River, NJ: Prentice Hall, 1996, pp. 480–481.

Alternatives

You can also compute the DFT with:

- `fft` less efficient than the Goertzel algorithm when you only need the DFT at a few frequencies.
- `czt` the chirp z-transform. `czt` calculates the z-transform of an input on a circular or spiral contour and includes the DFT as a special case.

Purpose

Average filter delay (group delay)

Syntax

```
[gd,w] = grpdelay(b,a)
[gd,w] = grpdelay(b,a,n)
[gd,w] = grpdelay(sos,n)
[gd,w] = grpdelay(Hd,n)
[gd,f] = grpdelay(b,a,n,fs)
[gd,w] = grpdelay(b,a,n,'whole')
[gd,f] = grpdelay(b,a,n,'whole', fs)
gd = grpdelay(b,a,w)
gd = grpdelay(b,a,f,fs)
grpdelay(...)
```

Description

The *group delay* of a filter is a measure of the average delay of the filter as a function of frequency. It is the negative first derivative of the phase response of the filter. If the frequency response of a filter is $H(e^{j\omega})$, then the group delay is

$$\tau_g(\omega) = -\frac{d\theta(\omega)}{d\omega}$$

where $\theta(\omega)$ is the phase, or argument, of phase $H(e^{j\omega})$.

`[gd,w] = grpdelay(b,a)` returns the group delay, `gd`, of the discrete-time filter specified by the input vectors, `b` and `a`. The input vectors are the coefficients for the numerator, `b`, and denominator, `a`, polynomials in z^{-1}. The Z-transform of the discrete-time filter is

$$H(z) = \frac{B(z)}{A(z)} = \frac{\sum_{l=0}^{N-1} b(n+1)z^{-l}}{\sum_{l=0}^{M-1} a(l+1)z^{-l}},$$

The filter's group delay is evaluated at 512 equally-spaced points in the interval $[0,\pi)$ on the unit circle. The evaluation points on the unit circle are returned in `w`.

`[gd,w] = grpdelay(b,a,n)` returns the group delay of the discrete-time filter evaluated at `n` equally-spaced points on the unit circle in the interval [0,π). `n` is a positive integer.

`[gd,w] = grpdelay(sos,n)` returns the group delay for the second order sections matrix, `sos`. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. If the number of sections is less than 2, `grpdelay` considers the input to be the numerator vector, `b`. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`[gd,w] = grpdelay(Hd,n)` returns the group delay for the `dfilt` filter object, `Hd`, or the array of `dfilt` filter objects. If `Hd` is an array of `dfilt` objects, each column of `gd` is the group delay of the corresponding `dfilt` object.

`[gd,f] = grpdelay(b,a,n,fs)` specifies a positive sampling frequency `fs` in hertz. It returns a length `n` vector `f` containing the frequency points in hertz at which the group delay is evaluated. `f` contains `n` points between 0 and `fs/2`.

`[gd,w] = grpdelay(b,a,n,'whole')` and

`[gd,f] = grpdelay(b,a,n,'whole', fs)` use `n` points around the whole unit circle (from 0 to 2π, or from 0 to `fs`).

`gd = grpdelay(b,a,w)` and

`gd = grpdelay(b,a,f,fs)` return the group delay evaluated at the angular frequencies in `w` (in radians/sample) or in `f` (in cycles/unit time)), respectively, where `fs` is the sampling frequency. `w` and `f` are vectors with at least two elements.

`grpdelay(...)` plots the group delay versus frequency. The plot is displayed in `fvtool`. If the input is the numerator and denominator coefficients, a second order sections matrix, or a single `dfilt` object, the group delay of the single filter is displayed. If the input is an array of `dfilt` objects, the group delays of all filters in the array are displayed.

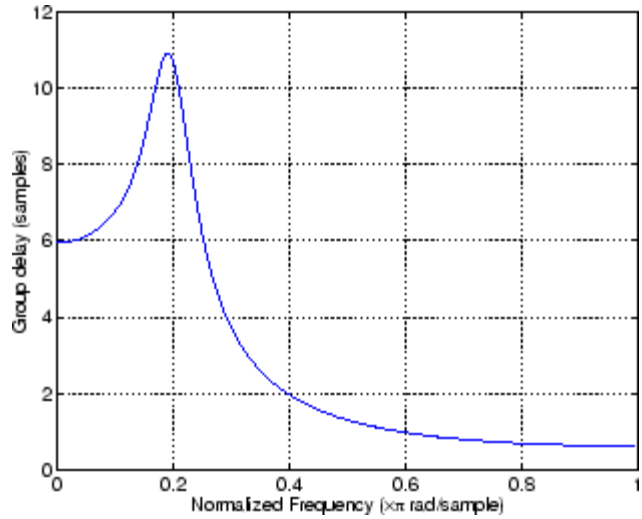
`grpdelay` works for both real and complex filters.

Note If the input to `grpdelay` is single precision, the group delay is calculated using single-precision arithmetic. The output, `gd`, is single precision.

Examples

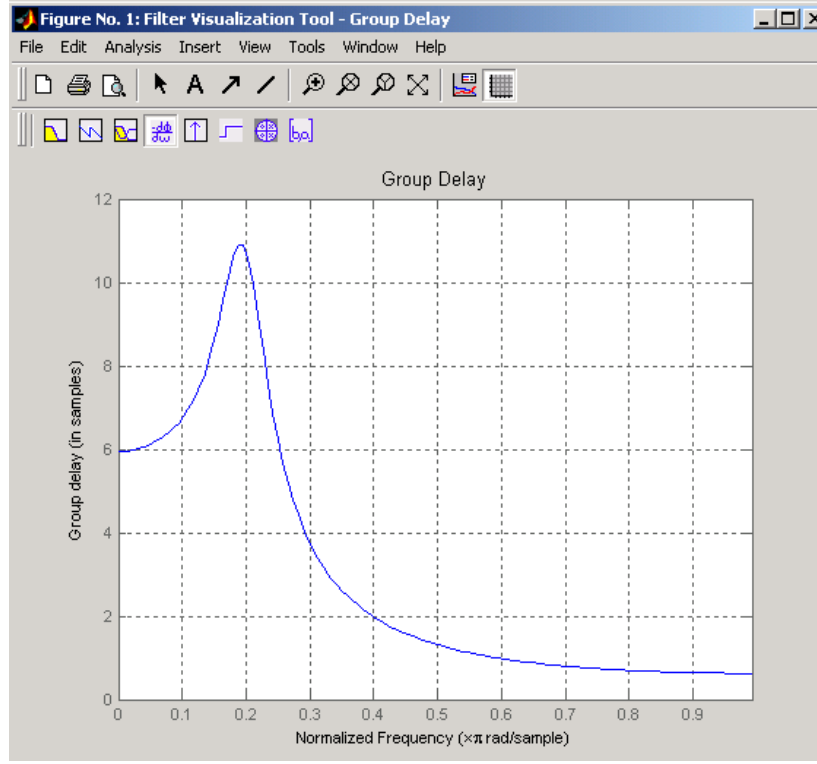
Plot the group delay of Butterworth filter `b(z)/a(z)`:

```
[b,a] = butter(6,0.2);
grpdelay(b,a,128)
```



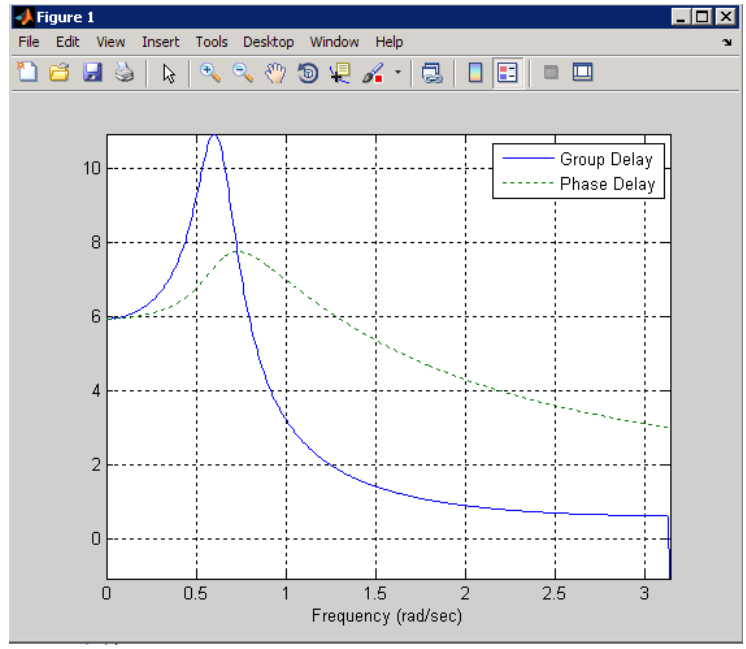
The same example using a `dfilt` object and displaying the result in the Filter Visualization Tool (`fvtool`) is

```
[b,a] = butter(6,0.2);
Hd=dfilt.df1(b,a);
grpdelay(Hd,128)
```



Plot both the group and phase delays of a system on the same graph:

```
[b,a] = butter(6,0.2);
gd = grpdelay(b,a,512);
gd(1) = [];       % Avoid NaNs
[h,w] = freqz(b,a,512); h(1) = []; w(1) = [];
pd = -unwrap(angle(h))./w;
plot(w,gd,w,pd,':')
axis([0 pi min(gd) max(gd)]);
xlabel('Frequency (rad/sec)'); grid;
legend('Group Delay','Phase Delay');
```



Algorithms

`grpdelay` multiplies the filter coefficients by a unit ramp. After Fourier transformation, this process corresponds to differentiation.

See Also

`cceps` | `fft` | `freqz` | `fvtool` | `hilbert` | `icceps` | `rceps`

Purpose

Hamming window

Syntax

```
w = hamming(L)
w = hamming(L,'sflag')
```

Description

`w = hamming(L)` returns an `L`-point symmetric Hamming window in the column vector `w`. `L` should be a positive integer. The coefficients of a Hamming window are computed from the following equation.

$$w(n) = 0.54 - 0.46\cos\left(2\pi\frac{n}{N}\right), \quad 0 \le n \le N$$

The window length is $L = N + 1$.

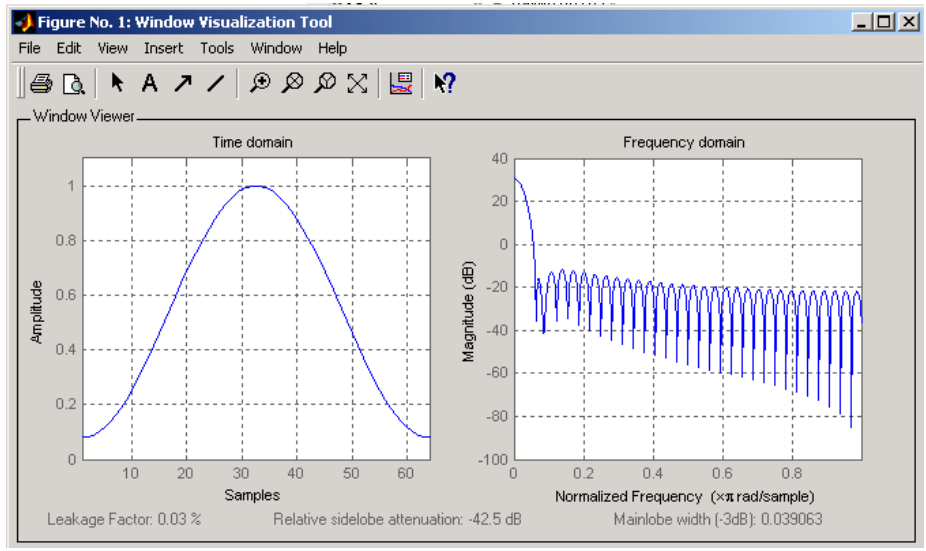
`w = hamming(L,'sflag')` returns an `L`-point Hamming window using the window sampling specified by `'sflag'`, which can be either `'periodic'` or `'symmetric'` (the default). The `'periodic'` flag is useful for DFT/FFT purposes, such as in spectral analysis. The DFT/FFT contains an implicit periodic extension and the periodic flag enables a signal windowed with a periodic window to have perfect periodic extension. When `'periodic'` is specified, `hamming` computes a length `L+1` window and returns the first `L` points. When using windows for filter design, the `'symmetric'` flag should be used.

Note If you specify a one-point window (`L=1`), the value `1` is returned.

Examples

Create a 64-point Hamming window and display the result in WVTool:

```
L=64;
wvtool(hamming(L))
```



References

[1] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, 1989, pp. 447-448.

See Also

`blackman` | `flattopwin` | `hann` | `window` | `wintool` | `wvtool`

Purpose

Hann (Hanning) window

Syntax

```
w = hann(L)
w = hann(L,'sflag')
```

Description

`w = hann(L)` returns an `L`-point symmetric Hann window in the column vector `w`. `L` must be a positive integer. The coefficients of a Hann window are computed from the following equation.

$$w(n) = 0.5\left(1 - \cos\left(2\pi\frac{n}{N}\right)\right), \quad 0 \le n \le N$$

The window length is $L = N + 1$.

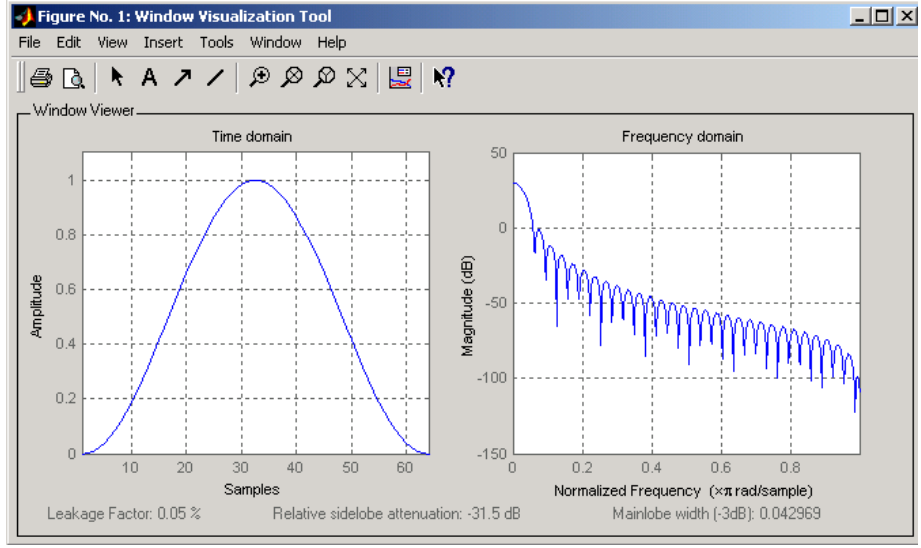
`w = hann(L,'sflag')` returns an `L`-point Hann window using the window sampling specified by `'sflag'`, which can be either `'periodic'` or `'symmetric'` (the default). The `'periodic'` flag is useful for DFT/FFT purposes, such as in spectral analysis. The DFT/FFT contains an implicit periodic extension and the periodic flag enables a signal windowed with a periodic window to have perfect periodic extension. When `'periodic'` is specified, `hann` computes a length `L+1` window and returns the first `L` points. When using windows for filter design, the `'symmetric'` flag should be used.

Note If you specify a one-point window (`L=1`), the value `1` is returned.

Examples

Create a 64-point Hann window and display the result in WVTool:

```
L=64;
wvtool(hann(L))
```



References

[1] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, 1989, pp. 447-448.

See Also

`blackman` | `flattopwin` | `hamming` | `window` | `wintool` | `wvtool`

Purpose

Discrete-time analytic signal using Hilbert transform

Syntax

```
x = hilbert(xr)
x = hilbert(xr,n)
```

Description

`x = hilbert(xr)` returns a complex helical sequence, sometimes called the *analytic signal*, from a real data sequence. The analytic signal `x = xr + i*xi` has a real part, `xr`, which is the original data, and an imaginary part, `xi`, which contains the Hilbert transform. The imaginary part is a version of the original real sequence with a 90° phase shift. Sines are therefore transformed to cosines and vice versa. The Hilbert transformed series has the same amplitude and frequency content as the original real data and includes phase information that depends on the phase of the original data.

If `xr` is a matrix, `x = hilbert(xr)` operates columnwise on the matrix, finding the Hilbert transform of each column.

`x = hilbert(xr,n)` uses an `n` point FFT to compute the Hilbert transform. The input data `xr` is zero-padded or truncated to length `n`, as appropriate.

The Hilbert transform is useful in calculating instantaneous attributes of a time series, especially the amplitude and frequency. The instantaneous amplitude is the amplitude of the complex Hilbert transform; the instantaneous frequency is the time rate of change of the instantaneous phase angle. For a pure sinusoid, the instantaneous amplitude and frequency are constant. The instantaneous phase, however, is a sawtooth, reflecting the way in which the local phase angle varies linearly over a single cycle. For mixtures of sinusoids, the attributes are short term, or local, averages spanning no more than two or three points.

Reference [1] describes the Kolmogorov method for minimum phase reconstruction, which involves taking the Hilbert transform of the logarithm of the spectral density of a time series. The toolbox function `rceps` performs this reconstruction.

For a discrete-time analytic signal `x`, the last half of `fft(x)` is zero, and the first (DC) and center (Nyquist) elements of `fft(x)` are purely real.

Examples

```
xr = [1 2 3 4];
x = hilbert(xr)
x
```

You can see that the imaginary part, `imag(x)` = `[1 -1 -1 1]`, is the Hilbert transform of `xr`, and the real part, `real(x)` = `[1 2 3 4]`, is simply `xr` itself. Note that the last half of `fft(x)` = `[10 -4+4i -2 0]` is zero (in this example, the last half is just the last element), and that the DC and Nyquist elements of `fft(x)`, 10 and -2 respectively, are purely real.

Algorithms

The analytic signal for a sequence `x` has a *one-sided Fourier transform*, that is, negative frequencies are 0. To approximate the analytic signal, `hilbert` calculates the FFT of the input sequence, replaces those FFT coefficients that correspond to negative frequencies with zeros, and calculates the inverse FFT of the result.

In detail, `hilbert` uses a four-step algorithm:

1. It calculates the FFT of the input sequence, storing the result in a vector `x`.
2. It creates a vector `h` whose elements `h(i)` have the values:
   - 1 for `i = 1, (n/2)+1`
   - 2 for `i = 2, 3, ... , (n/2)`
   - 0 for `i = (n/2)+2, ... , n`
3. It calculates the element-wise product of `x` and `h`.
4. It calculates the inverse FFT of the sequence obtained in step 3 and returns the first `n` elements of the result.

If the input data `xr` is a matrix, `hilbert` operates in a similar manner, extending each step above to handle the matrix case.

References

[1] Claerbout, J.F., *Fundamentals of Geophysical Data Processing*, McGraw-Hill, 1976, pp.59-62.

[2] Marple, S.L., "Computing the discrete-time analytic signal via FFT," IEEE Transactions on Signal Processing, Vol. 47, No. 9 (September 1999), pp. 2600-2603.

[3] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, 2nd ed., Prentice-Hall, 1998.

See Also

`fft | ifft | rceps`

icceps

Purpose

Inverse complex cepstrum

Syntax

```
x = icceps(xhat,nd)
```

Description

Note `icceps` only works on real data.

`x = icceps(xhat,nd)` returns the inverse complex cepstrum of the real data sequence `xhat`, removing `nd` samples of delay. If `xhat` was obtained with `cceps(x)`, then the amount of delay that was added to `x` was the element of `round(unwrap(angle(fft(x)))/pi)` corresponding to π radians.

References

[1] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, 1989.

See Also

`cceps` | `hilbert` | `rceps` | `unwrap`

Purpose

Inverse discrete cosine transform

Syntax

```
x = idct(y)
x = idct(y,n)
```

Description

The inverse discrete cosine transform reconstructs a sequence from its discrete cosine transform (DCT) coefficients. The `idct` function is the inverse of the `dct` function.

`x = idct(y)` returns the inverse discrete cosine transform of `y`

$$x(n) = \sum_{k=1}^{N} w(k)y(k)\cos\left(\frac{\pi(2n-1)(k-1)}{2N}\right) \quad n = 1,2,\ldots N$$

where

$$w(k) = \begin{cases} \dfrac{1}{\sqrt{N}} & k = 1 \\ \sqrt{\dfrac{2}{N}} & 2 \le k \le N \end{cases}$$

and N = `length(x)`, which is the same as `length(y)`. The series is indexed from n = 1 and k = 1 instead of the usual n = 0 and k = 0 because MATLAB vectors run from 1 to N instead of from 0 to N-1.

`x = idct(y,n)` appends zeros or truncates the vector `y` to length `n` before transforming.

If `y` is a matrix, `idct` transforms its columns.

References

[1] Jain, A.K., *Fundamentals of Digital Image Processing*, Prentice-Hall, 1989.

[2] Pennebaker, W.B., and J.L. Mitchell, *JPEG Still Image Data Compression Standard*, Van Nostrand Reinhold, 1993, Chapter 4.

See Also

`dct` | `dct2` | `idct2` | `ifft`

Purpose

Inverse Fast Walsh–Hadamard transform

Syntax

```
y = ifwht(x)
y = ifwht(x,n)
y = ifwht(x,n,ordering)
```

Description

`y = ifwht(x)` returns the coefficients of the inverse discrete fast Walsh–Hadamard transform of the input `x`. If `x` is a matrix, the inverse fast Walsh-Hadamard tranform is calculated on each column of `x`. The inverse fast Walsh-Hadamard tranform operates only on signals with length equal to a power of 2. If the length of `x` is less than a power of 2, its length is padded with zeros to the next greater power of two before processing.

`y = ifwht(x,n)` returns the `n`-point inverse discrete Walsh–Hadamard transform, where `n` must be a power of 2.

`y = ifwht(x,n,ordering)` specifies the ordering to use for the returned inverse Walsh–Hadamard transform coefficients. To specify ordering, you must enter a value for the length `n` or, to use the default behavior, specify an empty vector `[]` for `n`. Valid values for ordering are the following strings:

Ordering	Description
`'sequency'`	Coefficients in order of ascending sequency value, where each row has an additional zero crossing. This is the default ordering.
`'hadamard'`	Coefficients in normal Hadamard order.
`'dyadic'`	Coefficients in Gray code order, where a single bit change occurs from one coefficient to the next.

Algorithms

The inverse fast Walsh-Hadamard tranform algorithm is similar to the Cooley-Tukey algorithm used for the inverse FFT. Both use a butterfly structure to determine the transform coefficients. See the references below for details.

References

[1] Beauchamp, K.G., *Applications of Walsh and Related Functions*, Academic Press, 1984.

[2] Beer, T., *Walsh Transforms*, American Journal of Physics, Volume 49, Issue 5, May 1981.

See Also

`fwht` | `dct` | `idct` | `fft` | `ifft`

Purpose

Impulse invariance method for analog-to-digital filter conversion

Syntax

```
[bz,az] = impinvar(b,a,fs)
[bz,az] = impinvar(b,a,fs,tol)
```

Description

`[bz,az] = impinvar(b,a,fs)` creates a digital filter with numerator and denominator coefficients `bz` and `az`, respectively, whose impulse response is equal to the impulse response of the analog filter with coefficients `b` and `a`, scaled by `1/fs`. If you leave out the argument `fs`, or specify `fs` as the empty vector `[]`, it takes the default value of 1 Hz.

`[bz,az] = impinvar(b,a,fs,tol)` uses the tolerance specified by `tol` to determine whether poles are repeated. A larger tolerance increases the likelihood that `impinvar` interprets closely located poles as multiplicities (repeated ones). The default is 0.001, or 0.1% of a pole's magnitude. Note that the accuracy of the pole values is still limited to the accuracy obtainable by the `roots` function.

Examples

Example 1

Convert an analog lowpass filter to a digital filter using `impinvar` with a sampling frequency of 10 Hz:

```
[b,a] = butter(4,0.3,'s');
[bz,az] = impinvar(b,a,10);
```

Example 2

Illustrate the relationship between analog and digital impulse responses [2].

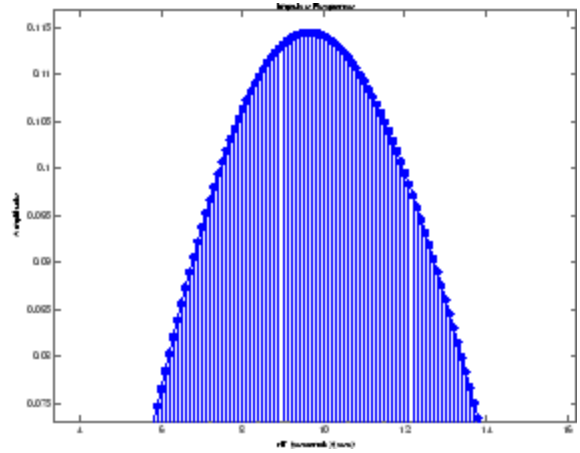
Note This example requires the `impulse` function from Control System Toolbox™ software.

The steps used in this example are:

1. Create an analog Butterworth filter
2. Use `impinvar` with a sampling frequency `Fs` of 10 Hz to scale the coefficients by 1/`Fs`. This compensates for the gain that will be introduced in Step 4 below.
3. Use Control System Toolbox `impulse` function to plot the continuous-time unit impulse response of an LTI system.
4. Plot the digital impulse response, multiplying the numerator by a constant (`Fs`) to compensate for the 1/`Fs` gain introduced in the impulse response of the derived digital filter.

```
[b,a] = butter(4,0.3,'s');
[bz,az] = impinvar(b,a,10);
sys = tf(b,a);
impulse(sys);
hold on;
impz(10*bz,az,[],10);
```

Zooming the resulting plot shows that the analog and digital impulse responses are the same.



Algorithms

`impinvar` performs the impulse-invariant method of analog-to-digital transfer function conversion discussed in reference [1]:

1. It finds the partial fraction expansion of the system represented by `b` and `a`.
2. It replaces the poles `p` by the poles `exp(p/fs)`.
3. It finds the transfer function coefficients of the system from the residues from step 1 and the poles from step 2.

References

[1] Parks, T.W., and C.S. Burrus, *Digital Filter Design*, John Wiley & Sons, 1987, pp.206-209.

[2] Antoniou, Andreas, *Digital Filters*, McGraw Hill, Inc, 1993, pp.221-224.

See Also

`bilinear` | `lp2bp` | `lp2bs` | `lp2hp` | `lp2lp`

Purpose

Impulse response of digital filter

Syntax

```
[h,t] = impz(b,a)
[h,t] = impz(sos)
[h,t] = impz(Hd)
[h,t] = impz(...,n)
[h,t] = impz(...,n,fs)
impz(...)
```

Description

`[h,t] = impz(b,a)` returns the impulse response of the filter with numerator coefficients `b` and denominator coefficients `a`. `impz` chooses the number of samples and returns the response in the column vector `h` and sample times in the column vector `t` (where `t = [0:n-1]'`, and `n = length(t)` is computed automatically).

Note If the input to `impz` is single precision, the impulse response is calculated using single-precision arithmetic. The output, `h`, is single precision.

`[h,t] = impz(sos)` returns the impulse response for the second order sections matrix, `sos`. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. If the number of sections is less than 2, `impz` considers the input to be the numerator vector, `b`. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`[h,t] = impz(Hd)` returns the impulse response for the `dfilt` filter object, `Hd`, or the array of `dfilt` filter objects. If `Hd` is an array of `dfilt` objects, each column of `h` is the impulse response of the corresponding `dfilt` object.

`[h,t] = impz(...,n)` computes n samples of the impulse response when n is an integer (`t = [0:n-1]'`). If n is a vector of integers, impz computes the impulse response at those integer locations, starting the response computation from 0 (and `t = n` or `t = [0 n]`). If, instead of `n`,

you include the empty vector `[]` for the second argument, the number of samples is computed automatically by default.

`[h,t] = impz(...,n,fs)` computes n samples and produces a vector `t` of length `n` so that the samples are spaced `1/fs` units apart.

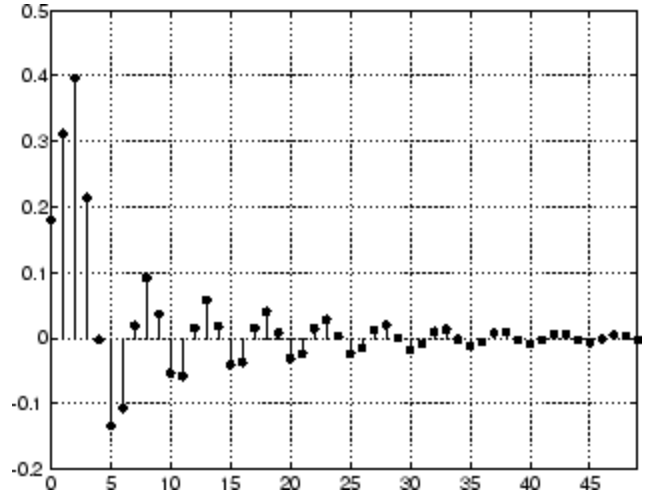
`impz(...)` with no output arguments plots the impulse response of the filter. If you input the filter coefficients or second order sections matrix, the current figure window is used. If you input a `dfilt` object or array of filter objects, `fvtool` is used to plot the impulse response.

Note `impz` works for both real and complex input systems.

Examples

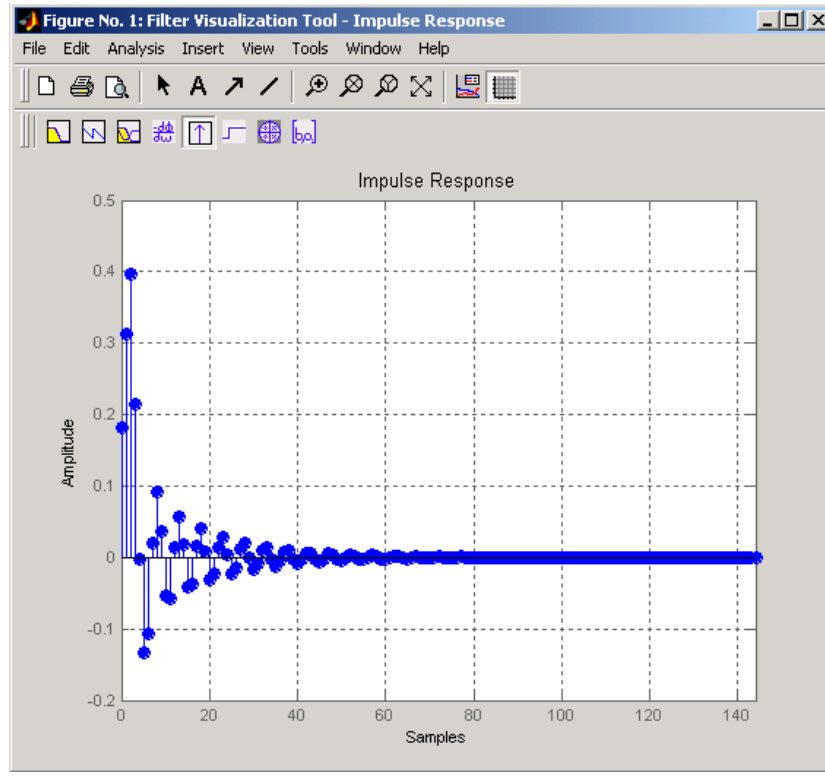
Plot the first 50 samples of the impulse response of a fourth-order lowpass elliptic filter with cutoff frequency of 0.4 times the Nyquist frequency:

```
[b,a] = ellip(4,0.5,20,0.4);
impz(b,a,50)
```



The same example using a `dfilt` object and displaying the result in the Filter Visualization Tool (`fvtool`) is

```
[b,a] = ellip(4,0.5,20,0.4);
Hd = dfilt.df1(b,a)
impz(Hd,50)
```



Algorithms

`impz` filters a length `n` impulse sequence using

```
filter(b,a,[1 zeros(1,n-1)])
```

and plots the results using `stem`.

To compute `n` in the auto-length case, `impz` either uses `n = length(b)` for the FIR case or first finds the poles using `p = roots(a)`, if `length(a)` is greater than 1.

If the filter is unstable, `n` is chosen to be the point at which the term from the largest pole reaches `10^6` times its original value.

If the filter is stable, `n` is chosen to be the point at which the term due to the largest amplitude pole is `5*10^-5` of its original amplitude.

If the filter is oscillatory (poles on the unit circle only), `impz` computes five periods of the slowest oscillation.

If the filter has both oscillatory and damped terms, `n` is chosen to equal five periods of the slowest oscillation or the point at which the term due to the largest (nonunity) amplitude pole is `5*10^-5` of its original amplitude, whichever is greater.

`impz` also allows for delays in the numerator polynomial. The number of delays is incorporated into the computation for the number of samples.

See Also

`impulse | stem`

Purpose

Impulse response length

Syntax

```
len = impzlength(b,a)
len = impzlength(sos)
len = impzlength(hd)
len = impzlength(hs)
len = impzlength(___,tol)
```

Description

`len = impzlength(b,a)` returns the impulse response length for the causal discrete-time filter with the rational system function specified by the numerator, `b`, and denominator, `a`, polynomials in z^{-1}. For stable IIR filters, `len` is the effective impulse response sequence length. Terms in the IIR filter's impulse response after the `len`-th term are essentially zero.

`len = impzlength(sos)` returns the effective impulse response length for the IIR filter specified by the second order sections matrix, `sos`. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. If the number of sections is less than 2, `impzlength` considers the input to be the numerator vector, `b`. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`len = impzlength(hd)` returns the impulse response length for the `dfilt` or `mfilt` filter object, `hd`. You must have the DSP System Toolbox software to use `impzlength` with an `mfilt` object. You can also input an array of filter objects. If `hd` is an array of filter objects, each column of `len` is the impulse response length of the corresponding filter object.

`len = impzlength(hs)` returns the impulse response length for the filter System object, `hs`. You must have the DSP System Toolbox software to use `impzlength` with a filter System object.

`len = impzlength( ___ ,tol)` specifies a tolerance for estimating the effective length of an IIR filter's impulse response. By default, `tol` is `5e-5`. Increasing the value of `tol` estimates a shorter effective length for an IIR filter's impulse response. Decreasing the value of `tol` produces a longer effective length for an IIR filter's impulse response.

Input Arguments

b - Numerator coefficients

vector | scalar

Numerator coefficients, specified as a scalar (allpole filter) or a vector.

Example: `b = fir1(20,0.25)`

Data Types
`single | double`
Complex Number Support: Yes

a - Denominator coefficients

vector | scalar

Denominator coefficients, specified as a scalar (FIR filter) or vector.

Data Types
`single | double`
Complex Number Support: Yes

sos - Matrix of second order sections

matrix

Matrix of second order sections, specified as a K-by-2 matrix. The system function of the K-th biquad filter has the rational z-transform

$$H_k(z) = \frac{B_k(1) + B_k(2)z^{-1} + B_k(3)z^{-2}}{A_k(1) + A_k(2)z^{-1} + A_k(3)z^{-2}}$$

The coefficients in the K-th row of the matrix, `sos`, are ordered as follows

$$[B_k(1)B_k(2)B_k(3)A_k(1)A_k(2)A_k(3)]$$

The frequency response of the filter is system function evaluated on the unit circle with

$$z = e^{i2\pi f}$$

hd - Filter object

Filter object, specified as a `dfilt` or `mfilt` object. You must have the DSP System Toolbox software to input an `mfilt` object.

tol - Tolerance for IIR filter effective impulse response length

`5e-5` (default) | positive scalar

Tolerance for IIR filter effective impulse response length, specified as a positive number. The tolerance determines the term in the absolutely summable sequence after which subsequent terms are considered to be 0. The default tolerance is `5e-5`. Increasing the tolerance returns a shorter effective impulse response sequence length. Decreasing the tolerance returns a longer effective impulse response sequence length.

hs - Filter System object

Filter System object, specified as one of the following:

- dsp.FIRFilter
- dsp.BiquadFilter
- dsp.FIRInterpolator
- dsp.CICInterpolator
- dsp.FIRDecimator
- dsp.CICDecimator
- dsp.FIRRateConverter

Using `impzlength` with a filter System object requires the DSP System Toolbox software.

Output Arguments

len - Length of impulse response

positive integer

Length of the impulse response, specified as a positive integer. For stable IIR filters with absolutely summable impulse responses, `impzlength` returns an effective length for the impulse response beyond which the coefficients are essentially zero. You can control this cutoff point by specifying the optional `tol` input argument.

Examples

IIR Filter Effective Impulse Response Length — — Coefficients

Create a lowpass allpole IIR filter with a pole at 0.9. Calculate the effective impulse response length, obtain the impulse response, and plot the result.

```
b = 1;
a = [1 -0.9];
len = impzlength(b,a)
[h,t] = impz(b,a);
stem(t,h)
h(len)
```

The value of the impulse response at the estimate length has decayed to approximately 10^{-6}.

Impulse Response Length — — Filter Objects

Design IIR Butterworth and FIR equiripple filters for data sampled at 1 kHz. The passband frequency is 100 Hz and the stopband frequency is 150 Hz. The passband ripple is 0.5 dB and there is 60 dB of stopband attenuation. Obtain `dfilt` objects for the filters and compare the filter impulse response sequence lengths.

```
d = fdesign.lowpass('Fp,Fst,Ap,Ast',100,150,0.5,60,1000);
Hd1 = design(d,'butter');
Hd2 = design(d,'equiripple');
len = impzlength([Hd1 Hd2])
```

IIR Filter Effective Impulse Response Length — — Second Order Sections

Design a 4-th order lowpass elliptic filter with a cutoff frequency of 0.4π radians/sample. Specify 1 dB of passband ripple and 60 dB of stopband attenuation. Design the filter in pole-zero-gain form and obtain the second order section matrix using `zp2sos`. Determine the effective impulse response sequence length from the second order sections matrix.

```
[z,p,k] = ellip(4,1,60,.4);
[sos,g] = zp2sos(z,p,k);
len = impzlength(sos)
```

Impulse Response Length of Filter System object

This example requires DSP System Toolbox software.

Design a 4-th order lowpass elliptic filter with a cutoff frequency of 0.4π radians/sample. Specify 1 dB of passband ripple and 60 dB of stopband attenuation. Design the filter in pole-zero-gain form and obtain the second order section matrix using `zp2sos`. Create a biquad filter System object and input the System object to `impzlength`.

```
[z,p,k] = ellip(4,1,60,.4);
[sos,g] = zp2sos(z,p,k);
hBqdFilt = dsp.BiquadFilter('Structure','Direct form I',...
                                              'SOSMatrix', sos,...
                                              'ScaleValues',g);
len = impzlength(hBqdFilt)
```

See Also

`impz` | `zp2sos`

interp

Purpose

Interpolation — increase sampling rate by integer factor

Syntax

```
y = interp(x,r)
y = interp(x,r,l,alpha)
[y,b] = interp(x,r,l,alpha)
```

Description

Interpolation increases the original sampling rate for a sequence to a higher rate. `interp` performs lowpass interpolation by inserting zeros into the original sequence and then applying a special lowpass filter. The filter returned by `intfilt` is identical to the filter used by `interp`.

`y = interp(x,r)` increases the sampling rate of `x` by a factor of `r`. The interpolated vector `y` is `r` times longer than the original input `x`.

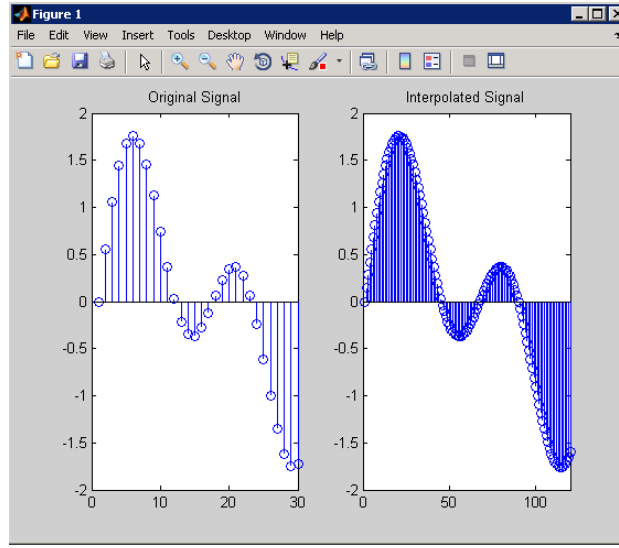
`y = interp(x,r,l,alpha)` specifies `l` (filter length) and `alpha` (cut-off frequency). The default value for `l` is 4 and the default value for `alpha` is 0.5.

`[y,b] = interp(x,r,l,alpha)` returns vector `b` containing the filter coefficients used for the interpolation.

Examples

Interpolate a signal by a factor of four:

```
t = 0:0.001:1;     % Time vector
x = sin(2*pi*30*t) + sin(2*pi*60*t);
y = interp(x,4);
subplot(121);
stem(x(1:30));
axis([0 30 -2 2]);
title('Original Signal');
subplot(122);
stem(y(1:120));
title('Interpolated Signal');
axis([0 120 -2 2]);
```



Algorithms

`interp` uses the lowpass interpolation Algorithm 8.1 described in [1]:

1. It expands the input vector to the correct length by inserting zeros between the original data values.
2. It designs a special symmetric FIR filter that allows the original data to pass through unchanged and interpolates between so that the mean-square errors between the interpolated points and their ideal values are minimized.
3. It applies the filter to the input vector to produce the interpolated output vector.

The length of the FIR lowpass interpolating filter is `2*l*r+1`. The number of original sample values used for interpolation is `2*l`. Ordinarily, `l` should be less than or equal to 10. The original signal is assumed to be band limited with normalized cutoff frequency $0 \le$ `alpha` ≤ 1, where 1 is half the original sampling frequency (the Nyquist

frequency). The default value for `l` is 4 and the default value for `alpha` is 0.5.

Diagnostics

If `r` is not an integer, `interp` gives the following error message:

```
Resampling rate R must be an integer.
```

References

[1] *Programs for Digital Signal Processing*, IEEE Press, New York, 1979, Algorithm 8.1.

See Also

`decimate` | `downsample` | `interp1` | `intfilt` | `resample` | `spline` | `upfirdn` | `upsample`

Purpose

Interpolation FIR filter design

Syntax

```
b = intfilt(l,p,alpha)
b = intfilt(l,n,'Lagrange')
```

Description

`b = intfilt(l,p,alpha)` designs a linear phase FIR filter that performs ideal bandlimited interpolation using the nearest `2*p` nonzero samples, when used on a sequence interleaved with `l-1` consecutive zeros every `l` samples. It assumes an original bandlimitedness of alpha times the Nyquist frequency. The returned filter is identical to that used by `interp`. `b` is length `2*l*p-1`

alpha is inversely proportional to the transition bandwidth of the filter and it also affects the bandwith of the don't-care regions in the stopband. Specifying alpha allows you to specify how much of the Nyquist interval your input signal occupies. This is beneficial, particularly for signals to be interpolated, because it allows you to increase the transition bandwidth without affecting the interpolation and results in better stopband attenuation for a given `l` and `p`. If you set alpha to 1, your signal is assumed to occupy the entire Nyquist interval. Setting alpha to less than one allows for don't-care regions in the stopband. For example, if your input occupies half the Nyquist interval, you could set alpha to 0.5.

`b = intfilt(l,n,'Lagrange')` designs an FIR filter that performs `n`th-order Lagrange polynomial interpolation on a sequence interleaved with `l-1` consecutive zeros every `r` samples. `b` has length `(n + 1)*l` for `n` even, and length `(n + 1)*l-1` for `n` odd. If both `n` and `l` are even, the filter designed is not linear phase.

Both types of filters are basically lowpass and have a gain of `l` in the passband..

Examples

Design a digital interpolation filter to upsample a signal by four, using the bandlimited method:

```
alpha = 0.5;                % "Bandlimitedness" factor
h1 = intfilt(4,2,alpha);    % Bandlimited interpolation
```

The filter `h1` works best when the original signal is bandlimited to alpha times the Nyquist frequency. Create a bandlimited noise signal:

```
x = filter(fir1(40,0.5),1,randn(200,1));   % Bandlimit
```

Now zero pad the signal with three zeros between every sample. The resulting sequence is four times the length of `x`:

```
xr = reshape([x zeros(length(x),3)]',4*length(x),1);
```

Interpolate using the `filter` command:

```
y = filter(h1,1,xr);
```

`y` is an interpolated version of `x`, delayed by seven samples (the group-delay of the filter). Zoom in on a section of one hundred samples to see this:

```
plot(100:200,y(100:200),7+(101:4:196),x(26:49),'o')
```

`intfilt` also performs Lagrange polynomial interpolation of the original signal. For example, first-order polynomial interpolation is just linear interpolation, which is accomplished with a triangular filter:

```
h2 = intfilt(4,1,'l');  % Lagrange interpolation
```

Algorithms

The bandlimited method uses `firls` to design an interpolation FIR filter. The polynomial method uses Lagrange's polynomial interpolation formula on equally spaced samples to construct the appropriate filter.

See Also

`decimate` | `downsample` | `interp` | `resample` | `upsample`

Purpose

Identify continuous-time filter parameters from frequency response data

Syntax

```
[b,a] = invfreqs(h,w,n,m)
[b,a] = invfreqs(h,w,n,m,wt)
[b,a] = invfreqs(h,w,n,m,wt,iter)
[b,a] = invfreqs(h,w,n,m,wt,iter,tol)
[b,a] = invfreqs(h,w,n,m,wt,iter,tol,'trace')
[b,a] = invfreqs(h,w,'complex',n,m,...)
```

Description

`invfreqs` is the inverse operation of `freqs`. It finds a continuous-time transfer function that corresponds to a given complex frequency response. From a laboratory analysis standpoint, `invfreqs` is useful in converting magnitude and phase data into transfer functions.

`[b,a] = invfreqs(h,w,n,m)` returns the real numerator and denominator coefficient vectors `b` and `a` of the transfer function

$$H(s) = \frac{B(s)}{A(s)} = \frac{b(1)s^n + b(2)s^{n-1} + \cdots + b(n+1)}{a(1)s^m + a(2)s^{m-1} + \cdots + a(m+1)}$$

whose complex frequency response is given in vector `h` at the frequency points specified in vector `w`. Scalars `n` and `m` specify the desired orders of the numerator and denominator polynomials.

The length of `h` must be the same as the length of `w`. `invfreqs` uses `conj(h)` at `-w` to ensure the proper frequency domain symmetry for a real filter.

`[b,a] = invfreqs(h,w,n,m,wt)` weights the fit-errors versus frequency, where `wt` is a vector of weighting factors the same length as `w`.

`[b,a] = invfreqs(h,w,n,m,wt,iter)` and

`[b,a] = invfreqs(h,w,n,m,wt,iter,tol)` provide a superior algorithm that guarantees stability of the resulting linear system and searches for the best fit using a numerical, iterative scheme. The `iter` parameter tells `invfreqs` to end the iteration when the solution has converged, or after `iter` iterations, whichever comes first. `invfreqs` defines convergence as occurring when the norm of the (modified) gradient

vector is less than `tol`, where `tol` is an optional parameter that defaults to 0.01. To obtain a weight vector of all ones, use

```
invfreqs(h,w,n,m,[],iter,tol)
```

`[b,a] = invfreqs(h,w,n,m,wt,iter,tol,'trace')` displays a textual progress report of the iteration.

`[b,a] = invfreqs(h,w,'complex',n,m,...)` creates a complex filter. In this case no symmetry is enforced, and the frequency is specified in radians between -π and π.

Tips

When building higher order models using high frequencies, it is important to scale the frequencies, dividing by a factor such as half the highest frequency present in `w`, so as to obtain well conditioned values of `a` and `b`. This corresponds to a rescaling of time.

Examples

Example 1

Convert a simple transfer function to frequency response data and then back to the original filter coefficients:

```
a = [1 2 3 2 1 4]; b = [1 2 3 2 3];
[h,w] = freqs(b,a,64);
[bb,aa] = invfreqs(h,w,4,5)
bb =
    1.0000   2.0000   3.0000   2.0000   3.0000
aa =
    1.0000   2.0000   3.0000   2.0000   1.0000   4.0000
```

Notice that `bb` and `aa` are equivalent to `b` and `a`, respectively. However, `aa` has poles in the right half-plane and thus the system is unstable. Use `invfreqs`'s iterative algorithm to find a stable approximation to the system:

```
[bbb,aaa] = invfreqs(h,w,4,5,[],30)
bbb =
    0.6816   2.1015   2.6694   0.9113  -0.1218
```

```
aaa =
    1.0000    3.4676    7.4060    6.2102    2.5413    0.0001
```

Example 2

Suppose you have two vectors, `mag` and `phase`, that contain magnitude and phase data gathered in a laboratory, and a third vector `w` of frequencies. You can convert the data into a continuous-time transfer function using `invfreqs`:

```
[b,a] = invfreqs(mag.*exp(j*phase),w,2,3);
```

Algorithms

By default, `invfreqs` uses an equation error method to identify the best model from the data. This finds `b` and `a` in

$$\min_{b,a} \sum_{k=1}^{n} wt(k)\left|h(k)A(w(k)) - B(w(k))\right|^2$$

by creating a system of linear equations and solving them with the MATLAB `\` operator. Here $A(w(k))$ and $B(w(k))$ are the Fourier transforms of the polynomials `a` and `b`, respectively, at the frequency $w(k)$, and n is the number of frequency points (the length of `h` and `w`). This algorithm is based on Levi [1]. Several variants have been suggested in the literature, where the weighting function `wt` gives less attention to high frequencies.

The superior ("output-error") algorithm uses the damped Gauss-Newton method for iterative search [2], with the output of the first algorithm as the initial estimate. This solves the direct problem of minimizing the weighted sum of the squared error between the actual and the desired frequency response points.

$$\min_{b,a} \sum_{k=1}^{n} wt(k)\left|h(k) - \frac{B(w(k))}{A(w(k))}\right|^2$$

References

[1] Levi, E.C., "Complex-Curve Fitting," *IRE Trans. on Automatic Control*, Vol.AC-4 (1959), pp.37-44.

[2] Dennis, J.E., Jr., and R.B. Schnabel, *Numerical Methods for Unconstrained Optimization and Nonlinear Equations*, Englewood Cliffs, NJ: Prentice-Hall, 1983.

See Also

`freqs` | `freqz` | `invfreqz` | `prony`

Purpose

Identify discrete-time filter parameters from frequency response data

Syntax

```
[b,a] = invfreqz(h,w,n,m)
[b,a] = invfreqz(h,w,n,m,wt)
[b,a] = invfreqz(h,w,n,m,wt,iter)
[b,a] = invfreqz(h,w,n,m,wt,iter,tol)
[b,a] = invfreqz(h,w,n,m,wt,iter,tol,'trace')
[b,a] = invfreqz(h,w,'complex',n,m,...)
```

Description

`invfreqz` is the inverse operation of `freqz`; it finds a discrete-time transfer function that corresponds to a given complex frequency response. From a laboratory analysis standpoint, `invfreqz` can be used to convert magnitude and phase data into transfer functions.

`[b,a] = invfreqz(h,w,n,m)` returns the real numerator and denominator coefficients in vectors `b` and `a` of the transfer function

$$H(z) = \frac{B(z)}{A(z)} = \frac{b(1) + b(2)z^{-1} + \cdots + b(n+1)z^{-n}}{a(1) + a(2)z^{-1} + \cdots + a(m+1)z^{-m}}$$

whose complex frequency response is given in vector `h` at the frequency points specified in vector `w`. Scalars `n` and `m` specify the desired orders of the numerator and denominator polynomials.

Frequency is specified in radians between 0 and π, and the length of `h` must be the same as the length of `w`. `invfreqz` uses `conj(h)` at `-w` to ensure the proper frequency domain symmetry for a real filter.

`[b,a] = invfreqz(h,w,n,m,wt)` weights the fit-errors versus frequency, where `wt` is a vector of weighting factors the same length as `w`.

`[b,a] = invfreqz(h,w,n,m,wt,iter)` and

`[b,a] = invfreqz(h,w,n,m,wt,iter,tol)` provide a superior algorithm that guarantees stability of the resulting linear system and searches for the best fit using a numerical, iterative scheme. The `iter` parameter tells `invfreqz` to end the iteration when the solution has converged, or after `iter` iterations, whichever comes first. `invfreqz` defines convergence as occurring when the norm of the (modified)

gradient vector is less than `tol`, where `tol` is an optional parameter that defaults to 0.01. To obtain a weight vector of all ones, use

```
invfreqz(h,w,n,m,[],iter,tol)
```

`[b,a] = invfreqz(h,w,n,m,wt,iter,tol,'trace')` displays a textual progress report of the iteration.

`[b,a] = invfreqz(h,w,'complex',n,m,...)` creates a complex filter. In this case no symmetry is enforced, and the frequency is specified in radians between -π and π.

Examples

Convert a simple transfer function to frequency response data and then back to the original filter coefficients:

```
a = [1 2 3 2 1 4]; b = [1 2 3 2 3];
[h,w] = freqz(b,a,64);
[bb,aa] = invfreqz(h,w,4,5)
bb =
    1.0000   2.0000   3.0000   2.0000   3.0000
aa =
    1.0000   2.0000   3.0000   2.0000   1.0000   4.0000
```

Notice that `bb` and `aa` are equivalent to `b` and `a`, respectively. However, `aa` has poles outside the unit circle and thus the system is unstable. Use `invfreqz`'s iterative algorithm to find a stable approximation to the system:

```
[bbb,aaa] = invfreqz(h,w,4,5,[],30)
bbb =
    0.2427   0.2788   0.0069   0.0971   0.1980
aaa =
    1.0000  -0.8944   0.6954   0.9997  -0.8933   0.6949
```

Algorithms

By default, `invfreqz` uses an equation error method to identify the best model from the data. This finds `b` and `a` in

$$\min_{b,a} \sum_{k=1}^{n} wt(k)\left|h(k)A(w(k)) - B(w(k))\right|^2$$

by creating a system of linear equations and solving them with the MATLAB `\` operator. Here $A(\omega(k))$ and $B(\omega(k))$ are the Fourier transforms of the polynomials `a` and `b`, respectively, at the frequency $\omega(k)$, and n is the number of frequency points (the length of `h` and `w`). This algorithm is a based on Levi [1].

The superior ("output-error") algorithm uses the damped Gauss-Newton method for iterative search [2], with the output of the first algorithm as the initial estimate. This solves the direct problem of minimizing the weighted sum of the squared error between the actual and the desired frequency response points.

$$\min_{b,a} \sum_{k=1}^{n} wt(k)\left|h(k) - \frac{B(w(k))}{A(w(k))}\right|^2$$

References

[1] Levi, E.C., "Complex-Curve Fitting," IRE Trans. on Automatic Control, Vol. AC-4 (1959), pp. 37-44.

[2] Dennis, J.E., Jr., and R.B. Schnabel, *Numerical Methods for Unconstrained Optimization and Nonlinear Equations*, Prentice-Hall, 1983.

See Also

`freqs` | `freqz` | `invfreqz` | `prony`

isallpass

Purpose

Determine whether filter is allpass

Syntax

```
flag = isallpass(b,a)
flag = isallpass(hd)
flag = isallpass(sos)
flag = islinphase(...,tol)
flag = isallpass(hs,...)
flag = isallpass(hs,'Arithmetic',arithtype)
```

Description

`flag = isallpass(b,a)` returns a logical output, `flag`, equal to `true` if the filter specified by numerator coefficients, `b`, and denominator coefficients, `a`, is an allpass filter. If the filter is not an allpass filter, `flag` is equal to `false`.

`flag = isallpass(hd)` returns `true`if the filter object, `hd`, is an allpass filter.

`flag = isallpass(sos)` returns `true` if the filter specified by second order sections matrix, `sos`, is an allpass filter. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`flag = islinphase(...,tol)` uses the tolerance, `tol`, to determine when two numbers are close enough to be considered equal. If not specified, `tol`, defaults to `eps^(2/3)`. Specifying a tolerance may be most helpful in fixed-point allpass filters.

`flag = isallpass(hs,...)` returns `true` if the filter System object `hs` is an allpass filter. You must have the DSP System Toolbox software to use this syntax.

`flag = isallpass(hs,'Arithmetic',arithtype)` analyzes the filter System object `hs` based on the specified *`arithtype`*. *`arithtype`* can be `'double'`, `'single'`, or `'fixed'`. When you specify `'double'` or `'single'`, the function performs double- or single-precision analysis. When you specify `'fixed'` , the arithmetic changes depending on the setting of the `CoefficientDataType` property and whether the System

object is locked or unlocked. You must have the DSP System Toolbox software to use this syntax.

Details for Fixed-Point Arithmetic

System Object State	Coefficient Data Type	Rule
Unlocked	`'Same as input'`	The function assumes that the coefficient data type is signed, 16 bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Unlocked	`'Custom'`	The function performs fixed-point analysis based on the setting of the `CustomCoefficientsDataType` property.
Locked	`'Same as input'`	When the input data type is `'double'` or `'fixed'`, the function assumes that the coefficient data type is signed, 16-bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Locked	`'Custom'`	The function performs fixed-point analysis based on

System Object State	Coefficient Data Type	Rule
		the setting of the `CustomCoefficientsDataType` property.

When you do not specify the arithmetic for non-CIC structures, the function uses double-precision arithmetic if the filter System object is in an unlocked state. If the System object is locked, the function performs analysis based on the locked input data type. CIC structures only support fixed-point arithmetic.

Examples

Create an allpass filter and verify that the frequency response is allpass.

```
b = [1/3 1/4 1/5 1];
a = fliplr(b);
flag = isallpass(b,a)
fvtool(b,a)
```

Create a lattice allpass filter and verify that the filter is allpass.

```
k = [1/2 1/3 1/4 1/5];
[b,a] = latc2tf(k,'allpass');
flag_isallpass = isallpass(b,a)
fvtool(b,a)
```

See Also

`islinphase` | `ismaxphase` | `isminphase` | `isstable`

Purpose

Determine whether filter has linear phase

Syntax

```
flag = islinphase(b,a)
flag = islinphase(sos)
flag = islinphase(h)
flag = islinphase(...,tol)
flag = islinphase(hs,...)
flag = islinphase(hs,'Arithmetic',arithtype)
```

Description

`flag = islinphase(b,a)` returns a logical output, `flag`, equal to `true` if the filter coefficients in `b` and `a` define a linear phase filter. `flag` is equal to `false` if the filter does not have linear phase.

`flag = islinphase(sos)` returns `true` if the filter specified by second order sections matrix, `sos`, is linear phase. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`flag = islinphase(h)` determines if the filter object `h` is linear phase. `islinphase` accepts an `adapfilt`, `dfilt`, or `mfilt` object. To create an `adapfilt` or `mfilt` object, you must have the DSP System Toolbox software.

`flag = islinphase(...,tol)` uses the tolerance, `tol`, to determine when two numbers are close enough to be considered equal. If not specified, `tol`, defaults to `eps^(2/3)`.

`flag = islinphase(hs,...)` determines whether the filter System object `hs` is linear phase. You must have the DSP System Toolbox to use `islinphase` with a System object.

`flag = islinphase(hs,'Arithmetic',arithtype)` analyzes the filter System object `hs` based on the specified *`arithtype`*. *`arithtype`* can be one of `'double'`, `'single'`, or `'fixed'`. When you specify `'double'` or `'single'`, the function performs double- or single-precision analysis. When you specify `'fixed'` , the arithmetic changes depending on the setting of the `CoefficientDataType` property and whether the System

object is locked or unlocked. You must have the DSP System Toolbox to use `islinphase` with a System object.

Details for Fixed-Point Arithmetic

System Object State	Coefficient Data Type	Rule
Unlocked	`'Same as input'`	The function assumes that the coefficient data type is signed, 16 bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Unlocked	`'Custom'`	The function performs fixed-point analysis based on the setting of the `CustomCoefficientsDataType` property.
Locked	`'Same as input'`	When the input data type is `'double'` or `'fixed'`, the function assumes that the coefficient data type is signed, 16-bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Locked	`'Custom'`	The function performs fixed-point analysis based on

System Object State	Coefficient Data Type	Rule
		the setting of the `CustomCoefficientsDataType` property.

When you do not specify the arithmetic for non-CIC structures, the function uses double-precision arithmetic if the filter System object is in an unlocked state. If the System object is locked, the function performs analysis based on the locked input data type. CIC structures only support fixed-point arithmetic.

Examples

This FIR filter has linear phase.

```
d = fdesign.lowpass('n,fc',10,0.55);
h = design(d,'window');
flag = islinphase(h)
```

Using the specification `nb,na,fp,fst` results in an IIR filter that is not linear phase in this design.

```
nb=15
na=10
d=fdesign.lowpass('nb,na,fp,fst',nb,na,0.45,0.55)
h=design(d);
flag = islinphase(h)
```

See Also

`isallpass` | `ismaxphase` | `isminphase` | `isstable`

isminphase

Purpose

Determine whether filter is minimum phase

Syntax

```
flag = isminphase(b,a)
flag = isminphase(sos)
flag = isminphase(h)
flag = isminphase(...,tol)
flag = isminphase(hs,...)
isminphase(hs,'Arithmetic',arithtype)
```

Description

`flag = isminphase(b,a)` returns a logical output, `flag`, equal to `true` if the filter specified by numerator coefficients, `b`, and denominator coefficients, `a`, is a minimum phase filter.

`flag = isminphase(sos)` returns `true` if the filter specified by second order sections matrix, `sos`, is minimum phase. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`flag = isminphase(h)` determines if the `dfilt` filter object `h` is minimum phase. If you have the DSP System Toolbox software, `isminphase` works with `adapfilt` and `mfilt` objects.

`flag = isminphase(...,tol)` uses the tolerance, `tol`, to determine when two numbers are close enough to be considered equal. If not specified, `tol`, defaults to `eps^(2/3)`.

A filter is *minimum phase* when all the zeros of its transfer function are on or inside the unit circle, or the numerator is a scalar. An equivalent definition for a minimum phase filter is a causal and stable system with a causal and stable inverse.

`flag = isminphase(hs,...)` determines whether the filter System object `hs` is minimum phase, returning `1` if true and `0` if false. You must have the DSP System Toolbox software to use this syntax.

`isminphase(hs,'Arithmetic',arithtype)` analyzes the filter System object `hs` based on the specified *`arithtype`*. *`arithtype`* can be `'double'`, `'single'`, or `'fixed'`. When you specify `'double'` or

'single', the function performs double- or single-precision analysis. When you specify 'fixed' , the arithmetic changes depending on the setting of the CoefficientDataType property and whether the System object is locked or unlocked. You must have the DSP System Toolbox software to use this syntax.

Details for Fixed-Point Arithmetic

System Object State	Coefficient Data Type	Rule
Unlocked	'Same as input'	The function assumes that the coefficient data type is signed, 16 bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Unlocked	'Custom'	The function performs fixed-point analysis based on the setting of the CustomCoefficientsDataType property.

System Object State	Coefficient Data Type	Rule
Locked	`'Same as input'`	When the input data type is `'double'` or `'fixed'`, the function assumes that the coefficient data type is signed, 16-bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Locked	`'Custom'`	The function performs fixed-point analysis based on the setting of the `CustomCoefficientsDataType` property.

When you do not specify the arithmetic for non-CIC structures, the function uses double-precision arithmetic if the filter System object is in an unlocked state. If the System object is locked, the function performs analysis based on the locked input data type. CIC structures only support fixed-point arithmetic.

Examples

Design a lowpass Butterworth IIR filter using second order sections and check if the filter is minimum phase.

```
[z,p,k] = butter(6,0.15);
SOS = zp2sos(z,p,k);
min_flag = isminphase(SOS)
```

For a filter defined with a set of single precision numerator and denominator coefficients, check if the filter is minimum phase for different tolerances.

```
b = single([1 1.00001]);
a = single([1 .45]);
min_flag1 = isminphase(b,a)
min_flag2 = isminphase(b,a,1e-3)
```

See Also

isallpass | islinphase | ismaxphase | isstable

ismaxphase

Purpose

Determine whether filter is maximum phase

Syntax

```
flag = ismaxphase(b,a)
flag = ismaxphase(sos)
flag = ismaxphase(h)
flag = ismaxphase(...,tol)
flag = ismaxphase(hs,...)
flag = ismaxphase(hs,'Arithmetic',arithtype)
```

Description

`flag = ismaxphase(b,a)` returns a logical output, `flag`, equal to `true` if the filter specified by numerator coefficients, `b`, and denominator coefficients, `a`, is a maximum phase filter.

`flag = ismaxphase(sos)` returns `true` if the filter specified by second order sections matrix, `sos`, is a maximum phase filter. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`flag = ismaxphase(h)` returns `true` if the `dfilt` filter object `h` is a maximum phase filter. If you have the DSP System Toolbox software, `ismaxphase` works with `adapfilt` and `mfilt` objects.

`flag = ismaxphase(...,tol)` uses the tolerance, `tol`, to determine when two numbers are close enough to be considered equal. If not specified, `tol`, defaults to `eps^(2/3)`.

`flag = ismaxphase(hs,...)` returns `true`if the filter System object `hs` is a maximum phase filter. You must have the DSP System Toolbox software to use this syntax.

`flag = ismaxphase(hs,'Arithmetic',arithtype)` analyzes the filter System object `hs` based on the specified *`arithtype`*. *`arithtype`* can be `'double'`, `'single'`, or `'fixed'`. When you specify `'double'` or `'single'`, the function performs double- or single-precision analysis. When you specify `'fixed'` , the arithmetic changes depending on the setting of the `CoefficientDataType` property and whether the System

object is locked or unlocked. You must have the DSP System Toolbox software to use this syntax.

Details for Fixed-Point Arithmetic

System Object State	Coefficient Data Type	Rule
Unlocked	`'Same as input'`	The function assumes that the coefficient data type is signed, 16 bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Unlocked	`'Custom'`	The function performs fixed-point analysis based on the setting of the `CustomCoefficientsDataType` property.
Locked	`'Same as input'`	When the input data type is `'double'` or `'fixed'`, the function assumes that the coefficient data type is signed, 16-bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Locked	`'Custom'`	The function performs fixed-point analysis based on

System Object State	Coefficient Data Type	Rule
		the setting of the `CustomCoefficientsDataType` property.

When you do not specify the arithmetic for non-CIC structures, the function uses double-precision arithmetic if the filter System object is in an unlocked state. If the System object is locked, the function performs analysis based on the locked input data type. CIC structures only support fixed-point arithmetic.

Examples

Design maximum-phase and minimum-phase lattice filters and verify their phase type.

```
k = [1/6 1/1.4];
bmax = latc2tf(k,'max');
bmin = latc2tf(k,'min');
max_flag = ismaxphase(bmax)
min_flag = isminphase(bmin)
```

For a filter defined with a set of single precision numerator and denominator coefficients, check if the filter is maximum phase for different tolerances.

```
b = single([1 -0.9999]);
a = single([1 0.45]);
max_flag1 = ismaxphase(b,a)
max_flag2 = ismaxphase(b,a,1e-3)
```

See Also

`isallpass` | `islinphase` | `isminphase` | `isstable`

Purpose

Determine whether filter is stable

Syntax

```
flag = isstable(b,a)
flag = isstable(sos)
flag = isstable(h)
flag = isstable(hs)
flag = isstable(hs,'Arithmetic',arithtype)
```

Description

`flag = isstable(b,a)` returns a logical output, `flag`, equal to `true` if the filter specified by numerator coefficients, `b`, and denominator coefficients, `a`, is a stable filter. If the poles lie on or outside the circle, `isstable` returns `false`. If the poles are inside the circle, `isstable` returns `true`.

`flag = isstable(sos)` returns `true` if the filter specified by second order sections matrix, `sos`, is stable. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`flag = isstable(h)` returns `true` if the filter object, `h`, is stable. If you have the DSP System Toolbox, you can use `isstable` with `adaptfilt` and `mfilt` objects.

`flag = isstable(hs)` returns `true` if the filter System object `hs` is stable. You must have the DSP System Toolbox software to use this syntax.

`flag = isstable(hs,'Arithmetic',arithtype)` analyzes the filter System object `hs` based on the specified *`arithtype`*. *`arithtype`* can be `'double'`, `'single'`, or `'fixed'`. When you specify `'double'` or `'single'`, the function performs double- or single-precision analysis. When you specify `'fixed'` , the arithmetic changes depending on the setting of the `CoefficientDataType` property and whether the System object is locked or unlocked. You must have the DSP System Toolbox software to use this syntax.

Details for Fixed-Point Arithmetic

System Object State	Coefficient Data Type	Rule
Unlocked	`'Same as input'`	The function assumes that the coefficient data type is signed, 16 bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Unlocked	`'Custom'`	The function performs fixed-point analysis based on the setting of the `CustomCoefficientsDataType` property.
Locked	`'Same as input'`	When the input data type is `'double'` or `'fixed'`, the function assumes that the coefficient data type is signed, 16-bit, and autoscaled. The function performs fixed-point analysis based on this assumption.
Locked	`'Custom'`	The function performs fixed-point analysis based on the setting of the `CustomCoefficientsDataType` property.

When you do not specify the arithmetic for non-CIC structures, the function uses double-precision arithmetic if the filter System object is in an unlocked state. If the System object is locked, the function performs analysis based on the locked input data type. CIC structures only support fixed-point arithmetic.

Examples

Design a Butterworth highpass IIR filter using second order sections and determine whether the filter is stable.

```
[z,p,k] = butter(6,0.7,'high');
SOS = zp2sos(z,p,k);
flag = isstable(SOS)
zplane(z,p)
```

Create a filter and determine the filter's stability for different coefficient data types and tolerances.

```
b = [1 -.5];
a = [1 -.999999999];
act_flag1 = isstable(b,a)
act_flag2 = isstable(single(b),single(a))
zplane(b,a)
```

See Also

`isallpass` | `islinphase` | `ismaxphase` | `isminphase` | `zplane`

is2rc

Purpose

Convert inverse sine parameters to reflection coefficients

Syntax

```
k = is2rc(isin)
```

Description

`k = is2rc(isin)` returns a vector of reflection coefficients `k` from a vector of inverse sine parameters `isin`.

Examples

```
isin = [0.2000 0.8727 0.0020 0.0052 -0.0052];
k = is2rc(isin);
```

References

[1] Deller, J.R., J.G. Proakis, and J.H.L. Hansen, *Discrete-Time Processing of Speech Signals*, Prentice-Hall, 1993.

See Also

`ac2rc` | `lar2rc` | `poly2rc` | `rc2is`

Purpose

Kaiser window

Syntax

```
w = kaiser(L,beta)
```

Description

`w = kaiser(L,beta)` returns an `L`-point Kaiser window in the column vector `w`. `beta` is the Kaiser window parameter that affects the sidelobe attenuation of the Fourier transform of the window. The default value for `beta` is 0.5.

To obtain a Kaiser window that designs an FIR filter with sidelobe attenuation of α dB, use the following ß.

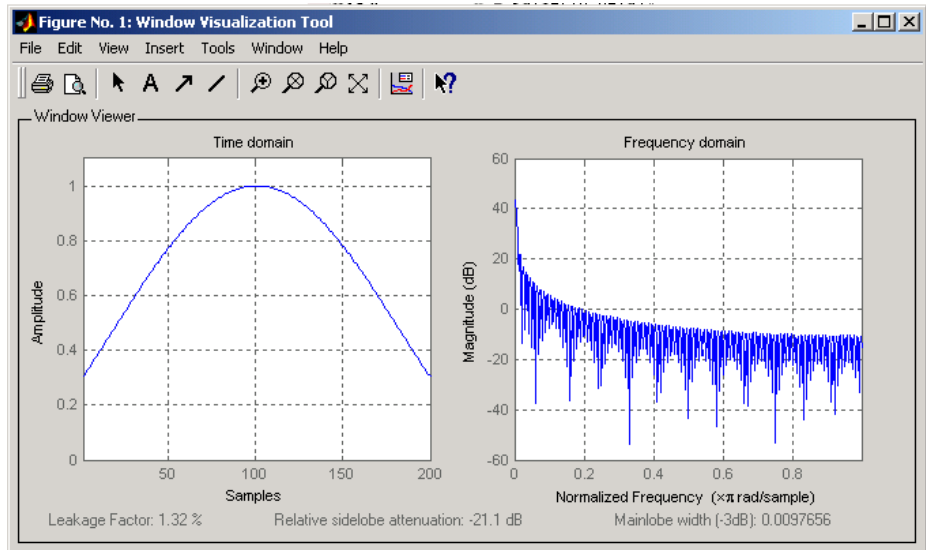
$$\beta = \begin{cases} 0.1102(\alpha - 8.7), & \alpha > 50 \\ 0.5842(\alpha - 21)^{0.4} + 0.07886(\alpha - 21), & 50 \geq \alpha \geq 21 \\ 0, & \alpha < 21 \end{cases}$$

Increasing `beta` widens the main lobe and decreases the amplitude of the sidelobes (i.e., increases the attenuation).

Examples

Create a 200-point Kaiser window with a beta of 2.5 and display the result using WVTool:

```
w = kaiser(200,2.5);
wvtool(w)
```



References

[1] Kaiser, J.F., "Nonrecursive Digital Filter Design Using the I0- sinh Window Function," Proc. 1974 *IEEE Symp. Circuits and Systems*, (April 1974), pp. 20-23.

[2] *Selected Papers in Digital Signal Processing II*, IEEE Press, New York, 1975.

[3] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, 1989, p. 453.

See Also

`chebwin` | `gausswin` | `kaiserord` | `tukeywin` | `window` | `wintool` | `wvtool`

Purpose

Kaiser window FIR filter design estimation parameters

Syntax

```
[n,Wn,beta,ftype] = kaiserord(f,a,dev)
[n,Wn,beta,ftype] = kaiserord(f,a,dev,fs)
c = kaiserord(f,a,dev,fs,'cell')
```

Description

`kaiserord` returns a filter order `n` and `beta` parameter to specify a Kaiser window for use with the `fir1` function. Given a set of specifications in the frequency domain, `kaiserord` estimates the minimum FIR filter order that will approximately meet the specifications. `kaiserord` converts the given filter specifications into passband and stopband ripples and converts cutoff frequencies into the form needed for windowed FIR filter design.

`[n,Wn,beta,ftype] = kaiserord(f,a,dev)` finds the approximate order `n`, normalized frequency band edges `Wn`, and weights that meet input specifications `f`, `a`, and `dev`. `f` is a vector of band edges and `a` is a vector specifying the desired amplitude on the bands defined by `f`. The length of `f` is twice the length of `a`, minus 2. Together, `f` and `a` define a desired piecewise constant response function. `dev` is a vector the same size as `a` that specifies the maximum allowable error or deviation between the frequency response of the output filter and its desired amplitude, for each band. The entries in `dev` specify the passband ripple and the stopband attenuation. You specify each entry in `dev` as a positive number, representing absolute filter gain (not in decibels).

Note If, in the vector `dev`, you specify unequal deviations across bands, the minimum specified deviation is used, since the Kaiser window method is constrained to produce filters with minimum deviation in all of the bands.

`fir1` can use the resulting order `n`, frequency vector `Wn`, multiband magnitude type `ftype`, and the Kaiser window parameter `beta`. The `ftype` string is intended for use with `fir1`; it is equal to `'high'` for a highpass filter and `'stop'` for a bandstop filter. For multiband filters,

it can be equal to `'dc-0'` when the first band is a stopband (starting at $f = 0$) or `'dc-1'` when the first band is a passband.

To design an FIR filter `b` that approximately meets the specifications given by `kaiser` parameters `f`, `a`, and `dev`, use the following command.

```
b = fir1(n,Wn,kaiser(n+1,beta),ftype,'noscale')
```

`[n,Wn,beta,ftype] = kaiserord(f,a,dev,fs)` uses a sampling frequency `fs` in Hz. If you don't specify the argument `fs`, or if you specify it as the empty vector `[]`, it defaults to 2 Hz, and the Nyquist frequency is 1 Hz. You can use this syntax to specify band edges scaled to a particular application's sampling frequency. The frequency band edges in `f` must be from 0 to `fs`/2.

`c = kaiserord(f,a,dev,fs,'cell')` is a cell-array whose elements are the parameters to `fir1`.

Note In some cases, `kaiserord` underestimates or overestimates the order `n`. If the filter does not meet the specifications, try a higher order such as `n+1`, `n+2`, and so on, or a try lower order.

Results are inaccurate if the cutoff frequencies are near 0 or the Nyquist frequency, or if `dev` is large (greater than 10%).

Tips

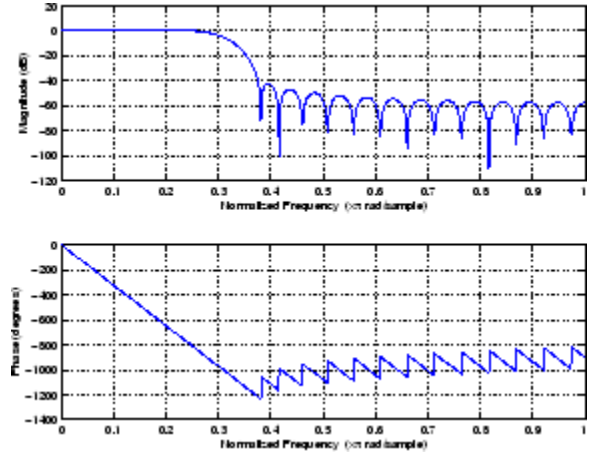
Be careful to distinguish between the meanings of filter length and filter order. The filter *length* is the number of impulse response samples in the FIR filter. Generally, the impulse response is indexed from $n = 0$ to $n = L–1$ where L is the filter length. The filter *order* is the highest power in a z-transform representation of the filter. For an FIR transfer function, this representation is a polynomial in z, where the highest power is z^{L-1} and the lowest power is z^0. The filter order is one less than the length ($L–1$) and is also equal to the number of zeros of the z polynomial.

Examples

Example 1

Design a lowpass filter with passband defined from 0 to 1 kHz and stopband defined from 1500 Hz to 4 kHz. Specify a passband ripple of 5% and a stopband attenuation of 40 dB:

```
fsamp = 8000;
fcuts = [1000 1500];
mags = [1 0];
devs = [0.05 0.01];
[n,Wn,beta,ftype] = kaiserord(fcuts,mags,devs,fsamp);
hh = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');
freqz(hh)
```

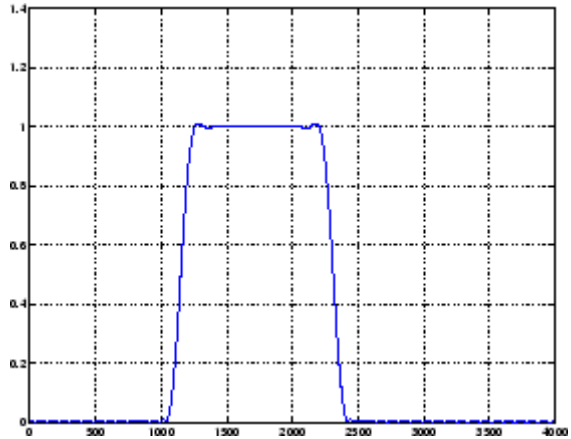


Example 2

Design an odd-length bandpass filter (note that odd length means even order, so the input to `fir1` must be an even integer):

```
fsamp = 8000;
fcuts = [1000 1300 2210 2410];
mags = [0 1 0];
devs = [0.01 0.05 0.01];
[n,Wn,beta,ftype] = kaiserord(fcuts,mags,devs,fsamp);
```

```
n = n + rem(n,2);
hh = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');
[H,f] = freqz(hh,1,1024,fsamp);
plot(f,abs(H)), grid on
```



Example 3

Design a lowpass filter with a passband cutoff of 1500 Hz, a stopband cutoff of 2000 Hz, passband ripple of 0.01, stopband ripple of 0.1, and a sampling frequency of 8000 Hz:

```
[n,Wn,beta,ftype] = kaiserord([1500 2000],[1 0],...
                                [0.01 0.1],8000);
b = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');
```

This is equivalent to

```
c = kaiserord([1500 2000],[1 0],[0.01 0.1],8000,'cell');
b = fir1(c{:});
```

Algorithms

`kaiserord` uses empirically derived formulas for estimating the orders of lowpass filters, as well as differentiators and Hilbert transformers. Estimates for multiband filters (such as bandpass filters) are derived from the lowpass design formulas.

The design formulas that underlie the Kaiser window and its application to FIR filter design are

$$\beta = \begin{cases} 0.1102(\alpha - 8.7), & \alpha > 50 \\ 0.5842(\alpha - 21)^{0.4} + 0.07886(\alpha - 21), & 50 \geq \alpha \geq 21 \\ 0, & \alpha < 21 \end{cases}$$

where $\alpha = -20\log_{10}\delta$ is the stopband attenuation expressed in decibels (recall that $\delta_p = \delta_s$ is required).

The design formula is

$$n = \frac{\alpha - 7.95}{2.285(\Delta\omega)}$$

where n is the filter order and $\Delta\omega$ is the width of the smallest transition region.

References

[1] Kaiser, J.F., "Nonrecursive Digital Filter Design Using the - sinh Window Function," Proc. 1974 IEEE Symp. *Circuits and Systems*, (April 1974), pp. 20-23.

[2] *Selected Papers in Digital Signal Processing II*, IEEE Press, New York, 1975, pp. 123-126.

[3] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, 1989, pp. 458-562.

See Also

`fir1` | `kaiser` | `firpmord`

kaiserwin

Purpose

Kaiser window filter from specification object

Syntax

```
h = design(d,'kaiserwin')
h = design(d,'kaiserwin',designoption,value,designoption,...
value,...)
```

Description

`h = design(d,'kaiserwin')` designs a digital filter `hd`, or a multirate filter `hm` that uses a Kaiser window. For `kaiserwin` to work properly, the filter order in the specifications 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.

`h = design(d,'kaiserwin',designoption,value,designoption,... value,...)` returns a filter where you specify design options as input arguments and the design process uses the Kaiser window technique.

To determine the available design options, use `designmethods` with the specification object and the design method as input arguments as shown.

```
designopts(d,'method')
```

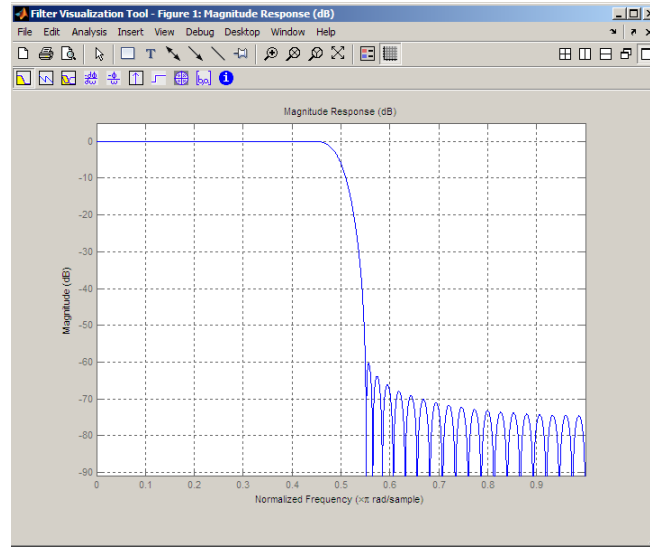
For complete help about using `kaiserwin`, refer to the command line help system. For example, to get specific information about using `kaiserwin` with `d`, the specification object, enter the following at the MATLAB prompt.

```
help(d,'kaiserwin')
```

Examples

This example designs a direct form FIR filter from a lowpass filter specification object.

```
d=fdesign.lowpass;
Hd=design(d,'kaiserwin');
fvtool(Hd)
```



See Also

`design` | `fdesign`

lar2rc

Purpose

Convert log area ratio parameters to reflection coefficients

Syntax

```
k = lar2rc(g)
```

Description

`k = lar2rc(g)` returns a vector of reflection coefficients `k` from a vector of log area ratio parameters `g`.

Examples

```
g = [0.6389    4.5989    0.0063    0.0163   -0.0163];
k = lar2rc(g);
```

References

[1] Deller, J.R., J.G. Proakis, and J.H.L. Hansen, *Discrete-Time Processing of Speech Signals*, Prentice-Hall, 1993.

See Also

`ac2rc` | `is2rc` | `poly2rc` | `rc2lar`

Purpose

Convert lattice filter parameters to transfer function form

Syntax

```
[num,den] = latc2tf(k,v)
[num,den] = latc2tf(k,'iiroption')
num = latc2tf(k,'firoption')
```

Description

`[num,den] = latc2tf(k,v)` finds the transfer function numerator `num` and denominator `den` from the IIR lattice coefficients `k` and ladder coefficients `v`.

`[num,den] = latc2tf(k,'iiroption')` produces an IIR filter transfer function according to the value of the string `'iiroption'`:

- `'allpole'`: Produces an all-pole filter transfer function from the associated all-pole IIR lattice filter coefficients `k`.
- `'allpass'`: Produces an allpass filter transfer function from the associated allpass IIR lattice filter coefficients `k`.

`num = latc2tf(k,'firoption')` produces an FIR filter according to the value of the string `'firoption'`:

- `'min'`: Produces a minimum-phase FIR filter numerator from the associated minimum-phase FIR lattice filter coefficients `k`.
- `'max'`: Produces a maximum-phase FIR filter numerator from the associated maximum-phase FIR lattice filter coefficients `k`.
- `'FIR'`: Produces a general FIR filter numerator from the lattice filter coefficients `k` (default, if you leave off the string altogether).

See Also

`latcfilt` | `tf2latc`

latcfilt

Purpose

Lattice and lattice-ladder filter implementation

Syntax

```
[f,g] = latcfilt(k,x)
[f,g] = latcfilt(k,v,x)
[f,g] = latcfilt(k,1,x)
[f,g,zf] = latcfilt(...,'ic',zi)
[f,g,zf] = latcfilt(...,dim)
```

Description

When filtering data, lattice coefficients can be used to represent

- FIR filters
- All-pole IIR filters
- Allpass IIR filters
- General IIR filters

`[f,g] = latcfilt(k,x)` filters `x` with the FIR lattice coefficients in the vector `k`. The forward lattice filter result is `f` and `g` is the backward filter result. If $|k| \le 1$, `f` corresponds to the minimum-phase output, and `g` corresponds to the maximum-phase output.

If `k` and `x` are vectors, the result is a (signal) vector. Matrix arguments are permitted under the following rules:

- If `x` is a matrix and `k` is a vector, each column of `x` is processed through the lattice filter specified by `k`.
- If `x` is a vector and `k` is a matrix, each column of `k` is used to filter `x`, and a signal matrix is returned.
- If `x` and `k` are both matrices with the same number of columns, then the *i*th column of `k` is used to filter the *i*th column of `x`. A signal matrix is returned.

`[f,g] = latcfilt(k,v,x)` filters `x` with the IIR lattice coefficients `k` and ladder coefficients `v`. Both `k` and `v` must be vectors, while `x` can be a signal matrix.

`[f,g] = latcfilt(k,1,x)` filters `x` with the IIR lattice specified by `k`, where `k` and `x` can be vectors or matrices. `f` is the all-pole lattice filter result and `g` is the allpass filter result.

`[f,g,zf] = latcfilt(...,'ic',zi)` accepts a length-`k` vector `zi` specifying the initial condition of the lattice states. Output `zf` is a length-`k` vector specifying the final condition of the lattice states.

`[f,g,zf] = latcfilt(...,dim)` filters `x` along the dimension `dim`. To specify a `dim` value, the FIR lattice coefficients `k` must be a vector and you must specify all previous input parameters in order. Use the empty vector [ ] for any parameters you do not want to specify. `zf` returns the final conditions in columns, regardless of the shape of `x`.

Examples

Filter data with an FIR lattice filter:

```
%create data
x=randn(512,1);
%reflection coefficients for 3-point MA filter
[f,g]=latcfilt([1/2 1],x);
%compare f vector to dfilt.latticemamin output
Hd=dfilt.latticemamin([1/2 1]);
y=filter(Hd,x);
isequal(y,f) %returns 1
%compare g vector to dfilt.latticemamax output
Hd1=dfilt.latticemamax([1/2 1]);
y1=filter(Hd1,x);
isequal(g,y1) %returns 1
```

See Also

`dfilt.latticemamax` | `dfilt.latticemamin` | `filter`

levinson

Purpose

Levinson-Durbin recursion

Syntax

```
a = levinson(r)
a = levinson(r,n)
[a,e] = levinson(r,n)
[a,e,k] = levinson(r,n)
```

Description

The Levinson-Durbin recursion is an algorithm for finding an all-pole IIR filter with a prescribed deterministic autocorrelation sequence. It has applications in filter design, coding, and spectral estimation. The filter that `levinson` produces is minimum phase.

`a = levinson(r)` finds the coefficients of a `length(r)-1` order autoregressive linear process which has `r` as its autocorrelation sequence. `r` is a real or complex deterministic autocorrelation sequence. If `r` is a matrix, `levinson` finds the coefficients for each column of `r` and returns them in the rows of `a`. `n=length(r)-1` is the default order of the denominator polynomial $A(z)$; that is, `a = [1 a(2) ... a(n+1)]`. The filter coefficients are ordered in descending powers of z^{-1}.

$$H(z) = \frac{1}{A(z)} = \frac{1}{1 + a(2)z^{-1} + \cdots + a(n+1)z^{-n}}$$

`a = levinson(r,n)` returns the coefficients for an autoregressive model of order n.

`[a,e] = levinson(r,n)` returns the prediction error, e, of order n.

`[a,e,k] = levinson(r,n)` returns the reflection coefficients `k` as a column vector of length `n`.

Note `k` is computed internally while computing the `a` coefficients, so returning `k` simultaneously is more efficient than converting `a` to `k` with `tf2latc`.

Algorithms

`levinson` solves the symmetric Toeplitz system of linear equations

$$\begin{bmatrix} r(1) & r(2)^* & \cdots & r(n)^* \\ r(2) & r(1) & \cdots & r(n-1)^* \\ \vdots & \ddots & \ddots & \vdots \\ r(n) & \cdots & r(2) & r(1) \end{bmatrix} \begin{bmatrix} a(2) \\ a(3) \\ \vdots \\ a(n+1) \end{bmatrix} = \begin{bmatrix} -r(2) \\ -r(3) \\ \vdots \\ -r(n+1) \end{bmatrix}$$

where `r = [`$r(1)$ ... $r(n+1)$`]` is the input autocorrelation vector, and $r(i)^*$ denotes the complex conjugate of $r(i)$. The input `r` is typically a vector of autocorrelation coefficients where lag 0 is the first element $r(1)$. The algorithm requires $O(n^2)$ flops and is thus much more efficient than the MATLAB `\` command for large `n`. However, the `levinson` function uses `\` for low orders to provide the fastest possible execution.

References

[1] Ljung, L., *System Identification: Theory for the User*, Prentice-Hall, 1987, pp. 278-280.

See Also

`lpc` | `prony` | `rlevinson` | `schurrc` | `stmcb`

lp2bp

Purpose

Transform lowpass analog filters to bandpass

Syntax

```
[bt,at] = lp2bp(b,a,Wo,Bw)
[At,Bt,Ct,Dt] = lp2bp(A,B,C,D,Wo,Bw)
```

Description

`lp2bp` transforms analog lowpass filter prototypes with a cutoff angular frequency of 1 rad/s into bandpass filters with desired bandwidth and center frequency. The transformation is one step in the digital filter design process for the `butter`, `cheby1`, `cheby2`, and `ellip` functions.

`lp2bp` can perform the transformation on two different linear system representations: transfer function form and state-space form. In both cases, the input system must be an analog filter prototype.

Transfer Function Form (Polynomial)

`[bt,at] = lp2bp(b,a,Wo,Bw)` transforms an analog lowpass filter prototype given by polynomial coefficients into a bandpass filter with center frequency `Wo` and bandwidth `Bw`. Row vectors `b` and `a` specify the coefficients of the numerator and denominator of the prototype in descending powers of *s*.

$$\frac{B(s)}{A(s)} = \frac{b(1)s^n + \cdots + b(n)s + b(n+1)}{a(1)s^m + \cdots + a(m)s + a(m+1)}$$

Scalars `Wo` and `Bw` specify the center frequency and bandwidth in units of rad/s. For a filter with lower band edge `w1` and upper band edge `w2`, use `Wo = sqrt(w1*w2)` and `Bw = w2-w1`.

`lp2bp` returns the frequency transformed filter in row vectors `bt` and `at`.

State-Space Form

`[At,Bt,Ct,Dt] = lp2bp(A,B,C,D,Wo,Bw)` converts the continuous-time state-space lowpass filter prototype in matrices `A`, `B`, `C`, `D` shown below

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

into a bandpass filter with center frequency `Wo` and bandwidth `Bw`. For a filter with lower band edge `w1` and upper band edge `w2`, use `Wo = sqrt(w1*w2)` and `Bw = w2-w1`.

The bandpass filter is returned in matrices `At`, `Bt`, `Ct`, `Dt`.

Algorithms

`lp2bp` is a highly accurate state-space formulation of the classic analog filter frequency transformation. Consider the state-space system

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

where u is the input, x is the state vector, and y is the output. The Laplace transform of the first equation (assuming zero initial conditions) is

$$sX(s) = AX(s) + BU(s)$$

Now if a bandpass filter is to have center frequency ω_0 and bandwidth B_w, the standard s-domain transformation is

$$s = Q(p^2 + 1) / p$$

where $Q = \omega_0/B_w$ and $p = s/\omega_0$. Substituting this for s in the Laplace transformed state-space equation, and considering the operator p as d/dt results in

$$Q\ddot{x} + Qx = \dot{A}x + B\dot{u}$$

or

$$Q\ddot{x} - \dot{A}x - B\dot{u} = -Qx$$

Now define

$$Q\dot{\omega} = -Qx$$

which, when substituted, leads to

$$Q\dot{x} = Ax + Q\omega + Bu$$

The last two equations give equations of state. Write them in standard form and multiply the differential equations by ω_0 to recover the time/frequency scaling represented by p and find state matrices for the bandpass filter:

```
Q = Wo/Bw; [ma,m] = size(A);
At = Wo*[A/Q eye(ma,m);-eye(ma,m) zeros(ma,m)];
Bt = Wo*[B/Q; zeros(ma,n)];
Ct = [C zeros(mc,ma)];
Dt = d;
```

If the input to `lp2bp` is in transfer function form, the function transforms it into state-space form before applying this algorithm.

See Also

`bilinear` | `impinvar` | `lp2bs` | `lp2hp` | `lp2lp`

Purpose

Transform lowpass analog filters to bandstop

Syntax

```
[bt,at] = lp2bs(b,a,Wo,Bw)
[At,Bt,Ct,Dt] = lp2bs(A,B,C,D,Wo,Bw)
```

Description

`lp2bs` transforms analog lowpass filter prototypes with a cutoff angular frequency of 1 rad/s into bandstop filters with desired bandwidth and center frequency. The transformation is one step in the digital filter design process for the `butter`, `cheby1`, `cheby2`, and `ellip` functions.

`lp2bs` can perform the transformation on two different linear system representations: transfer function form and state-space form. In both cases, the input system must be an analog filter prototype.

Transfer Function Form (Polynomial)

`[bt,at] = lp2bs(b,a,Wo,Bw)` transforms an analog lowpass filter prototype given by polynomial coefficients into a bandstop filter with center frequency `Wo` and bandwidth `Bw`. Row vectors `b` and `a` specify the coefficients of the numerator and denominator of the prototype in descending powers of *s*.

$$\frac{B(s)}{A(s)} = \frac{b(1)s^n + \cdots + b(n)s + b(n+1)}{a(1)s^m + \cdots + a(m)s + a(m+1)}$$

Scalars `Wo` and `Bw` specify the center frequency and bandwidth in units of radians/second. For a filter with lower band edge `w1` and upper band edge `w2`, use `Wo = sqrt(w1*w2)` and `Bw = w2-w1`.

`lp2bs` returns the frequency transformed filter in row vectors `bt` and `at`.

State-Space Form

`[At,Bt,Ct,Dt] = lp2bs(A,B,C,D,Wo,Bw)` converts the continuous-time state-space lowpass filter prototype in matrices `A`, `B`, `C`, `D` shown below

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

into a bandstop filter with center frequency `Wo` and bandwidth `Bw`. For a filter with lower band edge `w1` and upper band edge `w2`, use `Wo = sqrt(w1*w2)` and `Bw = w2-w1`.

The bandstop filter is returned in matrices `At`, `Bt`, `Ct`, `Dt`.

Algorithms

`lp2bs` is a highly accurate state-space formulation of the classic analog filter frequency transformation. If a bandstop filter is to have center frequency ω_0 and bandwidth B_w, the standard s-domain transformation is

$$s = \frac{p}{Q(p^2 + 1)}$$

where $Q = \omega_0/B_w$ and $p = s/\omega_0$. The state-space version of this transformation is

```
Q = Wo/Bw;
At = [Wo/Q*inv(A) Wo*eye(ma);-Wo*eye(ma) zeros(ma)];
Bt = -[Wo/Q*(A\B); zeros(ma,n)];
Ct = [C/A zeros(mc,ma)];
Dt = D - C/A*B;
```

See `lp2bp` for a derivation of the bandpass version of this transformation.

See Also

`bilinear` | `impinvar` | `lp2bp` | `lp2hp` | `lp2lp`

Purpose

Transform lowpass analog filters to highpass

Syntax

```
[bt,at] = lp2hp(b,a,Wo)
[At,Bt,Ct,Dt] = lp2hp(A,B,C,D,Wo)
```

Description

`lp2hp` transforms analog lowpass filter prototypes with a cutoff angular frequency of 1 rad/s into highpass filters with desired cutoff angular frequency. The transformation is one step in the digital filter design process for the `butter`, `cheby1`, `cheby2`, and `ellip` functions.

The `lp2hp` function can perform the transformation on two different linear system representations: transfer function form and state-space form. In both cases, the input system must be an analog filter prototype.

Transfer Function Form (Polynomial)

`[bt,at] = lp2hp(b,a,Wo)` transforms an analog lowpass filter prototype given by polynomial coefficients into a highpass filter with cutoff angular frequency `Wo`. Row vectors `b` and `a` specify the coefficients of the numerator and denominator of the prototype in descending powers of s.

$$\frac{B(s)}{A(s)} = \frac{b(1)s^n + \cdots + b(n)s + b(n+1)}{a(1)s^m + \cdots + a(m)s + a(m+1)}$$

Scalar `Wo` specifies the cutoff angular frequency in units of radians/second. The frequency transformed filter is returned in row vectors `bt` and `at`.

State-Space Form

`[At,Bt,Ct,Dt] = lp2hp(A,B,C,D,Wo)` converts the continuous-time state-space lowpass filter prototype in matrices `A`, `B`, `C`, `D` below

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

into a highpass filter with cutoff angular frequency `Wo`. The highpass filter is returned in matrices `At`, `Bt`, `Ct`, `Dt`.

Algorithms

`lp2hp` is a highly accurate state-space formulation of the classic analog filter frequency transformation. If a highpass filter is to have cutoff angular frequency ω_0, the standard s-domain transformation is

$$s = \frac{\omega_0}{p}$$

The state-space version of this transformation is

```
At = Wo*inv(A);
Bt = -Wo*(A\B);
Ct = C/A;
Dt = D - C/A*B;
```

See `lp2bp` for a derivation of the bandpass version of this transformation.

See Also

`bilinear` | `impinvar` | `lp2bp` | `lp2bs` | `lp2lp`

Purpose

Change cutoff frequency for lowpass analog filter

Syntax

```
[bt,at] = lp2lp(b,a,Wo)
[At,Bt,Ct,Dt] = lp2lp(A,B,C,D,Wo)
```

Description

`lp2lp` transforms an analog lowpass filter prototype with a cutoff angular frequency of 1 rad/s into a lowpass filter with any specified cutoff angular frequency. The transformation is one step in the digital filter design process for the `butter`, `cheby1`, `cheby2`, and `ellip` functions.

The `lp2lp` function can perform the transformation on two different linear system representations: transfer function form and state-space form. In both cases, the input system must be an analog filter prototype.

Transfer Function Form (Polynomial)

`[bt,at] = lp2lp(b,a,Wo)` transforms an analog lowpass filter prototype given by polynomial coefficients into a lowpass filter with cutoff angular frequency `Wo`. Row vectors `b` and `a` specify the coefficients of the numerator and denominator of the prototype in descending powers of *s*.

$$\frac{B(s)}{A(s)} = \frac{b(1)s^n + \cdots + b(n)s + b(n+1)}{a(1)s^m + \cdots + a(m)s + a(m+1)}$$

Scalar `Wo` specifies the cutoff angular frequency in units of radians/second. `lp2lp` returns the frequency transformed filter in row vectors `bt` and `at`.

State-Space Form

`[At,Bt,Ct,Dt] = lp2lp(A,B,C,D,Wo)` converts the continuous-time state-space lowpass filter prototype in matrices `A`, `B`, `C`, `D` below

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

into a lowpass filter with cutoff angular frequency `Wo`. `lp2lp` returns the lowpass filter in matrices `At`, `Bt`, `Ct`, `Dt`.

Algorithms

`lp2lp` is a highly accurate state-space formulation of the classic analog filter frequency transformation. If a lowpass filter is to have cutoff angular frequency ω_0, the standard s-domain transformation is

$$s = p / \omega_0$$

The state-space version of this transformation is

```
At = Wo*A;
Bt = Wo*B;
Ct = C;
Dt = D;
```

See `lp2bp` for a derivation of the bandpass version of this transformation.

See Also

`bilinear` | `impinvar` | `lp2bp` | `lp2bs` | `lp2hp`

Purpose

Linear prediction filter coefficients

Syntax

```
[a,g] = lpc(x,p)
```

Description

`lpc` determines the coefficients of a forward linear predictor by minimizing the prediction error in the least squares sense. It has applications in filter design and speech coding.

`[a,g] = lpc(x,p)` finds the coefficients of a `p`th-order linear predictor (FIR filter) that predicts the current value of the real-valued time series `x` based on past samples.

$$\hat{x}(n) = -a(2)x(n-1) - a(3)x(n-2) - \cdots - a(p+1)x(n-p)$$

`p` is the order of the prediction filter polynomial, `a = [1 a(2) ... a(p+1)]`. If `p` is unspecified, `lpc` uses as a default `p = length(x)-1`. If `x` is a matrix containing a separate signal in each column, `lpc` returns a model estimate for each column in the rows of matrix `a` and a column vector of prediction error variances `g`. The length of `p` must be less than or equal to the length of `x`.

Examples

Estimate a data series using a third-order forward predictor, and compare to the original signal.

First, create the signal data as the output of an autoregressive process driven by white noise. Use the last 4096 samples of the AR process output to avoid start-up transients:

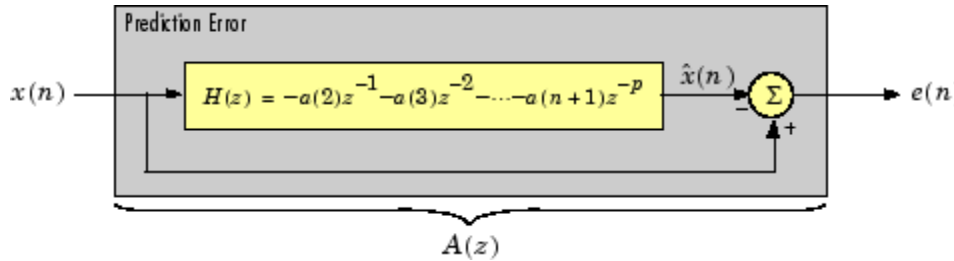
```
noise = randn(50000,1);  % Normalized white Gaussian noise
x = filter(1,[1 1/2 1/3 1/4],noise);
x = x(45904:50000);
```

Compute the predictor coefficients, estimated signal, prediction error, and autocorrelation sequence of the prediction error:

```
a = lpc(x,3);
est_x = filter([0 -a(2:end)],1,x);    % Estimated signal
e = x - est_x;                        % Prediction error
```

```
[acs,lags] = xcorr(e,'coeff');   % ACS of prediction error
```

The prediction error, $e(n)$, can be viewed as the output of the prediction error filter $A(z)$ shown below, where $H(z)$ is the optimal linear predictor, $x(n)$ is the input signal, and $\hat{x}(n)$ is the predicted signal.



Compare the predicted signal to the original signal.

```
plot(1:97,x(4001:4097),1:97,est_x(4001:4097),'--');
title('Original Signal vs. LPC Estimate');
xlabel('Sample Number'); ylabel('Amplitude'); grid;
legend('Original Signal','LPC Estimate')
```

Look at the autocorrelation of the prediction error:

```
plot(lags,acs);
title('Autocorrelation of the Prediction Error');
xlabel('Lags'); ylabel('Normalized Value'); grid;
```

The prediction error is approximately white Gaussian noise, as expected for a third-order AR input process.

Algorithms

`lpc` uses the autocorrelation method of autoregressive (AR) modeling to find the filter coefficients. The generated filter might not model the process exactly even if the data sequence is truly an AR process of the correct order. This is because the autocorrelation method implicitly windows the data, that is, it assumes that signal samples beyond the length of `x` are 0.

`lpc` computes the least squares solution to

$$Xa = b$$

where

$$X = \begin{bmatrix} x(1) & 0 & \cdots & 0 \\ x(2) & x(1) & \ddots & \vdots \\ \vdots & x(2) & \ddots & 0 \\ x(m) & \vdots & \ddots & x(1) \\ 0 & x(m) & \ddots & x(2) \\ \vdots & \ddots & \ddots & \vdots \\ 0 & \cdots & 0 & x(m) \end{bmatrix}, \quad a = \begin{bmatrix} 1 \\ a(2) \\ \vdots \\ a(p+1) \end{bmatrix}, \quad b = \begin{bmatrix} 1 \\ 0 \\ \vdots \\ 0 \end{bmatrix}$$

and `m` is the length of `x`. Solving the least squares problem via the normal equations

$$X^H Xa = X^H b$$

leads to the Yule-Walker equations

$$\begin{bmatrix} r(1) & r(2)^* & \cdots & r(p)^* \\ r(2) & r(1) & \ddots & \vdots \\ \vdots & \ddots & \ddots & r(2)^* \\ r(p) & \cdots & r(2) & r(1) \end{bmatrix} \begin{bmatrix} a(2) \\ a(3) \\ \vdots \\ a(p+1) \end{bmatrix} = \begin{bmatrix} -r(2) \\ -r(3) \\ \vdots \\ -r(p+1) \end{bmatrix}$$

where r = `[`$r(1)\ r(2)\ \ldots\ r(p+1)$`]` is an autocorrelation estimate for `x` computed using `xcorr`. The Yule-Walker equations are solved in $O(p^2)$ flops by the Levinson-Durbin algorithm (see `levinson`).

References

[1] Jackson, L.B., *Digital Filters and Signal Processing, Second Edition*, Kluwer Academic Publishers, 1989. pp. 255-257.

See Also

`aryule` | `levinson` | `prony` | `pyulear` | `stmcb`

lsf2poly

Purpose

Convert line spectral frequencies to prediction filter coefficients

Syntax

```
a = lsf2poly(lsf)
```

Description

`a = lsf2poly(lsf)` returns a vector `a` containing the prediction filter coefficients from the vector `lsf` of line spectral frequencies. If `lsf` is a matrix of size MxN with separate channels of line spectral frequencies in each column, the returned `a` matrix has the resulting prediction filter coefficients in rows and is of size Nx(M+1).

Examples

```
lsf = [0.7842    1.5605    1.8776    1.8984    2.3593];
a = lsf2poly(lsf)
a =
    1.0000    0.6148    0.9899    0.0001    0.0031   -0.0081
```

References

[1] Deller, J.R., J.G. Proakis, and J.H.L. Hansen, *Discrete-Time Processing of Speech Signals*, Prentice-Hall, 1993.

[2] Rabiner, L.R., and R.W. Schafer, *Digital Processing of Speech Signals*, Prentice-Hall, 1978.

See Also

`ac2poly` | `poly2lsf` | `rc2poly`

Purpose

Convert magnitude to decibels (dB)

Syntax

```
ydb = mag2db(y)
```

Description

`ydb = mag2db(y)` returns the corresponding decibel (dB) value `ydb` for a given magnitude `y`. The relationship between magnitude and decibels is `ydb` = $20*\log_{10}(y)$.

See Also

`db2mag`

Purpose

Generalized Marcum Q function

Syntax

```
Q = marcumq(a,b)
Q = marcumq(a,b,m)
```

Description

`Q = marcumq(a,b)` computes the Marcum Q function of `a` and `b`, defined by

$$Q(a,b) = \int_b^\infty x \exp\left(-\frac{(x^2+a^2)}{2}\right) I_0(ax)dx$$

where `a` and `b` are nonnegative real numbers. In this expression, I_0 is the modified Bessel function of the first kind of zero order.

`Q = marcumq(a,b,m)` computes the generalized Marcum Q, defined by

$$Q\ (a,b) = \frac{1}{a^{m-1}} \int_b^\infty x^m \exp\left(-\frac{(x^2+a^2)}{2}\right) I_{m-1}(ax)dx$$

where `a` and `b` are nonnegative real numbers, and `m` is a positive integer. In this expression, I_{m-1} is the modified Bessel function of the first kind of order m-1.

If any of the inputs is a scalar, it is expanded to the size of the other inputs.

Algorithms

`marcumq` uses the algorithm developed in [3]. The paper describes two error criteria: a relative error criterion and an absolute error criterion. `marcumq` utilizes the absolute error criterion.

References

[1] Cantrell, P. E., and A. K. Ojha, "Comparison of Generalized Q-Function Algorithms," *IEEE Transactions on Information Theory*, Vol. IT-33, July, 1987, pp. 591–596.

[2] Marcum, J. I., "A Statistical Theory of Target Detection by Pulsed Radar: Mathematical Appendix," RAND Corporation, Santa Monica, CA, Research Memorandum RM-753, July 1, 1948. Reprinted in *IRE Transactions on Information Theory*, Vol. IT-6, April, 1960, pp. 59–267.

[3] Shnidman, D. A., "The Calculation of the Probability of Detection and the Generalized Marcum Q-Function," *IEEE Transactions on Information Theory*, Vol. IT-35, March, 1989, pp. 389–400.

See Also

`besseli`

maxflat

Purpose

Generalized digital Butterworth filter design

Syntax

```
[b,a] = maxflat(n,m,Wn)
b = maxflat(n,'sym',Wn)
[b,a,b1,b2] = maxflat(n,m,Wn)
[b,a,b1,b2,sos,g] = maxflat(n,m,Wn)
[...] = maxflat(n,m,Wn,'design_flag')
```

Description

`[b,a] = maxflat(n,m,Wn)` is a lowpass Butterworth filter with numerator and denominator coefficients `b` and `a` of orders `n` and `m` respectively. `Wn` is the normalized cutoff frequency at which the magnitude response of the filter is equal to $1/\sqrt{2}$ (approx. -3 dB). `Wn` must be between 0 and 1, where 1 corresponds to the Nyquist frequency.

`b = maxflat(n,'sym',Wn)` is a symmetric FIR Butterworth filter. `n` must be even, and `Wn` is restricted to a subinterval of [0,1]. The function raises an error if `Wn` is specified outside of this subinterval.

`[b,a,b1,b2] = maxflat(n,m,Wn)` returns two polynomials `b1` and `b2` whose product is equal to the numerator polynomial `b` (that is, `b = conv(b1,b2)`). `b1` contains all the zeros at `z = -1`, and `b2` contains all the other zeros.

`[b,a,b1,b2,sos,g] = maxflat(n,m,Wn)` returns the second-order sections representation of the filter as the filter matrix `sos` and the gain `g`.

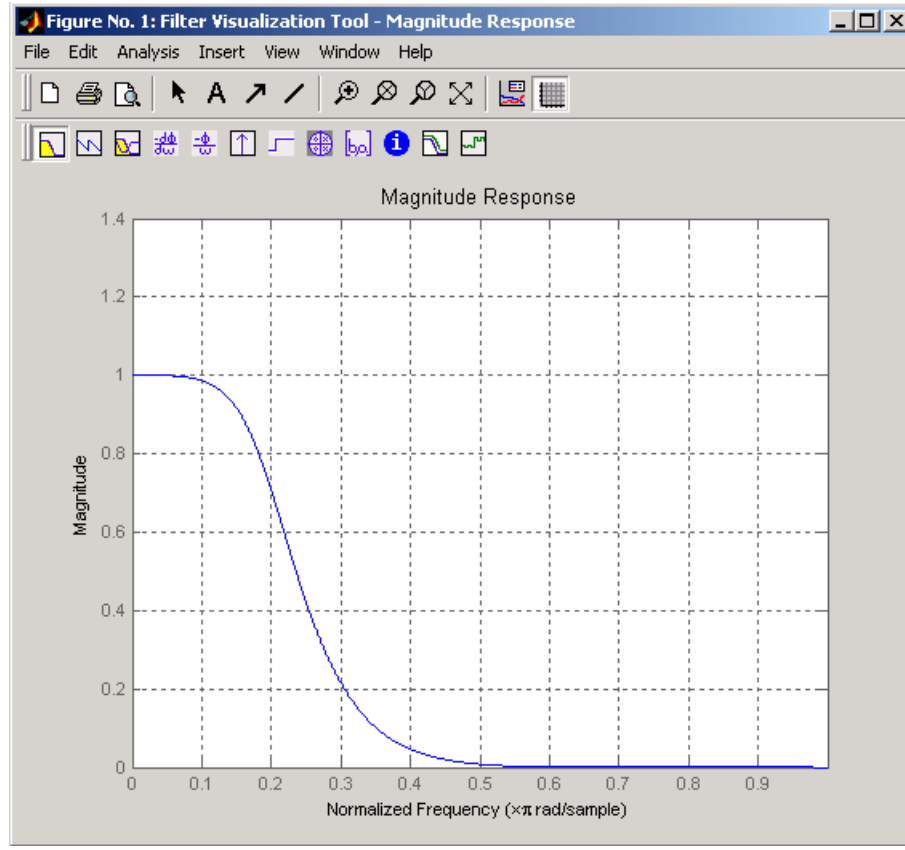
`[...] = maxflat(n,m,Wn,'design_flag')` enables you to monitor the filter design, where '*design_flag*' is

- `'trace'` for a textual display of the design table used in the design
- `'plots'` for plots of the filter's magnitude, group delay, and zeros and poles
- `'both'` for both the textual display and plots

Examples

```
n = 10; m = 2; Wn = 0.2;
[b,a] = maxflat(n,m,Wn)
```

```
fvtool(b,a)                    % Display the magnitude plot
```



Algorithms

The method consists of the use of formulae, polynomial root finding, and a transformation of polynomial roots.

References

[1] Selesnick, I.W., and C.S. Burrus, "Generalized Digital Butterworth Filter Design," *Proceedings of the IEEE Int. Conf. Acoust., Speech, Signal Processing, Vol. 3* (May 1996).

See Also

butter | filter | freqz

Purpose

1-D median filtering

Syntax

```
y = medfilt1(x,n)
y = medfilt1(x,n,blksz)
y = medfilt1(x,n,blksz,dim)
```

Description

`y = medfilt1(x,n)` applies an order `n` one-dimensional median filter to vector `x`; the function considers the signal to be 0 beyond the end points. Output `y` has the same length as `x`.

For n odd, `y(k)` is the median of `x(k-(n-1)/2:k+(n-1)/2)`.

For `n` even, `y(k)` is the median of `x(k-n/2)`, `x(k-(n/2)+1)`, ..., `x(k+(n/2)-1)`. In this case, `medfilt1` sorts the numbers, then takes the average of the `n/2` and `(n/2)+1` elements.

The default for n is 3.

`y = medfilt1(x,n,blksz)` uses a `for`-loop to compute `blksz` (block size) output samples at a time. Use `blksz << length(x)` if you are low on memory, since `medfilt1` uses a working matrix of size n-by-`blksz`. By default, `blksz = length(x)`; this provides the fastest execution if you have sufficient memory.

If `x` is a matrix, `medfilt1` median filters its columns using

```
y(:,i) = medfilt1(x(:,i),n,blksz)
```

in a loop over the columns of `x`.

`y = medfilt1(x,n,blksz,dim)` specifies the dimension, `dim`, along which the filter operates.

References

[1] Pratt, W.K., *Digital Image Processing*, John Wiley & Sons, 1978, pp. 330-333.

See Also

`filter` | `medfilt2` | `median`

midcross

Purpose

Mid-reference level crossing for bilevel waveform

Syntax

```
C = midcross(X)
C = midcross(X,FS)
C = midcross(X,T)
[C,MIDLEV] = midcross(...)
C = midcross(X,Name,Value)
midcross(...)
```

Description

`C = midcross(X)` returns a vector, `C`, of time instants where each transition of the input signal, `X`, crosses the 50% reference level. The sample instants correspond to the indices of the input vector. Because `midcross` uses interpolation to determine the crossing instant, `C` may contain values that do not correspond to sampling instants. To determine the transitions, `midcross` estimates the state levels of `X` by a histogram method. `midcross` identifies all intervals which cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-708.

`C = midcross(X,FS)` specifies the sample rate, `FS`, in hertz as a positive scalar. The first sample instant corresponds to t=0. Because `midcross` uses interpolation to determine the crossing instant, `C` may contain values that do not correspond to sampling instants.

`C = midcross(X,T)` specifies the sample instants, `T`, as a vector with the same number of elements as `X`. Because `midcross` uses interpolation to determine the crossing instant, `C` may contain values that do not correspond to sampling instants.

`[C,MIDLEV] = midcross(...)` returns the waveform value corresponding to the mid-reference level.

`C = midcross(X,Name,Value)` returns the time instants corresponding to mid-reference level crossings with additional options specified by one or more `Name,Value` pair arguments.

`midcross(...)` plots the signal and marks the location of the mid-crossings (mid-reference level instants) and the associated reference levels. `midcross` also plots the state levels with upper and lower state boundaries.

Input Arguments

X

Bilevel waveform. `X` is a real-valued row or column vector.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

Specify optional comma-separated pairs of `Name,Value` arguments. `Name` is the argument name and `Value` is the corresponding value. `Name` must appear inside single quotes (`' '`). You can specify several name and value pair arguments in any order as `Name1,Value1,...,NameN,ValueN`.

'MidPct'

Mid-reference level as a percentage of the waveform amplitude.

Default: `50`

'StateLevels'

Low and high state levels. `StateLevels` is a 1-by-2 real-valued vector. The first element is the low state level. The second element is the high state level. If you do not specify low- and high-state levels, `midcross` estimates the state levels from the input waveform using the histogram method.

'Tolerance'

Tolerance levels (lower- and upper-state boundaries) expressed as a percentage. See "State-Level Tolerances" on page 1-708.

Default: 2

Output Arguments

C

Time instants of the mid-reference level crossings.

MIDLEV

Mid-reference level.

Definitions

Mid-Reference Level

The mid-reference level in a bilevel waveform with low-state level, *S_1*, and high–state level, *S_2*, is

$$S_1 + \frac{1}{2}(S_2 - S_1)$$

Mid Reference Level Instant

Let $y_{50\%}$ denote the mid–reference level.

Let $t_{50\%-}$ and $t_{50\%+}$ denote the two consecutive sampling instants corresponding to the waveform values nearest in value to $y_{50\%}$.

Let $y_{50\%-}$ and $y_{50\%+}$ denote the waveform values at $t_{50\%-}$ and $t_{50\%+}$.

The mid-reference level instant is

$$t_{50\%} = t_{50\%} + \left(\frac{t_{50\%_+} - t_{50\%_-}}{y_{50\%_+} - y_{50\%_-}}\right)(y_{50\%_+} - y_{50\%_-})$$

State-Level Tolerances

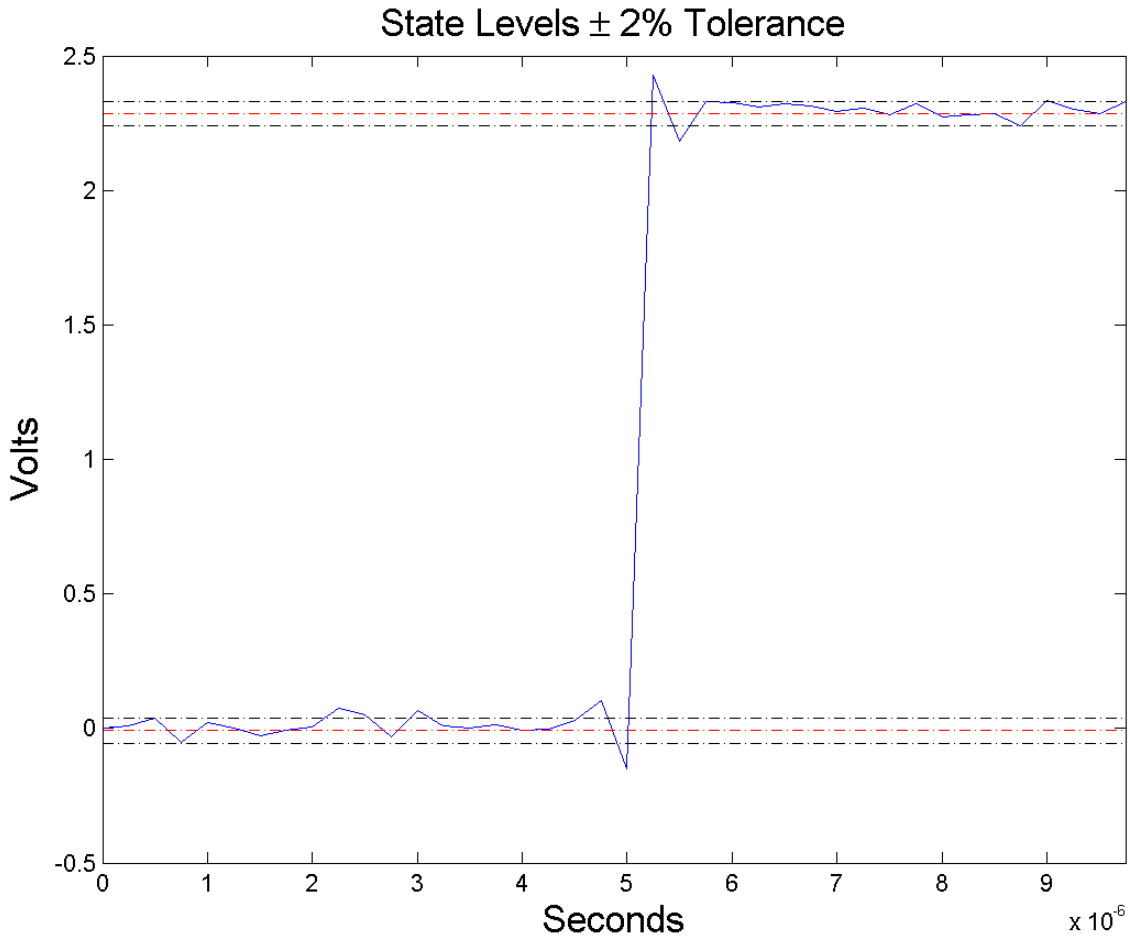
Each state level can have associated lower- and upper-state boundaries. These state boundaries are defined as the state level plus or minus a

scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The red dashed lines indicate the estimated state levels.

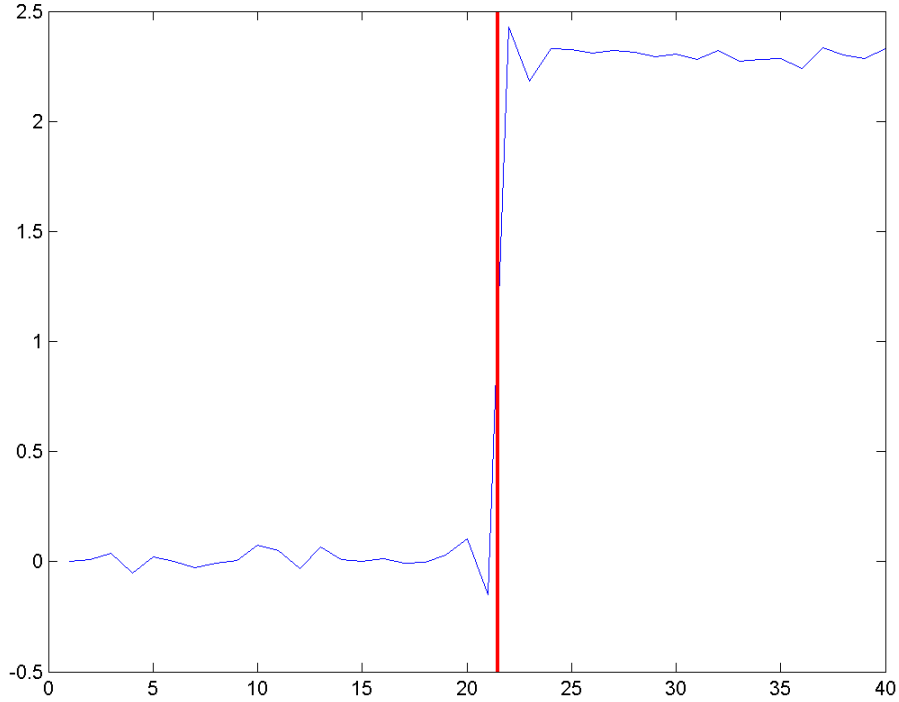


Examples

Mid-Reference Level Instant of Bilevel Waveform

Assuming a sampling interval of 1, compute the mid-reference level instant of a bilevel waveform and plot the result.

```
load('transitionex.mat', 'x');
C = midcross(x);
plot(x); hold on;
plot([C C],[-0.5 2.5],'r','linewidth',2);
```



The instant at which the waveform crosses the 50% reference level is 21.5. Note that this is not a sampling instant present in the input vector because `midcross` uses interpolation to identify the mid-reference level crossing.

Mid-Reference Level Instant with Sampling Frequency

Compute the mid-reference level instant using the sampling rate for a bilevel waveform sampled at 4 MHz.

```
load('transitionex.mat','x','t');
Fs = 1/(t(2)-t(1));
C = midcross(x,Fs);
```

Mid Reference Level Instant Using Sample Instants

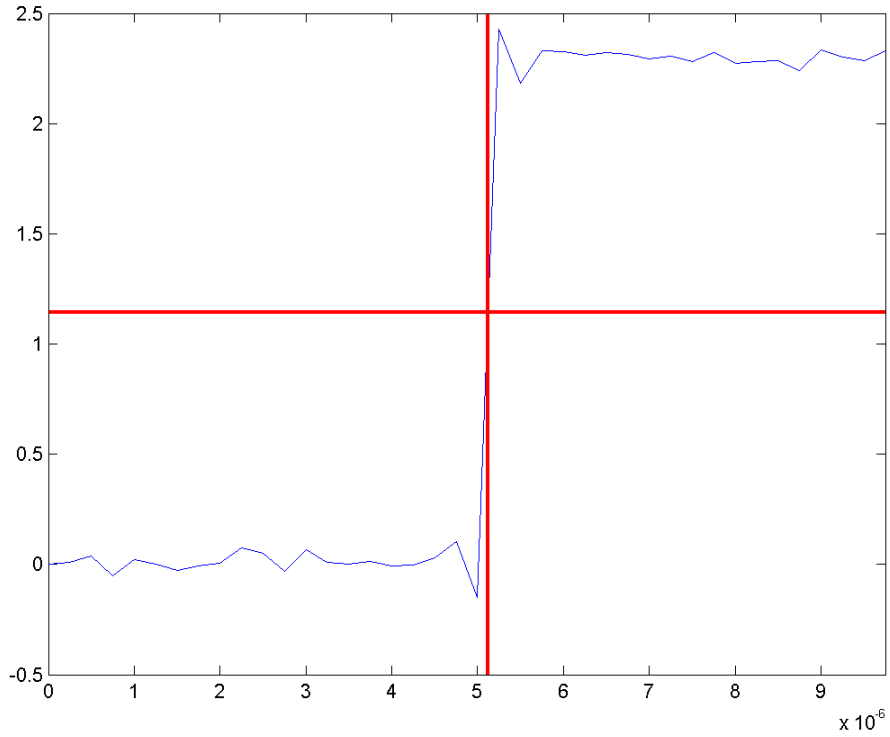
Compute the mid-reference level instants using a vector of sample times equal in length to the bilevel waveform. The sampling rate is 4 MHz.

```
load('transitionex.mat','x','t');
C = midcross(x,t);
```

Mid-Reference Level Value of Bilevel Waveform

Compute the level corresponding to the mid-reference level instant. Plot the result.

```
load('transitionex.mat','x','t');
[C,MIDLEV] = midcross(x,t);
plot(t,x); hold on;
plot([C C],[-0.5 2.5],'r','linewidth',2);
plot([0 t(end)],[MIDLEV MIDLEV],'r','linewidth',2);
axis tight;
```



60% Reference Level Instant and Waveform Value

Obtain the 60% reference level instant and value for a bilevel waveform.

```
load('transitionex.mat','x','t');
[C,Lev60] = midcross(x,t,'MidPct',60);
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003. p. 20.

See Also

`falltime` | `pulsewidth` | `risetime` | `settlingtime` | `statelevels`

Purpose

Modulation for communications simulation

Syntax

```
y = modulate(x,fc,fs,'method')
y = modulate(x,fc,fs,'method',opt)
[y,t] = modulate(x,fc,fs)
```

Description

`y = modulate(x,fc,fs,'method')` and

`y = modulate(x,fc,fs,'method',opt)` modulate the real message signal `x` with a carrier frequency `fc` and sampling frequency `fs`, using one of the options listed below for `'method'`. Note that some methods accept an option, `opt`.

Note Use `modulate` and `demod` in the Signal Processing Toolbox with real-valued signals to obtain real-valued outputs. `modulate` and `demod` are not intended to accept complex-valued inputs or produce complex-valued outputs.

Method	Description

`amdsb-sc`
or
`am`
	Amplitude modulation, double sideband, suppressed carrier. Multiplies `x` by a sinusoid of frequency `fc`.
	`y = x.*cos(2*pi*fc*t)`

`amdsb-tc`	Amplitude modulation, double sideband, transmitted carrier. Subtracts scalar `opt` from `x` and multiplies the result by a sinusoid of frequency `fc`. `y = (x-opt).*cos(2*pi*fc*t)` If the `opt` parameter is not present, `modulate` uses a default of `min(min(x))` so that the message signal `(x-opt)` is entirely nonnegative and has a minimum value of 0.

Method	Description
`amssb`	Amplitude modulation, single sideband. Multiplies `x` by a sinusoid of frequency `fc` and adds the result to the Hilbert transform of `x` multiplied by a phase shifted sinusoid of frequency `fc`. `y = x.*cos(2*pi*fc*t)+imag(hilbert(x)).*sin(2*pi*fc*t)` This effectively removes the upper sideband.

`fm`	Frequency modulation. Creates a sinusoid with instantaneous frequency that varies with the message signal `x`.
	
	`y = cos(2*pi*fc*t + opt*cumsum(x))`
	
	`cumsum` is a rectangular approximation to the integral of `x`. `modulate` uses `opt` as the constant of frequency modulation. If `opt` is not present, `modulate` uses a default of
	
	`opt = (fc/fs)*2*pi/(max(max(x)))`
	
	so the maximum frequency excursion from `fc` is `fc` Hz.

`pm`	Phase modulation. Creates a sinusoid of frequency `fc` whose phase varies with the message signal `x`.
	
	`y = cos(2*pi*fc*t + opt*x)`
	
	`modulate` uses `opt` as the constant of phase modulation. If `opt` is not present, `modulate` uses a default of
	
	`opt = pi/(max(max(x)))`
	
	so the maximum phase excursion is π radians.

Method	Description
`pwm`	Pulse-width modulation. Creates a pulse-width modulated signal from the pulse widths in `x`. The elements of `x` must be between 0 and 1, specifying the width of each pulse in fractions of a period. The pulses start at the beginning of each period, that is, they are left justified. `modulate(x,fc,fs,'pwm','centered')` yields pulses centered at the beginning of each period. `y` is length `length(x)*fs/fc`.
`ppm`	Pulse-position modulation. Creates a pulse-position modulated signal from the pulse positions in `x`. The elements of `x` must be between 0 and 1, specifying the left edge of each pulse in fractions of a period. `opt` is a scalar between 0 and 1 that specifies the length of each pulse in fractions of a period. The default for `opt` is `0.1`. `y` is length `length(x)*fs/fc`.
`qam`	Quadrature amplitude modulation. Creates a quadrature amplitude modulated signal from signals `x` and `opt`. `y = x.*cos(2*pi*fc*t) + opt.*sin(2*pi*fc*t)` `opt` must be the same size as `x`.

If you do not specify '*method*', then `modulate` assumes `am`. Except for the `pwm` and `ptm` cases, `y` is the same size as `x`.

If `x` is an array, `modulate` modulates its columns.

`[y,t] = modulate(x,fc,fs)` returns the internal time vector `t` that `modulate` uses in its computations.

See Also

`demod` | `vco`

mscohere

Purpose

Magnitude squared coherence

Syntax

```
Cxy = mscohere(x,y)
Cxy = mscohere(x,y,window)
Cxy = mscohere(x,y,window,noverlap)
[Cxy,W] = mscohere(x,y,window,noverlap,nfft)
[Cxy,F] = mscohere(x,y,window,noverlap,nfft,fs)
[...] = mscohere(x,y,...,'twosided')
mscohere(...)
```

Description

`Cxy = mscohere(x,y)` finds the magnitude squared coherence estimate Cxy of the input signals `x` and `y` using Welch's averaged, modified periodogram method. The magnitude squared coherence estimate is a function of frequency with values between 0 and 1 that indicates how well `x` corresponds to `y` at each frequency. The magnitude squared coherence is a function of the power spectral densities (P_{xx}(f) and P_{yy}(f)) of `x` and `y` and the cross power spectral density (P_{xy}(f)) of `x` and `y`.

$$C_{xy}(f) = \frac{|P_{xy}(f)|^2}{P_{xx}(f)P_{yy}(f)}$$

`x` and `y` must be the same length. For real `x` and `y`, `mscohere` returns a one-sided coherence estimate and for complex `x` or `y`, it returns a two-sided estimate.

`mscohere` uses the following default values:

Parameter	Description	Default Value
`nfft`	FFT length which determines the frequencies at which the coherence is estimated For real `x` and `y`, the length of `Cxy` is (`nfft`/2+1) if `nfft` is even or (`nfft`+1)/2 if `nfft` is odd. For complex `x` or `y`, the length of `Cxy` is `nfft`. If `nfft` is greater than the signal length, the data is zero-padded. If `nfft` is less than the signal length, the segment is wrapped using datawrap so that the length is equal to `nfft`.	Maximum of 256 or the next power of 2 greater than the length of each section of `x` or `y`
`fs`	Sampling frequency	1
`window`	Windowing function and number of samples to use for each section	Periodic Hamming window of length to obtain eight equal sections of x and `y`
`noverlap`	Number of samples by which the sections overlap	Value to obtain 50% overlap

Note You can use the empty matrix `[]` to specify the default value for any input argument except `x` or `y`. For example, `Pxy = mschoere(x,y,[],[],128)` uses a Hamming window, default `noverlap` to obtain 50% overlap, and the specified 128 `nfft`.

`Cxy = mscohere(x,y,window)` specifies a windowing function, divides `x` and `y` into equal overlapping sections of the specified window length, and windows each section using the specified window function. If you supply a scalar for `window`, `Cxy` uses a Hamming window of that length. `mscohere` zero pads the sections if the window length exceeds `nfft`.

`Cxy = mscohere(x,y,window,noverlap)` overlaps the sections of `x` by `noverlap` samples. `noverlap` must be an integer smaller than the length of `window`.

`[Cxy,W] = mscohere(x,y,window,noverlap,nfft)` uses the specified FFT length `nfft` to calculate the coherence estimate. It also returns `W`, which is the vector of normalized frequencies (in rad/sample) at which the coherence is estimated. For real `x` and `y`, `Cxy` length is (`nfft`/2 +1) if `nfft` is even and if `nfft` is odd, the length is (`nfft`+1)/2. For complex `x` or `y`, the length of `Cxy` is `nfft`. For real signals, the range of `W` is [0, pi] when `nfft` is even and [0, pi) when `nfft` is odd. For complex signals, the range of `W` is [0, 2*pi).

`[Cxy,F] = mscohere(x,y,window,noverlap,nfft,fs)` returns `Cxy` as a function of frequency and a vector `F` of frequencies at which the coherence is estimated. `fs` is the sampling frequency in Hz. For real signals, the range of `F` is [0, `fs`/2] when `nfft` is even and [0, `fs`/2) when `nfft` is odd. For complex signals, the range of `F` is [0, `fs`).

`[...] = mscohere(x,y,...,'twosided')` returns a coherence estimate with frequencies that range over the whole Nyquist interval. Specifying `'onesided'` uses half the Nyquist interval.

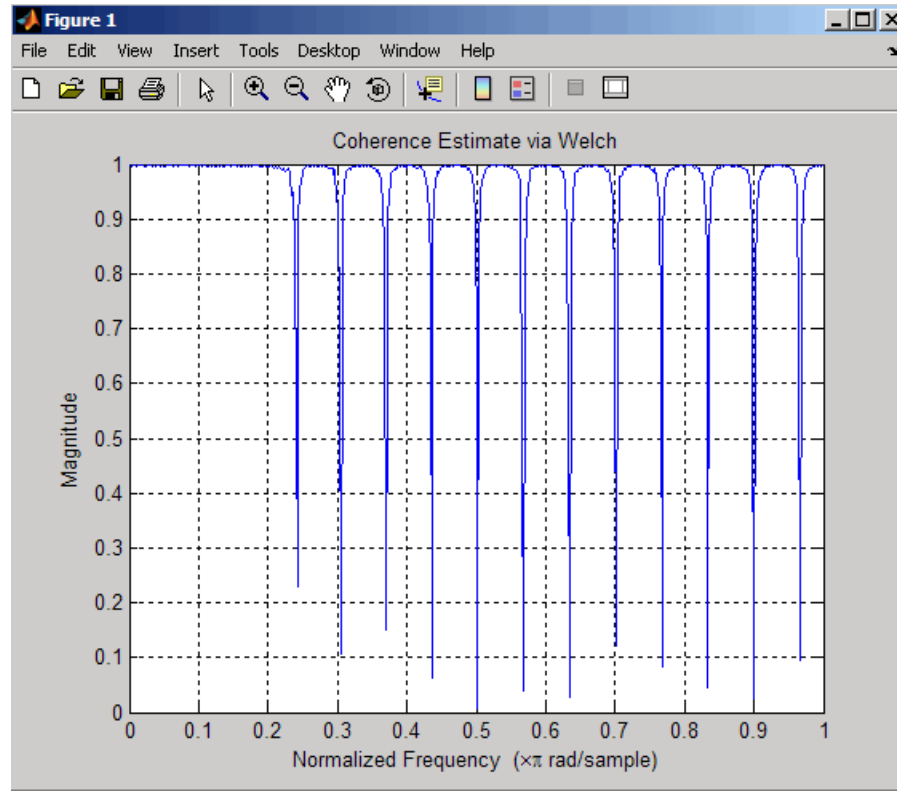
`mscohere(...)` plots the magnitude squared coherence versus frequency in the current figure window.

Note If you estimate the magnitude squared coherence with a single window, or section, the value is identically 1 for all frequencies [1] . You must use at least two sections.

Examples

Compute and plot the coherence estimate between two colored noise sequences `x` and `y`:

```
rng default;
h = fir1(30,0.2,rectwin(31));
h1 = ones(1,10)/sqrt(10);
r = randn(16384,1);
x = filter(h1,1,r);
y = filter(h,1,x);
mscohere(x,y,hanning(1024),512,1024)
```



Algorithms

`mscohere` estimates the magnitude squared coherence function [2] using Welch's overlapped averaged periodogram method (see references [3] and [4]).

References

[1] Stoica, P., and R. Moses. *Introduction to Spectral Analysis.* Upper Saddle River, NJ: Prentice-Hall, 2005. pp. 67-68.

[2] Kay, S.M. *Modern Spectral Estimation.* Englewood Cliffs, NJ: Prentice-Hall, 1988. pp. 453-455.

[3] Rabiner, L.R., and B. Gold. *Theory and Application of Digital Signal Processing.* Englewood Cliffs, NJ: Prentice-Hall, 1975.

[4] Welch, P.D. "The Use of Fast Fourier Transform for the Estimation of Power Spectra: A Method Based on Time Averaging Over Short, Modified Periodograms." *IEEE Trans. Audio Electroacoust. Vol. AU-15 (June 1967)*, pp. 70-73.

See Also

`cpsd` | `periodogram` | `pwelch` | `spectrum` | `tfestimate`

Purpose

Nuttall-defined minimum 4-term Blackman-Harris window

Syntax

```
w = nuttallwin(N)
w = nuttalwin(N,SFLAG)
```

Description

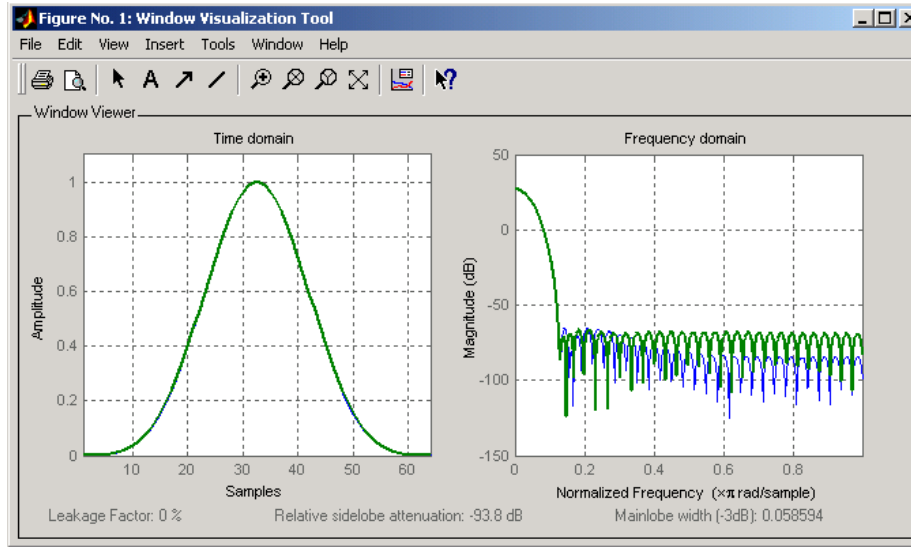
`w = nuttallwin(N)` returns a Nuttall defined `N`-point, 4-term symmetric Blackman-Harris window in the column vector `w`. The window is minimum in the sense that its maximum sidelobes are minimized. The coefficients for this window differ from the Blackman-Harris window coefficients computed with `blackmanharris` and produce slightly lower sidelobes.

`w = nuttalwin(N,SFLAG)` uses `SFLAG` window sampling. `SFLAG` can be `'symmetric'` or `'periodic'`. The default is `'symmetric'`. You can find the equations defining the symmetric and periodic windows in "Definitions" on page 1-724.

Examples

Compare 64-point Blackman-Harris and Nuttall's Blackman-Harris windows and plot them using WVTool:

```
L = 64;
w = blackmanharris(L);
y = nuttallwin(L);
wvtool(w,y)
```



The maximum difference between the two windows is

```
max(abs(y-w))

ans =

    0.0099
```

Definitions

The equation for the **symmetric** Nuttall defined 4-term Blackman-Harris window is

$$w(n) = a_0 - a_1 \cos\left(2\pi\frac{n}{N-1}\right) + a_2 \cos\left(4\pi\frac{n}{N-1}\right) - a_3 \cos\left(6\pi\frac{n}{N-1}\right)$$

where n= 0,1,2, ... N-1.

The equation for the **periodic** Nuttall defined 4-term Blackman-Harris window is

$$w(n) = a_0 - a_1 \cos\left(2\pi\frac{n}{N}\right) + a_2 \cos\left(4\pi\frac{n}{N}\right) - a_3 \cos\left(6\pi\frac{n}{N}\right)$$

where n= 0,1,2, ... N-1. The periodic window is N-periodic.

The coefficients for this window are

a_0 = 0.3635819

a_1 = 0.4891775

a_2 = 0.1365995

a_3 = .0106411

References

[1] Nuttall, Albert H. "Some Windows with Very Good Sidelobe Behavior." *IEEE Transactions on Acoustics, Speech, and Signal Processing*. Vol. ASSP-29 (February 1981). pp. 84-91.

See Also

`barthannwin` | `bartlett` | `blackmanharris` | `bohmanwin` | `parzenwin` | `rectwin` | `triang` | `window` | `wintool` | `wvtool`

overshoot

Purpose

Overshoot metrics of bilevel waveform transitions

Syntax

```
OS = overshoot(X)
OS = overshoot(X,FS)
OS = overshoot(X,T)
[OS,OSLEV,OSINST] = overshoot(...)
[...] = overshoot(...,Name,Value)
overshoot(...)
```

Description

`OS = overshoot(X)` returns the greatest absolute deviations larger than the final state levels of each transition in the bilevel waveform, `X`. The overshoots, `OS`, are expressed as a percentage of the difference between the state levels. The length of `OS` corresponds to the number of transitions detected in the input signal. The sample instants in `X` correspond to the vector indices. To determine the transitions, `overshoot` estimates the state levels of the input waveform by a histogram method. `overshoot` identifies all intervals which cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-731.

`OS = overshoot(X,FS)` specifies the sampling frequency in hertz. The sampling frequency determines the sample instants corresponding to the elements in `X`. The first sample instant in `X` corresponds to t=0.

`OS = overshoot(X,T)` specifies the sample instants, `T`, as a vector with the same number of elements as `X`.

`[OS,OSLEV,OSINST] = overshoot(...)` returns the levels, `OSLEV`, and sample instants,`OSINST`, of the overshoots for each transition.

`[...] = overshoot(...,Name,Value)` returns the greatest deviations larger than the final state level with additional options specified by one or more `Name,Value` pair arguments.

`overshoot(...)` plots the bilevel waveform and marks the location of the overshoot of each transition as well as the lower and upper

reference-level instants and the associated reference levels. The state levels and associated lower and upper-state boundaries are also plotted.

Input Arguments

X

Bilevel waveform. `X` is a real-valued row or column vector.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

'PctRefLevels'

Reference levels as a percentage of the waveform amplitude. The lower-state level is defined to be 0 percent. The upper-state level is defined to be 100 percent. The value of `'PCTREFLEVELS'` is a two-element real row vector whose elements correspond to the lower and upper percent reference levels.

Default: `[10 90]`

'Region'

Specifies the region over which to compute the overshoot. Valid values for `'Region'` are `'Preshoot'` or `'Postshoot'`. If you specify `'Preshoot'`, the end of the pretransition aberration region is defined as the last instant where the signal exits the first state. If you specify `'Postshoot'`, the start of the posttransition aberration region is defined as the instant when the signal enters the second state.

Default: `'Postshoot'`

'SeekFactor'

Aberration region duration. Specifies the duration of the region over which to compute the overshoot for each transition as a multiple of the corresponding transition duration. If the edge of the waveform is reached, or a complete intervening transition is detected before the duration aberration region duration elapses, the duration is truncated to the edge of the waveform or the start of the intervening transition.

Default: `3`

'StateLevels'

Lower and upper state levels. Specifies the levels to use for the lower and upper state levels as a two-element real row vector whose first and second elements correspond to the lower and upper state levels of the input waveform.

'Tolerance'

Specifies the tolerance that the initial and final levels of each transition must be within the respective state levels. The `'Tolerance'` value is a scalar expressed as the percentage of the difference between the upper and lower state levels.

Default: `2`

Output Arguments

OS

Overshoots expressed as a percentage of the state levels. The overshoot percentages are computed based on the greatest deviation from the final state level in each transition. By default overshoots are computed for posttransition aberration regions. See “Overshoot” on page 1-729.

OSLEV

Level of the pretransition or posttransition overshoot.

OSINST

Sample instants of pretransition or posttransition overshoots. If you specify the sampling frequency or sampling instants, the overshoot instants are in seconds. If you do not specify the sampling frequency or sampling instants, the overshoot instants are the indices of the input vector.

Definitions

Overshoot

For a positive-going (positive-polarity) pulse, overshoot expressed as a percentage is

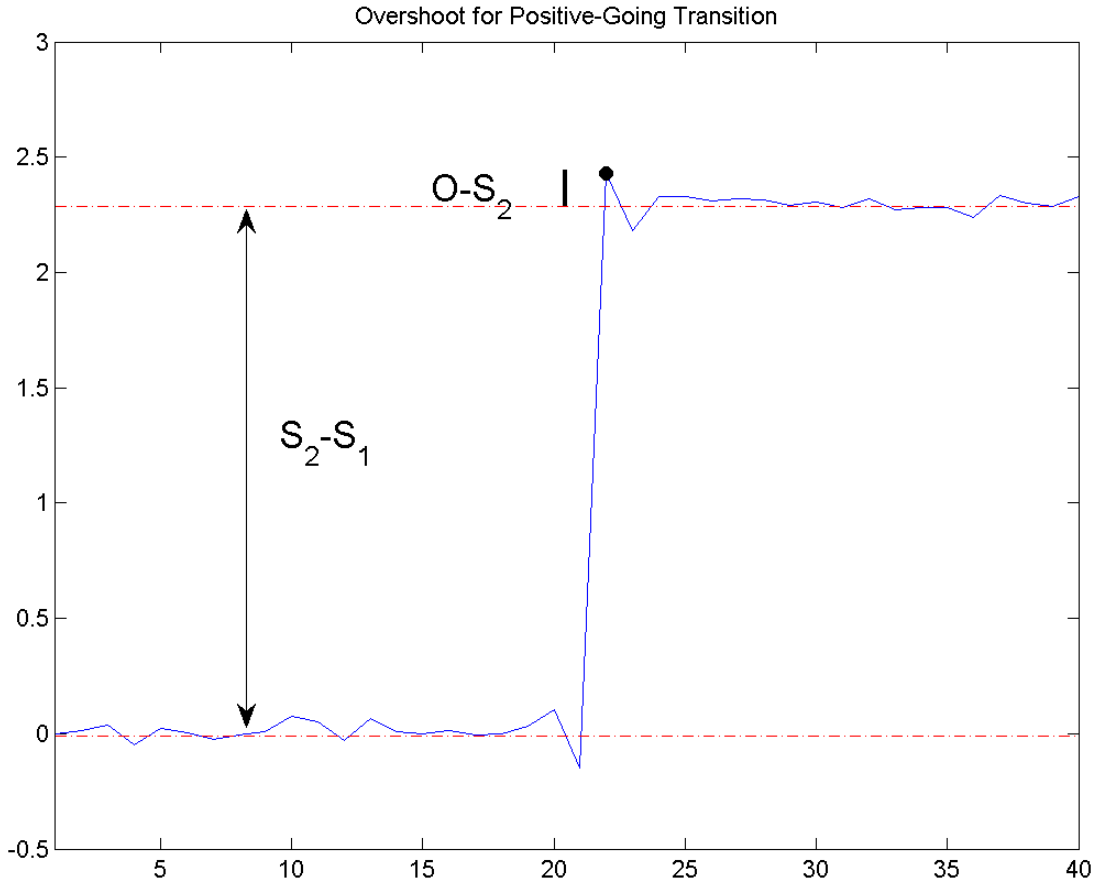
$$100\frac{(O - S_2)}{(S_2 - S_1)}$$

where O is the maximum deviation greater the high-state level, S_2 is the high state, and S_1 is the low state.

For a negative-going (negative-polarity) pulse, overshoot expressed as a percentage is

$$100\frac{(O - S_1)}{(S_2 - S_1)}$$

The following figure illustrates the calculation of overshoot for a positive-going transition.



The red dashed lines indicate the estimated state levels. The double-sided black arrow depicts the difference between the high and low-state levels. The solid black line indicates the difference between the overshoot value and the high-state level.

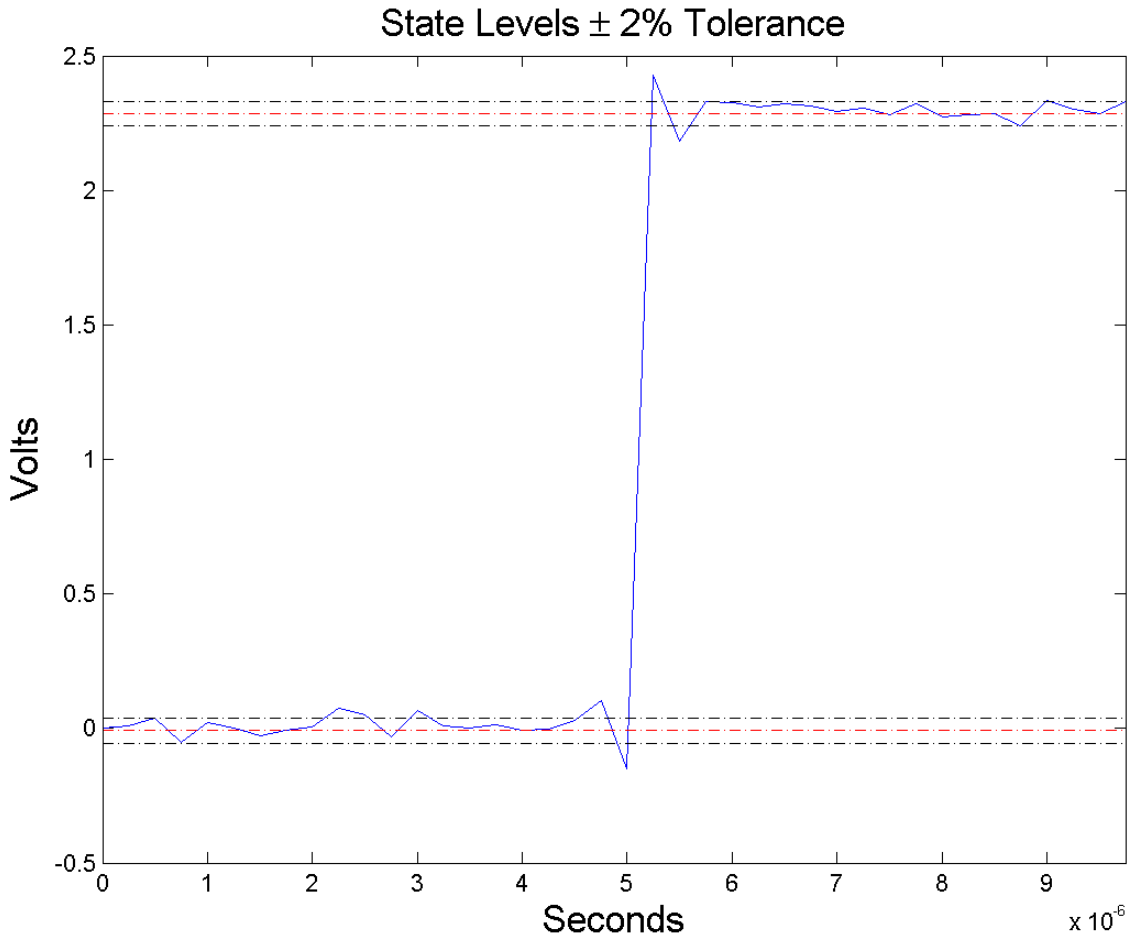
State-Level Tolerances

Each state level can have associated lower- and upper-state boundaries. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The estimated state levels are indicated by a dashed red line.



Examples

Overshoot Percentage in Posttransition Aberration Region

Determine the maximum percent overshoot relative to the high-state level in a 2.3 V clock waveform.

Load the 2.3 V clock data. Plot the waveform. In this example, you see that the maximum overshoot in the posttransition region occurs near index 22.

```
load('transitionex.mat', 'x');
plot(x);
set(gca,'xtick',[1 6 12 18 22 28 34 40]);
```

Determine the maximum percent overshoot.

```
os = overshoot(x);
```

Overshoot Percentage, Levels, and Sample Instant in Posttransition Aberration Region

Determine the maximum percent overshoot relative to the high-state level, the level of the overshoot, and the sample instant in a 2.3 V clock waveform.

Load the 2.3 V clock data with sampling instants. Plot the waveform. The clock data is sampled at 4 MHz.

```
load('transitionex.mat', 'x','t');
plot(t,x);
```

Determine the maximum percent overshoot, the level of the overshoot in volts, and the sampling instant where the maximum overshoot occurs. Plot the result.

```
[os,oslev,osinst] = overshoot(x,t);
plot(t.*1e6,x); xlabel('Microseconds');
hold on; grid on;
plot(osinst*1e6,oslev,'ro','markerfacecolor',[1 0 0]);
```

Overshoot Percentage, Levels, and Sample Instant in Pretransition Aberration Region

Determine the maximum percent overshoot relative to the low-state level, the level of the overshoot, and the sample instant in a 2.3 V clock

waveform. Specify the `'Region'` as `'Preshoot'` to output pretransition metrics.

Load the 2.3 V clock data with sampling instants. Plot the waveform. The clock data is sampled at 4 MHz.

```
load('transitionex.mat', 'x','t');
plot(t,x);
```

Determine the maximum percent overshoot, the level of the overshoot in volts, and the sampling instant where the maximum overshoot occurs. Plot the result.

```
load('transitionex.mat', 'x','t');
[os,oslev,osinst] = overshoot(x,t,'Region','Preshoot');
plot(t.*1e6,x); xlabel('Microseconds');
hold on; grid on;
plot(osinst*1e6,oslev,'ro','markerfacecolor',[1 0 0]);
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003, pp. 15–17.

See Also

`settlingtime` | `statelevels` | `overshoot`

Purpose

Parzen (de la Valle-Poussin) window

Syntax

```
w = parzenwin(L)
```

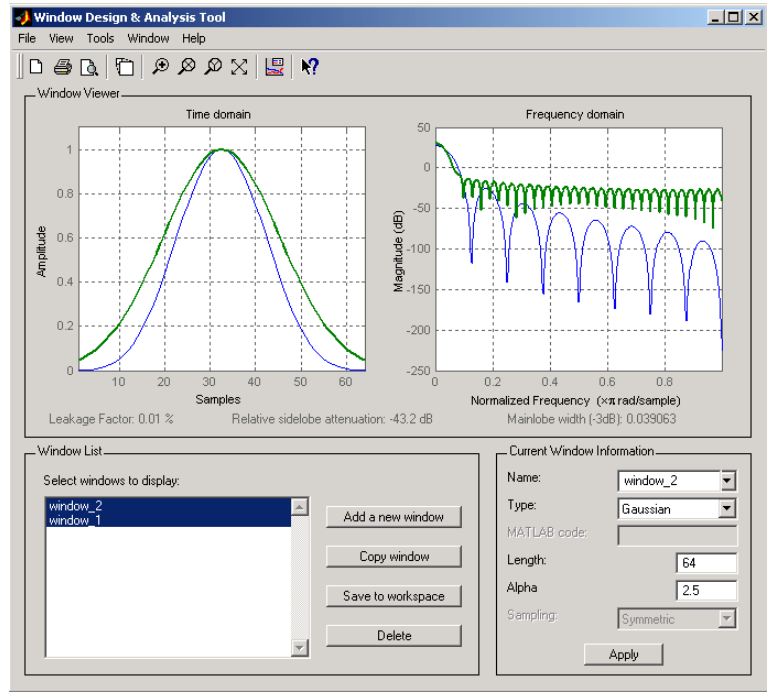
Description

`w = parzenwin(L)` returns the `L`-point Parzen (de la Valle-Poussin) window in column vector `w`. Parzen windows are piecewise cubic approximations of Gaussian windows. Parzen window sidelobes fall off as $1/\omega^4$.

Examples

Compare 64-point Parzen and Gaussian windows and display the result using `sigwin` window objects and `wintool`:

```
wintool(sigwin.parzenwin(64),sigwin.gausswin(64))
```



Algorithms

The following equation defines the *N*–point Parzen window over the interval $-\frac{(N-1)}{2} \le n \le \frac{(N-1)}{2}$:

$$w(n) = \begin{cases} 1 - 6\left(\frac{|n|}{N/2}\right)^2 + 6\left(\frac{|n|}{N/2}\right)^3 & 0 \le |n| \le (N-1)/4 \\ 2\left(1 - \frac{|n|}{N/2}\right)^3 & (N-1)/4 < |n| \le (N-1)/2 \end{cases}$$

References

[1] Harris, F.J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform.” *Proceedings of the IEEE*. Vol. 66, No. 1 (January 1978).

See Also

`barthannwin` | `bartlett` | `blackmanharris` | `bohmanwin` | `nuttallwin` | `rectwin` | `triang` | `window` | `wintool` | `wvtool`

Purpose

PSD using Burg method

Syntax

```
Pxx = pburg(x,order)
Pxx = pburg(x,order,nfft)
[Pxx,w] = pburg(...)
[Pxx,w] = pburg(x,order,w)
Pxx = pburg(x,order,nfft,fs)
Pxx = pburg(x,order,f,fs)
[Pxx,f] = pburg(x,order,nfft,fs)
[Pxx,f] = pburg(x,order,f,fs)
[Pxx,f] = pburg(x,order,nfft,fs,freqrange)
[Pxx,w] = pburg(x,order,nfft,freqrange)
[Pxx,f,Pxxc] = pburg(...,'ConfidenceLevel',P)
pburg(...)
```

Description

`Pxx = pburg(x,order)` implements the Burg algorithm, a parametric spectral estimation method, and returns `Pxx`, an estimate of the power spectral density (PSD) of the vector `x`. The entries of `x` represent samples of a discrete-time signal, and `order` is the integer specifying the order of an autoregressive (AR) prediction model for the signal, used in estimating the PSD.

The power spectral density is calculated in units of power per radians per sample. Real-valued inputs produce full power one-sided (in frequency) PSDs (by default), while complex-valued inputs produce two-sided PSDs.

In general, the length of the FFT and the values of the input `x` determine the length of `Pxx` and the range of the corresponding normalized frequencies. For this syntax, the (default) FFT length is 256. The following table indicates the length of `Pxx` and the range of the corresponding normalized frequencies for this syntax.

PSD Vector Characteristics for an FFT Length of 256 (Default)

Real/Complex Input Data	Length of Pxx	Range of the Corresponding Normalized Frequencies
Real-valued	129	[0, π]
Complex-valued	256	[0, 2π)

`Pxx = pburg(x,order,nfft)` uses the integer FFT length `nfft` to calculate the PSD vector `Pxx`.

`[Pxx,w] = pburg(...)` also returns `w`, a vector of normalized angular frequencies at which the two-sided PSD is estimated. `Pxx` and `w` have the same length. The units for `w` are rad/sample.

The length of `Pxx` and the frequency range for `w` depend on `nfft` and the values of the input `x`. The following table indicates the length of `Pxx` and the frequency range for `w` in this syntax.

PSD and Frequency Vector Characteristics

Real/Complex Input Data	nfft Even/Odd	Length of Pxx	Range of w
Real-valued	Even	`(nfft/2 + 1)`	[0, π]
Real-valued	Odd	`(nfft + 1)/2`	[0, π)
Complex-valued	Even or odd	`nfft`	[0, 2π)

`[Pxx,w] = pburg(x,order,w)` uses a vector of normalized frequencies `w` with two or more elements to compute the PSD at those frequencies and returns a two-sided PSD.

```
Pxx = pburg(x,order,nfft,fs)
```

or

`Pxx = pburg(x,order,f,fs)` uses the integer FFT length `nfft` to calculate the PSD vector `Pxx` or uses the vector of frequencies `f` in Hz and the sampling frequency `fs` to compute the two-sided PSD vector `Pxx` at those frequencies. If you specify `nfft` as the empty vector `[]`, it uses the default value of 256. If you specify `fs` as the empty vector `[]`, the sampling frequency `fs` defaults to 1 Hz. The spectral density produced is calculated in units of power per Hz.

`[Pxx,f] = pburg(x,order,nfft,fs)`

or

`[Pxx,f] = pburg(x,order,f,fs)` returns the frequency vector f. In this case, the units for the frequency vector are in Hz. The frequency range for `f` depends on `nfft`, `fs`, and the values of the input `x`. The length of `Pxx` is the same as in the table above. The following table indicates the frequency range for `f` for this syntax.

PSD and Frequency Vector Characteristics with fs Specified

Real/Complex Input Data	nfft Even/Odd	Range of f
Real-valued	Even	`[0,fs/2]`
Real-valued	Odd	`[0,fs/2)`
Complex-valued	Even or odd	`[0,fs)`

`[Pxx,f] = pburg(x,order,nfft,fs,freqrange)` or

`[Pxx,w] = pburg(x,order,nfft,freqrange)` specifies the range of frequency values to include in the output frequency vectors, `f` or `w`. This syntax is useful when `x` is real. `freqrange` can be either:

- `'onesided'` — returns the one-sided PSD of a real input signal, `x`. If `nfft` is even, `Pxx` has length `nfft/2+1` and is computed over the interval [0,π]. If `nfft` is odd, the length of `Pxx` is `(nfft+1)/2` and the frequency interval is [0,π). When you specify `fs` , the intervals are [0,`fs`/2] and [0,`fs`/2) for even and odd length `nfft` respectively.

- `'twosided'` — returns the two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval [0,2π). When you specify `fs`, the frequency interval is [0,`fs`).
- `'centered'` — returns the centered two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval (-π, π] for even length `nfft` and (-π, π) for odd length `nfft`. When you specify `fs`, the frequency intervals are (-`fs`/2, `fs`/2] and (-`fs`/2,`fs`/2) for even and odd length `nfft` respectively.

`[Pxx,f,Pxxc] = pburg(...,'ConfidenceLevel',P)` returns the P100% confidence interval for `Pxx`, where `P` is a nonnegative scalar between 0 and 1. The default value for `P` is 0.95. Large-sample confidence intervals are computed using a Gaussian probability density function. `Pxxc` is N-by-2 matrix, where N is the length of `Pxx`. The first column, `Pxxc(:,1)`, is the lower bound of the confidence interval. The second column, `Pxxc(:,2)`, is the upper bound. See [1] for a description of approximate large-sample confidence intervals for AR PSD estimates.

`pburg(...)` with no outputs plots the PSD in the current figure window. The frequency range on the plot is the same as the range of output `w` (or `f`) for a given set of parameters.

Tips

The power spectral density is computed as the distribution of power per unit frequency. This algorithm depends on your selecting an appropriate model order for your signal.

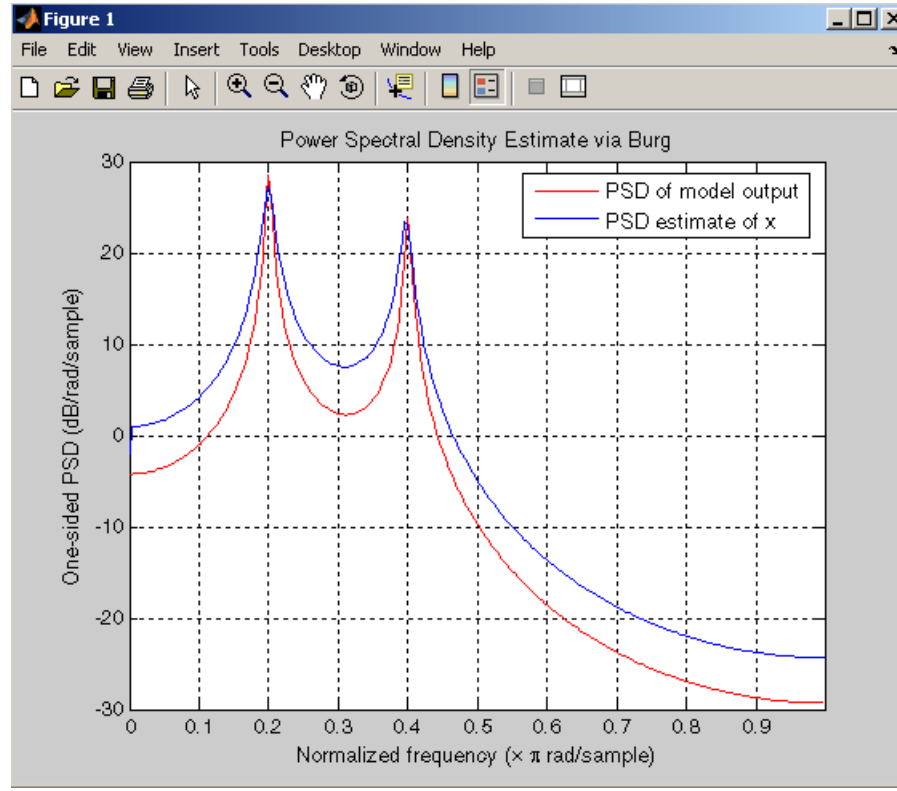
Examples

Compare Filter Response with Process Realization

The Burg method estimates the spectral density by fitting an AR prediction model of a given order to the signal, so first generate a signal from an AR (all-pole) model of a given order. Use `freqz` to check the magnitude of the frequency response of your AR filter. Then, generate the input signal `x` by filtering white noise through the AR filter. Estimate the PSD of `x` based on a fourth-order AR prediction model because in this case we know that the original AR system model `a` has order 4:

```
% Define AR filter coefficients
```

```
a = [1 -2.2137 2.9403 -2.1697 0.9606];
[H,w] = freqz(1,a,256);     % AR filter freq response
% Scale to make one-sided PSD
Hp = plot(w/pi,20*log10(2*abs(H)/(2*pi)),'r');
hold on;
rng default;
x = filter(1,a,randn(256,1));           % AR system output
pburg(x,4,511);
xlabel('Normalized frequency (\times \pi rad/sample)')
ylabel('One-sided PSD (dB/rad/sample)')
legend('PSD of model output','PSD estimate of x')
```

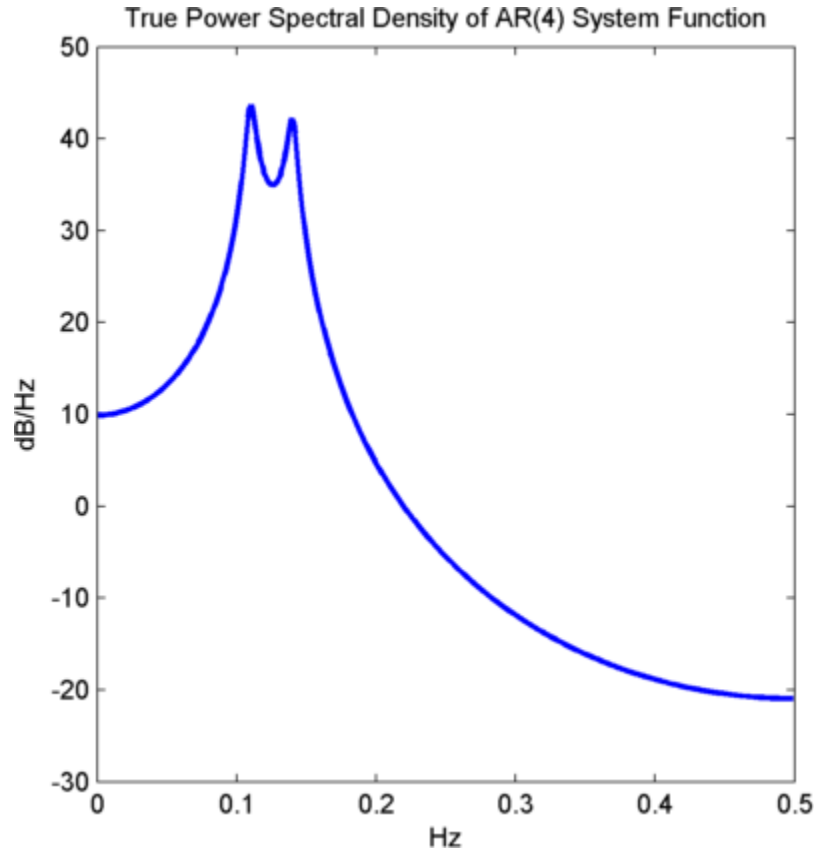


Large-Sample Confidence Intervals for AR PSD Estimate

This example shows you how to obtain and plot confidence intervals for an AR PSD estimate.

Create the coefficients for an AR(4) system function. Use `freqz` to obtain and plot the true power spectral density.

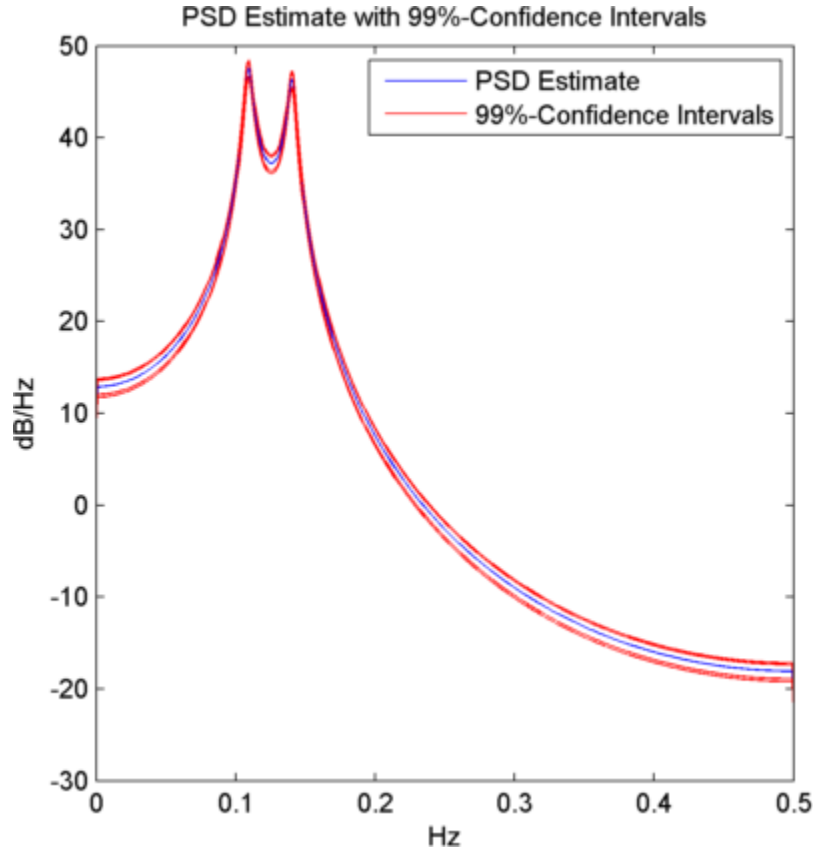
```
A = [1 -2.7607 3.8106 -2.6535 0.9238];
[H,F] = freqz(1,A,[],1);
plot(F,20*log10(abs(H)),'b','linewidth',2);
xlabel('Hz'); ylabel('dB/Hz');
title('True Power Spectral Density of AR(4) System Function')
```



Create a realization of the AR(4) process represented by the coefficients. Set the random number generator to the default settings for reproducible results. Obtain approximate large-sample 99%-confidence intervals for the PSD estimate.

```
rng default;
x = randn(1000,1);
y = filter(1,A,x);
[Pxx,F,Pxxc] = pburg(y,4,1024,1,'ConfidenceLevel',0.99);
plot(F,10*log10(Pxx),'b'); hold on;
```

```
plot(F,10*log10(Pxxc),'r'); xlabel('Hz'); ylabel('dB/Hz');
legend('PSD Estimate', '99%-Confidence Intervals')
title('PSD Estimate with 99%-Confidence Intervals')
```



Algorithms

You can use linear prediction filters to model the second-order statistical characteristics of a signal. The prediction filter output can be used to model the signal when the input is white noise.

The Burg method fits an AR linear prediction filter model of the specified order to the input signal by minimizing (using least squares)

the arithmetic mean of the forward and backward prediction errors. The spectral density then is computed from the frequency response of the prediction filter. The AR filter parameters are constrained to satisfy the Levinson-Durbin recursion.

References

[1] Kay, S.M. *Modern Spectral Estimation*, Englewood Cliffs, NJ, Prentice-Hall, 1988, pp. 194–195.

[2] Marple, S.L. *Digital Spectral Analysis*, Englewood Cliffs, NJ, Prentice-Hall, 1987, Chapter 7.

[3] Stoica, P., and R.L. Moses, *Introduction to Spectral Analysis*, Prentice-Hall, 1997.

See Also

`arburg` | `lpc` | `pcov` | `peig` | `periodogram` | `pmcov` | `pmtm` | `pmusic` | `pwelch` | `pyulear`

Purpose

PSD using covariance method

Syntax

```
Pxx = pcov(x,order)
Pxx = pcov(x,order,nfft)
[Pxx,w] = pcov(...)
[Pxx,w] = pcov(x,order,w)
Pxx = pcov(x,order,nfft,fs)
Pxx = pcov(x,order,f,fs)
[Pxx,f] = pcov(x,order,nfft,fs)
[Pxx,f] = pcov(x,order,f,fs)
[Pxx,f] = pcov(x,order,nfft,fs,freqrange)
[Pxx,w] = pcov(x,order,nfft,freqrange)
[Pxx,f,Pxxc] = pcov(...,'ConfidenceLevel',P)
pcov(...)
```

Description

`Pxx = pcov(x,order)` implements the covariance algorithm, a parametric spectral estimation method, and returns `Pxx`, an estimate of the power spectral density (PSD) of the vector `x`. The entries of `x` represent samples of a discrete-time signal, and where `order` is the integer specifying the order of an autoregressive (AR) prediction model for the signal, used in estimating the PSD.

The power spectral density is calculated in units of power per radians per sample. Real-valued inputs produce full power one-sided (in frequency) PSDs (by default), while complex-valued inputs produce two-sided PSDs.

In general, the length of the FFT and the values of the input `x` determine the length of `Pxx` and the range of the corresponding normalized frequencies. For this syntax, the (default) FFT length is 256. The following table indicates the length of `Pxx` and the range of the corresponding normalized frequencies for this syntax.

PSD Vector Characteristics for an FFT Length of 256 (Default)

Real/Complex Input Data	Length of Pxx	Range of the Corresponding Normalized Frequencies
Real-valued	129	[0, π]
Complex-valued	256	[0, 2π)

`Pxx = pcov(x,order,nfft)` uses the integer FFT length `nfft` to calculate the PSD vector `Pxx`.

`[Pxx,w] = pcov(...)` also returns `w`, a vector of normalized angular frequencies at which the two-sided PSD is estimated. `Pxx` and `w` have the same length. The units for `w` are rad/sample.

The length of `Pxx` and the frequency range for `w` depend on `nfft` and the values of the input `x`. The following table indicates the length of `Pxx` and the frequency range for `w` in this syntax.

PSD and Frequency Vector Characteristics

Real/Complex Input Data	nfft Even/Odd	Length of Pxx	Range of w
Real-valued	Even	`(nfft/2 + 1)`	[0, π]
Real-valued	Odd	`(nfft + 1)/2`	[0, π)
Complex-valued	Even or odd	`nfft`	[0, 2π)

`[Pxx,w] = pcov(x,order,w)` uses a vector of normalized frequencies `w` with two or more elements to compute the PSD at those frequencies and returns a two-sided PSD.

```
Pxx = pcov(x,order,nfft,fs)
```

or

`Pxx = pcov(x,order,f,fs)` uses the integer FFT length `nfft` to calculate the PSD vector `Pxx` or uses the vector of frequencies `f` in Hz and the sampling frequency `fs` to compute the two-sided PSD vector `Pxx` at those frequencies. If you specify `nfft` as the empty vector `[]`, it uses the default value of 256. If you specify `fs` as the empty vector `[]`, the sampling frequency `fs` defaults to 1 Hz. The spectral density produced is calculated in units of power per Hz.

`[Pxx,f] = pcov(x,order,nfft,fs)`

or

`[Pxx,f] = pcov(x,order,f,fs)` returns the frequency vector f. In this case, the units for the frequency vector are in Hz. The frequency range for `f` depends on `nfft`, `fs`, and the values of the input `x`. The length of `Pxx` is the same as in the table above. The following table indicates the frequency range for `f` for this syntax.

PSD and Frequency Vector Characteristics with fs Specified

Real/Complex Input Data	nfft Even/Odd	Range of f
Real-valued	Even	`[0,fs/2]`
Real-valued	Odd	`[0,fs/2)`
Complex-valued	Even or odd	`[0,fs)`

`[Pxx,f] = pcov(x,order,nfft,fs,freqrange)` or

`[Pxx,w] = pcov(x,order,nfft,freqrange)` specifies the range of frequency values to include in the output frequency vectors, `f` or `w`. This syntax is useful when `x` is real. `freqrange` can be either:

- `'onesided'` — returns the one-sided PSD of a real input signal, `x`. If `nfft` is even, `Pxx` has length `nfft/2+1` and is computed over the interval [0,π]. If `nfft` is odd, the length of `Pxx` is `(nfft+1)/2` and the frequency interval is [0,π). When your specify `fs` , the intervals are [0,`fs`/2] and [0,`fs`/2) for even and odd length `nfft` respectively.

- `'twosided'` — returns the two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval [0,2π). When you specify `fs`, the frequency interval is [0,`fs`).
- `'centered'` — returns the centered two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval (-π, π] for even length `nfft` and (-π, π) for odd length `nfft`. When you specify `fs`, the frequency intervals are (-`fs`/2, `fs`/2] and (-`fs`/2,`fs`/2) for even and odd length `nfft` respectively.

`[Pxx,f,Pxxc] = pcov(...,'ConfidenceLevel',P)` returns the P100% confidence interval for `Pxx`, where `P` is a nonnegative scalar between 0 and 1. The default value for `P` is 0.95. Large-sample confidence intervals are computed using a Gaussian probability density function. `Pxxc` is N-by-2 matrix, where N is the length of `Pxx`. The first column, `Pxxc(:,1)`, is the lower bound of the confidence interval. The second column, `Pxxc(:,2)`, is the upper bound. See [1] for a description of approximate large-sample confidence intervals for AR PSD estimates.

`pcov(...)` with no outputs plots the PSD in the current figure window. The frequency range on the plot is the same as the range of output `w` (or `f`) for a given set of parameters.

Tips

The power spectral density is computed as the distribution of power per unit frequency.

This algorithm depends on your selecting an appropriate model order for your signal.

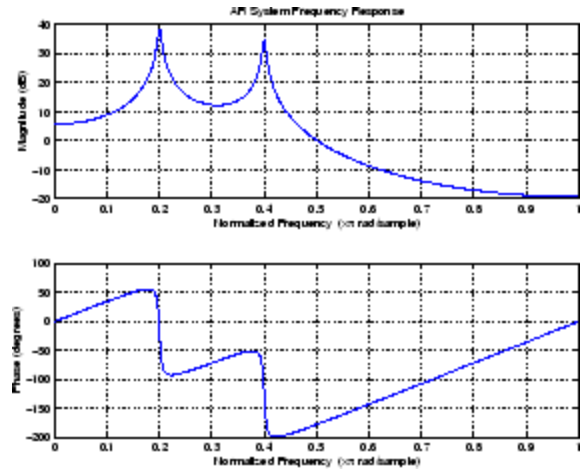
Examples

Covariance Method AR PSD Estimate

Because the covariance method estimates the spectral density by fitting an AR prediction model of a given order to the signal, first generate a signal from an AR (all-pole) model of a given order. You can use `freqz` to check the magnitude of the frequency response of your AR filter. This will give you an idea of what to expect when you estimate the PSD using `pcov`:

```
a = [1 -2.2137 2.9403 -2.1697 0.9606];    % AR filter coefficients
```

```
freqz(1,a)                                % AR filter frequency response
title('AR System Frequency Response')
```



Now generate the input signal `x` by filtering white noise through the AR filter. Estimate the PSD of `x` based on a fourth-order AR prediction model since in this case we know that the original AR system model `a` has order 4:

```
% Signal generated from AR filter
x = filter(1,a,randn(256,1));
% Fourth-order estimate
pcov(x,4)
```

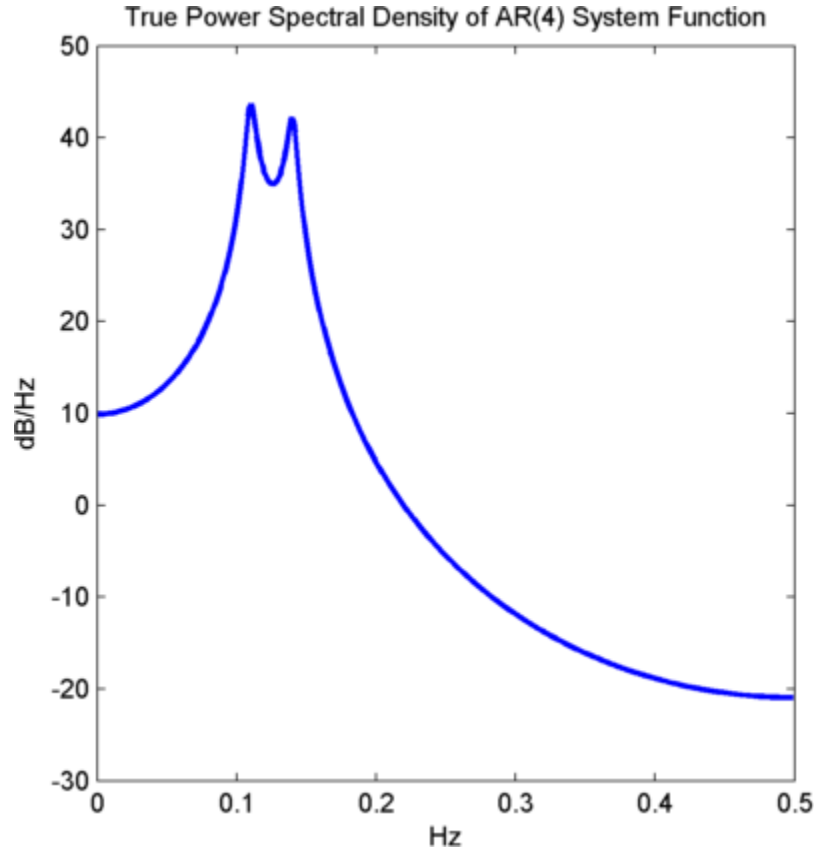
Large-Sample Confidence Intervals for AR PSD Estimate

This example shows you how to obtain and plot confidence intervals for an AR PSD estimate.

Create the coefficients for an AR(4) system function. Use `freqz` to obtain and plot the true power spectral density.

```
A = [1 -2.7607 3.8106 -2.6535 0.9238];
[H,F] = freqz(1,A,[],1);
```

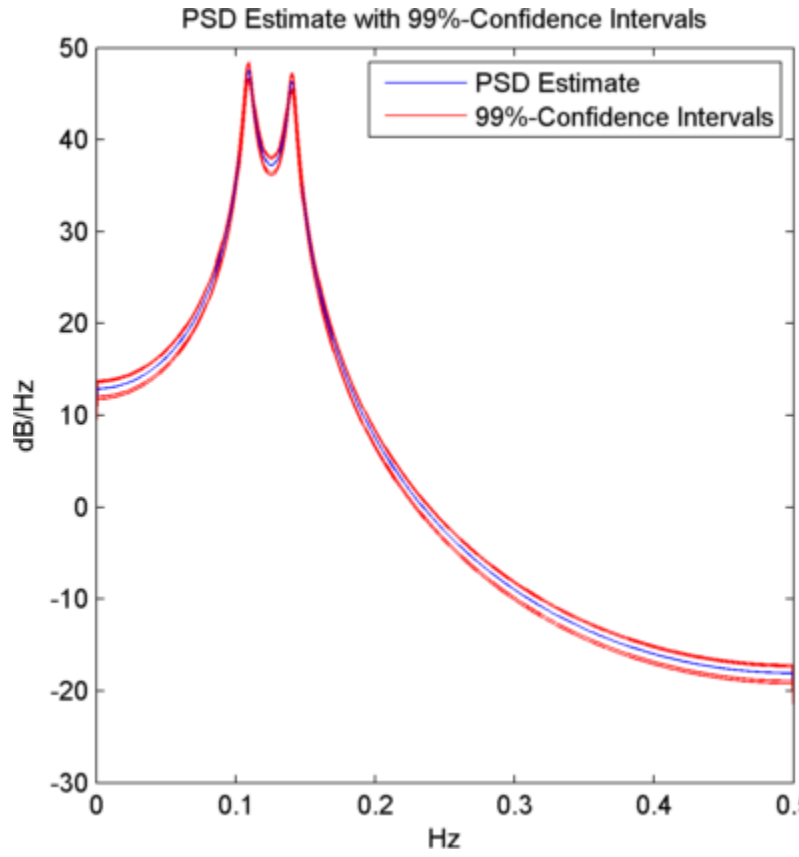
```
plot(F,20*log10(abs(H)),'b','linewidth',2);
xlabel('Hz'); ylabel('dB/Hz');
title('True Power Spectral Density of AR(4) System Function')
```



Create a realization of the AR(4) process represented by the coefficients. Set the random number generator to the default settings for reproducible results. Obtain approximate large-sample 99%-confidence intervals for the PSD estimate.

```
rng default;
```

```
x = randn(1000,1);
y = filter(1,A,x);
[Pxx,F,Pxxc] = pcov(y,4,1024,1,'ConfidenceLevel',0.99);
plot(F,10*log10(Pxx),'b'); hold on;
plot(F,10*log10(Pxxc),'r'); xlabel('Hz'); ylabel('dB/Hz');
legend('PSD Estimate', '99%-Confidence Intervals')
title('PSD Estimate with 99%-Confidence Intervals')
```



Algorithms

Linear prediction filters can be used to model the second-order statistical characteristics of a signal. The prediction filter output can be used to model the signal when the input is white noise.

The covariance method estimates the PSD of a signal using the covariance method. The covariance (or nonwindowed) method fits an AR linear prediction filter model to the signal by minimizing the forward prediction error (based on causal observations of your input signal) in the least squares sense. The spectral estimate returned by `pcov` is the squared magnitude of the frequency response of this AR model.

References

[1] Kay, S.M. *Modern Spectral Estimation*, Englewood Cliffs, NJ, Prentice-Hall, 1988, pp. 194–195.

[2] Marple, S.L. *Digital Spectral Analysis*, Englewood Cliffs, NJ, Prentice-Hall, 1987, Chapter 7.

[3] Stoica, P., and R.L. Moses, *Introduction to Spectral Analysis*, Prentice-Hall, 1997.

See Also

`arcov` | `lpc` | `pburg` | `peig` | `periodogram` | `pcov` | `pmtm` | `pmusic` | `pwelch` | `pyulear`

peak2peak

Purpose

Maximum-to-minimum difference

Syntax

```
Y = peak2peak(X)
Y = peak2peak(X,DIM)
```

Description

`Y = peak2peak(X)` returns the difference between the maximum and minimum values in `X`. `peak2peak` operates along the first nonsingleton dimension of `X` by default. For example, if `X` is a row or column vector, `Y` is a real-valued scalar. If `Y` is an N-by-M matrix with N>1, `Y` is a 1-by-M row vector containing the maximum-to-minimum differences of the columns of `X`.

`Y = peak2peak(X,DIM)` computes the maximum-to-minimum differences of `X` along the dimension, `DIM`.

Input Arguments

X

Real- or complex-valued input vector or matrix. By default, `peak2peak` acts along the first nonsingleton dimension of `X`. For complex-valued inputs, `peak2peak` identifies the maximum and minimum in absolute value. `peak2peak` subtracts the complex number with the minimum modulus from the complex number with the maximum modulus.

DIM

Dimension for maximum-to-minimum difference. The optional `DIM` input argument specifies the dimension along which to compute the maximum-to-minimum differences.

Default: First nonsingleton dimension

Output Arguments

Y

Maximum-to-minimum difference. For vectors, `Y` is a real-valued scalar. For matrices, `Y` contains the maximum-to-minimum differences computed along the specified dimension, `DIM`. By default, `DIM` is the first nonsingleton dimension.

Examples

Peak-to-Peak Difference of Sinusoid

Compute the maximum-to-minimum difference of a 100-Hz sinusoid sampled at 1 kHz.

```
t = 0:0.001:1-0.001;
X = cos(2*pi*100*t);
Y = peak2peak(X);
```

Peak-to-Peak Difference of Complex Exponential

Compute the maximum-to-minimum difference of a complex exponential with a frequency of π/4 radians/sample.

Create a complex exponential with a frequency of π/4 radians/sample. Find the peak-to-peak difference.

```
n = 0:99;
x = exp(1j*pi/4*n);
maxmin = peak2peak(x);
```

Peak-to-Peak Differences of 2-D Matrix

Create a matrix where each column is a 100-Hz sinusoid sampled at 1 kHz with a different amplitude. The amplitude is equal to the column index.

Compute the maximum-to-minimum differences of the columns.

```
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t)';
X = repmat(x,1,4);
amp = 1:4;
amp = repmat(amp,1e3,1);
X = X.*amp;
Y = peak2peak(X);
```

Peak-to-Peak Differences of 2-D Matrix Along Specified Dimension

Create a matrix where each row is a 100-Hz sinusoid sampled at 1 kHz with a different amplitude. The amplitude is equal to the row index.

Compute the maximum-to-minimum differences of the rows specifying the dimension equal to 2 with the `DIM` argument.

```
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t);
X = repmat(x,4,1);
amp = (1:4)';
amp = repmat(amp,1,1e3);
X = X.*amp;
Y = peak2peak(X,2);
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003.

See Also

`min` | `max` | `peak2rms`

Purpose

Peak-magnitude-to-RMS ratio

Syntax

```
Y = peak2rms(X)
Y = peak2rms(X,DIM)
```

Description

`Y = peak2rms(X)` returns the ratio of the largest absolute value in `X` to the root-mean-square (RMS) value of `X`. `peak2rms` operates along the first nonsingleton dimension of `X`. For example, if `X` is a row or column vector, `Y` is a real-valued scalar. If `Y` is an N-by-M matrix with N>1, `Y` is a 1-by-M row vector containing the peak-magnitude-to-RMS levels of the columns of `Y`.

`Y = peak2rms(X,DIM)` computes the peak-magnitude-to-RMS level of `X` along the dimension, `DIM`.

Input Arguments

X

Real– or complex-valued input vector or matrix. By default, `peak2rms` acts along the first nonsingleton dimension of `X`.

DIM

Dimension for peak-magnitude-to-RMS ratio. The optional `DIM` input argument specifies the dimension along which to compute the peak-magnitude-to-RMS level.

Default: First nonsingleton dimension

Output Arguments

Y

Peak-magnitude-to-RMS ratio. For vectors, `Y` is a real-valued scalar. For matrices, `Y` contains the peak-magnitude-to-RMS levels computed along the specified dimension, `DIM`. By default, `DIM` is the first nonsingleton dimension.

Definitions

Peak-magnitude-to-RMS Level

The peak-magnitude-to-RMS ratio is

$$\frac{||X||_\infty}{\sqrt{\frac{1}{N}\sum_{n=1}^{N}|X_n|^2}}$$

where the *l*-infinity norm and RMS values are computed along the specified dimension.

Examples

Peak-magnitude-to-RMS Ratio of Sinusoid

Compute the peak-magnitude-to-RMS ratio of a 100-Hz sinusoid sampled at 1 kHz.

```
t = 0:0.001:1-0.001;
X = cos(2*pi*100*t);
Y = peak2rms(X);
```

Peak-magnitude-to-RMS Ratio of Complex Exponential

Compute the peak-magnitude-to-RMS ratio of a complex exponential with a frequency of π/4 radians/sample.

Create a complex exponential with a frequency of π/4 radians/sample. Find the peak-magnitude-to-RMS ratio.

```
n = 0:99;
X = exp(1j*pi/4*n);
Y = peak2rms(X);
```

Peak-magnitude-to-RMS ratio of 2-D Matrix

Create a matrix where each column is a 100-Hz sinusoid sampled at 1 kHz with a different amplitude. The amplitude is equal to the column index.

Compute the peak-magnitude-to-RMS ratio of the columns.

```
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t)';
X = repmat(x,1,4);
```

```
amp = 1:4;
amp = repmat(amp,1e3,1);
X = X.*amp;
Y = peak2rms(X);
```

Peak-magnitude-to-RMS ratio of 2-D Matrix Along Specified Dimension

Create a matrix where each row is a 100-Hz sinusoid sampled at 1 kHz with a different amplitude. The amplitude is equal to the row index.

Compute the peak-magnitude-to-RMS ratio of the rows specifying the dimension equal to 2 with the `DIM` argument.

```
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t);
X = repmat(x,4,1);
amp = (1:4)';
amp = repmat(amp,1,1e3);
X = X.*amp;
Y = peak2rms(X,2);
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003.

peig

Purpose

Pseudospectrum using eigenvector method

Syntax

```
[S,w] = peig(x,p)
[S,w] = peig(x,p,w)
[S,w] = peig(...,nfft)
[S,f] = peig(x,p,nfft,fs)
[S,f] = peig(x,p,f,fs)
[S,f] = peig(...,'corr')
[S,f] = peig(x,p,nfft,fs,nwin,noverlap)
[...] = peig(...,freqrange)
[...,v,e] = peig(...)
peig(...)
```

Description

`[S,w] = peig(x,p)` implements the eigenvector spectral estimation method and returns `S`, the pseudospectrum estimate of the input signal `x`, and `w`, a vector of normalized frequencies (in rad/sample) at which the pseudospectrum is evaluated. The pseudospectrum is calculated using estimates of the eigenvectors of a correlation matrix associated with the input data `x`, where `x` is specified as either:

- A row or column vector representing one observation of the signal
- A rectangular array for which each row of `x` represents a separate observation of the signal (for example, each row is one output of an array of sensors, as in array processing), such that `x'*x` is an estimate of the correlation matrix

---

Note You can use the output of `corrmtx` to generate such an array `x`.

You can specify the second input argument `p` as either:

- A scalar integer. In this case, the signal subspace dimension is `p`.
- A two-element vector. In this case, `p(2)`, the second element of `p`, represents a threshold that is multiplied by λ_{min}, the smallest estimated eigenvalue of the signal's correlation matrix. Eigenvalues below the threshold λ_{min}`*p(2)` are assigned to the noise subspace.

In this case, `p(1)` specifies the maximum dimension of the signal subspace.

Note If the inputs to `peig` are real sinusoids, set the value of `p` to double the number of input signals. If the inputs are complex sinusoids, set `p` equal to the number of inputs.

The extra threshold parameter in the second entry in `p` provides you more flexibility and control in assigning the noise and signal subspaces.

`S` and `w` have the same length. In general, the length of the FFT and the values of the input `x` determine the length of the computed `S` and the range of the corresponding normalized frequencies. The following table indicates the length of `S` (and `w`) and the range of the corresponding normalized frequencies for this syntax.

S Characteristics for an FFT Length of 256 (Default)

Real/Complex Input Data	Length of S and w	Range of the Corresponding Normalized Frequencies
Real-valued	129	[0, π]
Complex-valued	256	[0, 2π)

`[S,w] = peig(x,p,w)` returns the pseudospectrum in the vector `S` computed at the normalized frequencies specified in vector `w`, which has two or more elements

`[S,w] = peig(...,nfft)` specifies the integer length of the FFT `nfft` used to estimate the pseudospectrum. The default value for `nfft` (entered as an empty vector `[]`) is 256.

The following table indicates the length of `S` and `w`, and the frequency range for `w` for this syntax.

S and Frequency Vector Characteristics

Real/Complex Input Data	nfft Even/Odd	Length of S and w	Range of w
Real-valued	Even	`(nfft/2 + 1)`	[0, π]
Real-valued	Odd	`(nfft + 1)/2`	[0, π)
Complex-valued	Even or odd	`nfft`	[0, 2π)

`[S,f] = peig(x,p,nfft,fs)` returns the pseudospectrum in the vector `S` evaluated at the corresponding vector of frequencies `f` (in Hz). You supply the sampling frequency `fs` in Hz. If you specify `fs` with the empty vector `[]`, the sampling frequency defaults to 1 Hz.

The frequency range for `f` depends on `nfft`, `fs`, and the values of the input `x`. The length of `S` (and `f`) is the same as in the S and Frequency Vector Characteristics on page 1-762 above. The following table indicates the frequency range for `f` for this syntax.

S and Frequency Vector Characteristics with fs Specified

Real/Complex Input Data	nfft Even/Odd	Range of f
Real-valued	Even	`[0,fs/2]`
Real-valued	Odd	`[0,fs/2)`
Complex-valued	Even or odd	`[0,fs)`

`[S,f] = peig(x,p,f,fs)` returns the pseudospectrum in the vector `S` computed at the frequencies specified in vector `f`, which has two or more elements

`[S,f] = peig(...,'corr')` forces the input argument `x` to be interpreted as a correlation matrix rather than matrix of signal data. For this syntax `x` must be a square matrix, and all of its eigenvalues must be nonnegative.

`[S,f] = peig(x,p,nfft,fs,nwin,noverlap)` allows you to specify `nwin`, a scalar integer indicating a rectangular window length, or a real-valued vector specifying window coefficients. Use the scalar integer `noverlap` in conjunction with `nwin` to specify the number of input sample points by which successive windows overlap. `noverlap` is not used if `x` is a matrix. The default value for `nwin` is `2*p(1)` and `noverlap` is `nwin-1`.

With this syntax, the input data `x` is segmented and windowed before the matrix used to estimate the correlation matrix eigenvalues is formulated. The segmentation of the data depends on `nwin`, `noverlap`, and the form of `x`. Comments on the resulting windowed segments are described in the following table.

Windowed Data Depending on x and nwin

Input data x	Form of nwin	Windowed Data
Data vector	Scalar	Length is `nwin`
Data vector	Vector of coefficients	Length is `length(nwin)`
Data matrix	Scalar	Data is not windowed.
Data matrix	Vector of coefficients	`length(nwin)` must be the same as the column length of `x`, and `noverlap` is not used.

See the table, Eigenvector Length Depending on Input Data and Syntax on page 1-765, for related information on this syntax.

Note The arguments `nwin` and `noverlap` are ignored when you include the string `'corr'` in the syntax.

`[...] = peig(...,freqrange)` specifies the range of frequency values to include in `f` or `w`. This syntax is useful when `x` is real. `freqrange` can be either:

- `'onesided'` — returns the one-sided PSD of a real input signal, `x`. If `nfft` is even, `Pxx` has length`nfft`/2+1 and is computed over the interval [0,π]. If `nfft` is odd, the length of `Pxx` is (`nfft`+1)/2 and the frequency interval is [0,π). When your specify `fs` , the intervals are [0,`fs`/2) and [0,`fs`/2] for even and odd length`nfft`respectively.
- `'twosided'` — returns the two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval [0,2π). When you specify `fs`, the frequency interval is [0,`fs`).
- `'centered'` — returns the centered two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval (-π, π] for even length `nfft` and (-π, π]) for odd length `nfft`. When you specify `fs`, the frequency intervals are (-`fs`/2, `fs`/2] and (-`fs`/2,`fs`/2) for even and odd length `nfft` respectively.

Note You can put the string arguments `freqrange` or `'corr'` anywhere in the input argument list after `p`.

`[...,v,e] = peig(...)` returns the matrix `v` of noise eigenvectors, along with the associated eigenvalues in the vector `e`. The columns of `v` span the noise subspace of dimension `size(v,2)`. The dimension of the signal subspace is `size(v,1)-size(v,2)`. For this syntax, `e` is a vector of estimated eigenvalues of the correlation matrix.

`peig(...)` with no output arguments plots the pseudospectrum in the current figure window.

Tips

In the process of estimating the pseudospectrum, `peig` computes the noise and signal subspaces from the estimated eigenvectors v_j and eigenvalues λ_j of the signal's correlation matrix. The smallest of these eigenvalues is used in conjunction with the threshold parameter `p(2)` to affect the dimension of the noise subspace in some cases.

The length n of the eigenvectors computed by `peig` is the sum of the dimensions of the signal and noise subspaces. This eigenvector length depends on your input (signal data or correlation matrix) and the syntax you use.

The following table summarizes the dependency of the eigenvector length on the input argument.

Eigenvector Length Depending on Input Data and Syntax

Form of Input Data x	Comments on the Syntax	Length n of Eigenvectors
Row or column vector	`nwin` is specified as a scalar integer.	`nwin`
Row or column vector	`nwin` is specified as a vector.	`length(nwin)`
Row or column vector	`nwin` is not specified.	`2*p(1)`
l-by-m matrix	If `nwin` is specified as a scalar, it is not used. If `nwin` is specified as a vector, `length(nwin)` must equal m.	m
m-by-m nonnegative definite matrix	The string `'corr'` is specified and `nwin` is not used.	m

You should specify `nwin` > `p(1)` or `length(nwin)` > `p(1)` if you want `p(2)` > `1` to have any effect.

Examples

Implement the eigenvector method to find the pseudospectrum of the sum of three sinusoids in noise, using the default FFT length of 256. The inputs are complex sinusoids so you set `p` equal to the number of inputs. Use the modified covariance method for the correlation matrix estimate:

```
n=0:99;
```

```
s=exp(i*pi/2*n)+2*exp(i*pi/4*n)+exp(i*pi/3*n)+randn(1,100);
X=corrmtx(s,12,'mod');
peig(X,3,'whole')            % Uses default NFFT of 256
```

Algorithms

The eigenvector method estimates the pseudospectrum from a signal or a correlation matrix using a weighted version of the MUSIC algorithm derived from Schmidt's eigenspace analysis method [1] [2]. The algorithm performs eigenspace analysis of the signal's correlation matrix in order to estimate the signal's frequency content. The eigenvalues and eigenvectors of the signal's correlation matrix are estimated using `svd` if you don't supply the correlation matrix. This algorithm is particularly suitable for signals that are the sum of sinusoids with additive white Gaussian noise.

The eigenvector method produces a pseudospectrum estimate given by

$$P_{ev}(f) = \frac{1}{\sum_{k=p+1}^{N} |\, \mathrm{v}_k^H \mathrm{e}(f) \,|^2 \,/ \lambda_k}$$

where N is the dimension of the eigenvectors and v_k is the kth eigenvector of the correlation matrix of the input signal. The integer p is the dimension of the signal subspace, so the eigenvectors v_k used in the sum correspond to the smallest eigenvalues λ_k of the correlation matrix. The eigenvectors used span the noise subspace. The vector $\mathrm{e}(f)$ consists of complex exponentials, so the inner product

$$\mathrm{v}_k^H \mathrm{e}(f)$$

amounts to a Fourier transform. This is used for computation of the pseudospectrum. The FFT is computed for each v_k and then the squared magnitudes are summed and scaled.

References

[1] Marple, S.L. *Digital Spectral Analysis*, Englewood Cliffs, NJ, Prentice-Hall, 1987, pp. 373-378.

[2] Schmidt, R.O, “Multiple Emitter Location and Signal Parameter Estimation,” *IEEE Trans. Antennas Propagation*, Vol. AP-34 (March 1986), pp.276-280.

[3] Stoica, P., and R.L. Moses, *Introduction to Spectral Analysis*, Prentice-Hall, 1997.

See Also

`corrmtx` | `dspdata` | `pburg` | `periodogram` | `pmtm` | `pmusic` | `prony` | `pwelch` | `rooteig` | `rootmusic` | `spectrum.eigenvector`

periodogram

Purpose

Periodogram power spectral density estimate

Syntax

```
pxx = periodogram(x)
pxx = periodogram(x,window)
pxx = periodogram(x,window,nfft)

[pxx,w] = periodogram( ___ )
[pxx,f] = periodogram( ___ ,fs)

[pxx,w] = periodogram(x,window,w)
[pxx,f] = periodogram(x,window,f,fs)

[ ___ ] = periodogram(x,window, ___ ,freqrange)
[ ___ ] = periodogram(x,window, ___ ,spectrumtype)

[pxx,f,pxxc] =
periodogram( ___ ,'ConfidenceLevel',probability)

periodogram( ___ )
```

Description

`pxx = periodogram(x)` returns the periodogram power spectral density (PSD) estimate of the input signal, `x`, using a rectangular window. If `x` is real-valued, `pxx` is a one-sided PSD estimate. If `x` is complex-valued, `pxx` is a two-sided PSD estimate. The number of points, `nfft`, in the discrete Fourier transform (DFT) is the maximum of 256 or the next power of two greater than the signal length.

`pxx = periodogram(x,window)` returns the modified periodogram PSD estimate using the window, `window`. `window` is a vector the same length as `x`.

`pxx = periodogram(x,window,nfft)` uses `nfft` points in the discrete Fourier transform (DFT). If `nfft` is greater than the signal length, `x` is zero-padded to length `nfft`. If `nfft` is less than the signal length, the signal is wrapped modulo `nfft` and summed using `datawrap`. For

example, the input signal `[1 2 3 4 5 6 7 8]` with `nfft` equal to 4 results in the periodogram of `sum([1 5; 2 6; 3 7; 4 8],2)`.

`[pxx,w] = periodogram( ___ )` returns the normalized frequency vector, `w`. If `pxx` is a one-sided periodogram, `w` spans the interval [0,π] if `nfft` is even and [0,π) if `nfft` is odd. If `pxx` is a two-sided periodogram, `w` spans the interval [0,2π).

`[pxx,f] = periodogram( ___ ,fs)` returns a frequency vector, `f`, in cycles per unit time. The sampling frequency, `fs`, is the number of samples per unit time. If the unit of time is seconds, then `f` is in cycles/sec (Hz). For real–valued signals, `f` spans the interval [0,`fs`/2] when `nfft` is even and [0,`fs`/2) when `nfft` is odd. For complex-valued signals, `f` spans the interval [0,`fs`).

`[pxx,w] = periodogram(x,window,w)` returns the two-sided periodogram estimates at the normalized frequencies specified in the vector, `w`. The vector, `w`, must contain at least 2 elements.

`[pxx,f] = periodogram(x,window,f,fs)` returns the two-sided periodogram estimates at the frequencies specified in the vector, `f`. The vector, `f`, must contain at least 2 elements. The frequencies in `f` are in cycles per unit time. The sampling frequency, `fs`, is the number of samples per unit time. If the unit of time is seconds, then `f` is in cycles/sec (Hz).

`[ ___ ] = periodogram(x,window, ___ ,freqrange)` returns the periodogram over the frequency range specified by `freqrange`. Valid options for `freqrange` are: `'onesided'`, `'twosided'`, or `'centered'`.

`[ ___ ] = periodogram(x,window, ___ ,spectrumtype)` returns the PSD estimate if `spectrumtype` is specified as `'psd'` and returns the power spectrum if `spectrumtype` is specified as `'power'`.

`[pxx,f,pxxc] = periodogram( ___ ,'ConfidenceLevel',probability)` returns the `probability`x100% confidence intervals for the PSD estimate in `pxxc`.

`periodogram( ___ )` with no output arguments plots the periodogram PSD estimate in dB per unit frequency in the current figure window.

Input Arguments

x - Input signal

vector

Input signal, specified as a row or column vector.

Data Types
`single | double`
Complex Number Support: Yes

window - Window

rectwin(length(x)) (default) | `[]` | vector

Window, specified as a row or column vector the same length as the input signal. If you specify `window` as empty, the default rectangular window is used.

Data Types
`single | double`

nfft - Number of DFT points

`max(256,2^nextpow2(length(x))` (default) | integer | `[]`

Number of DFT points, specified as a positive integer. For a real-valued input signal, `x`, the PSD estimate, `pxx` has length (`nfft`/2+1) if `nfft` is even, and (`nfft`+1)/2 if `nfft` is odd. For a complex-valued input signal,`x`, the PSD estimate always has length `nfft`. If `nfft` is specified as empty, the default `nfft` is used.

Data Types
`single | double`

fs - Sampling frequency

positive scalar

Sampling frequency, specified as a positive scalar. The sampling frequency is the number of samples per unit time. If the unit of time is seconds, the sampling frequency has the units hertz.

w - Normalized frequencies for Goertzel algorithm

vector

Normalized frequencies for Goertzel algorithm, specified as a row or column vector with at least 2 elements. Normalized frequencies are in radians/sample.

Example: `w = [pi/4 pi/2]`

Data Types
`double`

f - Cyclical frequencies for Goertzel algorithm

vector

Cyclical frequencies for Goertzel algorithm, specified as a row or column vector with at least 2 elements. The frequencies are in cycles per unit time. The unit time is specified by the sampling frequency, `fs`. If `fs` has units of samples/second, then `f` has units of Hz.

Example: `fs = 1000; f= [100 200]`

Data Types
`double`

freqrange - Frequency range for PSD estimate

`'onesided' | 'twosided' | 'centered'`

Frequency range for the PSD estimate, specified as a one of `'onesided'`, `'twosided'`, or `'centered'`. The default is `'onesided'` for real-valued signals and `'twosided'` for complex-valued signals. The frequency ranges corresponding to each option are

- `'onesided'` — returns the one-sided PSD estimate of a real-valued input signal, `x`. If `nfft` is even, `pxx` will have length `nfft`/2+1 and is

computed over the interval [0,π] radians/sample. If `nfft` is odd, the length of `pxx` is (`nfft`+1)/2 and the interval is [0,π) radians/sample. When `fs` is optionally specified, the corresponding intervals are [0,`fs`/2] cycles/unit time and [0,`fs`/2) cycles/unit time for even and odd length `nfft` respectively.

- `'twosided'` — returns the two-sided PSD estimate for either the real-valued or complex-valued input, `x`. In this case, `pxx` has length `nfft` and is computed over the interval [0,2π) radians/sample. When `fs` is optionally specified, the interval is [0,`fs`) cycles/unit time.
- `'centered'` — returns the centered two-sided PSD estimate for either the real-valued or complex-valued input, `x`. In this case, `pxx` has length `nfft` and is computed over the interval (-π,π] radians/sample for even length `nfft` and (-π,π) radians/sample for odd length `nfft`. When `fs` is optionally specified, the corresponding intervals are (-`fs`/2, `fs`/2] cycles/unit time and (-`fs`/2, `fs`/2) cycles/unit time for even and odd length `nfft` respectively.

Data Types
`char`

spectrumtype - Power spectrum scaling

`'psd'` (default) | `'power'`

Power spectrum scaling, specified as one of `'psd'` or `'power'`. Omitting the `spectrumtype`, or specifying `'psd'`, returns the power spectral density. Specifying `'power'` scales each estimate of the PSD by the equivalent noise bandwidth of the window. Use the `'power'` option to obtain an estimate of the power at each frequency.

Data Types
`char`

probability - Confidence interval for PSD estimate

0.95 (default) | Scalar in the range (0,1)

Coverage probability for the true PSD, specified as a scalar in the range (0,1). The output, `pxxc`, contains the lower and upper bounds of the `probability`x100% interval estimate for the true PSD.

Output Arguments

pxx - PSD estimate

vector

PSD estimate, specified as a real-valued, nonnegative column vector.

Data Types
`single | double`

w - Normalized frequencies

vector

Normalized frequencies, specified as a real-valued column vector. If `pxx` is a one-sided PSD estimate, `w` spans the interval [0,π] if `nfft` is even and [0,π) if `nfft` is odd. If `pxx` is a two-sided PSD estimate, `w` spans the interval [0,2π). For a DC-centered PSD estimate, `f` spans the interval (-π,π] radians/sample for even length `nfft` and (-π,π) radians/sample for odd length `nfft`.

Data Types
`double`

f - Cyclical frequencies

vector

Cyclical frequencies, specified as a real-valued column vector. For a one-sided PSD estimate, `f` spans the interval [0,`fs`/2] when `nfft` is even and [0,`fs`/2) when `nfft` is odd. For a two-sided PSD estimate, `f` spans the interval [0,`fs`). For a DC-centered PSD estimate, `f` spans the interval (-`fs`/2, `fs`/2] cycles/unit time for even length `nfft` and (-`fs`/2, `fs`/2) cycles/unit time for odd length `nfft` .

Data Types
`double`

pxxc - Confidence bounds

matrix

Confidence bounds, specified as an N-by-2 matrix with real-valued elements. The row dimension of the matrix is equal to the length of the PSD estimate, `pxx`. The first column contains the lower confidence

bound and the second column contains the upper confidence bound for the corresponding PSD estimates in the rows of `pxx`. The coverage probability of the confidence intervals is determined by the value of the `probability` input.

Data Types
`single | double`

Definitions

Periodogram

The periodogram is a nonparametric estimate of the power spectral density (PSD) of a wide-sense stationary random process. The periodogram is the Fourier transform of the biased estimate of the autocorrelation sequence. For a signal, x_n, sampled at `fs` samples per unit time, the periodogram is defined as

$$\hat{P}(f) = \frac{\Delta t}{N} \left| \sum_{n=0}^{N-1} x_n e^{-i2\pi fn} \right|^2 \quad -1/2\Delta t < f \le 1/2\Delta t$$

where Δt is the sampling interval. For a one-sided periodogram, the values at all frequencies except 0 and the Nyquist, 1/2Δt, are multiplied by 2 so that the total power is conserved.

If the frequencies are in radians/sample, the periodogram is defined as

$$\hat{P}(f) = \frac{1}{2\pi N} \left| \sum_{n=0}^{N-1} x_n e^{-i\omega n} \right|^2 \quad -\pi < \omega \le \pi$$

The frequency range in the preceding equations has variations depending on the value of the `freqrange` argument. See the description of `freqrange` in “Input Arguments” on page 1-770.

The integral of the true PSD, *P(f)*, over one period, *1/Δt* for cyclical frequency and 2π for normalized frequency, is equal to the variance of the wide-sense stationary random process.

$$\sigma^2 = \int_{-1/2\Delta t}^{1/2\Delta t} P(f)\, df$$

For normalized frequencies, replace the limits of integration appropriately.

Modified Periodogram

The modified periodogram multiplies the input time series by a window function. A suitable window function is nonnegative and decays to zero at the beginning and end points. Multiplying the time series by the window function *tapers* the data gradually on and off and helps to alleviate the leakage in the periodogram. See "Bias and Variability in the Periodogram" for an example.

If h_n is a window function, the modified periodogram is defined by

$$\hat{P}(f) = \frac{\Delta t}{N} \left| \sum_{n=0}^{N-1} h_n x_n e^{-i2\pi fn} \right|^2 \quad -1/2\Delta t < f \le 1/2\Delta t$$

where Δt is the sampling interval.

If the frequencies are in radians/sample, the modified periodogram is defined as

$$\hat{P}(f) = \frac{1}{2\pi N} \left| \sum_{n=0}^{N-1} h_n x_n e^{-i\omega n} \right|^2 \quad -\pi < \omega \le \pi$$

The frequency range in the preceding equations has variations depending on the value of the `freqrange` argument. See the description of `freqrange` in "Input Arguments" on page 1-770.

Examples

Periodogram Using Default Inputs

Obtain the periodogram of an input signal consisting of a discrete-time sinusoid with an angular frequency of π/4 radians/sample with additive N(0,1) white noise.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the periodogram using the default rectangular window and DFT length. The DFT length is the next power of two greater than the signal length, or 512 points. Because the signal is real-valued and has even length, the periodogram is one-sided and there are 512/2+1 points.

```
n = 0:319;
x = cos(pi/4*n)+randn(size(n));
pxx = periodogram(x);
plot(10*log10(pxx))
```

Modified Periodogram with Hamming Window

Obtain the modified periodogram of an input signal consisting of a discrete-time sinusoid with an angular frequency of π/4 radians/sample with additive N(0,1) white noise.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the modified periodogram using a Hamming window and default DFT length. The DFT length is the next power of two greater than the signal length, or 512 points. Because the signal is real-valued and has even length, the periodogram is one-sided and there are 512/2+1 points.

```
n = 0:319;
x = cos(pi/4*n)+randn(size(n));
pxx = periodogram(x,hamming(length(x)));
plot(10*log10(pxx))
```

DFT Length Equal to Signal Length

Obtain the periodogram of an input signal consisting of a discrete-time sinusoid with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. Use a DFT length equal to the signal length.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the periodogram using the default rectangular window and DFT length equal to the signal length. Because the signal is real-valued,

the one-sided periodogram is returned by default with a length equal to 320/2+1.

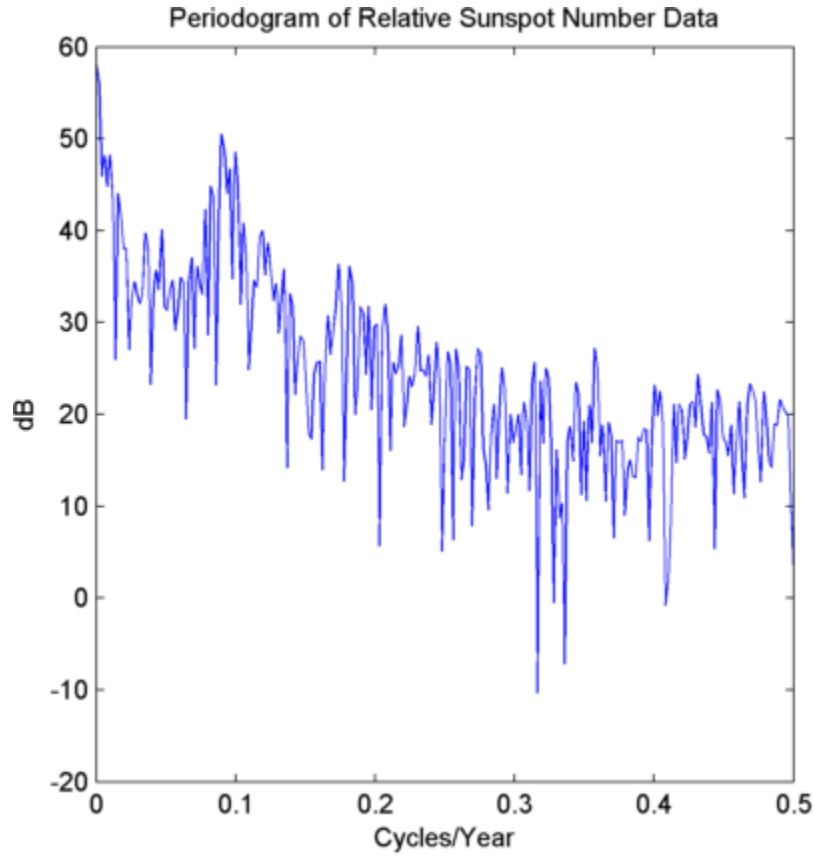
```
n = 0:319;
x = cos(pi/4*n)+randn(size(n));
nfft = length(x);
pxx = periodogram(x,[],nfft);
plot(10*log10(pxx))
```

Periodogram of Relative Sunspot Numbers

Obtain the periodogram of the Wolf (relative sunspot) number data sampled yearly between 1700 and 1987.

Load the relative sunspot number data. Obtain the periodogram using the default rectangular window and number of DFT points (512 in this example). The sampling rate for these data is 1 sample/year. Plot the periodogram.

```
load sunspot.dat
relNums=sunspot(:,2);
[pxx,f] = periodogram(relNums,[],[],1);
plot(f,10*log10(pxx))
xlabel('Cycles/Year'); ylabel('dB');
title('Periodogram of Relative Sunspot Number Data');
```



You see in the preceding figure that there is a peak in the periodogram at approximately 0.1 cycles/year, which indicates a period of approximately 10 years.

Periodogram Using Goertzel's Algorithm — Normalized Frequency

Obtain the periodogram of an input signal consisting of two discrete-time sinusoids with an angular frequencies of π/4 and π/2 radians/sample in additive N(0,1) white noise. Use Goertzel's

algorithm to obtain the two-sided periodogram estimates at π/4 and π/2 radians/sample. Compare the result to the one-sided periodogram.

```
n = 0:319;
x = cos(pi/4*n)+0.5*sin(pi/2*n)+randn(size(n));
[pxx,w] = periodogram(x,[],[pi/4 pi/2]);
pxx
[pxx1,w1] = periodogram(x);
plot(w1,pxx1)
```

You see that the periodogram values obtained using Goertzel's algorithm are 1/2 the values in the one-sided periodogram. This is consistent with the fact that using Goertzel's algorithm returns the two-sided periodogram.

Periodogram Using Goertzel's Algorithm — Frequency in Hz

Create a signal consisting of two sine waves with frequencies of 100 and 200 Hz in N(0,1) white additive noise. The sampling frequency is 1 kHz. Use Goertzel's algorithm to obtain the two-sided periodogram at 100 and 200 Hz.

```
fs = 1000;
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t)+sin(2*pi*200*t)+randn(size(t));
freq = [100 200];
[pxx,f] = periodogram(x,[],freq,fs);
```

Upper and Lower 95%-Confidence Bounds

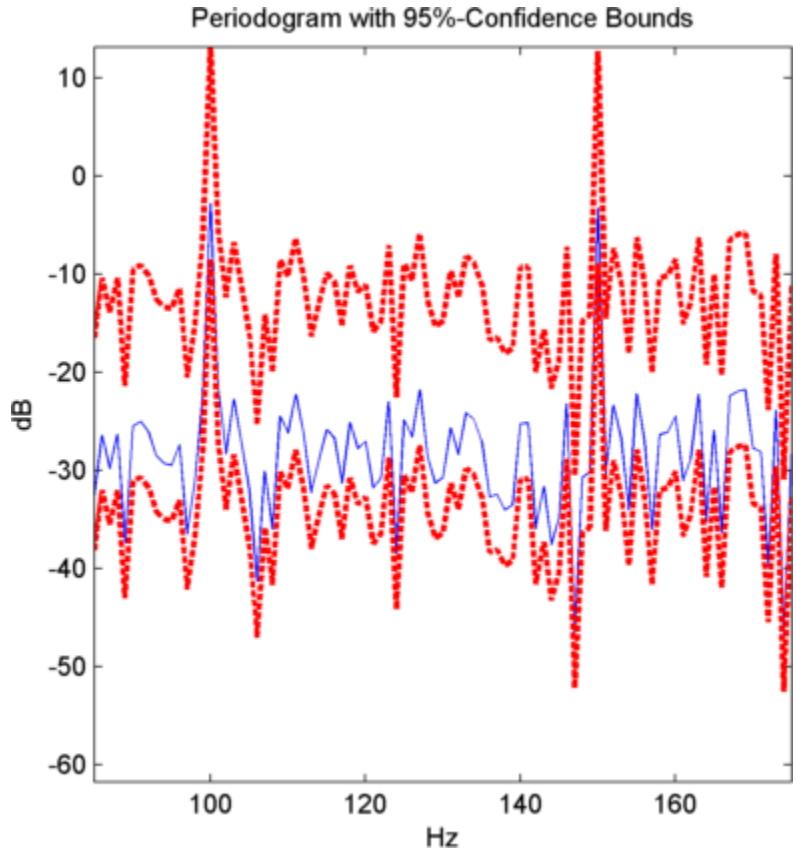
The following example illustrates the use of confidence bounds with the periodogram. While not a necessary condition for statistical significance, frequencies in the periodogram where the lower confidence bound exceeds the upper confidence bound for surrounding PSD estimates clearly indicate significant oscillations in the time series.

Create a signal consisting of the superposition of 100-Hz and 150-Hz sine waves in additive white N(0,1) noise. The amplitude of the two sine waves is 1. The sampling frequency is 1 kHz.

```
t = 0:0.001:1-0.001;
fs = 1000;
x = cos(2*pi*100*t)+sin(2*pi*150*t)+randn(size(t));
```

Obtain the periodogram with 95%-confidence bounds. Plot the periodogram along with the confidence interval and zoom in on the frequency region of interest near 100 and 150 Hz.

```
[pxx,f,pxxc] = periodogram(x,rectwin(length(x)),length(x),fs,...
'ConfidenceLevel', 0.95);
plot(f,10*log10(pxx)); hold on;
plot(f,10*log10(pxxc),'r--','linewidth',2);
axis([85 175 min(min(10*log10(pxxc))) max(max(10*log10(pxxc)))]);
xlabel('Hz'); ylabel('dB');
title('Periodogram with 95%-Confidence Bounds');
```



At 100 and 150 Hz, the lower confidence bound exceeds the upper confidence bounds for surrounding PSD estimates.

Power Estimate of Sinusoid

Estimate the power of sinusoid at a specific frequency using the `'power'` option.

Create a 100-Hz sinusoid one second in duration sampled at 1 kHz. The amplitude of the sine wave is 1.8, which equates to a power of $1.8^2/2 = 1.62$. Estimate the power using the `'power'` option.

```
fs = 1000;
t = 0:1/fs:1-1/fs;
x = 1.8*cos(2*pi*100*t);
[pxx,f] = periodogram(x,hamming(length(x)),length(x),fs,'power');
[pwrest,idx] = max(pxx);
fprintf('The maximum power occurs at %3.1f Hz\n',f(idx));
fprintf('The power estimate is %2.2f\n',pwrest);
```

DC-Centered Periodogram

Obtain the periodogram of a 100-Hz sine wave in additive N(0,1) noise. The data are sampled at 1 kHz. Use the `'centered'` option to obtain the DC-centered periodogram and plot the result.

```
fs = 1000;
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t)+randn(size(t));
[pxx,f] = periodogram(x,[],length(x),fs,'centered');
plot(f,10*log10(pxx))
xlabel('Hz'); ylabel('dB')
```

See Also

`bandpower` | `pcov` | `pburg` | `pmcov` | `pmtm` | `pwelch` | `sfdr`

Related Examples

- “Bias and Variability in the Periodogram”
- “Power Spectral Density Estimates Using FFT”

Concepts

- “Nonparametric Methods”

Purpose

Phase delay of digital filter

Syntax

```
[phi,w] = phasedelay(b,a,n)
[phi,w] = phasedelay(sos,n)
[phi,w] = phasedelay(Hd,n)
[phi,w] = phasedelay(...,n,'whole')
phi = phasedelay(...,w)
[phi,f] = phasedelay(...,n,fs)
[phi,f] = phasedelay(...,n,'whole',fs)
phi = phasedelay(...,f,fs)
[phi,w,s] = phasedelay(...)
[phi,f,s] = phasedelay(...)
phasedelay(...)
```

Description

`[phi,w] = phasedelay(b,a,n)` returns the `n`-point phase delay response vector `phi` and the `n`-point frequency reponse vector `w` (in radians/sample) of the filter defined by numerator coefficients `b` and denominator coefficients `a`. The phase delay response is evaluated at `n` equally spaced points around the upper half of the unit circle. If `n` is omitted, it defaults to 512.

`[phi,w] = phasedelay(sos,n)` returns the `n`-point phase delay response for the second order sections matrix, `sos`. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. If the number of sections is less than 2, `phasedelay` considers the input to be the numerator vector, `b`. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`[phi,w] = phasedelay(Hd,n)` returns the `n`-point phase delay response for the `dfilt` filter object, `Hd`, or the array of `dfilt` filter objects. If `Hd` is an array of `dfilt` objects, each column of `phi` is the step response of the corresponding `dfilt` object.

`[phi,w] = phasedelay(...,n,'whole')` uses `n` equally spaced points around the whole unit circle.

`phi = phasedelay(...,w)` returns the phase delay response at frequencies specified in vector `w` (in radians/sample). The frequencies are normally between 0 and π.`w` must contain at least two elements.

`[phi,f] = phasedelay(...,n,fs)` and `[phi,f] = phasedelay(...,n,'whole',fs)` return the phase delay vector `f` (in Hz), using the sampling frequency `fs` (in Hz). `f` must contain at least two elements.

`phi = phasedelay(...,f,fs)` returns the phase delay response at the frequencies specified in vector `f` (in Hz), using the sampling frequency `fs` (in Hz)..

`[phi,w,s] = phasedelay(...)` and `[phi,f,s] = phasedelay(...)` return plotting information, where `s` is a structure with fields you can change to display different frequency response plots.

`phasedelay(...)` with no output arguments plots the phase delay response of the filter. If you input the filter coefficients or second order sections matrix, the current figure window is used. If you input a `dfilt` object or array of filter objects, `fvtool` is used to plot the phase delay response.

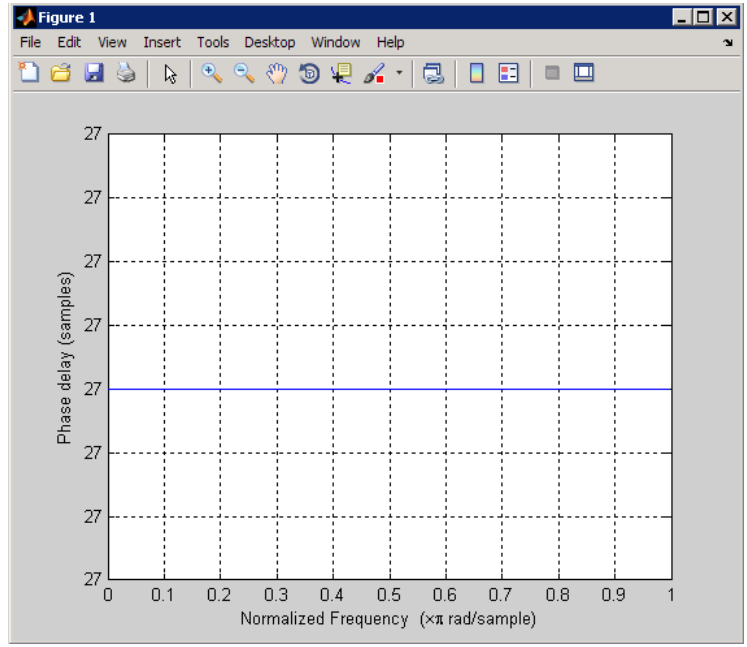
Note If the input to `phasedelay` is single precision, the phase delay response is calculated using single-precision arithmetic. The output, `phi`, is single precision.

Examples

Example 1

Plot the phase delay response of a constrained least squares FIR filter:

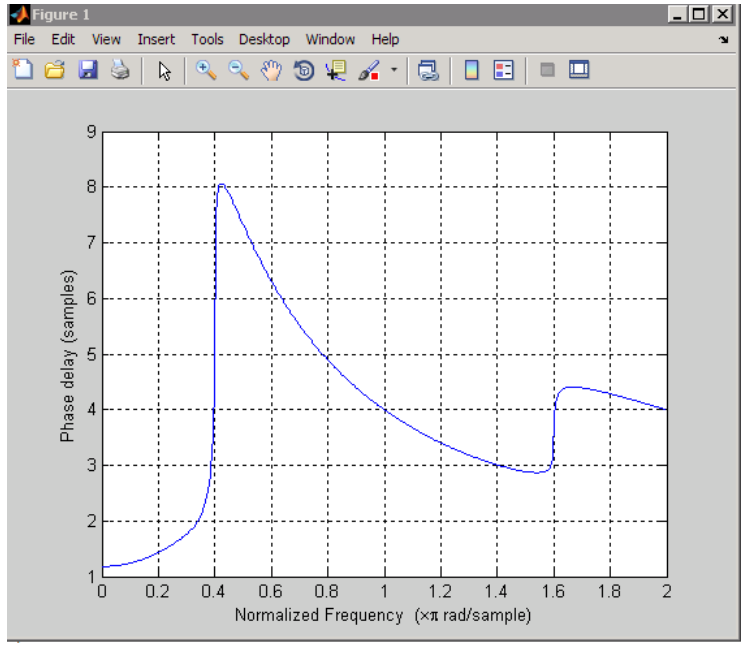
```
b=fircls1(54,.3,.02,.008);
phasedelay(b)
```



Example 2

Plot the phase delay response of an elliptic filter:

```
[b,a] = ellip(10,.5,20,.4);
phasedelay(b,a,512,'whole')
```



See Also

`freqz` | `fvtool` | `phasez` | `grpdelay`

Purpose

Phase response of digital filter

Syntax

```
[phi,w] = phasez(b,a,n)
[phi,w] = phasez(sos,n)
[phi,w]=phasez(Hd,n)
[phi,w] = phasez(...,n,'whole')
phi = phasez(...,w)
[phi,f] = phasez(...,n,fs)
phi = phasez(...f,fs)
[phi,w,s] = phasez(...)
phasez(...)
```

Description

`[phi,w] = phasez(b,a,n)` returns the `n`-point unwrapped phase response vector, `phi`, in radians and frequency vector, `w`, in radians/sample for the filter coefficients specified in `b` and `a`. The values of the frequency vector, `w`, range from 0 to `pi`. If `n` is omitted, the length of the phase response vector defaults to 512.

`[phi,w] = phasez(sos,n)` returns the unwrapped phase response for the second order sections matrix, `sos`. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. If the number of sections is less than 2, `phasez` considers the input to be the numerator vector, `b`. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`[phi,w]=phasez(Hd,n)` returns the unwrapped phase response for the `dfilt` filter object, `Hd`, or the array of `dfilt` filter objects. If `Hd` is an array of `dfilt` objects, each column of `phi` is the group delay of the corresponding `dfilt` object. If `n` is unspecified for discrete-time filter objects, the length of the phase response vector defaults to 8192.

`[phi,w] = phasez(...,n,'whole')` returns frequency and unwrapped phase response vectors evaluated at `n` equally-spaced points around the unit circle from 0 to `2*pi` radians/sample.

`phi = phasez(...,w)` returns the unwrapped phase response in radians at frequencies specified in `w` (radians/sample). The frequencies

are normally between 0 and pi.The vector w must have at least two elements.

[phi,f] = phasez(...,n,fs) return the unwrapped phase vector phi in radians and the frequency vector in Hz. The frequency vector ranges from 0 to the Nyquist frequency, fs/2. If the 'whole' option is used, the frequency vector ranges from 0 to the sampling frequency.

phi = phasez(...f,fs) return the phase response in radians at the frequencies specified in the vector f (in Hz) using the sampling frequency fs (in Hz). The vector f must have at least two elements.

[phi,w,s] = phasez(...) return plotting information, where s is a structure array with fields you can change to display different frequency response plots.

phasez(...) with no output arguments plots the phase response of the filter. If you input the filter coefficients or second order sections matrix, the current figure window is used. If you input a discrete-time filter object or array of filter objects, fvtool is used to plot the phase response.

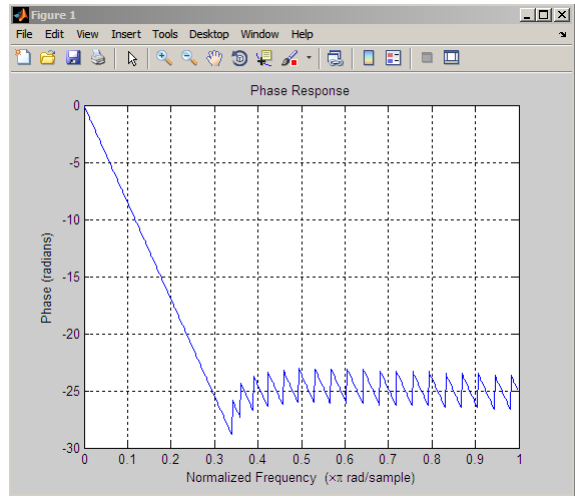
Note If the input to phasez is single precision, the phase response is calculated using single-precision arithmetic. The output, phi, is single precision.

Examples

Example 1

Plot the phase response of a constrained least squares FIR filter:

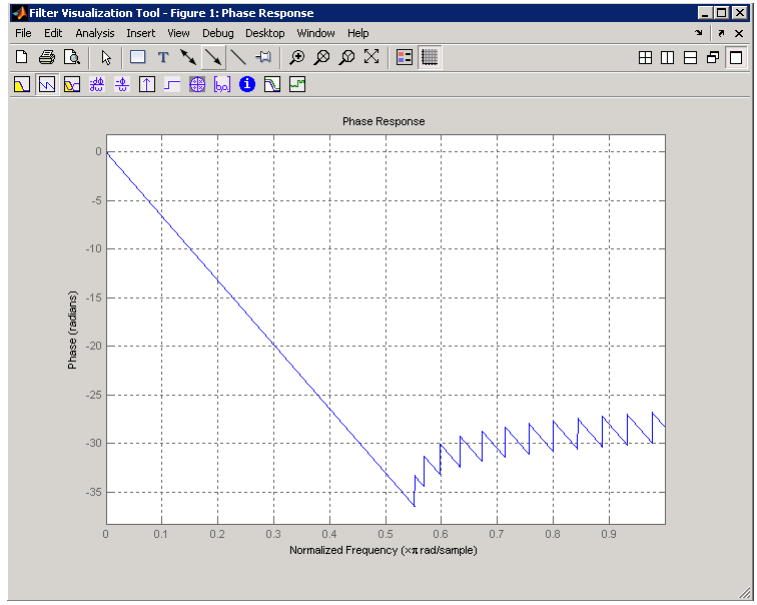
```
b=fircls1(54,.3,.02,.008);
phasez(b)
```



Example 2

In the next example, we design an equiripple lowpass default filter object and display the result:

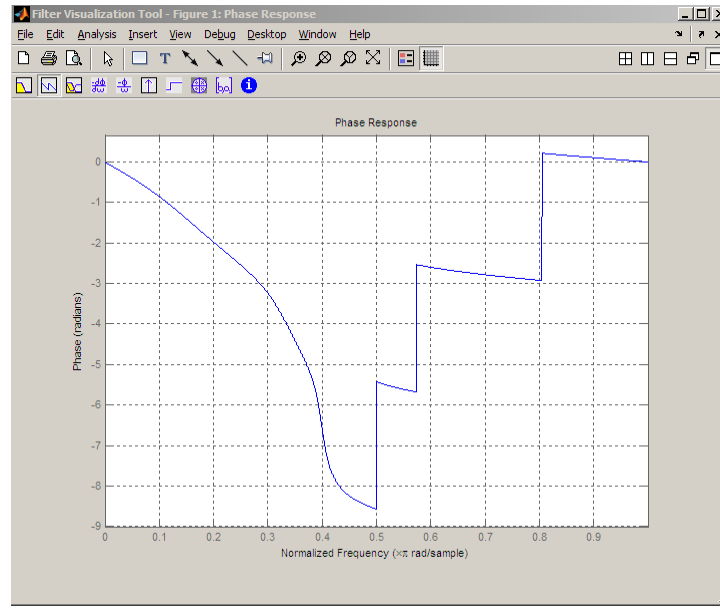
```
d=fdesign.lowpass;
Hd=design(d,'equiripple');
phasez(Hd)
```



Example 3

Plot the phase response of an elliptic filter:

```
d=fdesign.lowpass('Fp,Fst,Ap,Ast',0.4,0.5,1,60);
Hd=design(d,'ellip');
phasez(Hd)
```



See Also

`freqz` | `fvtool` | `phasedelay` | `grpdelay`

Purpose

PSD using modified covariance method

Syntax

```
Pxx = pmcov(x,order)
Pxx = pmcov(x,order,nfft)
[Pxx,w] = pmcov(...)
[Pxx,w] = pmcov(x,order,w)
Pxx = pmcov(x,order,nfft,fs)
Pxx = pmcov(x,order,f,fs)
[Pxx,f] = pmcov(x,order,nfft,fs)
[Pxx,f] = pmcov(x,order,f,fs)
[Pxx,f] = pmcov(x,order,nfft,fs,freqrange)
[Pxx,w] = pmcov(x,order,nfft,freqrange)
[Pxx,f,Pxxc] = pmcov(...,'ConfidenceLevel',P)
pmcov(...)
```

Description

`Pxx = pmcov(x,order)` implements the modified covariance algorithm, a parametric spectral estimation method, and returns `Pxx`, an estimate of the power spectral density (PSD) of the vector `x`. The entries of `x` represent samples of a discrete-time signal, and `order` is the integer specifying the order of an autoregressive (AR) prediction model for the signal, used in estimating the PSD.

The power spectral density is calculated in units of power per radians per sample. Real-valued inputs produce full power one-sided (in frequency) PSDs (by default), while complex-valued inputs produce two-sided PSDs.

In general, the length of the FFT and the values of the input `x` determine the length of `Pxx` and the range of the corresponding normalized frequencies. For this syntax, the (default) FFT length is 256. The following table indicates the length of `Pxx` and the range of the corresponding normalized frequencies for this syntax.

PSD Vector Characteristics for an FFT Length of 256 (Default)

Real/Complex Input Data	Length of Pxx	Range of the Corresponding Normalized Frequencies
Real-valued	129	[0, π]
Complex-valued	256	[0, 2π)

`Pxx = pmcov(x,order,nfft)` uses the integer FFT length `nfft` to calculate the PSD vector `Pxx`.

`[Pxx,w] = pmcov(...)` also returns `w`, a vector of normalized angular frequencies at which the two-sided PSD is estimated. `Pxx` and `w` have the same length. The units for `w` are rad/sample.

The length of `Pxx` and the frequency range for `w` depend on `nfft` and the values of the input `x`. The following table indicates the length of `Pxx` and the frequency range for `w` in this syntax.

PSD and Frequency Vector Characteristics

Real/Complex Input Data	nfft Even/Odd	Length of Pxx	Range of w
Real-valued	Even	`(nfft/2 + 1)`	[0, π]
Real-valued	Odd	`(nfft + 1)/2`	[0, π)
Complex-valued	Even or odd	`nfft`	[0, 2π)

`[Pxx,w] = pmcov(x,order,w)` uses a vector of normalized frequencies `w` with two or more elements to compute the PSD at those frequencies and returns a two-sided PSD.

```
Pxx = pmcov(x,order,nfft,fs)
```

or

`Pxx = pmcov(x,order,f,fs)` uses the integer FFT length `nfft` to calculate the PSD vector `Pxx` or uses the vector of frequencies `f` in Hz

and the sampling frequency `fs` to compute the two-sided PSD vector `Pxx` at those frequencies. If you specify `nfft` as the empty vector `[]`, it uses the default value of 256. If you specify `fs` as the empty vector `[]`, the sampling frequency `fs` defaults to 1 Hz. The spectral density produced is calculated in units of power per Hz.

```
[Pxx,f] = pmcov(x,order,nfft,fs)
```

or

`[Pxx,f] = pmcov(x,order,f,fs)` returns the frequency vector f. In this case, the units for the frequency vector are in Hz. The frequency range for `f` depends on `nfft`, `fs`, and the values of the input `x`. The length of `Pxx` is the same as in the table above. The following table indicates the frequency range for `f` for this syntax.

PSD and Frequency Vector Characteristics with fs Specified

Real/Complex Input Data	nfft Even/Odd	Range of f
Real-valued	Even	`[0,fs/2]`
Real-valued	Odd	`[0,fs/2)`
Complex-valued	Even or odd	`[0,fs)`

`[Pxx,f] = pmcov(x,order,nfft,fs,freqrange)` or

`[Pxx,w] = pmcov(x,order,nfft,freqrange)` specifies the range of frequency values to include in the output frequency vectors, `f` or `w`. This syntax is useful when `x` is real. `freqrange` can be either:

- `'onesided'` — returns the one-sided PSD of a real input signal, `x`. If `nfft` is even, `Pxx` has length `nfft/2+1` and is computed over the interval [0,π]. If `nfft` is odd, the length of `Pxx` is `(nfft+1)/2` and the frequency interval is [0,π). When your specify `fs` , the intervals are [0,`fs`/2] and [0,`fs`/2) for even and odd length `nfft` respectively.
- `'twosided'` — returns the two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval [0,2π). When you specify `fs`, the frequency interval is [0,`fs`).

- `'centered'` — returns the centered two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval (-π, π] for even length `nfft` and (-π, π) for odd length `nfft`. When you specify `fs`, the frequency intervals are (-`fs`/2, `fs`/2] and (-`fs`/2,`fs`/2) for even and odd length `nfft` respectively.

`[Pxx,f,Pxxc] = pmcov(...,'ConfidenceLevel',P)` returns the P100% confidence interval for `Pxx`, where `P` is a nonnegative scalar between 0 and 1. The default value for `P` is 0.95. Large-sample confidence intervals are computed using a Gaussian probability density function. `Pxxc` is N-by-2 matrix, where N is the length of `Pxx`. The first column, `Pxxc(:,1)`, is the lower bound of the confidence interval. The second column, `Pxxc(:,2)`, is the upper bound. See [1] for a description of approximate large-sample confidence intervals for AR PSD estimates.

`pmcov(...)` with no outputs plots the PSD in the current figure window. The frequency range on the plot is the same as the range of output `w` (or `f`) for a given set of parameters.

Tips

The power spectral density is computed as the distribution of power per unit frequency.

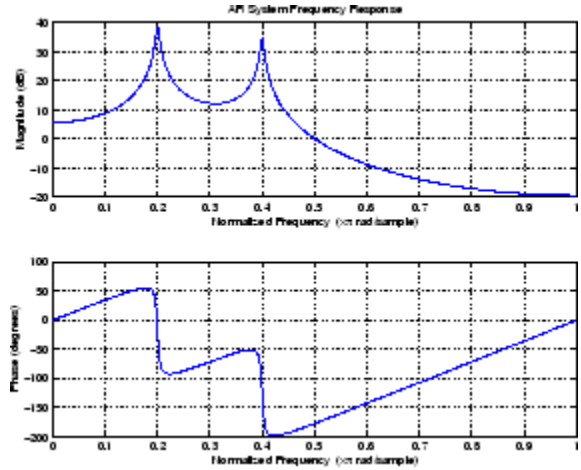
This algorithm depends on your selecting an appropriate model order for your signal.

Examples

Modified Covariance AR PSD Estimate

Because the modified covariance method estimates the spectral density by fitting an AR prediction model of a given order to the signal, first generate a signal from an AR (all-pole) model of a given order. You can use `freqz` to check the magnitude of the frequency response of your AR filter. This will give you an idea of what to expect when you estimate the PSD using `pmcov`:

```
a = [1 -2.2137 2.9403 -2.1697 0.9606];    % AR filter coefficients
freqz(1,a)                                % AR filter frequency response
title('AR System Frequency Response')
```



Now generate the input signal `x` by filtering white noise through the AR filter. Estimate the PSD of `x` based on a fourth-order AR prediction model since in this case we know that the original AR system model `a` has order 4:

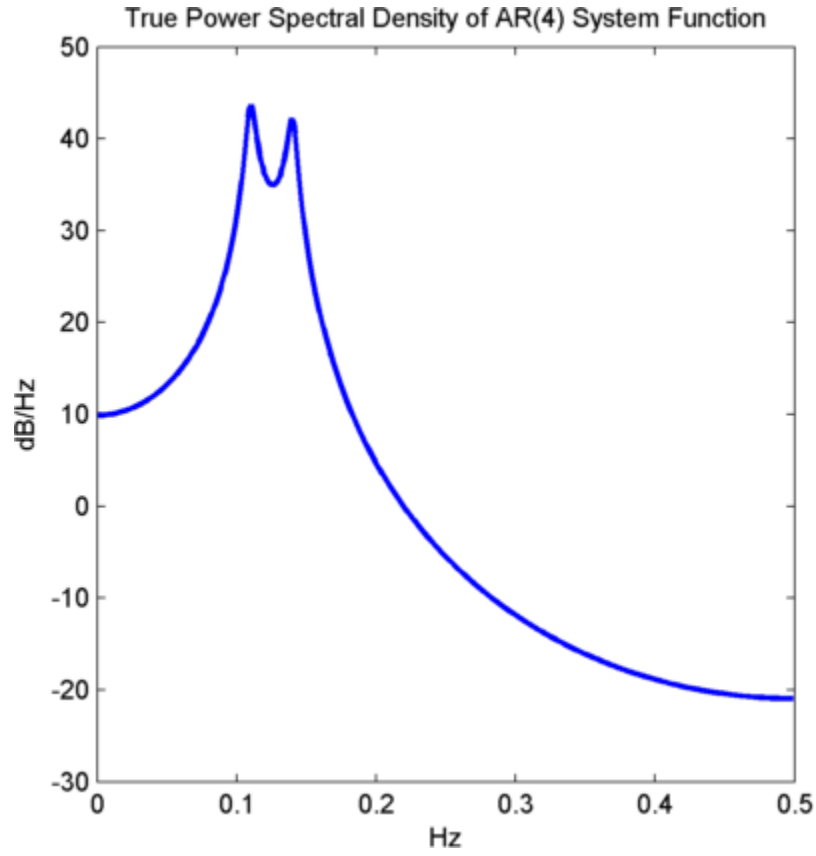
```
x = filter(1,a,randn(256,1));     % AR filter output
pmcov(x,4)                        % Fourth-order estimate
```

Large-Sample Confidence Intervals for AR PSD Estimate

This example shows you how to obtain and plot confidence intervals for an AR PSD estimate.

Create the coefficients for an AR(4) system function. Use `freqz` to obtain and plot the true power spectral density.

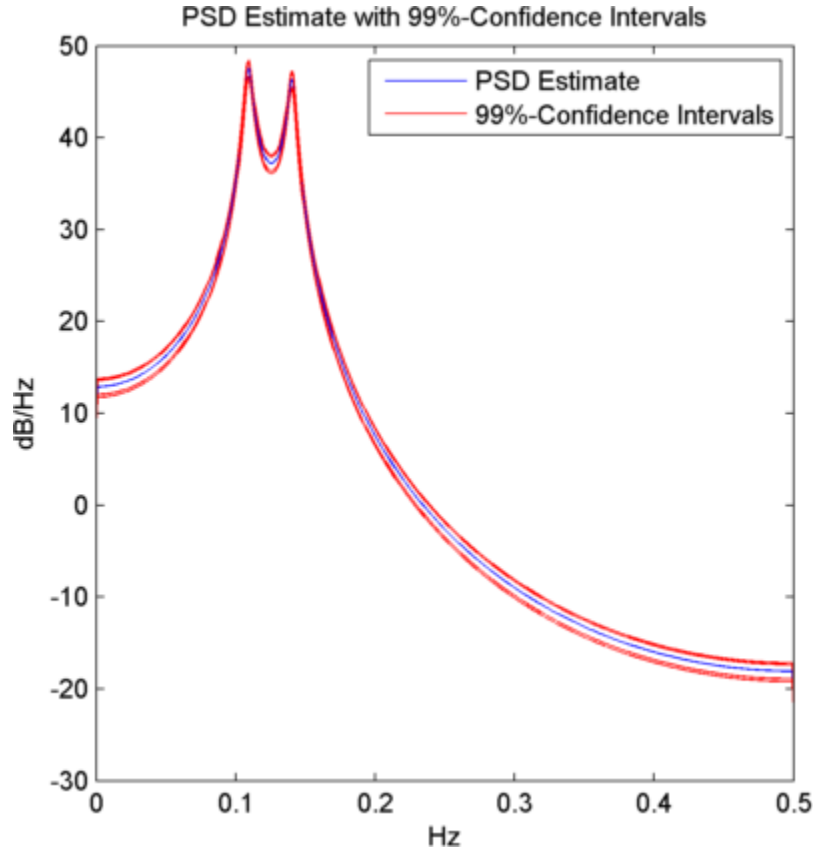
```
A = [1 -2.7607 3.8106 -2.6535 0.9238];
[H,F] = freqz(1,A,[],1);
plot(F,20*log10(abs(H)),'b','linewidth',2);
xlabel('Hz'); ylabel('dB/Hz');
title('True Power Spectral Density of AR(4) System Function')
```



sCreate a realization of the AR(4) process represented by the coefficients. Set the random number generator to the default settings for reproducible results. Obtain approximate large-sample 99%-confidence intervals for the PSD estimate.

```
rng default;
x = randn(1000,1);
y = filter(1,A,x);
[Pxx,F,Pxxc] = pmcov(y,4,1024,1,'ConfidenceLevel',0.99);
plot(F,10*log10(Pxx),'b'); hold on;
```

```
plot(F,10*log10(Pxxc),'r'); xlabel('Hz'); ylabel('dB/Hz');
legend('PSD Estimate', '99%-Confidence Intervals')
title('PSD Estimate with 99%-Confidence Intervals')
```



Algorithms

Linear prediction filters can be used to model the second-order statistical characteristics of a signal. The prediction filter output can be used to model the signal when the input is white noise.

`pmcov` estimates the PSD of the signal vector using the modified covariance method. This method fits an autoregressive (AR) linear

prediction filter model to the signal by simultaneously minimizing the forward and backward prediction errors (based on causal observations of your input signal) in the least squares sense. The spectral estimate returned by `pmcov` is the magnitude squared frequency response of this AR model.

References

[1] Kay, S.M. *Modern Spectral Estimation*, Englewood Cliffs, NJ, Prentice-Hall, 1988, pp. 194–195.

[2] Marple, S.L. *Digital Spectral Analysis*, Englewood Cliffs, NJ, Prentice-Hall, 1987, Chapter 7.

[3] Stoica, P., and R.L. Moses, *Introduction to Spectral Analysis*, Prentice-Hall, 1997.

See Also

`armcov` | `lpc` | `pburg` | `pcov` | `peig` | `periodogram` | `pmtm` | `pmusic` | `pwelch` | `prony` | `pmcov`

Purpose

Multitaper power spectral density estimate

Syntax

```
pxx = pmtm(x)
pxx = pmtm(x,nw)
pxx = pmtm(x,nw,nfft)

[pxx,w] = pmtm(___)
[pxx,f] = pmtm(___,fs)

[pxx,w] = pmtm(x,nw,w)
[pxx,f] = pmtm(x,nw,f,fs)

[___] = pmtm(___,method)

[___] = pmtm(x,e,v)
[___] = pmtm(x,dpss_params)

[___] = pmtm(___,'DropLastTaper',dropflag)
[___] = pmtm(___,freqrange)
[pxx,f,pxxc] = pmtm(___,'ConfidenceLevel',probability)

pmtm(___)
```

Description

`pxx = pmtm(x)` returns Thomson's multitaper power spectral density (PSD) estimate of the input signal, `x`. The tapers are the discrete prolate spheroidal (DPSS), or Slepian, sequences. The time-halfbandwidth, `nw`, product is 4. By default, `pmtm` uses the first 2`nw`–1 DPSS sequences. If `x` is real-valued, `pxx` is a one-sided PSD estimate. If `x` is complex-valued, `pxx` is a two-sided PSD estimate. The number of points, `nfft`, in the discrete Fourier transform (DFT) is the maximum of 256 or the next power of two greater than the signal length.

`pxx = pmtm(x,nw)` use the time-halfbandwidth product, `nw`, to obtain the multitaper PSD estimate. The time-halfbandwidth product controls the frequency resolution of the multitaper estimate. `pmtm` uses 2`nw`-1 Slepian tapers in the PSD estimate.

`pxx = pmtm(x,nw,nfft)` uses `nfft` points in the DFT. If `nfft` is greater than the signal length, `x` is zero-padded to length `nfft`. If `nfft` is less than the signal length, the signal is wrapped modulo `nfft` .

`[pxx,w] = pmtm( ___ )` returns the normalized frequency vector, `w`. If `pxx` is a one-sided PSD estimate, `w` spans the interval [0,π] if `nfft` is even and [0,π) if `nfft` is odd. If `pxx` is a two-sided PSD estimate, `w` spans the interval [0,2π).

`[pxx,f] = pmtm( ___ ,fs)` returns a frequency vector, `f`, in cycles per unit time. The sampling frequency, `fs`, is the number of samples per unit time. If the unit of time is seconds, then `f` is in cycles/sec (Hz). For real–valued signals, `f` spans the interval [0,`fs`/2] when `nfft` is even and [0,`fs`/2) when `nfft` is odd. For complex-valued signals, `f` spans the interval [0,`fs`).

`[pxx,w] = pmtm(x,nw,w)` returns the two-sided multitaper PSD estimates at the normalized frequencies specified in the vector, `w`. The vector, `w`, must contain at least 2 elements.

`[pxx,f] = pmtm(x,nw,f,fs)` returns the two-sided multitaper PSD estimates at the frequencies specified in the vector, `f`. The vector, `f`, must contain at least 2 elements. The frequencies in `f` are in cycles per unit time. The sampling frequency, `fs`, is the number of samples per unit time. If the unit of time is seconds, then `f` is in cycles/sec (Hz).

`[ ___ ] = pmtm( ___ ,method)` combines the individual tapered PSD estimates using the method, `method`. `method` can be one of: `'adapt'` (default), `'eigen'`, or `'unity'`.

`[ ___ ] = pmtm(x,e,v)` uses the tapers in the N-by-K matrix `e` with concentrations `v` in the frequency band [-*w*,*w*]. N is the length of the input signal, `x`. Use `dpss` to obtain the Slepian tapers and corresponding concentrations.

`[ ___ ] = pmtm(x,dpss_params)` uses the cell array, `dpss_params`, to pass input arguments to `dpss` except the number of elements in the sequences. The number of elements in the sequences is the first input argument to `dpss` and is not included in `dpss_params`. For example

```
x = randn(1000,1);
pxx = pmtm(x,{2.5,3});
```

`[ ___ ] = pmtm( ___ ,'DropLastTaper',dropflag)` specifies whether `pmtm` drops the last taper in the computation of the multitaper PSD estimate. `dropflag` is a logical. The default value of `dropflag` is `true` and the last taper is not used in the PSD estimate.

`[ ___ ] = pmtm( ___ ,freqrange)` returns the multitaper PSD estimate over the frequency range specified by `freqrange`. Valid options for `freqrange` are

: `'onesided'`, `'twosided'`, or `'centered'`.

`[pxx,f,pxxc] = pmtm( ___ ,'ConfidenceLevel',probability)` returns the `probability`x100% confidence intervals for the PSD estimate in `pxxc`.

`pmtm( ___ )` with no output arguments plots the multitaper PSD estimate in the current figure window.

Input Arguments

x - Input signal

vector

Input signal, specified as a row or column vector.

Data Types
`single | double`
Complex Number Support: Yes

nw - Time-halfbandwidth product

4 (default) | positive scalar

Time-halfbandwidth product, specified as a positive scalar. In multitaper spectral estimation, the user specifies the resolution bandwidth of the multitaper estimate [-*W*,*W*] where *W*=*j*/NΔt for some small *j*>1. Equivalently, *W* is some small multiple of the frequency resolution of the DFT. The time-halfbandwidth product is the product of the resolution halfbandwidth and the number of samples in the input signal, *N*. The number of Slepian tapers whose Fourier transforms are well-concentrated in [-*W*,*W*] (eigenvalues close to unity) is 2*NW*-1.

nfft - Number of DFT points

`max(256,2^nextpow2(length(x))` (default) | integer | `[]`

Number of DFT points, specified as a positive integer. For a real-valued input signal, `x`, the PSD estimate, `pxx` has length (`nfft`/2+1) if `nfft` is even, and (`nfft`+1)/2 if `nfft` is odd. For a complex-valued input signal,`x`, the PSD estimate always has length `nfft`. If `nfft` is specified as empty, the default `nfft` is used.

Data Types
`single | double`

fs - Sampling frequency

positive scalar

Sampling frequency, specified as a positive scalar. The sampling frequency is the number of samples per unit time. If the unit of time is seconds, the sampling frequency has the units hertz.

w - Normalized frequencies for Goertzel algorithm

vector

Normalized frequencies for Goertzel algorithm, specified as a row or column vector with at least 2 elements. Normalized frequencies are in radians/sample.

Example: `w = [pi/4 pi/2]`

Data Types
`double`

f - Cyclical frequencies for Goertzel algorithm

vector

Cyclical frequencies for Goertzel algorithm, specified as a row or column vector with at least 2 elements. The frequencies are in cycles per unit time. The unit time is specified by the sampling frequency, `fs`. If `fs` has units of samples/second, then `f` has units of Hz.

Example: `fs = 1000; f= [100 200]`

Data Types
`double`

method - Weights on individual tapered PSD estimates

`'adapt'` (default) | `'eigen'` | `'unity'`

Weights on individual tapered PSD estimates, specified as one of `'adapt'`, `'eigen'`, or `'unity'`. The default is Thomson's adaptive frequency-dependent weights, `'adapt'`. The calculation of these weights is detailed on pp. 368–370 in [1]. The `'eigen'` method weights each tapered PSD estimate by the eigenvalue (frequency concentration) of the corresponding Slepian taper. The `'unity'` method weights each tapered PSD estimate equally.

e - DPSS (Slepian) sequences

matrix

DPSS (Slepian) sequences, specified as a N-by-K matrix where N is the length of the input signal, `x`. The matrix `e` is the output of `dpss`.

v - Eigenvalues for DPSS (Slepian) sequences

vector

Eigenvalues for DPSS (Slepian) sequences, specified as a column vector. The eigenvalues for the DPSS sequences indicate the proportion of the sequence energy concentrated in the resolution bandwidth, [-*W*, *W*]. The eigenvalues range lie in the interval (0,1) and generally the first 2*NW*-1 eigenvalues are close to 1 and then decrease toward 0.

dpss_params - Input arguments for `dpss`

cell array

Input arguments for `dpss`, specified as a cell array. The first input argument to `dpss` is the length of the DPSS sequences and is omitted from `dpss_params`. The length of the DPSS sequences is obtained from the length of the input signal, `x`.

Example: `{3.5,5}`

dropflag - Flag indicating whether to drop or keep the last DPSS sequence

`true` (default) | `false`

Flag indicating whether to drop or keep the last DPSS sequence, specified as a logical. The default is `true` and `pmtm` drops the last taper. In a multitaper estimate, the first 2*NW*-1 DPSS sequences have eigenvalues close to unity. If you use less than 2*NW*-1 sequences, it is likely that all the tapers have eigenvalues close to 1 and you can specify `dropflag` as `false` to keep the last taper.

freqrange - Frequency range for PSD estimate

`'onesided'` | `'twosided'` | `'centered'`

Frequency range for the PSD estimate, specified as a one of `'onesided'`, `'twosided'`, or `'centered'`. The default is `'onesided'` for real-valued signals and `'twosided'` for complex-valued signals. The frequency ranges corresponding to each option are

- `'onesided'` — returns the one-sided PSD estimate of a real-valued input signal, `x`. If `nfft` is even, `pxx` will have length `nfft`/2+1 and is computed over the interval [0,π] radians/sample. If `nfft` is odd, the length of `pxx` is (`nfft`+1)/2 and the interval is [0,π) radians/sample. When `fs` is optionally specified, the corresponding intervals are [0,`fs`/2] cycles/unit time and [0,`fs`/2) cycles/unit time for even and odd length `nfft` respectively.
- `'twosided'` — returns the two-sided PSD estimate for either the real-valued or complex-valued input, `x`. In this case, `pxx` has length `nfft` and is computed over the interval [0,2π) radians/sample. When `fs` is optionally specified, the interval is [0,`fs`) cycles/unit time.

- `'centered'` — returns the centered two-sided PSD estimate for either the real-valued or complex-valued input, `x`. In this case, `pxx` has length `nfft` and is computed over the interval (-π,π] radians/sample for even length `nfft` and (-π,π) radians/sample for odd length `nfft`. When `fs` is optionally specified, the corresponding intervals are (-`fs`/2, `fs`/2] cycles/unit time and (-`fs`/2, `fs`/2) cycles/unit time for even and odd length `nfft` respectively.

Data Types
`char`

probability - Confidence interval for PSD estimate

0.95 (default) | Scalar in the range (0,1)

Coverage probability for the true PSD, specified as a scalar in the range (0,1). The output, `pxxc`, contains the lower and upper bounds of the `probability`x100% interval estimate for the true PSD.

Output Arguments

pxx - PSD estimate

vector

PSD estimate, specified as a real-valued, nonnegative column vector.

Data Types
`single | double`

w - Normalized frequencies

vector

Normalized frequencies, specified as a real-valued column vector. If `pxx` is a one-sided PSD estimate, `w` spans the interval [0,π] if `nfft` is even and [0,π) if `nfft` is odd. If `pxx` is a two-sided PSD estimate, `w` spans the interval [0,2π). For a DC-centered PSD estimate, `f` spans the interval (-π,π] radians/sample for even length `nfft` and (-π,π) radians/sample for odd length `nfft`.

Data Types
`double`

f - Cyclical frequencies

vector

Cyclical frequencies, specified as a real-valued column vector. For a one-sided PSD estimate, `f` spans the interval [0,`fs`/2] when `nfft` is even and [0,`fs`/2) when `nfft` is odd. For a two-sided PSD estimate, `f` spans the interval [0,`fs`). For a DC-centered PSD estimate, `f` spans the interval (-`fs`/2, `fs`/2] cycles/unit time for even length `nfft` and (-`fs`/2, `fs`/2) cycles/unit time for odd length `nfft` .

Data Types
`double`

pxxc - Confidence bounds

matrix

Confidence bounds, specified as an N-by-2 matrix with real-valued elements. The row dimension of the matrix is equal to the length of the PSD estimate, `pxx`. The first column contains the lower confidence bound and the second column contains the upper confidence bound for the corresponding PSD estimates in the rows of `pxx`. The coverage probability of the confidence intervals is determined by the value of the `probability` input.

Data Types
`single | double`

Examples

Multitaper Estimate Using Default Inputs

Obtain the multitaper PSD estimate of an input signal consisting of a discrete-time sinusoid with an angular frequency of π/4 radians/sample with additive N(0,1) white noise.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the multitaper PSD estimate using the default time-halfbandwidth product of 4 and DFT length. The default number of DFT points is 512. Because the signal is real-valued, the PSD estimate is one-sided and there are 512/2+1 points in the PSD estimate.

```
n = 0:319;
```

```
x = cos(pi/4*n)+randn(size(n));
pxx = pmtm(x);
plot(10*log10(pxx))
```

Specify Time-Halfbandwidth Product

Obtain the multitaper PSD estimate with a specified time-halfbandwidth product.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the multitaper PSD estimate with a time–halfbandwidth product of 2.5. The resolution bandwidth is [-2.5π/320,2.5π/320] radians/sample. The default number of DFT points is 512. Because the signal is real-valued, the PSD estimate is one-sided and there are 512/2+1 points in the PSD estimate.

```
n = 0:319;
x = cos(pi/4*n)+randn(size(n));
pxx = pmtm(x,2.5);
plot(10*log10(pxx))
```

DFT Length Equal to Signal Length

Obtain the multitaper PSD estimate of an input signal consisting of a discrete-time sinusoid with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. Use a DFT length equal to the signal length.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the multitaper PSD estimate with a time-halfbandwidth product of 3 and a DFT length equal to the signal length. Because the signal is real-valued, the one-sided PSD estimate is returned by default with a length equal to 320/2+1.

```
n = 0:319;
x = cos(pi/4*n)+randn(size(n));
pxx = pmtm(x,3,length(x));
plot(10*log10(pxx))
```

Multitaper Estimate with Sampling Frequency

Obtain the multitaper PSD estimate of a signal sampled at 1 kHz. The signal is a 100-Hz sine wave in additive N(0,1) white noise. The signal duration is 2 seconds. Use a time-halfbandwidth product of 3 and DFT length equal to the signal length.

```
fs = 1000;
t = 0:1/fs:2-1/fs;
x = cos(2*pi*100*t)+randn(size(t));
[pxx,f] = pmtm(x,3,length(x),fs);
plot(f,10*log10(pxx))
```

Average Single-Taper Estimates with Unity Weights

Obtain a multitaper PSD estimate where the individual tapered direct spectral estimates are given equal weight in the average.

Obtain the multitaper PSD estimate of a signal sampled at 1 kHz. The signal is a 100-Hz sine wave in additive N(0,1) white noise. The signal duration is 2 seconds. Use a time-halfbandwidth product of 3 and DFT length equal to the signal length. Use the `'unity'` option to give equal weight in the average to each of the individual tapered direct spectral estimates.

```
fs = 1000;
t = 0:1/fs:2-1/fs;
x = cos(2*pi*100*t)+randn(size(t));
[pxx,f] = pmtm(x,3,length(x),fs,'unity');
plot(f,10*log10(pxx))
```

DPSS Sequences and Their Frequency-Domain Concentrations

This example examines the frequency-domain concentrations of the DPSS sequences. The example produces a multitaper PSD estimate of an input signal by precomputing the Slepian sequences and selecting only those with more than 99% of their energy concentrated in the resolution bandwidth.

The signal is a 100-Hz sine wave in additive N(0,1) white noise. The signal duration is 2 seconds.

```
fs = 1000;
t = 0:1/fs:2-1/fs;
x = cos(2*pi*100*t)+randn(size(t));
```

Set the time-halfbandwidth product to 3.5. For the signal length of 2000 samples and a sampling interval of 0.001 seconds, this results in a resolution bandwidth of [-1.75,1.75] Hz. Calculate the first 10 Slepian sequences and examine their frequency concentrations in the specified resolution bandwidth. Determine the number of Slepian sequences with energy concentrations greater than 90%.

```
[e,v] = dpss(length(x),3.5,10);
stem(1:length(v),v,'markerfacecolor',[0 0 1]); set(gca,'ylim',[0 1.2])
title('Energy Concentrations in [-w,w] of k-th Slepian Sequence');
xlabel('k-th sequence'); ylabel('Proportion of Energy in [-w,w]');
h = line(1:length(v),0.990*ones(length(v),1));
set(h,'color','r','linewidth',2)
idx = find(v>0.99,1,'last');
```

Using the selected DPSS sequences, obtain the multitaper PSD estimate. Set `'DropLastTaper'` to `false` to use all the selected tapers.

```
[pxx,f] = pmtm(x,e(:,1:idx),v(1:idx),length(x),fs,'DropLastTaper',false);
plot(f,10*log10(pxx))
```

DC-Centered Multitaper PSD Estimate

Obtain the multitaper PSD estimate of a 100-Hz sine wave in additive N(0,1) noise. The data are sampled at 1 kHz. Use the `'centered'` option to obtain the DC-centered PSD and plot the result.

```
fs = 1000;
t = 0:1/fs:2-1/fs;
x = cos(2*pi*100*t)+randn(size(t));
[pxx,f] = pmtm(x,3.5,length(x),fs,'centered');
plot(f,10*log10(pxx))
```

```
xlabel('Hz'); ylabel('dB')
```

Upper and Lower 95%-Confidence Bounds

The following example illustrates the use of confidence bounds with the multitaper PSD estimate. While not a necessary condition for statistical significance, frequencies in the multitaper PSD estimate where the lower confidence bound exceeds the upper confidence bound for surrounding PSD estimates clearly indicate significant oscillations in the time series.

Create a signal consisting of the superposition of 100-Hz and 150-Hz sine waves in additive white N(0,1) noise. The amplitude of the two sine waves is 1. The sampling frequency is 1 kHz. The signal is 2 seconds in duration.

```
fs = 1000;
t = 0:1/fs:2-1/fs;
x = cos(2*pi*100*t)+cos(2*pi*150*t)+randn(size(t));
```

Obtain the multitaper PSD estimate with 95%-confidence bounds. Plot the PSD estimate along with the confidence interval and zoom in on the frequency region of interest near 100 and 150 Hz.

```
[pxx,f,pxxc] = pmtm(x,3.5,length(x),fs,...
'ConfidenceLevel', 0.95);
plot(f,10*log10(pxx)); hold on;
plot(f,10*log10(pxxc),'r--','linewidth',2);
axis([85 175 min(min(10*log10(pxxc))) max(max(10*log10(pxxc)))]);
xlabel('Hz'); ylabel('dB');
title('Multitaper PSD Estimate with 95%-Confidence Bounds');
```

At 100 and 150 Hz, the lower confidence bound exceeds the upper confidence bounds for surrounding PSD estimates.

Definitions

Discrete Prolate Spheroidal (Slepian) Sequences

The derivation of the Slepian sequences proceeds from the discrete-time — continuous frequency concentration problem. For all ℓ^2 sequences

index-limited to 0,1,...,N-1, the problem seeks the sequence having the maximal concentration of its energy in a frequency band [-*W*,*W*] with |*W*| < 1/2Δt.

This amounts to finding the eigenvalues and corresponding eigenvectors of an N-by-N self-adjoint positive semi-definite operator. Therefore, the eigenvalues are real and nonnegative and eigenvectors corresponding to distinct eigenvalues are mutually orthogonal. In this particular problem, the eigenvalues are bounded by 1 and the eigenvalue is the measure of the sequence's energy concentration in the frequency interval [-*W*,*W*].

The eigenvalue problem is given by

$$\sum_{n=0}^{N-1} \frac{\sin(2\pi W(n-m))}{\pi(n-m)} g_n = \lambda_k(N,W) g_m \quad m = 0,1,2,\ldots,N-1$$

The 0-th order DPSS sequence, g_0 is the eigenvector corresponding to the largest eigenvalue. The 1-st order DPSS sequence, g_1 is the eigenvector corresponding to the next largest eigenvalue and is orthogonal to the 0-th order sequence. The 2-nd order DPSS sequence, g_2, is the eigenvector corresponding to the third largest eigenvalue and is orthogonal to the 0-th order and 1-st order DPSS sequences. Because the operator is N-by-N, there are N eigenvectors. However, it can be shown that for a given sequence length N and a specified bandwidth [-*W*,*W*], there are approximately 2N*W*–1 DPSS sequences with eigenvalues very close to unity.

Multitaper Spectral Estimation

The periodogram is not a consistent estimator of the true power spectral density of a wide-sense stationary process. To produce a consisten estimate of the PSD, the multitaper method averages modified periodograms obtained using a family of mutually orthogonal tapers (windows). In addition to mutual orthogonality, the tapers also have optimal time-frequency concentration properties. Both the orthogonality and time-frequency concentration of the tapers is critical to the success of the multitaper technique. See "Discrete Prolate

Spheroidal (Slepian) Sequences" on page 1-811 for a brief description of the Slepian sequences used in Thomson's multitaper method.

The multitaper method uses K modified periodograms with each one obtained using a different Slepian sequence as the window. Let

$$S_k(f) = \Delta t \left| \sum_{n=0}^{N-1} g_{k,n} x_n e^{-i2\pi fn\Delta t} \right|^2$$

denote the modified periodogram obtained with the k-th Slepian sequence, $g_{k,n}$.

In the simplest form, the multitaper method simply averages the K modified periodograms to produce the multitaper PSD estimate.

$$S^{(\mathrm{MT})}(f) = \frac{1}{K} \sum_{k=0}^{K-1} S_k(f)$$

Note the difference between the multitaper PSD estimate and Welch's method. Both methods reduce the variability in the periodogram by averaging over approximately uncorrelated estimates of the PSD. However, the two approaches differ in how they produce these uncorrelated PSD estimates. The multitaper method uses the entire signal in each modified periodogram. The orthogonality of the Slepian tapers decorrelates the different modified periodograms. Welch's overlapped segment averaging approach uses segments of the signal in each modified periodogram and the segmenting decorrelates the different modified periodograms.

The preceding equation corresponds to the `'unity'` option in `pmtm`. However, as explained in "Discrete Prolate Spheroidal (Slepian) Sequences" on page 1-811, the Slepian sequences do not possess equal energy concentration in the frequency band of interest. The higher the order of the Slepian sequence, the less concentrated the sequence energy is in the band $[-W,W]$ with the concentration given by the eigenvalue. Consequently, it can be beneficial to use the eigenvalues to weight the K modified periodograms prior to averaging. This corresponds to the `'eigen'` option in `pmtm`.

Using the sequence eigenvalues to produce a weighted average of modified periodograms accounts for the frequency concentration properties of the Slepian sequences. However, it does not account for the interaction between the power spectral density of the random process and the frequency concentration of the Slepian sequences. Specifically, frequency regions where the random process has little power are less reliably estimated in the modified periodograms using higher order Slepian sequences. This argues for an frequency-dependent adaptive process, which accounts not only for the frequency concentration of the Slepian sequence, but also for the power distribution in the time series. This adaptive weighting corresponds to the `'adapt'` option in `pmtm` and is the default for computing the multitaper estimate.

References

[1] Percival, D.B., and A.T. Walden, *Spectral Analysis for Physical Applications: Multitaper and Conventional Univariate Techniques*, Cambridge University Press, 1993.

[2] Thomson, D.J., "Spectrum estimation and harmonic analysis," *Proceedings of the IEEE, Vol. 70* (1982), pp. 1055-1096.

See Also

`dpss | periodogram | pwelch`

Purpose

Pseudospectrum using MUSIC algorithm

Syntax

```
[S,w] = pmusic(x,p)
[S,w] = pmusic(x,p,w)
[S,w] = pmusic(...,nfft)
[S,f] = pmusic(x,p,nfft,fs)
[S,f] = pmusic(x,p,f,fs)
[S,f] = pmusic(...,'corr')
[S,f] = pmusic(x,p,nfft,fs,nwin,noverlap)
[...] = pmusic(...,freqrange)
[...,v,e] = pmusic(...)
pmusic(...)
```

Description

`[S,w] = pmusic(x,p)` implements the MUSIC (Multiple Signal Classification) algorithm and returns `S`, the pseudospectrum estimate of the input signal `x`, and a vector `w` of normalized frequencies (in rad/sample) at which the pseudospectrum is evaluated. The pseudospectrum is calculated using estimates of the eigenvectors of a correlation matrix associated with the input data `x`, where `x` is specified as either:

- A row or column vector representing one observation of the signal
- A rectangular array for which each row of `x` represents a separate observation of the signal (for example, each row is one output of an array of sensors, as in array processing), such that `x'*x` is an estimate of the correlation matrix

Note You can use the output of `corrmtx` to generate such an array `x`.

You can specify the second input argument `p` as either:

- A scalar integer. In this case, the signal subspace dimension is `p`.
- A two-element vector. In this case, `p(2)`, the second element of `p`, represents a threshold that is multiplied by λ_{min}, the smallest estimated eigenvalue of the signal's correlation matrix. Eigenvalues

below the threshold λ_{min}*p(2) are assigned to the noise subspace. In this case, p(1) specifies the maximum dimension of the signal subspace.

Note If the inputs to pmusic are real sinusoids, set the value of p to double the number of input signals. If the inputs are complex sinusoids, set p equal to the number of inputs.

The extra threshold parameter in the second entry in p provides you more flexibility and control in assigning the noise and signal subspaces.

S and w have the same length. In general, the length of the FFT and the values of the input x determine the length of the computed S and the range of the corresponding normalized frequencies. The following table indicates the length of S (and w) and the range of the corresponding normalized frequencies for this syntax.

S Characteristics for an FFT Length of 256 (Default)

Real/Complex Input Data	Length of S and w	Range of the Corresponding Normalized Frequencies
Real-valued	129	[0, π]
Complex-valued	256	[0, 2π)

[S,w] = pmusic(x,p,w) returns the pseudospectrum in the vector S computed at the normalized frequencies specified in vector w, which has two or more elements

[S,w] = pmusic(...,nfft) specifies the integer length of the FFT nfft used to estimate the pseudospectrum. The default value for nfft (entered as an empty vector []) is 256.

The following table indicates the length of S and w, and the frequency range for w in this syntax.

S and Frequency Vector Characteristics

Real/Complex Input Data	nfft Even/Odd	Length of S and w	Range of w
Real-valued	Even	`(nfft/2 + 1)`	[0, π]
Real-valued	Odd	`(nfft + 1)/2`	[0, π)
Complex-valued	Even or odd	`nfft`	[0, 2π)

`[S,f] = pmusic(x,p,nfft,fs)` returns the pseudospectrum in the vector `S` evaluated at the corresponding vector of frequencies `f` (in Hz). You supply the sampling frequency `fs` in Hz. If you specify `fs` with the empty vector `[]`, the sampling frequency defaults to 1 Hz.

The frequency range for `f` depends on `nfft`, `fs`, and the values of the input `x`. The length of `S` (and `f`) is the same as in the S and Frequency Vector Characteristics on page 1-817 above. The following table indicates the frequency range for `f` for this syntax.

S and Frequency Vector Characteristics with fs Specified

Real/Complex Input Data	nfft Even/Odd	Range of f
Real-valued	Even	`[0,fs/2]`
Real-valued	Odd	`[0,fs/2)`
Complex-valued	Even or odd	`[0,fs)`

`[S,f] = pmusic(x,p,f,fs)` returns the pseudospectrum in the vector `S` computed at the frequencies specified in vector `f`, which has two or more elements

`[S,f] = pmusic(...,'corr')` forces the input argument `x` to be interpreted as a correlation matrix rather than matrix of signal data. For this syntax `x` must be a square matrix, and all of its eigenvalues must be nonnegative.

`[S,f] = pmusic(x,p,nfft,fs,nwin,noverlap)` allows you to specify `nwin`, a scalar integer indicating a rectangular window length, or a real-valued vector specifying window coefficients. Use the scalar integer `noverlap` in conjunction with `nwin` to specify the number of input sample points by which successive windows overlap. `noverlap` is not used if `x` is a matrix. The default value for `nwin` is `2*p(1)` and `noverlap` is `nwin-1`.

With this syntax, the input data `x` is segmented and windowed before the matrix used to estimate the correlation matrix eigenvalues is formulated. The segmentation of the data depends on `nwin`, `noverlap`, and the form of `x`. Comments on the resulting windowed segments are described in the following table.

Windowed Data Depending on x and nwin

Input data x	Form of nwin	Windowed Data
Data vector	Scalar	Length is `nwin`
Data vector	Vector of coefficients	Length is `length(nwin)`
Data matrix	Scalar	Data is not windowed.
Data matrix	Vector of coefficients	`length(nwin)` must be the same as the column length of `x`, and `noverlap` is not used.

See the Eigenvector Length Depending on Input Data and Syntax on page 1-820 below for related information on this syntax.

Note The arguments `nwin` and `noverlap` are ignored when you include the string `'corr'` in the syntax.

`[...] = pmusic(...,freqrange)` specifies the range of frequency values to include in `f` or `w`. This syntax is useful when `x` is real. `freqrange` can be either:

- `'onesided'` — returns the one-sided PSD of a real input signal, `x`. If `nfft` is even, `Pxx` has length`nfft`/2+1 and is computed over the interval [0,π]. If `nfft` is odd, the length of `Pxx` is (`nfft`+1)/2 and the frequency interval is [0,π). When your specify `fs` , the intervals are [0,`fs`/2) and [0,`fs`/2] for even and odd length`nfft`respectively.
- `'twosided'` — returns the two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval [0,2π). When you specify `fs`, the frequency interval is [0,`fs`).
- `'centered'` — returns the centered two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval (-π, π] for even length `nfft` and (-π, π]) for odd length `nfft`. When you specify `fs`, the frequency intervals are (-`fs`/2, `fs`/2] and (-`fs`/2,`fs`/2) for even and odd length `nfft` respectively.

Note You can put the string arguments `freqrange` or `'corr'` anywhere in the input argument list after `p`.

`[...,v,e] = pmusic(...)` returns the matrix `v` of noise eigenvectors, along with the associated eigenvalues in the vector `e`. The columns of `v` span the noise subspace of dimension `size(v,2)`. The dimension of the signal subspace is `size(v,1)-size(v,2)`. For this syntax, `e` is a vector of estimated eigenvalues of the correlation matrix.

`pmusic(...)` with no output arguments plots the pseudospectrum in the current figure window.

Tips

In the process of estimating the pseudospectrum, `pmusic` computes the noise and signal subspaces from the estimated eigenvectors v_j and eigenvalues λ_j of the signal's correlation matrix. The smallest of these eigenvalues is used in conjunction with the threshold parameter `p(2)` to affect the dimension of the noise subspace in some cases.

The length n of the eigenvectors computed by `pmusic` is the sum of the dimensions of the signal and noise subspaces. This eigenvector

length depends on your input (signal data or correlation matrix) and the syntax you use.

The following table summarizes the dependency of the eigenvector length on the input argument.

Eigenvector Length Depending on Input Data and Syntax

Form of Input Data x	Comments on the Syntax	Length n of Eigenvectors
Row or column vector	`nwin` is specified as a scalar integer.	`nwin`
Row or column vector	`nwin` is specified as a vector.	`length(nwin)`
Row or column vector	`nwin` is not specified.	`2*p(1)`
l-by-*m* matrix	If `nwin` is specified as a scalar, it is not used. If `nwin` is specified as a vector, `length(nwin)` must equal *m*.	*m*
m-by-*m* nonnegative definite matrix	The string `'corr'` is specified and `nwin` is not used.	*m*

You should specify `nwin > p(1)` or `length(nwin) > p(1)` if you want `p(2) > 1` to have any effect.

Examples

Example 1: pmusic with no Sampling Specified

This example analyzes a signal vector `x`, assuming that two real sinusoidal components are present in the signal subspace. In this case, the dimension of the signal subspace is 4 because each real sinusoid is the sum of two complex exponentials:

```
n = 0:199;
```

```
x = cos(0.257*pi*n) + sin(0.2*pi*n) + 0.01*randn(size(n));
pmusic(x,4)      % Set p to 4 because two real inputs
```

Example 2: Specifying Sampling Frequency and Subspace Dimensions

This example analyzes the same signal vector `x` with an eigenvalue cutoff of 10% above the minimum. Setting `p(1) = Inf` forces the signal/noise subspace decision to be based on the threshold parameter `p(2)`. Specify the eigenvectors of length 7 using the `nwin` argument, and set the sampling frequency `fs` to 8 kHz:

```
n = 0:199;
x = cos(0.257*pi*n) + sin(0.2*pi*n) + 0.01*randn(size(n));
[P,f] = pmusic(x,[Inf,1.1],[],8000,7); % Window length = 7
```

Example 3: Entering a Correlation Matrix

Supply a positive definite correlation matrix `R` for estimating the spectral density. Use the default 256 samples:

```
R = toeplitz(cos(0.1*pi*[0:6])) + 0.1*eye(7);
[P,f] = pmusic(R,4,'corr');
```

Example 4: Entering a Signal Data Matrix Generated from corrmtx

Enter a signal data matrix `Xm` generated from data using `corrmtx`:

```
n = 0:699;
x = cos(0.257*pi*(n)) + 0.1*randn(size(n));
Xm = corrmtx(x,7,'mod');
[P,w] = pmusic(Xm,2);
```

Example 5: Using Windowing to Create the Effect of a Signal Data Matrix

Use the same signal, but let `pmusic` form the 100-by-7 data matrix using its windowing input arguments. In addition, specify an FFT of length 512:

```
n = 0:699;
x = cos(0.257*pi*(n)) + 0.1*randn(size(n));
[PP,ff] = pmusic(x,2,512,[],7,0);
```

Algorithms

The name MUSIC is an acronym for MUltiple SIgnal Classification. The MUSIC algorithm estimates the pseudospectrum from a signal or a correlation matrix using Schmidt's eigenspace analysis method [1]. The algorithm performs eigenspace analysis of the signal's correlation matrix in order to estimate the signal's frequency content. This algorithm is particularly suitable for signals that are the sum of sinusoids with additive white Gaussian noise. The eigenvalues and eigenvectors of the signal's correlation matrix are estimated if you don't supply the correlation matrix.

The MUSIC pseudospectrum estimate is given by

$$P_{music}(f) = \frac{1}{e^H(f)\left(\sum_{k=p+1}^{N} \mathrm{v}_k \mathrm{v}_k^H\right)e(f)} = \frac{1}{\sum_{k=p+1}^{N} |\,\mathrm{v}_k^H e(f)\,|^2}$$

where N is the dimension of the eigenvectors and v_k is the k-th eigenvector of the correlation matrix. The integer p is the dimension of the signal subspace, so the eigenvectors v_k used in the sum correspond to the smallest eigenvalues and also span the noise subspace. The vector e(f) consists of complex exponentials, so the inner product

$$\mathrm{v}_k^H \mathrm{e}(f)$$

amounts to a Fourier transform. This is used for computation of the pseudospectrum estimate. The FFT is computed for each v_k and then the squared magnitudes are summed.

References

[1] Marple, S.L. *Digital Spectral Analysis*, Englewood Cliffs, NJ, Prentice-Hall, 1987, pp. 373-378.

[2] Schmidt, R.O, "Multiple Emitter Location and Signal Parameter Estimation," *IEEE Trans. Antennas Propagation, Vol. AP-34* (March 1986), pp. 276-280.

[3] Stoica, P., and R.L. Moses, *Introduction to Spectral Analysis*, Prentice-Hall, Englewood Cliffs, NJ, 1997.

See Also

`corrmtx` | `dspdata` | `pburg` | `peig` | `periodogram` | `pmtm` | `prony` | `pwelch` | `rooteig` | `rootmusic` | `spectrum.music`

poly2ac

Purpose

Convert prediction filter polynomial to autocorrelation sequence

Syntax

```
r = poly2ac(a,efinal)
```

Description

`r = poly2ac(a,efinal)` finds the autocorrelation vector `r` corresponding to the prediction filter polynomial `a`. The autocorrelation sequence produced is approximately the same as that of the output of the autoregressive prediction filter whose coefficients are determined by `a`. `poly2ac` also produces the final `length(r)` step prediction error `efinal`. If `a(1)` is not equal to `1`, poly2ac normalizes the prediction filter polynomial by `a(1)`. `a(1)` cannot be 0.

Tips

You can apply this function to both real and complex polynomials.

Examples

```
a = [1.0000 0.6147 0.9898 0.0004 0.0034 -0.0077];
efinal = 0.2;
r = poly2ac(a,efinal)
```

References

[1] Kay, S.M. *Modern Spectral Estimation*, Englewood Cliffs, NJ, Prentice-Hall, 1988

See Also

`ac2poly` | `poly2rc` | `rc2ac`

Purpose

Convert prediction filter coefficients to line spectral frequencies

Syntax

```
lsf = poly2lsf(a)
```

Description

`lsf = poly2lsf(a)` returns a vector `lsf` of line spectral frequencies from a vector `a` of prediction filter coefficients.

Examples

```
a = [1.0000  0.6149  0.9899  0.0000  0.0031 -0.0082];
lsf = poly2lsf(a)
```

References

[1] Deller, J.R., J.G. Proakis, and J.H.L. Hansen, *Discrete-Time Processing of Speech Signals*, Prentice-Hall, 1993.

[2] Rabiner, L.R., and R.W. Schafer, *Digital Processing of Speech Signals*, Prentice-Hall, 1978.

See Also

`lsf2poly`

Purpose

Convert prediction filter polynomial to reflection coefficients

Syntax

```
k = poly2rc(a)
[k,r0] = poly2rc(a,efinal)
```

Description

`k = poly2rc(a)` converts the prediction filter polynomial `a` to the reflection coefficients of the corresponding lattice structure. `a` can be real or complex, and `a(1)` cannot be 0. If `a(1)` is not equal to `1`, poly2rc normalizes the prediction filter polynomial by `a(1)`. `k` is a row vector of size `length(a)-1`.

`[k,r0] = poly2rc(a,efinal)` returns the zero-lag autocorrelation, `r0`, based on the final prediction error, `efinal`.

A simple, fast way to check if `a` has all of its roots inside the unit circle is to check if each of the elements of `k` has magnitude less than 1.

```
stable = all(abs(poly2rc(a))<1)
```

Examples

```
a = [1.0000   0.6149   0.9899   0.0000   0.0031  -0.0082];
efinal = 0.2;
[k,r0] = poly2rc(a,efinal)
```

Limitations

If `abs(k(i)) == 1` for any `i`, finding the reflection coefficients is an ill-conditioned problem. `poly2rc` returns some `NaN`s and provide a warning message in this case.

Algorithms

`poly2rc` implements this recursive relationship:

$$k(n) = a_n(n)$$

$$a_{n-1}(m) = \frac{a_n(m) - k(n)a_n(n-m)}{1-k(n)^2}, \quad m = 1,2,\cdots,n-1$$

This relationship is based on Levinson's recursion [1]. To implement it, `poly2rc` loops through `a` in reverse order after discarding its first element. For each loop iteration `i`, the function:

1 Sets `k(i)` equal to `a(i)`

2 Applies the second relationship above to elements 1 through `i` of the vector `a`.

```
a = (a-k(i)*fliplr(a))/(1-k(i)^2);
```

References

[1] Kay, S.M. *Modern Spectral Estimation*, Englewood Cliffs, NJ, Prentice-Hall, 1988.

See Also

`ac2rc` | `latc2tf` | `latcfilt` | `poly2ac` | `rc2poly` | `tf2latc`

polyscale

Purpose

Scale roots of polynomial

Syntax

```
b = polyscale(a,alpha)
```

Description

`b = polyscale(a,alpha)` scales the roots of a polynomial in the *z*-plane, where `a` is a vector containing the polynomial coefficients and `alpha` is the scaling factor.

If `alpha` is a real value in the range `[0 1]`, then the roots of `a` are radially scaled toward the origin in the *z*-plane. Complex values for `alpha` allow arbitrary changes to the root locations.

Tips

By reducing the radius of the roots in an autoregressive polynomial, the bandwidth of the spectral peaks in the frequency response is expanded (flattened). This operation is often referred to as *bandwidth expansion*.

See Also

`polystab | roots`

Purpose

Stabilize polynomial

Syntax

```
b = polystab(a)
```

Description

`polystab` stabilizes a polynomial with respect to the unit circle; it reflects roots with magnitudes greater than 1 inside the unit circle.

`b = polystab(a)` returns a row vector `b` containing the stabilized polynomial, where `a` is a vector of polynomial coefficients, normally in the *z*-domain.

$$A(z) = a(1) + a(2)z^{-1} + \ldots + a(m+1)z^{-m}$$

Examples

`polystab` can convert a linear-phase filter into a minimum-phase filter with the same magnitude response:

```
h = fir1(25,0.4);
hmin = polystab(h) * norm(h)/norm(polystab(h));
```

Algorithms

`polystab` finds the roots of the polynomial and maps those roots found outside the unit circle to the inside of the unit circle:

```
v = roots(a);
vs = 0.5*(sign(abs(v)-1)+1);
v = (1-vs).*v + vs./conj(v);
b = a(1)*poly(v);
```

See Also

`roots`

pow2db

Purpose

Convert power to decibels (dB)

Syntax

```
ydb = pow2db(y)
```

Description

`ydb = pow2db(y)` returns the corresponding decibel (dB) value `ydb` for a given power value `y`. The relationship between power and decibels is `ydb` = $10*\log_{10}(y)$.

See Also

`db2pow`

Purpose

Prony method for filter design

Syntax

```
[Num,Den] = prony(impulse_resp,num_ord,denom_ord)
```

Description

`[Num,Den] = prony(impulse_resp,num_ord,denom_ord)` returns the numerator `Num` and denominator `Den` coefficients for a causal rational system function with impulse response `impulse_resp`. The system function has numerator order `num_ord` and denominator order `denom_ord`. The lengths of `Num` and `Den` are `num_ord+1` and `denom_ord+1`. If the length of `impulse_resp` is less than the largest order (`num_ord` or `denom_ord`), `impulse_resp` is padded with zeros. Enter 0 in `num_ord` for an all-pole system function. For an all-zero system function, enter a 0 for `denom_ord`.

Definitions

The *system function* is the z-transform of the impulse response *h[n]*:

$$H(z) = \sum_{n=-\infty}^{\infty} h[n] z^{-n}$$

A *rational system function* is a ratio of polynomials in z^{-1}. By convention the numerator polynomial is *B(z)* and the denominator is *A(z)*. The following equation describes a causal rational system function of numerator order `num_ord` q and denominator order `denom_ord` p :

$$H(z) = \frac{\sum_{k=0}^{q} b[k] z^{-k}}{1 + \sum_{l=1}^{p} a[l] z^{-l}}$$

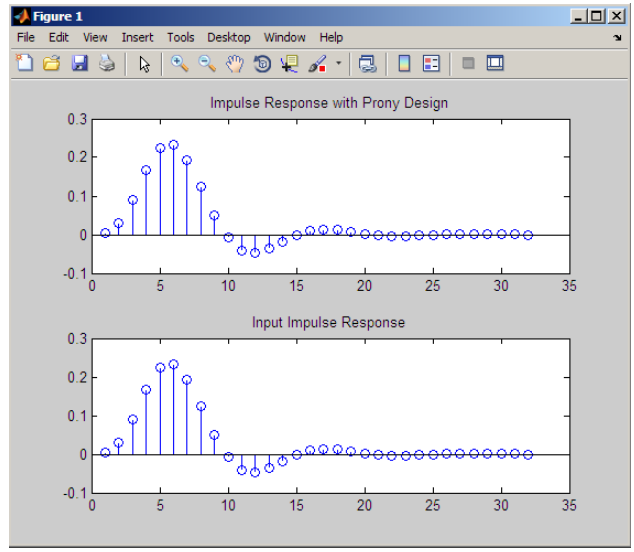
where *a[0]=1*.

Examples

Fit IIR model to an impulse response of lowpass filter:

```
d=fdesign.lowpass('Nb,Na,F3dB',4,4,0.2);
% Butterworth filter design
Hd=design(d,'butter');
```

```
% Obtain impulse response
impulse_resp=filter(Hd,[1 zeros(1,31)]);
% Find system function of order 4
denom_order=4; num_order=4;
[Num,Den]=prony(impulse_resp,num_order,denom_order);
% Compare impulse response with input
subplot(211);
stem(impz(Num,Den,length(impulse_resp)));
title('Impulse Response with Prony Design');
subplot(212);
stem(impulse_resp); title('Input Impulse Response');
```



Fit FIR model to an impulse response of highpass filter:

```
d=fdesign.highpass('N,F3dB',10,0.8);
Hd=design(d,'maxflat');
% Impulse response
impulse_resp=filter(Hd,[1 zeros(1,31)]);
% Find all-zero system function of order 10
```

```
num_order=10; denom_order=0;
[Num,Den]=prony(impulse_resp,num_order,denom_order);
% Compare Num to Hd.Numerator.
% Num and Hd.Numerator will not be identical but the
% coefficients will be close in value.
```

References

Parks, T.W., and C.S. Burrus *Digital Filter Design*, John Wiley & Sons, 1987, pp., 226–228.

See Also

`design` | `fdesign` | `impz` | `levinson` | `lpc`

How To

- "Parametric Modeling"

pulseperiod

Purpose

Period of bilevel pulse

Syntax

```
P = pulseperiod(X)
P = pulseperiod(X,FS)
P = pulseperiod(X,T)
[P,INITCROSS] = pulseperiod(...)
[P,INITCROSS,FINALCROSS] = pulseperiod(...)
[P,INITCROSS,FINALCROSS,NEXTCROSS] = pulseperiod(...)
[P,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] =
pulseperiod(...)
[P,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] =
pulseperiod(...,
   Name,Value)
```

Description

`P = pulseperiod(X)` returns a vector, `P`, containing the difference between the mid-reference level instants of the initial transition of each positive-polarity pulse and the next positive-going transition in the bilevel waveform,`X`. If `pulseperiod` does not find two positive-polarity transitions, `P` is empty. To determine the transitions for each pulse, `pulseperiod` estimates the state levels of the input waveform by a histogram method and identifies all regions which cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See “State-Level Tolerances” on page 1-839. Because `pulseperiod` uses interpolation to determine the mid-reference level instants, `P` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`P = pulseperiod(X,FS)` specifies the sampling rate in hertz as a positive scalar. The first sample instant in `X` corresponds to t=0. Because `pulseperiod` uses interpolation to determine the mid-reference level instants, `P` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`P = pulseperiod(X,T)` specifies the sampling instants in a vector equal in length to `X`. Because `pulseperiod` uses interpolation to

determine the mid-reference level instants, `P` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`[P,INITCROSS] = pulseperiod(...)` returns the mid-reference level instants of the first transition of each pulse.

`[P,INITCROSS,FINALCROSS] = pulseperiod(...)` returns the mid-reference level instants of the final transition of each pulse.

`[P,INITCROSS,FINALCROSS,NEXTCROSS] = pulseperiod(...)` returns the mid-reference level instants of next detected transition after each pulse.

`[P,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] = pulseperiod(...)` returns the mid-reference level,`MIDLEV`.

`[P,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] = pulseperiod(...,Name,Value)` returns the pulse periods with additional options specified by one or more `Name,Value` pair arguments.

Input Arguments

X

Bilevel waveform. If the waveform, `X`, does not contain at least two transitions, `pulseperiod` outputs an empty matrix.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

'MidPct'

Mid-reference level as a percentage of the waveform amplitude.

Default: 50

'Polarity'

Pulse polarity. Specify the polarity as `'positive'` or `'negative'`. If you specify `'positive'`, `pulseperiod` looks for pulses whose initial transition is positive-going (positive polarity). If you specify `'negative'`, `pulseperiod` looks for pulses whose initial transition is negative-going (negative polarity).

Default: `'positive'`

'StateLevels'

Low- and high-state levels. `StateLevels` is a 1-by-2 real-valued vector. The first element is the low-state level. The second element is the high-state level. If you do not specify low and high-state levels, `pulseperiod` estimates the state levels from the input waveform using the histogram method.

'Tolerance'

Tolerance levels (lower and upper state boundaries) expressed as a percentage. See "State-Level Tolerances" on page 1-839.

Default: 2

Output Arguments

P

Pulse period in seconds. The pulse period is defined as the time between the mid-reference level instants of two consecutive transitions.

INITCROSS

Mid-reference level instant of initial transition.

FINALCROSS

Mid-reference level instant of final transition.

NEXTCROSS

Mid-reference level instant of the first pulse transition after the final transition of the preceding pulse.

MIDLEV

Waveform value that corresponds to the mid-reference level.

Definitions

Mid-Reference Level

The mid-reference level in a bilevel waveform with low-state level, *S_1*, and high- state level, *S_2*, is

$$S_1 + \frac{1}{2}(S_2 - S_1)$$

Mid-Reference Level Instant

Let $y_{50\%}$ denote the mid-reference level.

Let $t_{50\%-}$ and $t_{50\%+}$ denote the two consecutive sampling instants corresponding to the waveform values nearest in value to $y_{50\%}$.

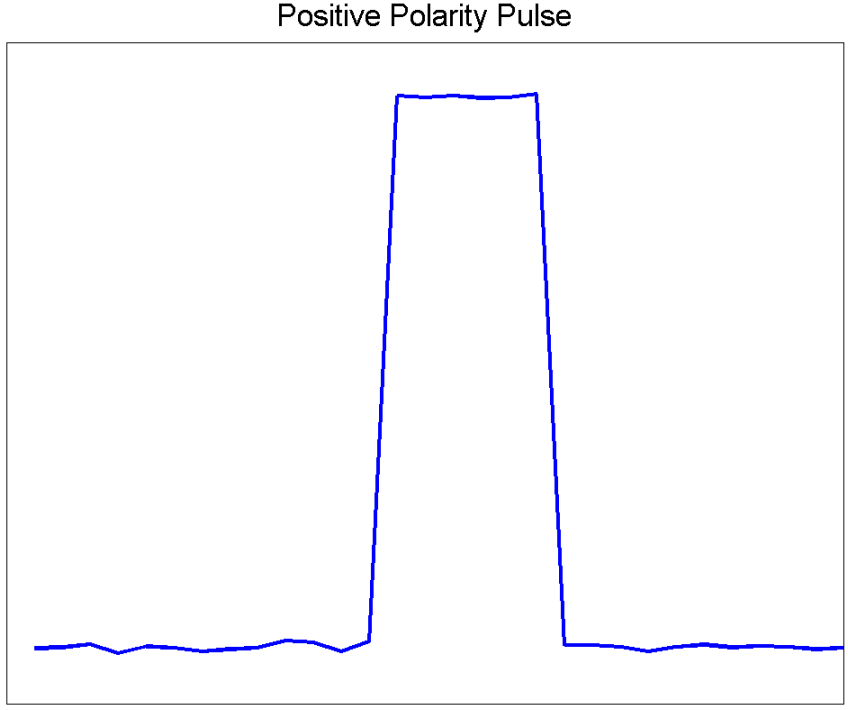
Let $y_{50\%-}$ and $y_{50\%+}$ denote the waveform values at $t_{50\%-}$ and $t_{50\%+}$.

The mid-reference level instant is

$$t_{50\%} = t_{50\%} + \left(\frac{t_{50\%_+} - t_{50\%_-}}{y_{50\%_+} - y_{50\%_-}}\right)\left(y_{50\%_+} - y_{50\%_-}\right)$$

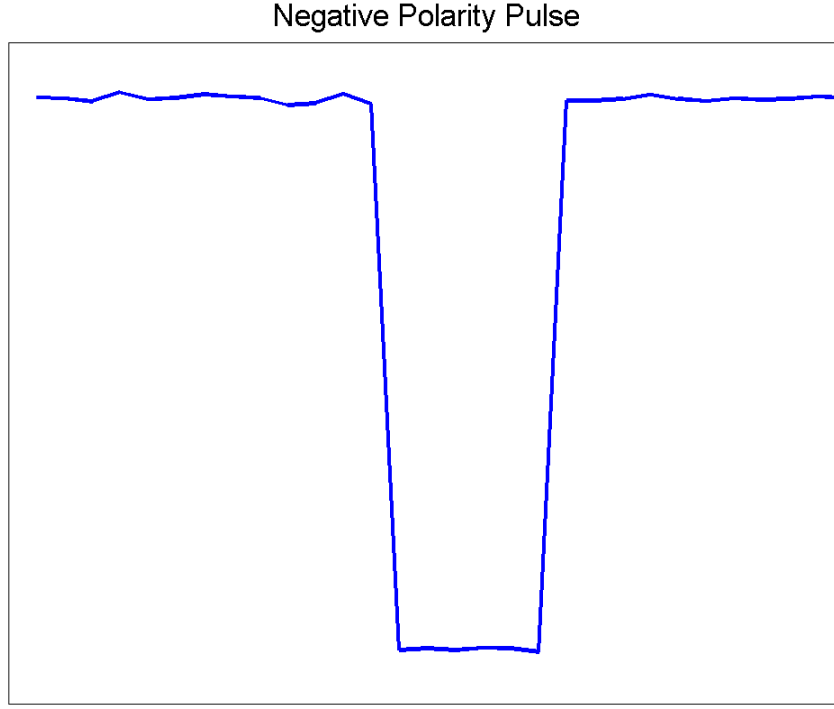
Pulse Polarity

If the initial transition of a pulse is positive-going, the pulse has positive polarity. The following figure shows a positive-polarity pulse.



Equivalently, a positive-polarity (positive-going) pulse has a terminating state more positive than the originating state.

If the initial transition of a pulse is negative-going, the pulse has negative polarity. The following figure shows a negative-polarity pulse.



Equivalently, a negative-polarity (negative-going) pulse has a originating state more positive than the terminating state.

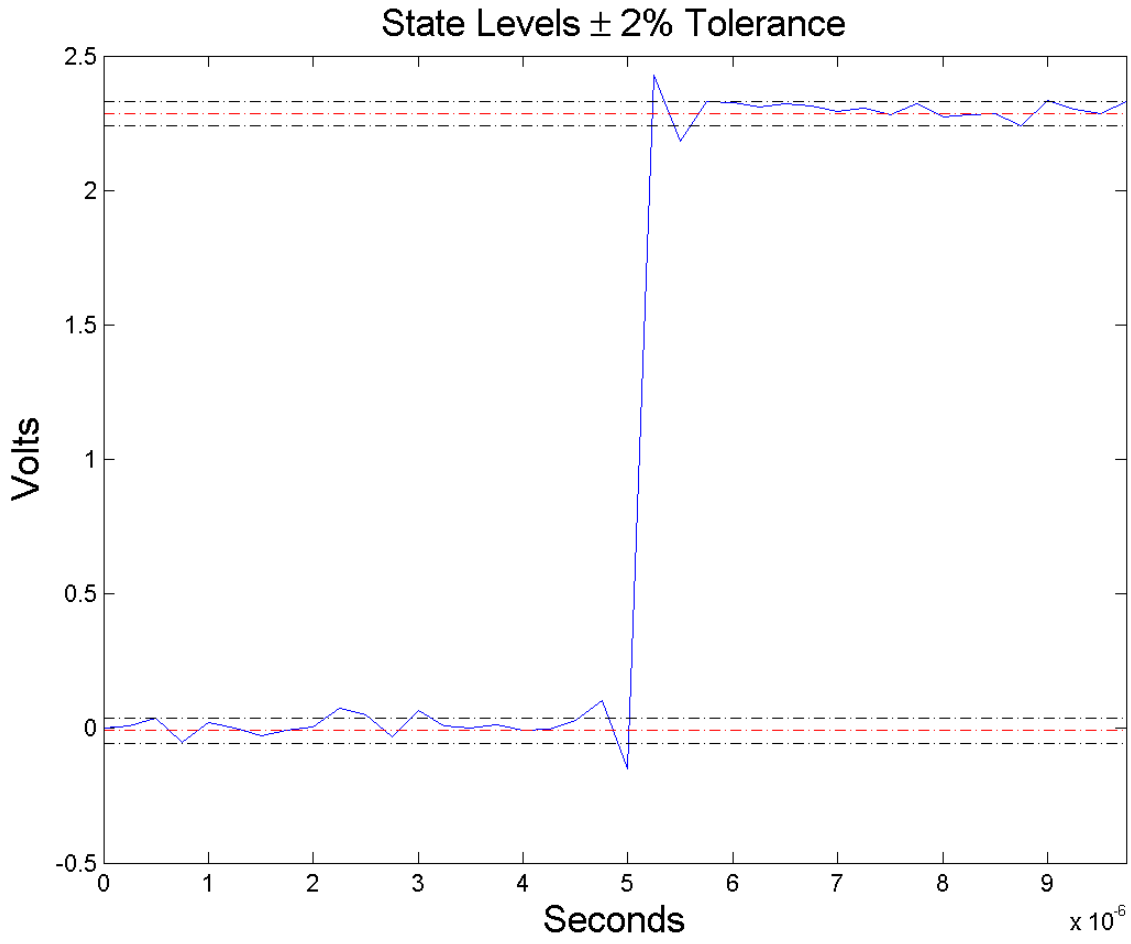
State-Level Tolerances

Each state level can have associated lower- and upper-state boundaries. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The red dashed lines indicate the estimated state levels.



Examples

Pulse Period of Bilevel Waveform

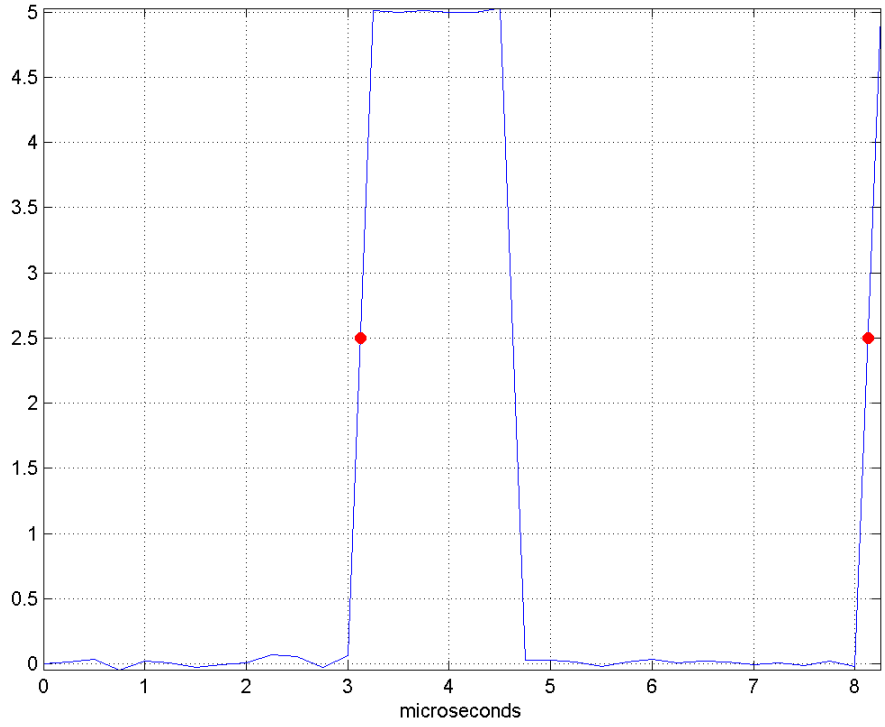
Compute the pulse period of a bilevel waveform with two positive-polarity transitions. The sampling rate is 4 MHz.

```
load('pulseex.mat', 'x', 't');
p = pulseperiod(x, t);
```

Determine Mid-Reference Level Instants of Pulse Period

Determine the mid-reference level instants, which define the pulse period for a bilevel waveform. Mark the mid-reference level instants on a plot of the data.

```
load('pulseex.mat', 'x', 't');
[p,initcross,~,nextcross,midlev] = pulseperiod(x,t);
fprintf('Pulse period is %2.3f microseconds \n',p*1e6);
plot(t.*1e6,x); hold on;
grid on; axis tight; xlabel('microseconds');
plot(initcross.*1e6,midlev,'ro','markerfacecolor',[1 0 0]);
plot(nextcross.*1e6,midlev,'ro','markerfacecolor',[1 0 0]);
```



References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003.

See Also

`dutycycle` | `pulsesep` | `pulsewidth` | `statelevels`

pulsesep

Purpose

Separation between bilevel waveform pulses

Syntax

```
S = pulsesep(X)
S = pulsesep(X,FS)
S = pulsesep(X,T)
[S,INITCROSS] = pulsesep(...)
[S,INITCROSS,FINALCROSS] = pulsesep(...)
[S,INITCROSS,FINALCROSS,NEXTCROSS] = pulsesep(...)
[S,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] = pulsesep(...)
[S,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] =
pulsesep(...,Name,
   Value)
```

Description

`S = pulsesep(X)` returns the differences, `S`, between the mid-reference level instants of the final negative-going transitions of every positive-polarity pulse and the next positive-going transition. `X` is a bilevel waveform. To determine the transitions that compose each pulse, `pulsesep` estimates the state levels of `X` by a histogram method. `pulsesep` identifies all regions that cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-849. Because `pulsesep` uses interpolation to determine the mid-reference level instants, `S` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`S = pulsesep(X,FS)` specifies the sampling rate, `FS`, in Hz as a positive scalar. The first time instant corresponds to t=0. Because `pulsesep` uses interpolation to determine the mid-reference level instants, `S` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`S = pulsesep(X,T)` specifies the sampling instants, `T`, in a vector equal in length to `X`. Because `pulsesep` uses interpolation to determine the mid-reference level instants, `S` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`[S,INITCROSS] = pulsesep(...)` returns the mid-reference level instants, `INITCROSS`, of the first positive-polarity transitions.

`[S,INITCROSS,FINALCROSS] = pulsesep(...)` returns the mid-reference level instants, `FINALCROSS`, of the final transition of each pulse.

`[S,INITCROSS,FINALCROSS,NEXTCROSS] = pulsesep(...)` returns the mid-reference level instants, `NEXTCROSS`, of the next detected transition after each pulse.

`[S,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] = pulsesep(...)` returns the mid-reference level, `MIDLEV`.

`[S,INITCROSS,FINALCROSS,NEXTCROSS,MIDLEV] = pulsesep(...,Name,Value)` returns the pulse separations with additional options specified by one or more `Name,Value` pair arguments.

Input Arguments

X

Bilevel waveform. If the waveform, `X`, does not contain at least two transitions, `pulsesep` outputs an empty matrix.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

'MidPct'

Mid-reference level as a percentage of the waveform amplitude.

Default: `50`

'Polarity'

Pulse polarity. Specify the polarity as `'positive'` or `'negative'`. If you specify `'positive'`, `pulsesep` looks for pulses with positive-going (positive polarity) initial transitions. If you specify `'negative'`, `pulsesep` looks for pulses with negative-going (negative polarity) initial transitions. See "Pulse Polarity" on page 1-847.

Default: `'positive'`

'StateLevels'

Low- and high-state levels. `StateLevels` is a 1-by-2 real-valued vector. The first element is the low-state level. The second element is the high-state level. If you do not specify low- and high-state levels, `pulsesep` estimates the state levels from the input waveform using the histogram method.

'Tolerance'

Tolerance levels (lower- and upper-state boundaries) expressed as a percentage. See "State-Level Tolerances" on page 1-849.

Default: `2`

Output Arguments

S

Pulse separations in seconds. The *pulse separation* is defined as the time between the mid-reference level instants of the final transition of one pulse and the initial transition of the next pulse. See "Pulse Separation" on page 1-851.

INITCROSS

Mid-reference level instants of initial transition.

FINALCROSS

Mid-reference level instants of final transition.

NEXTCROSS

Mid-reference level instants of the initial transition after the final transition of the preceding pulse.

MIDLEV

Waveform value that corresponds to the mid-reference level.

Definitions

Mid-Reference Level

The mid-reference level in a bilevel waveform with low-state level, *S_1*, and high-state level, *S_2*, is

$$S_1 + \frac{1}{2}(S_2 - S_1)$$

Mid-Reference Level Instant

Let $y_{50\%}$ denote the mid-reference level.

Let $t_{50\%_-}$ and $t_{50\%_+}$ denote the two consecutive sampling instants corresponding to the waveform values nearest in value to $y_{50\%}$.

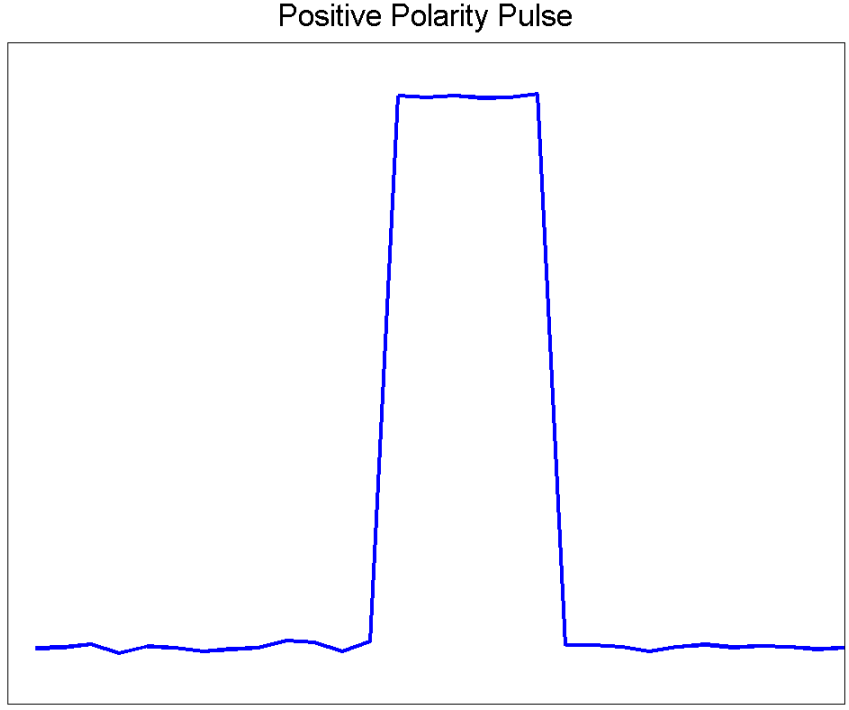
Let $y_{50\%_-}$ and $y_{50\%_+}$ denote the waveform values at $t_{50\%_-}$ and $t_{50\%_+}$.

The mid-reference level instant is

$$t_{50\%} = t_{50\%} + (\frac{t_{50\%_+} - t_{50\%_-}}{y_{50\%_+} - y_{50\%_-}})(y_{50\%_+} - y_{50\%_-})$$

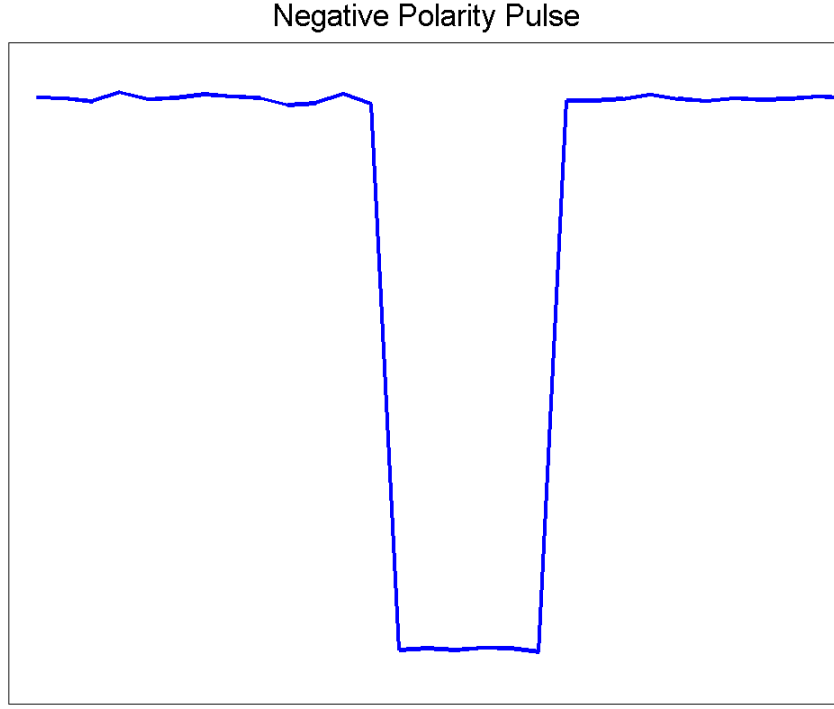
Pulse Polarity

If the pulse has an initial positive-going transition, the pulse has positive polarity. The following figure shows a positive-polarity pulse.



Equivalently, a positive-polarity (positive-going) pulse has a terminating state more positive than the originating state.

If the pulse has an initial negative-going transition, the pulse has negative polarity. The following figure shows a negative-polarity pulse.



Equivalently, a negative-polarity (negative-going) pulse has a originating state more positive than the terminating state.

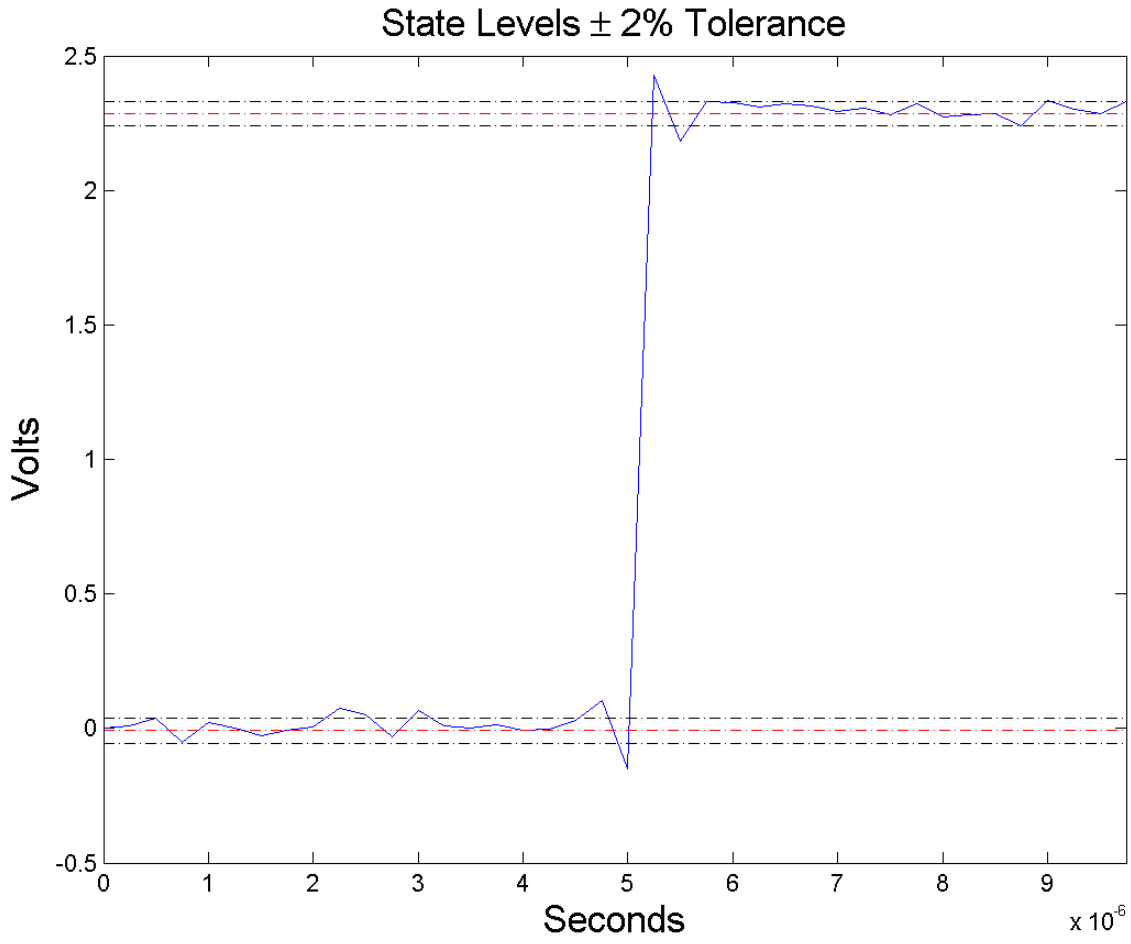
State-Level Tolerances

Each state level can have an associated lower- and upper-state boundary. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

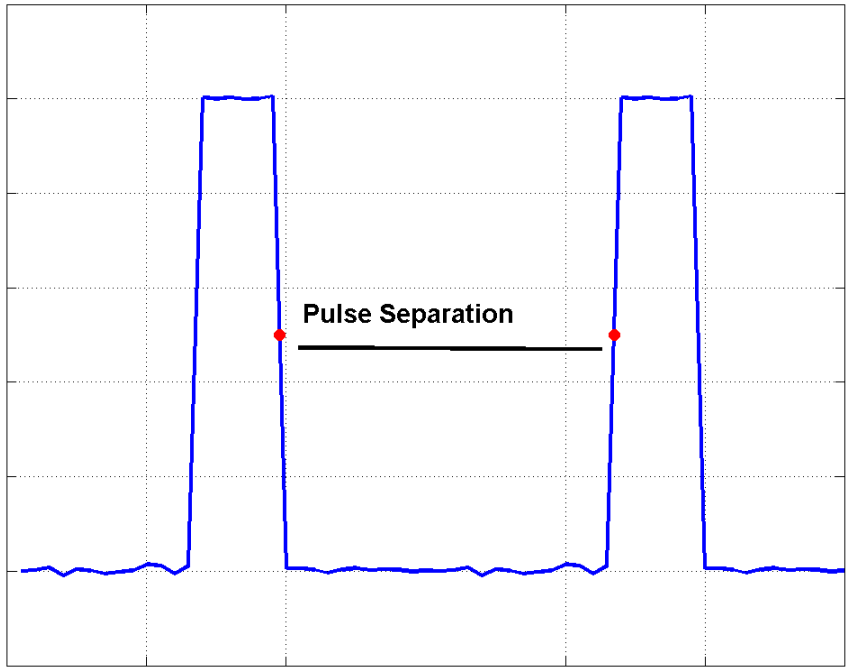
where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The red dashed lines indicate the estimated state levels.



Pulse Separation

Pulse separation is the time difference between the mid-reference level instant of the final transition of one pulse and the mid-reference level instant of the initial transition of the next pulse. The following figure illustrates pulse separation.



Examples

Pulse Separation in Bilevel Waveform

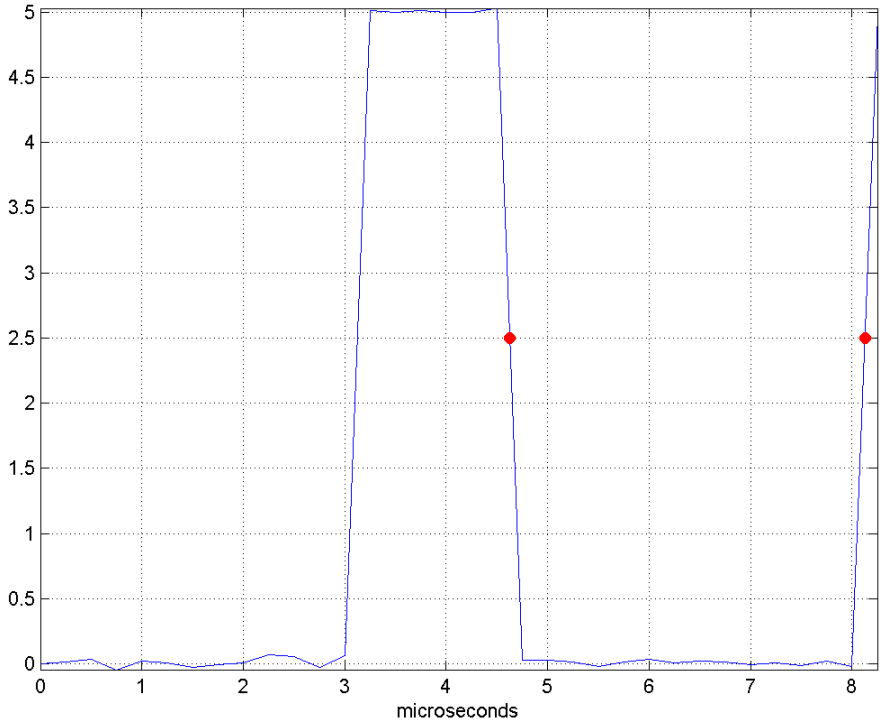
Compute the pulse separation in a bilevel waveform with two positive-polarity transitions. The sampling rate is 4 MHz.

```
load('pulseex.mat', 'x', 't');
s = pulsesep(x, t);
```

Determine Mid-Reference Level Instants Defining Pulse Separation

Determine the mid-reference level instants, which define the pulse separation for a bilevel waveform. Mark the mid-reference level instants on a plot of the data.

```
load('pulseex.mat', 'x', 't');
[s,~,finalcross,nextcross,midlev] = pulsesep(x,t);
fprintf('Pulse separation is %2.3f microseconds \n',s*1e6);
plot(t.*1e6,x); hold on;
grid on; axis tight; xlabel('microseconds');
plot(finalcross.*1e6,midlev,'ro','markerfacecolor',[1 0 0]);
plot(nextcross.*1e6,midlev,'ro','markerfacecolor',[1 0 0]);
```



References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003.

See Also

`dutycycle` | `pulseperiod` | `pulsewidth` | `statelevels`

Purpose

Bilevel waveform pulse width

Syntax

```
W = pulsewidth(X)
W = pulsewidth(X,FS)
W = pulsewidth(X,T)
[W,INITCROSS] = pulsewidth(...)
[W,INITCROSS,FINALCROSS] = pulsewidth(...)
[W,INITCROSS,FINALCROSS,MIDLEV] = pulsewidth(...)
W = pulsewidth(...,Name,Value)
```

Description

`W = pulsewidth(X)` returns a vector, `W`, containing the time differences between the mid-reference level instants of the initial and final transitions of each positive-polarity pulse in the bilevel waveforml, `X`. To determine the transitions, `pulsewidth` estimates the low- and high-state levels of `X` by a histogram method. `pulsewidth` identifies all regions that cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-860. Because `pulsewidth` uses interpolation to determine the mid-reference level instants, `W` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`W = pulsewidth(X,FS)` specifies the sample rate, `FS`, in hertz as a positive scalar. The first sample in the waveform corresponds to t=0. Because `pulsewidth` uses interpolation to determine the mid-reference level instants, `W` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`W = pulsewidth(X,T)` specifies the sample instants, `T`, as a vector with the same number of elements as `X`. Because `pulsewidth` uses interpolation to determine the mid-reference level instants, `W` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`[W,INITCROSS] = pulsewidth(...)` returns a column vector, `INITCROSS`, whose elements correspond to the mid-reference level instants of the initial transition of each pulse.

`[W,INITCROSS,FINALCROSS] = pulsewidth(...)` returns a column vector, `FINALCROSS`, whose elements correspond to the mid-reference level instants of the final transition of each pulse.

`[W,INITCROSS,FINALCROSS,MIDLEV] = pulsewidth(...)` returns the waveform value, `MIDLEV`, which corresponds to the mid-reference level.

`W = pulsewidth(...,Name,Value)` returns the pulse widths with additional options specified by one or more `Name,Value` pair arguments.

Input Arguments

X

Bilevel waveform. `X` is a real-valued row or column vector.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

'MidPct'

Mid-reference level as percentage of the waveform amplitude. See "Mid-Reference Level" on page 1-857.

Default: `50`

'Polarity'

Pulse polarity. Specify the polarity as `'positive'` or `'negative'`. If you specify `'positive'`, `pulsewidth` looks for pulses with positive-going (positive polarity) initial transitions. If you specify `'negative'`, `pulsewidth` looks for pulses with negative-going (negative polarity) initial transitions. See "Pulse Polarity" on page 1-858.

Default: `'positive'`

'StateLevels'

Low- and high-state levels. `StateLevels` is a 1-by-2 real-valued vector. The first element is the low-state level. The second element is the high-state level. If you do not specify low- and high-state levels, `pulsewidth` estimates the state levels from the input waveform using the histogram method.

'Tolerance'

Tolerance levels (lower and upper state boundaries) expressed as a percentage. See "State-Level Tolerances" on page 1-860.

Default: `2`

Output Arguments

W

Pulse widths in seconds. The pulse width is the time difference between the initial and final transitions of a pulse. The times of the initial and final transitions are referred to as *transition occurence instants* in [1].

INITCROSS

Mid-reference level instants of the initial transition

FINALCROSS

Mid-reference level instants of the final transition

MIDLEV

Waveform value corresponding to the mid-reference level

Definitions

Mid-Reference Level

The mid-reference level in a bilevel waveform with low-state level, *S_1*, and high- state level, *S_2*, is

$$S_1 + \frac{1}{2}(S_2 - S_1)$$

Mid-Reference Level Instant

Let $y_{50\%}$ denote the mid-reference level.

Let $t_{50\%-}$ and $t_{50\%+}$ denote the two consecutive sampling instants corresponding to the waveform values nearest in value to $y_{50\%}$.

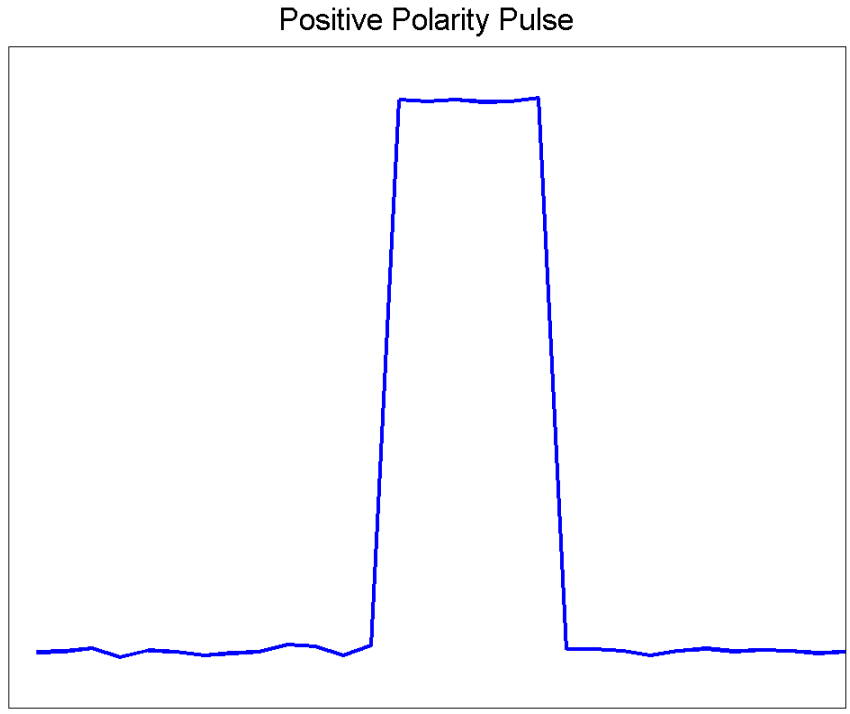
Let $y_{50\%-}$ and $y_{50\%+}$ denote the waveform values at $t_{50\%-}$ and $t_{50\%+}$.

The mid-reference level instant is

$$t_{50\%} = t_{50\%} + (\frac{t_{50\%_+} - t_{50\%_-}}{y_{50\%_+} - y_{50\%_-}})(y_{50\%_+} - y_{50\%_-})$$

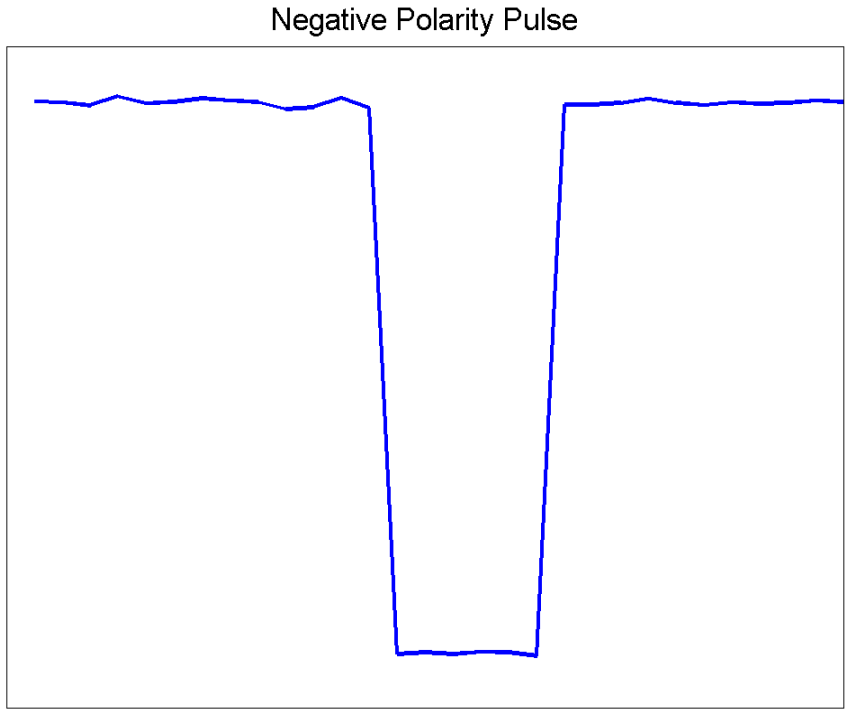
Pulse Polarity

If the pulse has a positive-going initial transition, the pulse has positive polarity. The following figure shows a positive-polarity pulse.



Equivalently, a positive-polarity (positive-going) pulse has a terminating state more positive than the originating state.

If the pulse has a negative-going initial transition, the pulse has negative polarity. The following figure shows a negative-polarity pulse.



Equivalently, a negative-polarity (negative-going) pulse has a originating state more positive than the terminating state.

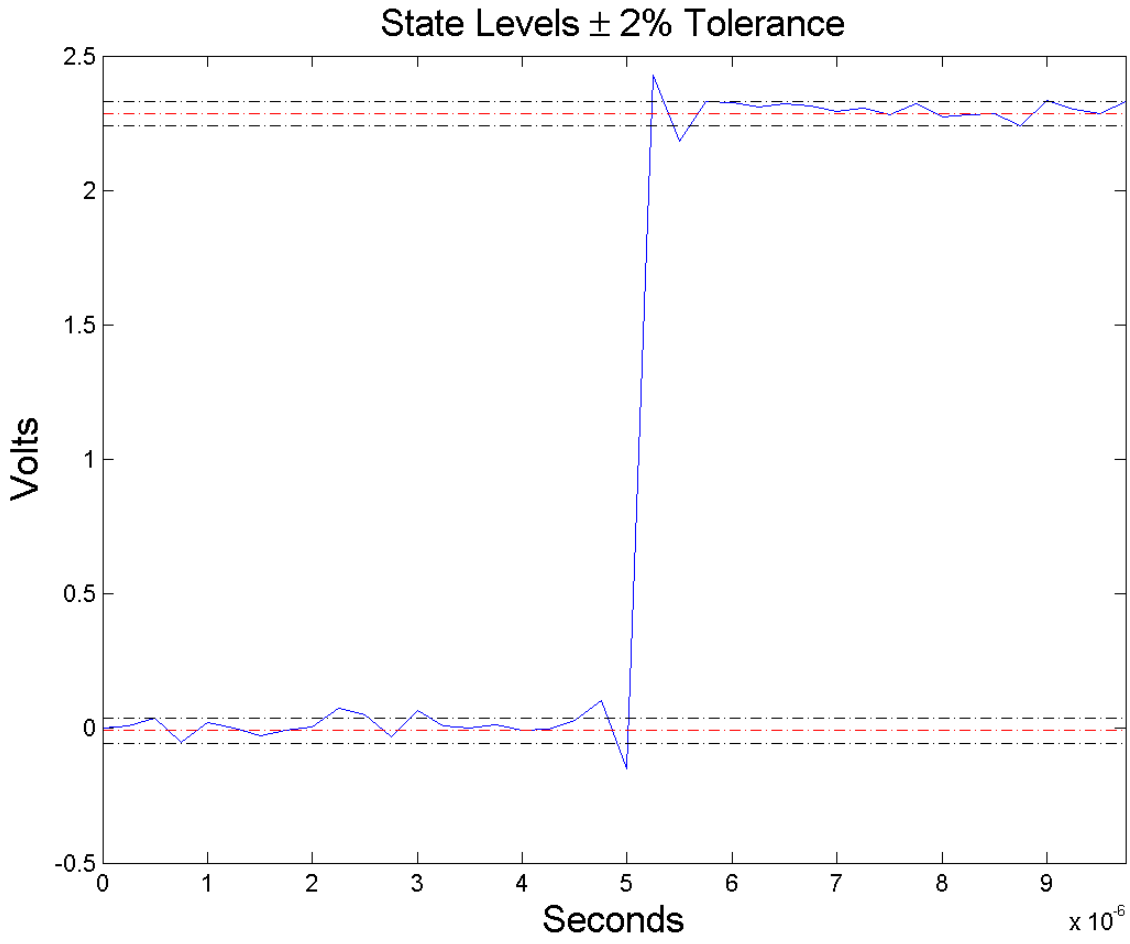
State-Level Tolerances

Each state level can have an associated lower- and upper-state boundary. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The red dashed lines indicate the estimated state levels.



Examples

Pulse Width of Bilevel Waveform

Compute the pulse width of a bilevel waveform sampled at 4 MHz.

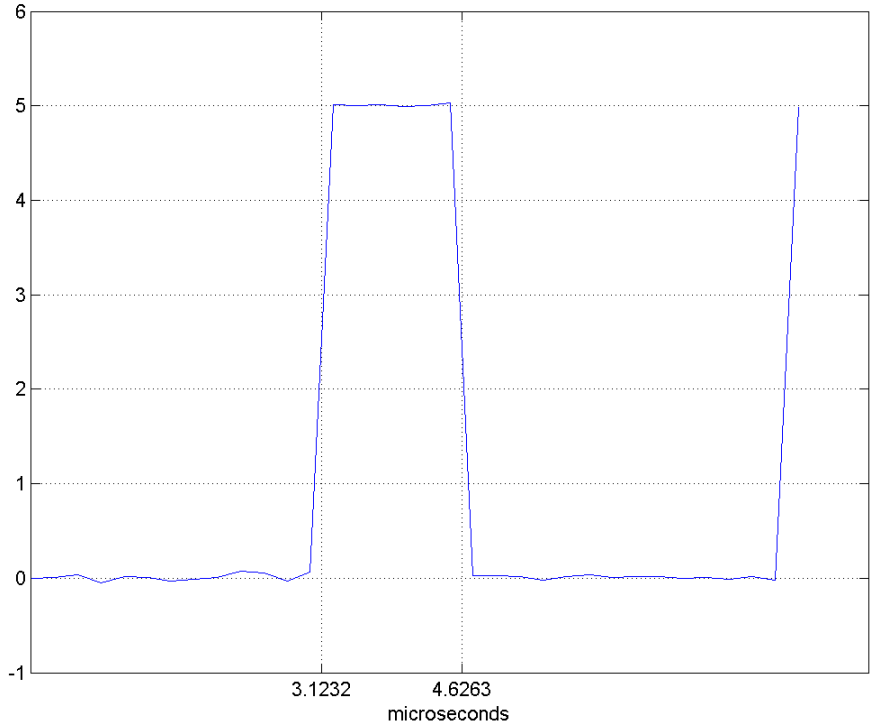
```
load('pulseex.mat', 'x', 't');
```

```
w = pulsewidth(x, t);
plot(t,x); grid on;
```

First and Second Transition Times for Bilevel Waveform

Compute the initial and final transition occurrences for a bilevel waveform sampled at 4 MHz. Plot the result annotated with the transition occurrences.

```
load('pulseex.mat', 'x', 't');
fs = 4e6;
[w,initcross,finalcross] = pulsewidth(x,fs);
plot(t.*1e6,x);
set(gca,'xtick',[initcross*1e6 finalcross*1e6]);
grid on;
xlabel('microseconds');
```



Specify State Levels for Bilevel Waveform

Specify the state levels for the bilevel waveform instead of estimating the levels from the data. Use the `'StateLevels'` name-value pair to enter the low-state level as 0 and the high-state level as 5.

```
load('pulseex.mat', 'x', 't');
[w,initcross,finalcross] = pulsewidth(x,fs,'StateLevels',[0 5]);
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003.

See Also

dutycycle | pulseperiod | pulsesep | statelevels

pulstran

Purpose

Pulse train

Syntax

```
pulstran
y = pulstran(t,d,'func')
pulstran(t,d,'func',p1,p2,...)
pulstran(t,d,p,fs)
pulstran(t,d,p)
pulstran(...,'func')
```

Description

`pulstran` generates pulse trains from continuous functions or sampled prototype pulses.

`y = pulstran(t,d,'func')` generates a pulse train based on samples of a continuous function, `'func'`, where `'func'` is

- `'gauspuls'`, for generating a Gaussian-modulated sinusoidal pulse
- `'rectpuls'`, for generating a sampled aperiodic rectangle
- `'tripuls'`, for generating a sampled aperiodic triangle

`pulstran` is evaluated `length(d)` times and returns the sum of the evaluations `y = func(t-d(1)) + func(t-d(2)) + ...`

The function is evaluated over the range of argument values specified in array `t`, after removing a scalar argument offset taken from the vector `d`. Note that *`func`* must be a vectorized function that can take an array `t` as an argument.

An optional gain factor may be applied to each delayed evaluation by specifying `d` as a two-column matrix, with the offset defined in column 1 and associated gain in column 2 of `d`. Note that a row vector will be interpreted as specifying delays only.

`pulstran(t,d,'func',p1,p2,...)` allows additional parameters to be passed to `'func'` as necessary. For example:

```
func(t-d(1),p1,p2,...) + func(t-d(2),p1,p2,...) + ...
```

`pulstran(t,d,p,fs)` generates a pulse train that is the sum of multiple delayed interpolations of the prototype pulse in vector `p`, sampled at the

rate `fs`, where `p` spans the time interval `[0,(length(p)-1)/fs]`, and its samples are identically 0 outside this interval. By default, linear interpolation is used for generating delays.

`pulstran(t,d,p)` assumes that the sampling rate `fs` is equal to 1 Hz.

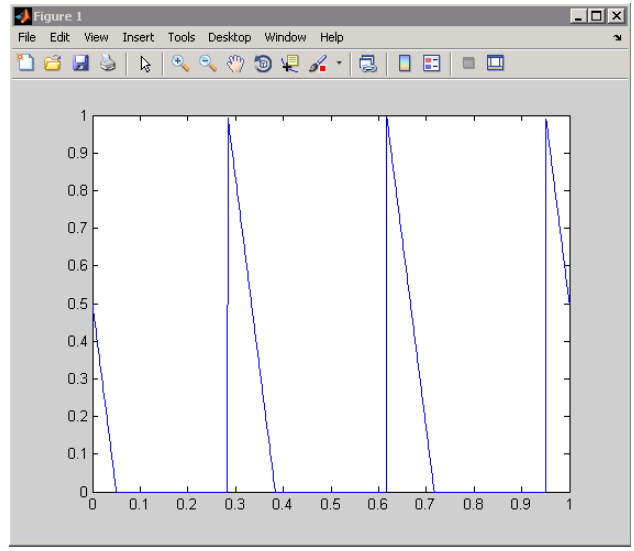
`pulstran(...,'func')` specifies alternative interpolation methods. See `interp1` for a list of available methods.

Examples

Example 1

This example generates an asymmetric sawtooth waveform with a repetition frequency of 3 Hz and a sawtooth width of 0.1s. It has a signal length of 1s and a 1 kHz sample rate:

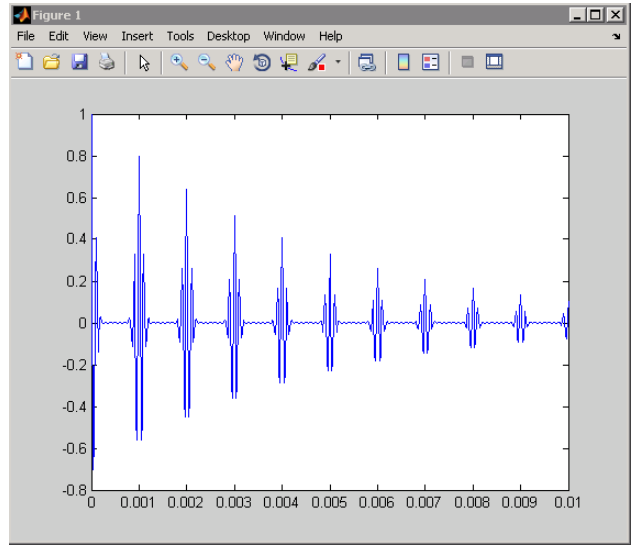
```
t = 0 : 1/1e3 : 1;          % 1 kHz sample freq for 1 sec
d = 0 : 1/3 : 1;            % 3 Hz repetition freq
y = pulstran(t,d,'tripuls',0.1,-1);
plot(t,y)
```



Example 2

This example generates a periodic Gaussian pulse signal at 10 kHz, with 50% bandwidth. The pulse repetition frequency is 1 kHz, sample rate is 50 kHz, and pulse train length is 10 msec. The repetition amplitude should attenuate by 0.8 each time:

```
t = 0 : 1/50E3 : 10e-3;
d = [0 : 1/1E3 : 10e-3 ; 0.8.^(0:10)]';
y = pulstran(t,d,'gauspuls',10e3,0.5);
plot(t,y)
```



See Also

chirp | cos | diric | gauspuls | rectpuls | sawtooth | sin | sinc | square | tripuls

Purpose

Welch's power spectral density estimate

Syntax

```
pxx = pwelch(x)
pxx = pwelch(x,window)
pxx = pwelch(x,window,noverlap)
pxx = pwelch(x,window,noverlap,nfft)

[pxx,w] = pwelch( ___ )
[pxx,f] = pwelch( ___ ,fs)

[pxx,w] = pwelch(x,window,noverlap,w)
[pxx,f] = pwelch(x,window,noverlap,f,fs)

[ ___ ] = pwelch(x,window, ___ ,freqrange)
[ ___ ] = pwelch(x,window, ___ ,spectrumtype)

[pxx,f,pxxc] = pwelch( ___ ,'ConfidenceLevel', probability)

pwelch( ___ )
```

Description

`pxx = pwelch(x)` returns the power spectral density (PSD) estimate, `pxx`, of the input signal, `x` using Welch's overlapped segment averaging estimator. If `x` is real-valued, `pxx` is a one-sided PSD estimate. If `x` is complex-valued, `pxx` is a two-sided PSD estimate. By default, `x` is divided into 8 segments with 50% overlap and each section is windowed with a Hamming window. The modified periodograms are averaged to obtain the PSD estimate. If you cannot divide the length of `x` exactly into eight sections with 50% overlap, `x` is truncated accordingly.

`pxx = pwelch(x,window)` uses the input vector or integer, `window`, to divide the signal into sections. If `window` is a vector, `pwelch` divides the signal into sections equal in length to the length of `window`. The modified periodograms are computed using the signal sections multiplied by the vector, `window`. If `window` is an integer, the signal is divided into sections of length `window`. The modified periodograms are computed using a Hamming window of length `window`.

`pxx = pwelch(x,window,noverlap)` uses `noverlap` samples of overlap from section to section. `noverlap` must be an positive integer smaller than `window` if `window` is an integer. `noverlap` must be a positive integer less than the length of `window` if `window` is a vector. If you do not specify `noverlap`, or specify `noverlap` as empty, the default number of overlapped samples is 50% of the window length.

`pxx = pwelch(x,window,noverlap,nfft)` specifies the number of discrete Fourier transform (DFT) points to use in the PSD estimate. The default `nfft` is the greater of 256 or the next power of 2 greater than the length of the segments.

`[pxx,w] = pwelch( ___ )` returns the normalized frequency vector, `w`. If `pxx` is a one-sided PSD estimate, `w` spans the interval [0,π] if `nfft` is even and [0,π) if `nfft` is odd. If `pxx` is a two-sided PSD estimate, `w` spans the interval [0,2π).

`[pxx,f] = pwelch( ___ ,fs)` returns a frequency vector, `f`, in cycles per unit time. The sampling frequency, `fs`, is the number of samples per unit time. If the unit of time is seconds, then `f` is in cycles/sec (Hz). For real–valued signals, `f` spans the interval [0,`fs`/2] when `nfft` is even and [0,`fs`/2) when `nfft` is odd. For complex-valued signals, `f` spans the interval [0,`fs`).

`[pxx,w] = pwelch(x,window,noverlap,w)` returns the two-sided Welch PSD estimates at the normalized frequencies specified in the vector, `w`. The vector, `w`, must contain at least 2 elements.

`[pxx,f] = pwelch(x,window,noverlap,f,fs)` returns the two-sided Welch PSD estimates at the frequencies specified in the vector, `f`. The vector, `f`, must contain at least 2 elements. The frequencies in `f` are in cycles per unit time. The sampling frequency, `fs`, is the number of samples per unit time. If the unit of time is seconds, then `f` is in cycles/sec (Hz).

`[ ___ ] = pwelch(x,window, ___ ,freqrange)` returns the Welch PSD estimate over the frequency range specified by `freqrange`. Valid options for `freqrange` are: `'onesided'`, `'twosided'`, or `'centered'`.

`[ ___ ] = pwelch(x,window, ___ ,spectrumtype)` returns the PSD estimate if `spectrumtype` is specified as `'psd'` and returns the power spectrum if `spectrumtype` is specified as `'power'`.

`[pxx,f,pxxc] = pwelch( ___ ,'ConfidenceLevel', probability)` returns the `probability`x100% confidence intervals for the PSD estimate in `pxxc`.

`pwelch( ___ )` with no output arguments plots the Welch PSD estimate in the current figure window.

Input Arguments

x - Input signal

vector

Input signal, specified as a row or column vector.

Data Types
`single | double`
Complex Number Support: Yes

window - Window

integer | vector | `[]`

Window, specified as a row or column vector or an integer. If `window` is a vector, `pwelch` divides `x` into overlapping sections of length equal to the length of `window`, and then multiplies each signal section with the vector specified in `window`. If `window` is an integer, `pwelch` is divided into sections of length equal to the integer value, and a Hamming window of equal length is used. If the length of `x` cannot be divided exactly into an integer number of sections with `noverlap` number of overlapping samples, `x` is truncated accordingly. If you specify `window` as empty, the default Hamming window is used to obtain eight sections of `x` with `noverlap` overlapping samples.

Data Types
`single | double`

noverlap - Number of overlapped samples

positive integer | `[]`

Number of overlapped samples, specified as a positive integer smaller than the length of `window`. If you omit `noverlap` or specify `noverlap` as empty, a value is used to obtain 50% overlap between segments.

nfft - Number of DFT points

`max(256,2^nextpow2(length(window))` (default) | integer | `[]`

Number of DFT points, specified as a positive integer. For a real-valued input signal, `x`, the PSD estimate, `pxx` has length (`nfft`/2+1) if `nfft` is even, and (`nfft`+1)/2 if `nfft` is odd. For a complex-valued input signal,`x`, the PSD estimate always has length `nfft`. If `nfft` is specified as empty, the default `nfft` is used.

If `nfft` is greater than the segment length, the data is zero-padded. If `nfft` is less than the segment length, the segment is wrapped using `datawrap` to make the length equal to `nfft`.

Data Types
`single | double`

fs - Sampling frequency

positive scalar

Sampling frequency, specified as a positive scalar. The sampling frequency is the number of samples per unit time. If the unit of time is seconds, the sampling frequency has the units hertz.

w - Normalized frequencies for Goertzel algorithm

vector

Normalized frequencies for Goertzel algorithm, specified as a row or column vector with at least 2 elements. Normalized frequencies are in radians/sample.

Example: `w = [pi/4 pi/2]`

Data Types
`double`

f - Cyclical frequencies for Goertzel algorithm

vector

Cyclical frequencies for Goertzel algorithm, specified as a row or column vector with at least 2 elements. The frequencies are in cycles per unit time. The unit time is specified by the sampling frequency, `fs`. If `fs` has units of samples/second, then `f` has units of Hz.

Example: `fs = 1000; f= [100 200]`

Data Types
`double`

freqrange - Frequency range for PSD estimate

`'onesided' | 'twosided' | 'centered'`

Frequency range for the PSD estimate, specified as a one of `'onesided'`, `'twosided'`, or `'centered'`. The default is `'onesided'` for real-valued signals and `'twosided'` for complex-valued signals. The frequency ranges corresponding to each option are

- `'onesided'` — returns the one-sided PSD estimate of a real-valued input signal, `x`. If `nfft` is even, `pxx` will have length `nfft`/2+1 and is computed over the interval [0,π] radians/sample. If `nfft` is odd, the length of `pxx` is (`nfft`+1)/2 and the interval is [0,π) radians/sample. When `fs` is optionally specified, the corresponding intervals are [0,`fs`/2] cycles/unit time and [0,`fs`/2) cycles/unit time for even and odd length `nfft` respectively.
- `'twosided'` — returns the two-sided PSD estimate for either the real-valued or complex-valued input, `x`. In this case, `pxx` has length `nfft` and is computed over the interval [0,2π) radians/sample. When `fs` is optionally specified, the interval is [0,`fs`) cycles/unit time.
- `'centered'` — returns the centered two-sided PSD estimate for either the real-valued or complex-valued input, `x`. In this case,

`pxx` has length `nfft` and is computed over the interval (-π,π] radians/sample for even length `nfft` and (-π,π) radians/sample for odd length `nfft`. When `fs` is optionally specified, the corresponding intervals are (-`fs`/2, `fs`/2] cycles/unit time and (-`fs`/2, `fs`/2) cycles/unit time for even and odd length `nfft` respectively.

Data Types
`char`

spectrumtype - Power spectrum scaling
`'psd'` (default) | `'power'`

Power spectrum scaling, specified as one of `'psd'` or `'power'`. Omitting the `spectrumtype`, or specifying `'psd'`, returns the power spectral density. Specifying `'power'` scales each estimate of the PSD by the equivalent noise bandwidth of the window. Use the `'power'` option to obtain an estimate of the power at each frequency.

Data Types
`char`

probability - Confidence interval for PSD estimate
0.95 (default) | Scalar in the range (0,1)

Coverage probability for the true PSD, specified as a scalar in the range (0,1). The output, `pxxc`, contains the lower and upper bounds of the `probability`x100% interval estimate for the true PSD.

Output Arguments

pxx - PSD estimate
vector

PSD estimate, specified as a real-valued, nonnegative column vector.

Data Types
`single | double`

w - Normalized frequencies
vector

Normalized frequencies, specified as a real-valued column vector. If `pxx` is a one-sided PSD estimate, `w` spans the interval [0,π] if `nfft` is even and [0,π) if `nfft` is odd. If `pxx` is a two-sided PSD estimate, `w` spans the interval [0,2π). For a DC-centered PSD estimate, `f` spans the interval (-π,π] radians/sample for even length `nfft` and (-π,π) radians/sample for odd length `nfft`.

Data Types
`double`

f - Cyclical frequencies
vector

Cyclical frequencies, specified as a real-valued column vector. For a one-sided PSD estimate, `f` spans the interval [0,`fs`/2] when `nfft` is even and [0,`fs`/2) when `nfft` is odd. For a two-sided PSD estimate, `f` spans the interval [0,`fs`). For a DC-centered PSD estimate, `f` spans the interval (-`fs`/2, `fs`/2] cycles/unit time for even length `nfft` and (-`fs`/2, `fs`/2) cycles/unit time for odd length `nfft` .

Data Types
`double`

pxxc - Confidence bounds
matrix

Confidence bounds, specified as an N-by-2 matrix with real-valued elements. The row dimension of the matrix is equal to the length of the PSD estimate, `pxx`. The first column contains the lower confidence bound and the second column contains the upper confidence bound for the corresponding PSD estimates in the rows of `pxx`. The coverage probability of the confidence intervals is determined by the value of the `probability` input.

Data Types
`single | double`

Examples

Welch Estimate Using Default Inputs

Obtain the Welch PSD estimate of an input signal consisting of a discrete-time sinusoid with an angular frequency of π/4 radians/sample with additive N(0,1) white noise.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the Welch PSD estimate using the default Hamming window and DFT length. The default segment length is 71 samples and the DFT length is the 256 points yielding a frequency resolution of 2π/256 radians/sample. Because the signal is real-valued, the periodogram is one-sided and there are 256/2+1 points.

```
n = 0:319;
x = cos(pi/4*n)+randn(size(n));
pxx = pwelch(x);
plot(10*log10(pxx))
```

Welch Estimate Using Specified Segment Length

Obtain the Welch PSD estimate of an input signal consisting of a discrete-time sinusoid with an angular frequency of π/4 radians/sample with additive N(0,1) white noise.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the Welch PSD estimate dividing the signal into segments 100 samples in length. The signal segments are multiplied by a Hamming window 100 samples in length. The number of overlapped samples is 50. The DFT length is 256 points yielding a frequency resolution of 2π/256 radians/sample. Because the signal is real-valued, the PSD estimate is one-sided and there are 256/2+1 points.

```
n = 0:319;
x = cos(pi/4*n)+randn(size(n));
segmentLength = 100;
pxx = pwelch(x,segmentLength);
plot(10*log10(pxx))
```

Welch Estimate Specifying Segment Overlap

Obtain the Welch PSD estimate of an input signal consisting of a discrete-time sinusoid with an angular frequency of π/4 radians/sample with additive N(0,1) white noise.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the Welch PSD estimate dividing the signal into segments 100 samples in length. The signal segments are multiplied by a Hamming window 100 samples in length. The number of overlapped samples is 25. The DFT length is 256 points yielding a frequency resolution of 2π/256 radians/sample. Because the signal is real-valued, the PSD estimate is one-sided and there are 256/2+1 points.

```
n = 0:319;
x = cos(pi/4*n)+randn(size(n));
segmentLength = 100;
noverlap = 25;
pxx = pwelch(x,segmentLength,noverlap);
plot(10*log10(pxx))
```

Welch Estimate Using Specified DFT Length

Obtain the Welch PSD estimate of an input signal consisting of a discrete-time sinusoid with an angular frequency of π/4 radians/sample with additive N(0,1) white noise.

Create a sine wave with an angular frequency of π/4 radians/sample with additive N(0,1) white noise. The signal is 320 samples in length. Obtain the Welch PSD estimate dividing the signal into segments 100 samples in length. Use the default overlap of 50%. Specify the DFT length to be 640 points so that the frequency of π/4 radians/sample corresponds to a DFT bin (bin 81).Because the signal is real-valued, the PSD estimate is one-sided and there are 640/2+1 points.

```
n = 0:319;
x = cos(pi/4*n)+randn(size(n));
segmentLength = 100;
```

```
nfft = 640;
pxx = pwelch(x,segmentLength,[],nfft);
plot(10*log10(pxx));
xlabel('Radians/sample'); ylabel('dB');
```

Welch PSD Estimate of Signal with Frequency in Hz

Create a signal consisting of a 100-Hz sinusoid in additive N(0,1) white noise. The sampling rate is 1 kHz and the signal is 5 seconds in duration.

```
fs = 1000;
t = 0:1/fs:5-1/fs;
x = cos(2*pi*100*t)+randn(size(t));
```

Obtain Welch's overlapped segment averaging PSD estimate of the preceding signal. Use a segment length of 500 samples with 300 overlapped samples. Use 500 DFT points so that 100 Hz falls directly on a DFT bin. Input the sampling frequency to output a vector of frequencies in Hz. Plot the result.

```
[pxx,f] = pwelch(x,500,300,500,fs);
plot(f,10*log10(pxx))
xlabel('Hz'); ylabel('dB');
```

DC-Centered Power Spectrum

Create a signal consisting of a 100-Hz sinusoid in additive N(0,1/4) white noise. The sampling rate is 1 kHz and the signal is 5 seconds in duration.

```
fs = 1000;
t = 0:1/fs:5-1/fs;
noisevar = 1/4;
x = cos(2*pi*100*t)+sqrt(noisevar)*randn(size(t));
```

Obtain the DC-centered power spectrum using Welch's method. Use a segment length of 500 samples with 300 overlapped samples and a DFT length of 500 points. Plot the result.

```
[pxx,f] = pwelch(x,500,300,500,fs,'centered','power');
plot(f,pxx);
xlabel('Hz'); ylabel('dB');
grid on;
```

You see that the power at –100 and 100 Hz is close to the expected power of 1/4 for a real-valued sine wave with an amplitude of 1. The deviation from 1/4 is due to the effect of the additive noise.

Upper and Lower 95%-Confidence Bounds

The following example illustrates the use of confidence bounds with Welch's overlapped segment averaging (WOSA) PSD estimate. While not a necessary condition for statistical significance, frequencies in Welch's estimate where the lower confidence bound exceeds the upper confidence bound for surrounding PSD estimates clearly indicate significant oscillations in the time series.

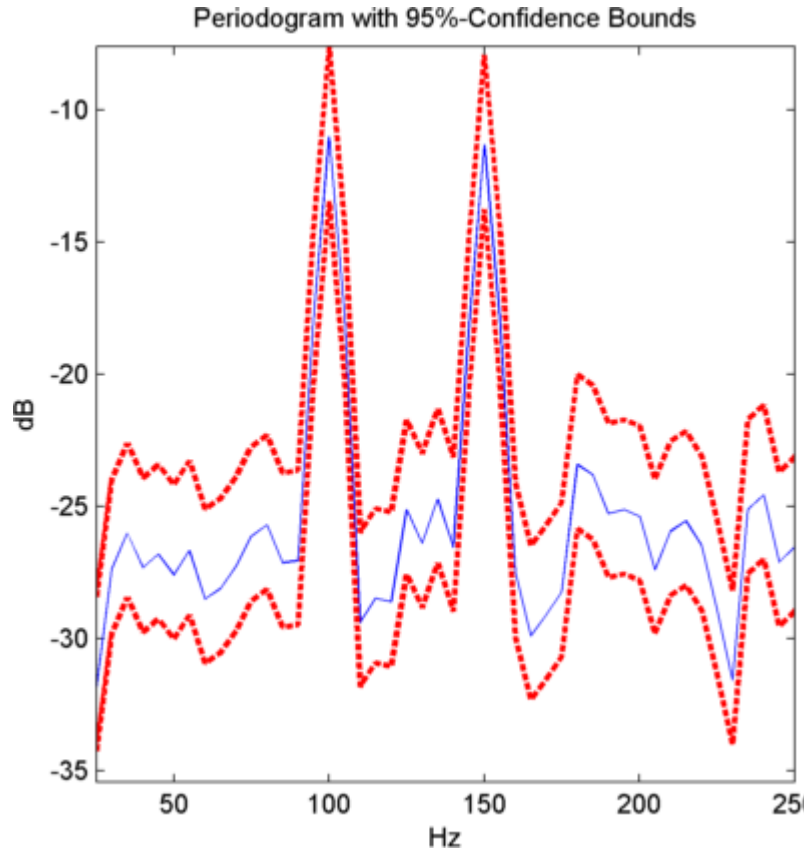
Create a signal consisting of the superposition of 100-Hz and 150-Hz sine waves in additive white N(0,1) noise. The amplitude of the two sine waves is 1. The sampling frequency is 1 kHz.

```
t = 0:0.001:1-0.001;
fs = 1000;
x = cos(2*pi*100*t)+sin(2*pi*150*t)+randn(size(t));
```

Obtain the WOSA estimate with 95%-confidence bounds. Set the segment length equal to 200 and overlap the segments by 50% (100 samples). Plot the WOSA PSD estimate along with the confidence interval and zoom in on the frequency region of interest near 100 and 150 Hz.

```
L = 200;
noverlap = 100;
[pxx,f,pxxc] = pwelch(x,hamming(L),noverlap,200,fs,...
    'ConfidenceLevel',0.95);
plot(f,10*log10(pxx)); hold on;
plot(f,10*log10(pxxc),'r--','linewidth',2);
axis([25 250 min(min(10*log10(pxxc))) max(max(10*log10(pxxc)))]);
```

```
xlabel('Hz'); ylabel('dB');
title('Welch Estimate with 95%-Confidence Bounds');
```



At 100 and 150 Hz, the lower confidence bound exceeds the upper confidence bounds for surrounding PSD estimates.

Definitions

Welch's Overlapped Segment Averaging (WOSA) Spectral Estimation

The periodogram is not a consistent estimator of the true power spectral density of a wide-sense stationary process. Welch's technique to reduce the variance of the periodogram breaks the time series into segments, usually overlapping. Welch's method computes a modified periodogram for each segment and then averages these estimates to produce the estimate of the power spectral density. Because the process is wide-sense stationary and Welch's method uses PSD estimates of different segments of the time series, the modified periodograms represent approximately uncorrelated estimates of the true PSD and averaging reduces the variability.

The segments are typically multiplied by a window function, such as a Hamming window, so that Welch's method amounts to averaging modified periodograms. Because the segments usually overlap, data values at the beginning and end of the segment tapered by the window in one segment, occur away from the ends of adjacent segments. This guards against the loss of information caused by windowing.

See Also

`periodogram | pmtm`

Related Examples

- "Bias and Variability in the Periodogram"

Concepts

- "Spectral Analysis"

Purpose

PSD using Yule-Walker AR method

Syntax

```
Pxx = pyulear(x,order)
Pxx = pyulear(x,order,nfft)
[Pxx,w] = pyulear(...)
[Pxx,w] = pyulear(x,order,w)
Pxx = pyulear(x,order,nfft,fs)
Pxx = pyulear(x,order,f,fs)
[Pxx,f] = pyulear(x,order,nfft,fs)
[Pxx,f] = pyulear(x,order,f,fs)
[Pxx,f] = pyulear(x,order,nfft,fs,freqrange)
[Pxx,w] = pyulear(x,order,nfft,freqrange)
[Pxx,f,Pxxc] = pyulear(...,'ConfidenceLevel',P)
pyulear(...)
```

Description

`Pxx = pyulear(x,order)` implements the Yule-Walker algorithm, a parametric spectral estimation method, and returns `Pxx`, an estimate of the power spectral density (PSD) of the vector `x`. The entries of `x` represent samples of a discrete-time signal. `order` is the integer specifying the order of an autoregressive (AR) prediction model for the signal, used in estimating the PSD. This estimate is also an estimate of the maximum entropy.

The power spectral density is calculated in units of power per radians per sample. Real-valued inputs produce full power one-sided (in frequency) PSDs (by default), while complex-valued inputs produce two-sided PSDs.

In general, the length of the FFT and the values of the input `x` determine the length of `Pxx` and the range of the corresponding normalized frequencies. For this syntax, the (default) FFT length is 256. The following table indicates the length of `Pxx` and the range of the corresponding normalized frequencies for this syntax.

PSD Vector Characteristics for an FFT Length of 256 (Default)

Real/Complex Input Data	Length of Pxx	Range of the Corresponding Normalized Frequencies
Real-valued	129	[0, π]
Complex-valued	256	[0, 2π)

`Pxx = pyulear(x,order,nfft)` uses the integer FFT length `nfft` to calculate the PSD vector `Pxx`.

`[Pxx,w] = pyulear(...)` also returns `w`, a vector of normalized angular frequencies at which the two-sided PSD is estimated. `Pxx` and `w` have the same length. The units for `w` are rad/sample.

The length of `Pxx` and the frequency range for `w` depend on `nfft` and the values of the input `x`. The following table indicates the length of `Pxx` and the frequency range for `w` in this syntax.

PSD and Frequency Vector Characteristics

Real/Complex Input Data	nfft Even/Odd	Length of Pxx	Range of w
Real-valued	Even	`(nfft/2 + 1)`	[0, π]
Real-valued	Odd	`(nfft + 1)/2`	[0, π)
Complex-valued	Even or odd	`nfft`	[0, 2π)

`[Pxx,w] = pyulear(x,order,w)` uses a vector of normalized frequencies `w` with two or more elements to compute the PSD at those frequencies and returns a two-sided PSD.

```
Pxx = pyulear(x,order,nfft,fs)
```

or

`Pxx = pyulear(x,order,f,fs)` uses the integer FFT length `nfft` to calculate the PSD vector `Pxx` or uses the vector of frequencies `f` in Hz and the sampling frequency `fs` to compute the two-sided PSD vector `Pxx` at those frequencies. If you specify `nfft` as the empty vector `[]`, it uses the default value of 256. If you specify `fs` as the empty vector `[]`, the sampling frequency `fs` defaults to 1 Hz. The spectral density produced is calculated in units of power per Hz.

`[Pxx,f] = pyulear(x,order,nfft,fs)`

or

`[Pxx,f] = pyulear(x,order,f,fs)` returns the frequency vector f. In this case, the units for the frequency vector are in Hz. The frequency range for `f` depends on `nfft`, `fs`, and the values of the input `x`. The length of `Pxx` is the same as in the table above. The following table indicates the frequency range for `f` for this syntax.

PSD and Frequency Vector Characteristics with fs Specified

Real/Complex Input Data	nfft Even/Odd	Range of f
Real-valued	Even	`[0,fs/2]`
Real-valued	Odd	`[0,fs/2)`
Complex-valued	Even or odd	`[0,fs)`

`[Pxx,f] = pyulear(x,order,nfft,fs,freqrange)` or

`[Pxx,w] = pyulear(x,order,nfft,freqrange)` specifies the range of frequency values to include in the output frequency vectors, `f` or `w`. This syntax is useful when `x` is real. `freqrange` can be either:

- `'onesided'` — returns the one-sided PSD of a real input signal, `x`. If `nfft` is even, `Pxx` has length `nfft/2+1` and is computed over the interval [0,π]. If `nfft` is odd, the length of `Pxx` is `(nfft+1)/2` and the frequency interval is [0,π). When your specify `fs` , the intervals are [0,`fs`/2] and [0,`fs`/2) for even and odd length `nfft` respectively.

- `'twosided'` — returns the two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval [0,2π). When you specify `fs`, the frequency interval is [0,`fs`).
- `'centered'` — returns the centered two-sided PSD for either real or complex input, `x`. In this case, `Pxx` has length `nfft` and is computed over the interval (-π, π] for even length `nfft` and (-π, π) for odd length `nfft`. When you specify `fs`, the frequency intervals are (-`fs`/2, `fs`/2] and (-`fs`/2,`fs`/2) for even and odd length `nfft` respectively.

`[Pxx,f,Pxxc] = pyulear(...,'ConfidenceLevel',P)` returns the P100% confidence interval for `Pxx`, where `P` is a nonnegative scalar between 0 and 1. The default value for `P` is 0.95. Large-sample confidence intervals are computed using a Gaussian probability density function. `Pxxc` is N-by-2 matrix, where N is the length of `Pxx`. The first column, `Pxxc(:,1)`, is the lower bound of the confidence interval. The second column, `Pxxc(:,2)`, is the upper bound. See [1] for a description of approximate large-sample confidence intervals for AR PSD estimates.

`pyulear(...)` with no outputs plots the PSD in the current figure window. The frequency range on the plot is the same as the range of output `w` (or `f`) for a given set of parameters.

Tips

The power spectral density is computed as the distribution of power per unit frequency.

This algorithm depends on your selecting an appropriate model order for your signal.

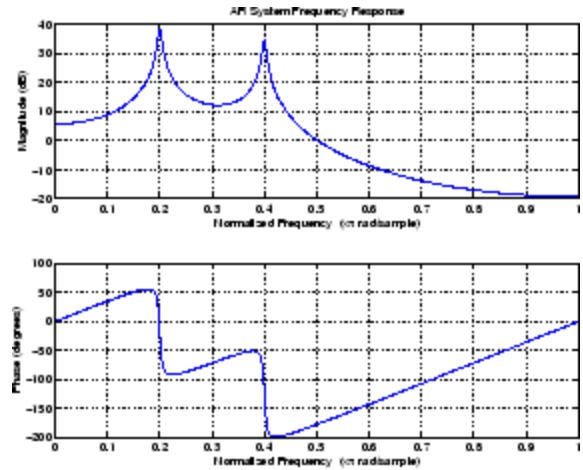
Examples

Yule-Walker AR PSD Estimate

Because the Yule-walker method estimates the spectral density by fitting an AR prediction model of a given order to the signal, first generate a signal from an AR (all-pole) model of a given order. You can use `freqz` to check the magnitude of the frequency response of your AR filter. This will give you an idea of what to expect when you estimate the PSD using `pyulear`:

```
% AR filter coefficients
```

```
a = [1 -2.2137 2.9403 -2.1697 0.9606];
% AR filter frequency response
freqz(1,a)
title('AR System Frequency Response')
```



Now generate the input signal `x` by filtering white noise through the AR filter. Estimate the PSD of `x` based on a fourth-order AR prediction model, since in this case, we know that the original AR system model `a` has order 4:

```
x = filter(1,a,randn(256,1));     % AR system output
pyulear(x,4)                      % Fourth-order estimate
```

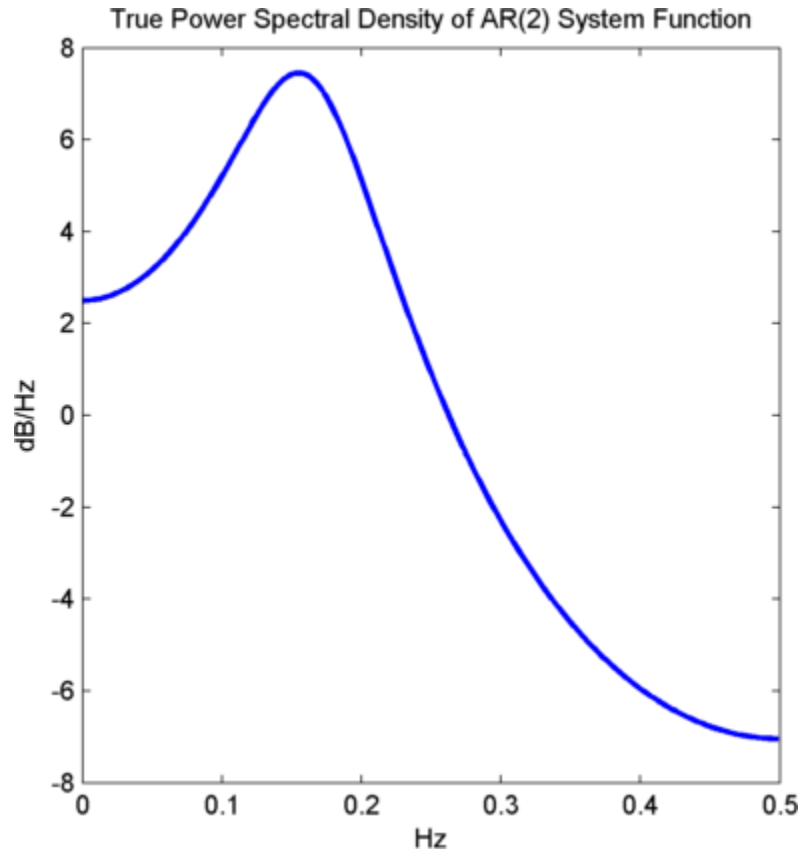
Large-Sample Confidence Intervals for AR PSD Estimate

This example shows you how to obtain and plot confidence intervals for an AR PSD estimate.

Create the coefficients for an AR(2) system function. Use `freqz` to obtain and plot the true power spectral density.

```
A = [1 -0.75 0.5];
[H,F] = freqz(1,A,[],1);
```

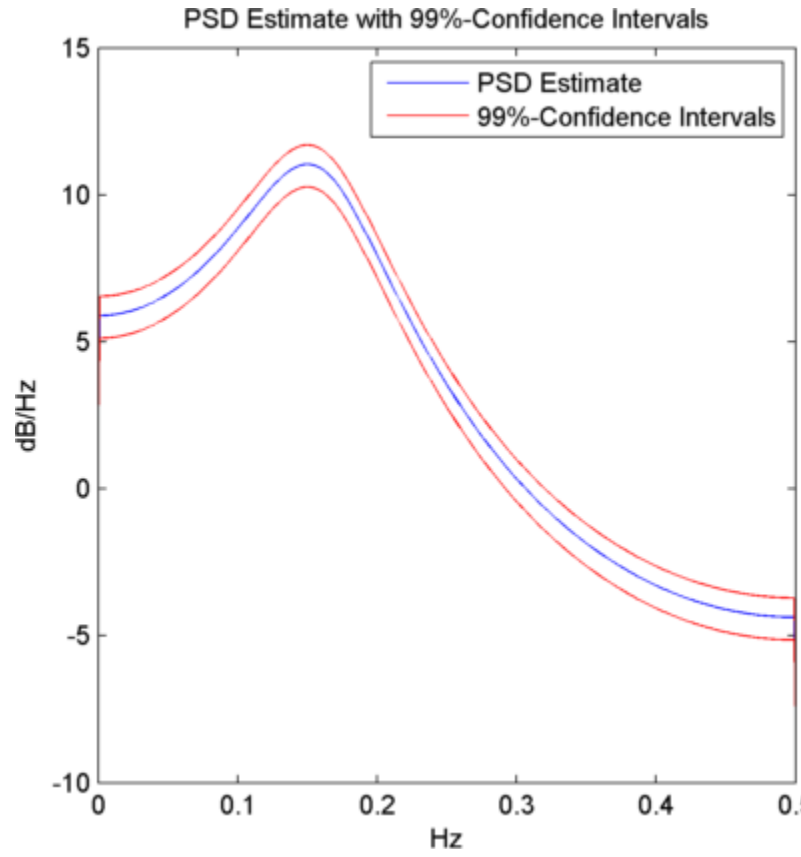
```
plot(F,20*log10(abs(H)),'b','linewidth',2);
xlabel('Hz'); ylabel('dB/Hz');
title('True Power Spectral Density of AR(2) System Function')
```



Create a realization of the AR(2) process represented by the coefficients. Set the random number generator to the default settings for reproducible results. Obtain approximate large-sample 99%-confidence intervals for the PSD estimate.

```
rng default;
```

```
x = randn(1000,1);
y = filter(1,A,x);
[Pxx,F,Pxxc] = pyulear(y,2,1024,1,'ConfidenceLevel',0.99);
plot(F,10*log10(Pxx),'b'); hold on;
plot(F,10*log10(Pxxc),'r'); xlabel('Hz'); ylabel('dB/Hz');
legend('PSD Estimate', '99%-Confidence Intervals')
title('PSD Estimate with 99%-Confidence Intervals')
```



Algorithms

Linear prediction filters can be used to model the second-order statistical characteristics of a signal. The prediction filter output can be used to model the signal when the input is white noise.

`pyulear` estimates the PSD of an input signal vector using the Yule-Walker AR method. This method, also called the autocorrelation or windowed method, fits an autoregressive (AR) linear prediction filter model to the signal by minimizing the forward prediction error (based on all observations of the in put sequence) in the least squares sense. This formulation leads to the Yule-Walker equations, which are solved by the Levinson-Durbin recursion. The spectral estimate returned by `pyulear` is the squared magnitude of the frequency response of this AR model.

References

[1] Kay, S.M. *Modern Spectral Estimation*, Englewood Cliffs, NJ, Prentice-Hall, 1988, pp. 194–195.

[2] Marple, S.L., *Digital Spectral Analysis*, Prentice-Hall, 1987, Chapter 7.

[3] Stoica, P., and R.L. Moses, *Introduction to Spectral Analysis*, Prentice-Hall, 1997.

See Also

`aryule` | `lpc` | `pyulear` | `pcov` | `peig` | `periodogram` | `pmcov` | `pmtm` | `pmusic` | `prony` | `pwelch`

realizemdl

Purpose

Simulink subsystem block for filter

Syntax

```
realizemdl(FiltObject)
realizemdl(FiltObject,propertyname1,propertyvalue1,...)
```

Description

`realizemdl(FiltObject)` generates a model of the filter object `FiltObject` in a Simulink subsystem block using sum, gain, and delay blocks from Simulink. The properties and values of `FiltObject` define the resulting subsystem block parameters.

`realizemdl` requires Simulink. To accurately realize models of quantized filters, use Fixed-Point Designer.

`realizemdl(FiltObject,propertyname1,propertyvalue1,...)` generates the model for`FiltObject` with the associated `propertyname`/`propertyvalue` pairs, and any other values you set in `FiltObject`.

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'`or *`Subsystemname`*	Specify whether to add the block to your current Simulink model or create a new model to contain the block. If you provide the name of a current subsystem in *`subsystemname`*, `realizemdl` adds the new block to the specified subsystem.
`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`*'.
`MapCoeffstoPorts`	`'off'` (default) or `'on'`	Specify whether to map the coefficients of the filter to the ports of the block.
`MapStates`	`'off'` (default) or `'on'`	Specifies whether to apply the current filter states to the realized model. 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 model. Setting the property to `'on'` preserves the current filter states in the realized model.

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.
`CoeffNames`	`{'Num'}` (default FIR),`{'Num','Den'}` (default direct form IIR),`{'Num','Den','g'}` (default IIR SOS), `{'K'}` (default form lattice)	Specify the coefficient variable names as string variables in a cell array.`MapCoeffsToPorts` must be set to `'on'` for this property to apply.

Property Name	Property Values	Description
`InputProcessing`	`'columnsaschannels'` (default), `'elementsaschannels'` , or `'inherited'`	Specify frame-based (`'columnsaschannels'`) or sample-based (`'elementsaschannels'`) processing. The `Inherited (this choice will be removed - see release notes)` option will be removed in a future release. For more information, see “” for frame-based processing in the *DSP System Toolbox Release Notes*.
`RateOption`	`'enforcesinglerate'` (default) or `'allowmultirate'`	Specify how the block adjusts the rate at the output to accommodate the reduced number of samples. This parameter applies only when `InputProcessing` is `'columnsaschannels'`.

Examples

Realize Simulink model of lowpass Butterworth filter:

```
Hd = fdesign.lowpass('N,F3dB',4,0.25);
d = design(Hd,'butter');
realizemdl(d);
```

Realize Simulink model with coefficients mapped to ports:

```
Hd = fdesign.lowpass('N,F3dB',4,0.25);
d = design(Hd,'butter');
%Realize Simulink model and export coefficients
```

```
realizemdl(d,'MapCoeffsToPorts','on');
```

In this case, the filter is an IIR filter with a direct form II second-order sections structure. Setting `MapCoeffstoPorts` to `'on'` exports the numerator coefficients, the denominator coefficients, and the gains to the MATLAB workspace using the default variable names `Num`, `Den`, and `g`. Each column of `Num` and `Den` represents one second-order section. You can modify the filter coefficients directly in the MATLAB workspace providing tunability to the realized Simulink model.

See Also

`block` | `design` | `fdesign`

Purpose

Convert reflection coefficients to autocorrelation sequence

Syntax

```
r = rc2ac(k,r0)
```

Description

`r = rc2ac(k,r0)` finds the autocorrelation coefficients, `r`, of the output of the discrete-time prediction error filter from the lattice-form reflection coefficients `k` and initial zero-lag autocorrelation `r0`.

Examples

```
k = [0.3090    0.9800    0.0031    0.0082   -0.0082];
r0 = 0.1;
a = rc2ac(k,r0)
```

References

[1] Kay, S.M., *Modern Spectral Estimation*, Prentice-Hall, Englewood Cliffs, NJ, 1988.

See Also

`ac2rc` | `poly2ac` | `rc2poly`

rc2is

Purpose

Convert reflection coefficients to inverse sine parameters

Syntax

```
isin = is2rc(k)
```

Description

`isin = is2rc(k)` returns a vector of inverse sine parameters `isin` from a vector of reflection coefficients `k`.

Examples

```
k = [0.3090 0.9801 0.0031 0.0082 -0.0082];
isin = rc2is(k)
```

References

[1] Deller, J.R., J.G. Proakis, and J.H.L. Hansen, *Discrete-Time Processing of Speech Signals*, Prentice-Hall, 1993.

See Also

`is2rc`

Purpose

Convert reflection coefficients to log area ratio parameters

Syntax

```
g = rc2lar(k)
```

Description

`g = rc2lar(k)` returns a vector of log area ratio parameters `g` from a vector of reflection coefficients `k`.

Examples

```
k = [0.3090 0.9801 0.0031 0.0082 -0.0082];
g = rc2lar(k)
```

References

[1] Deller, J.R., J.G. Proakis, and J.H.L. Hansen, *Discrete-Time Processing of Speech Signals*, Prentice-Hall, 1993.

See Also

`lar2rc`

rc2poly

Purpose

Convert reflection coefficients to prediction filter polynomial

Syntax

```
a = rc2poly(k)
[a,efinal] = rc2poly(k,r0)
```

Description

`a = rc2poly(k)` converts the reflection coefficients `k` corresponding to the lattice structure to the prediction filter polynomial `a`, with `a(1) = 1`. The output `a` is row vector of length `length(k)+1`.

`[a,efinal] = rc2poly(k,r0)` returns the final prediction error `efinal` based on the zero-lag autocorrelation, `r0`.

Examples

Consider a lattice IIR filter given by reflection coefficients `k`:

```
k = [0.3090    0.9800    0.0031    0.0082   -0.0082];
```

Its equivalent prediction filter representation is given by

```
a = rc2poly(k)
```

Algorithms

`rc2poly` computes output `a` using Levinson's recursion [1]. The function

1. Sets the output vector `a` to the first element of `k`.
2. Loops through the remaining elements of `k`.

   For each loop iteration `i`, `a = [a + a(i-1:-1:1)*k(i) k(i)]`.
3. Implements `a = [1 a]`.

References

[1] Kay, S.M., *Modern Spectral Estimation*, Prentice-Hall, Englewood Cliffs, NJ, 1988.

See Also

`ac2poly` | `latc2tf` | `latcfilt` | `poly2rc` | `rc2ac` | `rc2is` | `rc2lar` | `tf2latc`

Purpose

Real cepstrum and minimum phase reconstruction

Syntax

```
rceps(x)
[y,ym] = rceps(x)
```

Description

The *real cepstrum* is the inverse Fourier transform of the real logarithm of the magnitude of the Fourier transform of a sequence.

Note `rceps` only works on real data.

`rceps(x)` returns the real cepstrum of the real sequence `x`. The real cepstrum is a real-valued function.

`[y,ym] = rceps(x)` returns both the real cepstrum `y` and a minimum phase reconstructed version `ym` of the input sequence.

Algorithms

`rceps` is an implementation of algorithm 7.2 in [2], that is,

```
y = real(ifft(log(abs(fft(x)))));
```

Appropriate windowing in the cepstral domain forms the reconstructed minimum phase signal:

```
w = [1;2*ones(n/2-1,1);ones(1-rem(n,2),1);zeros(n/2-1,1)];
ym = real(ifft(exp(fft(w.*y))));
```

References

[1] Oppenheim, A.V., and R.W. Schafer, *Digital Signal Processing*, Englewood Cliffs, NJ, Prentice-Hall, 1975.

[2] *Programs for Digital Signal Processing*, IEEE Press, New York, 1979.

See Also

`cceps` | `fft` | `hilbert` | `icceps` | `unwrap`

rectpuls

Purpose

Sampled aperiodic rectangle

Syntax

```
y = rectpuls(t)
y = rectpuls(t,w)
```

Description

`y = rectpuls(t)` returns a continuous, aperiodic, unity-height rectangular pulse at the sample times indicated in array `t`, centered about `t = 0` and with a default width of 1. Note that the interval of non-zero amplitude is defined to be open on the right, that is, `rectpuls(-0.5) = 1` while `rectpuls(0.5) = 0`.

`y = rectpuls(t,w)` generates a rectangle of width `w`.

`rectpuls` is typically used in conjunction with the pulse train generating function `pulstran`.

See Also

`chirp` | `cos` | `diric` | `gauspuls` | `pulstran` | `sawtooth` | `sin` | `sinc` | `square` | `tripuls`

Purpose

Rectangular window

Syntax

```
w = rectwin(L)
```

Description

`w = rectwin(L)` returns a rectangular window of length `L` in the column vector `w`. This function is provided for completeness; a rectangular window is equivalent to no window at all.

Algorithms

```
w = ones(L,1);
```

References

[1] Oppenheim, A.V., and R.W. Schafer. *Discrete-Time Signal Processing*. Upper Saddle River, NJ: Prentice-Hall, 1999, pp. 468-471.

See Also

`barthannwin` | `bartlett` | `blackmanharris` | `bohmanwin` | `nuttallwin` | `parzenwin` | `triang` | `window` | `wintool` | `wvtool`

resample

Purpose

Change sampling rate by rational factor

Syntax

```
y = resample(x,p,q)
y = resample(x,p,q,n)
y = resample(x,p,q,n,beta)
y = resample(x,p,q,b)
[y,b] = resample(x,p,q)
```

Description

`y = resample(x,p,q)` resamples the sequence in vector `x` at `p/q` times the original sampling rate, using a polyphase filter implementation. `p` and `q` must be positive integers. The length of `y` is equal to `ceil(length(x)*p/q)`. If `x` is a matrix, `resample` works down the columns of `x`.

`resample` applies an anti-aliasing (lowpass) FIR filter to `x` during the resampling process. It designs the filter using `firls` with a Kaiser window.

`y = resample(x,p,q,n)` uses `n` terms on either side of the current sample, `x(k)`, to perform the resampling. The length of the FIR filter `resample` uses is proportional to `n`; larger values of `n` provide better accuracy at the expense of more computation time. The default for `n` is 10. If you let `n = 0`, resample performs a nearest-neighbor interpolation

```
y(k) = x(round((k-1)*q/p)+1)
```

where `y(k) = 0` if the index to `x` is greater than `length(x)`.

`y = resample(x,p,q,n,beta)` uses beta as the design parameter for the Kaiser window that `resample` employs in designing the lowpass filter. The default for `beta` is 5.

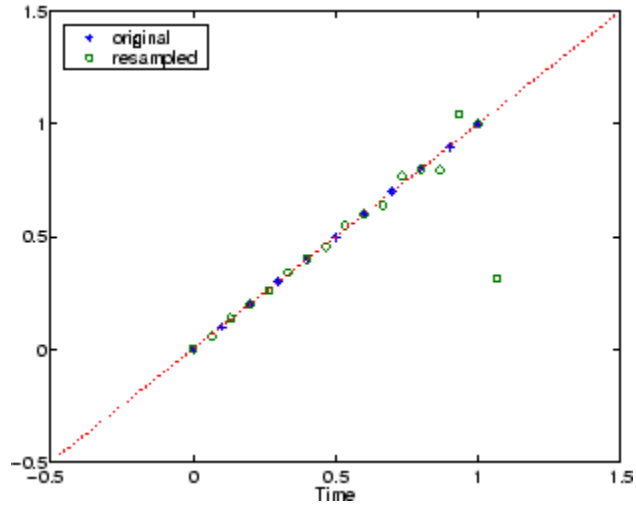
`y = resample(x,p,q,b)` filters `x` using the vector of filter coefficients `b`.

`[y,b] = resample(x,p,q)` returns the vector `b`, which contains the coefficients of the filter applied to `x` during the resampling process.

Examples

Resample a simple linear sequence at 3/2 the original rate:

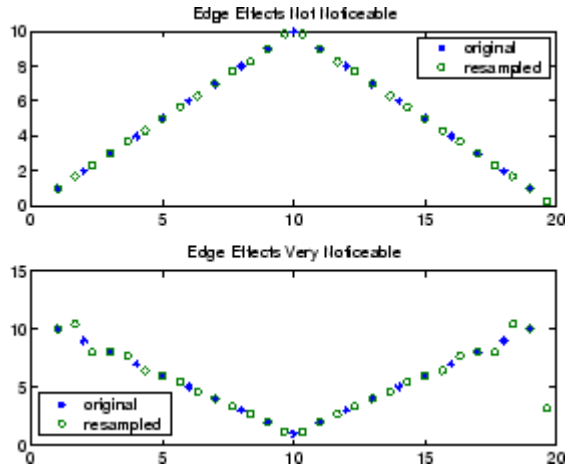
```
fs1 = 10;              % Original sampling frequency in Hz
t1 = 0:1/fs1:1;        % Time vector
x = t1;                % Define a linear sequence
y = resample(x,3,2);  % Now resample it
t2 = (0:(length(y)-1))*2/(3*fs1);  % New time vector
plot(t1,x,'*',t2,y,'o',-0.5:0.01:1.5,-0.5:0.01:1.5,':')
legend('original','resampled'); xlabel('Time')
```



Notice that the last few points of the output `y` are inaccurate. In its filtering process, `resample` assumes the samples at times before and after the given samples in `x` are equal to zero. Thus large deviations from zero at the end points of the sequence `x` can cause inaccuracies in `y` at its end points. The following two plots illustrate this side effect of `resample`:

```
x = [1:10 9:-1:1]; y = resample(x,3,2);
subplot(2,1,1);
plot(1:19,x,'*',(0:28)*2/3 + 1,y,'o');
title('Edge Effects Not Noticeable');
```

```
legend('original','resampled');
x = [10:-1:1 2:10]; y = resample(x,3,2);
subplot(2,1,2);
plot(1:19,x,'*',(0:28)*2/3 + 1,y,'o')
title('Edge Effects Very Noticeable');
legend('original','resampled');
```



Algorithms

`resample` performs an FIR design using `firls`, followed by rate changing implemented with `upfirdn`.

See Also

`decimate` | `downsample` | `firls` | `interp` | `interp1` | `intfilt` | `kaiser` | `mfilt` | `spline` | `upfirdn` | `upsample`

Purpose

z-transform partial-fraction expansion

Syntax

```
[r,p,k] = residuez(b,a)
[b,a] = residuez(r,p,k)
```

Description

`residuez` converts a discrete time system, expressed as the ratio of two polynomials, to partial fraction expansion, or residue, form. It also converts the partial fraction expansion back to the original polynomial coefficients.

Note Numerically, the partial fraction expansion of a ratio of polynomials is an ill-posed problem. If the denominator polynomial is near a polynomial with multiple roots, then small changes in the data, including roundoff errors, can cause arbitrarily large changes in the resulting poles and residues. You should use state-space (or pole-zero representations instead.

`[r,p,k] = residuez(b,a)` finds the residues, poles, and direct terms of a partial fraction expansion of the ratio of two polynomials, $b(z)$ and $a(z)$. Vectors `b` and `a` specify the coefficients of the polynomials of the discrete-time system $b(z)/a(z)$ in descending powers of z.

$$B(z) = b_0 + b_1 z^{-1} + b_2 z^{-2} + \cdots + b_m z^{-m}$$
$$A(z) = a_0 + a_1 z^{-1} + a_2 z^{-2} + \cdots + a_n z^{-n}$$

If there are no multiple roots and `a > n-1`,

$$\frac{B(z)}{A(z)} = \frac{r(1)}{1 - p(1)z^{-1}} + \cdots + \frac{r(n)}{1 - p(n)z^{-1}} + k(1) + k(2)z^{-1} + \cdots + k(m-n+1)z^{-(m-n)}$$

The returned column vector `r` contains the residues, column vector `p` contains the pole locations, and row vector `k` contains the direct terms. The number of poles is

```
n = length(a)-1 = length(r) = length(p)
```

The direct term coefficient vector `k` is empty if `length(b)` is less than `length(a)`; otherwise:

```
length(k) = length(b) - length(a) + 1
```

If `p(j) = ... = p(j+s-1)` is a pole of multiplicity `s`, then the expansion includes terms of the form

$$\frac{r(j)}{1-p(j)z^{-1}}+\frac{r(j+1)}{(1-p(j)z^{-1})^2}+\cdots+\frac{r(j+s_r-1)}{(1-p(j)z^{-1})^s}$$

`[b,a] = residuez(r,p,k)` with three input arguments and two output arguments, converts the partial fraction expansion back to polynomials with coefficients in row vectors `b` and `a`.

The `residue` function in the standard MATLAB language is very similar to `residuez`. It computes the partial fraction expansion of continuous-time systems in the Laplace domain (see reference [1]), rather than discrete-time systems in the *z*-domain as does `residuez`.

Algorithms

`residuez` applies standard MATLAB functions and partial fraction techniques to find `r`, `p`, and `k` from `b` and `a`. It finds

- The direct terms `a` using `deconv` (polynomial long division) when `length(b)` > `length(a)-1`.
- The poles using `p` = `roots(a)`.
- Any repeated poles, reordering the poles according to their multiplicities.
- The residue for each nonrepeating pole p_i by multiplying $b(z)/a(z)$ by $1/(1 - piz^{-1})$ and evaluating the resulting rational function at $z = p_i$.
- The residues for the repeated poles by solving

  ```
  S2*r2 = h - S1*r1
  ```

for `r2` using `\`. `h` is the impulse response of the reduced $b(z)/a(z)$, `S1` is a matrix whose columns are impulse responses of the first-order systems made up of the nonrepeating roots, and `r1` is a column containing the residues for the nonrepeating roots. Each column of matrix `S2` is an impulse response. For each root p_j of multiplicity s_j, `S2` contains s_j columns representing the impulse responses of each of the following systems.

$$\frac{1}{1-pjz^{-1}}, \frac{1}{(1-pjz^{-1})^2}, \cdots, \frac{1}{(1-pjz^{-1})^{s_j}}$$

The vector `h` and matrices `S1` and `S2` have `n + xtra` rows, where `n` is the total number of roots and the internal parameter `xtra`, set to 1 by default, determines the degree of over-determination of the system of equations.

References

[1] Oppenheim, A.V., and R.W. Schafer, *Digital Signal Processing,* Prentice-Hall, Englewood Cliffs, NJ, 1975, pp. 166-170.

See Also

`convmtx` | `deconv` | `poly` | `prony` | `residue` | `roots` | `ss2tf` | `tf2ss` | `tf2zp` | `tf2zpk` | `zp2ss`

risetime

Purpose

Rise time of positive-going bilevel waveform transitions

Syntax

```
R = risetime(X)
R = risetime(X,FS)
R = risetime(X,T)
[R,LT,UT] = risetime(...)
[R,LT,UT,LL,UL] = risetime(...)
[...] = risetime(...,Name,Value)
risetime(...)
```

Description

`R = risetime(X)` returns a vector, `R`, containing the time each transition of the input bilevel waveform, `X`, takes to cross from the 10% to 90% reference levels. To determine the transitions, `risetime` estimates the state levels of the input waveform by a histogram method. `risetime` identifies all regions that cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-913. Because `risetime` uses interpolation, `R` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`R = risetime(X,FS)` specifies the sampling frequency in hertz. The sampling frequency determines the sample instants corresponding to the elements in `X`. The first sample instant in `X` corresponds to t=0. Because `risetime` uses interpolation, `R` may contain values that do not correspond to sampling instants of the bilevel waveform, `X`.

`R = risetime(X,T)` specifies the sample instants, `T`, as a vector with the same number of elements as `X`.

`[R,LT,UT] = risetime(...)` returns vectors, `LT` and `UT`, whose elements correspond to the time instants where `X` crosses the lower- and upper-percent reference levels.

`[R,LT,UT,LL,UL] = risetime(...)` returns the levels, `LL` and `UL`, that correspond to the lower- and upper-percent reference levels.

`[...] = risetime(...,Name,Value)` returns the rise times with additional options specified by one or more `Name,Value` pair arguments.

`risetime(...)` plots the signal and darkens the regions of each transition where rise time is computed. The plot marks the lower and upper crossings and the associated reference levels. The state levels and the corresponding associated lower- and upper-state boundaries are also plotted.

Input Arguments

X

Bilevel waveform. `X` is a real-valued row or column vector.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

'PctRefLevels'

Reference levels as a percentage of the waveform amplitude. The low-state level is defined to be 0 percent. The high-state level is defined to be 100 percent. The value of `'PCTREFLEVELS'` is a two-element real row vector whose elements correspond to the lower and upper percent reference levels.

Default: `[10 90]`

'StateLevels'

Low- and high-state levels. Specifies the levels to use for the low- and high-state levels as a 2-element real row vector. The first element is the low-state level. The second element is the high-state level.

'Tolerance'

Tolerance levels (lower- and upper-state boundaries) expressed as a percentage. See “State-Level Tolerances” on page 1-913.

Default: 2

Output Arguments

R

Rise times. R is a vector containing the duration of each positive-going transition. If you specify the sampling rate, FS, or the sampling instants, T, rise times are in seconds. If you do not specify a sampling rate, or sampling instants, rise times are in samples.

LT

Instants when positive-going transition crosses the lower-reference level. By default, the lower reference level is the 10% reference level. The upper reference level is the 90% reference level. You can change the default reference levels by specifying the 'PctRefLevels' name-value pair.

UT

Instants when positive-going transition crosses the upper-reference level. By default, the lower reference level is the 10% reference level. The upper reference level is the 90% reference level. You can change the default reference levels by specifying the 'PctRefLevels' name-value pair.

LL

Lower reference level in waveform amplitude units.LL is a vector containing the waveform value corresponding to the lower reference level in each positive-going transition. By default, the lower reference level is the 10% reference level. You can change the default reference levels by specifying the 'PctRefLevels' name-value pair.

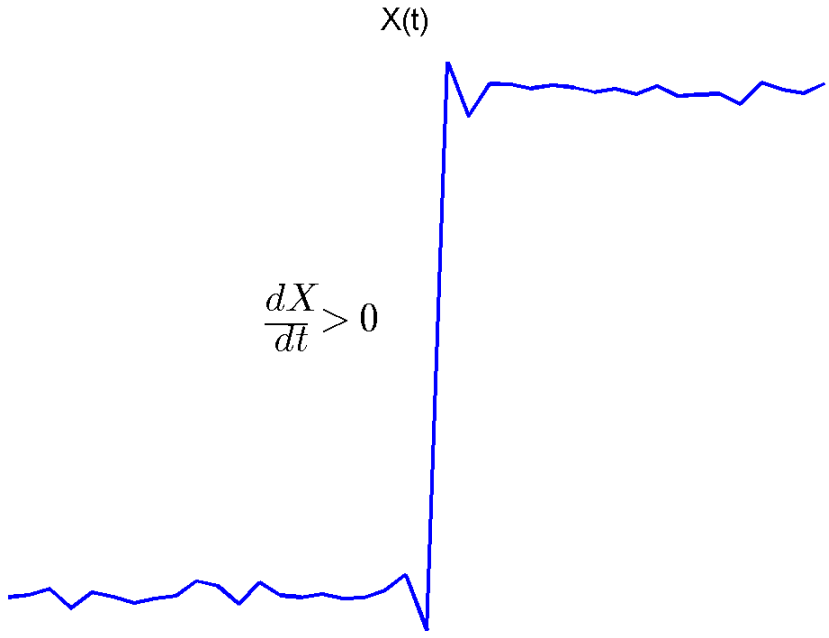
UL

Upper reference level in waveform amplitude units. `LL` is a vector containing the waveform value corresponding to the upper reference level in each positive-going transition. By default, the upper reference level is the 90% reference level. You can change the default reference levels by specifying the `'PctRefLevels'` name-value pair.

Definitions

Positive-Going Transition

A *positive-going transition* in a bilevel waveform is a transition from the low-state level to the high-state level. A positive-polarity (positive-going) pulse has a terminating state more positive than the originating state. If the waveform is differentiable in the neighborhood of the transition, an equivalent definition is a transition with a positive first derivative. The following figure shows a positive-going transition.



In the preceding figure, the amplitude values of the waveform do not appear because a positive-going transition does not depend on the actual waveform values. A positive-going transition is defined by the direction of the transition.

Percent Reference Levels

If S_1 is the low state, S_2 is the high state, and U is the *upper*-percent reference level. The waveform value corresponding to the upper percent reference level is

$$S_1 + \frac{U}{100}(S_2 - S_1)$$

If L is the *lower*-percent reference level, the waveform value corresponding to the lower percent reference level is

$$S_1 + \frac{L}{100}(S_2 - S_1)$$

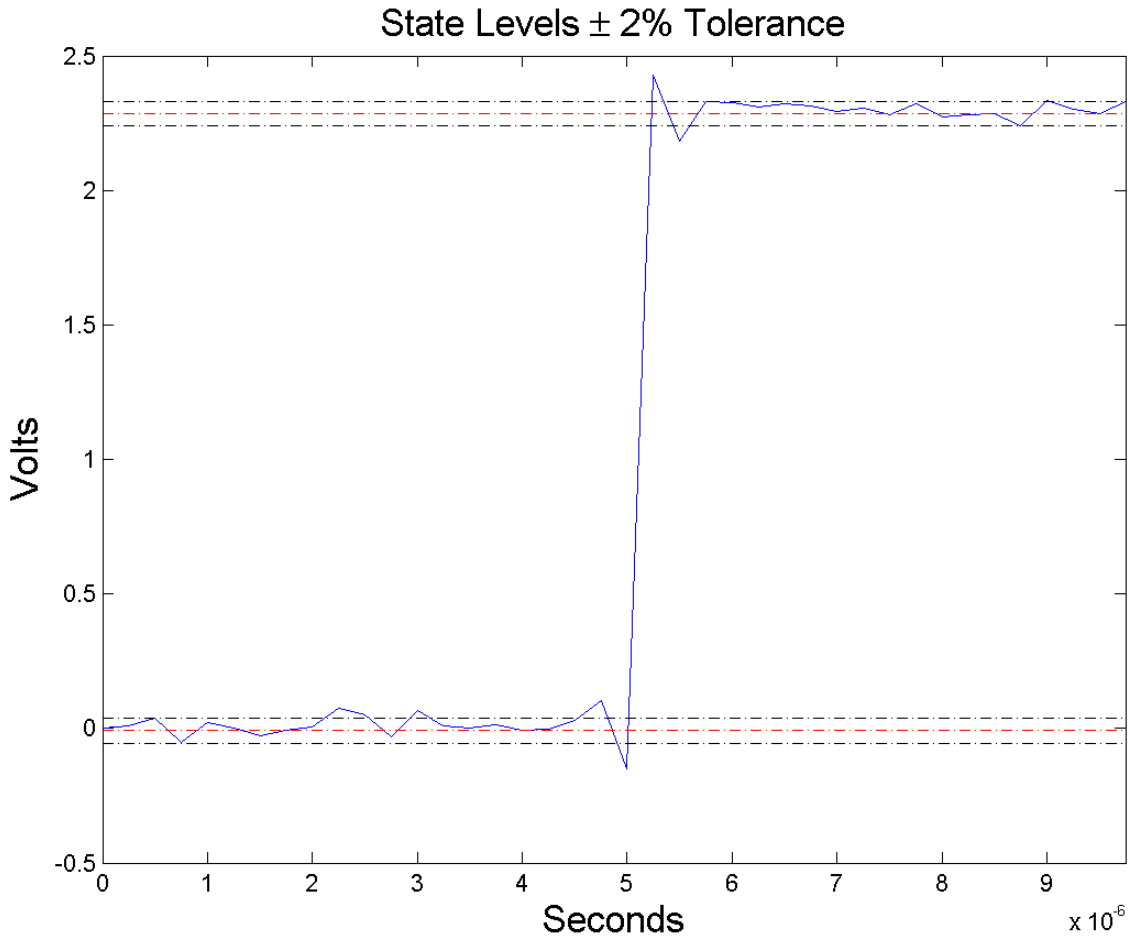
State-Level Tolerances

Each state level can have associated lower- and upper-state boundaries. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The red dashed lines indicate the estimated state levels.



Examples

Rise Time in a Bilevel Waveform

Determine the rise time in samples for a 2.3 V clock waveform.

Load the 2.3 V clock data. Determine the rise time in samples. Use the default `[10 90]` percent reference levels.

```
load('transitionex.mat','x');
R = risetime(x);
```

The rise time is less than 1, indicating that the transition occurred in a fraction of a sample.

Rise Time with 20% and 80% Reference Levels

Determine the rise time in a 2.3 V clock waveform sampled at 4 MHz. Compute the rise time using the 20% and 80% reference levels.

Load the 2.3 V clock data with sampling instants. Plot the waveform.

```
load('transitionex.mat','x','t');
plot(t,x);
```

Determine the rise time using the 20% and 80% reference levels.

```
R = risetime(x,'PctRefLevels',[20 80]);
```

Rise Time, Reference-Level Instants, and Reference Levels

Determine the rise time, reference-level instants, and reference levels in a 2.3 V clock waveform sampled at 4 MHz.

Load the 2.3 V clock waveform along with the sampling instants.

```
load('transitionex.mat','x','t');
```

Determine the rise time, reference-level instants, and reference levels.

```
[R,LT,UT,LL,UL] = risetime(x,t);
```

Plot the waveform in microseconds with the lower- and upper-reference levels and reference-level instants. Show that the rise time is the difference between the upper- and lower-reference level instants.

```
plot(t.*1e6,x);
```

```
xlabel('microseconds'); ylabel('Volts');
hold on; grid on;
plot(LT.*1e6,LL,'ro','markerfacecolor',[1 0 0]);
plot(UT.*1e6,UL,'ro','markerfacecolor',[1 0 0]);
fprintf('Rise time is %1.4f microseconds.\n',(UT-LT)*1e6)
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003, pp. 15–17.

See Also

`falltime` | `slewrate` | `statelevels`

Purpose

Reverse Levinson-Durbin recursion

Syntax

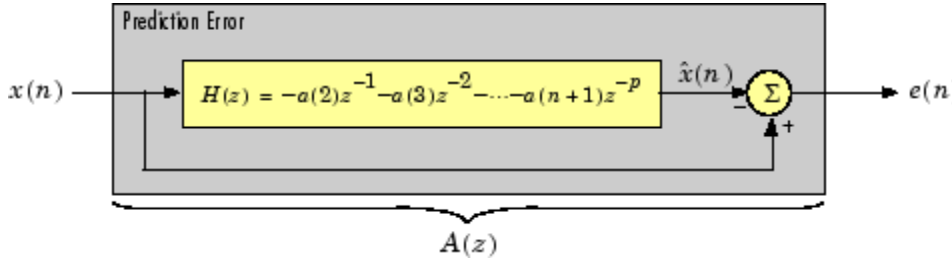
```
r = rlevinson(a,efinal)
[r,u] = rlevinson(a,efinal)
[r,u,k] = rlevinson(a,efinal)
[r,u,k,e] = rlevinson(a,efinal)
```

Description

The reverse Levinson-Durbin recursion implements the step-down algorithm for solving the following symmetric Toeplitz system of linear equations for r, where $r = [r(1)\ L r(p+1)]$ and $r(i)^*$ denotes the complex conjugate of $r(i)$.

$$\begin{bmatrix} r(1) & r(2)^* & \cdots & r(p)^* \\ r(2) & r(1) & \cdots & r(p-1)^* \\ \vdots & \ddots & \ddots & \vdots \\ r(p) & \cdots & r(2) & r(1) \end{bmatrix} \begin{bmatrix} a(2) \\ a(3) \\ \vdots \\ a(p+1) \end{bmatrix} = \begin{bmatrix} -r(2) \\ -r(3) \\ \vdots \\ -r(p+1) \end{bmatrix}$$

`r = rlevinson(a,efinal)` solves the above system of equations for r given vector `a`, where `a = [1` $a(2)$ L $a(p+1)$`]`. In linear prediction applications, `r` represents the autocorrelation sequence of the input to the prediction error filter, where `r(1)` is the zero-lag element. The figure below shows the typical filter of this type, where $H(z)$ is the optimal linear predictor, $x(n)$ is the input signal, $\hat{x}(n)$ is the predicted signal, and $e(n)$ is the prediction error.



Input vector `a` represents the polynomial coefficients of this prediction error filter in descending powers of z.

$$A(z) = 1 + a(2)z^{-1} + \cdots + a(n+1)z^{-p}$$

The filter must be minimum phase to generate a valid autocorrelation sequence. `efinal` is the scalar prediction error power, which is equal to the variance of the prediction error signal, $\sigma^2(e)$.

`[r,u] = rlevinson(a,efinal)` returns upper triangular matrix U from the UDU^* decomposition

$$R^{-1} = UE^{-1}U^*$$

where

$$R = \begin{bmatrix} r(1) & r(2)^* & \cdots & r(p)^* \\ r(2) & r(1) & \cdots & r(p-1)^* \\ \vdots & \ddots & \ddots & \vdots \\ r(p) & \cdots & r(2) & r(1) \end{bmatrix}$$

and E is a diagonal matrix with elements returned in output `e` (see below). This decomposition permits the efficient evaluation of the inverse of the autocorrelation matrix, R^{-1}.

Output matrix `u` contains the prediction filter polynomial, `a`, from each iteration of the reverse Levinson-Durbin recursion

$$U = \begin{bmatrix} a_1(1)^* & a_2(2)^* & \cdots & a_{p+1}(p+1)^* \\ 0 & a_2(1)^* & \ddots & a_{p+1}(p)^* \\ 0 & 0 & \ddots & a_{p+1}(p-1)^* \\ \vdots & \ddots & \ddots & \vdots \\ 0 & \cdots & 0 & a_{p+1}(1)^* \end{bmatrix}$$

where $a_i(j)$ is the jth coefficient of the ith order prediction filter polynomial (i.e., step i in the recursion). For example, the 5th order prediction filter polynomial is

```
a5 = u(5:-1:1,5)'
```

Note that `u(p+1:-1:1,p+1)'` is the input polynomial coefficient vector `a`.

`[r,u,k] = rlevinson(a,efinal)` returns a vector `k` of length (p+1) containing the reflection coefficients. The reflection coefficients are the conjugates of the values in the first row of `u`.

```
k = conj(u(1,2:end))
```

`[r,u,k,e] = rlevinson(a,efinal)` returns a vector of length p+1 containing the prediction errors from each iteration of the reverse Levinson-Durbin recursion: `e(1)` is the prediction error from the first-order model, `e(2)` is the prediction error from the second-order model, and so on.

These prediction error values form the diagonal of the matrix E in the UDU^* decomposition of R^{-1}.

$$R^{-1} = UE^{-1}U^*$$

References

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

See Also

`levinson` | `lpc` | `prony` | `stmcb`

Purpose

Root-mean-square level

Syntax

```
Y = rms(X)
Y = rms(X,DIM)
```

Description

`Y = rms(X)` returns the root-mean-square (RMS) level of the input, `X`. If `X` is a row or column vector, `Y` is a real-valued scalar. For matrices, `Y` contains the RMS levels computed along the first nonsingleton dimension. For example, if `X` is an N-by-M matrix with N>1, `Y` is a 1-by-M row vector containing the RMS levels of the columns of `X`.

`Y = rms(X,DIM)` computes the RMS level of `X` along the dimension, `DIM`.

Input Arguments

X

Real or complex-valued input vector or matrix. By default, `rms` acts along the first nonsingleton dimension of `X`.

DIM

Dimension for RMS levels. The optional `DIM` input argument specifies the dimension along which to compute the RMS levels.

Default: First nonsingleton dimension

Output Arguments

Y

Root-mean-square level. For vectors, `Y` is a real-valued scalar. For matrices, `Y` contains the RMS levels computed along the specified dimension `DIM`. By default, `DIM` is the first nonsingleton dimension.

Definitions

Root-Mean-Square Level

The root-mean-square level of a vector, X, is

$$X_{\mathrm{RMS}} = \sqrt{\frac{1}{N}\sum_{n=1}^{N}|X_n|^2}$$

with the summation performed along the specified dimension.

Examples

RMS Level of Sinusoid

Compute the RMS level of a 100-Hz sinusoid sampled at 1 kHz.

```
t = 0:0.001:1-0.001;
X = cos(2*pi*100*t);
Y = rms(X);
```

RMS Levels of 2-D Matrix

Create a matrix where each column is a 100-Hz sinusoid sampled at 1 kHz with a different amplitude. The amplitude is equal to the column index.

Compute the RMS levels of the columns.

```
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t)';
X = repmat(x,1,4);
amp = 1:4;
amp = repmat(amp,1e3,1);
X = X.*amp;
Y = rms(X);
```

RMS Levels of 2-D Matrix Along Specified Dimension

Create a matrix where each row is a 100-Hz sinusoid sampled at 1 kHz with a different amplitude. The amplitude is equal to the row index.

Compute the RMS levels of the rows specifying the dimension equal to 2 with the `DIM` argument.

```
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t);
X = repmat(x,4,1);
amp = (1:4)';
amp = repmat(amp,1,1e3);
X = X.*amp;
```

```
Y = rms(X,2);
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Std 181, 2003.

See Also

`mean` | `peak2rms` | `std`

Purpose

Frequency and power content using eigenvector method

Syntax

```
[w,pow] = rooteig(x,p)
[f,pow] = rooteig(...,fs)
[w,pow] = rooteig(...,'corr')
```

Description

`[w,pow] = rooteig(x,p)` estimates the frequency content in the time samples of a signal `x`, and returns `w`, a vector of frequencies in rad/sample, and the corresponding signal power in the vector `pow` in units of power, such as volts^2. The input signal `x` is specified either as:

- A row or column vector representing one observation of the signal
- A rectangular array for which each row of `x` represents a separate observation of the signal (for example, each row is one output of an array of sensors, as in array processing), such that `x'*x` is an estimate of the correlation matrix

Note You can use the output of `corrmtx` to generate such an array `x`.

You can specify the second input argument `p` as either:

- A scalar integer. In this case, the signal subspace dimension is `p`.
- A two-element vector. In this case, `p(2)`, the second element of `p`, represents a threshold that is multiplied by λ_{min}, the smallest estimated eigenvalue of the signal's correlation matrix. Eigenvalues below the threshold λ_{min}`*p(2)` are assigned to the noise subspace. In this case, `p(1)` specifies the maximum dimension of the signal subspace.

The extra threshold parameter in the second entry in `p` provides you more flexibility and control in assigning the noise and signal subspaces.

The length of the vector `w` is the computed dimension of the signal subspace. For real-valued input data `x`, the length of the corresponding power vector `pow` is given by

```
length(pow) = 0.5*length(w)
```

For complex-valued input data `x`, `pow` and `w` have the same length.

`[f,pow] = rooteig(...,fs)` returns the vector of frequencies `f` calculated in Hz. You supply the sampling frequency `fs` in Hz. If you specify `fs` with the empty vector `[]`, the sampling frequency defaults to 1 Hz.

`[w,pow] = rooteig(...,'corr')` forces the input argument `x` to be interpreted as a correlation matrix rather than a matrix of signal data. For this syntax, you must supply a square matrix for `x`, and all of its eigenvalues must be nonnegative.

Note You can place the string `'corr'` anywhere after `p`.

Examples

Find the frequency content in a signal composed of three complex exponentials in noise. Use the modified covariance method to estimate the correlation matrix used by the eigenvector method:

```
n=0:99;
s = exp(i*pi/2*n)+2*exp(i*pi/4*n)+...
  exp(i*pi/3*n)+randn(1,100);
% Estimate correlation matrix using
%  modified covariance method.
X=corrmtx(s,12,'mod');
[W,P] = rooteig(X,3)
```

Algorithms

The eigenvector method used by `rooteig` is the same as that used by `peig`. The algorithm performs eigenspace analysis of the signal's correlation matrix in order to estimate the signal's frequency content.

The difference between `peig` and `rooteig` is:

- `peig` returns the pseudospectrum at all frequency samples.

- `rooteig` returns the estimated discrete frequency spectrum, along with the corresponding signal power estimates.

`rooteig` is most useful for frequency estimation of signals made up of a sum of sinusoids embedded in additive white Gaussian noise.

See Also

`corrmtx` | `peig` | `pmusic` | `spectrum` | `rootmusic` | `spectrum.eigenvector`

Purpose

Root MUSIC algorithm

Syntax

```
W = rootmusic(X,P)
[W,POW] = rootmusic(X,P)
[F, POW] = rootmusic(...,Fs)
[W,POW] = rootmusic(...,'corr')
```

Description

`W = rootmusic(X,P)` returns the frequencies in radians/sample for the `P` complex exponentials (sinusoids) that make up the signal `X`.

The input `X` is specified either as:

- A row or column vector representing one realization of signal
- A rectangular array for which each row of `X` represents a separate observation of the signal (for example, each row is one output of an array of sensors, as in array processing), such that `X'*X` is an estimate of the correlation matrix

Note You can use the output of `corrmtx` to generate such an array `X`.

`[W,POW] = rootmusic(X,P)` returns the estimated absolute value squared amplitudes of the sinusoids at the frequencies `W`.

The second input argument, `P` is the number of complex sinusoids in `X`. You can specify `P` as either:

- A positive integer. In this case, the signal subspace dimension is `P`.
- A two-element vector. In this case, `P(2)`, the second element of `P`, represents a threshold that is multiplied by λ_{min}, the smallest estimated eigenvalue of the signal's correlation matrix. Eigenvalues below the threshold λ_{min}`*P(2)` are assigned to the noise subspace. In this case, `P(1)` specifies the maximum dimension of the signal subspace.

The extra threshold parameter in the second entry in `P` provides you more flexibility and control in assigning the noise and signal subspaces.

The length of the vector `W` is the computed dimension of the signal subspace. For real-valued input data `X`, the length of the corresponding power vector `POW` is given by

```
length(POW) = 0.5*length(W)
```

For complex-valued input data `X`, `POW` and `W` have the same length.

`[F, POW] = rootmusic(...,Fs)` returns the vector of frequencies `F` calculated in Hz. You supply the sampling frequency `Fs` in Hz. If you specify `Fs` with the empty vector `[]`, the sampling frequency defaults to 1 Hz.

`[W,POW] = rootmusic(...,'corr')` forces the input argument `X` to be interpreted as a correlation matrix rather than a matrix of signal data. For this syntax, you must supply a square matrix for `X`, and all of its eigenvalues must be nonnegative.

Note You can place the string `'corr'` anywhere after `P`.

Examples

Estimate the amplitudes for 2 sinusoids in noise. The separation between the sinusoids is less than the resolution of the periodogram, 2π/N radians/sample.

```
s1 = RandStream.create('mrg32k3a');
n=0:99; % N is equal to 100. Periodogram resolution is (2*pi)/100
freqs = [pi/4 pi/4+0.06];
s = 2*exp(1j*freqs(1)*n)+1.5*exp(1j*freqs(2)*n)+...
0.5*randn(s1,1,100)+1j*0.5*randn(s1,1,100);
X = corrmtx(s,12,'mod');
[W,P] = rootmusic(X,2);
```

Algorithms

The MUSIC algorithm used by `rootmusic` is the same as that used by `pmusic`. The algorithm performs eigenspace analysis of the signal's correlation matrix in order to estimate the signal's frequency content.

The difference between `pmusic` and `rootmusic` is:

- `pmusic` returns the pseudospectrum at all frequency samples.
- `rootmusic` returns the estimated discrete frequency spectrum, along with the corresponding signal power estimates.

`rootmusic` is most useful for frequency estimation of signals made up of a sum of sinusoids embedded in additive white Gaussian noise.

Diagnostics

If the input signal, `x` is real and an odd number of sinusoids, `p` is specified, this error message is displayed

```
Real signals require an even number p of complex sinusoids.
```

See Also

`corrmtx` | `peig` | `pmusic` | `spectrum` | `rooteig` | `spectrum.music`

Purpose

Root-sum-of-squares level

Syntax

```
Y = rssq(X)
Y = rssq(X,DIM)
```

Description

`Y = rssq(X)` returns the root-sum-of-squares (RSS) level, `Y`, of the input, `X`. If `X` is a row or column vector, `Y` is a real-valued scalar. For matrices, `Y` contains the RSS levels computed along the first nonsingleton dimension. For example, if `Y` is an N-by-M matrix with N>1, `Y` is a 1-by-M row vector containing the RSS levels of the columns of `Y`.

`Y = rssq(X,DIM)` computes the RSS level of `X` along the dimension, `DIM`.

Input Arguments

X

Real- or complex-valued input vector or matrix. By default, `rssq` acts along the first nonsingleton dimension of `X`.

DIM

Dimension for root-sum-of-squares (RSS) level. The optional `DIM` input argument specifies the dimension along which to compute the RSS level.

Default: First nonsingleton dimension

Output Arguments

Y

Root-sum-of-squares level. For vectors, `Y` is a real-valued scalar. For matrices, `Y` contains the RSS levels computed along the specified dimension, `DIM`. By default, `DIM` is the first nonsingleton dimension.

Definitions

Root-Sum-of-Squares Level

The root-sum-of-squares (RSS) level of a vector, *X*, is

$$X_{\mathrm{RSS}} = \sqrt{\sum_{n=1}^{N} |X_n|^2}$$

with the summation performed along the specified dimension. The RSS is also referred to as the ℓ^2 norm.

Examples

RSS Level of Sinusoid

Compute the RSS level of a 100-Hz sinusoid sampled at 1 kHz.

```
t = 0:0.001:1-0.001;
X = cos(2*pi*100*t);
Y = rssq(X);
```

RSS Level of 2-D Matrix

Create a matrix where each column is a 100-Hz sinusoid sampled at 1 kHz with a different amplitude. The amplitude is equal to the column index.

Compute the RSS level of the columns.

```
t = 0:0.001:1-0.001;
x = cos(2*pi*100*t)';
X = repmat(x,1,4);
amp = 1:4;
amp = repmat(amp,1e3,1);
X = X.*amp;
Y = rssq(X);
```

RSS Level of 2-D Matrix Along Specified Dimension

Create a matrix where each row is a 100-Hz sinusoid sampled at 1 kHz with a different amplitude. The amplitude is equal to the row index.

Compute the RSS level of the rows specifying the dimension equal to 2 with the `DIM` argument.

```
t = 0:0.001:1-0.001;
```

```
x = cos(2*pi*100*t);
X = repmat(x,4,1);
amp = (1:4)';
amp = repmat(amp,1,1e3);
X = X.*amp;
Y = rssq(X,2);
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003.

sawtooth

Purpose

Sawtooth or triangle wave

Syntax

```
sawtooth(t)
sawtooth(t,width)
```

Description

`sawtooth(t)` generates a sawtooth wave with period 2π for the elements of time vector `t`. `sawtooth(t)` is similar to `sin(t)`, but creates a sawtooth wave with peaks of -1 and 1 instead of a sine wave. The sawtooth wave is defined to be -1 at multiples of 2π and to increase linearly with time with a slope of 1/π at all other times.

`sawtooth(t,width)` generates a modified triangle wave where `width`, a scalar parameter between 0 and 1, determines the point between 0 and 2π at which the maximum occurs. The function increases from -1 to 1 on the interval 0 to 2π*`width`, then decreases linearly from 1 to -1 on the interval 2π*`width` to 2π. Thus a parameter of 0.5 specifies a standard triangle wave, symmetric about time instant π with peak-to-peak amplitude of 1. `sawtooth(t,1)` is equivalent to `sawtooth(t)`.

See Also

`chirp` | `cos` | `diric` | `gauspuls` | `pulstran` | `rectpuls` | `sin` | `sinc` | `square` | `tripuls`

Purpose

Compute reflection coefficients from autocorrelation sequence

Syntax

```
k = schurrc(r)
[k,e] = schurrc(r)
```

Description

`k = schurrc(r)` uses the Schur algorithm to compute a vector `k` of reflection coefficients from a vector `r` representing an autocorrelation sequence. `k` and `r` are the same size. The reflection coefficients represent the lattice parameters of a prediction filter for a signal with the given autocorrelation sequence, `r`. When `r` is a matrix, `schurrc` treats each column of `r` as an independent autocorrelation sequence, and produces a matrix `k`, the same size as `r`. Each column of `k` represents the reflection coefficients for the lattice filter for predicting the process with the corresponding autocorrelation sequence `r`.

`[k,e] = schurrc(r)` also computes the scalar `e`, the prediction error variance. When `r` is a matrix, `e` is a column vector. The number of rows of `e` is the same as the number of columns of `r`.

Examples

Create an autocorrelation sequence from the MATLAB speech signal contained in `mtlb.mat`, and use the Schur algorithm to compute the reflection coefficients of a lattice prediction filter for this autocorrelation sequence:

```
load mtlb
r = xcorr(mtlb(1:5),'unbiased');
k = schurrc(r(5:end))
k =
   -0.7583
    0.1384
    0.7042
   -0.3699
```

References

[1] Proakis, J. and D. Manolakis, *Digital Signal Processing: Principles, Algorithms, and Applications*, Third edition, Prentice-Hall, 1996, pp. 868-873.

See Also

levinson

Purpose

Settling time for bilevel waveform

Syntax

```
S = settlingtime(X,D)
S = settlingtime(X,FS,D)
S = settlingtime(X,T,D)
[S,SLEV,SINST] = settlingtime(...)
[S,SLEV,SINST] = settlingtime(...,Name,Value)
settlingtime(...)
```

Description

`S = settlingtime(X,D)` returns the time, `S`, from the mid-reference level instant to the time instant each transition enters and remains within a 2% tolerance region of the final state over the duration, `D`. `D` is a positive scalar. Because `settlingtime` uses interpolation to determine the mid-reference level instant, `S` may contain values that do not correspond to sampling instants. The length of `S` is equal to the number of detected transitions in the input signal, `X`. If for any transition, the level of the waveform does not remain within the lower and upper tolerance boundaries, the requested duration is not present, or an intervening transition is detected, `settlingtime` marks the corresponding element in `S` as `NaN`. See "Settle Seek Duration" on page 1-940 for cases in which `settlingtime` returns a `NaN`. To determine the transitions, `settlingtime` estimates the state levels of the input waveform by a histogram method. `settlingtime` identifies all regions that cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-939.

`S = settlingtime(X,FS,D)` specifies the sampling rate for the bilevel waveform, `X` in hertz. The first sample instant in `X` is equal to t=0. Because `settlingtime` uses interpolation to determine the mid-reference level instant, `S` may contain values that do not correspond to sampling instants.

`S = settlingtime(X,T,D)` specifies the sample instants, `T`, as a vector with the same number of elements as `X`.

`[S,SLEV,SINST] = settlingtime(...)` returns vectors, `SLEV`, and `SINST`, whose elements correspond to the levels and sample instants of the settling points for each transition.

`[S,SLEV,SINST] = settlingtime(...,Name,Value)` returns the settling times, levels, and corresponding sample instants with additional options specified by one or more `Name,Value` pair arguments.

`settlingtime(...)` plots the signal and darkens the regions of each transition where settling time is computed. The plot marks the location of the settling time of each transition, the mid-crossings, and the associated reference levels. The plot also displays the state levels with the corresponding lower and upper tolerance boundaries.

Input Arguments

X

Bilevel waveform. `X` is a real-valued row or column vector.

D

Settle-seek duration. `D` is a positive scalar, which defines the duration after the mid-reference level instant that `settlingtime` looks for a settling time. If no settling time occurs in `D` seconds after the mid-reference level instant, `settlingtime` returns a `NaN`. See "Settling Time" on page 1-937 and "Settle Seek Duration" on page 1-940.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

'MidPct'

Mid-reference level as a percentage of the waveform amplitude. See "Mid-Reference Level" on page 1-938.

Default: 50

'StateLevels'

Low and high-state levels. `StateLevels` is a 1-by-2 real-valued vector. The first element is the low-state level. The second element is the high-state level. If you do not specify low and high-state levels, `settlingtime` estimates the state levels from the input waveform using the histogram method.

'Tolerance'

Tolerance levels (lower and upper state boundaries) expressed as a percentage. See "State-Level Tolerances" on page 1-939.

Default: 2

Output Arguments

S

The time from the mid-reference level instant to the time instant each transition enters and remains within a 2% tolerance region of the final state over duration, `D`.

SLEV

Waveform values at the settling points.

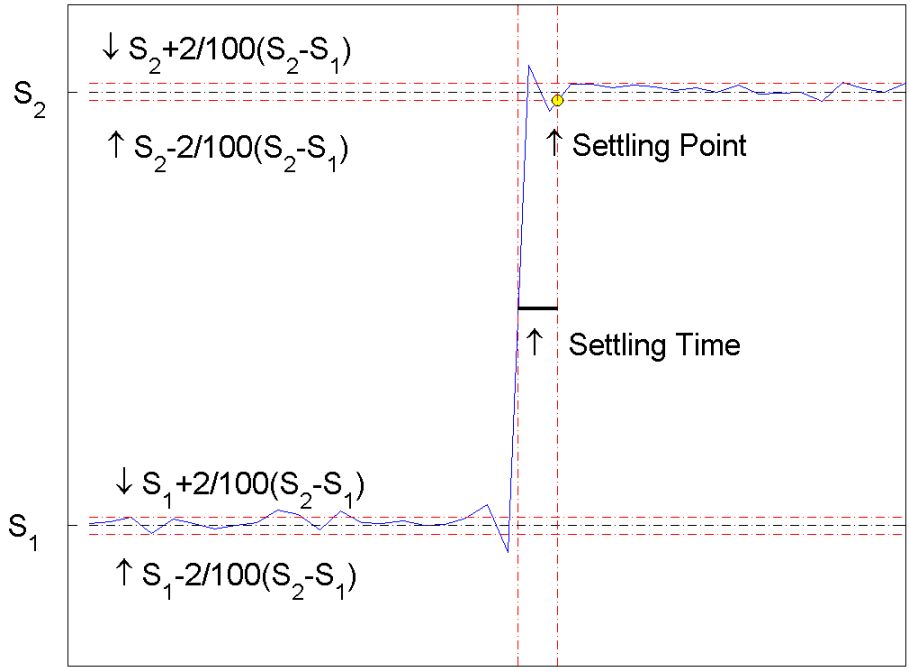
SINST

Time instants of the settling points.

Definitions

Settling Time

The settling time is the time after the mid-reference level instant when the signal crosses into and remains in the 2%-tolerance region around the state level. The settling time is illustrated in the following figure. The low- and high-state levels are the dashed black lines. The 2% tolerances above and below the state levels are shown by the red dashed lines and the settling time is indicated by the yellow circle.



Mid-Reference Level

The mid-reference level in a bilevel waveform with low-state level, *S_1*, and high- state level, *S_2*, is

$$S_1 + \frac{1}{2}(S_2 - S_1)$$

Mid-Reference Level Instant

Let $y_{50\%}$ denote the mid reference level.

Let $t_{50\%-}$ and $t_{50\%+}$ denote the two consecutive sampling instants corresponding to the waveform values nearest in value to $y_{50\%}$.

Let $y_{50\%-}$ and $y_{50\%+}$ denote the waveform values at $t_{50\%-}$ and $t_{50\%+}$.

The mid-reference level instant is

$$t_{50\%} = t_{50\%} + \left(\frac{t_{50\%_+} - t_{50\%_-}}{y_{50\%_+} - y_{50\%_-}}\right)\left(y_{50\%_+} - y_{50\%_-}\right)$$

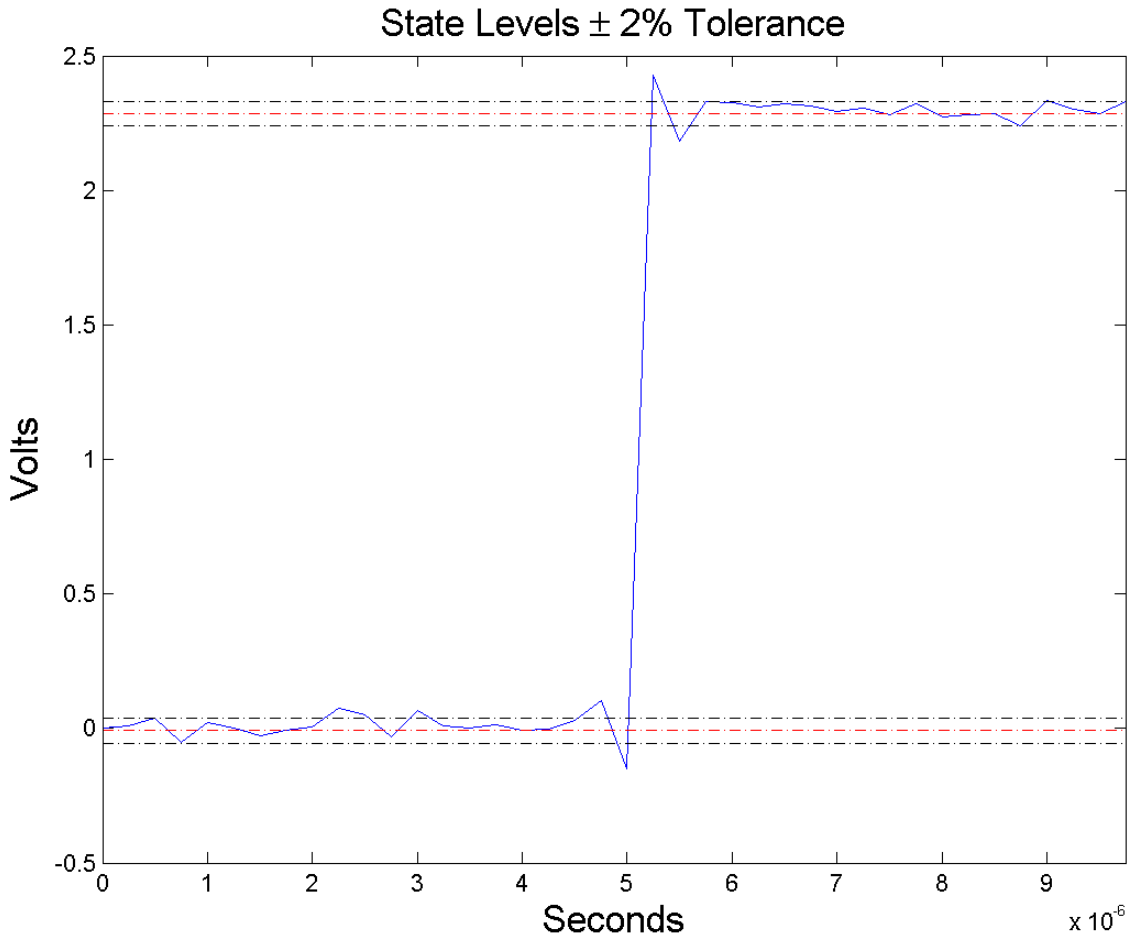
State-Level Tolerances

Each state level can have associated lower- and upper-state boundaries. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

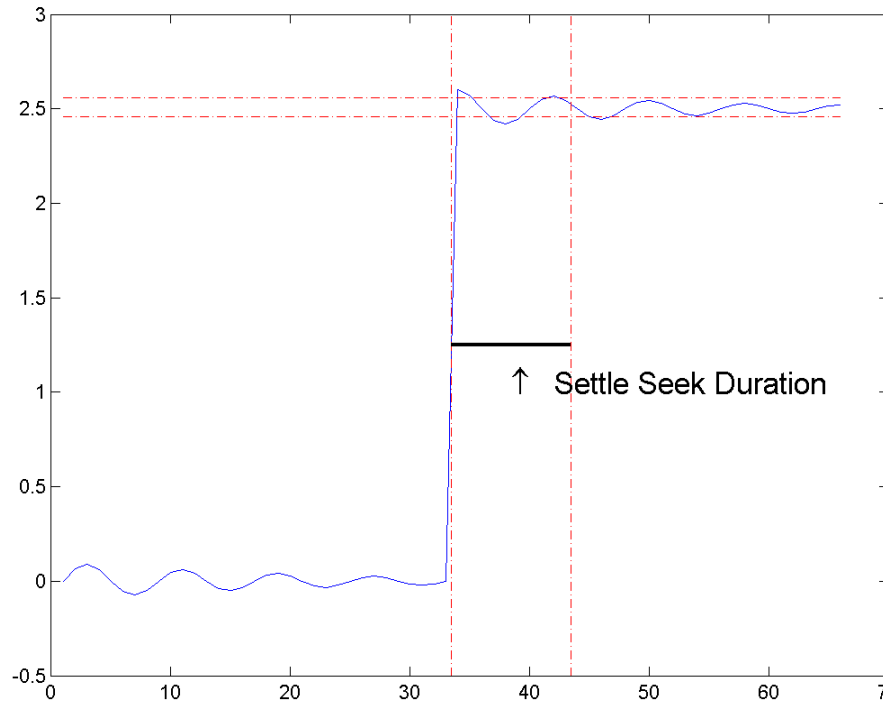
The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The estimated state levels are indicated by a dashed red line.



Settle Seek Duration

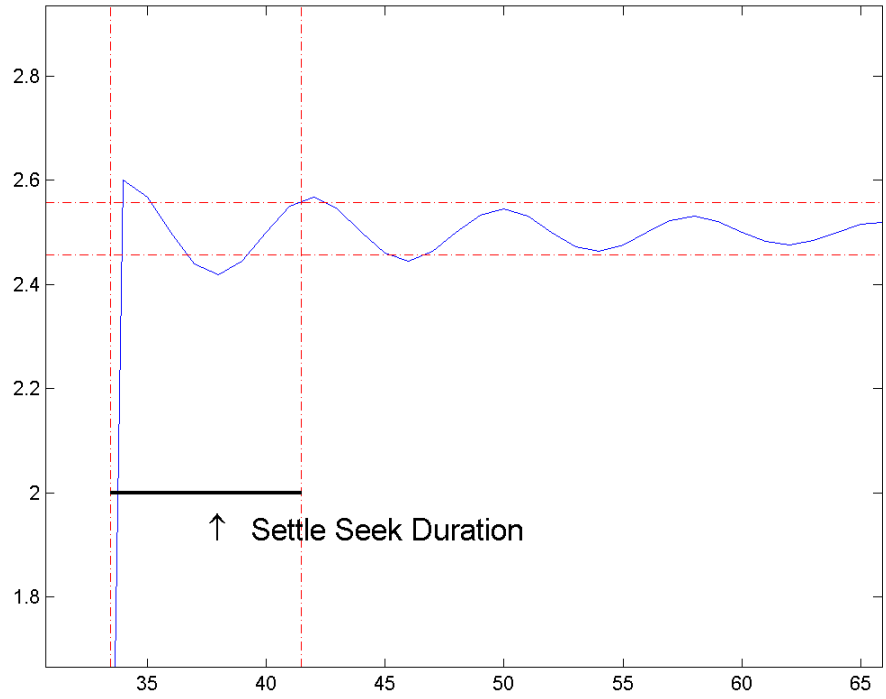
The settle seek duration defines the interval of time after the mid-reference level instant that `settlingtime` looks for a settling point. If `settlingtime` does not find a settling point within the settle seek

duration, `settlingtime` returns `NaN` for the settling time. The following figure illustrates a settle seek duration of 10 samples.



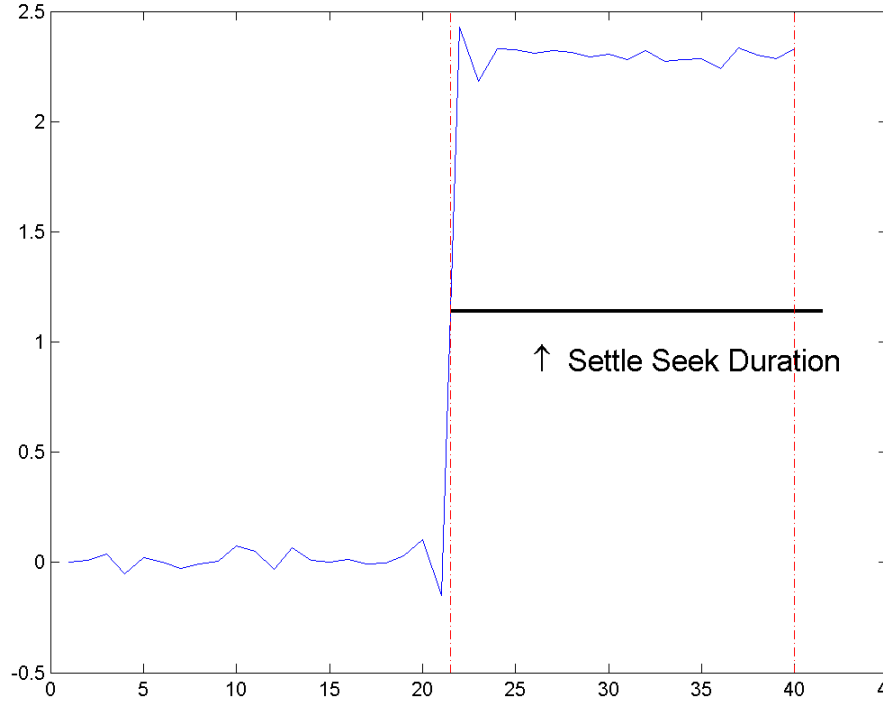
`settlingtime` may fail to find a settling point in the specified settle seek duration if any one of the following conditions occurs:

- The last waveform value in the settle seek interval is not within the upper- and lower-state boundaries determined by the specified tolerance. The following figure illustrates this condition for a settle seek duration of 8 samples and a 2% tolerance region.

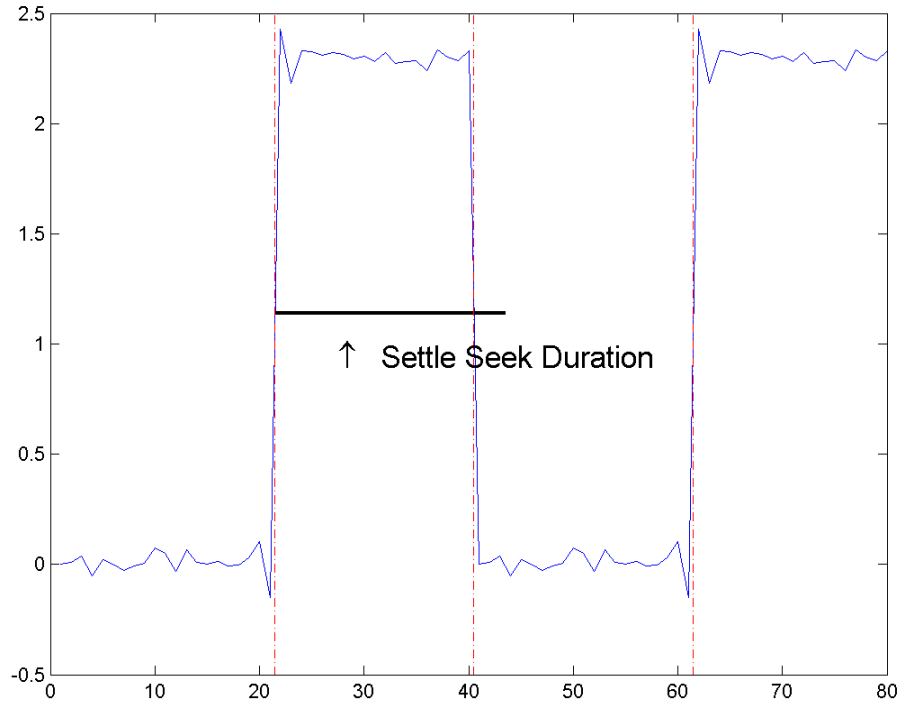


In the preceding figure, you see that the last sample in the settle seek interval exceeds the upper state boundary. In this example, reducing or increasing the settle seek duration can result in a valid settling time.

- There is an insufficient number of waveform samples for the specified settle seek duration. The following figure illustrates this condition for a settle seek duration of 20 samples. The settle seek duration extends beyond the final sample of the waveform.



- An intervening transition is detected before the end of the specified settle seek duration. The following figure illustrates this condition for a settle seek duration of 22 samples. An intervening transition is detected before the end of the 22–sample settle seek duration.

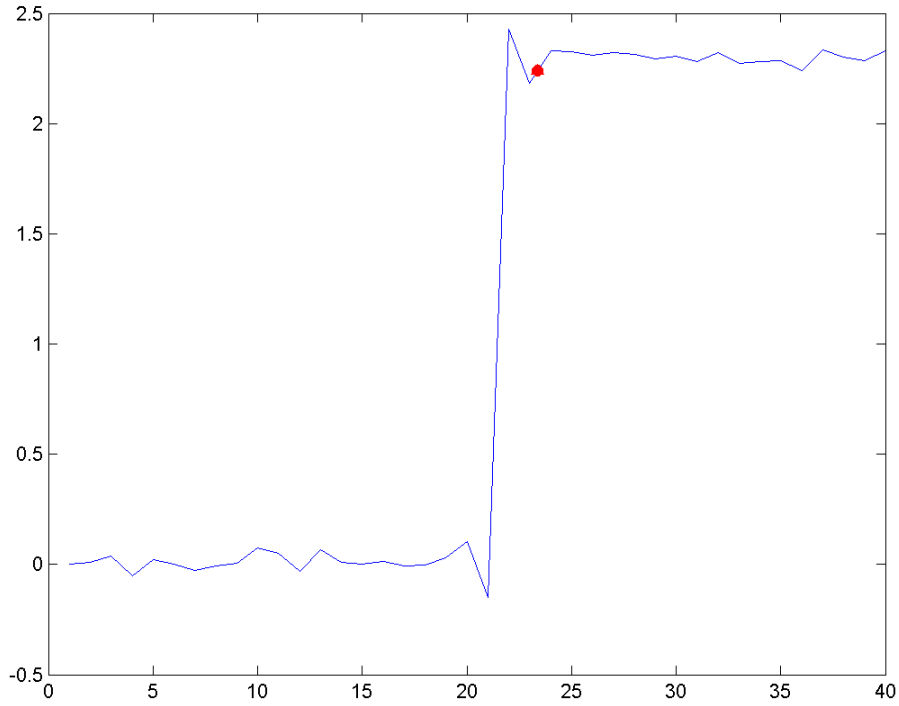


Examples

Determine Settling Point and Settling Level

Determine the settling point and corresponding waveform value for a bilevel waveform. Plot the waveform and mark the settling point.

```
load('transitionex.mat', 'x');
[S,SLEV,SINST] = settlingtime(x,10);
plot(x); hold on;
plot(SINST,SLEV,'ro','markerfacecolor',[1 0 0]);
```

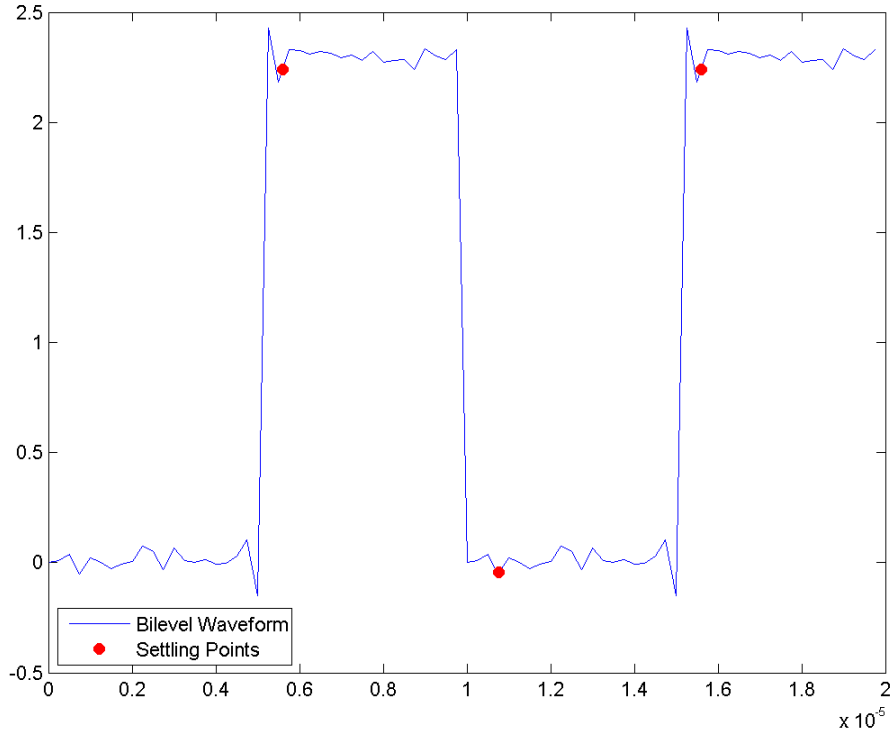


Determine Settling Points for a Three-Transition Bilevel Waveform

Determine the settling points for a three-transition bilevel waveform. The data is sampled at 4 MHz. Use a one-microsecond settle-seek duration. Plot the settling points.

```
load('transitionex.mat', 'x');
y = [x; fliplr(x)];
fs = 4e6;
t = 0:1/fs:(length(y)*1/fs)-1/fs;
```

```
[S,SLEV,SINST] = settlingtime(y,fs,1e-6);
% equivalent to [S,SLEV,SINST] = settlingtime(y,t);
plot(t,y); hold on;
plot(SINST,SLEV,'ro','markerfacecolor',[1 0 0]);
legend('Bilevel Waveform','Settling Points','Location','SouthWest');
```



References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003, pp. 23–24.

See Also

`falltime` | `midcross` | `pulsewidth` | `risetime` | `statelevels`

Purpose

Compute period of sequence

Syntax

```
p = seqperiod(x)
[p,num] = seqperiod(x)
```

Description

`p = seqperiod(x)` returns the integer `p` that corresponds to the period of the sequence in a vector `x`. The period `p` is computed as the minimum length of a subsequence `x(1:p)` of `x` that repeats itself continuously every `p` samples in `x`. The length of `x` does not have to be a multiple of `p`, so that an incomplete repetition is permitted at the end of `x`. If the sequence `x` is not periodic, then `p = length(x)`.

- If `x` is a matrix, then `seqperiod` checks for periodicity along each column of `x`. The resulting output p is a row vector with the same number of columns as `x`.
- If `x` is a multidimensional array, then `seqperiod` checks for periodicity along the first nonsingleton dimension of `x`. In this case:
  - `p` is a multidimensional array of integers with a leading singleton dimension.
  - The lengths of the remaining dimensions of `p` correspond to those of the dimensions of `x` after the first nonsingleton one.

`[p,num] = seqperiod(x)` also returns the number `num` of repetitions of `x(1:p)` in `x`. `num` might not be an integer.

Examples

```
x = [4 0 1 6;
     2 0 2 7;
     4 0 1 5;
     2 0 5 6];
p = seqperiod(x)
p =
     2     1     4     3
```

The result implies:

- The first column of `x` has period 2.

- The second column of `x` has period 1.
- The third column of `x` is not periodic, so `p(3)` is just the number of rows of `x`.
- The fourth column of `x` has period 3, although the last (second) repetition of the periodic sequence is incomplete.

Purpose

Specifications for filter specification object

Syntax

```
setspecs(D,specvalue1,specvalue2,...)
setspecs(D,Specification,specvalue1,specvalue2,...)
setspecs(...Fs)
setspecs(...,MAGUNITS)
```

Description

`setspecs(D,specvalue1,specvalue2,...)` sets the specifications in filter specification object, `D`, in the same order they appear in the `Specification` property.

`setspecs(D,Specification,specvalue1,specvalue2,...)` changes the specifications for an existing filter specification object and sets values for the new `Specification` property.

`setspecs(...Fs)` specifies the sampling frequency, `Fs`, in Hz. The sampling frequency must be a scalar trailing all other specifications. Entering a sampling frequency causes all other frequency specifications to be in Hz.

`setspecs(...,MAGUNITS)` specifies the units for any magnitude specifications. `MAGUNITS` can be one of the following: `'linear'`, `'dB'`, or `'squared'`. The default is `'dB'`. The magnitude specifications are always converted and stored in dB regardless of how the units are specified.

Use `SET(D,'SPECIFICATION')` to get the list of all available specification types for the filter specification object, `D`.

Examples

Construct a lowpass filter with specifications for the filter order and cutoff frequency (-6 dB). Use `setspecs` after construction to set the values of the filter order and cutoff frequency. Display the values in the MATLAB command window.

```
D = fdesign.lowpass('N,Fc');
setspecs(D,10,0.2);
D.FilterOrder
D.Fcutoff
```

Construct a highpass filter with specifications for the numerator order, denominator order, and 3-dB frequency. Assume the sampling frequency is 1 kHz. Use `setspecs` to set the numerator and denominator orders to 6. Set the 3-dB frequency to 250 Hz. In order to use frequency specifications in Hz, specify the sampling frequency as a trailing scalar.

```
D = fdesign.highpass('Nb,Na,F3dB');
setspecs(D,6,6,250,1000);
```

See Also

`design` | `designmethods` | `designopts` | `fdesign`

Purpose

Spurious free dynamic range

Syntax

```
r = sfdr(x)
r = sfdr(x,fs,msd)

r = sfdr(sxx,f,pwrflag)
r = sfdr(sxx,f,msd,pwrflag)

[r,spurpow,spurfreq] = sfdr(___)
```

Description

`r = sfdr(x)` returns the spurious free dynamic range (SFDR), `r`, in dB of the real sinusoidal signal, `x`. `sfdr` computes the power spectrum using a modified periodogram with a Hamming window. The mean is subtracted from `x` before computing the power spectrum. The number of points used in the computation of the discrete Fourier transform (DFT) is the same as the length of the signal, `x`.

`r = sfdr(x,fs,msd)` returns the SFDR considering only spurs that are separated from the fundamental (carrier) frequency by the minimum spur distance, `msd`, specified in cycles/unit time. The sampling frequency is `fs`. If the carrier frequency is `Fc`, then all spurs in the interval (`Fc-msd`, `Fc+msd`) are ignored.

`r = sfdr(sxx,f,pwrflag)` returns the SFDR of the one-sided power spectrum of a real-valued signal, `sxx`. `f` is the vector of frequencies corresponding to the power estimates in `sxx`. The first element of `f` must equal 0 and the power in the corresponding element of `sxx` (DC) is ignored in the SFDR computation.

`r = sfdr(sxx,f,msd,pwrflag)` returns the SFDR considering only spurs that are separated from the fundamental (carrier) frequency by the minimum spur distance, `msd`. If the carrier frequency is `Fc`, then all spurs in the interval (`Fc-msd`, `Fc+msd`) are ignored. When the input to `sfdr` is a power spectrum, specifying `msd` can prevent high sidelobe levels from being identified as spurs.

`[r,spurpow,spurfreq] = sfdr( ___ )` returns the power and frequency of the largest spur.

Input Arguments

x - Real-valued sinusoidal signal

row vector | column vector

Real-valued sinusoidal signal, specified as a row or column vector. The mean is subtracted from `x` prior to obtaining the power spectrum for SFDR computation.

Example: `x = cos(pi/4*(0:79))+1e-4*cos(pi/2*(0:79));`

Data Types
`double`

fs - Sampling rate

1 (default) | positive scalar

The sampling rate of the signal in cycles/unit time, specified as a positive scalar. When the unit of time is seconds, `fs` is in Hz.

Data Types
`double`

msd - Minimum spur distance

0 (default) | positive scalar

Minimum number of discrete Fourier transform (DFT) bins to ignore in the SFDR computation, specified as a positive scalar. You can use this argument to ignore spurs or sidelobes that occur in close proximity to the carrier, or fundamental frequency. For example, if the carrier frequency is `Fc`, then all spurs in the range `(Fc-msd, Fc+msd)` are ignored.

Data Types
`double`

sxx - One-sided power spectrum

row or column vector of positive numbers

One-sided power spectrum to use in the SFDR computation, specified as row or column vector. The first element is the DC power (0 frequency)

and is removed prior to computing the SFDR. However, depending on the bandwidth of the window used in obtaining the power spectrum and the frequency resolution, leakage from a DC shift may be present in adjacent DFT bins. The presence of leakage from DC can affect the SFDR computation, see "DC Leakage Affects SFDR" on page 1-955. In such cases, input the time-domain data instead of the power spectrum or compute the power spectrum after subtracting the mean from the signal.

Data Types
`double`

f - Vector of frequencies

row or column vector of nonnegative numbers

Vector of frequencies corresponding to the power estimates in `sxx`, specified as a row or column vector.

pwrflag - Power spectrum input flag

`'power'`

Flag indicating that the input is a one-sided power spectrum, `sxx`, specified as the string `'power'`.

Output Arguments

r - Spurious free dynamic range

real-valued scalar

Spurious free dynamic range in dB, specified as a real-valued scalar. The spurious free dynamic range is the difference in dB between the power at the peak frequency and the power at the next largest frequency (spur). If the input is time series data, the power estimates are obtained from a modified periodogram using a Hamming window. The length of the DFT used in the periodogram is equal to the length of the input signal, `x`. If you want to use a different power spectrum as the basis for the SFDR measurement, you can input your power spectrum using the `'power'` flag.

Data Types
`double`

spurpow - power of largest spur

Power in dB of the largest spur, specified as a real-valued scalar.

Data Types
`double`

spurfreq - frequency of largest spur

Frequency in Hz of the largest spur, specified as a real-valued scalar. If you do not supply the sampling frequency as an input argument, `sfdr` assumes a sampling frequency of 1 Hz.

Data Types
`double`

Examples

SFDR of Sinusoid

Obtain the SFDR for a 10 MHz tone with amplitude 1 sampled at 100 MHz. There is a spur at the 1st harmonic (20 MHz) with an amplitude of 3.16×10^{-4}.

```
deltat = 1e-8;
t = 0:deltat:1e-6-deltat;
x = cos(2*pi*10e6*t)+3.16e-4*cos(2*pi*20e6*t);
r = sfdr(x);
```

Minimum Spur Distance

Obtain the SFDR for a 10 MHz tone with amplitude 1 sampled at 100 MHz. There is a spur at the 1st harmonic (20 MHz) with an amplitude of 3.16×10^{-4}. Use a minimum spur distance of 1 MHz.

```
deltat = 1e-8;
fs = 1/deltat;
t = 0:deltat:1e-6-deltat;
x = cos(2*pi*10e6*t)+3.16e-4*cos(2*pi*20e6*t);
r = sfdr(x,fs,1e6);
```

SFDR from Periodogram

Obtain the power spectrum of a 10 MHz tone with amplitude 1 sampled at 100 MHz. There is a spur at the 1st harmonic (20 MHz) with an amplitude of 3.16×10^{-4}. Use the one-sided power spectrum and a vector of corresponding frequencies in Hz to compute the SFDR.

```
deltat = 1e-8;
fs = 1/deltat;
t = 0:deltat:1e-6-deltat;
x = cos(2*pi*10e6*t)+3.16e-4*cos(2*pi*20e6*t);
[sxx,f] = periodogram(x,rectwin(length(x)),length(x),fs,'power');
r = sfdr(sxx,f,'power');
```

Determine Frequency and Power of Largest Spur

Determine the frequency in MHz for the largest spur. The input signal is a 10 MHz tone with amplitude 1 sampled at 100 MHz. There is a spur at the 1st harmonic (20 MHz) with an amplitude of 3.16×10^{-4}.

```
deltat = 1e-8;
t = 0:deltat:1e-6-deltat;
x = cos(2*pi*10e6*t)+3.16e-4*cos(2*pi*20e6*t);
[r,spurpow,spurfreq] = sfdr(x,1/deltat);
spur_MHz = spurfreq/1e6;
```

DC Leakage Affects SFDR

This example shows how leakage from a DC signal shift can affect the SFDR when the input to `sfdr` is a power spectrum.

Create a signal sampled at 44.1 kHz. The signal consists of a superposition of two sinusoids with frequencies of 9.8 and 14.7 kHz in white Gaussian additive noise. Assuming the signal values are in volts, the 9.8-kHz sine wave has an amplitude of 1 volt and the 14.7-kHz sine wave has an amplitude of 100 microvolts. Equivalently, the power in the 14.7-kHz sine wave is 80 dB below the power of the 9.8-kHz sine wave. The additive white Gaussian noise has a mean of 0 and a variance of 0.001 microvolts. Additionally, the signal has a DC shift of 0.1 volts.

```
fs = 44.1e3;
f1 = 9.8e3;
f2 = 14.7e3;
N = 900;
nT = (1:N)/fs;
x = 0.1+sin(2*pi*f1*nT) + 100e-6*sin(2*pi*f2*nT) + sqrt(1e-9)*randn(1,N);
```

Determine the SFDR using both the time series data input and the power spectrum. The power spectrum in both cases is obtained using a Hamming window. Compare the results.

```
[sfd1, spur1, frq1] = sfdr(x, fs)
[sxx,f]=periodogram(x,hamming(length(x)),length(x),fs,'power');
[sfd2, spur2, frq2] = sfdr(sxx, f,'power')
```

The frequency resolution is 49 Hz. When the input to `sfdr` is the power spectrum obtained without first removing the mean, a spur is detected at the first nonzero-frequency DFT bin. In this case, the frequency of the first nonzero DFT bin is 49 Hz. However, when the input to `sfdr` is the time series data, the mean is subtracted prior to obtaining the power spectrum and the leakage from the DC component is avoided. `sfdr` correctly detects the spur at 14.7 kHz.

See Also

`bandpower` | `enbw` | `periodogram`

Purpose

Savitzky-Golay filter design

Syntax

```
b = sgolay(k,f)
b = sgolay(k,f,w)
[b,g] = sgolay(...)
```

Description

`b = sgolay(k,f)` designs a Savitzky-Golay FIR smoothing filter `b`. The polynomial order `k` must be less than the frame size, `f`, which must be odd. If `k = f-1`, the designed filter produces no smoothing. The output, `b`, is an `f`-by-`f` matrix whose rows represent the time-varying FIR filter coefficients. In a smoothing filter implementation (for example, `sgolayfilt`), the last `(f-1)/2` rows (each an FIR filter) are applied to the signal during the startup transient, and the first `(f-1)/2` rows are applied to the signal during the terminal transient. The center row is applied to the signal in the steady state.

`b = sgolay(k,f,w)` specifies a weighting vector `w` with length `f`, which contains the real, positive-valued weights to be used during the least-squares minimization.

`[b,g] = sgolay(...)` returns the matrix `g` of differentiation filters. Each column of `g` is a differentiation filter for derivatives of order `p`-1 where `p` is the column index. Given a signal `x` of length `f`, you can find an estimate of the p^{th} order derivative, `xp`, of its middle value from:

```
xp((f+1)/2) = (factorial(p)) * g(:,p+1)' * x
```

Tips

Savitzky-Golay smoothing filters (also called digital smoothing polynomial filters or least squares smoothing filters) are typically used to “smooth out” a noisy signal whose frequency span (without noise) is large. In this type of application, Savitzky-Golay smoothing filters perform much better than standard averaging FIR filters, which tend to filter out a significant portion of the signal’s high frequency content along with the noise. Although Savitzky-Golay filters are more effective at preserving the pertinent high frequency components of the signal, they are less successful than standard averaging FIR filters at rejecting noise when noise levels are particularly high. The particular formulation of Savitzky-Golay filters preserves various moment orders

better than other smoothing methods, which tend to preserve peak widths and heights better than Savitzky-Golay.

Savitzky-Golay filters are optimal in the sense that they minimize the least-squares error in fitting a polynomial to each frame of noisy data.

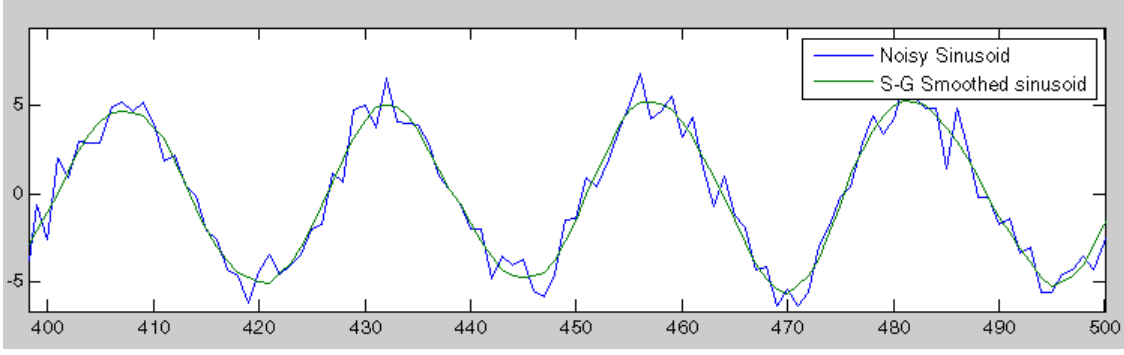
Examples

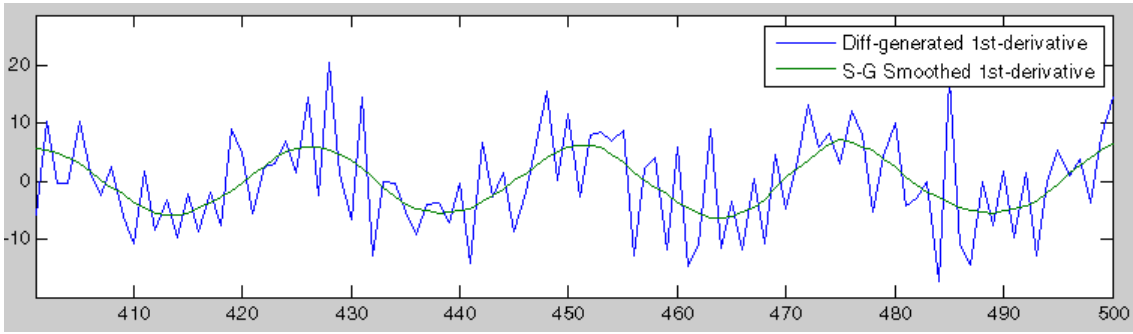
Use `sgolay` to smooth a noisy sinusoid and compare the resulting first and second derivatives to the first and second derivatives computed using `diff`. Notice how using `diff` amplifies the noise and generates useless results.

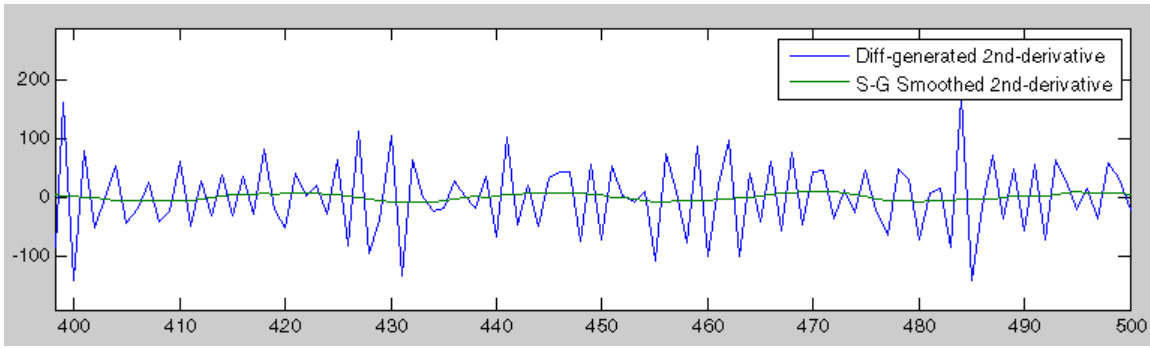
```
N = 4;                  % Order of polynomial fit
F = 21;                 % Window length
[b,g] = sgolay(N,F);   % Calculate S-G coefficients

dx = .2;
xLim = 200;
x = 0:dx:xLim-1;

y = 5*sin(0.4*pi*x)+randn(size(x));  % Sinusoid with noise

HalfWin  = ((F+1)/2) -1;
for n = (F+1)/2:996-(F+1)/2,
  % Zero-th derivative (smoothing only)
  SG0(n) =   dot(g(:,1), y(n - HalfWin: n + HalfWin));

  % 1st differential
  SG1(n) =   dot(g(:,2), y(n - HalfWin: n + HalfWin));

  % 2nd differential
  SG2(n) = 2*dot(g(:,3)', y(n - HalfWin: n + HalfWin))';
end

SG1 = SG1/dx;         % Turn differential into derivative
SG2 = SG2/(dx*dx);    % and into 2nd derivative

% Scale the "diff" results
```

```
DiffD1 = (diff(y(1:length(SG0)+1)))/ dx;
DiffD2 = (diff(diff(y(1:length(SG0)+2)))) / (dx*dx);

subplot(3,1,1);
plot([y(1:length(SG0))', SG0'])
legend('Noisy Sinusoid','S-G Smoothed sinusoid')

subplot(3, 1, 2);
plot([DiffD1',SG1'])
legend('Diff-generated 1st-derivative', ...
'S-G Smoothed 1st-derivative')

subplot(3, 1, 3);
plot([DiffD2',SG2'])
legend('Diff-generated 2nd-derivative',...
'S-G Smoothed 2nd-derivative')
```

Note The figures below are zoomed in each figure window panel to show more detail.







References

[1] Orfanidis, S.J., *Introduction to Signal Processing*, Prentice-Hall, Englewood Cliffs, NJ, 1996.

See Also

`fir1` | `firls` | `filter` | `sgolayfilt`

Purpose

Savitzky-Golay filtering

Syntax

```
y = sgolayfilt(x,k,f)
y = sgolayfilt(x,k,f,w)
y = sgolayfilt(x,k,f,w,dim)
```

Description

`y = sgolayfilt(x,k,f)` applies a Savitzky-Golay FIR smoothing filter to the data in vector `x`. If `x` is a matrix, `sgolayfilt` operates on each column. The polynomial order `k` must be less than the frame size, `f`, which must be odd. If `k = f-1`, the filter produces no smoothing.

`y = sgolayfilt(x,k,f,w)` specifies a weighting vector `w` with length `f`, which contains the real, positive-valued weights to be used during the least-squares minimization. If `w` is not specified or if it is specified as empty, `[]`, `w` defaults to an identity matrix.

`y = sgolayfilt(x,k,f,w,dim)` specifies the dimension, `dim`, along which the filter operates. If `dim` is not specified, `sgolayfilt` operates along the first non-singleton dimension; that is, dimension 1 for column vectors and nontrivial matrices, and dimension 2 for row vectors.

Tips

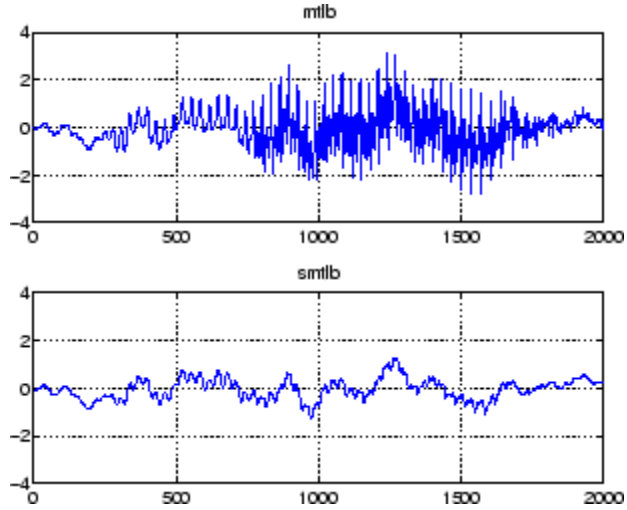
Savitzky-Golay smoothing filters (also called digital smoothing polynomial filters or least-squares smoothing filters) are typically used to "smooth out" a noisy signal whose frequency span (without noise) is large. In this type of application, Savitzky-Golay smoothing filters perform much better than standard averaging FIR filters, which tend to filter out a significant portion of the signal's high frequency content along with the noise. Although Savitzky-Golay filters are more effective at preserving the pertinent high frequency components of the signal, they are less successful than standard averaging FIR filters at rejecting noise.

Savitzky-Golay filters are optimal in the sense that they minimize the least-squares error in fitting a polynomial to frames of noisy data.

Examples

Smooth the `mtlb` signal by applying a cubic Savitzky-Golay filter to data frames of length 41:

```
load mtlb                           % Load data
smtlb = sgolayfilt(mtlb,3,41);   % Apply 3rd-order filter
subplot(2,1,1)
plot([1:2000],mtlb(1:2000)); axis([0 2000 -4 4]);
title('mtlb'); grid;
subplot(2,1,2)
plot([1:2000],smtlb(1:2000)); axis([0 2000 -4 4]);
title('smtlb'); grid;
```



References

[1] Orfanidis, S.J., *Introduction to Signal Processing*, Prentice-Hall, Englewood Cliffs, NJ, 1996.

See Also

`medfilt1` | `filter` | `sgolay` | `sosfilt`

Purpose

Shift data to operate on specified dimension

Syntax

```
[x,perm,nshifts] = shiftdata(x,dim)
```

Description

`[x,perm,nshifts] = shiftdata(x,dim)` shifts data `x` to permute dimension `dim` to the first column using the same permutation as the built-in `filter` function. The vector `perm` returns the permutation vector that is used.

If `dim` is missing or empty, then the first non-singleton dimension is shifted to the first column, and the number of shifts is returned in `nshifts`.

`shiftdata` is meant to be used in tandem with `unshiftdata`, which shifts the data back to its original shape. These functions are useful for creating functions that work along a certain dimension, like `filter`, `goertzel`, `sgolayfilt`, and `sosfilt`.

Examples

Example 1

This example shifts `x`, a 3-x-3 magic square, permuting dimension `2` to the first column. `unshiftdata` shifts `x` back to its original shape.

1. Create a 3-x-3 magic square:

```
x = fi(magic(3))
```

```
x =

     8     1     6
     3     5     7
     4     9     2
```

2. Shift the matrix `x` to work along the second dimension:

```
[x,perm,nshifts] = shiftdata(x,2)
```

The permutation vector, `perm`, and the number of shifts, `nshifts`, are returned along with the shifted matrix, `x`:

```
x =

     8     3     4
     1     5     9
     6     7     2


perm =

     2     1


nshifts =

     []
```

3. Shift the matrix back to its original shape:

```
y = unshiftdata(x,perm,nshifts)

y =

     8     1     6
     3     5     7
     4     9     2
```

Example 2

This example shows how `shiftdata` and `unshiftdata` work when you define `dim` as empty.

1. Define `x` as a row vector:

```
x = 1:5
```

```
x =

     1     2     3     4     5
```

2. Define `dim` as empty to shift the first non-singleton dimension of `x` to the first column:

```
[x,perm,nshifts] = shiftdata(x,[])
```

`x` is returned as a column vector, along with `perm`, the permutation vector, and `nshifts`, the number of shifts:

```
x =

     1
     2
     3
     4
     5


perm =

     []


nshifts =

     1
```

3. Using `unshiftdata`, restore `x` to its original shape:

```
y = unshiftdata(x,perm,nshifts)
```

```
y =

     1     2     3     4     5
```

See Also

permute | shiftdim | unshiftdata

Purpose

Signal processing window object

Syntax

```
w=sigwin.window
```

Description

`w=sigwin.window` returns a window object, `w`, of type *`window`*. Each window type takes one or more inputs. If you specify a `sigwin.window` with no inputs, a default window of length 64 is created.

Note You must specify a *`window`* type with `sigwin`.

Constructors

`window` for `sigwin` specifies the type of window. The following table lists the supported window functions with links to the corresponding class reference page for the window object.

Window	Window object
Modified Bartlett-Hanning Window	`sigwin.barthannwin`
Bartlett Window	`sigwin.bartlett`
Blackman Window	`sigwin.blackman`
Blackman-Harris Window	`sigwin.blackmanharris`
Bohman Window	`sigwin.bohmanwin`
Dolph-Chebyshev Window	`sigwin.chebwin`
Flat Top Window	`sigwin.flattopwin`
Gaussian Window	`sigwin.gausswin`
Hamming Window	`sigwin.hamming`
Hann (Hanning) Window	`sigwin.hann`
Kaiser Window	`sigwin.kaiser`

Window	Window object
Nuttall defined 4–term Blackman-Harris Window	`sigwin.nuttallwin`
Parzen Window	`sigwin.parzenwin`
Rectangular Window	`sigwin.rectwin`
Taylor Window	`sigwin.taylorwin`
Triangular Window	`sigwin.triang`
Tukey Window	`sigwin.tukeywin`

Methods

Methods provide ways of performing functions directly on your `sigwin` object without having to specify the window parameters again. You can apply this method directly on the variable you assigned to your `sigwin` object.

Method	Description
`generate`	Returns a column vector of values representing the window.
`info`	Returns information about the window object.
`winwrite`	Writes an ASCII file that contains window weights for a single window object or a vector of window objects. Default filename is `untitled.wf`. `winwrite(Hd,filename)` writes to a disk file named `filename` in the current working directory. The `.wf` extension is added automatically.

Viewing Object Parameters

As with any object, you can use `get` to view a `sigwin` object's parameters. To see a specific parameter,

```
get(w,'parameter')
```

or to see all parameters for an object,

```
get(w)
```

Changing Object Parameters

To set specific parameters,

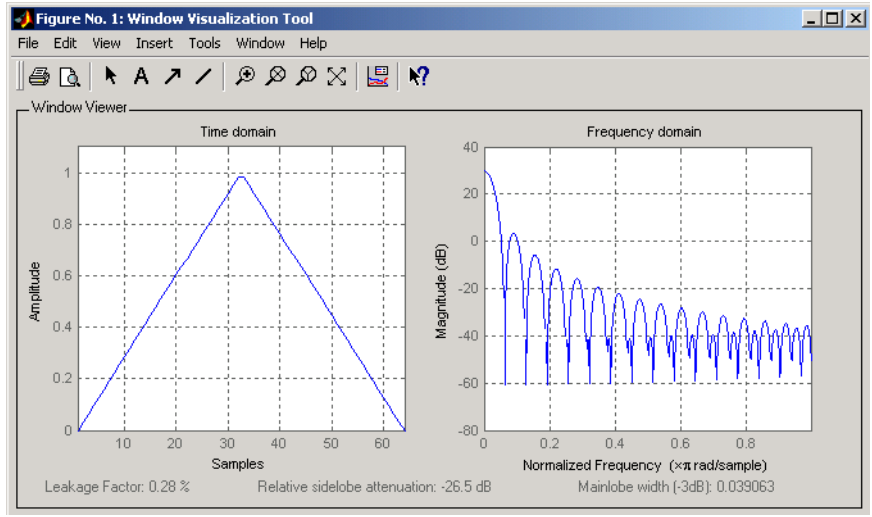
```
set(w,'parameter1',value,'parameter2',value,...)
```

Note that you must use single quotation marks around the parameter name.

Examples

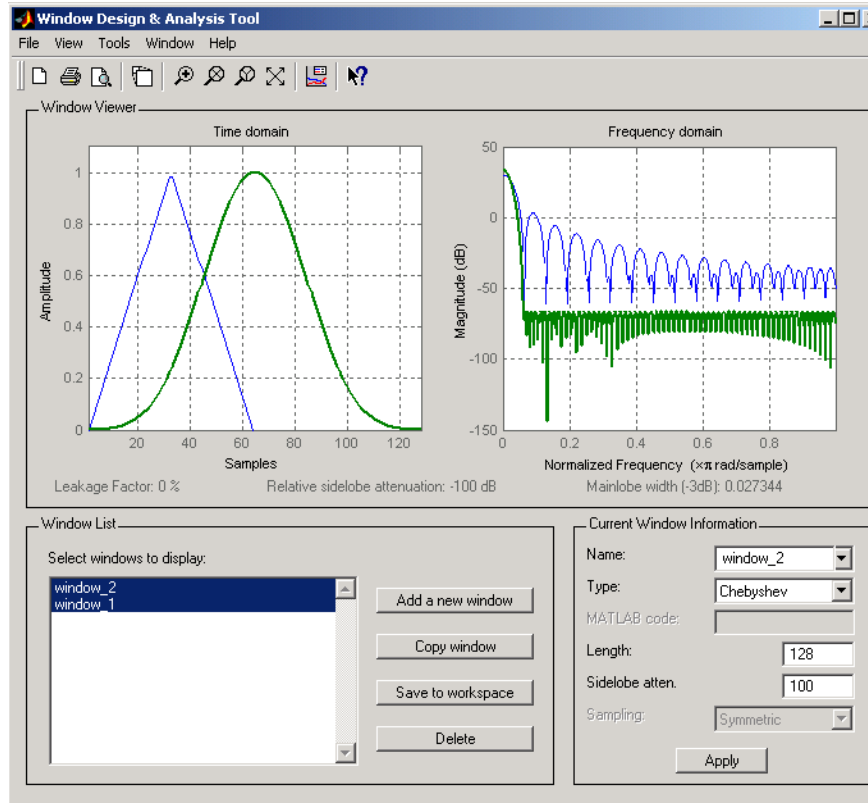
Create a default Bartlett window and view the results in the Window Visualization Tool (`wvtool`). See `bartlett` for information on Bartlett windows:

```
w=sigwin.bartlett

w =
    Length: 64
      Name: 'Bartlett'

wvtool(w)
```



Create a 128-point Chebyshev window with 100 dB of sidelobe attenuation. (See `chebwin` for information on Chebyshev windows.) View the results of this and the above Bartlett window in the Window Design and Analysis Tool (`wintool`):

```
w1=sigwin.chebwin(128,100)

w1 =

            Length: 128
              Name: 'Chebyshev'
     SidelobeAtten: 100

wintool(w,w1)
```



To save the window values in a vector, use:

```
d = generate(w);
```

See Also

`window` | `wintool` | `wvtool`

sigwin.barthannwin

Purpose

Construct Bartlett-Hanning window object

Description

`sigwin.barthannwin` creates a handle to a Bartlett-Hanning window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines a modified Bartlett-Hanning window of length *N*:

$$w(x) = 0.62 - 0.48\,|\,x\,| + 0.38\cos(2\pi x) \quad -1/2 \le x \le 1/2$$

where *x* is an *N*-point linearly spaced vector over the interval [1/2, 1/2].

Construction

`H = sigwin.barthannwin` returns a modified Bartlett-Hanning window object `H` of length 64.

`H = sigwin.barthannwin(`*`Length`*`)` returns a modified Bartlett-Hanning window object `H` of length *`Length`*. *`Length`* requires a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

Properties

Length

Modified Bartlett-Hanning window length. The window length requires a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

Methods

`generate`	Generates modified Bartlett-Hanning window
`info`	Display information about modified Bartlett-Hanning window object
`winwrite`	Save Bartlett window object values in ASCII file

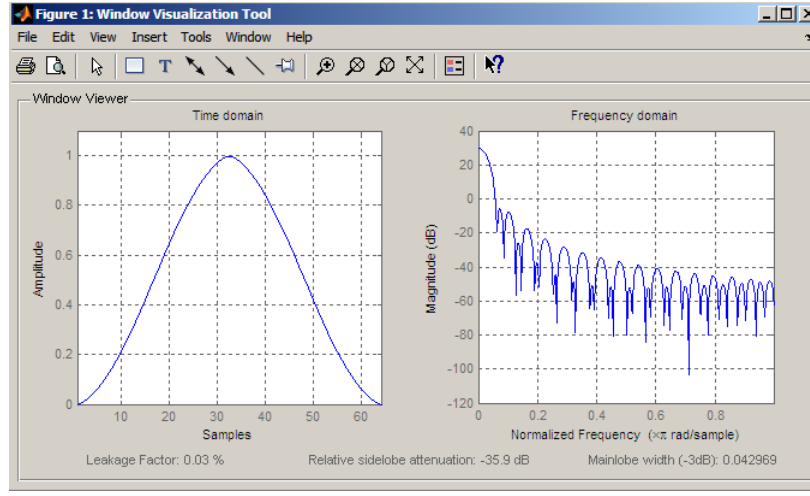
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` modified Bartlett-Hanning window:

```
H=sigwin.barthannwin;
wvtool(H);
```



Generate length `N=128` modified Bartlett-Hanning window, return values, and write ASCII file with window values:

```
H=sigwin.barthannwin(128);
% Return window with generate
win=generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'barthannwin_128')
```

References

Yeong, H.H., and Pearce, J.A. “A New Window and Comparison to Standard Windows,” *IEEE Transactions on Acoustics, Speech and Signal Processing*, Vol. 37, 1989, pp. 298–301.

See Also

`sigwin` | `window` | `wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

Purpose

Generates modified Bartlett-Hanning window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the modified Bartlett-Hanning window object `H` as a double-precision column vector.

Examples

Extract values from modified Bartlett-Hanning window object:

```
H=sigwin.barthannwin(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.barthannwin.info

Purpose

Display information about modified Bartlett-Hanning window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length information for the modified Bartlett-Hanning window object `H`.

`info_win = info(H)` returns length information for the modified Bartlett-Hanning window object `H` in the character array `info_win`.

Examples

Return information about a modified Bartlett-Hanning window object:

```
H = sigwin.barthannwin(256);
info_win = info(H);
```

Purpose

Save Bartlett window object values in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog box that enables you to export the values of the modified Bartlett-Hanning window object `H` to an ASCII file with filename extension `wf` .

`winwrite(H,'filename')` saves the values of the modified Bartlett-Hanning window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The filename extension is `wf`.

Examples

Write modified Bartlett-Hanning window values to ASCII file:

```
H = sigwin.barthannwin;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.bartlett

Purpose

Construct Bartlett window object

Description

`sigwin.bartlett` creates a handle to a Bartlett window object for use in spectral analysis and filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

For *N* even, the following equation defines the Bartlett window:

$$w(n) = \begin{cases} \dfrac{2n}{N-1} & 0 \le n \le N/2 - 1 \\ 2 - \dfrac{2n}{N-1} & N/2 \le n \le N-1 \end{cases}$$

For *N* odd, the equation for the Bartlett window is:

$$w(n) = \begin{cases} \dfrac{2n}{N-1} & 0 \le n \le (N-1)/2 \\ 2 - \dfrac{2n}{N-1} & (N-1)/2 + 1 \le n \le N-1 \end{cases}$$

Construction

`H = sigwin.bartlett` returns a Bartlett window object `H` of length 64.

`H = sigwin.bartlett(Length)` returns a Bartlett window object `H` of length *Length*. *Length* must be a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

Properties

Length

Bartlett window length. The length requires a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

Methods

`generate`	Generates Bartlett window
`info`	Display information about Bartlett window object
`winwrite`	Save Bartlett window object values in ASCII file

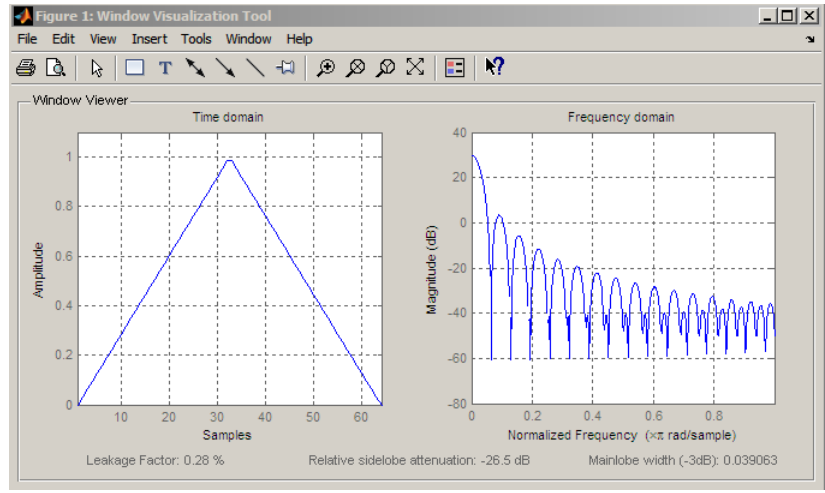
Copy Semantics

Handle. To learn how this affects your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Create default length `N=64` Bartlett window:

```
H = sigwin.bartlett;
wvtool(H);
```



Generate length `N=128` Bartlett window, return values, and write ASCII file with window values:

```
H = sigwin.bartlett(128);
% Return window with generate
```

```
win = generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'bartlett_128')
```

References

Oppenheim, A.V., and Schafer, R.W. *Discrete-time Signal Processing*, Upper Saddle River, N.J: Prentice Hall, 1989, pp. 444–447.

See Also

`sigwin | window | wvtool`

Tutorials

- "Windows"

How To

- Class Attributes
- Property Attributes

Purpose

Generates Bartlett window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Bartlett window object `H` as a double-precision column vector.

Examples

Extract values from Bartlett window object:

```
H = sigwin.bartlett(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.bartlett.info

Purpose

Display information about Bartlett window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length information for the Bartlett window object `H`.

`info_win = info(H)` returns length information for the Bartlett window object `H` in the character array `info_win`.

Examples

Return information about a Bartlett window object:

```
H = sigwin.bartlett(256);
info_win = info(H);
```

Purpose

Save Bartlett window object values in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog box that enables you to export the values of the Bartlett window object `H` to an ASCII file with filename extension `wf`.

`winwrite(H,'filename')` saves the values of the Bartlett window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The filename extension is `wf`.

Examples

Write Bartlett window values to ASCII file:

```
H=sigwin.bartlett;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.blackman

Purpose

Construct Blackman window object

Description

`sigwin.blackman` creates a handle to a Blackman window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines the Blackman window of length *N*:

$$w(n) = 0.42 - 0.5\cos(2\pi n / (N-1)) + 0.08\cos(4\pi n / (N-1)) \quad 0 \le n \le M-1$$

where *M* is *N*/2 for *N* even and *(N+1)*/2 for *N* odd.

In the symmetric case, the second half of the Blackman window $M \le n \le N-1$ is obtained by flipping the first half around the midpoint. The symmetric option is the preferred method when using a Blackman window in FIR filter design.

The periodic Blackman window is constructed by extending the desired window length by one sample to *N+1*, constructing a symmetric window, and removing the last sample. The periodic version is the preferred method when using a Blackman window in spectral analysis because the discrete Fourier transform assumes periodic extension of the input vector.

Construction

`H = sigwin.blackman` returns a Blackman window object `H` of length 64 with symmetric sampling.

`H = sigwin.blackman(`*`Length`*`)` returns a Blackman window object `H` of length *`Length`* with symmetric sampling. *`Length`* requires a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

`H = sigwin.blackman(`*`Length`*`,`*`SamplingFlag`*`)` returns a Blackman window object `H` with sampling *`Sampling_Flag`*. The *`Sampling_Flag`* can be either `'symmetric'` or `'periodic'`.

Properties

Length

Blackman window length. Must be a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

SamplingFlag

`'symmetric'` is the default and forces exact symmetry between the first and second halves of the Blackman window. A symmetric window is preferred in FIR filter design by the window method.

`'periodic'` designs a symmetric Blackman window of length *Length*+1 and truncates the window to length *Length*. This design is preferred in spectral analysis where the window is treated as one period of a *Length*-point periodic sequence.

Methods

`generate`	Generates Blackman window
`info`	Display information about Blackman window object
`winwrite`	Save Blackman window in ASCII file

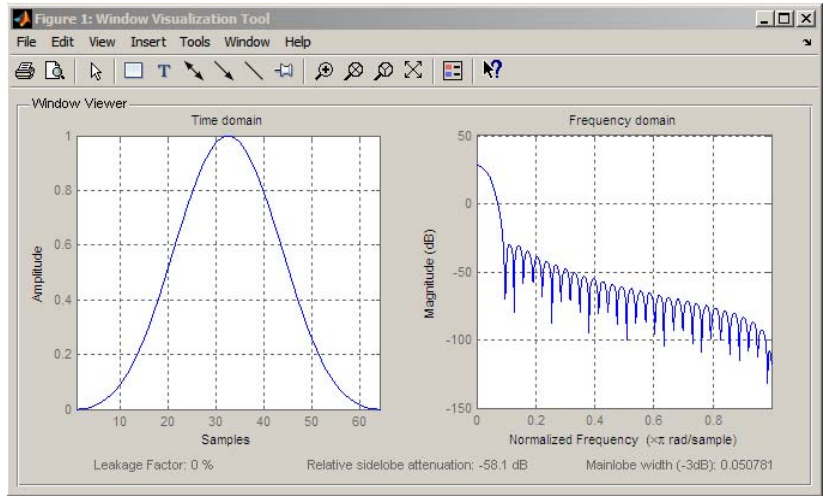
Copy Semantics

Handle. To learn how this affects your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` symmetric Blackman window:

```
H = sigwin.blackman;
wvtool(H);
```



Generate length `N=128` periodic Blackman window, return values, and write ASCII file:

```
H = sigwin.blackman(128,'periodic');
% Return window with generate
win = generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'blackman_128')
```

References

Oppenheim, A.V. and Schafer, R.W. *Discrete-time Signal Processing*, Upper Saddle River, N.J: Prentice Hall, 1989, pp. 444–447.

See Also

`sigwin | window | wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

Purpose

Generates Blackman window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Blackman window object `H` as a double-precision column vector.

Examples

Extract values from Blackman window object:

```
H = sigwin.blackman(128);
% Extract window values as column vector
win = generate(H);
```

sigwin.blackman.info

Purpose

Display information about Blackman window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length and sampling information about the Blackman window object `H`.

`info_win = info(H)` returns length and sampling information about the Blackman window object `H` in the character array `info_win`.

Examples

Return information about a Blackman window object:

```
H = sigwin.blackman(256);
info_win = info(H);
```

Purpose

Save Blackman window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog box that enables you to export the values of the Blackman window object `H` to an ASCII file with filename extension `wf` .

`winwrite(H,'filename')` saves the values of the Blackman window object `H` in the current folder as a column vector in the ASCII file `'filename'` with filename extension `wf`.

Examples

Write Blackman window values to ASCII file:

```
H=sigwin.blackman;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.blackmanharris

Purpose

Construct Blackman–Harris window object

Description

`sigwin.blackmanharris` creates a handle to a Blackman-Harris window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines the **symmetric** Blackman-Harris window of length *N*:

$$w(n) = a_0 - a_1 \cos(\frac{2\pi n}{N-1}) + a_2 \cos(\frac{4\pi n}{N-1}) - a_3 \cos(\frac{6\pi n}{N-1}) \quad 0 \le n \le N-1$$

The following equation defines the **periodic** Blackman-Harris window of length *N*:

$$w(n) = a_0 - a_1 \cos(\frac{2\pi n}{N}) + a_2 \cos(\frac{4\pi n}{N}) - a_3 \cos(\frac{6\pi n}{N}) \quad 0 \le n \le N-1$$

The following table lists the coefficients:

Coefficient	Value
a0	0.35875
a1	0.48829
a2	0.14128
a3	0.01168

Construction

`H = sigwin.blackmanharris` returns a Blackman-Harris window object `H` of length 64.

`H = sigwin.blackmanharris(Length)` returns a Blackman-Harris window object `H` of length `Length`. `Length` must be a positive integer. Entering a positive noninteger value for `Length` rounds the length to the nearest integer. Entering a 1 for `Length` results in a window with a single value of 1.

Properties

Length

Blackman-Harris window length. The window length requires a positive integer. Entering a positive noninteger value for `Length` rounds the length to the nearest integer. Entering a 1 for `Length` results in a window with a single value of 1.

SamplingFlag

The type of window returned as one of `'symmetric'` or `'periodic'`. The default is `'symmetric'`. A symmetric window exhibits perfect symmetry between halves of the window. Setting the `SamplingFlag` property to `'periodic'` results in a N-periodic window. The equations for the Blackman-Harris window differ slightly based on the value of the `SamplingFlag` property. See "Description" on page 1-990 for details.

Methods

`generate`	Generates Blackman–Harris window
`info`	Display information about Blackman–Harris window object
`winwrite`	Save Blackman–Harris window in ASCII file

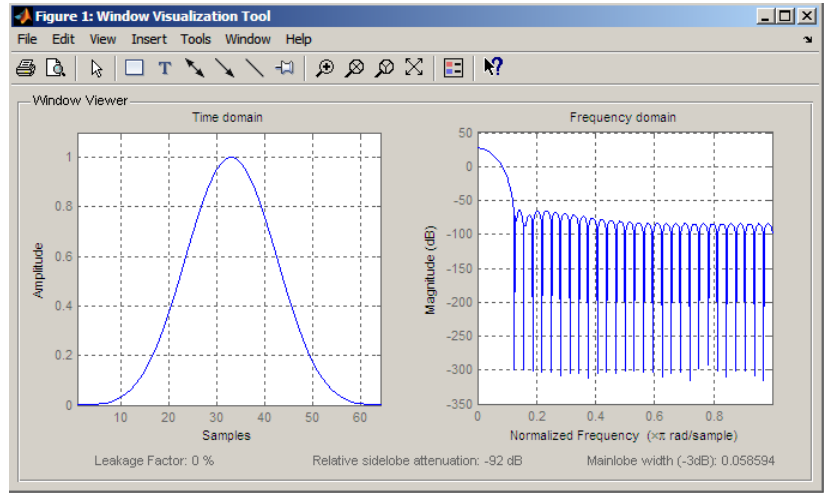
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` Blackman-Harris window:

```
H=sigwin.blackmanharris;
wvtool(H);
```



Generate length `N=128` periodic Blackman-Harris window, return values, and write ASCII file:

```
H=sigwin.blackmanharris(128);
H.SamplingFlag = 'periodic';
% Return window with generate
win=generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'blackmanharris_128')
```

References

Harris, F. J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform,” *Proceedings of the IEEE.*. Vol. 66, 1978.

See Also

`sigwin` | `window` | `wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

Purpose

Generates Blackman–Harris window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Blackman–Harris window object `H` as a double-precision column vector.

Examples

Extract values from Blackman–Harris window object:

```
H=sigwin.blackmanharris(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.blackmanharris.info

Purpose

Display information about Blackman–Harris window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length information for the Blackman–Harris window object `H`.

`info_win = info(H)` returns length information for the Blackman–Harris window object `H` in the character array `info_win`.

Examples

Return information about a Blackman–Harris window object:

```
H = sigwin.blackmanharris(256);
info_win = info(H);
```

Purpose

Save Blackman–Harris window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog box that enables you to export the values of the Blackman–Harris window object `H` to an ASCII file with filename extension `wf`.

`winwrite(H,'filename')` saves the values of the Blackman–Harris window object `H` in the current folder as a column vector in the ASCII file `'filename'` with filename extension `wf`.

Examples

Write Blackman–Harris window values to ASCII file:

```
H=sigwin.blackmanharris;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.bohmanwin

Purpose

Construct Bohman window object

Description

`sigwin.bohmanwin` creates a handle to a Bohman window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines the Bohman window of length *N*:

$$w(x) = (1 - |x|)\cos(\pi |x|) + \frac{1}{\pi}\sin(\pi |x|) \quad -1 \le x \le 1$$

where *x* is a length *N* vector of linearly spaced values generated using `linspace`. The first and last elements of the Bohman window are forced to be identically zero.

Construction

`H = sigwin.bohmanwin` returns a Bohman window object `H` of length 64.

`H = sigwin.bohmanwin(`*`Length`*`)` returns a Bohman window object `H` of length *`Length`*. *`Length`* is a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

Properties

Length

Bohman window length. Must be a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

Methods

`generate`	Generates Bohman window
`info`	Display information about Bohman window object
`winwrite`	Save Bohman window object values in ASCII file

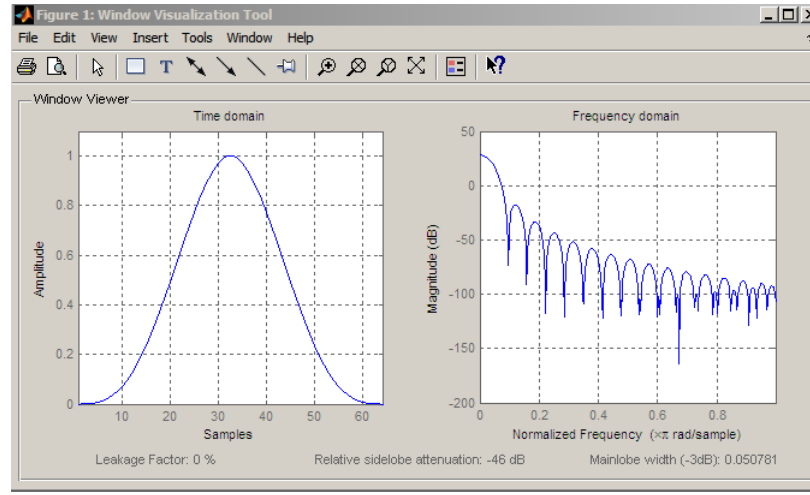
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` Bohman window:

```
H=sigwin.bohmanwin;
wvtool(H);
```



Generate length `N=128` Bohman window, return values, and write ASCII file:

```
H=sigwin.bohmanwin(128);
% Return window with generate
win=generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'bohmanwin_128')
```

References

Harris, F.J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform.”*Proceedings of the IEEE*.Vol. 66, 1978, pp. 51–83.

See Also

`sigwin` | `window` | `wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

Purpose

Generates Bohman window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Bohman window object as a double-precision column vector.

Examples

Extract values from Bohman window object:

```
H=sigwin.bohmanwin(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.bohmanwin.info

Purpose

Display information about Bohman window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length information for the Bohman window object `H`.

`info_win = info(H)` returns length information for the Bohman window object `H` in the character array `info_win`.

Examples

Return information for a Bohman window object:

```
H=sigwin.bohmanwin(256);
info_win=info(H);
```

Purpose

Save Bohman window object values in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of the Bohman window object `H` to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Bohman window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Bohman window values to ASCII file:

```
H=sigwin.bohmanwin;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.chebwin

Purpose

Construct Dolph-Chebyshev window object

Description

`sigwin.chebwin` creates a handle to a Dolph–Chebyshev window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The Dolph-Chebyshev window is constructed in the frequency domain by taking samples of the window's Fourier transform:

$$\hat{W}(k) = (-1)^k \frac{\cos[N \cos^{-1}[\beta \cos(\pi k / N)]]}{\cosh[N \cosh^{-1}(\beta)]} \quad 0 \le k \le N-1$$

where

$$\beta = \cos[1 / N \cosh^{-1}(10^{\alpha})]$$

α determines the level of the sidelobe attenuation. The level of the sidelobe attenuation is equal to -20α. For example, 100 dB of attenuation results from setting $\alpha = 5$

The discrete-time Dolph-Chebyshev window is obtained by taking the inverse DFT of $\hat{W}(k)$ and scaling the result to have a peak value of 1.

Construction

`H = sigwin.chebwin` returns a Dolph-Chebyshev window object `H` of length 64 with relative sidelobe attenuation of 100 dB.

`H = sigwin.chebwin(`*`Length`*`)` returns a Dolph–Chebyshev window object `H` of length *`Length`* with relative sidelobe attenuation of 100 dB. *`Length`* requires a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. A window length of 1 results in a window with a single value equal to 1.

`H = sigwin.chebwin(`*`Length`*`,`*`SidelobeAtten`*`)` returns a Dolph-Chebyshev window object with relative sidelobe attenuation of *`atten_param`* dB.

Properties

Length

Dolph-Chebyshev window length.

SidelobeAtten

The attenuation parameter in dB. The attenuation parameter is a positive real number that determines the relative sidelobe attenuation of the window.

Methods

`generate`	Generates Dolph-Chebyshev window
`info`	Display information about Dolph–Chebyshev window object
`winwrite`	Save Dolph-Chebyshev window object values in ASCII file

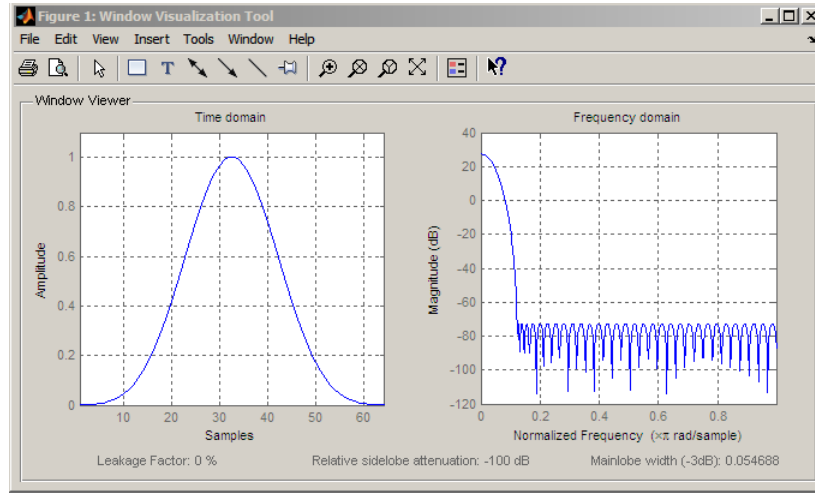
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` Dolph–Chebyshev window with 100 dB relative sidelobe attenuation:

```
H=sigwin.chebwin;
wvtool(H);
```



Generate length `N=128` Chebyshev window with 120 dB attenuation, return values, and write ASCII file:

```
H=sigwin.chebwin(128,120);
% Return window with generate
win=generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'chebwin_128_100')
```

References

Harris.F.J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform.” *Proceedings of the IEEE*. Vol. 66, 1978, pp. 51–83.

See Also

`sigwin | window | wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

Purpose

Generates Dolph-Chebyshev window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Dolph-Chebyshev window object `H` as a double-precision column vector.

Examples

Extract values from Dolph-Chebyshev window object:

```
H=sigwin.chebwin(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.chebwin.info

Purpose

Display information about Dolph–Chebyshev window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length and relative sidelobe attenuation information for the Dolph-Chebyshev window object `H`.

`info_win = info(H)` returns length information for the Dolph-Chebyshev window object `H` in the character array `info_win`.

Examples

Return information about a Dolph-Chebyshev window object:

```
H=sigwin.chebwin(256);
info_win=info(H);
```

Purpose

Save Dolph-Chebyshev window object values in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of the Dolph-Chebyshev window object `H` to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Dolph-Chebyshev window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Dolph-Chebyshev window values to ASCII file:

```
H=sigwin.chebwin;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.flattopwin

Purpose

Construct flat top window object

Description

`sigwin.flattopwin` creates a handle to a flat top window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

Construction

`H = sigwin.flattopwin` returns a flat top window object `H` of length 64 with symmetric sampling.

`H = sigwin.flattopwin(`*`Length`*`)` returns a flat top window object of length *`Length`* with symmetric sampling. *`Length`* must be a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

`H = sigwin.flattopwin(`*`Length`*`,`*`SamplingFlag`*`)` returns a flat top window object `H` of length *`Length`* with sampling *`SamplingFlag`*. The *`SamplingFlag`* can be either `'symmetric'` or `'periodic'`.

Properties

Length

Flat top window length. Must be a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

SamplingFlag

`'symmetric'` is the default and forces exact symmetry between the first and second halves of the flat top window. A symmetric window is preferred in FIR filter design.

`'periodic'` designs a symmetric flat top window of length *`Length`*`+1` and truncates the window to length *`Length`*. This design is preferred in spectral analysis where the window is treated as one period of a *`Length`*-point periodic sequence.

Methods

`generate`	Generates flat top window
`info`	Display information about flat top window object
`winwrite`	Save flat top window in ASCII file

Definitions

The following equation defines the flat top window of length *N*:

$$w(n) = a_0 - a_1 \cos(2\pi n / (N-1)) + a_2 \cos(4\pi n / (N-1)) - a_3 \cos(6\pi n / (N-1)) + a_4 \co$$

where *M* is *N*/2 for *N* even and *(N+1)*/2 for *N* odd.

The second half of the symmetric flat top window $M \le n \le N-1$ is obtained by flipping the first half around the midpoint. The symmetric option is the preferred method when using a flat top window in FIR filter design by the window method.

The periodic flat top window is constructed by extending the desired window length by one sample, constructing a symmetric window, and removing the last sample. The periodic version is the preferred method when using a flat top window in spectral analysis because the discrete Fourier transform assumes periodic extension of the input vector.

The coefficients are listed in the following table:

Coefficient	Value
a0	0.21557895
a1	0.41663158
a2	0.277263158
a3	0.083578947
a4	0.006947368

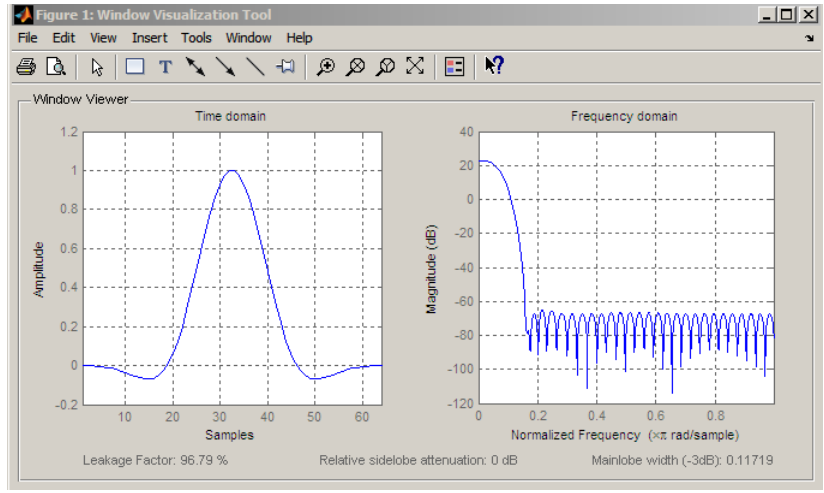
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` symmetric flat top window:

```
H=sigwin.flattopwin;
wvtool(H);
```



Generate length `N=128` periodic flat top window, return values, and write ASCII file:

```
H=sigwin.flattopwin(128,'periodic');
% Return window with generate
win=generate(H);
% Write ascii file in current directory
% with window values
winwrite(H,'flattopwin_128')
```

References

Oppenheim, A.V. and Schafer, R.W. *Discrete-time Signal Processing*, Upper Saddle River, N.J: Prentice Hall, 1989, pp. 444–447.

See Also

`sigwin` | `window` | `wvtool`

Tutorials

- "Windows"

How To

- Class Attributes
- Property Attributes

sigwin.flattopwin.generate

Purpose

Generates flat top window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the flat top window object as a double-precision column vector.

Examples

Extract values from flat top window object:

```
H=sigwin.flattopwin(128);
% Extract window values as column vector
win=generate(H);
```

Purpose

Display information about flat top window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length and sampling information for the flat top window object `H`.

`info_win = info(H)` returns length and sampling information for the flat top window object `H` in the character array `info_win`.

Examples

Return information about a flat top window object:

```
H=sigwin.flattopwin(256);
info_win=info(H);
```

sigwin.flattopwin.winwrite

Purpose

Save flat top window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the flat top window values to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the flat top window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write flat top window values to ASCII file:

```
H=sigwin.flattopwin;
% Open dialog for ASCII file
winwrite(H);
```

Purpose

Construct Gaussian window object

Description

`sigwin.gausswin` creates a handle to a Gaussian window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines the Gaussian window of length `N`:

$$w(x) = e^{-1/2(\alpha^2 x^2/M^2)} \quad -M \le x \le M$$

where `M=(N-1)/2` and `x` is a linearly spaced vector of length `N`.

Equating α with the usual standard deviation of a Gaussian value, σ, note:

$$\alpha = \frac{(N-1)}{2\sigma}$$

Construction

`H = sigwin.gausswin` returns a Gaussian window object `H` of length 64 and dispersion parameter `alpha` of 2.5.

`H = sigwin.gausswin(Length)` returns a Gaussian window object `H` of length `Length` and dispersion parameter `alpha` of 2.5. `Length` requires a positive integer. Entering a positive noninteger value for `Length` rounds the length to the nearest integer. Entering a 1 for `Length` results in a window with a single value of 1.

`H = sigwin.gausswin(Length,Alpha)` returns a Gaussian window object with dispersion parameter `alpha`. `alpha` requires a nonnegative real number and is inversely proportional to the standard deviation of a Gaussian value.

Properties

Length

Gaussian window length. The window length requires a positive integer. Entering a positive noninteger value for `Length` rounds the length to the nearest integer. Entering a 1 for `Length` results in a window with a single value of 1.

Alpha

Width of Gaussian window. `Alpha` is inversely proportional to the standard deviation of a Gaussian. Larger values of `Alpha` produce Gaussian windows with inflection points closer to the peak value, or narrower windows. In the frequency domain, larger values of `Alpha` produce a Gaussian window with increased spread of the main lobe in frequency but decreased sidelobe energy.

Methods

`generate`	Generates Gaussian window
`info`	Display information about Gaussian window object
`winwrite`	Save Gaussian window in ASCII file

Copy Semantics

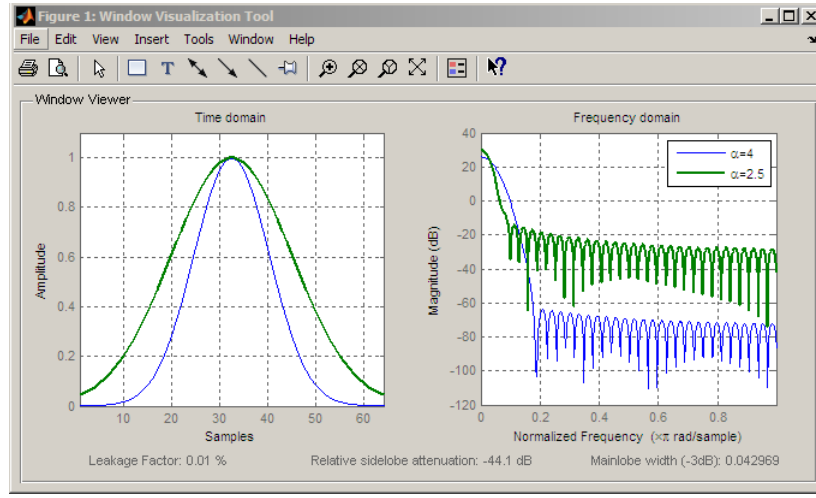
Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Compare two Gaussian windows with different `alpha` values:

```
H=sigwin.gausswin(64,4);
H1=sigwin.gausswin(64,2.5);
% Plot comparison
fwvt=wvtool(H,H1);
legend(get(fwvt,'currentaxes'),'\alpha=4','\alpha=2.5');
```

The main lobe is wider for `alpha=4` but the window, with `alpha=4`, demonstrates reduced sidelobe energy.



References

Harris, F.J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform,” *Proceedings of the IEEE*. Vol. 66, 1978, pp. 51–83.

See Also

`sigwin` | `window` | `wvtool`

Tutorials

- “Windows”

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sigwin.gausswin.generate

Purpose

Generates Gaussian window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Gaussian window object `H` as a double-precision column vector.

Examples

Extract values from Gaussian window object:

```
H=sigwin.gausswin(128,4);
% Extract window values as column vector
win=generate(H);
```

Purpose

Display information about Gaussian window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length and dispersion information for the Gaussian window object `H`.

`info_win = info(H)` returns length and dispersion information for the Gaussian window object `H` in the character array `info_win`.

Examples

Return information about a Gaussian window object:

```
H=sigwin.gausswin(256);
info_win=info(H);
```

sigwin.gausswin.winwrite

Purpose

Save Gaussian window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of Gaussian window object `H` to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Gaussian window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Gaussian window values to ASCII file:

```
H=sigwin.gausswin;
% Open dialog  for ASCII file
winwrite(H);
```

Purpose

Construct Hamming window object

Description

`sigwin.hamming` creates a handle to a Hamming window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines the Hamming window of length *N*:

$$w(n) = 0.54 - 0.46\cos(2\pi n / N - 1) \quad 0 \le n \le M - 1$$

where *M* is *N*/2 for *N* even and *(N+1)*/2 for *N* odd.

The second half of the symmetric Hamming window $M \le n \le N - 1$ is obtained by flipping the first half around the midpoint. The symmetric option is the preferred method when using a Hamming window in FIR filter design.

The periodic Hamming window is constructed by extending the desired window length by one sample, constructing a symmetric window, and removing the last sample. The periodic version is the preferred method when using a Hamming window in spectral analysis because the discrete Fourier transform assumes periodic extension of the input vector.

Construction

`H = sigwin.hamming` returns a symmetric Hamming window object `H` of length 64.

`H = sigwin.hamming(`*`Length`*`)` returns a symmetric Hamming window object with length *`Length`* . *`Length`* must be a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

`H = sigwin.hamming(`*`Length`*`,`*`SamplingFlag`*`)` returns a Hamming window with sampling *`Sampling_Flag`*. The *`SamplingFlag`* can be either `'symmetric'` or `'periodic'`.

Properties

Length

Hamming window length. The window length must be a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

SamplingFlag

`'symmetric'` is the default and forces exact symmetry between the first and second halves of the Hamming window. A symmetric window is preferred in FIR filter design by the window method.

`'periodic'` designs a symmetric Hamming window of length *Length+1* and truncates the window to length *Length*. This design is preferred in spectral analysis where the window is treated as one period of a *Length*-point periodic sequence.

Methods

`generate`	Generates Hamming window
`info`	Display information about Hamming window object
`winwrite`	Save Hamming window in ASCII file

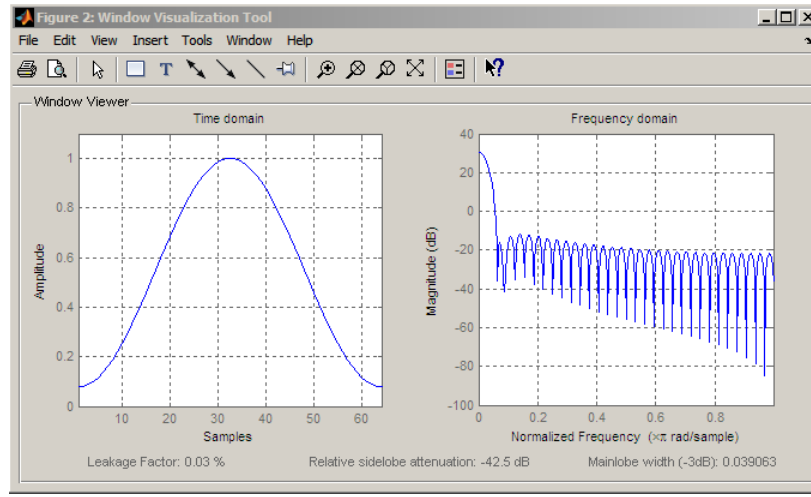
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` symmetric Hamming window:

```
H=sigwin.hamming;
wvtool(H);
```



Generate a length `N=128` periodic Hamming window, return the values, and write ASCII file:

```
H=sigwin.hamming(128,'periodic');
% Return window values with generate
win=generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'hamming_128')
```

References

Oppenheim, A.V. and Schafer, R.W. *Discrete-time Signal Processing*, Upper Saddle River, N.J: Prentice Hall, 1989, pp. 444–447.

See Also

`sigwin | window | wvtool`

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- “Windows”

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sigwin.hamming.generate

Purpose

Generates Hamming window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Hamming window object as a double-precision column vector.

Examples

Extract values from Hamming window object:

```
H=sigwin.hamming(128);
% Extract window values as column vector
win=generate(H);
```

Purpose

Display information about Hamming window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length and sampling information for the Hamming window object `H`.

`info_win = info(H)` returns length and sampling information for the Hamming window object `H` in the character array `info_win`.

Examples

Return information about a Hamming window object:

```
H=sigwin.hamming(256);
info_win=info(H);
```

sigwin.hamming.winwrite

Purpose

Save Hamming window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the Hamming window values to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Hamming window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Hamming window values to ASCII file:

```
H=sigwin.hamming;
% Open dialog box for ASCII file
winwrite(H);
```

Purpose

Construct Hann (Hanning) window object

Description

`sigwin.hann` creates a handle to a Hann window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The symmetric Hann window of length `N` is defined as:

$$w(n) = 0.5(1 - \cos(2\pi n / N - 1)) \quad 0 \le n \le M - 1$$

where `M` is `N/2` for `N` even and `(N+1)/2` for `N` odd.

The second half of the symmetric Hann window $M \le n \le N - 1$ is obtained by flipping the first half around the midpoint. The symmetric option is the preferred method when using a Hann window in FIR filter design.

The periodic Hann window is constructed by extending the desired window length by one sample, constructing a symmetric window, and removing the last sample. The periodic version is the preferred method when using a Hann window in spectral analysis because the discrete Fourier transform assumes periodic extension of the input vector.

Construction

`H = sigwin.hann` returns a symmetric Hann window object `H` of length 64.

`H = sigwin.hann(Length)` returns a symmetric Hann window object with length `Length`. `Length` requires a positive integer. Entering a positive noninteger value for `Length` rounds the length to the nearest integer. Entering a 1 for `Length` results in a window with a single value of 1.

`H = sigwin.hann(Length,SamplingFlag)` returns a Hann window object with sampling `Sampling_Flag`. The `SamplingFlag` can be either `'symmetric'` or `'periodic'`.

Properties

Length

Hann window length. Must be a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

SamplingFlag

`'symmetric'` is the default and forces exact symmetry between the first and second halves of the Hann window. A symmetric window is preferred in FIR filter design by the window method.

`'periodic'` designs a symmetric Hann window of length *Length+1* and truncates the window to length *Length*. This design is preferred in spectral analysis where the window is treated as one period of a *Length*-point periodic sequence.

Methods

`generate`	Generates Hann window
`info`	Display information about Hann window object
`winwrite`	Save Hann window object values in ASCII file

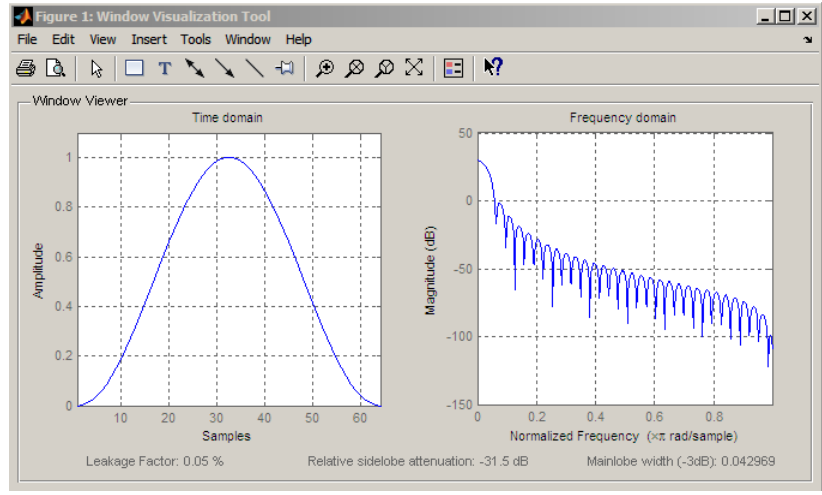
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` symmetric Hann window:

```
H=sigwin.hann;
wvtool(H);
```



Generate length `N=128` periodic Hann window, return values, and write ASCII file:

```
H=sigwin.hann(128,'periodic');
% Return window with generate
win=generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'hann_128')
```

References

Oppenheim, A.V. and Schafer, R.W. *Discrete-time Signal Processing*, Upper Saddle River, N.J: Prentice Hall, 1989, pp. 444–447.

See Also

`sigwin | window | wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

sigwin.hann.generate

Purpose

Generates Hann window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Hann window object `H` as a double-precision column vector.

Examples

Extract values from Hann window object:

```
H=sigwin.hann(128);
% Extract window values as column vector
win=generate(H);
```

Purpose

Display information about Hann window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length and sampling information for the Hann window object `H`.

`info_win = info(H)` returns length and sampling information for the Hann window object `H` in the character array `info_win`.

Examples

Return information about a Hann window object:

```
H=sigwin.hann(256);
info_win=info(H);
```

sigwin.hann.winwrite

Purpose

Save Hann window object values in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of the Hann window object `H` to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Hann window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Hann window values to ASCII file:

```
H=sigwin.hann;
% Open dialog box for ASCII file
winwrite(H);
```

Purpose

Construct Kaiser window object

Description

`sigwin.kaiser` creates a handle to a Kaiser window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines the Kaiser window of length *N*:

$$w(x) = I_0\left(\beta\sqrt{1-\frac{4x^2}{(N-1)^2}}\right) / I_0(\beta) \quad -(N-1)/2 \le x \le (N-1)/2$$

where *x* is linearly spaced *N*-point vector and $I_0()$ is the modified zero-th order Bessel function of the first kind. β is the attenuation parameter.

Construction

`H = sigwin.kaiser` returns a Kaiser window object `H` of length 64 and attenuation parameter *beta* of 0.5.

`H = sigwin.kaiser(`*Length*`)` returns a Kaiser window object `H` of length *Length* and attenuation parameter *beta* of 0.5. *Length* requires a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

`H = sigwin.kaiser(`*Length*`,`*Beta*`)` returns a Kaiser window object with real-valued attenuation parameter *beta*.

Properties

Length

Kaiser window length. The window length requires a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

Beta

Attenuation parameter. `Beta` requires a real number. Larger absolute values of `Beta` result in greater stopband attenuation,

or equivalently greater attenuation between the main lobe and first side lobe.

Methods

`generate`	Generates Kaiser window
`info`	Display information about Kaiser window object
`winwrite`	Save Kaiser window in ASCII file

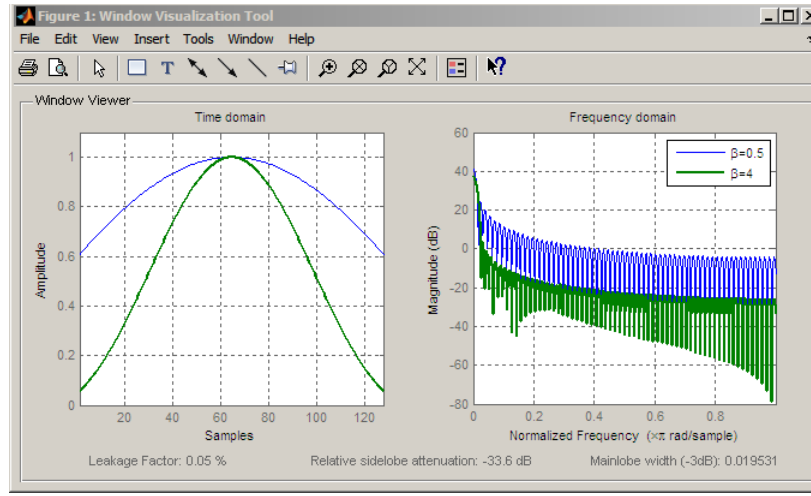
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Compare two Kaiser windows with different `Beta` values:

```
H = sigwin.kaiser(128,1.5);
% Kaiser window with Beta=4.5
H1 = sigwin.kaiser(128,4.5);
% Plot comparison
fwvt = wvtool(H,H1);
legend(get(fwvt,'currentaxes'),'\beta=1.5','\beta=4.5');
```



References

Oppenheim, A.V., and Schafer, R.W. *Discrete-time Signal Processing*, Upper Saddle River, N.J: Prentice Hall, 1989, pp. 444–447.

See Also

`besseli` | `sigwin` | `window` | `wvtool` |

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

sigwin.kaiser.generate

Purpose

Generates Kaiser window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Kaiser window object as a double-precision column vector.

Examples

Extract values from Kaiser window object:

```
H=sigwin.kaiser(128,4);
% Extract window values as column vector
win=generate(H);
```

Purpose

Display information about Kaiser window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length and attenuation information for the Kaiser window object `H`.

`info_win = info(H)` returns length and attenuation information for the Kaiser window object `H` in the character array `info_win`.

Examples

Return information about a Kaiser window object:

```
H=sigwin.kaiser(256);
info_win=info(H);
```

sigwin.kaiser.winwrite

Purpose

Save Kaiser window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the Kaiser window values to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Kaiser window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Kaiser window values to ASCII file:

```
H=sigwin.kaiser;
% Open dialog box for ASCII file
winwrite(H);
```

Purpose

Construct Nuttall defined 4–term Blackman-Harris window object

Description

`sigwin.nuttallwin` creates a handle to a Nuttall defined 4–term Blackman-Harris window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

Construction

`H = sigwin.nuttallwin` returns a Nuttall defined 4–term Blackman-Harris window object window object `H` of length 64.

`H = sigwin.nuttallwin(Length)` returns a Nuttall defined 4–term Blackman-Harris window object `H` of length `Length`. Entering a positive noninteger value for `Length` rounds the length to the nearest integer. Entering a 1 for `Length` results in a window with a single value of 1. The `SamplingFlag` property defaults to `'symmetric'`.

Properties

Length

Nuttall defined 4–term Blackman-Harris window length. The window length must be a positive integer. Entering a positive noninteger value for `Length` rounds the length to the nearest integer. Entering a 1 for `Length` results in a window with a single value of 1.

SamplingFlag

The type of window returned as one of `'symmetric'` or `'periodic'`. The default is `'symmetric'`. A symmetric window exhibits perfect symmetry between halves of the window. Setting the `SamplingFlag` property to `'periodic'` results in a N-periodic window. The equations for the Nuttall defined 4–term Blackman-Harris window differ slightly based on the value of the `SamplingFlag` property. See “Definitions” on page 1-1040 for details.

Methods

`generate`	Generates Nuttall defined 4–term Blackman-Harris window
`info`	Display information about Nuttall defined 4–term Blackman-Harris window object
`winwrite`	Save Nuttall defined 4-term Blackman-Harris window object values in ASCII file

Definitions

The following equation defines the symmetric Nuttall defined 4–term Blackman-Harris window of length *N*.

$$w(n) = a_0 - a_1 \cos(\frac{2\pi n}{N-1}) + a_2 \cos(\frac{4\pi n}{N-1}) - a_3 \cos(\frac{6\pi n}{N-1}) \quad 0 \le n \le N-1$$

The following equation defines the periodic Nuttall defined 4–term Blackman-Harris window of length *N*.

$$w(n) = a_0 - a_1 \cos(\frac{2\pi n}{N}) + a_2 \cos(\frac{4\pi n}{N}) - a_3 \cos(\frac{6\pi n}{N}) \quad 0 \le n \le N-1$$

The following table lists the coefficients:

Coefficient	Value
a0	0.3635819
a1	0.4891775
a2	0.1365995
a3	0.0106411

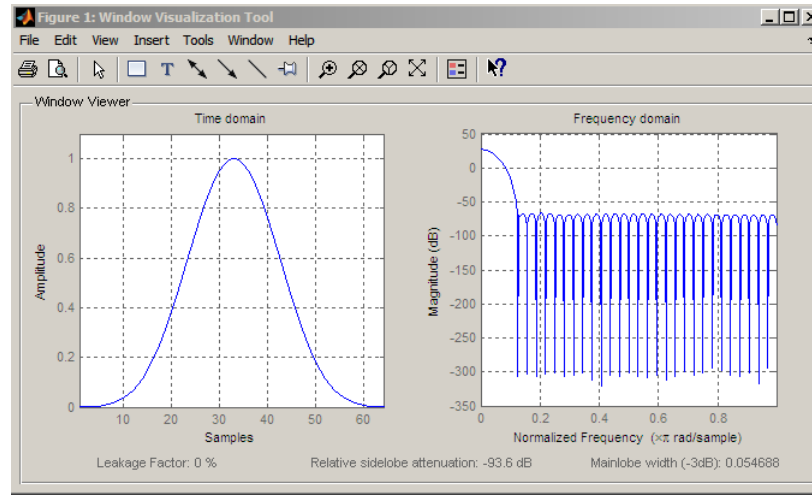
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Construct a length `N=64` symmetric Nuttall defined 4–term Blackman-Harris window:

```
H=sigwin.nuttallwin;
wvtool(H);
```



Generate a length `N=128` periodic Nuttall defined 4–term Blackman-Harris window, return values, and write ASCII file:

```
H=sigwin.nuttallwin(128);
H.SamplingFlag = 'periodic';
% Return window with generate
win=generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'nuttallwin_128')
```

References

Nuttall, A.H. “Some Windows with Very Good Sidelobe Behavior." *IEEE Transactions on Acoustics, Speech, and Signal Processing*. Vol. 29, 1981, pp. 84–91.

See Also

`sigwin` | `window` | `wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

Purpose

Generates Nuttall defined 4–term Blackman-Harris window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Nuttall defined 4–term Blackman-Harris window object as a double-precision column vector.

Examples

Extract values from Nuttall defined 4–term Blackman-Harris window object:

```
H=sigwin.nuttallwin(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.nuttalwin.info

Purpose

Display information about Nuttall defined 4–term Blackman-Harris window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length information about the Nuttall defined 4–term Blackman-Harris window object `H`.

`info_win = info(H)` returns length information about the Nuttall defined 4–term Blackman-Harris window object `H` in the character array `info_win`.

Examples

Return information about Nuttall defined 4–term Blackman-Harris window object:

```
H=sigwin.nuttallwin(256);
info_win=info(H);
```

Purpose

Save Nuttall defined 4-term Blackman-Harris window object values in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of the Nuttall defined 4-term Blackman-Harris window object `H` to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Nuttall defined 4-term Blackman-Harris window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Nuttall defined 4-term Blackman-Harris window values to ASCII file:

```
H=sigwin.nuttallwin;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.parzenwin

Purpose

Construct Parzen window object

Description

`sigwin.parzenwin` creates a handle to a Parzen window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines the *N*–point Parzen window over the interval $-\frac{(N-1)}{2} \le n \le \frac{(N-1)}{2}$:

$$w(n) = \begin{cases} 1 - 6\left(\frac{|n|}{N/2}\right)^2 + 6\left(\frac{|n|}{N/2}\right)^3 & 0 \le |n| \le (N-1)/4 \\ 2\left(1 - \frac{|n|}{N/2}\right)^3 & (N-1)/4 < |n| \le (N-1)/2 \end{cases}$$

Construction

`H = sigwin.parzenwin` returns a Parzen window object `H` of length 64.

`H = sigwin.parzenwin(`*`Length`*`)` returns a Parzen window object `H` of length *`Length`*. *`Length`* requires a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

Properties

Length

`Length` requires a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

Methods

`generate`	Generate Parzen window
`info`	Display information about Parzen window object
`winwrite`	Save Parzen window in ASCII file

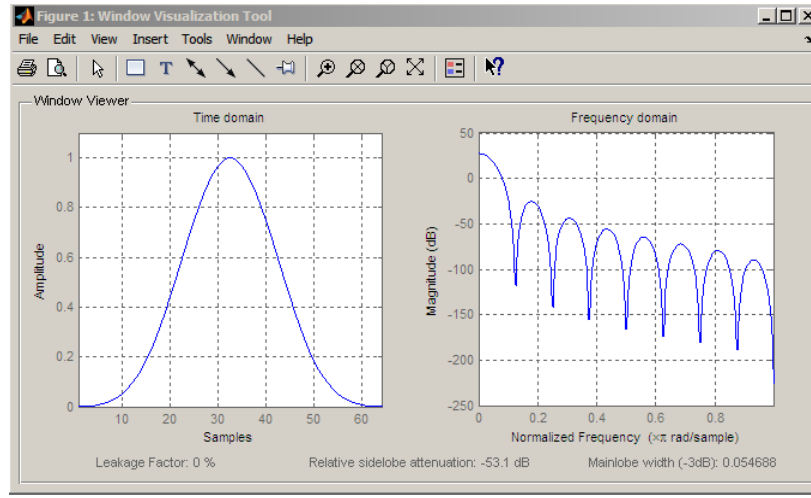
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` Parzen window:

```
H=sigwin.parzenwin;
wvtool(H);
```



Generate length `N=128` Parzen window object, return values, and write ASCII file:

```
H=sigwin.parzenwin(128);
% Return window with generate
win=generate(H);
% Write ascii file in current directory
% with window values
winwrite(H,'parzenwin_128')
```

References

Harris, F.J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform.” *Proceedings of the IEEE*, Vol. 66. 1978, pp. 51–83.

See Also

`sigwin | window | wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

Purpose

Generate Parzen window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Parzen window object as a double-precision column vector.

Examples

Extract values from Parzen window object:

```
H=sigwin.parzenwin(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.parzenwin.info

Purpose

Display information about Parzen window object

Syntax

```
info(H)
info_win=info(H)
```

Description

`info(H)` displays length information about the Parzen window object `H`.

`info_win=info(H)` returns length information about the Parzen window object `H` in the character array `info_win`.

Examples

Return information about a Parzen window object:

```
% 256-point Parzen window
H=sigwin.parzenwin(256);
info_win=info(H);
```

Purpose

Save Parzen window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of the Parzen window object `H` to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Parzen window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Parzen window values to ASCII file:

```
H=sigwin.parzenwin;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.rectwin

Purpose

Construct rectangular window object

Description

`sigwin.rectwin` creates a handle to a rectangular window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines the rectangular window of length *N*:

$$w(n) = 1 \quad 0 \le n \le N - 1$$

Construction

`H = sigwin.rectwin` returns a rectangular window object `H` of length 64.

`H = sigwin.rectwin(Length)` returns a rectangular window object `H` of length *Length*. *Length* requires a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

Properties

Length

Rectangular window length. The window length requires a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

Methods

`generate`	Generates rectangular window
`info`	Display information about rectangular window object
`winwrite`	Save rectangular window in ASCII file

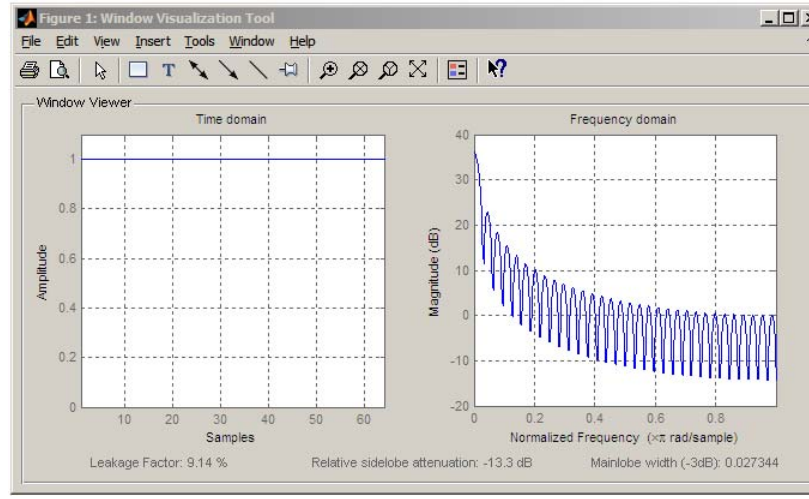
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Create default length `N=64` rectangular window:

```
H=sigwin.rectwin;
wvtool(H);
```



Generate length `N=128` rectangular window, return values, and write ASCII file:

```
H=sigwin.rectwin(128);
% Return window with generate
win=generate(H);
% Write ascii file in current directory
% with window values
winwrite(H,'rectwin_128')
```

References

Oppenheim, A.V., and Schafer, R.W. *Discrete-time Signal Processing*, Upper Saddle River, N.J: Prentice Hall, 1989, pp. 444–447.

See Also

`sigwin` | `window` | `wvtool`

Tutorials

- "Windows"

How To

- Class Attributes
- Property Attributes

Purpose

Generates rectangular window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the rectangular window object `H` as a double-precision column vector.

Examples

Extract values from rectangular window object:

```
H=sigwin.rectwin(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.rectwin.info

Purpose

Display information about rectangular window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length information for the rectangular window object `H`.

`info_win = info(H)` returns length information for the rectangular window object `H` in the character array `info_win`.

Examples

Return information about a rectangular window object:

```
H=sigwin.rectangular(256);
info_win=info(H);
```

Purpose

Save rectangular window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of the rectangular window object `H` to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the rectangular window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write rectangular window values to ASCII file:

```
H=sigwin.rectwin;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.taylorwin

Purpose

Construct Taylor window object

Description

`sigwin.taylorwin` creates a handle to a Taylor window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

Taylor windows are similar to Dolph-Chebyshev windows. The Taylor window approximates the minimization of the main lobe width in the Dolph-Chebyshev window, but allows the sidelobe levels to decrease beyond a certain frequency. Taylor windows are typically used in radar applications, such as weighting synthetic aperature radar images and antenna design.

Construction

`H = sigwin.taylorwin` returns a Taylor window object `H` of length 64, with a maximum sidelobe level of 30 dB and 4 constant-level sidelobes adjacent to the main lobe.

`H = sigwin.taylorwin(`*`Length`*`)` returns a Taylor window object `H` of length *`Length`* with a maximum sidelobe level of 30 dB and 4 constant-level sidelobes adjacent to the main lobe. *`Length`* must be a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

`H = sigwin.taylorwin(`*`Length`*`,`*`Nbar`*`)` returns a Taylor window object with *`Nbar`* nearly constant-level sidelobes adjacent to the main lobe. *`Nbar`* must be a positive integer.

`H = sigwin.taylorwin(`*`Length`*`,`*`Nbar`*`,`*`SidelobeLevel`*`)` returns a Taylor window object with a maximum sidelobe level *`SidelobeLevel`* dB below the main lobe level.

Properties

Length

Taylor window length. The window length must be a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

Nbar

Number of nearly constant-level sidelobes. Must be a positive integer.

SidelobeLevel

Maximum sidelobe level relative to the main lobe peak. The maximum sidelobe level is a nonnegative number which gives side lobes `SidelobeLevel` dB down from the main lobe peak.

Methods

`generate`	Generates Taylor window
`info`	Display information about Taylor window object
`winwrite`	Save Taylor window object values in ASCII file

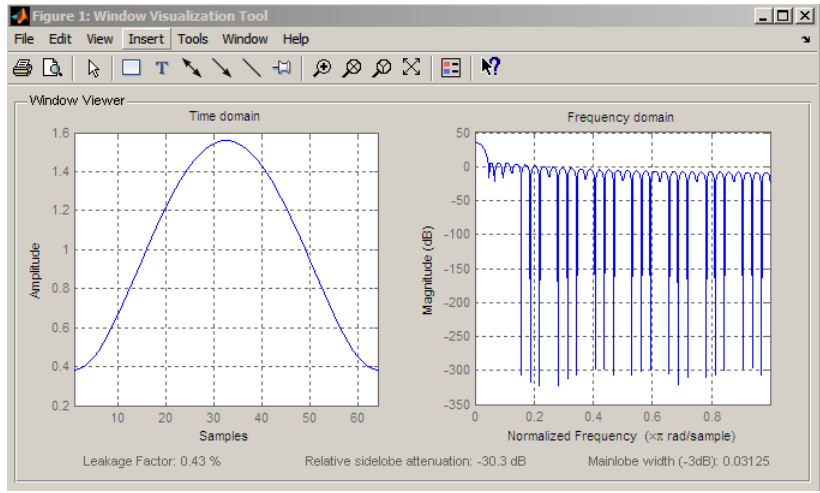
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `N=64` Taylor window:

```
H=sigwin.taylorwin;
wvtool(H);
```



Generate length `N=128` Taylor window, return values, and write ASCII file with window values:

```
H=sigwin.taylorwin(128);
% Return window with generate
win=generate(H);
% Write ASCII file in current directory
% with window values
winwrite(H,'taylorwin_128')
```

References

Carrara, W.G., R.M. Majewski and R.S. Goodman. *Spotlight Synthetic Aperature Radar: Signal Processing Algorithms*, Artech House Publishers, Boston, 1995, Appendix D.2.

See Also

`sigwin | window | wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
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Purpose

Generates Taylor window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Taylor window object `H` as a double-precision column vector.

Examples

Extract values from Taylor window object:

```
H=sigwin.taylorwin(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.taylorwin.info

Purpose

Display information about Taylor window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length and sidelobe information for the Taylor window object `H`.

`info_win = info(H)` returns length and sidelobe information for the Taylor window object `H` in the character array `info_win`.

Examples

Return information about a Taylor window object:

```
H=sigwin.taylorwin(256);
info_win=info(H);
```

Purpose

Save Taylor window object values in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of the Taylor window object `H` to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Taylor window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Taylor window values to ASCII file:

```
H=sigwin.taylorwin;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.triang

Purpose

Construct triangular window object

Description

`sigwin.triang` is a triangular window object.

`sigwin.triang` creates a handle to a triangular window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

For L odd, the triangular window is defined as:

$$w(n) = \begin{cases} \dfrac{2n}{L+1} & 1 \le n \le (L+1)/2 \\ 2 - \dfrac{2n}{L+1} & (L+1)/2 + 1 \le n \le L \end{cases}$$

For L even, the triangular window is defined as:

$$w(n) = \begin{cases} \dfrac{(2n-1)}{L} & 1 \le n \le L/2 \\ 2 - \dfrac{(2n-1)}{L} & L/2 + 1 \le n \le L \end{cases}$$

Construction

`H = sigwin.triang` returns a triangular window object `H` of length 64.

`H = sigwin.triang(`*`Length`*`)` returns a triangular window object `H` of length *`Length`*. Entering a positive non-integer value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

Properties

Length

Triangular window length. The window length requires a positive integer. Entering a positive non-integer value for *`Length`* rounds the length to the nearest integer. Entering a 1 for *`Length`* results in a window with a single value of 1.

Methods

`generate`	Generates triangular window
`info`	Display information about triangular window
`winwrite`	Save triangular window in ASCII file

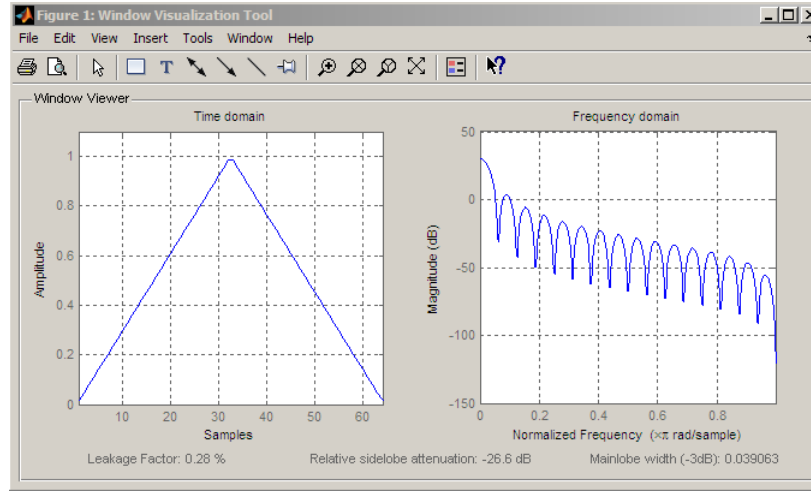
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length `L = 64` triangular window:

```
H=sigwin.triang;
wvtool(H);
```



Generate length `L = 128` triangular window, return values, and write ASCII file:

```
H=sigwin.triang(128);
```

```
% Return window with generate
win=generate(H);
% Write ascii file in current directory
% with window values
winwrite(H,'triang_128')
```

References

Oppenheim, A.V., and Schafer, R.W. *Discrete-time Signal Processing*, Upper Saddle River, N.J: Prentice Hall, 1989, pp. 444–447.

See Also

`sigwin` | `window` | `wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

Purpose

Generates triangular window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the triangular window object `H` as a double-precision column vector.

Examples

Extract values from triangular window object:

```
H=sigwin.triang(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.triang.info

Purpose

Display information about triangular window

Syntax

```
info(H)
info_array = info(H)
```

Description

`info(H)` displays length information for the triangular window object `H`.

`info_array = info(H)` returns length information for the triangular window object `H` in the character array `info_array`.

Examples

Return information about a triangular window object:

```
H=sigwin.triangular(256);
info_win=info(H);
```

Purpose

Save triangular window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of the triangular window object `H` to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the triangular window object `H` as a column vector in the ASCII file `'filename'` in the current folder. The file extension `.wf` is automatically appended to `filename`.

Examples

Write triangular window values to ASCII file:

```
H=sigwin.triang;
% Open dialog box for ASCII file
winwrite(H);
```

sigwin.tukeywin

Purpose

Construct Tukey window object

Description

`sigwin.tukeywin` creates a handle to a Tukey window object for use in spectral analysis and FIR filtering by the window method. Object methods enable workspace import and ASCII file export of the window values.

The following equation defines the *N*–point Tukey window:

$$w(x) = \begin{cases} \frac{1}{2}\{1+\cos(\frac{2\pi}{\alpha}[x-\alpha/2])\} & 0 \le x < \frac{\alpha}{2} \\ 1 & \frac{\alpha}{2} \le x < 1-\frac{\alpha}{2} \\ \frac{1}{2}\{1+\cos(\frac{2\pi}{\alpha}[x-1+\alpha/2])\} & 1-\frac{\alpha}{2} \le x \le 1 \end{cases}$$

where x is a N–point linearly spaced vector generated using `linspace`. The parameter α is the ratio of cosine-tapered section length to the entire window length with $0 \le \alpha \le 1$. For example, setting $\alpha=0.5$ produces a Tukey window where 1/2 of the entire window length consists of segments of a phase-shifted cosine with period $2\alpha=1$. If you specify $\alpha \le 0$, an N-point rectangular window is returned. If you specify $\alpha \ge 1$, a von Hann window (`sigwin.hann`) is returned.

Construction

`H = sigwin.tukeywin` returns a Tukey or cosine-tapered window object `H` of length 64 with *`Alpha`* parameter equal to 0.5.

`H = sigwin.tukeywin(`*`Length`*`)` returns a Tukey window object `H` of length *`Length`* with *`Alpha`* parameter equal to 0.5. *`Length`* requires a positive integer. Entering a positive noninteger value for *`Length`* rounds the length to the nearest integer.

`H = sigwin.tukeywin(`*`Length`*`,`*`Alpha`*`)` returns a Tukey window object with the ratio of the tapered section length to the entire window length *`Alpha`*. *`Alpha`* defaults to 0.5. As *`Alpha`* approaches zero, the Tukey window approaches a rectangular window. As *`Alpha`* approaches one, the Tukey window approaches a Hann window.

Properties

Length

Tukey window length. The window length must be a positive integer. Entering a positive noninteger value for *Length* rounds the length to the nearest integer. Entering a 1 for *Length* results in a window with a single value of 1.

Alpha

The ratio of tapered window section to constant section. As a ratio, Alpha satisfies the inequality $0 \le \alpha \le 1$. As *Alpha* approaches zero, the Tukey window approaches a rectangular window. As *Alpha* approaches one, the Tukey window approaches a Hann window. Specifying *Alpha* less than zero or greater than one replaces *Alpha* with 0 and 1 respectively.

Methods

generate	Generates Tukey window
info	Display information about Tukey window object
winwrite	Save Tukey window in ASCII file

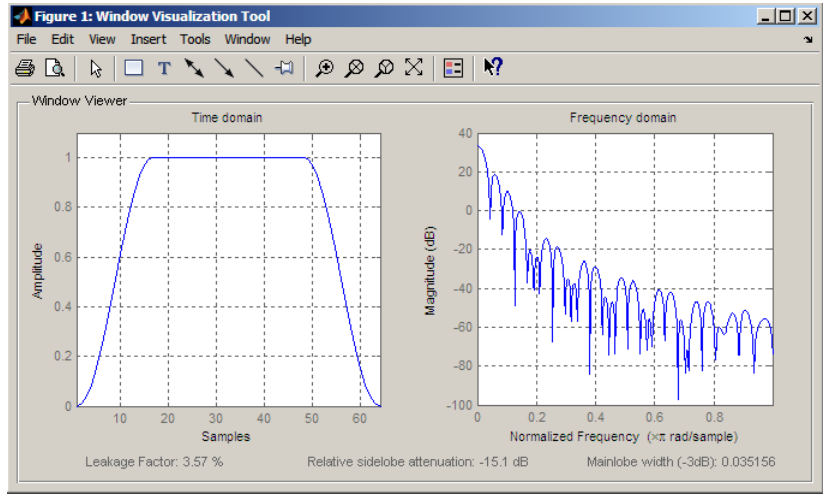
Copy Semantics

Handle. To learn how copy semantics affect your use of the class, see Copying Objects in the MATLAB Programming Fundamentals documentation.

Examples

Default length N=64 Tukey window:

```
H=sigwin.tukeywin;
wvtool(H);
```



Generate length `N=128` Tukey window, return values, and write ASCII file:

```
H=sigwin.tukeywin(128,1/4);
% Return window with generate
win=generate(H);
% Write ascii file in current directory
% with window values
winwrite(H,'tukeywin_128')
```

References

[1] Bloomfield P. *Fourier Analysis of Time Series: An Introduction*, New York: Wiley-Interscience, 2000, p.69.

See Also

`sigwin | window | wvtool`

Tutorials

- “Windows”

How To

- Class Attributes
- Property Attributes

Purpose

Generates Tukey window

Syntax

```
win = generate(H)
```

Description

`win = generate(H)` returns the values of the Tukey window object `H` as a double-precision column vector.

Examples

Extract values from Tukey window object:

```
H=sigwin.tukeywin(128);
% Extract window values as column vector
win=generate(H);
```

sigwin.tukeywin.info

Purpose

Display information about Tukey window object

Syntax

```
info(H)
info_win = info(H)
```

Description

`info(H)` displays length and tapered–to–constant section ratio information for the Tukey window object `H`.

`info_win = info(H)` returns length and tapered–to–constant section ratio information for the Tukey window object `H` in the character array `info_win`.

Examples

Return information about a Tukey window object:

```
H=sigwin.tukey(256);
info_win=info(H);
```

Purpose

Save Tukey window in ASCII file

Syntax

```
winwrite(H)
winwrite(H,'filename')
```

Description

`winwrite(H)` opens a dialog to export the values of the Tukey window object to an ASCII file. The file extension `.wf` is automatically appended.

`winwrite(H,'filename')` saves the values of the Tukey window object `H` in the current folder as a column vector in the ASCII file `'filename'`. The file extension `.wf` is automatically appended to `filename`.

Examples

Write Tukey window values to ASCII file:

```
H=sigwin.tukeywin;
% Open dialog box for ASCII file
winwrite(H);
```

Purpose

Sinc vector or matrix

Syntax

```
y = sinc(x)
```

Description

`sinc` computes the sinc function of an input vector or array, where the sinc function is

$$\mathrm{sinc}(t) = \begin{cases} 1, & t = 0 \\ \dfrac{\sin(\pi t)}{\pi t} & t \neq 0 \end{cases}$$

This function is the continuous inverse Fourier transform of the rectangular pulse of width 2π and height 1.

$$\mathrm{sinc}(t) = \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{j\omega t} d\omega$$

`y = sinc(x)` returns an array `y` the same size as `x`, whose elements are the `sinc` function of the elements of `x`.

The space of functions bandlimited in the frequency range $\omega \in [-\pi, \pi]$ is spanned by the infinite (yet countable) set of sinc functions shifted by integers. Thus any such bandlimited function $g(t)$ can be reconstructed from its samples at integer spacings.

$$g(t) = \sum_{n=-\infty}^{\infty} g(n)\mathrm{sinc}(t - n)$$

Examples

Perform ideal bandlimited interpolation by assuming that the signal to be interpolated is 0 outside of the given time interval and that it has been sampled at exactly the Nyquist frequency:

```
t = (1:10)';                % Column vector of time samples
x = randn(size(t));         % Column vector of data
ts = linspace(-5,15,600)';  % Times at which to interpolate
```

```
y = sinc(ts(:,ones(size(t))) - t(:,ones(size(ts)))')*x;
plot(t,x,'o',ts,y)
```

See Also

chirp | cos | diric | gauspuls | pulstran | rectpuls | sawtooth | sin | square | tripuls

slewrate

Purpose

Slew rate of bilevel waveform

Syntax

```
S = slewrate(X)
S = slewrate(X,Fs)
S = slewrate(X,T)
[S,LT,UT] = slewrate(...)
[S,LT,UT,LL,UL] = slewrate(...)
S = slewrate(...,Name,Value)
slewrate(...)
```

Description

`S = slewrate(X)` returns the slew rate for all transitions found in the bilevel waveform, `X`. The slew rate is the slope of the line connecting the 10% and 90% reference levels. The sample instants of `X` are the indices of the vector. To determine the transitions, `slewrate` estimates the state levels of the input waveform by a histogram method. `slewrate` identifies all regions that cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-1081.

`S = slewrate(X,Fs)` specifies the sample rate, `Fs`, in hertz. The first time instant in `X` corresponds to t=0.

`S = slewrate(X,T)` specifies the sample instants in the vector, `T`. The length of `T` must equal the length of `X`.

`[S,LT,UT] = slewrate(...)` returns the time instants when the waveform crosses the lower-percent reference level, `LT`, and upper-percent reference level, `UT`. If you do not specify lower- and upper-percent reference levels, the levels default to 10% and 90%.

`[S,LT,UT,LL,UL] = slewrate(...)` returns the waveform values that correspond to the lower-reference levels, `LL`, and upper-reference levels, `UL`.

`S = slewrate(...,Name,Value)` returns the slew rate for all transitions with additional options specified by one or more `Name,Value` pair arguments.

`slewrate(...)` plots the bilevel waveform and darkens the regions of each transition where the slew rate is computed. The plot marks the lower- and upper-reference level crossings and associated reference levels. The plot indicates the state levels and associated lower and upper tolerances.

Input Arguments

X

Bilevel waveform as a real-valued column or row vector. If the input waveform does not have at least one transition, `slewrate` returns an empty matrix.

Fs

Sampling rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

'PctRefLevels'

Percent reference levels. See "Percent Reference Levels" on page 1-1080 for a definition.

Default: `[10,90]`

'StateLevels'

Low- and high-state levels. `StateLevels` is a 1-by-2 real-valued vector. The first element is the low-state level. The second element is the high-state level. If you do not specify low- and high-state levels, `slewrate` estimates the state levels from the input waveform using the histogram method.

'Tolerance'

Tolerance levels (lower and upper state boundaries) expressed as a percentage. See “State-Level Tolerances” on page 1-1081.

Default: `2`

Output Arguments

S

Slew rates as real-valued scalars. A positive slew rate indicates that the upper-percent reference level occurs later than the lower-percent reference level. A negative slew rate indicates that the upper-percent reference level occurs before the lower-percent reference level.

LT

Time instants when signal crosses the lower percent reference level. If you do not specify the lower percent reference levels with the `'PctRefLevels'` name-value pair, the lower percent reference level is 10%.

UT

Time instants when signal crosses the upper-percent reference level. If you do not specify the upper-percent reference levels with the `'PctRefLevels'` name-value pair, the upper-percent reference level is 90%.

LL

Waveform values at the lower-reference level.

UL

Waveform values at the upper-reference level.

Definitions

Percent Reference Levels

If S_1 is the low state, S_2 is the high state, and U is the *upper*-percent reference level. The waveform value corresponding to the upper-percent reference level is

$$S_1 + \frac{U}{100}(S_2 - S_1)$$

If L is the *lower*-percent reference level, the waveform value corresponding to the lower percent reference level is

$$S_1 + \frac{L}{100}(S_2 - S_1)$$

Slew Rate

The slew rate is the slope of a line connecting the upper- and lower-percent reference levels. Let t_L denote the time instant when the waveform crosses the lower reference level and t_U denote the time instant when the waveform crosses the upper percent reference level. Using the definitions for the upper and lower percent reference levels given in "Percent Reference Levels" on page 1-1080, the slew rate is

$$\frac{S_1 + \frac{U}{100}(S_2 - S_1) - \{S_1 + \frac{L}{100}(S_2 - S_1)\}}{t_U - t_L}$$

$$\frac{\frac{U-L}{100}(S_2 - S_1)}{t_U - t_L}$$

When t_L occurs earlier than t_U, the slew rate is positive. When t_U occurs earlier than t_L, the slew rate is negative.

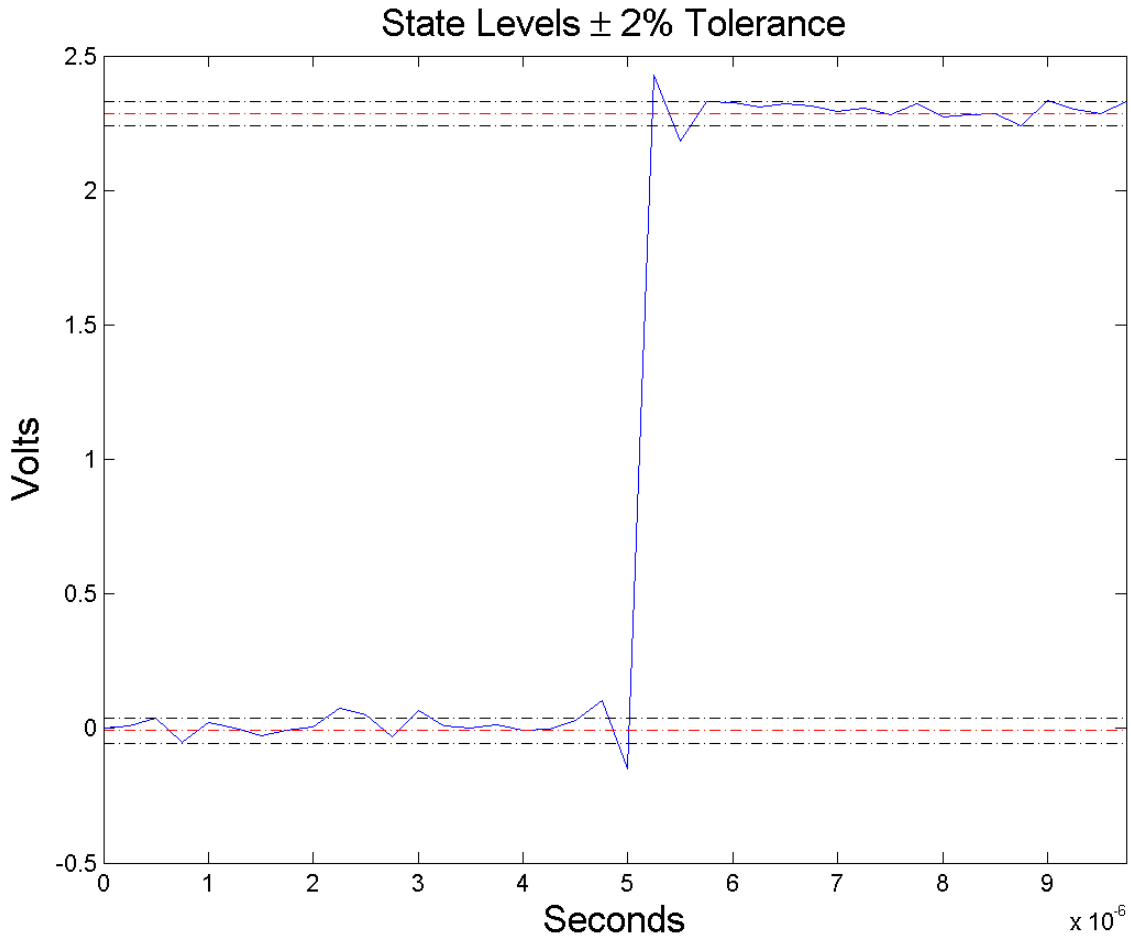
State-Level Tolerances

Each state level can have associated lower- and upper-state boundaries. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The red dashed lines indicate the estimated state levels.



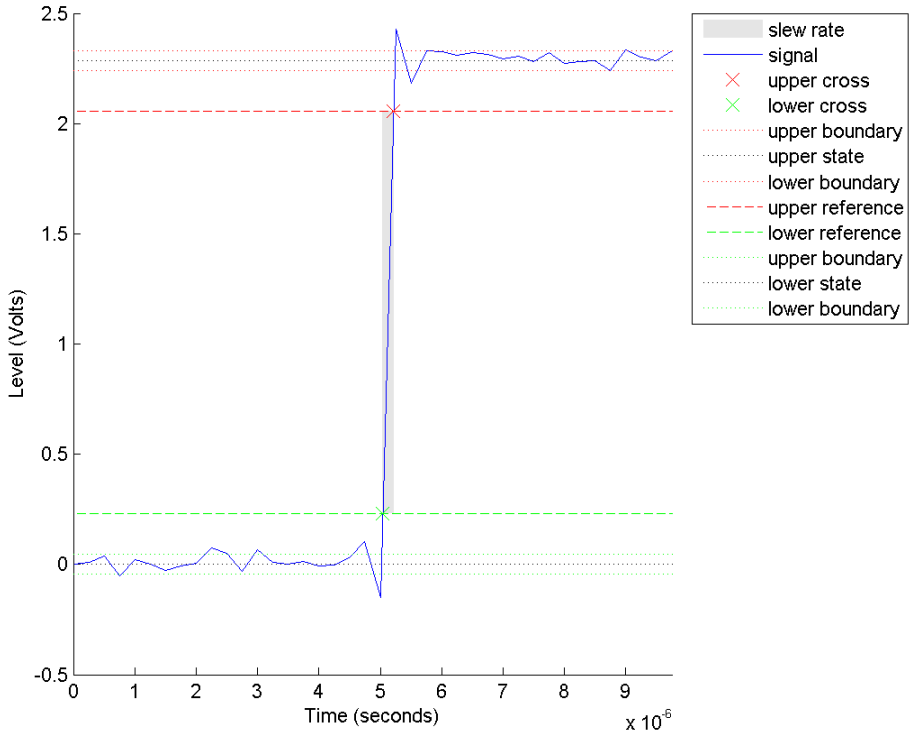
Examples

Slew Rate For One-Transition Waveform

Use `slewrate` with no output arguments to plot the slew rate information for a step waveform sampled at 4 MHz.

Load the `transitionex.mat` file and compute the slew rate.

```
load('transitionex.mat', 'x', 't');
slewrate(x, t)
```



Slew Rates for Three-Transition Waveform

Use the waveform in "Slew Rate For One-Transition Waveform" on page 1-1083 to create a three-transition (two positive and one negative) bilevel waveform. Obtain the slew rates for the three transitions.

```
load('transitionex.mat', 'x');
```

```
y = [x ; fliplr(x)];
t = 0:1/4e6:(length(y)*(1/4e6))-1/4e6;
S = slewrate(y, t);
```

Lower and Upper Transition Times

Return the lower- and upper-transition times for the three-transition waveform in"Slew Rates for Three-Transition Waveform" on page 1-1084 .

```
load('transitionex.mat', 'x');
y = [x ; fliplr(x)];
t = 0:1/4e6:(length(y)*(1/4e6))-1/4e6;
[S,LT,UT] = slewrate(y, t);
% or [S,LT,UT] = slewrate(y,4e6);
```

Lower and Upper Reference Levels

Return the waveform values corresponding to the lower- and upper-reference levels for the three-transition waveform in"Slew Rates for Three-Transition Waveform" on page 1-1084 . Compute these values for the default 10% and 90% and for 20% and 80%.

```
load('transitionex.mat', 'x');
y = [x ; fliplr(x)];
t = 0:1/4e6:(length(y)*(1/4e6))-1/4e6;
[~,LT_1090,UT_1090,LL_1090,UL_1090] = slewrate(y, t);
[~,LT_2080,UT_2080,LL_2080,UL_2080] = slewrate(y, t,...
    'PctRefLevels',[20 80]);
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003.

See Also

`falltime` | `midcross` | `pulsewidth` | `risetime` | `settlingtime` | `statelevels`

sos2cell

Purpose

Convert second-order sections matrix to cell array

Syntax

```
c = sos2cell(m)
c = sos2cell(m,g)
```

Description

`c = sos2cell(m)` changes an L-by-6 second-order section matrix m generated by `tf2sos` into a 1-by-L cell array of 1-by-2 cell arrays `c`. You can use `c` to specify a quantized filter with L cascaded second-order sections.

The matrix `m` should have the form

```
m = [b1 a1;b2 a2; ... ;bL aL]
```

where both `bi` and `ai`, with $i = 1, ..., L$, are 1-by-3 row vectors. The resulting `c` is a 1-by-L cell array of cells of the form

```
c = { {b1 a1} {b2 a2} ... {bL aL} }
```

`c = sos2cell(m,g)` with the optional gain term `g`, prepends the constant value `g` to `c`. When you use the added gain term in the command, `c` is a 1-by-L cell array of cells of the form

```
c = {{g,1} {b1,a1} {b2,a2}...{bL,aL} }
```

Examples

Use `sos2cell` to convert the 2-by-6 second-order section matrix produced by `tf2sos` into a 1-by-2 cell array `c` of cells. Display the second entry in the first cell in `c`:

```
[b,a] = ellip(4,0.5,20,0.6);
m = tf2sos(b,a);
c = sos2cell(m);
c{1}{2}
ans =
    1.0000    0.1677    0.2575
```

See Also

`tf2sos` | `cell2sos`

Purpose

Convert digital filter second-order section parameters to state-space form

Syntax

```
[A,B,C,D] = sos2ss(sos)
[A,B,C,D] = sos2ss(sos,g)
```

Description

`sos2ss` converts a second-order section representation of a given digital filter to an equivalent state-space representation.

`[A,B,C,D] = sos2ss(sos)` converts the system `sos`, in second-order section form, to a single-input, single-output state-space representation.

$$\begin{aligned} x[n+1] &= Ax[n] + Bu[n] \\ y[n] &= Cx[n] + Du[n] \end{aligned}$$

The discrete transfer function in second-order section form is given by

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

`sos` is a L-by-6 matrix organized as

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

The entries of `sos` must be real for proper conversion to state space. The returned matrix `A` is size N-by-N, where $N = L -$ `B` is a length $N-1$ column vector, `C` is a length $N-1$ row vector, and `D` is a scalar.

`[A,B,C,D] = sos2ss(sos,g)` converts the system `sos` in second-order section form with gain `g`.

$$H(z) = g\prod_{k=1}^{L} H_k(z)$$

Examples

Compute the state-space representation of a simple second-order section system with a gain of 2:

```
sos = [1  1  1  1  0 -1; -2  3  1  1 10  1];
[A,B,C,D] = sos2ss(sos)
A =
   -10     0    10     1
     1     0     0     0
     0     1     0     0
     0     0     1     0
B =
     1
     0
     0
     0
C =
    21     2   -16    -1
D =
    -2
```

Algorithms

`sos2ss` first converts from second-order sections to transfer function using `sos2tf`, and then from transfer function to state-space using `tf2ss`.

See Also

`sos2tf` | `sos2zp` | `ss2sos` | `tf2ss` | `zp2ss`

Purpose

Convert digital filter second-order section data to transfer function form

Syntax

```
[b,a] = sos2tf(sos)
[b,a] = sos2tf(sos,g)
```

Description

`sos2tf` converts a second-order section representation of a given digital filter to an equivalent transfer function representation.

`[b,a] = sos2tf(sos)` returns the numerator coefficients `b` and denominator coefficients `a` of the transfer function that describes a discrete-time system given by `sos` in second-order section form. The second-order section format of $H(z)$ is given by

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

`sos` is an L-by-6 matrix that contains the coefficients of each second-order section stored in its rows.

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

Row vectors `b` and `a` contain the numerator and denominator coefficients of $H(z)$ stored in descending powers of z.

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2z^{-1} + \cdots + b_{n+1}z^{-n}}{a_1 + a_2z^{-1} + \cdots + a_{m+1}z^{-m}}$$

`[b,a] = sos2tf(sos,g)` returns the transfer function that describes a discrete-time system given by `sos` in second-order section form with gain `g`.

$$H(z) = g\prod_{k=1}^{L} H_k(z)$$

Examples

Compute the transfer function representation of a simple second-order section system:

```
sos = [1  1  1  1  0 -1; -2  3  1  1 10  1];
[b,a] = sos2tf(sos)
b =
   -2     1     2     4     1
a =
    1    10     0   -10    -1
```

Algorithms

`sos2tf` uses the `conv` function to multiply all of the numerator and denominator second-order polynomials together. For higher order filters (possibly starting as low as order 8), numerical problems due to roundoff errors may occur when forming the transfer function.

See Also

`latc2tf` | `sos2ss` | `sos2zp` | `ss2tf` | `tf2sos` | `zp2tf`

Purpose

Convert digital filter second-order section parameters to zero-pole-gain form

Syntax

```
[z,p,k] = sos2zp(sos)
[z,p,k] = sos2zp(sos,g)
```

Description

`sos2zp` converts a second-order section representation of a given digital filter to an equivalent zero-pole-gain representation.

`[z,p,k] = sos2zp(sos)` returns the zeros `z`, poles `p`, and gain `k` of the system given by `sos` in second-order section form. The second-order section format of $H(z)$ is given by

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

`sos` is an L-by-6 matrix that contains the coefficients of each second-order section in its rows.

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

Column vectors `z` and `p` contain the zeros and poles of the transfer function $H(z)$.

$$H(z) = k\frac{(z - z_1)(z - z_2)\cdots(z - z_n)}{(p - p_1)(p - p_2)\cdots(p - p_m)}$$

where the orders n and m are determined by the matrix `sos`.

`[z,p,k] = sos2zp(sos,g)` returns the zeros `z`, poles `p`, and gain `k` of the system given by `sos` in second-order section form with gain `g`.

$$H(z) = g\prod_{k=1}^{L} H_k(z)$$

Examples

Compute the poles, zeros, and gain of a simple system in second-order section form:

```
sos = [1  1  1  1  0 -1; -2  3  1  1 10  1];
[z,p,k] = sos2zp(sos)
z =
  -0.5000 + 0.8660i
  -0.5000 - 0.8660i
   1.7808
  -0.2808
p =
   -1.0000
    1.0000
   -9.8990
   -0.1010
k =
    -2
```

Algorithms

`sos2zp` finds the poles and zeros of each second-order section by repeatedly calling `tf2zp`.

See Also

`sos2ss` | `sos2tf` | `ss2zp` | `tf2zp` | `tf2zpk` | `zp2sos`

Purpose

Second-order (biquadratic) IIR digital filtering

Syntax

```
y = sosfilt(sos,x)
y = sosfilt(sos,x,dim)
```

Description

`y = sosfilt(sos,x)` applies the second-order section digital filter `sos` to the vector `x`. The output, `y`, is the same length as `x`.

Note If either input to `sosfilt` is single precision, filtering is implemented using single-precision arithmetic. The output, `y`, is single precision.

`sos` represents the second-order section digital filter $H(z)$

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

by an L-by-6 matrix containing the coefficients of each second-order section in its rows.

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

If `x` is a matrix, `sosfilt` applies the filter to each column of `x` independently. The output `y` is a matrix of the same size, containing the filtered data corresponding to each column of `x`.

If `x` is a multidimensional array, `sosfilt` filters along the first nonsingleton dimension. The output `y` is a multidimensional array of the same size as `x`, containing the filtered data corresponding to each row and column of `x`.

The second order sections matrix,`sos`, the input signal,`x`, or both can be double or single precision. If at least one input is single precision, filtering is done with single precision arithmetic.

`y = sosfilt(sos,x,dim)` operates along the dimension `dim`.

References

[1] Orfanidis, S.J., *Introduction to Signal Processing*, Prentice-Hall, Englewood Cliffs, NJ, 1996.

See Also

`filter` | `medfilt1` | `sgolayfilt`

Purpose

Spectrogram using short-time Fourier transform

Syntax

```
S = spectrogram(x)
S = spectrogram(x,window)
S = spectrogram(x,window,noverlap)
S = spectrogram(x,window,noverlap,nfft)
S = spectrogram(x,window,noverlap,nfft,fs)
[S,F,T] = spectrogram(...)
[S,F,T] = spectrogram(x,window,noverlap,F)
[S,F,T] = spectrogram(x,window,noverlap,F,fs)
[S,F,T,P] = spectrogram(...)
spectrogram(...,FREQLOCATION)
spectrogram(...)
```

Description

`spectrogram`, when used without any outputs, plots a spectrogram or, when used with an `S` output, returns the short-time Fourier transform of the input signal. To create a spectrogram from the returned short-time Fourier transform data, refer to the `[S,F,T,P]` syntax described below.

`S = spectrogram(x)` returns `S`, the short time Fourier transform of the input signal vector `x`. By default, `x` is divided into eight segments. If `x` cannot be divided exactly into eight segments, it is truncated. These default values are used.

- `window` is a Hamming window of length `nfft`.
- `noverlap` is the number of samples that each segment overlaps. The default value is the number producing 50% overlap between segments.
- `nfft` is the FFT length and is the maximum of 256 or the next power of 2 greater than the length of each segment of `x`. Instead of `nfft`, you can specify a vector of frequencies, `F`. See below for more information.
- `fs` is the sampling frequency, which defaults to normalized frequency.

Each column of `S` contains an estimate of the short-term, time-localized frequency content of `x`. Time increases across the columns of `S` and frequency increases down the rows.

If x is a length Nx complex signal, S is a complex matrix with nfft rows and k columns, where for a scalar window

```
k = fix((Nx-noverlap)/(window-noverlap))
```

or if window is a vector

```
k = fix((Nx-noverlap)/(length(window)-noverlap))
```

For real x, the output S has (nfft/2+1) rows if nfft is even, and (nfft+1)/2 rows if nfft is odd.

`S = spectrogram(x,window)` uses the window specified. If window is an integer, x is divided into segments equal to that integer value and a Hamming window is used. If window is a vector, x is divided into segments equal to the length of window and then the segments are windowed using the window functions specified in the window vector. For a list of available windows see "Windows".

Note To obtain the same results for the removed specgram function, specify a 'Hann' window of length 256.

`S = spectrogram(x,window,noverlap)` overlaps noverlap samples of each segment. noverlap must be an integer smaller than window or if window is a vector, smaller than the length of window.

`S = spectrogram(x,window,noverlap,nfft)` uses the nfft number of sampling points to calculate the discrete Fourier transform. nfft must be a scalar.

`S = spectrogram(x,window,noverlap,nfft,fs)` uses fs sampling frequency in Hz. If fs is specified as empty [], it defaults to 1 Hz.

`[S,F,T] = spectrogram(...)` returns a vector of frequencies, F, and a vector of times, T, at which the spectrogram is computed. F has length equal to the number of rows of S. T has length k (defined above) and the values in T correspond to the center of each segment.

`[S,F,T] = spectrogram(x,window,noverlap,F)` uses a vector `F` of frequencies in Hz. `F` must be a vector with at least two elements. This case computes the spectrogram at the frequencies in `F` using the Goertzel algorithm. The specified frequencies are rounded to the nearest DFT bin commensurate with the signal's resolution. In all other syntax cases where `nfft` or a default for `nfft` is used, the short-time Fourier transform is used. The `F` vector returned is a vector of the rounded frequencies. `T` is a vector of times at which the spectrogram is computed. The length of `F` is equal to the number of rows of `S`. The length of `T` is equal to `k`, as defined above and each value corresponds to the center of each segment.

`[S,F,T] = spectrogram(x,window,noverlap,F,fs)` uses a vector `F` of frequencies in Hz as above and uses the `fs` sampling frequency in Hz. If `fs` is specified as empty `[]`, it defaults to 1 Hz.

`[S,F,T,P] = spectrogram(...)` returns a matrix `P` containing the power spectral density (PSD) of each segment. For real `x`, `P` contains the one-sided modified periodogram estimate of the PSD of each segment. For complex `x` and when you specify a vector of frequencies `F`, `P` contains the two-sided PSD.

`spectrogram(...,FREQLOCATION)` specifies which axis to use as the frequency axis in displaying the spectrogram. Specify `FREQLOCATION` as a trailing string argument. Valid options are `'xaxis'` or `'yaxis'`. The strings are not case sensitive. If you do not specify `FREQLOCATION`, `spectrogram` uses the x-axis as the frequency axis by default.

The elements of the PSD matrix `P` are given by $P(i,j) = k\,|\,S(i,j)\,|^2$ where k is a real-valued scalar defined as follows

- For the one-sided PSD,

$$k = \frac{2}{Fs\sum_{n=1}^{L} |\,w(n)\,|\,2}$$

where $w(n)$ denotes the window function (Hamming by default) and Fs is the sampling frequency. At zero and the Nyquist frequencies, the factor of 2 in the numerator is replaced by 1.

- For the two-sided PSD,

$$k = \frac{1}{Fs\sum_{n=1}^{L} |w(n)|^2}$$

at all frequencies.

- If the sampling frequency is not specified, Fs is replaced in the denominator by 2π.

`spectrogram(...)` plots the PSD estimate for each segment on a surface in a figure window. The plot is created using

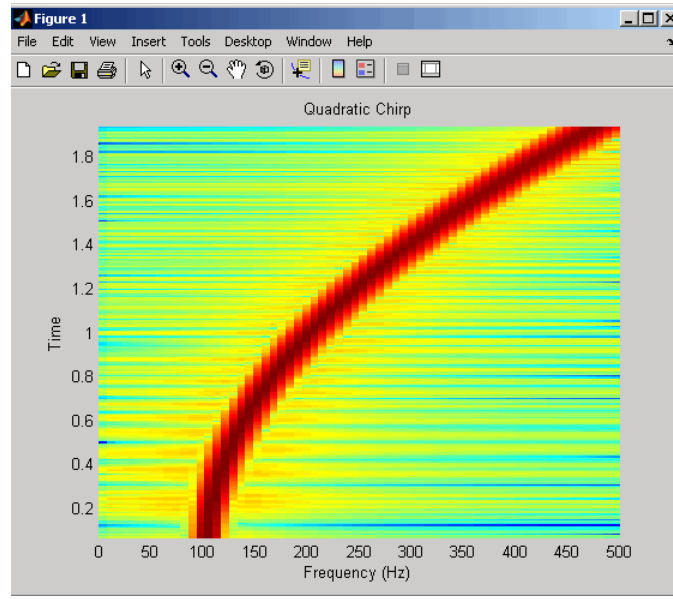
```
surf(T,F,10*log10(abs(P)));
axis tight;
view(0,90);
```

Using `spectrogram(...,'freqloc')` syntax and adding a `'freqloc'` string (either `'xaxis'` or `'yaxis'`) controls where the frequency axis is displayed. Using `'xaxis'` displays the frequency on the *x*-axis. Using `'yaxis'` displays frequency on the *y*-axis and time on the *x*-axis. The default is `'xaxis'`. If you specify both a `'freqloc'` string and output arguments, `'freqloc'` is ignored.

Examples

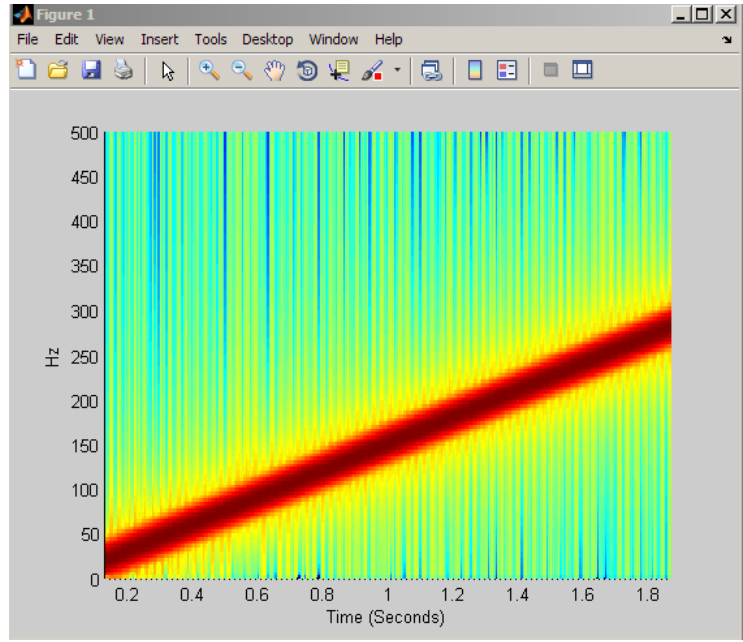
Compute and display the PSD of each segment of a quadratic chirp, which starts at 100 Hz and crosses 200 Hz at `t` = 1 sec.

```
T = 0:0.001:2;
X = chirp(T,100,1,200,'q');
spectrogram(X,128,120,128,1E3);
title('Quadratic Chirp');
```



Compute and display the PSD of each segment of a linear chirp, which starts at DC and crosses 150 Hz at `t` = 1 sec.

```
T = 0:0.001:2;
X = chirp(T,0,1,150);
[S,F,T,P] = spectrogram(X,256,250,256,1E3);
surf(T,F,10*log10(P),'edgecolor','none'); axis tight;
view(0,90);
xlabel('Time (Seconds)'); ylabel('Hz');
```



References

[1] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, Englewood Cliffs, NJ, 1989, pp. 713-718.

[2] Rabiner, L.R., and R.W. Schafer, *Digital Processing of Speech Signals*, Prentice-Hall, Englewood Cliffs, NJ, 1978.

See Also

`goertzel` | `periodogram` | `pwelch` | `spectrum.periodogram` | `spectrum.welch`

How To

- “Windows”

Purpose

Spectral estimation

Syntax

```
Hs = spectrum.estmethod(input1,...)
```

Description

`Hs = spectrum.estmethod(input1,...)` returns a spectral estimation object `Hs` of type *`estmethod`*. This object contains all the parameter information needed for the specified estimation method. Each estimation method takes one or more inputs, which are described on the individual reference pages.

Estimation Methods

Estimation methods for `spectrum` specify the type of spectral estimation method to use. Available estimation methods for `spectrum` are listed below.

Note You must use a spectral *`estmethod`* with `spectrum`.

Spectrum Estimation Methods

spectrum.estmethod	Description	Corresponding Function
`spectrum.burg`	Burg	`pburg`
`spectrum.cov`	Covariance	`pcov`
`spectrum.eigenvector`	Eigenvector	`peig`
`spectrum.mcov`	Modified covariance	`pmcov`
`spectrum.mtm`	Thompson multitaper	`pmtm`
`spectrum.music`	Multiple Signal Classification	`pmusic`
`spectrum.periodogram`	Periodogram	`periodogram`
`spectrum.welch`	Welch	`pwelch`
`spectrum.yulear`	Yule-Walker	`pyulear`

For more information on each estimation method, use the syntax `help spectrum.estmethod` at the MATLAB prompt or refer to its reference page.

Note For estimation methods that use overlap and window length inputs, you specify the number of overlap samples as a percent overlap and you specify the segment length instead of the window length.

For estimation methods that use windows, if the window uses an additional parameter, a property is dynamically added to the spectrum object for that parameter. You can change that property using `set` (see "Changing Object Properties" on page 1-1112).

Methods

Methods provide ways of performing functions directly on your `spectrum` object without having to specify the spectral estimation parameters again. You can apply these methods directly on the variable you assigned to your `spectrum` object. For more information on any of these methods, use the syntax `help spectrum/method` at the MATLAB prompt or refer to the table below.

Spectrum Methods

Method	Description
`msspectrum`	Note that the `msspectrum` method is only available for the `periodogram` and `welch` spectrum estimation objects. The mean-squared spectrum is intended for discrete spectra (from periodic, discrete-time signals). The distribution of the mean square value across frequency is the msspectrum. Unlike the power spectral density (see `psd` below), the peaks in the mean-square spectrum reflect the power in the signal at a given frequency. For the PSD, the power is reflected as the area in a frequency band. The units of the mean-squared spectrum are units of power.

Spectrum Methods (Continued)

Method	Description

	`Hmss = msspectrum(Hs,X)` returns a mean-square spectrum object containing the mean-square (power) estimate of the discrete-time signal `X` using the spectrum object `Hs`. Default for real `X` is the `'onesided'` Nyquist frequency range and for complex `X` the default is the `'twosided'` Nyquist frequency range.
	
	`Hmss` contains a vector of normalized frequencies `W`, at which the mean-square spectrum is estimated. For real signals, the range of `W` is [0,π] if the number of FFT points (`NFFT`) is even, and [0,π) if `NFFT` is odd. For complex signals, the range of `W` is [0,2π). To estimate the spectrum on a vector of specific frequencies, see `FreqPoints` property below.
	
	The `msspectrum` method includes these properties, which you can set using this `msspectrum` method or via the `msspectrumopts` method. These properties are listed here and described in the `msspectrumopts` section below:
	
	`SpectrumType` — `'onesided'` or `'twosided'`
	`NormalizedFrequency` – normalizes frequency between 0 and 1
	`Fs` — sampling frequency in Hz
	`NFFT` — number of FFT points
	`CenterDC` — shifts data and frequencies to center DC component
	`FreqPoints` — `'All'` or `'User Defined'`
	`FrequencyVector` — frequencies at which to compute spectrum
	`ConfLevel` — confidence level to calculate the confidence interval. Value must be from 0 to 1.
	
	For example, `Hmss = msspectrum(Hs,X,'FreqPoints','User Defined', FreqVector,fvect)` returns a mean-square spectrum object where the spectrum is calculated only on the frequency points defined in the frequency vector, `fvect`.

Spectrum Methods (Continued)

Method	Description
	`msspectrum(...)` with no output arguments plots the mean-square spectrum in dB.
`msspectrumopts`	`Hopts = msspectrumopts(Hs)` returns an object that contains options for the spectrum object `Hs`. `Hopts = msspectrumopts(Hs,X)` returns an object with data-specific options and defaults. You can pass an `Hopts` options object as an argument to the `msspectrum` method. Any individual option you specify after the `Hopts` object overrides the value in `Hopts`. For example, `Hmss = msspectrum(Hs,X,Hopts, 'SpectrumType', 'twosided')` overrides the default `SpectrumType` value in `Hopts`. The following properties apply to both `msspectrumopts` and `msspectrum` methods. `Hmss = msspectrum (..., 'SpectrumType', 'twosided')` returns the two-sided mean-square spectrum. The spectrum length (NFFT) is computed over [0,2π), or if `Fs` is specified, [0,`Fs`) . Entering `'onesided'` returns the one-sided mean-square spectrum, which contains the total signal power in half the Nyquist range. Default is `'onesided'`. `Hmss = msspectrum(Hs,X,'NormalizedFrequency',true)` returns a mean-square spectrum object with frequency values normalized between 0 and 1. Default is `true`. `Hmss = msspectrum(Hs,X,'Fs',Fs)` returns a mean-square spectrum object computed as a function of frequency, where `Fs` is the sampling frequency in Hz. Note that you can set `Fs` only if `NormalizedFrequency` is set to `false`. `Hmss = msspectrum(...,'NFFT',nfft)` specifies the number of FFT points to use. Valid values are a positive integer, `'Nextpow2'` or `'Auto'`. `'Nextpow2'` uses the next power of 2 greater than the

Spectrum Methods (Continued)

Method	Description

	input length or 256, whichever is greater. `'Auto'` uses the input length or 256, whichever is greater. Default is `'Nextpow2'`. Note that for `spectrum.welch`, `'Nextpow2'` and `'Auto'` are compared to the `SegmentLength` instead of the input length.
	
	`Hmss = msspectrum (..., 'Centerdc', true)` shifts the data and frequency values so that the DC component is at the center of the spectrum. Default is `false`.
	
	To estimate the spectrum on a vector of specific frequencies, first set the number of frequency points to `'User Defined'`, which replaces the `NFFT` property of `msspectrum` with a `FrequencyVector` property.
	`Hopts.FreqPoints = 'User Defined'`
	(Note that the default for `FreqPoints` is `'All'`, which causes `msspectrum` to use the `NFFT` property as described above.)
	
	Then, specify the frequency vector `F` to use.
	`Hopts.FrequencyVector = F`
	(Note that the default value for `FrequencyVector` is `'Auto'`. In this case, the number of frequency points used follows the same rule as described for `NFFT 'Auto'` above.)
	
	`Hmms = msspectrum(...,'ConfLevel',p)` specifies the confidence level `p` for computing the confidence interval, which is an estimate of the error in the calculated mean-squared spectrum. The confidence level (`p`) is between 0 and 1. For example, `Hmss = msspectrum(Hs,X,'ConfLevel',0.95)` returns the 95% confidence interval.

Spectrum Methods (Continued)

Method	Description
psd	Note that music and eigenvector spectrum objects do not support the psd method. See the pseudospectrum method below. The power spectral density (PSD) is intended for continuous spectra. The integral of the PSD over a given frequency band computes the average power in the signal in that frequency band. In contrast to the msspectrum, the peaks in this spectra do not reflect the power at a given frequency. The units of the PSD are power per unit of frequency. See the avgpower method of dspdata for more information. Hpsd = psd (Hs,X) returns a power spectral density object containing the power spectral density estimate of the discrete-time signal X using the spectrum object Hs. The PSD is the distribution of power per unit frequency. Default for real X is 'onesided' and for complex X is 'twosided'. Hpsd contains a vector of normalized frequencies W, at which the PSD is estimated. For real signals, the range of W is [0,π] if the number of FFT points (NFFT) is even, and [0,π) if NFFT is odd. For complex signals, the range of W is [0,2π). The psd method includes these properties, which you can set using this psd method or via the psdopts method. These properties are listed here and described in the psdopts section below: SpectrumType — 'onesided' or 'twosided' NormalizedFrequency — normalizes frequency between 0 and 1 Fs — sampling frequency in Hz NFFT — number of FFT points CenterDC — shifts data and frequencies to center DC component FreqPoints — 'All' or 'User Defined' FrequencyVector – frequencies at which to compute spectrum ConfLevel — confidence level to calculate the confidence interval. Value must be from 0 to 1.

Spectrum Methods (Continued)

Method	Description
	For example, `Hmss = psd(Hs,X,'FreqPoints','User Defined', FreqVector,fvect)` returns a PSD object where the spectrum is calculated only on the frequency points defined in the frequency vector, `fvect`. `psd(...)` with no output arguments plots PSD in dB per unit frequency.
`psdopts`	`Hopts = psdopts(Hs)` returns an object that contains options for the spectrum object `Hs`. `Hopts = psdopts(Hs,X)` returns an object with data-specific options and defaults. You can pass an `Hopts` options object as an argument to the `psd` method. Any individual option you specify after the `Hopts` object overrides the value in `Hopts`. For example, `Hpsd = psd(Hs,X,Hopts,'SpectrumType', 'twosided')` overrides the `SpectrumType` value in `Hopts`. The following properties apply to both `psdmopts` and `psd` methods. `Hpsd = psd (Hs,X,'SpectrumType','twosided')` returns the two-sided power spectral density of `X`. The spectrum length is NFFT and is computed over [0,2π) if `Fs` is not specified or [0,`Fs`) if `Fs` is specified. Entering `'onesided'` returns the one-sided PSD, which contains the total signal power. `Hmss = psd(Hs,X,'NormalizedFrequency',true)` returns a power spectral density object with frequency values normalized between 0 and 1. Default is `true`. `Hpsd = psd (...,'Fs',Fs)` returns a power spectral density object computed as a function of frequency, where `Fs` is the sampling frequency in Hz.

Spectrum Methods (Continued)

Method	Description
	`Hmss = psd(...,'NFFT',nfft)` specifies the number of FFT points to use. Valid values are a positive integer, `'Nextpow2'` or `'Auto'`. `'Nextpow2'` uses the next power of 2 greater than the input length or 256, whichever is greater. `'Auto'` uses the input length or 256, whichever is greater. Default is `'Nextpow2'`. Note that for `spectrum.welch`, `'Nextpow2'` and `'Auto'` are compared to the `SegmentLength` instead of the input length. `Hmss = psd (..., 'Centerdc', true)` shifts the data and frequency values so that the DC component is at the center of the spectrum. Default is `false`. To estimate the spectrum on a vector of specific frequencies, first set the number of frequency points to `'User Defined'`, which replaces the `NFFT` property of `psd` with a `FrequencyVector` property. `Hopts.FreqPoints = 'User Defined'` (Note that the default for `FreqPoints` is `'All'` which causes `psd` to use the `NFFT` property as described above.) `Hmms = psd(...,'ConfLevel',p)` specifies the confidence level `p` for computing the confidence interval, which is an estimate of the error in the calculated PSD. The confidence level (`p`) is between 0 and 1. For example, `Hmss = psd(Hs,X,'ConfLevel',0.95)` returns the 95% confidence interval.

Spectrum Methods (Continued)

Method	Description

`pseudospectrum`	Note that this method is used for only `music` or `eigenvector` spectrum objects.
	
	`Hps = pseudospectrum(Hs,X)` returns an object containing the pseudospectrum estimate of the discrete-time signal `X` using the spectrum object `Hs`. `Hs` must be a `music` or `eigenvector` object. Default for real `X` is `'half'` and for complex `X` is the `'whole'` Nyquist frequency range.
	
	`Hps` contains a vector of normalized frequencies `W`, at which the pseudospectrum is estimated. For real signals, the range of `W` is [0,π] if the number of FFT points (`NFFT`) is even, and [0,π) if `NFFT` is odd. For complex signals, the range of `W` is [0,2π).
	
	The `pseudospectrum` method includes these properties, which you can set using this `pseudospectrum` method or via the `pseudospectrumopts` method. These properties are described below:
	
	`SpectrumRange` — `'half'` or `'whole'`
	`NormalizedFrequency` — normalizes frequency between 0 and 1
	`Fs` — sampling frequency in Hz
	`NFFT` — number of FFT points
	`CenterDC` — shifts data and frequencies to center DC component
	`FreqPoints` — `'All'` or `'User Defined'`
	`FrequencyVector` — frequencies at which to compute spectrum
	
	For example, `Hmss = psd(Hs,X,'FreqPoints','User Defined', FreqVector,fvect)` returns a PSD object where the spectrum is calculated only on the frequency points defined in the frequency vector, `fvect`.
	
	`pseudospectrum(...)` with no output arguments plots the pseudospectrum in dB.

Spectrum Methods (Continued)

Method	Description
`pseudo-spectrumopts`	`Hopts = pseudospectrumopts(Hs)` returns an object that contains options for the spectrum object `Hs`. `Hopts = pseudospectrumopts(Hs,X)` returns an object with data-specific options and defaults. You can pass an `Hopts` options object as an argument to the `pseudospectrum` method. Any individual option you specify after the `Hopts` object overrides the value in `Hopts`. For example, `Hpseudospectrum= pseudospectrum(Hs,X, Hopts,'SpectrumRange', 'whole')` overrides the `SpectrumRange` value in `Hopts`. `Hmps = pseudospectrum (..., 'SpectrumRange', 'whole')` returns the pseudospectrum over the whole Nyquist range. The spectrum length is NFFT and is computed over [0,2π) if `Fs` is not specified or [0,`Fs`) if `Fs` is specified. Entering `'half'` returns the pseudospectrum calculated over half the Nyquist range. `Hmss = pseudospectrum(Hs,X,'NormalizedFrequency',true)` returns a pseudospectrum object with frequency values normalized between 0 and 1. Default is `true`. `Hps = pseudospectrum(Hs,X,'Fs',Fs)` returns a pseudospectrum object computed as a function of frequency, where `Fs` is the sampling frequency in Hz. `Hps = pseudospectrum(...,'NFFT',nfft)` specifies the number of FFT points to use. Valid values are a positive integer, `'Nextpow2'` or `'Auto'`. `'Nextpow2'` uses the next power of 2 greater than the input length or 256, whichever is greater. `'Auto'` uses the input length or 256, whichever is greater. Default is `'Nextpow2'`. `Hps = pseudospectrum(...,'Centerdc',true)` shifts the data and frequency values so that the DC component is at the center of the spectrum. The default value is `false`. To estimate the spectrum on a vector of specific frequencies, first set the number of frequency points to `'User Defined'`, which replaces

Spectrum Methods (Continued)

Method	Description
	the NFFT property of pseudospectrum with a FrequencyVector property. Hopts.FreqPoints = 'User Defined' (Note that the default for FreqPoints is 'All', which causes pseudospectrum to use the NFFT property as described above.)
powerest	Note that powerest is available only for music and eigenvector spectrum objects. POW = powerest(Hs,X) returns a vector POW containing estimates of the powers of the complex sinusoids in X. The input X can be a vector or a matrix. If it is a matrix it can be a data matrix, where $X'^{*}X = R$ or a correlation matrix R. The value the InputType property of Hs determines how X is interpreted. Hs must be a music or eigenvector spectrum object. [POW,W]=powerest(Hs,X) returns POW and a vector W of the frequencies in rad/sample of the sinusoids in X. [POW,F]=powerest(Hs,X,Fs) returns POW and a vector F of the frequencies in Hz of the sinusoids in X. Fs is the sampling frequency.

Viewing Object Properties

As with any object, you can use get to view a spectrum object's properties. To see a specific property, use

```
get(Hs,'property')
```

where 'property' is the specific property name.

To see all properties for an object, use

```
get(Hs)
```

Changing Object Properties

To set specific properties, use

```
set(Hs,'property1',value, 'property2',value,...)
```

where 'property1', 'property2', etc. are the specific property names.

To view the options for a property use set without specifying a value

```
set(Hs,'property')
```

Note that you must use single quotation marks around the property name. For example, to change the order of a Burg spectrum object Hs to 6, use

```
set(Hs,'order',6)
```

Another example of using set to change an object's properties is this example of changing the dynamically created window property of a periodogram spectrum object.

```
Hs=spectrum.periodogram       % Create periodogram object

Hs =

    EstimationMethod: 'Periodogram'
          WindowName: 'Rectangular'

set(Hs,'WindowName','Chebyshev')   % Change window type
Hs                                 % View changed object

Hs =

    EstimationMethod: 'Periodogram'
          WindowName: 'Chebyshev'  % Note changed property
       SidelobeAtten: 100

set(Hs,'SidelobeAtten',150)   % Change dynamic property
```

```
Hs                              % View changed object

Hs =

    EstimationMethod: 'Periodogram'
          WindowName: 'Chebyshev'
       SidelobeAtten: 150
```

All `spectrum` object properties can be changed using the `set` command, except for the `EstimationMethod` property.

Another way to change an object's properties is by using the `inspect` command which opens the Property Inspector window where you can edit any property, except dynamic properties, such as those used with windows.

```
inspect(Hs)
```

Copying an Object

To create a copy of an object, use the `copy` method.

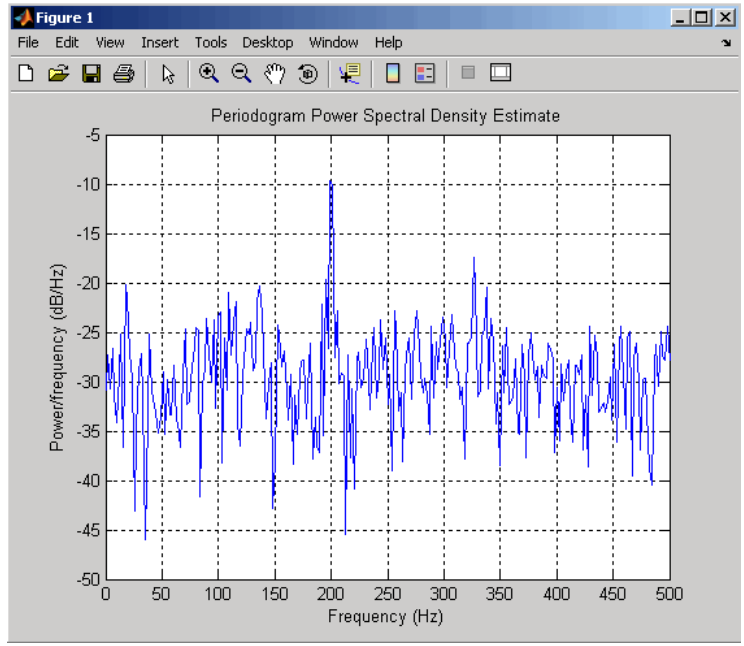
```
H2 = copy(Hs)
```

Note Using the syntax `H2 = Hs` copies only the object handle and does not create a new object.

Examples

Define a cosine of 200 Hz, add some noise and then view its power spectral density estimate generated with the periodogram algorithm.

```
Fs = 1000;
t = 0:1/Fs:.3;
x=cos(2*pi*t*200)+randn(size(t));
Hs=spectrum.periodogram;
psd(Hs,x,'Fs',Fs)
```



Refer to the reference pages for each estimation method for more examples.

See Also

`dspdata` | `spectrum.burg` | `spectrum.cov` | `spectrum.mcov` | `spectrum.yulear` | `spectrum.periodogram` | `spectrum.welch` | `spectrum.mtm` | `spectrum.eigenvector` | `spectrum.music`

Purpose

Burg spectrum

Syntax

```
Hs = spectrum.burg
Hs = spectrum.burg(order)
```

Description

`Hs = spectrum.burg` returns a default Burg spectrum object, `Hs`, that defines the parameters for the Burg parametric spectral estimation algorithm. The Burg algorithm estimates the spectral content by fitting an autoregressive (AR) linear prediction filter model of a given order to the signal.

`Hs = spectrum.burg(order)` returns a spectrum object, `Hs` with the specified `order`. The default value for `order` is 4.

Note See `pburg` for more information on the Burg algorithm.

Examples

Define a fourth order autoregressive model and view its power spectral density using the Burg algorithm.

```
x=randn(100,1);
x=filter(1,[1 1/2 1/3 1/4 1/5],x);   % 4th order AR filter
Hs=spectrum.burg;                    % 4th order AR model
psd(Hs,x,'NFFT',512)
```

See Also

`dspdata` | `spectrum` | `spectrum.cov` | `spectrum.mcov` | `spectrum.yulear` | `spectrum.periodogram` | `spectrum.welch` | `spectrum.mtm` | `spectrum.eigenvector` | `spectrum.music`

Purpose

Covariance spectrum

Syntax

```
Hs = spectrum.cov
Hs = spectrum.cov(order)
```

Description

`Hs = spectrum.cov` returns a default covariance spectrum object, `Hs`, that defines the parameters for the covariance spectral estimation algorithm. The covariance algorithm estimates the spectral content by fitting an autoregressive (AR) linear prediction model of a given `order` to the signal.

`Hs = spectrum.cov(order)` returns a spectrum object, `Hs` with the specified `order`. The default value for `order` is 4.

Note See `pcov` for more information on the covariance algorithm.

Examples

Define a fourth order autoregressive model and view its power spectral density using the covariance algorithm.

```
x=randn(100,1);
x=filter(1,[1 1/2 1/3 1/4 1/5],x);   % 4th order AR filter
Hs=spectrum.cov;                     % 4th order AR model
psd(Hs,x,'NFFT',512)
```

See Also

`dspdata` | `spectrum` | `spectrum.burg` | `spectrum.mcov` | `spectrum.yulear` | `spectrum.periodogram` | `spectrum.welch` | `spectrum.mtm` | `spectrum.eigenvector` | `spectrum.music`

Purpose

Eigenvector spectrum

Syntax

```
Hs = spectrum.eigenvector
Hs = spectrum.eigenvector(NSinusoids)
Hs = spectrum.eigenvector(NSinusoids,SegmentLength)
Hs = spectrum.eigenvector(NSinusoids,SegmentLength,...
OverlapPercent)
Hs = spectrum.eigenvector(NSinusoids,SegmentLength,...
OverlapPercent,WindowName)
Hs = spectrum.eigenvector(NSinusoids,SegmentLength,...
OverlapPercent,WindowName,SubspaceThreshold)
Hs = spectrum.eigenvector(NSinusoids,SegmentLength,...
OverlapPercent,WindowName,SubspaceThreshold,InputType)
```

Description

`Hs = spectrum.eigenvector` returns a default eigenvector spectrum object, `Hs`, that defines the parameters for an eigenanalysis spectral estimation method. This object uses the following default values:

Default Values

Property Name	Default Value	Description
`NSinusoids`	`2`	Number of complex sinusoids
`SegmentLength`	`4`	Length of each of the time-based segments into which the input signal is divided.
`OverlapPercent`	`50`	Percent overlap between segments

Default Values (Continued)

Property Name	Default Value	Description
`WindowName`	`'Rectangular'`	Window name string or `'User Defined'` (see `window` for valid window names). For more information on each window, refer to its reference page. This argument can also be a cell array containing the window name string or `'User Defined'` and, if used for the particular window, an optional parameter value. The syntax is `{wname,wparam}`. You can use `set` to change the value of the additional parameter or to define the MATLAB expression and parameters for a user-defined window (see `spectrum` for information on using `set`).
`SubspaceThreshold`	`0`	Threshold is the cutoff for signal and noise separation. The threshold is multiplied by λ_{min} , the smallest estimated eigenvalue of the signal's correlation matrix. Eigenvalues below the threshold (λ_{min}`*threshold`) are assigned to the noise subspace.
`InputType`	`'Vector'`	Type of input that will be used with this spectrum object. Valid values are `'Vector'`, `'DataMatrix'` and `'CorrelationMatrix'`.

`Hs = spectrum.eigenvector(NSinusoids)` returns a spectrum object, `Hs`, with the specified number of sinusoids and default values for all other properties. Refer to the table above for default values.

`Hs = spectrum.eigenvector(NSinusoids,SegmentLength)` returns a spectrum object, `Hs`, with the specified segment length.

`Hs = spectrum.eigenvector(NSinusoids,SegmentLength,...
OverlapPercent)` returns a spectrum object, `Hs`, with the specified overlap between segments.

`Hs = spectrum.eigenvector(NSinusoids,SegmentLength,...
OverlapPercent,WindowName)` returns a spectrum object, `Hs`, with the specified window.

Note Window names must be enclosed in single quotes, such as `spectrum.eigenvector(3,32,50,'chebyshev')` or `spectrum.eigenvector(3,32,50,{'chebyshev',60})`.

`Hs = spectrum.eigenvector(NSinusoids,SegmentLength,...
OverlapPercent,WindowName,SubspaceThreshold)` returns a spectrum object, `Hs`, with the specified subspace threshold.

`Hs = spectrum.eigenvector(NSinusoids,SegmentLength,...
OverlapPercent,WindowName,SubspaceThreshold,InputType)` returns a spectrum object, `Hs`, with the specified input type.

Note See `peig` for more information on the eigenanalysis algorithm.

Examples

Define a complex signal with three sinusoids, add noise, and view its pseudospectrum using eigenanalysis. Set the FFT length to 128.

```
n=0:99;
s=exp(i*pi/2*n)+2*exp(i*pi/4*n)+exp(i*pi/3*n)+randn(1,100);
Hs=spectrum.eigenvector(3,32,95,'rectangular',5);
```

```
pseudospectrum(Hs,s,'NFFT',128)
```

References

[1] Harris, F. J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform.” *Proceedings of the IEEE*. Vol. 66 (January 1978).

See Also

dspdata | spectrum | spectrum.music | spectrum.burg | spectrum.cov | spectrum.mcov | spectrum.yulear | spectrum.periodogram | spectrum.welch | spectrum.mtm

Purpose

Modified covariance spectrum

Syntax

```
Hs = spectrum.mcov
Hs = spectrum.mcov(order)
```

Description

`Hs = spectrum.mcov` returns a default modified covariance spectrum object, `Hs`, that defines the parameters for the modified covariance spectral estimation algorithm. The modified covariance algorithm estimates the spectral content by fitting an autoregressive (AR) linear prediction filter model of a given order to the signal.

`Hs = spectrum.mcov(order)` returns a spectrum object, `Hs` with the specified `order`. The default value for `order` is 4.

Note See `pmcov` for more information on the modified covariance algorithm.

Examples

Define a fourth order autoregressive model and view its power spectral density using the modified covariance algorithm.

```
x=randn(100,1);
x=filter(1,[1 1/2 1/3 1/4 1/5],x);   % 4th order AR filter
Hs=spectrum.mcov;                    % 4th order AR model
psd(Hs,x,'NFFT',512)
```

See Also

`dspdata` | `spectrum` | `spectrum.burg` | `spectrum.cov` | `spectrum.yulear` | `spectrum.periodogram` | `spectrum.welch` | `spectrum.mtm` | `spectrum.eigenvector` | `spectrum.music`

Purpose

Thomson multitaper spectrum

Syntax

```
Hs = spectrum.mtm
Hs = spectrum.mtm(TimeBW)
Hs = spectrum.mtm(DPSS,Concentrations)
Hs = spectrum.mtm(...,CombineMethod)
```

Description

`Hs = spectrum.mtm` returns a default Thomson multitaper spectrum object, `Hs` that defines the parameters for the Thomson multitaper spectral estimation algorithm, which uses a linear or nonlinear combination of modified periodograms. The periodograms are computed using a sequence of orthogonal tapers (windows in the frequency domain) specified from discrete prolate spheroidal sequences (`dpss`). This object uses the following default values:

Property Name	Default Value	Description
`TimeBW`	`4`	Product of time and bandwidth for the discrete prolate spheroidal sequences (or Slepian sequences) used as data windows
`CombineMethod`	`'adaptive'`	Algorithm for combining the individual spectral estimates. Valid values are `'adaptive'` — adaptive (nonlinear) `'unity'` — unity weights (linear) `'eigenvector'` — Eigenvalue weights (linear)

`Hs = spectrum.mtm(TimeBW)` returns a spectrum object, `Hs` with the specified time-bandwidth product.

`Hs = spectrum.mtm(DPSS,Concentrations)` returns a spectrum object, `Hs` with the specified dpss data tapers and their concentrations.

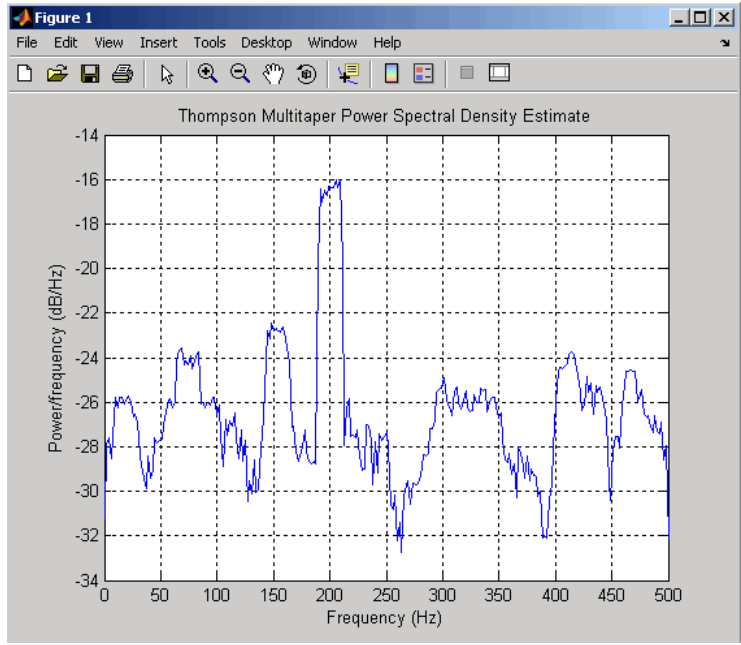
Note You can either specify the time-bandwidth product (`TimeBW`) or the `DPSS` data tapers and their `Concentrations`. See `dpss` and `pmtm` for more information.

`Hs = spectrum.mtm(...,CombineMethod)` returns a spectrum object, `Hs`, with the specified method for combining the spectral estimates. Refer to the table above for valid `CombineMethod` values.

Examples

Define a cosine of 200 Hz, add noise and view its power spectral density using the Thomson multitaper algorithm with a time-bandwidth product of 3.5.

```
Fs=1000;
t=0:1/Fs:.3;
x=cos(2*pi*t*200)+randn(size(t));
Hs=spectrum.mtm(3.5);
psd(Hs,x,'Fs',Fs)
```



The above example could be done by specifying the data tapers and concentrations instead of the time-bandwidth product.

```
Fs=1000;
t=0:1/Fs:.3;
x=cos(2*pi*t*200)+randn(size(t));
[e,v]=dpss(length(x),3.5);
Hs=spectrum.mtm(e,v);
psd(Hs,x,'Fs',Fs)
```

See Also

dspdata | spectrum | spectrum.periodogram | spectrum.welch | spectrum.burg | spectrum.cov | spectrum.mcov | spectrum.yulear | spectrum.eigenvector | spectrum.music

Purpose

Multiple signal classification spectrum

Syntax

```
Hs = spectrum.music
Hs = spectrum.music(NSinusoids)
Hs = spectrum.eigenvector(NSinusoids,SegmentLength)
Hs = spectrum.music(NSinusoids,SegmentLength,...
OverlapPercent)
Hs = spectrum.music(NSinusoids,SegmentLength,...
OverlapPercent,WindowName)
Hs = spectrum.music(NSinusoids,SegmentLength,...
OverlapPercent,WindowName,SubspaceThreshold)
Hs = spectrum.music(NSinusoids,SegmentLength,...
OverlapPercent,WindowName,SubspaceThreshold,InputType)
```

Description

`Hs = spectrum.music` returns a default multiple signal classification (MUSIC) spectrum object, `Hs`, that defines the parameters for the MUSIC spectral estimation algorithm, which uses Schmidt's eigenspace analysis algorithm. This object uses the following default values.

Default Values

Property Name	Default Value	Description
`NSinusoids`	`2`	Number of complex sinusoids
`SegmentLength`	`4`	Length of each of the time-based segments into which the input signal is divided.
`OverlapPercent`	`50`	Percent overlap between segments

Default Values (Continued)

Property Name	Default Value	Description
`WindowName`	`'Rectangular'`	Window name string or `'User Defined'` (see `window` for valid window names). For more information on each window, refer to its reference page). This argument can also be a cell array containing the window name string or `'User Defined'` and, if used for the particular window, an optional parameter value. The syntax is `{wname,wparam}`. You can use `set` to change the value of the additional parameter or to define the MATLAB expression and parameters for a user-defined window (see `spectrum` for information on using `set`).

Default Values (Continued)

Property Name	Default Value	Description
`SubspaceThreshold`	`0`	Threshold is the cutoff for signal and noise separation. The threshold is multiplied by λ_{min} , the smallest estimated eigenvalue of the signal's correlation matrix. Eigenvalues below the threshold (λ_{min}`*threshold`) are assigned to the noise subspace.
`InputType`	`'Vector'`	Type of input that will be used with this spectrum object. Valid values are `'Vector'`, `'DataMatrix'` and `'CorrelationMatrix'`.

`Hs = spectrum.music(NSinusoids)` returns a spectrum object, `Hs`, with the specified number of sinusoids and default values for all other properties. Refer to the table above for default values.

`Hs = spectrum.eigenvector(NSinusoids,SegmentLength)` returns a spectrum object, `Hs`, with the specified segment length.

`Hs = spectrum.music(NSinusoids,SegmentLength,...
OverlapPercent)` returns a spectrum object, `Hs`, with the specified overlap between segments.

`Hs = spectrum.music(NSinusoids,SegmentLength,...
OverlapPercent,WindowName)` returns a spectrum object, `Hs`, with the specified window.

Note Window names must be enclosed in single quotes, such as `spectrum.music(3,32,50,'chebyshev')` or `spectrum.music(3,32,50,{'chebyshev',60})`

`Hs = spectrum.music(NSinusoids,SegmentLength,...
OverlapPercent,WindowName,SubspaceThreshold)` returns a spectrum object, `Hs`, with the specified subspace threshold.

`Hs = spectrum.music(NSinusoids,SegmentLength,...
OverlapPercent,WindowName,SubspaceThreshold,InputType)` returns a spectrum object, `Hs`, with the specified input type.

Note See `pmusic` for more information on the MUSIC algorithm.

Examples

Define a complex signal with three sinusoids, add noise, and estimate its pseudospectrum using the MUSIC algorithm.

```
n=0:99;
s=exp(i*pi/2*n)+2*exp(i*pi/4*n)+exp(i*pi/3*n)+randn(1,100);
Hs=spectrum.music(3,20);
pseudospectrum(Hs,s)
```

References

[1] Harris, F. J. “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform.” *Proceedings of the IEEE*. Vol. 66 (January 1978).

See Also

`dspdata | spectrum | spectrum.eigenvector | spectrum.burg | spectrum.cov | spectrum.mcov | spectrum.yulear | spectrum.periodogram | spectrum.welch | spectrum.mtm`

Purpose

Periodogram spectrum

Syntax

```
Hs = spectrum.periodogram
Hs = spectrum.periodogram(winname)
Hs = spectrum.periodogram({winname,winparameter})
```

Description

`Hs = spectrum.periodogram` returns a default periodogram spectrum object, `Hs`, that defines the parameters for the periodogram spectral estimation method. This default object uses a rectangular window and a default FFT length equal to the next power of 2 (`NextPow2`) that is greater than the input length.

`Hs = spectrum.periodogram(winname)` returns a spectrum object, `Hs`, that uses the specified window. If the window uses an optional associated window parameter, it is set to the default value. This object uses the default FFT length.

`Hs = spectrum.periodogram({winname,winparameter})` returns a spectrum object, `Hs`, that uses the specified window and optional associated window parameter, if any. You specify the window and window parameter in a cell array with a `windowname` string and the parameter value. This object uses the default FFT length.

Valid `windowname` strings are:

```
'Bartlett'
'Bartlett-Hanning'
'Blackman'
'Blackman-Harris'
'Bohman'
'Chebyshev'
'Flat Top'
'Gaussian'
'Hamming'
'Hann'
'Kaiser'
'Nuttall'
'Parzen'
```

```
'Rectangular'
'Triangular'
'Tukey'
'User Defined'
```

See `window` and the corresponding window function page for window parameter information.

You can use `set` to change the value of the additional parameter or to define the MATLAB expression and parameters for a user-defined window (see `spectrum` for information on using `set`).

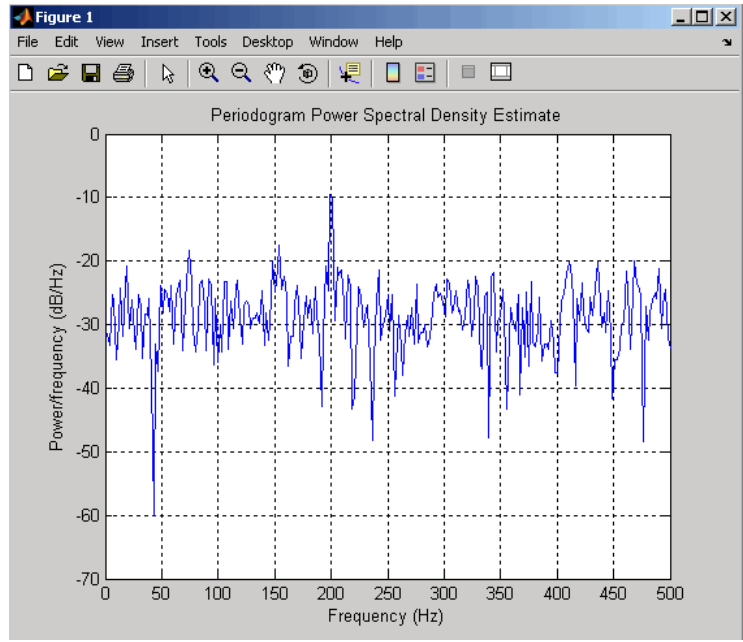
Note Window names must be enclosed in single quotes, such as `spectrum.periodogram('tukey')` or `spectrum.periodogram({'tukey',0.7})`.

Note See `periodogram` for more information on the periodogram algorithm.

Examples

Define a cosine of 200 Hz, add noise and view its spectral content using the periodogram spectral estimation technique.

```
Fs=1000;
t=0:1/Fs:.3;
x=cos(2*pi*t*200)+randn(size(t));
Hs=spectrum.periodogram;      % Use default values
psd(Hs,x,'Fs',Fs)
```



References

[1] Harris, F. J. "On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform." *Proceedings of the IEEE*. Vol. 66 (January 1978).

See Also

`dspdata` | `spectrum` | `spectrum.welch` | `spectrum.mtm` | `spectrum.burg` | `spectrum.cov` | `spectrum.mcov` | `spectrum.yulear` | `spectrum.eigenvector` | `spectrum.music`

spectrum.welch

Purpose

Welch spectrum

Syntax

```
Hs = spectrum.welch
Hs = spectrum.welch(WindowName)
Hs = spectrum.welch(WindowName,SegmentLength)
Hs = spectrum.welch(WindowName,SegmentLength,OverlapPercent)
```

Description

`Hs = spectrum.welch` returns a default Welch spectrum object, `Hs`, that defines the parameters for Welch's averaged, modified periodogram spectral estimation method. The object uses these default values.

Property Name	Default Value	Description
`{WindowName, winparam}` Cell array containing `WindowName` and optional window parameter	`'Hamming'`, `SamplingFlag: symmetric`	Cell array containing the window name string or `'User Defined'` and, if used for the particular window, an optional parameter value. (See `window` for valid window names and for more information on each window, refer to its reference page.) You can use `set` to change the value of the additional parameter or to define the MATLAB expression and parameters for a user-defined window. (See `spectrum` for information on using `set`.)
`WindowName`	`'Hamming'`, `SamplingFlag: symmetric`	Valid `windowname` strings are: `'Bartlett'`

Property Name	Default Value	Description
		'Bartlett-Hanning' 'Blackman' 'Blackman-Harris' 'Bohman' 'Chebyshev' 'Flat Top' 'Gaussian' 'Hamming' 'Hann' 'Kaiser' 'Nuttall' 'Parzen' 'Rectangular' 'Triangular' 'Tukey' 'User Defined' Window names must be enclosed in single quotes, such as `spectrum.welch('tukey')` or `spectrum.welch({'tukey',0.7})`. See `window` and the corresponding window function page for window parameter information. You can use `set` to change the value of the additional window parameter or to define the MATLAB expression and parameters for a user-defined window (see

Property Name	Default Value	Description
		`spectrum` for information on using `set`).
`SegmentLength`	`64`	Length of each of the time-based segments into which the input signal is divided. A modified periodogram is computed on each segment and the average of the periodograms forms the spectral estimate. Choosing the segment length is a compromise between estimate reliability (shorter segments) and frequency resolution (longer segments). A long segment length produces better resolution while a short segment length produces more averages, and therefore a decrease in the variance.
`OverlapPercent`	`50%`	Percent overlap between segments

`Hs = spectrum.welch(WindowName)` returns a spectrum object, `Hs`, using Welch's method with the specified window and the default values for all other parameters. To specify parameters for a window, use a cell array formatted as `spectrum.welch({WindowName,winparam})`.

`Hs = spectrum.welch(WindowName,SegmentLength)` returns a spectrum object, `Hs` with the specified segment length.

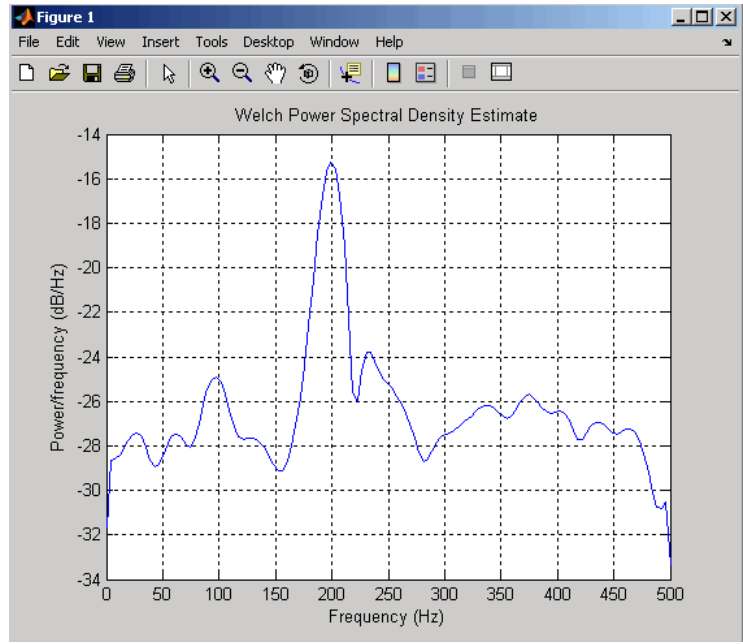
`Hs = spectrum.welch(WindowName,SegmentLength,OverlapPercent)` returns a spectrum object, `Hs` with the specified percentage overlap between segments.

Note See `pwelch` for more information on the Welch algorithm.

Examples

Define a cosine of 200 Hz, add noise and view its spectral content using the Welch algorithm.

```
Fs=1000;
t=0:1/Fs:.3;
x=cos(2*pi*t*200)+randn(size(t));
Hs=spectrum.welch;
psd(Hs,x,'Fs',Fs)
```



The following example produces a result similar to the obsoleted `spectrum` function, which used a Hann window as the default.

```
Fs = 1000;
t = 0:1/Fs:.3;
x=cos(2*pi*t*200)+randn(size(t));
window=33;
noverlap=32;
nfft=4097;
h = spectrum.welch('Hann',window,100*noverlap/window);
hpsd = psd(h,x,'NFFT',nfft,'Fs',Fs);
Pw = hpsd.Data;
Fw = hpsd.Frequencies;
```

References

[1] Harris, F. J. "On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform." *Proceedings of the IEEE*. Vol. 66 (January 1978).

See Also

`dspdata` | `spectrum` | `spectrum.periodogram` | `spectrum.mtm` | `spectrum.burg` | `spectrum.cov` | `spectrum.mcov` | `spectrum.yulear` | `spectrum.eigenvector` | `spectrum.music`

Purpose

Yule-Walker spectrum object

Syntax

```
Hs = spectrum.yulear
Hs = spectrum.yulear(order)
```

Description

`Hs = spectrum.yulear` returns a default Yule-Walker spectrum object, `Hs`, that defines the parameters for the Yule-Walker spectral estimation algorithm. This method is also called the auto-correlation or windowed method. The Yule-Walker algorithm estimates the spectral content by fitting an autoregressive (AR) linear prediction filter model of a given `order` to the signal. This leads to a set of Yule-Walker equations, which are solved using Levinson-Durbin recursion.

`Hs = spectrum.yulear(order)` returns a spectrum object, `Hs`, with the specified `order`. The default value for `order` is 4.

Note See `pyulear` for more information on the Yule-Walker algorithm.

Examples

Define a fourth order autoregressive model and view its spectral content using the Yule-Walker algorithm.

```
x=randn(100,1);
x=filter(1,[1 1/2 1/3 1/4 1/5],x);   % 4th order AR filter
Hs=spectrum.yulear;                  % 4th order AR model
psd(Hs,x,'NFFT',512)
```

See Also

`dspdata` | `spectrum` | `spectrum.burg` | `spectrum.cov` | `spectrum.mcov` | `spectrum.periodogram` | `spectrum.welch` | `spectrum.mtm` | `spectrum.eigenvector` | `spectrum.music`

Purpose

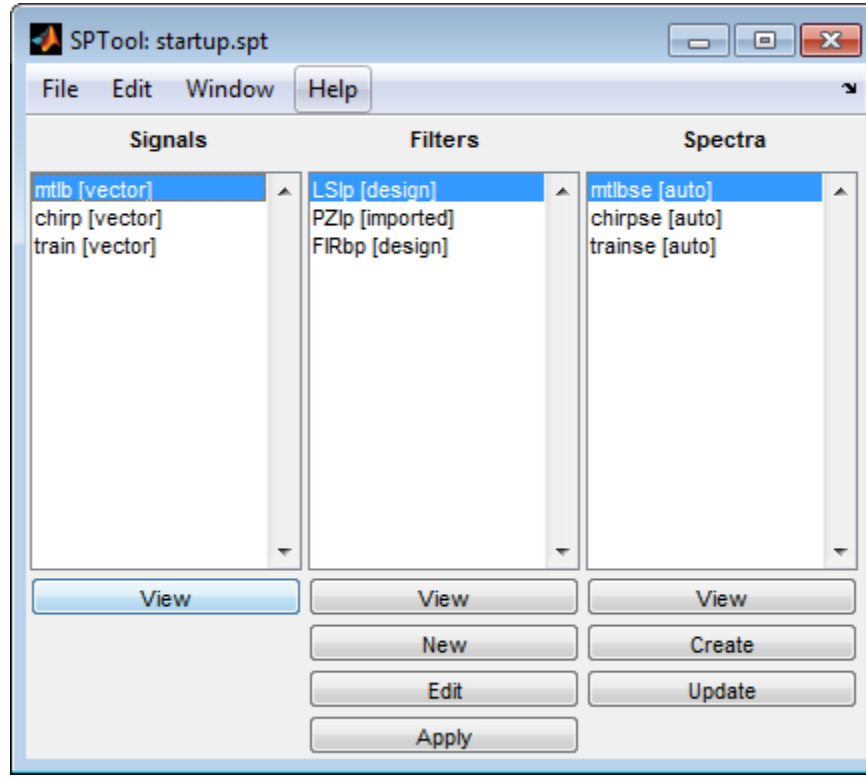
Open interactive digital signal processing tool

Syntax

```
sptool
```

Description

The command, `sptool`, opens SPTool, a suite of four tools: Signal Browser, Filter Design and Analysis Tool, FVTool, and Spectrum Viewer. These tools provide access to many of the signal, filter, and spectral analysis functions in the toolbox. When you type `sptool` at the command line, the SPTool suite opens.



Using SPTool, you can:

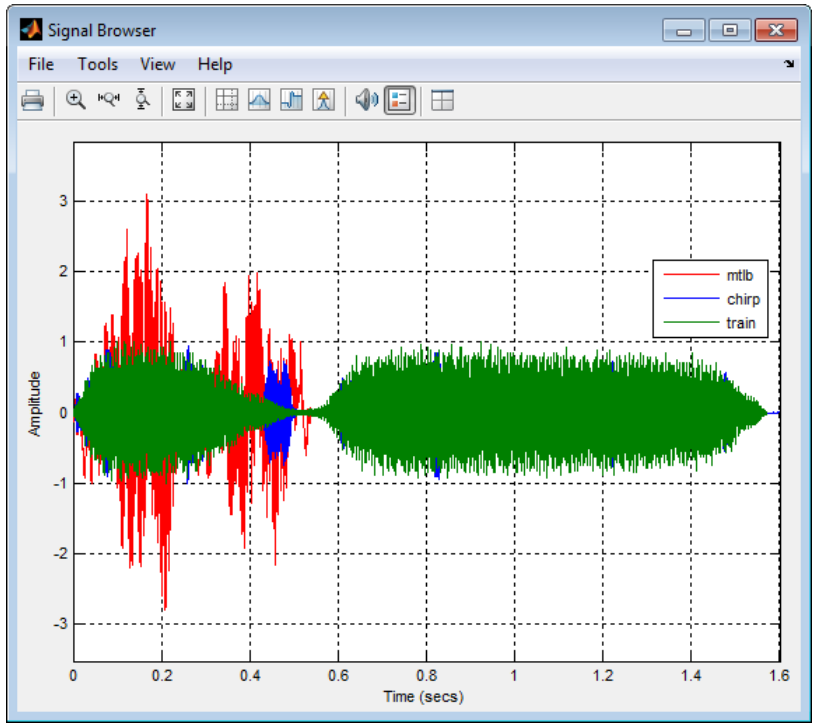
- Analyze signals listed in the **Signals** list box with the Signal Browser.
- Design or edit filters with the Filter Design and Analysis Tool (includes a Pole/Zero Editor).
- Analyze filter responses for filters listed in the **Filters** list box with FVTool.
- Apply filters in the **Filters** list box to signals in the **Signals** list box.
- Create and analyze signal spectra with the Spectrum Viewer.
- Print the Signal Browser, Filter Design and Analysis Tool, and Spectrum Viewer.

You can activate all four integrated signal processing tools from SPTool.

- "Signal Browser" on page 1-1139
- "Filter Design and Analysis Tool" on page 1-1183
- "Filter Visualization Tool" on page 1-1184
- "Spectrum Viewer" on page 1-1185

Signal Browser

The Signal Browser, hereafter referred to as the scope, allows you to view, measure, and analyze the time-domain information of one or more signals. To activate the Signal Browser, press the **View** button under the **Signals** list box in SPTool.



See the following sections for more information on the Signal Browser:

- “Displaying Multiple Signals” on page 1-1141
- “Signal Display” on page 1-1144
- “Toolbar” on page 1-1147
- “Measurements Panels” on page 1-1151
- “Visuals — Time Domain Options” on page 1-1174
- “Style Dialog Box” on page 1-1180

Displaying Multiple Signals

Multiple Signal Input

Select more than one signal in the **Signals** list box to show multiple signals within the same display or on separate displays. By default, the signals appear as different-colored lines on the same display. The signals can have different dimensions, sample rates, and data types. Each signal can be either real or complex valued.

Multiple Signal Colors

By default, Signal Browser has a white axes background and chooses line colors for each channel in a manner similar to the MATLAB `plot` function. Signal Browser considers each of the real and imaginary components of the input signals to be a channel, and assigns each channel a line color in the following order:

1 Blue

2 Dark Green

3 Red

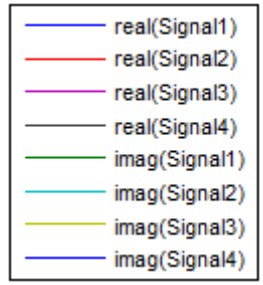
4 Cyan

5 Purple

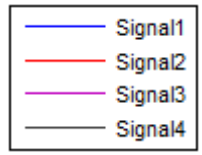
6 Dark Yellow

7 Black

If there are more than 7 channels, the scope repeats this order to assign line colors to the remaining channels. For example, if you select 4 complex-valued input signals, the following legend appears in the display.



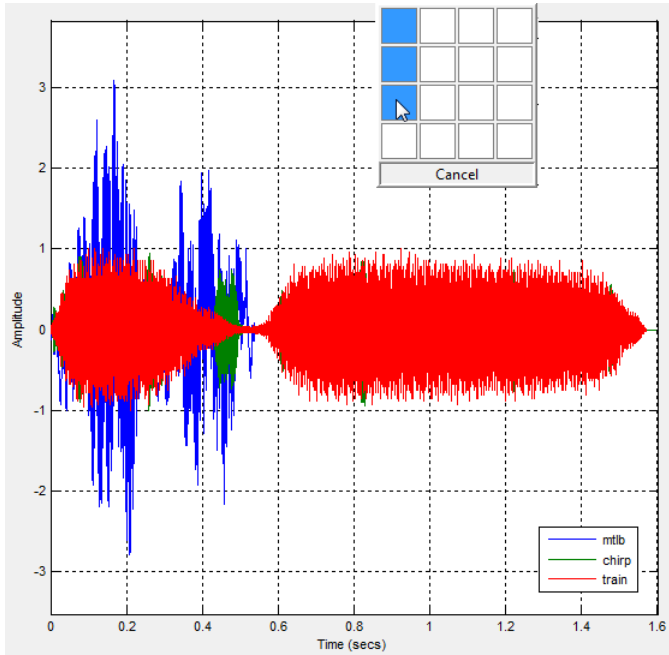
If all the input signals are real-valued, Signal Browser skips the line colors that would be associated with their imaginary components. For example, if you select 4 real-valued input signals, the following legend appears in the display.



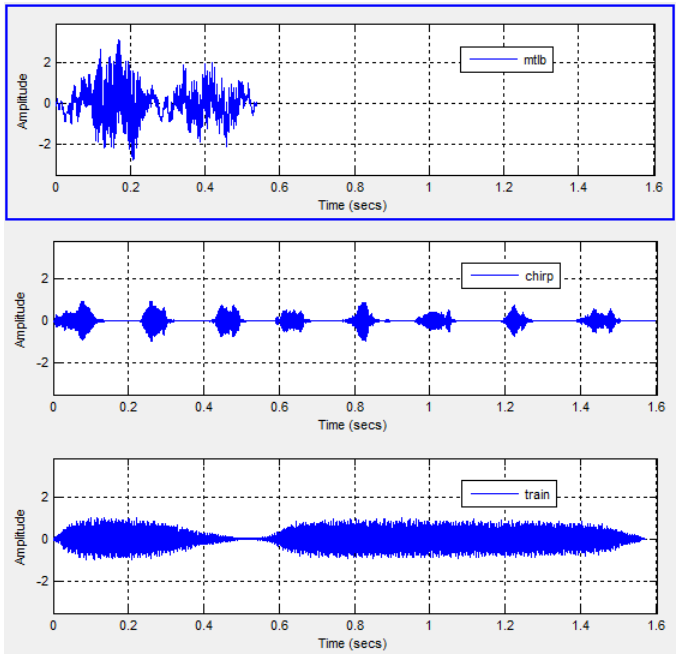
To manually modify any line color, select **View > Style** to open the Style dialog box. Next to **Properties for line**, select the signal name whose color you want to change. Then, next to **Line**, click the Line color button () and select any color from the palette. To change the axes background color, click the Axes background color button (), and select any color from the palette.

Multiple Displays

You can display multiple channels of data on different displays in the scope window. In the scope toolbar, select **View > Layout**, or select the Layout button (). This feature allows you to tile the window into a number of separate displays, up to a grid of 4 rows and 4 columns. For example, if there are three inputs to the scope, you can display the signals in separate displays by selecting row 3, column 1, as shown in the following figure.



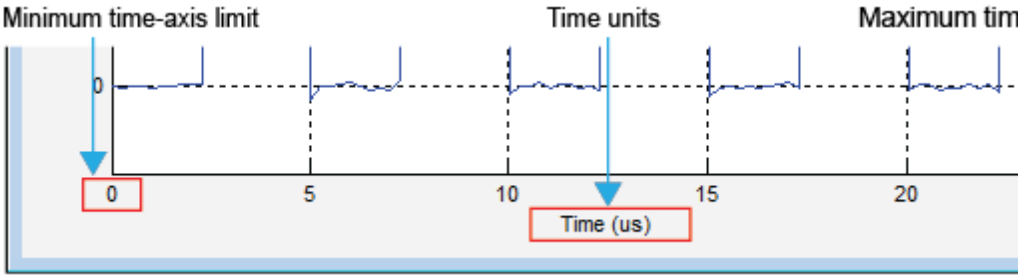
After you select row 3, column 1, the scope window is partitioned into three separate displays, as shown in the following figure.



When you use the Layout option to tile the window into multiple displays, the display highlighted in blue is referred to as the *active display*. The scope dialog boxes reference the active display.

Signal Display

The Signal Browser uses the longest time length of all the input signals selected in the **Signals** list box for the time range. To communicate the array of times that corresponds to the current display, the scope uses the **Minimum time-axis limit**, **Time units**, and **Maximum time-axis limit** indicators on the scope window. The following figure highlights these aspects of the Signal Browser window.



- **Minimum time-axis limit** — The Signal Browser sets the minimum *time*-axis limit to 0.
- **Maximum time-axis limit** — The Signal Browser sets the maximum *time*-axis limit to the final time step of the longest input signal.
- **Time units** — The units used to describe the *time*-axis. The Signal Browser sets the time units using the value of the **Time Units** parameter on the **Main** tab of the Visuals:Time Domain Options dialog box. By default, this parameter is set to `Metric (based on Time Span)` and displays in metric units such as microseconds, milliseconds, minutes, days, etc. You can change the unit of measure to `Seconds` to always display the *time*-axis values in units of seconds. You can change it to `None` to suppress the display of units of measure on the *time*-axis. When you set this parameter to `None`, then the Signal Browser shows only the word `Time` on the *time*-axis.

  To hide both the word `Time` and the values on the *time*-axis, set the **Show time-axis labels** parameter to `None`. To hide both the word `Time` and the values on the *time*-axis in all displays except the bottom ones in each column of displays, set this parameter to `Bottom Displays Only`. This behavior differs from that of the Simulink Scope block, which always shows the values but never shows a label on the *x*-axis.

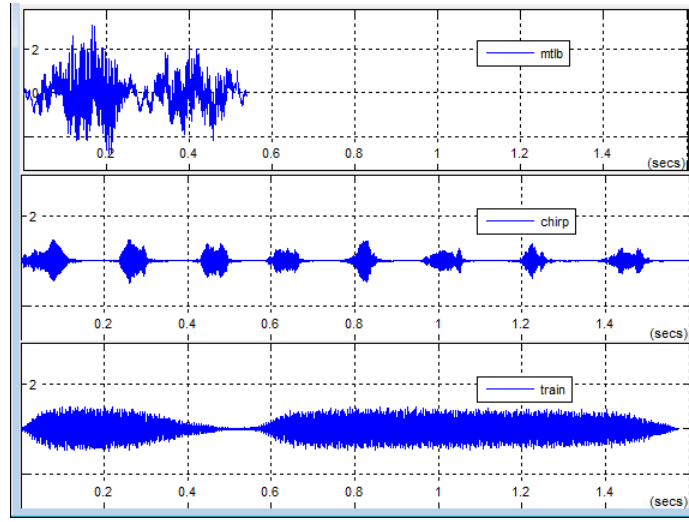
Signal Names and Legend Strings

Signal Browser uses the names of the signals in the SPTool as the strings displayed in the legends. If you change the name of any selected

signal in the **Signals** list box, its corresponding legend string in Signal Browser changes immediately. To change the name of any selected signal, from the SPTool menu, select **Edit > Name**. Signal Browser automatically updates the legend string to reflect the new signal name you entered. Similarly, if you modify any string in a legend in Signal Browser, SPTool updates the corresponding signal name in the **Signals** list box.

Axes Maximization

You can specify whether to display the Signal Browser in maximized axes mode. In this mode, the axes are expanded to fill the entire display. In each display, there is no space to show titles or axis labels. The minimum and maximum *time*-axis limits are located at the far-left and far-right edges of the display. The values at the axis tick marks appear as grid lines on top of the axes. The following figure highlights how three displays appear in maximized axes mode in the Signal Browser window.



To enable or disable this mode, in the Signal Browser menu, select **View > Properties** to bring up the Visuals:Time Domain Options

dialog box. In the **Main** pane, you can set the **Maximize axes** parameter to one of the following options:

- `Auto` — In this mode, the axes appear maximized in all displays only if the **Title** and **Y-Axis label** parameters are empty for every display. If you enter any value in any display for either of these parameters, the axes are not maximized.
- `On` — In this mode, the axes appear maximized in all displays. Any values entered into the **Title** and **Y-Axis label** parameters are hidden.
- `Off` — In this mode, none of the axes appear maximized.

See the "Visuals — Time Domain Options" on page 1-1174 section for more information.

Toolbar

The Signal Browser toolbar contains the following buttons.

Print Button

Button	Menu Location	Shortcut Keys	Description
	**File > Print**	**Ctrl+P**	Print the current scope window. To print the current scope window to a figure rather than sending it to your printer, select **File > Print to figure**.

Axes Control Buttons

	**Tools > Zoom In**	N/A	When this tool is active, you can zoom in on the scope window. To do so, click in the center of your area of interest, or click and drag your cursor to draw a rectangular area of interest inside the scope window.
	**Tools > Zoom X**	N/A	When this tool is active, you can zoom in on the *x*-axis. To do so, click inside the scope window, or click and drag your cursor along the *x*-axis over your area of interest.
	**Tools > Zoom Y**	N/A	When this tool is active, you can zoom in on the *y*-axis. To do so, click inside the scope window, or click and drag your cursor along the *y*-axis over your area of interest.
	**Tools > Pan**	N/A	When this tool is active, you can pan on the scope window. To do so, click in the center of your area of interest and drag your cursor to the left, right, up,

			or down, to move the position of the display.
	**Tools > Scale Axes Limits**	**Ctrl+A**	Click this button to scale the axes in the active scope window. Alternatively, you can enable automatic axes scaling by selecting one of the following options from the **Tools** menu: • **Automatically Scale Axes Limits** — When you select this option, the scope scales the axes as needed during simulation. • **Scale Axes Limits after 10 Updates** — When you select this option, the scope scales the axes after 10 updates. The scope does not scale the axes again during the simulation. • **Scale Axes Limits at Stop** — When you select this option, the scope scales the axes each time the simulation is stopped.

Measurements Buttons

	**Tools > Measurements > Cursor Measurements**	N/A	Open or close the **Cursor Measurements** panel. This panel puts screen cursors on all the displays. See the "Cursor Measurements Panel" on page 1-1153 section for more information.
	**Tools > Measurements > Signal Statistics**	N/A	Open or close the **Signal Statistics** panel. This panel displays the maximum, minimum, peak-to-peak difference, mean, median, RMS values of a selected signal, and the times at which the maximum and minimum occur. See the "Signal Statistics Panel" on page 1-1155 section for more information.
	**Tools > Measurements > Bilevel Measurements**	N/A	Open or close the **Bilevel Measurements** panel. This panel displays information about a selected signal's transitions, overshoots or undershoots, and cycles. See the "Bilevel Measurements Panel" on page 1-1156 section for more information.
	**Tools > Measurements > Peak Finder**	N/A	Open or close the **Peak Finder** panel. This panel displays maxima and the times at which they occur, allowing the settings for peak threshold, maximum number of peaks, and peak excursion to be modified.

			See the “Peak Finder Panel” on page 1-1171 section for more information.

Other Buttons

	**Tools > Play Selected Signal**	N/A	Play an audio signal. The function `soundsc` is used to play the signal.
	**View > Show All Legends**	N/A	Show a legend that matches each line style to a signal name in every display.
	**View > Layout**	N/A	Arrange the layout of displays in the Signal Browser. This feature allows you to tile your screen into a number of separate displays, up to a grid of 4 rows and 4 columns. You may find multiple displays useful when you select multiple input signals in SPTool. The default display is 1 row and 1 column. See the “Multiple Displays” on page 1-1142 section for more information.

You can control whether this toolbar appears in the Signal Browser window. From the Signal Browser menu, select **View > Toolbar**.

Measurements Panels

The Measurements panels are the five panels that appear at the right side of the Signal Browser. These panels are labeled **Trace selection**, **Cursor measurements**, **Signal statistics**, **Bilevel measurements**, and **Peak finder**.

Measurements Panel Buttons

Each of the Measurements panels contains the following buttons that enable you to modify the appearance of the current panel.

Button	Description
	Move the current panel to the top. When you are displaying more than one panel, this action moves the current panel above all the other panels.
	Collapse the current panel. When you first enable a panel, by default, it displays one or more of its panes. Click this button to hide all of its panes to conserve space. After you click this button, it becomes the expand button  .
	Expand the current panel. This button appears after you click the collapse button to hide the panes in the current panel. Click this button to display the panes in the current panel and show measurements again. After you click this button, it becomes the collapse button  again.
	Undock the current panel. This button lets you move the current panel into a separate window that can be relocated anywhere on your screen. After you click this button, it becomes the dock button  in the new window.
	Dock the current panel. This button appears only after you click the undock button. Click this button to put the current panel back into the right side of the Scope window. After you click this button, it becomes the undock button  again.
	Close the current panel. This button lets you remove the current panel from the right side of the Scope window.

Some panels have their measurements separated by category into a number of panes. Click the pane expand button  to show each pane that is hidden in the current panel. Click the pane collapse button  to hide each pane that is shown in the current panel.

Trace Selection Panel

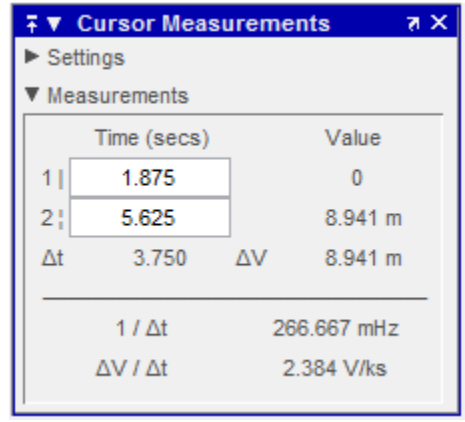
When you use the scope to view multiple signals, the Trace Selection panel appears if you have more than one signal displayed and you click on any of the other Measurements panels. The Measurements

panels display information about only the signal chosen in this panel. Choose the signal name for which you would like to display time domain measurements. See the following figure.

You can choose to hide or display the **Trace Selection** panel. In the Scope menu, select **Tools > Measurements > Trace Selection**.

Cursor Measurements Panel

The **Cursor Measurements** panel displays screen cursors. You can choose to hide or display the **Cursor Measurements** panel. In the Scope menu, select **Tools > Measurements > Cursor Measurements**. Alternatively, in the Scope toolbar, click the Cursor Measurements button.



Settings Pane

The **Settings** pane enables you to modify the type of screen cursors used to calculate time and value measurements.

- **Screen Cursors**— Shows screen cursors.
- **Horizontal**— Shows horizontal screen cursors.
- **Vertical**— Shows vertical screen cursors.
- **Waveform Cursors**— Shows cursors that attach to the input signals.
- **Lock Cursor Spacing**— Locks the time difference between the two cursors.

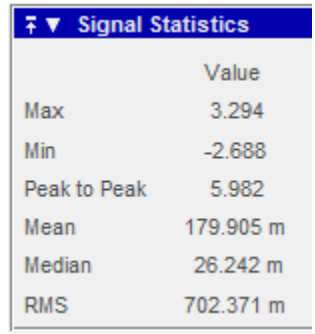
Measurements Pane

The **Measurements** pane shows the time and value measurements.

- **1 |**— Shows or enables you to modify the time or value at cursor number one, or both.
- **2 :**— Shows or enables you to modify the time or value at cursor number two, or both.
- **Δt**— Shows the absolute value of the difference in the times between cursor number one and cursor number two.
- **ΔV**— Shows the absolute value of the difference in signal amplitudes between cursor number one and cursor number two.
- **1/Δt**— Shows the rate, the reciprocal of the absolute value of the difference in the times between cursor number one and cursor number two.
- **ΔV/Δt**— Shows the scope, the ratio of the absolute value of the difference in signal amplitudes between cursors to the absolute value of the difference in the times between cursors.

Signal Statistics Panel

The **Signal Statistics** panel displays the maximum, minimum, peak-to-peak difference, mean, median, and RMS values of a selected signal. It also shows the *x*-axis indices at which the maximum and minimum values occur. You can choose to hide or display the **Signal Statistics** panel. In the Scope menu, select **Tools > Measurements > Signal Statistics**. Alternatively, in the scope toolbar, click the Signal Statistics  button.



Signal Statistics Measurements

The **Signal Statistics** panel shows statistics about the portion of the input signal within the *x*-axis and *y*-axis limits of the active display. The statistics shown are:

- **Max** — Shows the maximum or largest value within the displayed portion of the input signal. For more information on the algorithm this measurement uses, see the MATLAB `max` function reference.
- **Min** — Shows the minimum or smallest value within the displayed portion of the input signal. For more information on the algorithm this measurement uses, see the MATLAB `min` function reference.
- **Peak to Peak** — Shows the difference between the maximum and minimum values within the displayed portion of the input signal. For

more information on the algorithm this measurement uses, see the Signal Processing Toolbox `peak2peak` function reference.

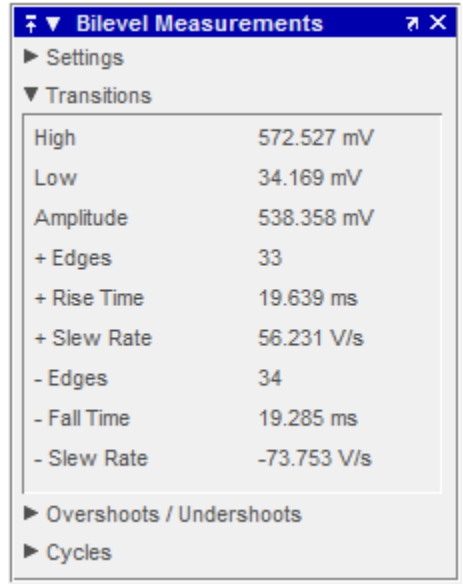
- **Mean** — Shows the average or mean of all the values within the displayed portion of the input signal. For more information on the algorithm this measurement uses, see the MATLAB `mean` function reference.
- **Median** — Shows the median value within the displayed portion of the input signal. For more information on the algorithm this measurement uses, see the MATLAB `median` function reference.
- **RMS** — Shows the difference between the maximum and minimum values within the displayed portion of the input signal. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `rms` function reference.

When you use the zoom options in the Scope, the Signal Statistics measurements automatically adjust to the time range shown in the display. In the Scope toolbar, click the **Zoom In** or **Zoom X** button to constrict the *x*-axis range of the display, and the statistics shown reflect this time range. For example, you can zoom in on one pulse to make the **Signal Statistics** panel display information about only that particular pulse.

The Signal Statistics measurements are valid for any units of the input signal. The letter after the value associated with each measurement represents the appropriate International System of Units (SI) prefix, such as *m* for *milli-*. For example, if the input signal is measured in volts, an *m* next to a measurement value indicates that this value is in units of millivolts.

Bilevel Measurements Panel

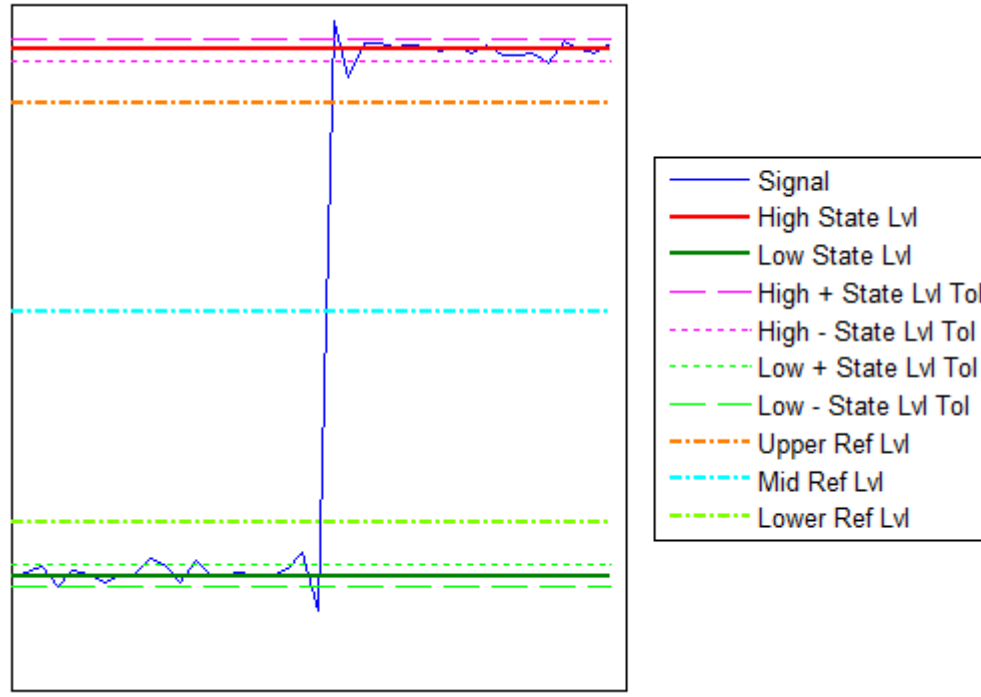
The **Bilevel Measurements** panel shows information about a selected signal's transitions, overshoots or undershoots, and cycles. You can choose to hide or display the **Bilevel Measurements** panel. In the Scope menu, select **Tools > Measurements > Bilevel Measurements**. Alternatively, in the Scope toolbar, you can select the Bilevel Measurements  button.



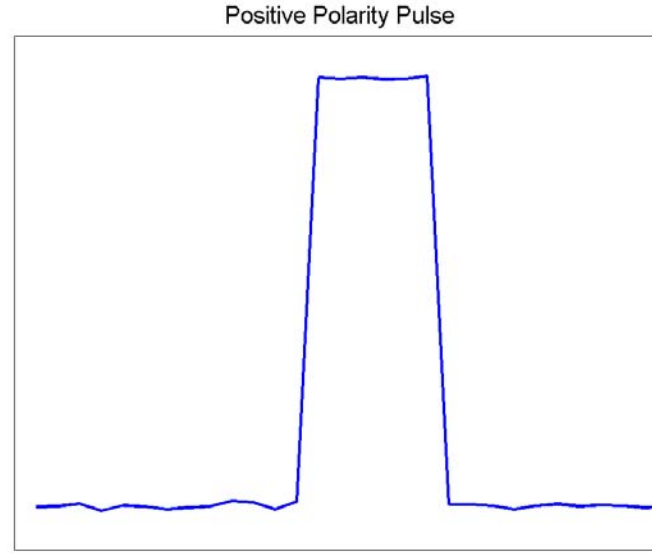
The **Bilevel Measurements** panel is separated into four panes, labeled **Settings**, **Transitions**, **Overshoots / Undershoots**, and **Cycles**. You can expand each pane to see the available options.

Settings Pane

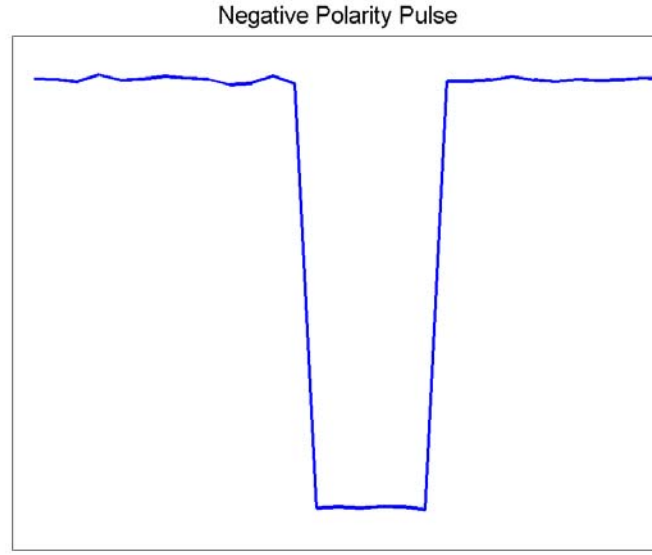
The **Settings** pane enables you to modify the properties used to calculate various measurements involving transitions, overshoots, undershoots, and cycles. You can modify the high-state level, low-state level, state-level tolerance, upper-reference level, mid-reference level, and lower-reference level, as shown in the following figure.



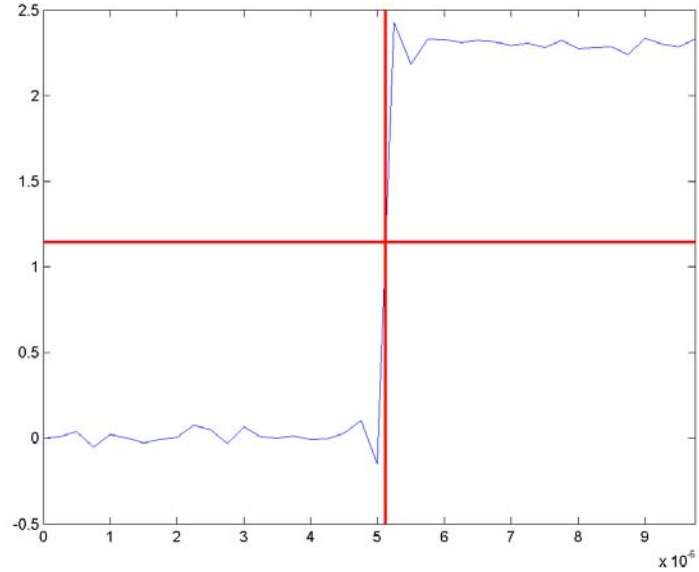
- **Auto State Level** — When this check box is selected, the Bilevel measurements panel autodetects the high- and low- state levels of a bilevel waveform. For more information on the algorithm this option uses, see the Signal Processing Toolbox `statelevels` function reference. When this check box is cleared, you may enter in values for the high- and low- state levels manually.
  - **High** — Used to manually specify the value that denotes a positive polarity, or high-state level, as shown in the following figure.



- **Low** — Used to manually specify the value that denotes a negative polarity, or low-state level, as shown in the following figure.



- **State Level Tolerance** — Tolerance within which the initial and final levels of each transition must be within their respective state levels. This value is expressed as a percentage of the difference between the high- and low-state levels.
- **Upper Ref Level** — Used to compute the end of the rise-time measurement or the start of the fall time measurement. This value is expressed as a percentage of the difference between the high- and low-state levels.
- **Mid Ref Level** — Used to determine when a transition occurs. This value is expressed as a percentage of the difference between the high- and low- state levels. In the following figure, the mid-reference level is shown as the horizontal line, and its corresponding mid-reference level instant is shown as the vertical line.

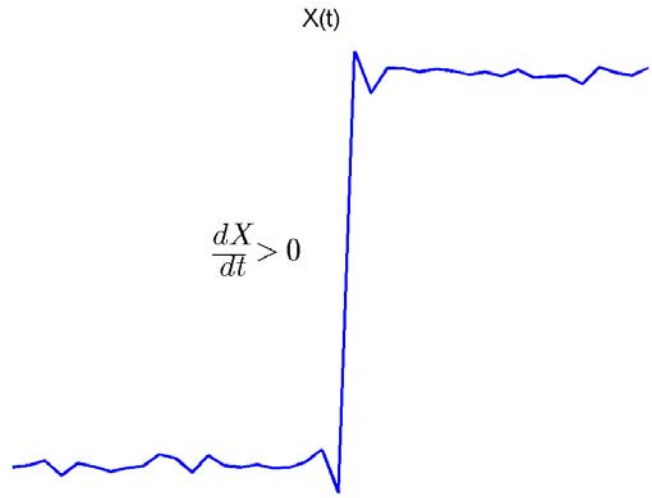


- **Lower Ref Level** — Used to compute the end of the fall-time measurement or the start of the rise-time measurement. This value is expressed as a percentage of the difference between the high- and low-state levels.
- **Settle Seek** — The duration after the mid-reference level instant when each transition occurs used for computing a valid settling time. This value is equivalent to the input parameter, `D`, which you can set when you run the `settlingtime` function. The settling time is displayed in the **Overshoots/Undershoots** pane.

Transitions Pane

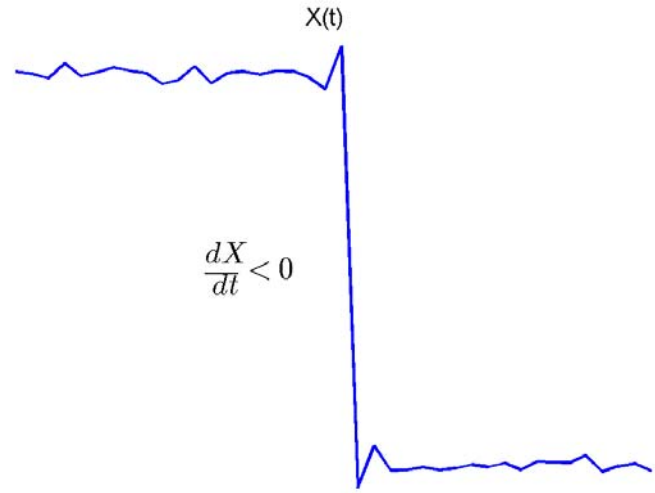
The **Transitions** pane displays calculated measurements associated with the input signal changing between its two possible state level values, high and low.

A positive-going transition, or *rising edge*, in a bilevel waveform is a transition from the low-state level to the high-state level. A positive-going transition has a slope value greater than zero. The following figure shows a positive-going transition.



Whenever there is a plus sign (+) next to a text label, this symbol refers to measurement associated with a rising edge, a transition from a low-state level to a high-state level.

A negative-going transition, or falling edge, in a bilevel waveform is a transition from the high-state level to the low-state level. A negative-going transition has a slope value less than zero. The following figure shows a negative-going transition.



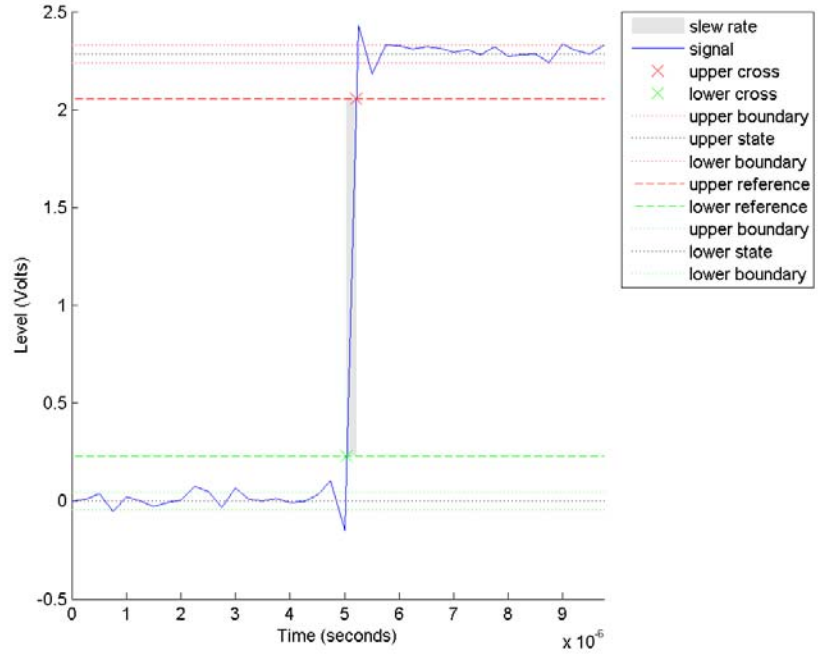
Whenever there is a minus sign (–) next to a text label, this symbol refers to measurement associated with a falling edge, a transition from a high-state level to a low-state level.

The Transition measurements assume that the amplitude of the input signal is in units of volts. You must convert all input signals to volts for the Transition measurements to be valid.

- **High** — The high-amplitude state level of the input signal over the duration of the **Time Span** parameter. You can set **Time Span** in the **Main** pane of the Visuals:Time Domain Options dialog box. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `statelevels` function reference.
- **Low** — The low-amplitude state level of the input signal over the duration of the **Time Span** parameter. You can set **Time Span** in the **Main** pane of the Visuals:Time Domain Options dialog box. For

more information on the algorithm this measurement uses, see the Signal Processing Toolbox `statelevels` function reference.

- **Amplitude** — Difference in amplitude between the high-state level and the low-state level.
- **+ Edges** — Total number of positive-polarity, or rising, edges counted within the displayed portion of the input signal.
- **+ Rise Time** — Average amount of time required for each rising edge to cross from the lower-reference level to the upper-reference level. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `risetime` function reference.
- **+ Slew Rate** — Average slope of each rising-edge transition line within the upper- and lower-percent reference levels in the displayed portion of the input signal. The region in which the slew rate is calculated appears in gray in the following figure.



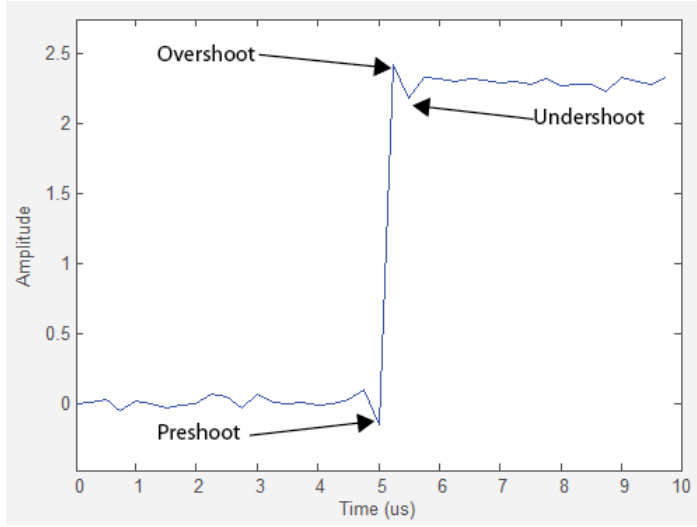
For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `slewrate` function reference.

- – **Edges** — Total number of negative-polarity or falling edges counted within the displayed portion of the input signal.
- – **Fall Time** — Average amount of time required for each falling edge to cross from the upper-reference level to the lower-reference level. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `falltime` function reference.
- – **Slew Rate** — Average slope of each falling edge transition line within the upper- and lower-percent reference levels in the displayed portion of the input signal. For more information on the algorithm

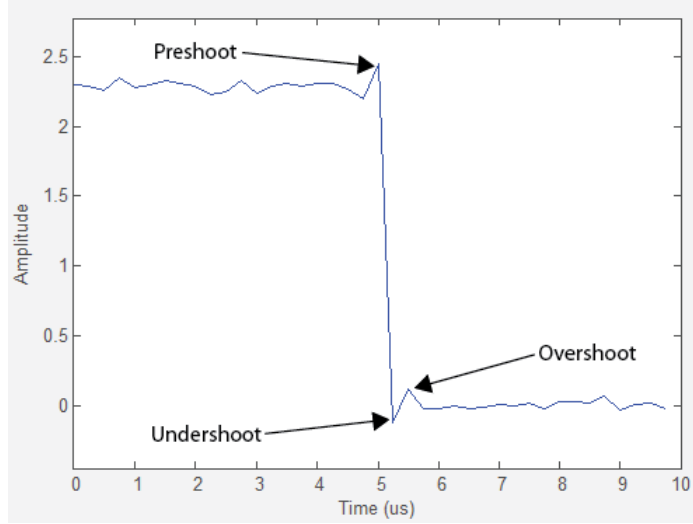
this measurement uses, see the Signal Processing Toolbox `slewrate` function reference.

Overshoots/Undershoots

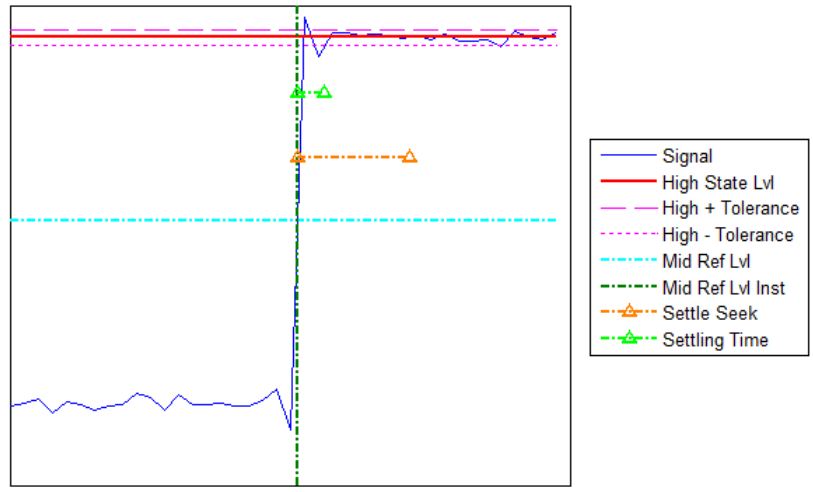
The **Overshoots/Undershoots** pane displays calculated measurements involving the distortion and damping of the input signal. *Overshoot* and *undershoot* refer to the amount that a signal respectively exceeds and falls below its final steady-state value. *Preshoot* refers to the amount prior to a transition that a signal varies from its initial steady-state value. This figure shows preshoot, overshoot, and undershoot for a rising-edge transition.



The next figure shows preshoot, overshoot, and undershoot for a falling-edge transition.



- **+ Preshoot** — Average lowest aberration in the region immediately preceding each rising transition.
- **+ Overshoot** — Average highest aberration in the region immediately following each rising transition. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `overshoot` function reference.
- **+ Undershoot** — Average lowest aberration in the region immediately following each rising transition. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `undershoot` function reference.
- **+ Settling Time** — Average time required for each rising edge to enter and remain within the tolerance of the high-state level for the remainder of the settle seek duration. The settling time is the time after the mid-reference level instant when the signal crosses into and remains in the tolerance region around the high-state level. This crossing is illustrated in the following figure.



You can modify the settle seek duration parameter in the **Settings** pane. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `settlingtime` function reference.

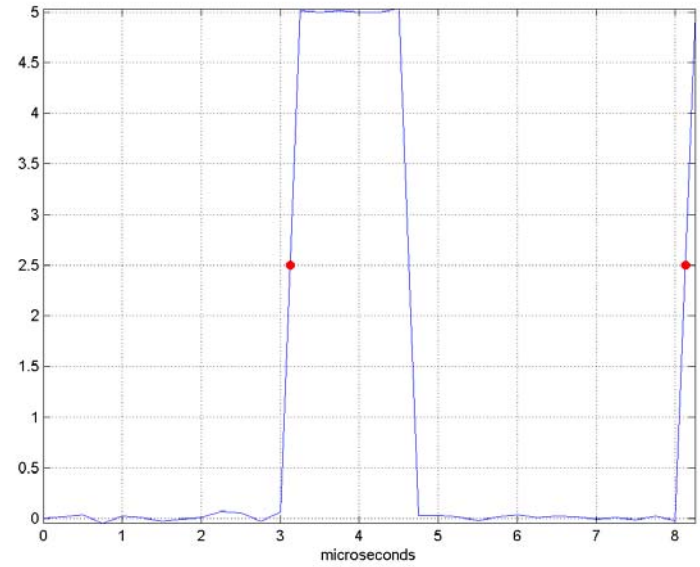
- – **Preshoot** — Average highest aberration in the region immediately preceding each falling transition.
- – **Overshoot** — Average highest aberration in the region immediately following each falling transition. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `overshoot` function reference.
- – **Undershoot** — Average lowest aberration in the region immediately following each falling transition. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `undershoot` function reference.
- – **Settling Time** — Average time required for each falling edge to enter and remain within the tolerance of the low-state level for the remainder of the settle seek duration. The settling time is the time after the mid-reference level instant when the signal crosses into and remains in the tolerance region around the low-state level. You can

modify the settle seek duration parameter in the **Settings** pane. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `settlingtime` function reference.

Cycles

The **Cycles** pane displays calculated measurements pertaining to repetitions or trends in the displayed portion of the input signal.

- **Period** — Average duration between adjacent edges of identical polarity within the displayed portion of the input signal. The Bilevel measurements panel calculates period as follows. It takes the difference between the mid-reference level instants of the initial transition of each positive-polarity pulse and the next positive-going transition. These mid-reference level instants appear as red dots in the following figure.



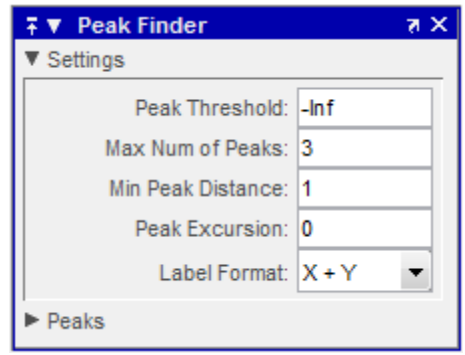
For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `pulseperiod` function reference.

- **Frequency** — Reciprocal of the average period. Whereas period is typically measured in some metric form of seconds, or seconds per cycle, frequency is typically measured in hertz or cycles per second.
- **+ Pulses** — Number of positive-polarity pulses counted.
- **+ Width** — Average duration between rising and falling edges of each positive-polarity pulse within the displayed portion of the input signal. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `pulsewidth` function reference.
- **+ Duty Cycle** — Average ratio of pulse width to pulse period for each positive-polarity pulse within the displayed portion of the input signal. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `dutycycle` function reference.
- **– Pulses** — Number of negative-polarity pulses counted.
- **– Width** — Average duration between rising and falling edges of each negative-polarity pulse within the displayed portion of the input signal. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `pulsewidth` function reference.
- **– Duty Cycle** — Average ratio of pulse width to pulse period for each negative-polarity pulse within the displayed portion of the input signal. For more information on the algorithm this measurement uses, see the Signal Processing Toolbox `dutycycle` function reference.

When you use the zoom options in the Scope, the bilevel measurements automatically adjust to the time range shown in the display. In the Scope toolbar, click the **Zoom In** or **Zoom X** button to constrict the *x*-axis range of the display, and the statistics shown reflect this time range. For example, you can zoom in on one rising edge to make the **Bilevel Measurements** panel display information about only that particular rising edge. However, this feature does not apply to the **High** and **Low** measurements.

Peak Finder Panel

The **Peak Finder** panel displays maxima and the *x*-axis values at which they occur, allowing the settings for peak threshold, maximum number of peaks, and peak excursion to be modified. You can choose to hide or display the **Peak Finder** panel. In the scope menu, select **Tools > Measurements > Peak Finder**. Alternatively, in the scope toolbar, select the Peak Finder button.



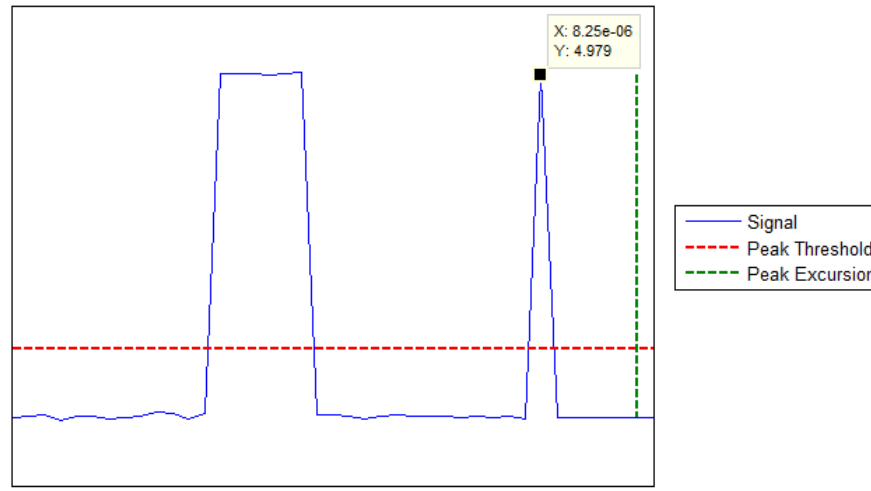
The **Peak finder** panel is separated into two panes, labeled **Settings** and **Peaks**. You can expand each pane to see the available options.

Settings Pane

The **Settings** pane enables you to modify the parameters used to calculate the peak values within the displayed portion of the input signal. For more information on the algorithms this pane uses, see the Signal Processing Toolbox `findpeaks` function reference.

- **Peak Threshold** — The level above which peaks are detected. This setting is equivalent to the `MINPEAKHEIGHT` parameter, which you can set when you run the `findpeaks` function.
- **Max Num of Peaks** — The maximum number of peaks to show. The value you enter must be a scalar integer between 1 and 99. This setting is equivalent to the `NPEAKS` parameter, which you can set when you run the `findpeaks` function.

- **Min Peaks Distance** — The minimum number of samples between adjacent peaks. This setting is equivalent to the `MINPEAKDISTANCE` parameter, which you can set when you run the `findpeaks` function.
- **Peak Excursion** — The minimum height difference between a peak and its neighboring samples. Peak excursion is illustrated alongside peak threshold in the following figure.



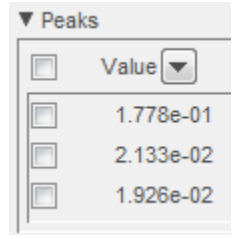
The *peak threshold* is a minimum value necessary for a sample value to be a peak. The *peak excursion* is the minimum difference between a peak sample and the samples to its left and right in the time domain. In the figure, the green vertical line illustrates the lesser of the two height differences between the labeled peak and its neighboring samples. This height difference must be greater than the **Peak Excursion** value for the labeled peak to be classified as a peak. Compare this setting to peak threshold, which is illustrated by the red horizontal line. The amplitude must be above this horizontal line for the labeled peak to be classified as a peak.

The peak excursion setting is equivalent to the `THRESHOLD` parameter, which you can set when you run the `findpeaks` function.

- **Label Format** — The coordinates to display next to the calculated peak values on the plot. To see peak values, you must first expand the **Peaks** pane and select the check boxes associated with individual peaks of interest. By default, both *x*-axis and *y*-axis values are displayed on the plot. Select which axes values you want to display next to each peak symbol on the display.
  - `X+Y` — Display both *x*-axis and *y*-axis values.
  - `X` — Display only *x*-axis values.
  - `Y` — Display only *y*-axis values.

Peaks Pane

The **Peaks** pane displays all of the largest calculated peak values. It also shows the coordinates at which the peaks occur, using the parameters you define in the **Settings** pane. You set the **Max Num of Peaks** parameter to specify the number of peaks shown in the list.



The numerical values displayed in the **Value** column are equivalent to the `pks` output argument returned when you run the `findpeaks` function. The numerical values displayed in the second column are similar to the `locs` output argument returned when you run the `findpeaks` function.

The Peak Finder displays the peak values in the **Peaks** pane. By default, the **Peak Finder** panel displays the largest calculated peak values in the **Peaks** pane in decreasing order of peak height. Use the sort descending button ( ) to rearrange the category and order by which Peak Finder displays peak values. Click this button again to

sort the peaks in ascending order instead. When you do so, the arrow changes direction to become the sort ascending button (). A filled sort button indicates that the peak values are currently sorted by this category. If the sort button is not filled (), then the peak values are sorted by the other category. The **Max Num of Peaks** parameter still controls the number of peaks listed.

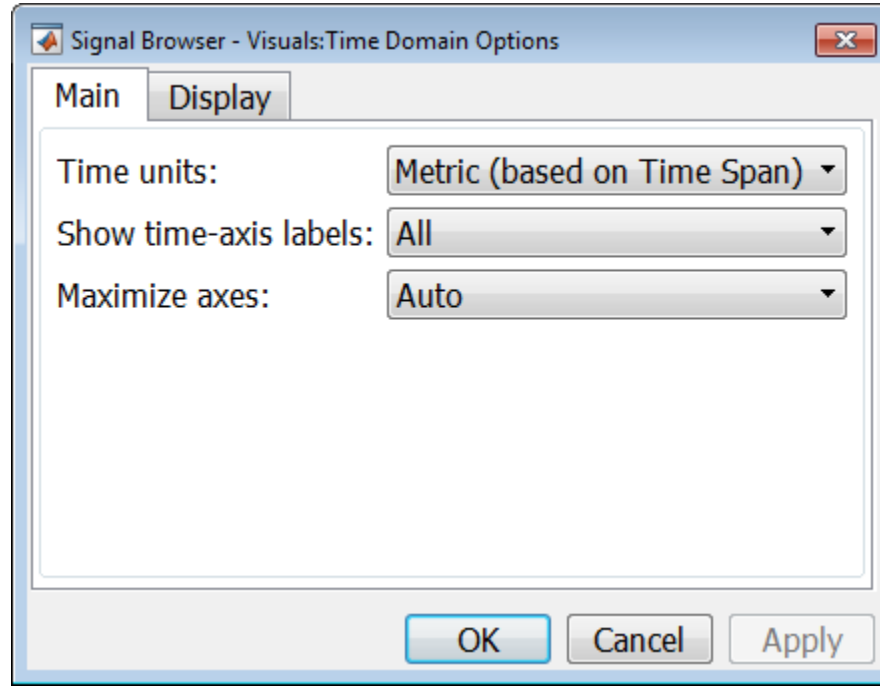
Use the check boxes to control which peaks are shown on the display. By default, all check boxes are cleared and the **Peak Finder** panel hides all the peaks. To show all the peak values on the display, select the check box in the top-left corner of the **Peaks** pane. To hide all the peak values on the display, clear this check box. To show an individual peak, select the check box directly to the left of its **Value** listing. To hide an individual peak, clear the check box directly to the left of its **Value** listing.

Visuals — Time Domain Options

The Visuals:Time Domain Options dialog box controls the visual configuration settings of the Scope displays. From the Scope menu, select **View > Properties** to open this dialog box.

Main Pane

The **Main** pane of the Visuals:Time Domain Options dialog box appears as follows.



Time units

Specify the units used to describe the *time*-axis. The default setting is `Metric`. You can select one of the following options:

- `Metric` — In this mode, the Scope converts the times on the *time*-axis to some metric units such as milliseconds, microseconds, days, etc. The Scope chooses the appropriate metric units, based on the minimum *time*-axis limit and the maximum *time*-axis limit of the scope window.
- `Seconds` — In this mode, the Scope always displays the units on the *time*-axis as seconds.
- `None` — In this mode, the Scope displays no units on the *time*-axis. The Scope shows only the word `Time` on the *time*-axis.

This parameter is Tunable.

Show time-axis labels

Specify how to display the time units used to describe the *time*-axis. The default setting is `All`. You can select one of the following options:

- `All` — In this mode, the *time*-axis labels appear in all displays.
- `None` — In this mode, the *time*-axis labels do not appear in the displays.
- `Bottom Displays Only` — In this mode, the *time*-axis labels appear in only the bottom row of the displays.

Tunable.

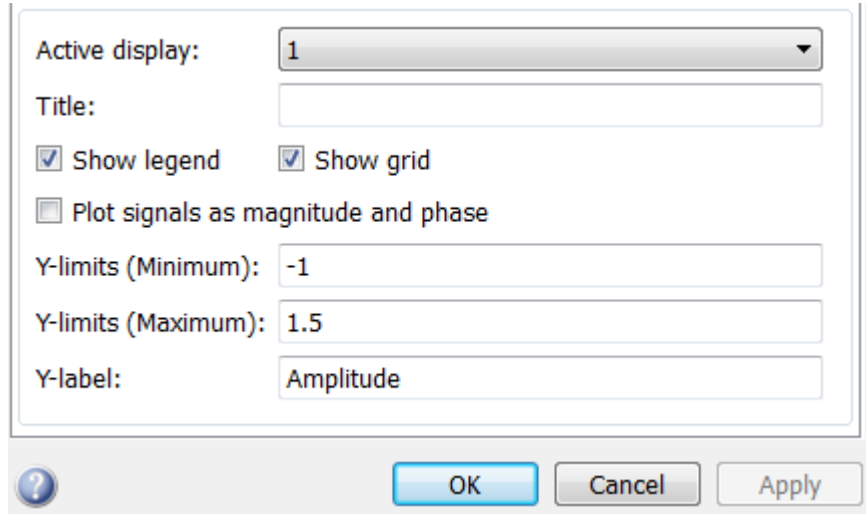
Maximize axes

Specify whether to display the Scope in maximized axes mode. In this mode, each of the axes are expanded to fit into the entire display. In each display, there is no space to show labels. Tick mark values are shown on top of the plotted data. The default setting is `Auto`. You can select one of the following options:

- `Auto` — In this mode, the axes appear maximized in all displays only if the **Title** and **Y-Axis label** parameters are empty for every display. If you enter any value in any display for either of these parameters, the axes are not maximized.
- `On` — In this mode, the axes appear maximized in all displays. Any values entered into the **Title** and **Y-Axis label** parameters are hidden.
- `Off` — In this mode, none of the axes appear maximized.

Display Pane

The **Display** pane of the Visuals—Time Domain Options dialog box appears as follows.



Active display

Specify the active display as an integer to get and set relevant properties. The number of a display corresponds to its column-wise placement index. Set this parameter to control which display should have its axes colors, line properties, marker properties, and visibility changed. This property is Tunable.

When you use the Layout option to tile the window into multiple displays, the display highlighted in blue is referred to as the *active display*. The default setting is `1`.

Title

Specify the active display title as a string. By default, the active display has no title. Tunable.

Show legend

Select this check box to show the legend in the display. The channel legend displays a name for each channel of each input signal. When the legend appears, you can place it anywhere inside of the scope window. To turn the legend off, clear the **Show legend** check box. Tunable.

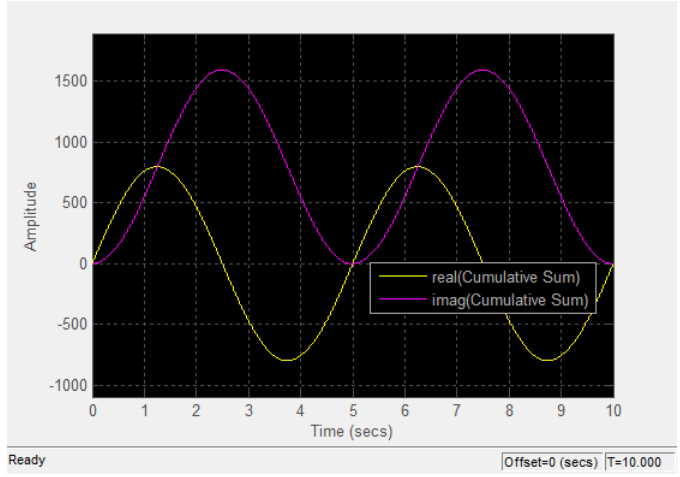
You can edit the name of any channel in the legend. To do so, double-click the current name, and enter a new channel name. By default, if the signal has multiple channels, the scope uses an index number to identify each channel of that signal. To change the appearance of any channel of any input signal in the scope window, from the scope menu, select **View > Style**.

Show grid

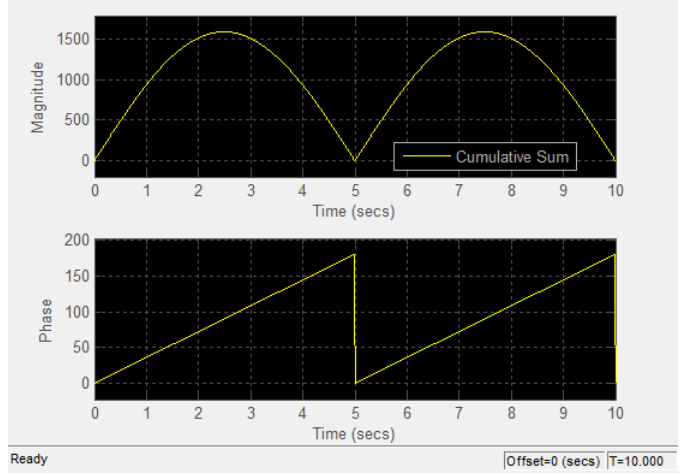
When you select this check box, a grid appears in the display of the scope figure. To hide the grid, clear this check box. Tunable.

Plot signals as magnitude and phase

When you select this check box, the scope splits the display into a magnitude plot and a phase plot. By default, this check box is cleared. If the input signal is complex valued, the scope plots the real and imaginary portions on the same axes. These real and imaginary portions appear as different-colored lines on the same axes, as shown in the following figure.



Selecting this check box and clicking the **Apply** or **OK** button changes the display. The magnitude of the input signal appears on the top axes and its phase, in degrees, appears on the bottom axes. See the following figure.



This feature is particularly useful for complex-valued input signals. If the input is a real-valued signal, selecting this check box always produces the same behavior. The phase is 0 for nonnegative input and 180 degrees for negative input. Tunable.

Y-limits (Minimum)

Specify the minimum value of the *y*-axis. Tunable.

When you select the **Plot signal(s) as magnitude and phase** check box, the value of this parameter always applies to the magnitude plot on the top axes. The phase plot on the bottom axes is always limited to a minimum value of –180 degrees.

Y-limits (Maximum)

Specify the maximum value of the *y*-axis. Tunable.

When you select the **Plot signal(s) as magnitude and phase** check box, the value of this parameter always applies to the magnitude plot on the top axes. The phase plot on the bottom axes is always limited to a maximum value of 180 degrees.

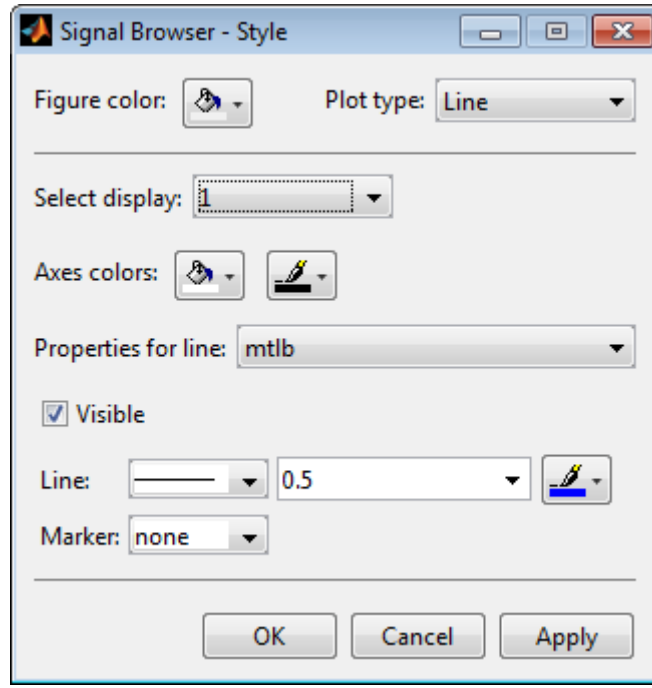
Y-label

Specify as a string the text for the scope to display to the left of the *y*-axis. This property is Tunable.

This parameter becomes invisible when you select the **Plot signal(s) as magnitude and phase** check box. When you enable that parameter, the *y*-axis label always appears as `Magnitude` on the top axes and `Phase` on the bottom axes.

Style Dialog Box

In the **Style** dialog box, you can customize the style of displays. You can change the color of the figure containing the displays, the background and foreground colors of display axes, and properties of lines in a display. From the Signal Browser menu, select **View > Style** to open this dialog box.



Properties

The **Style** dialog box allows you to modify the following properties of the Signal Browser:

Figure color

Specify the color that you want to apply to the background of the Signal Browser. By default, the figure color is gray.

Plot type

Specify the type of plot to use. The default setting is `Line`. Valid values for **Plot type** are:

- `Line` — Displays input signal as lines connecting each of the sampled values. This approach is similar to the functionality of the MATLAB `line` or `plot` function.
- `Stairs` — Displays input signal as a *stairstep* graph. A stairstep graph is made up of only horizontal lines and vertical lines. Each horizontal line represents the signal value for a discrete sample period and is connected to two vertical lines. Each vertical line represents a change in values occurring at a sample. This approach is equivalent to the MATLAB `stairs` function. Stairstep graphs are useful for drawing time history graphs of digitally sampled data.

This parameter is Tunable.

Select display

Specify the active display as a number, where a display number corresponds to the index of the input signal. The number of a display corresponds to its column-wise placement index. The default setting is `1`. Set this parameter to control which display should have its axes colors, line properties, marker properties, and visibility changed. Tunable.

Axes colors

Specify the color that you want to apply to the background of the axes for the active display.

Properties for line

Specify the signal for which you want to modify the visibility, line properties, and marker properties.

Visible

Specify whether the selected signal on the active display should be visible. If you clear this check box, the line disappears.

Line

Specify the line style, line width, and line color for the selected signal on the active display.

Marker

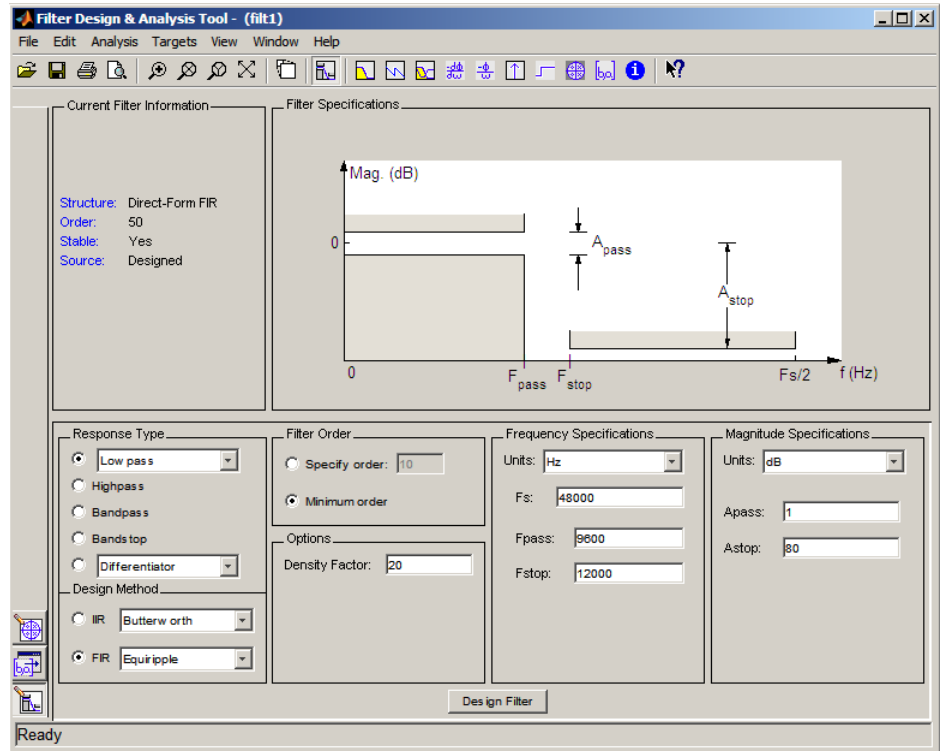
Specify marks for the selected signal on the active display to show at data points. This parameter is similar to the `Marker` property for the MATLAB Handle Graphics plot objects. You can choose any of the marker symbols from the following table.

Specifier	Marker Type
none	No marker (default)
○	Circle
□	Square
×	Cross
•	Point
+	Plus sign
*	Asterisk
◇	Diamond
▽	Downward-pointing triangle
△	Upward-pointing triangle
◁	Left-pointing triangle
▷	Right-pointing triangle

Specifier	Marker Type
☆	Five-pointed star (pentagram)
✡	Six-pointed star (hexagram)

Filter Design and Analysis Tool

The Filter Design and Analysis Tool `fdatool` allows you to design and edit FIR and IIR filters. To launch `fdatool`, press either the **New** button or the **Edit** button under the **Filters** list box in SPTool.

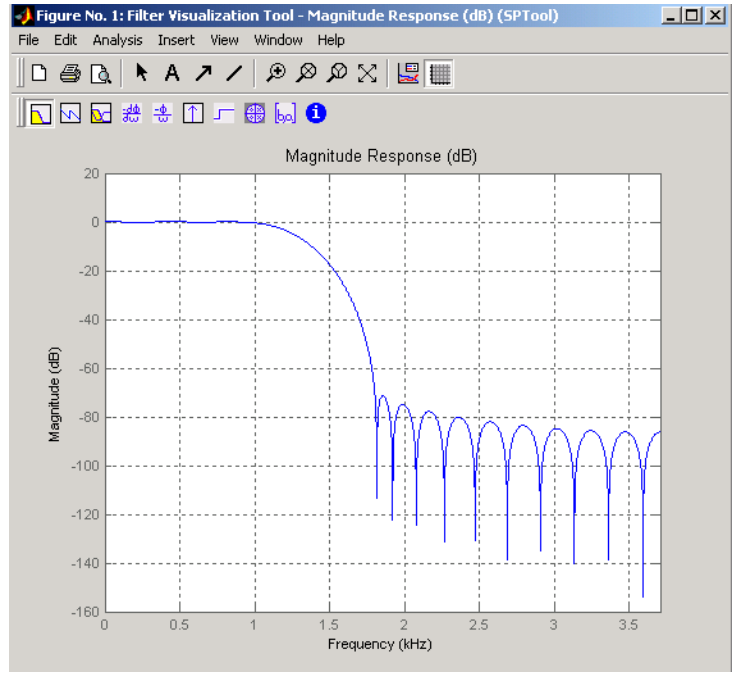


Note When you open FDATool from SPTool, a reduced version of FDATool that is compatible with SPTool opens.

The Filter Design and Analysis Tool has a Pole/Zero Editor you can access by selecting the  icon in the left column of FDATool.

Filter Visualization Tool

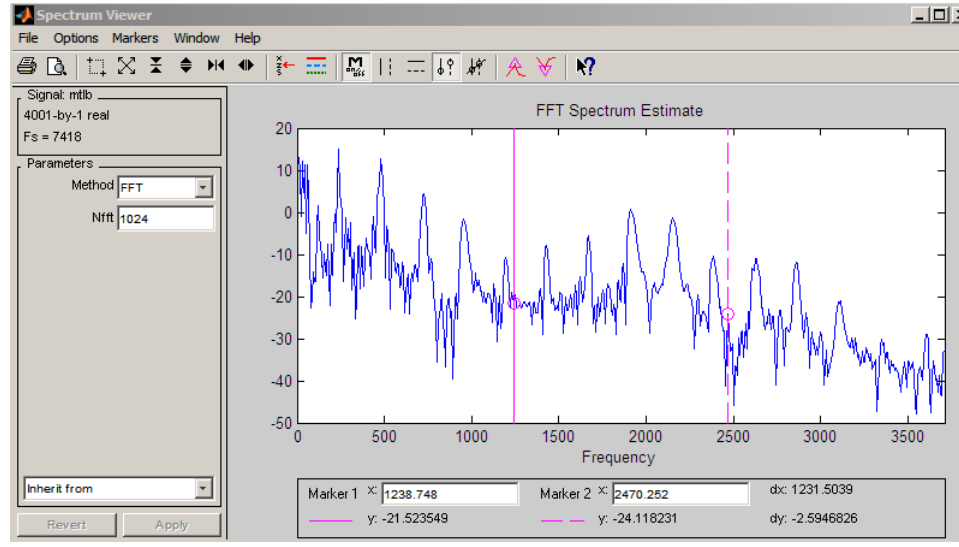
The Filter Visualization Tool (`fvtool`) allows you to view the characteristics of a designed or imported filter, including its magnitude response, phase response, group delay, phase delay, pole-zero plot, impulse response, and step response. To activate FVTool, click the **View** button under the **Filters** list box in SPTool.



Spectrum Viewer

The Spectrum Viewer allows you to analyze frequency-domain data graphically using a variety of methods of spectral density estimation, including the Burg method, the FFT method, the multitaper method, the MUSIC eigenvector method, Welch's method, and the Yule-Walker autoregressive method. To activate the Spectrum Viewer:

- Click the **Create** button under the **Spectra** list box to compute the power spectral density for a signal selected in the **Signals** list box in SPTool. You may need to click **Apply** to view the spectra.
- Click the **View** button to analyze spectra selected under the **Spectra** list box in SPTool.
- Click the **Update** button under the **Spectra** list box in SPTool to modify a selected power spectral density signal.



In addition, you can right-click in any plot display area to modify signal properties.

See Also

`fdatool` | `findpeaks` | `fvtool`

square

Purpose

Square wave

Syntax

```
x = square(t)
x = square(t,duty)
```

Description

`x = square(t)` generates a square wave with period 2π for the elements of time vector `t`. `square(t)` is similar to `sin(t)`, but creates a square wave with peaks of ±1 instead of a sine wave.

`x = square(t,duty)` generates a square wave with specified duty cycle, `duty`, which is a number between 0 and 100. The *duty cycle* is the percent of the period in which the signal is positive.

See Also

`chirp` | `cos` | `diric` | `gauspuls` | `pulstran` | `rectpuls` | `sawtooth` | `sin` | `tripuls`

Purpose

Convert digital filter state-space parameters to second-order sections form

Syntax

```
[sos,g] = ss2sos(A,B,C,D)
[sos,g] = ss2sos(A,B,C,D,iu)
[sos,g] = ss2sos(A,B,C,D,'order')
[sos,g] = ss2sos(A,B,C,D,iu,'order')
[sos,g] = ss2sos(A,B,C,D,iu,'order','scale')
sos = ss2sos(...)
```

Description

`ss2sos` converts a state-space representation of a given digital filter to an equivalent second-order section representation.

`[sos,g] = ss2sos(A,B,C,D)` finds a matrix `sos` in second-order section form with gain `g` that is equivalent to the state-space system represented by input arguments `A`, `B`, `C`, and `D`. The input system must be single output and real. `sos` is an L-by-6 matrix

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

whose rows contain the numerator and denominator coefficients b_{ik} and a_{ik} of the second-order sections of $H(z)$.

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

`[sos,g] = ss2sos(A,B,C,D,iu)` specifies a scalar `iu` that determines which input of the state-space system `A`, `B`, `C`, `D` is used in the conversion. The default for `iu` is 1.

`[sos,g] = ss2sos(A,B,C,D,'order')` and

`[sos,g] = ss2sos(A,B,C,D,iu,'order')` specify the order of the rows in `sos`, where `'order'` is

- `'down'`, to order the sections so the first row of `sos` contains the poles closest to the unit circle
- `'up'`, to order the sections so the first row of `sos` contains the poles farthest from the unit circle (default)

The zeros are always paired with the poles closest to them.

`[sos,g] = ss2sos(A,B,C,D,iu,'order','scale')` specifies the desired scaling of the gain and the numerator coefficients of all second-order sections, where `'scale'` is

- `'none'`, to apply no scaling (default)
- `'inf'`, to apply infinity-norm scaling
- `'two'`, to apply 2-norm scaling

Using infinity-norm scaling in conjunction with `up`-ordering minimizes the probability of overflow in the realization. Using 2-norm scaling in conjunction with `down`-ordering minimizes the peak round-off noise.

Note Infinity-norm and 2-norm scaling are appropriate only for direct-form II implementations.

`sos = ss2sos(...)` embeds the overall system gain, `g`, in the first section, $H_1(z)$, so that

$$H(z) = \prod_{k=1}^{L} H_k(z)$$

Note Embedding the gain in the first section when scaling a direct-form II structure is not recommended and may result in erratic scaling. To avoid embedding the gain, use `ss2sos` with two outputs.

Examples

Find a second-order section form of a Butterworth lowpass filter:

```
[A,B,C,D] = butter(5,0.2);
sos = ss2sos(A,B,C,D)
sos =
    0.0013   0.0013        0   1.0000  -0.5095        0
    1.0000   2.0008   1.0008   1.0000  -1.0966   0.3554
    1.0000   1.9979   0.9979   1.0000  -1.3693   0.6926
```

Algorithms

`ss2sos` uses a four-step algorithm to determine the second-order section representation for an input state-space system:

1. It finds the poles and zeros of the system given by `A`, `B`, `C`, and `D`.
2. It uses the function `zp2sos`, which first groups the zeros and poles into complex conjugate pairs using the `cplxpair` function. `zp2sos` then forms the second-order sections by matching the pole and zero pairs according to the following rules:
   a. Match the poles closest to the unit circle with the zeros closest to those poles.
   b. Match the poles next closest to the unit circle with the zeros closest to those poles.
   c. Continue until all of the poles and zeros are matched.

   `ss2sos` groups real poles into sections with the real poles closest to them in absolute value. The same rule holds for real zeros.
3. It orders the sections according to the proximity of the pole pairs to the unit circle. `ss2sos` normally orders the sections with poles closest to the unit circle last in the cascade. You can tell `ss2sos` to order the sections in the reverse order by specifying the `'down'` flag.
4. `ss2sos` scales the sections by the norm specified in the `'scale'` argument. For arbitrary $H(\omega)$, the scaling is defined by

$$\| H \|_p = \left[\frac{1}{2\pi} \int_0^{2\pi} | H(\omega) |^p \, d\omega \right]^{\frac{1}{p}}$$

where p can be either ∞ or 2. See the references for details. This scaling is an attempt to minimize overflow or peak round-off noise in fixed point filter implementations.

Diagnostics

If there is more than one input to the system, `ss2sos` gives the following error message:

```
State-space system must have only one input.
```

References

[1] Jackson, L.B., *Digital Filters and Signal Processing, 3rd ed.*, Kluwer Academic Publishers, Boston, 1996. Chapter 11.

[2] Mitra, S.K., *Digital Signal Processing: A Computer-Based Approach*, McGraw-Hill, New York, 1998. Chapter 9.

[3] Vaidyanathan, P.P., "Robust Digital Filter Structures," *Handbook for Digital Signal Processing*, S.K. Mitra and J.F. Kaiser, ed., John Wiley & Sons, New York, 1993, Chapter 7.

See Also

`cplxpair` | `sos2ss` | `ss2tf` | `ss2zp` | `tf2sos` | `zp2sos`

Purpose

Convert state-space filter parameters to transfer function form

Syntax

```
[b,a] = ss2tf(A,B,C,D,iu)
```

Description

`ss2tf` converts a state-space representation of a given system to an equivalent transfer function representation.

`[b,a] = ss2tf(A,B,C,D,iu)` returns the transfer function

$$H(s) = \frac{B(s)}{A(s)} = C(sI - A)^{-1}B + D$$

of the system

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

from the `iu`-th input. Vector `a` contains the coefficients of the denominator in descending powers of s. The numerator coefficients are returned in array `b` with as many rows as there are outputs y. `ss2tf` also works with systems in discrete time, in which case it returns the z-transform representation.

The `ss2tf` function is part of the standard MATLAB language.

Algorithms

The `ss2tf` function uses `poly` to find the characteristic polynomial $\det(sI–A)$ and the equality:

$$H(s) = C(sI - A)^{-1}B = \frac{\det(sI - A + BC) - \det(sI - A)}{\det(sI - A)}$$

See Also

`latc2tf` | `sos2tf` | `ss2sos` | `ss2zp` | `tf2ss` | `zp2tf`

Purpose

Convert state-space filter parameters to zero-pole-gain form

Syntax

```
[z,p,k] = ss2zp(A,B,C,D,i)
```

Description

`ss2zp` converts a state-space representation of a given system to an equivalent zero-pole-gain representation. The zeros, poles, and gains of state-space systems represent the transfer function in factored form.

`[z,p,k] = ss2zp(A,B,C,D,i)` calculates the transfer function in factored form

$$H(s) - \frac{Z(s)}{P(s)} - k\frac{(s-z_1)(s-z_2)\cdots(s-z_n)}{(s-p_1)(s-p_2)\cdots(s-p_n)}$$

of the continuous-time system

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

from the `i`th input (using the `i`th columns of `B` and `D`). The column vector `p` contains the pole locations of the denominator coefficients of the transfer function. The matrix `z` contains the numerator zeros in its columns, with as many columns as there are outputs y (rows in `C`). The column vector `k` contains the gains for each numerator transfer function.

`ss2zp` also works for discrete time systems. The input state-space system must be real.

The `ss2zp` function is part of the standard MATLAB language.

Examples

Here are two ways of finding the zeros, poles, and gains of a discrete-time transfer function:

$$H(z) = \frac{2 + 3z^{-1}}{1 + 0.4z^{-1} + z^{-2}}$$

```
b = [2 3 0];
```

```
a = [1 0.4 1];
[z,p,k] = tf2zp(b,a)
z =
    0.0000
   -1.5000
p =
    -0.2000 + 0.9798i
    -0.2000 - 0.9798i
k =
    2
[A,B,C,D] = tf2ss(b,a);
[z,p,k] = ss2zp(A,B,C,D,1)
z =
    0.0000
   -1.5000
p =
    -0.2000 + 0.9798i
    -0.2000 - 0.9798i
k =
    2
```

Algorithms

`ss2zp` finds the poles from the eigenvalues of the `A` array. The zeros are the finite solutions to a generalized eigenvalue problem:

```
z = eig([A B;C D], diag([ones(1,n) 0]);
```

In many situations this algorithm produces spurious large, but finite, zeros. `ss2zp` interprets these large zeros as infinite.

`ss2zp` finds the gains by solving for the first nonzero Markov parameters.

References

[1] Laub, A.J., and B.C. Moore, "Calculation of Transmission Zeros Using QZ Techniques," *Automatica 14* (1978), p. 557.

See Also

`sos2zp` | `ss2sos` | `ss2tf` | `tf2zp` | `tf2zpk` | `zp2ss`

statelevels

Purpose

State-level estimation for bilevel waveform with histogram method

Syntax

```
LEVELS = statelevels(X)
LEVELS = statelevels(X,NBINS)
LEVELS = statelevels(X,NBINS,METHOD)
[LEVELS,HISTOGRAM] = statelevels(...)
[LEVELS,HISTOGRAM,BINLEVELS] = statelevels(...)
statelevels(...)
```

Description

`LEVELS = statelevels(X)` estimates the low- and high-state levels in the bilevel waveform, `X`, using the histogram method. See "Algorithms" on page 1-1199.

`LEVELS = statelevels(X,NBINS)` specifies the number of bins to use in the histogram as a positive scalar. If unspecified, `NBINS` defaults to 100.

`LEVELS = statelevels(X,NBINS,METHOD)` estimates state levels using the mean or mode of the subhistograms. Valid entries for `METHOD` are `'mean'` or `'mode'`. `METHOD` defaults to `'mode'`. See "Algorithms" on page 1-1199.

`[LEVELS,HISTOGRAM] = statelevels(...)` returns the histogram, `HISTOGRAM`, of the values in `X`.

`[LEVELS,HISTOGRAM,BINLEVELS] = statelevels(...)` returns the centers of the histogram bins.

`statelevels(...)` displays a plot of the signal and the corresponding computed histogram.

Input Arguments

X

Bilevel waveform. `X` is a real-valued row or column vector.

NBINS

Number of histogram bins

Default: 100

METHOD

State-level estimation method in the subhistograms. METHOD is a string indicating the statistic to use for the estimation of the low- and high-state levels. Valid entries for METHOD are 'mode' or 'mean'. See "Algorithms" on page 1-1199.

Default: 'mode'

Output Arguments

LEVELS

Levels of low and high states. LEVELS is a 1-by-2 row vector of state levels estimated by the histogram method. The first element of LEVELS is the low-state level. The second element of LEVELS is the high-state level.

HISTOGRAM

Histogram counts (frequencies). HISTOGRAM is a column vector with NBINS elements containing the number of data values in each histogram bin.

BINLEVELS

Histogram bin centers. BINLEVELS is a column vector containing the bin centers for the histogram counts in HISTOGRAM.

Definitions

State

A particular level, which can be associated with an upper- and lower-state boundary. States are ordered from the most negative to the most positive. In a bilevel waveform, the most negative state is the low state. The most positive state is the high state.

State-Level Tolerances

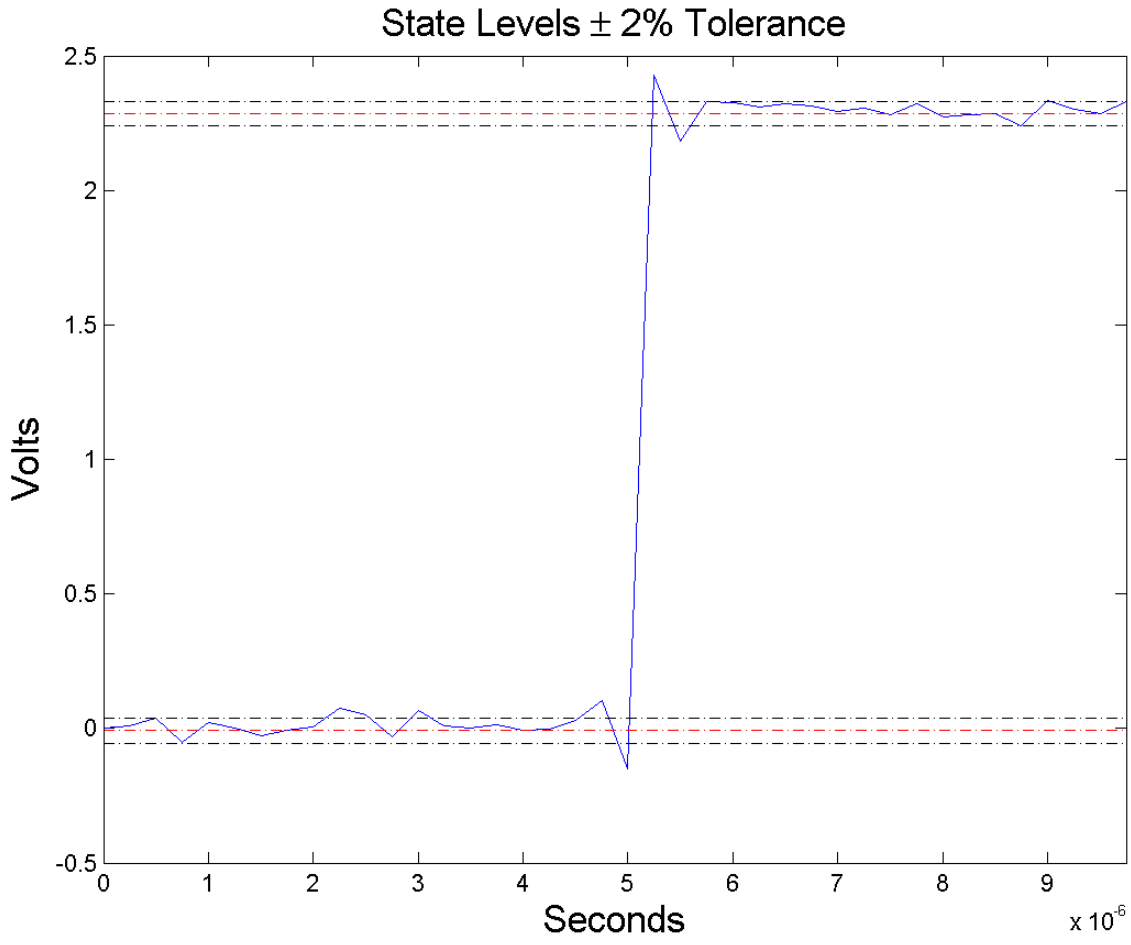
Each state level can have associated lower- and upper-state boundaries. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small

number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The red dashed lines indicate the estimated state levels.



Examples

Display State Levels and Subhistograms

Estimate the low- and high-state levels of 2.3 V underdamped clock data. Plot the data with the estimated state levels and subhistograms.

```
load('clockex.mat', 'x');
statelevels(x);
```

State Levels with 100 Bins and Modes of Subhistograms

Estimate the low and high-state levels of 2.3 V underdamped clock data sampled at 4 MHz.

Use the default number of bins and modes of the subhistograms to estimate the state levels. Plot the clock data with the lines indicating the estimated low and high-state levels.

```
load('clockex.mat', 'x', 't');
LEVELS = statelevels(x);
plot(t,x);
hold on;
plot(t,LEVELS(1).*ones(length(x)),'r--');
plot(t,LEVELS(2).*ones(length(x)),'r--');
```

State Levels Using Means of Subhistograms

Estimate the low and high-state levels of 2.3 V underdamped clock data sampled at 4 MHz.

Use the default number of bins and means of the subhistograms to estimate the state levels. Plot the clock data with the lines indicating the estimated low and high-state levels.

```
load('clockex.mat', 'x', 't');
LEVELS = statelevels(x,1e3,'mean');
```

Histogram Counts and Histogram Bin Centers

Estimate the low- and high-state levels of 2.3 V underdamped clock data sampled at 4 MHz. Return the histogram counts and histogram bin centers used in the histogram method.

```
load('clockex.mat', 'x', 't');
[LEVELS,HISTOGRAM,BINLEVELS] = statelevels(x);
```

Algorithms

`statelevels` uses the histogram method to estimate the states of a bilevel waveform. The histogram method is described in [1]. To summarize the method:

1. Determine the maximum and minimum amplitudes and amplitude range of the data.
2. For the specified number of histogram bins, determine the bin width as the ratio of the amplitude range to the number of bins.
3. Sort the data values into the histogram bins.
4. Identify the lowest-indexed histogram bin, i_{low}, and highest-indexed histogram bin, i_{high}, with nonzero counts.
5. Divide the histogram into two subhistograms.

   The indices of the lower histogram bins are $i_{low} \le i \le 1/2(i_{high} — i_{low})$.

   The indices of the upper histogram bins are $i_{low} + 1/2(i_{high} - i_{low}) \le i \le i_{high}$.
6. Compute the state levels by determining the mode or mean of the lower and upper histograms.

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003, pp. 15–17.

See Also

`midcross` | `overshoot` | `risetime` | `undershoot`

stepz

Purpose

Step response of digital filter

Syntax

```
[h,t] = stepz(b,a)
[h,t] = stepz(sos)
[h,t] = stepz(Hd)
[h,t] = stepz(...,n)
[h,t] = stepz(...,n,fs)
stepz(...)
```

Description

`[h,t] = stepz(b,a)` returns the step response of the filter with numerator coefficients `b` and denominator coefficients `a`. `stepz` chooses the number of samples and returns the response in the column vector `h` and sample times in the column vector `t` (where `t = [0:n-1]'`, and `n = length(t)` is computed automatically).

Note If the input to `stepz` is single precision, the step response is calculated using single-precision arithmetic. The output, `h`, is single precision.

`[h,t] = stepz(sos)` returns the step response for the second order sections matrix, `sos`. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. If the number of sections is less than 2, `stepz` considers the input to be the numerator vector, `b`. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`[h,t] = stepz(Hd)` returns the step response for the `dfilt` filter object, `Hd`, or the array of `dfilt` filter objects. If `Hd` is an array of `dfilt` objects, each column of `h` is the step response of the corresponding `dfilt` object.

`[h,t] = stepz(...,n)` computes the first n samples of the step response when n is an integer (`t = [0:n-1]'`). I

`[h,t] = stepz(...,n,fs)` computes n samples and produces a vector `t` of length `n` so that the samples are spaced `1/fs` units apart. `fs` is assumed to be in Hz.

`stepz(...)` with no output arguments plots the step response of the filter. If you input the filter coefficients or second order sections matrix, the current figure window is used. If you input a `dfilt` object or array of filter objects, `fvtool` is used to plot the step response.

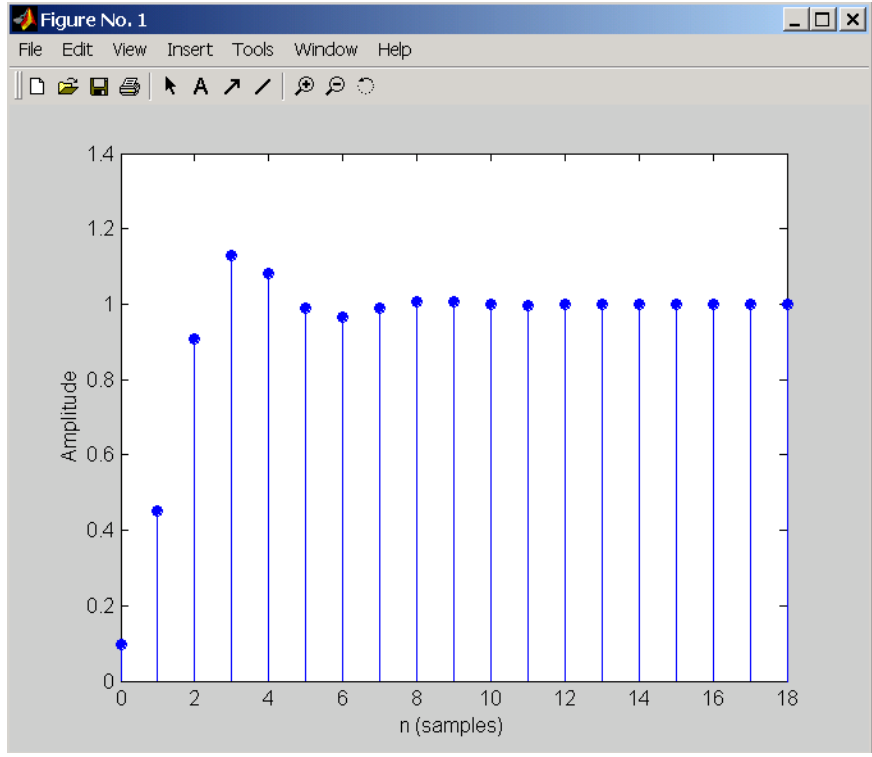
Note If you have the DSP System Toolbox product installed and are using a `dfilt` object with fixed-point properties, the filter internals are not used when calculating the step response.

Examples

Example 1

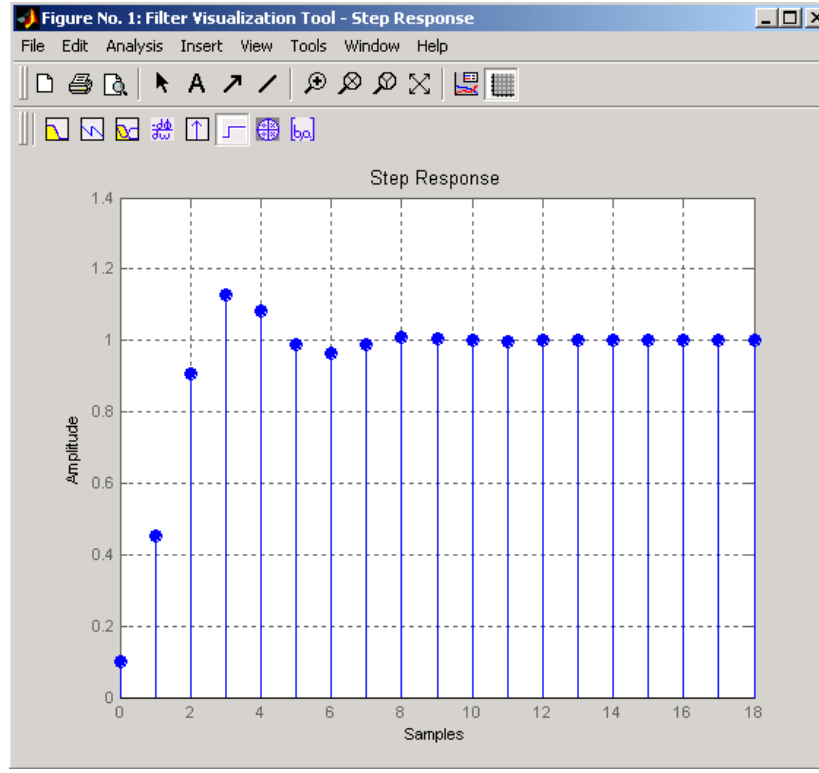
Plot the step response of a Butterworth filter:

```
[b,a] = butter(3,.4);
stepz(b,a)
```



The same example using a `dfilt` object and displaying the result in the Filter Visualization Tool (`fvtool`) is

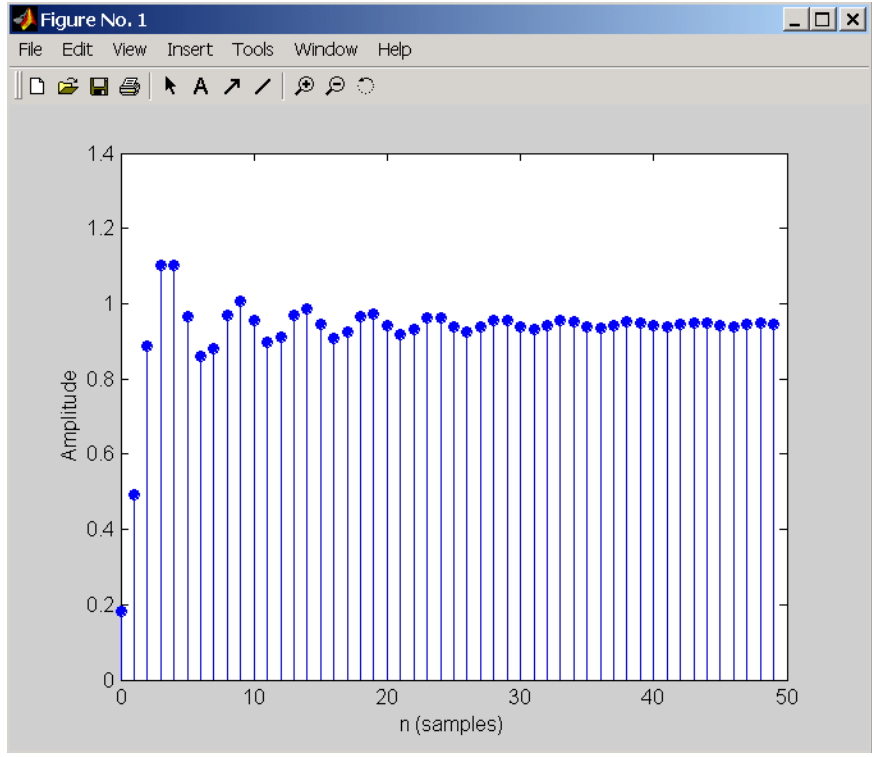
```
[b,a] = butter(3,.4);
Hd=dfilt.df1(b,a);
stepz(Hd)
```



Example 2

Plot the first 50 samples of the step response of a fourth-order lowpass elliptic filter with cutoff frequency of 0.4 times the Nyquist frequency:

```
[b,a] = ellip(4,0.5,20,0.4);
stepz(b,a,50)
```



The same example using a `dfilt` object and displaying the result in the Filter Visualization Tool (`fvtool`) is

```
[b,a] = ellip(4,0.5,20,0.4);
Hd=dfilt.df1(b,a);
stepz(Hd,50)
```

Algorithms

`stepz` filters a length `n` step sequence using

```
filter(b,a,ones(1,n))
```

and plots the results using `stem`.

To compute `n` in the auto-length case, `stepz` either uses `n = length(b)` for the FIR case or first finds the poles using `p = roots(a)`, if `length(a)` is greater than `1`.

If the filter is unstable, `n` is chosen to be the point at which the term from the largest pole reaches `10^6` times its original value.

If the filter is stable, `n` is chosen to be the point at which the term due to the largest amplitude pole is `5*10^-5` of its original amplitude.

If the filter is oscillatory (poles on the unit circle only), `stepz` computes five periods of the slowest oscillation.

If the filter has both oscillatory and damped terms, `n` is chosen to equal five periods of the slowest oscillation or the point at which the term due to the largest (nonunity) amplitude pole is `5*10^-5` of its original amplitude, whichever is greater.

`stepz` also allows for delays in the numerator polynomial. The number of delays is incorporated into the computation for the number of samples.

See Also

`freqz | grpdelay | impz | phasez | zplane`

Purpose

Compute linear model using Steiglitz-McBride iteration

Syntax

```
[b,a] = stmcb(h,nb,na)
[b,a] = stmcb(y,x,nb,na)
[b,a] = stmcb(h,nb,na,niter)
[b,a] = stmcb(y,x,nb,na,niter)
[b,a] = stmcb(h,nb,na,niter,ai)
[b,a] = stmcb(y,x,nb,na,niter,ai)
```

Description

Steiglitz-McBride iteration is an algorithm for finding an IIR filter with a prescribed time domain impulse response. It has applications in both filter design and system identification (parametric modeling).

`[b,a] = stmcb(h,nb,na)` finds the coefficients `b` and `a` of the system $b(z)/a(z)$ with approximate impulse response `h`, exactly `nb` zeros, and exactly `na` poles.

`[b,a] = stmcb(y,x,nb,na)` finds the system coefficients `b` and `a` of the system that, given `x` as input, has `y` as output. `x` and `y` must be the same length.

`[b,a] = stmcb(h,nb,na,niter)` and

`[b,a] = stmcb(y,x,nb,na,niter)` use `niter` iterations. The default for `niter` is 5.

`[b,a] = stmcb(h,nb,na,niter,ai)` and

`[b,a] = stmcb(y,x,nb,na,niter,ai)` use the vector `ai` as the initial estimate of the denominator coefficients. If `ai` is not specified, `stmcb` uses the output argument from `[b,ai] = prony(h,0,na)` as the vector `ai`.

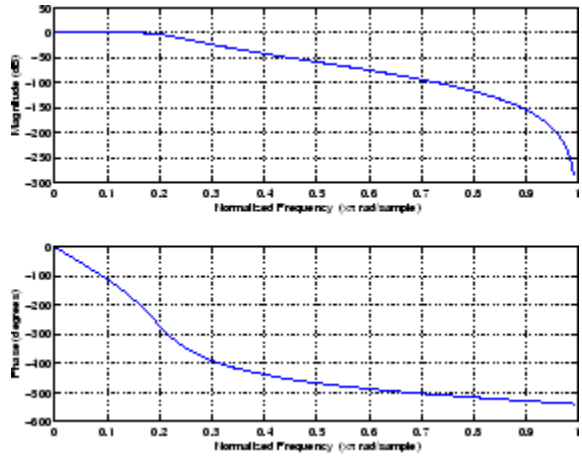
`stmcb` returns the IIR filter coefficients in length `nb+1` and `na+1` row vectors `b` and `a`. The filter coefficients are ordered in descending powers of z.

$$H(z) = \frac{B(z)}{A(z)} = \frac{b(1) + b(2)z^{-1} + \cdots + b(nb+1)z^{-nb}}{a(1) + a(2)z^{-1} + \cdots + a(na+1)z^{-na}}$$

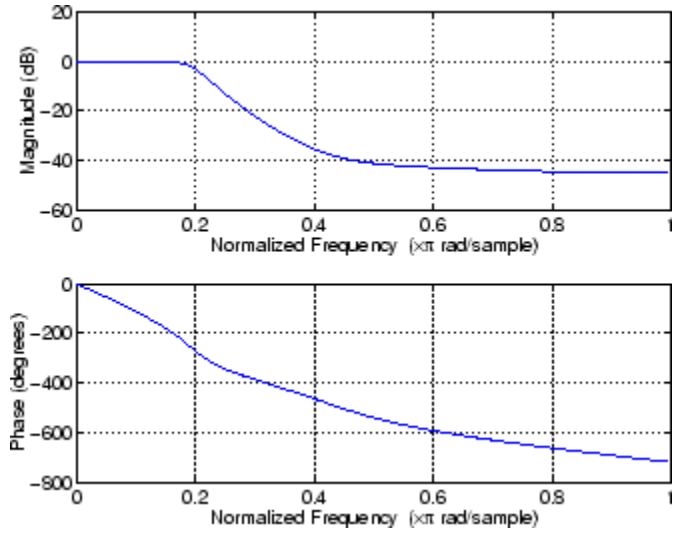
Examples

Approximate the impulse response of a Butterworth filter with a system of lower order:

```
[b,a] = butter(6,0.2);
h = filter(b,a,[1 zeros(1,100)]);
freqz(b,a,128)
```



```
[bb,aa] = stmcb(h,4,4);
freqz(bb,aa,128)
```



Algorithms

`stmcb` attempts to minimize the squared error between the impulse response h of $b(z)/a(z)$ and the input signal x.

$$\min_{a,b} \sum_{i=0}^{\infty} |x(i) - h(i)|^2$$

`stmcb` iterates using two steps:

1. It prefilters `h` and `x` using $1/a(z)$.
2. It solves a system of linear equations for `b` and `a` using `\`.

`stmcb` repeats this process `niter` times. No checking is done to see if the `b` and `a` coefficients have converged in fewer than `niter` iterations.

Diagnostics

If `x` and `y` have different lengths, `stmcb` produces this error message,

```
Input signal X and output signal Y must
have the same length.
```

References

[1] Steiglitz, K., and L.E. McBride, "A Technique for the Identification of Linear Systems," *IEEE Trans. Automatic Control, Vol. AC-10* (1965), pp. 461-464.

[2] Ljung, L., *System Identification: Theory for the Use*r, Prentice-Hall, Englewood Cliffs, NJ, 1987, p. 297.

See Also

`levinson` | `lpc` | `aryule` | `prony`

strips

Purpose

Strip plot

Syntax

```
strips(x)
strips(x,n)
strips(x,sd,fs)
strips(x,sd,fs,scale)
```

Description

`strips(x)` plots vector `x` in horizontal strips of length 250. If `x` is a matrix, `strips(x)` plots each column of `x`. The left-most column (column 1) is the top horizontal strip.

`strips(x,n)` plots vector `x` in strips that are each `n` samples long.

`strips(x,sd,fs)` plots vector `x` in strips of duration `sd` seconds, given a sampling frequency of `fs` samples per second.

`strips(x,sd,fs,scale)` scales the vertical axes.

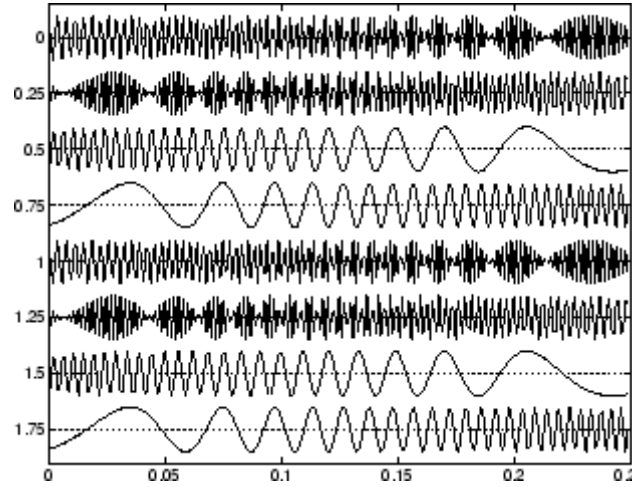
If `x` is a matrix, `strips(x,n)`, `strips(x,sd,fs)`, and `strips(x,sd,fs,scale)` plot the different columns of `x` on the same strip plot.

`strips` ignores the imaginary part of complex-valued `x`.

Examples

Plot two seconds of a frequency modulated sinusoid in 0.25 second strips:

```
fs = 1000;                          % Sampling frequency
t = 0:1/fs:2;                       % Time vector
x = vco(sin(2*pi*t),[10 490],fs);  % FM waveform
strips(x,0.25,fs)
```



See Also

`plot` | `stem`

taylorwin

Purpose

Taylor window

Syntax

```
w = taylorwin(n)
w = taylorwin(n,nbar)
w = taylorwin(n,nbar,sll)
```

Description

Taylor windows are similar to Chebyshev windows. While a Chebyshev window has the narrowest possible mainlobe for a specified sidelobe level, a Taylor window allows you to make tradeoffs between the mainlobe width and sidelobe level. The Taylor distribution avoids edge discontinuities, so Taylor window sidelobes decrease monotonically. Taylor window coefficients are not normalized. Taylor windows are typically used in radar applications, such as weighting synthetic aperature radar images and antenna design.

`w = taylorwin(n)` returns an `n`-point Taylor window in a column vector `w`. The values in this vector are the window weights or coefficients. `n` must be a positive integer. The default value for the number of approximately equal height sidelobes (`nbar`) is 4 and for the maximum sidelobe level (`sll`) is -30.

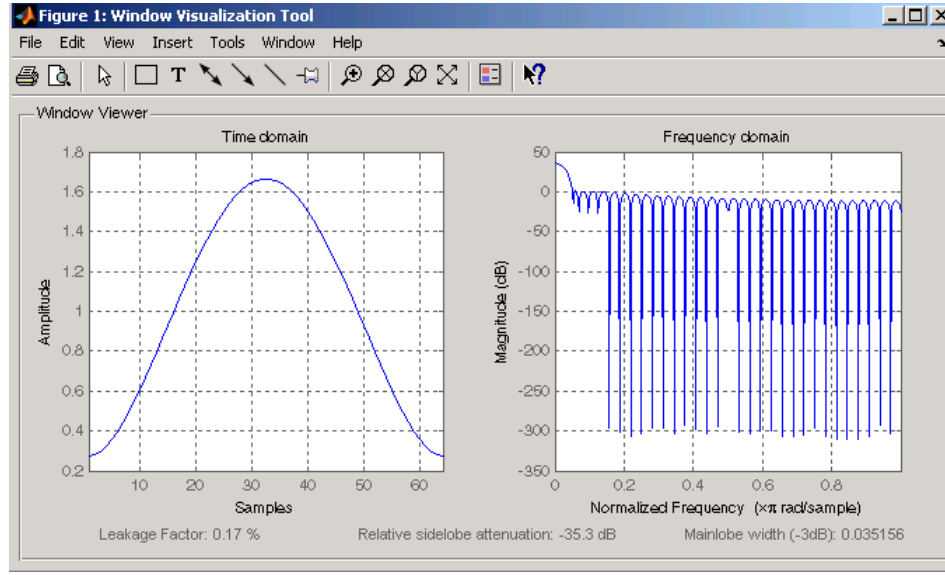
`w = taylorwin(n,nbar)` returns an `n`-point Taylor window with `nbar` nearly constant-level sidelobes adjacent to the mainlobe. These sidelobes are "nearly constant-level" because some decay occurs in the transition region. `nbar` must be a positive integer.

`w = taylorwin(n,nbar,sll)` returns an `n`-point Taylor window with a maximum sidelobe level of `sll` dB relative to the mainlobe peak.`sll` must be a negative value, such as -30, which produces sidelobes with peaks 30 dB down from the mainlobe peak.

Examples

Generate a 64-point Taylor window with four nearly constant-level sidelobes and a peak sidelobe level of -35 dB relative to the mainlobe peak.

```
w = taylorwin(64,4,-35);
wvtool(w);
```



References

[1] Carrara, W.G., R.M. Majewski and R.S. Goodman, *Spotlight Synthetic Aperature Radar: Signal Processing Algorithms*, Artech House Publishers, Boston, 1995, Appendix D.2.

[2] Brookner, Eli, *Practical Phased Array Antenna Systems*, Lex Book, Lexington, MA, 1991.

tf2latc

Purpose

Convert transfer function filter parameters to lattice filter form

Syntax

```
[k,v] = tf2latc(b,a)
k = tf2latc(1,a)
[k,v] = tf2latc(1,a)
k = tf2latc(b)
k = tf2latc(b, 'phase')
```

Description

`[k,v] = tf2latc(b,a)` finds the lattice parameters `k` and the ladder parameters `v` for an IIR (ARMA) lattice-ladder filter, normalized by `a(1)`. Note that an error is generated if one or more of the lattice parameters are exactly equal to 1.

`k = tf2latc(1,a)` finds the lattice parameters `k` for an IIR all-pole (AR) lattice filter.

`[k,v] = tf2latc(1,a)` returns the scalar ladder coefficient at the correct position in vector `v`. All other elements of `v` are zero.

`k = tf2latc(b)` finds the lattice parameters `k` for an FIR (MA) lattice filter, normalized by `b(1)`.

`k = tf2latc(b, 'phase')` specifies the type of FIR (MA) lattice filter, where `'phase'` is

- `'max'`, for a maximum phase filter.
- `'min'`, for a minimum phase filter.

See Also

`latc2tf` | `latcfilt` | `tf2sos` | `tf2ss` | `tf2zp` | `tf2zpk`

Purpose

Convert digital filter transfer function data to second-order sections form

Syntax

```
[sos,g] = tf2sos(b,a)
[sos,g] = tf2sos(b,a,'order')
[sos,g] = tf2sos(b,a,'order','scale')
sos = tf2sos(...)
```

Description

`tf2sos` converts a transfer function representation of a given digital filter to an equivalent second-order section representation.

`[sos,g] = tf2sos(b,a)` finds a matrix `sos` in second-order section form with gain `g` that is equivalent to the digital filter represented by transfer function coefficient vectors `a` and `b`.

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2 z^{-1} + \cdots + b_{n+1} z^{-n}}{a_1 + a_2 z^{-1} + \cdots + a_{m+1} z^{-m}}$$

`sos` is an L-by-6 matrix

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

whose rows contain the numerator and denominator coefficients b_{ik} and a_{ik} of the second-order sections of $H(z)$.

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

`[sos,g] = tf2sos(b,a,'order')` specifies the order of the rows in `sos`, where `'order'` is

- `'down'`, to order the sections so the first row of `sos` contains the poles closest to the unit circle

- `'up'`, to order the sections so the first row of `sos` contains the poles farthest from the unit circle (default)

`[sos,g] = tf2sos(b,a,'order','scale')` specifies the desired scaling of the gain and numerator coefficients of all second-order sections, where `'scale'` is:

- `'none'`, to apply no scaling (default)
- `'inf'`, to apply infinity-norm scaling
- `'two'`, to apply 2-norm scaling

Using infinity-norm scaling in conjunction with `up`-ordering minimizes the probability of overflow in the realization. Using 2-norm scaling in conjunction with `down`-ordering minimizes the peak round-off noise.

Note Infinity-norm and 2-norm scaling are appropriate only for direct-form II implementations.

`sos = tf2sos(...)` embeds the overall system gain, `g`, in the first section, $H_1(z)$, so that

$$H(z) = \prod_{k=1}^{L} H_k(z)$$

Note Embedding the gain in the first section when scaling a direct-form II structure is not recommended and may result in erratic scaling. To avoid embedding the gain, use `ss2sos` with two outputs.

Algorithms

`tf2sos` uses a four-step algorithm to determine the second-order section representation for an input transfer function system:

1. It finds the poles and zeros of the system given by `b` and `a`.

2. It uses the function `zp2sos`, which first groups the zeros and poles into complex conjugate pairs using the `cplxpair` function. `zp2sos` then forms the second-order sections by matching the pole and zero pairs according to the following rules:

   a. Match the poles closest to the unit circle with the zeros closest to those poles.

   b. Match the poles next closest to the unit circle with the zeros closest to those poles.

   c. Continue until all of the poles and zeros are matched.

   `tf2sos` groups real poles into sections with the real poles closest to them in absolute value. The same rule holds for real zeros.

3. It orders the sections according to the proximity of the pole pairs to the unit circle. `tf2sos` normally orders the sections with poles closest to the unit circle last in the cascade. You can tell `tf2sos` to order the sections in the reverse order by specifying the `'down'` flag.

4. `tf2sos` scales the sections by the norm specified in the `'scale'` argument. For arbitrary $H(\omega)$, the scaling is defined by

$$\| H \|_p = \left[\frac{1}{2\pi} \int_0^{2\pi} | H(\omega) |^p \, d\omega \right]^{\frac{1}{p}}$$

   where p can be either ∞ or 2. See the references for details on the scaling. This scaling is an attempt to minimize overflow or peak round-off noise in fixed point filter implementations.

References

[1] Jackson, L.B., *Digital Filters and Signal Processing, 3rd ed.*, Kluwer Academic Publishers, Boston, 1996, Chapter 11.

[2] Mitra, S.K., *Digital Signal Processing: A Computer-Based Approach*, McGraw-Hill, New York, 1998, Chapter 9.

[3] Vaidyanathan, P.P., "Robust Digital Filter Structures," *Handbook for Digital Signal Processing*, S.K. Mitra and J.F. Kaiser, ed., John Wiley & Sons, New York, 1993, Chapter 7.

See Also

`cplxpair` | `sos2tf` | `ss2sos` | `tf2ss` | `tf2zp` | `tf2zpk` | `zp2sos`

Purpose

Convert transfer function filter parameters to state-space form

Syntax

```
[A,B,C,D] = tf2ss(b,a)
```

Description

`tf2ss` converts the parameters of a transfer function representation of a given system to those of an equivalent state-space representation.

`[A,B,C,D] = tf2ss(b,a)` returns the `A`, `B`, `C`, and `D` matrices of a state space representation for the single-input transfer function

$$H(s) = \frac{B(s)}{A(s)} = \frac{b_1 s^{n-1} + \cdots + b_{n-1}s + b_n}{a_1 s^{m-1} + \cdots + a_{m-1}s + a_m} = C(sI - A)^{-1}B + D$$

in controller canonical form

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

The input vector `a` contains the denominator coefficients in descending powers of s. The rows of the matrix `b` contain the vectors of numerator coefficients (each row corresponds to an output). In the discrete-time case, you must supply `b` and `a` to correspond to the numerator and denominator polynomials with coefficients in descending powers of z.

For discrete-time systems you must make `b` have the same number of columns as the length of `a`. You can do this by padding each numerator represented in `b` (and possibly the denominator represented in the vector `a`) with trailing zeros. You can use the function `eqtflength` to accomplish this if `b` and `a` are vectors of unequal lengths.

The `tf2ss` function is part of the standard MATLAB language.

Examples

Consider the system:

$$H(s) = \frac{\begin{bmatrix} 2s+3 \\ s^2+2s+1 \end{bmatrix}}{s^2+0.4s+1}$$

To convert this system to state-space, type

```
b = [0 2 3; 1 2 1];
a = [1 0.4 1];
[A,B,C,D] = tf2ss(b,a)
A =
   -0.4000   -1.0000
    1.0000         0
B =
     1
     0
C =
    2.0000    3.0000
    1.6000         0
D =
     0
     1
```

Note There is disagreement in the literature on naming conventions for the canonical forms. It is easy, however, to generate similarity transformations that convert these results to other forms.

See Also

`sos2ss` | `ss2tf` | `tf2sos` | `tf2zp` | `tf2zpk` | `zp2ss`

Purpose

Convert transfer function filter parameters to zero-pole-gain form

Syntax

```
[z,p,k] = tf2zp(b,a)
```

Description

`tf2zp` finds the zeros, poles, and gains of a continuous-time transfer function.

Note You should use `tf2zp` when working with positive powers ($s^2 + s + 1$), such as in continuous-time transfer functions. A similar function, `tf2zpk`, is more useful when working with transfer functions expressed in inverse powers ($1 + z^{-1} + z^{-2}$), which is how transfer functions are usually expressed in DSP.

`[z,p,k] = tf2zp(b,a)` finds the matrix of zeros `z`, the vector of poles `p`, and the associated vector of gains `k` from the transfer function parameters `b` and `a`:

- The numerator polynomials are represented as columns of the matrix `b`.
- The denominator polynomial is represented in the vector `a`.

Given a SIMO continuous-time system in polynomial transfer function form

$$H(s) = \frac{B(s)}{A(s)} = \frac{b_1 s^{n-1} + \cdots + b_{n-1}s + b_n}{a_1 s^{m-1} + \cdots + a_{m-1}s + a_m}$$

you can use the output of `tf2zp` to produce the single-input, multioutput (SIMO) factored transfer function form

$$H(s) = \frac{Z(s)}{P(s)} = k\frac{(s - z_1)(s - z_2)\cdots(s - z_m)}{(s - p_1)(s - p_2)\cdots(s - p_n)}$$

The following describes the input and output arguments for `tf2zp`:

- The vector `a` specifies the coefficients of the denominator polynomial $A(s)$ (or $A(z)$) in descending powers of s (z^{-1}).
- The ith row of the matrix `b` represents the coefficients of the ith numerator polynomial (the ith row of $B(s)$ or $B(z)$). Specify as many rows of `b` as there are outputs.
- For continuous-time systems, choose the number nb of columns of `b` to be less than or equal to the length na of the vector `a`.
- For discrete-time systems, choose the number nb of columns of `b` to be equal to the length na of the vector `a`. You can use the function `eqtflength` to provide equal length vectors in the case that `b` and `a` are vectors of unequal lengths. Otherwise, pad the numerators in the matrix `b` (and, possibly, the denominator vector `a`) with zeros.
- The zero locations are returned in the columns of the matrix `z`, with as many columns as there are rows in `b`.
- The pole locations are returned in the column vector `p` and the gains for each numerator transfer function in the vector `k`.

The `tf2zp` function is part of the standard MATLAB language.

Examples

Find the zeros, poles, and gains of this continuous-time system:

$$H(s) = \frac{2s^2 + 3s}{s^2 + 0.4s + 1}$$

```
b = [2 3];
a = [1 0.4 1];
[b,a] = eqtflength(b,a);      % Make lengths equal
[z,p,k] = tf2zp(b,a)          % Obtain zero-pole-gain form
z =
         0
   -1.5000
p =
   -0.2000 + 0.9798i
   -0.2000 - 0.9798i
```

```
k =
    2
```

See Also

sos2zp | ss2zp | tf2sos | tf2ss | tf2zpk | zp2tf

tf2zpk

Purpose

Convert transfer function filter parameters to zero-pole-gain form

Syntax

```
[z,p,k] = tf2zpk(b,a)
```

Description

`tf2zpk` finds the zeros, poles, and gains of a discrete-time transfer function.

Note You should use `tf2zpk` when working with transfer functions expressed in inverse powers ($1 + z^{-1} + z^{-2}$), which is how transfer functions are usually expressed in DSP. A similar function, `tf2zp`, is more useful for working with positive powers ($s^2 + s + 1$), such as in continuous-time transfer functions.

`[z,p,k] = tf2zpk(b,a)` finds the matrix of zeros `z`, the vector of poles `p`, and the associated vector of gains `k` from the transfer function parameters `b` and `a`:

- The numerator polynomials are represented as columns of the matrix `b`.
- The denominator polynomial is represented in the vector `a`.

Given a single-input, multiple output (SIMO) discrete-time system in polynomial transfer function form

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_1 + b_2 z^{-1} \cdots + b_{n-1} z^{-n} + b_n z^{-n-1}}{a_1 + a_2 z^{-1} \cdots + a_{m-1} z^{-m} + a_m z^{-m-1}}$$

you can use the output of `tf2zpk` to produce the single-input, multioutput (SIMO) factored transfer function form

$$H(z) = \frac{Z(z)}{P(z)} = k \frac{(z - z_1)(z - z_2) \cdots (z - z_m)}{(z - p_1)(z - p_2) \cdots (z - p_n)}$$

The following describes the input and output arguments for `tf2zpk`:

- The vector `a` specifies the coefficients of the denominator polynomial $A(z)$ in descending powers of z.
- The ith row of the matrix `b` represents the coefficients of the `i`th numerator polynomial (the ith row of $B(s)$ or $B(z)$). Specify as many rows of `b` as there are outputs.
- The zero locations are returned in the columns of the matrix `z`, with as many columns as there are rows in `b`.
- The pole locations are returned in the column vector `p` and the gains for each numerator transfer function in the vector `k`.

Examples

Find the poles, zeros, and gain of a Butterworth filter:

```
[b,a] = butter(3,.4);
[z,p,k] = tf2zpk(b,a)
z =
  -1.0000
  -1.0000 + 0.0000i
  -1.0000 - 0.0000i

p =
   0.2094 + 0.5582i
   0.2094 - 0.5582i
   0.1584

k =
    0.0985
```

See Also

`sos2zp` | `ss2zp` | `tf2sos` | `tf2ss` | `tf2zp` | `zp2tf`

tfestimate

Purpose

Transfer function estimate

Syntax

```
Txy = tfestimate(x,y)
Txy = tfestimate(x,y,window)
Txy = tfestimate(x,y,window,noverlap)
[Txy,W] = tfestimate(x,y,window,noverlap,nfft)
[Txy,F] = tfestimate(x,y,window,noverlap,nfft,fs)
[...] = tfestimate(x,y,...,'twosided')
tfestimate(...)
```

Description

`Txy = tfestimate(x,y)` finds a transfer function estimate `Txy` given input signal vector `x` and output signal vector `y`. Vectors `x` and `y` must be the same length. The relationship between the input `x` and output `y` is modeled by the linear, time-invariant transfer function `Txy`. The *transfer function* is the quotient of the cross power spectral density (*Pyx*) of `x` and `y` and the power spectral density (*Pxx*) of `x`.

$$T_{xy}(f) = \frac{P_{yx}(f)}{P_{xx}(f)}$$

If `x` is real, `tfestimate` estimates the transfer function at positive frequencies only; in this case, the output `Txy` is a column vector of length `nfft/2+1` for `nfft` even and `(nfft+1)/2` for `nfft` odd. If `x` or `y` is complex, `tfestimate` estimates the transfer function for both positive and negative frequencies and `Txy` has length `nfft`.

`tfestimate` uses the following default values:

Default Values

Parameter	Description	Default Value
`nfft`	FFT length which determines the frequencies at which the PSD is estimated For real `x` and `y`, the length of `Txy` is (`nfft`/2+1) if `nfft` is even or (`nfft`+1)/2 if `nfft` is odd. For complex `x` or `y`, the length of `Txy` is `nfft`.	Maximum of 256 or the next power of 2 greater than the length of each section of `x` or `y`
`fs`	Sampling frequency	1
`window`	Windowing function and number of samples to use to section x and y	Periodic Hamming window with length equal to the signal segment length that results from dividing the signal `x` into eight sections and then applying the default or specified overlap.
`noverlap`	Number of samples by which the sections overlap	Value to obtain 50% overlap

Note You can use the empty matrix `[]` to specify the default value for any input argument except `x` or `y`. For example, `Txy = tfestimate(x,y,[],[],128)` uses a Hamming window with default length, as described above, default `noverlap` to obtain 50% overlap, and the specified 128 `nfft`.

`Txy = tfestimate(x,y,window)` specifies a windowing function, divides `x` and `y` into overlapping sections of the specified window length, and windows each section using the specified window function. If you supply a scalar for `window`, `Txy` uses a Hamming window of that length. The length of the window must be less than or equal to `nfft`. If the length of the window exceeds `nfft`, `tfestimate` zero pads the sections. To replicate the output of the obsoleted `tfe` function, specify `'hanning(nfft)'` as the window.

`Txy = tfestimate(x,y,window,noverlap)` overlaps the sections of `x` by `noverlap` samples. `noverlap` must be an integer smaller than the length of `window`.

`[Txy,W] = tfestimate(x,y,window,noverlap,nfft)` uses the specified FFT length `nfft` in estimating the PSD and CPSD estimates for the transfer function. It also returns `W`, which is the vector of normalized frequencies (inrad/sample) at which the `tfestimate` is estimated. For real signals, the range of `W` is [0, π] when `nfft` is even and [0, π) when `nfft` is odd. For complex signals, the range of `W` is [0, 2π).

`[Txy,F] = tfestimate(x,y,window,noverlap,nfft,fs)` returns `Txy` as a function of frequency and a vector `F` of frequencies at which `tfestimate` estimates the transfer function. `fs` is the sampling frequency in Hz. `F` is the same size as `Txy`, so `plot(f,Txy)` plots the transfer function estimate versus properly scaled frequency. For real signals, the range of `F` is [0, `fs`/2] when `nfft` is even and [0, `fs`/2) when `nfft` is odd. For complex signals, the range of `F` is [0, `fs`).

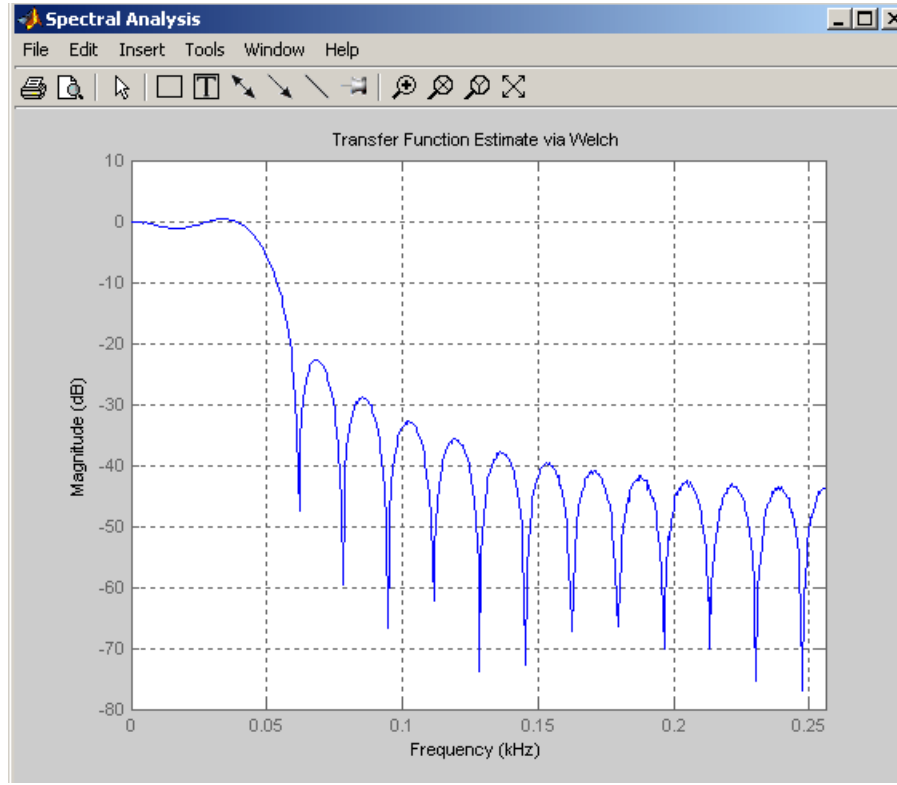
`[...] = tfestimate(x,y,...,'twosided')` returns a transfer function estimate with frequencies that range over the entire interval from 0 to the sampling frequency, [0,Fs). Specifying `'onesided'` uses from 0 to the Nyquist frequency.

`tfestimate(...)` with no output arguments plots the transfer function estimate in the current figure window.

Examples

Compute and plot the transfer function estimate between two colored noise sequences `x` and `y`:

```
h = fir1(30,0.2,rectwin(31));
x = randn(16384,1);
y = filter(h,1,x);
tfestimate(x,y,1024,[],[],512)
```



Algorithms

`tfestimate` uses Welch's averaged periodogram method. See `pwelch` for details.

See Also

`cpsd` | `mscohere` | `periodogram` | `pwelch` | `spectrum`

Purpose

Triangular window

Syntax

```
triang(L)
```

Description

`triang(L)` returns an `L`-point triangular window in the column vector `w`. The coefficients of a triangular window are:

For `L` odd:

$$w(n) = \begin{cases} \dfrac{2n}{L+1} & 1 \le n \le (L+1)/2 \\ 2 - \dfrac{2n}{L+1} & (L+1)/2 + 1 \le n \le L \end{cases}$$

For `L` even:

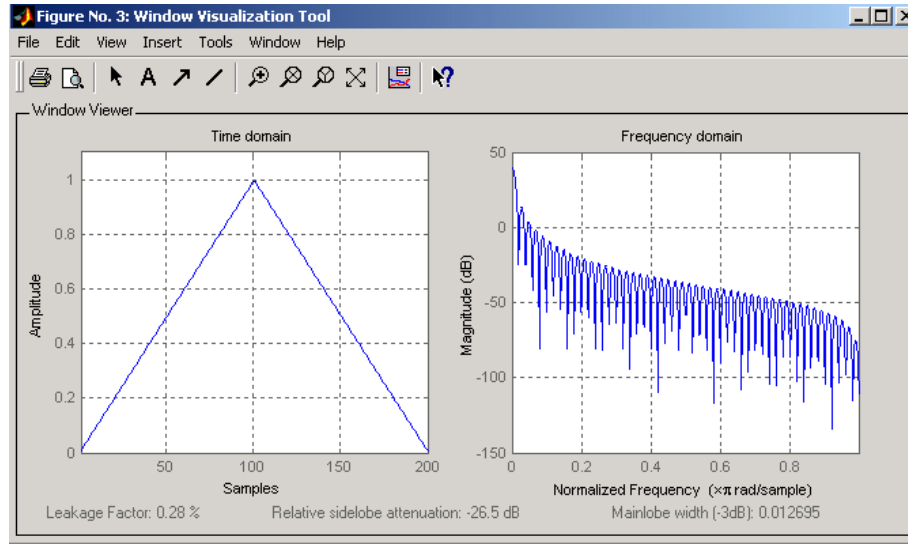
$$w(n) = \begin{cases} \dfrac{(2n-1)}{L} & 1 \le n \le L/2 \\ 2 - \dfrac{(2n-1)}{L} & L/2 + 1 \le n \le L \end{cases}$$

The triangular window is very similar to a Bartlett window. The Bartlett window always ends with zeros at samples 1 and `L`, while the triangular window is nonzero at those points. For `L` odd, the center `L-2` points of `triang(L-2)` are equivalent to `bartlett(L)`.

Examples

Create a 200-point triangular window and plot the result using WVTool.

```
L = 200;
wvtool(triang(L))
```



References

[1] Oppenheim, A.V., and R.W. Schafer, *Discrete-Time Signal Processing*, Prentice-Hall, Englewood Cliffs, NJ, 1989, pp. 447-448.

See Also

`barthannwin` | `bartlett` | `blackmanharris` | `bohmanwin` | `nuttallwin` | `parzenwin` | `rectwin` | `window` | `wintool` | `wvtool`

tripuls

Purpose

Sampled aperiodic triangle

Syntax

```
y = tripuls(T)
y = tripuls(T,w)
y = tripuls(T,w,s)
```

Description

`y = tripuls(T)` returns a continuous, aperiodic, symmetric, unity-height triangular pulse at the times indicated in array `T`, centered about `T=0` and with a default width of 1.

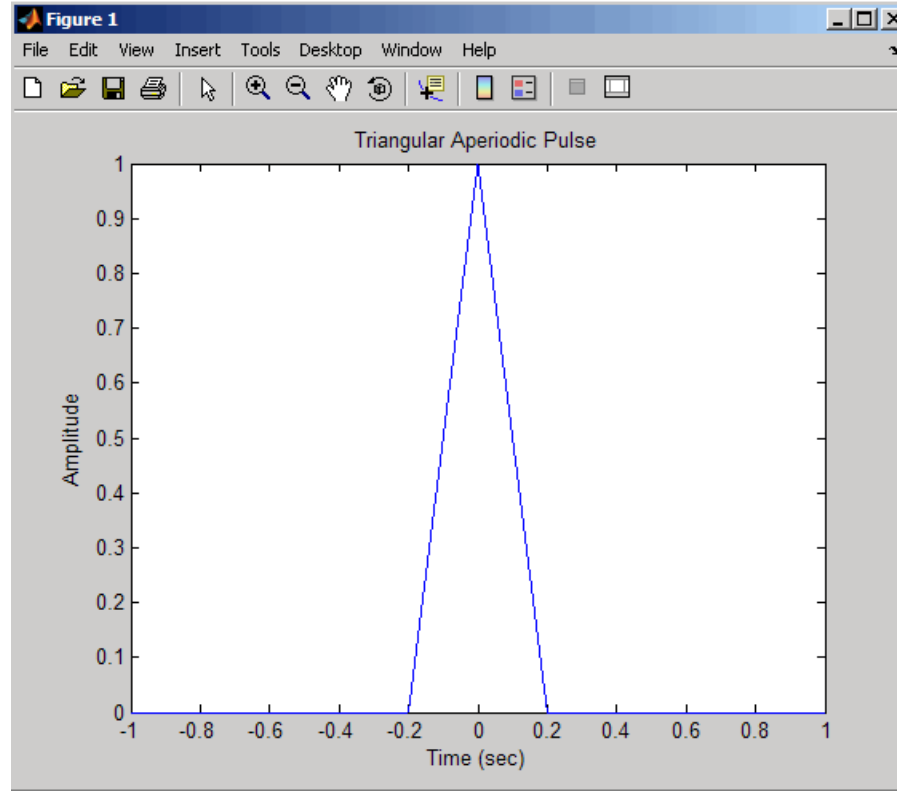
`y = tripuls(T,w)` generates a triangular pulse of width `w`.

`y = tripuls(T,w,s)` generates a triangular pulse with skew `s`, where `-1 < s < 1`. When `s` is 0, a symmetric triangular pulse is generated.

Examples

Create a triangular pulse with width 0.4.

```
fs = 10000;
t = -1:1/fs:1;
w = .4;
x = tripuls(t,w);
figure,plot(t,x)
xlabel('Time (sec)');ylabel('Amplitude');
title('Triangular Aperiodic Pulse')
```



See Also

`chirp` | `cos` | `diric` | `gauspuls` | `pulstran` | `rectpuls` | `sawtooth` | `sin` | `square` | `tripuls`

tukeywin

Purpose

Tukey (tapered cosine) window

Syntax

```
w = tukeywin(L,r)
```

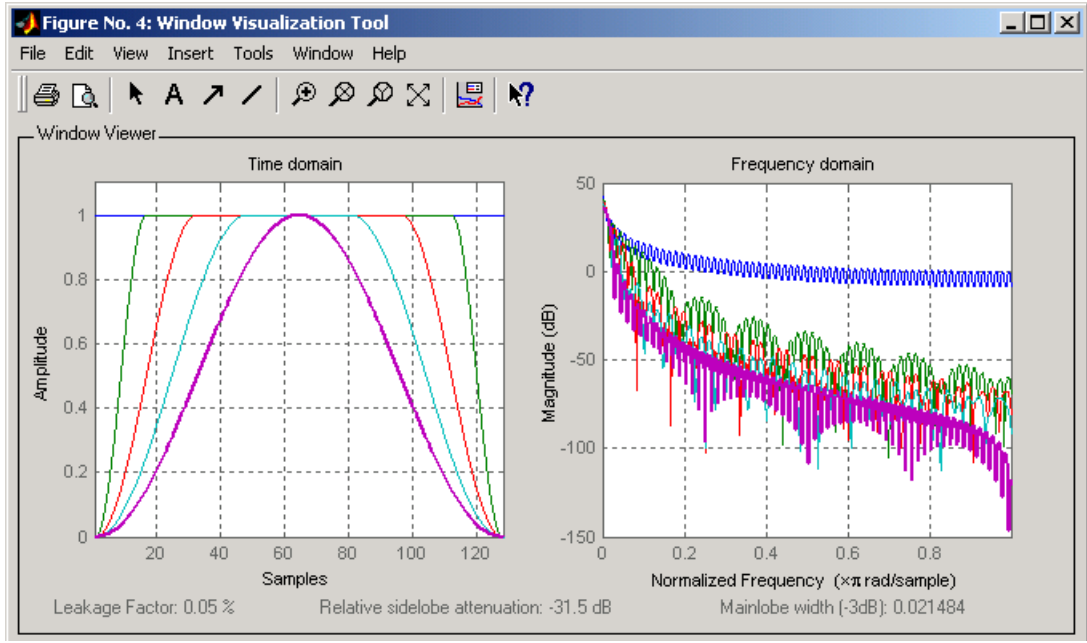
Description

`w = tukeywin(L,r)` returns an `L`-point Tukey window in the column vector `w`. A Tukey window is a rectangular window with the first and last `r/2` percent of the samples equal to parts of a cosine. See "Definitions" on page 1-1235 for the equation of the Tukey window. `r` is a real number between 0 and 1. If you input $r \leq 0$, you obtain a `rectwin` window. If you input $r \geq 1$, you obtain a `hann` window. `r` defaults to 0.5.

Examples

Compute 128-point Tukey windows with five different values for `r` and display the results using WVTool.

```
L=128;
t0=tukeywin(L,0);        % Equivalent to rectangular window
t25=tukeywin(L,0.25);
t5=tukeywin(L);          % r=0.5
t75=tukeywin(L,0.75);
t1=tukeywin(L,1);        % Equivalent to Hann window
wvtool(t0,t25,t5,t75,t1)
```



Definitions

The following equation defines the L–point Tukey window:

$$w(x) = \begin{cases} \frac{1}{2}\{1+\cos(\frac{2\pi}{r}[x - r/2])\} & 0 \le x < \frac{r}{2} \\ 1 & \frac{r}{2} \le x < 1 - \frac{r}{2} \\ \frac{1}{2}\{1+\cos(\frac{2\pi}{r}[x - 1 + r/2])\} & 1 - \frac{r}{2} \le x \le 1 \end{cases}$$

where x is a L–point linearly spaced vector generated using `linspace`. The parameter r is the ratio of cosine-tapered section length to the entire window length with $0 \le r \le 1$. For example, setting $r = 0.5$ produces a Tukey window where 1/2 of the entire window length consists of segments of a phase-shifted cosine with period $2r = 1$. If you specify

r≤0, an *L*-point rectangular window is returned. If you specify r≥1, an *L*-point von Hann window (`hann`) is returned.

References

[1] Bloomfield P. *Fourier Analysis of Time Series: An Introduction*, New York: Wiley-Interscience, 2000, p.69.

See Also

`chebwin` | `gausswin` | `kaiser` | `window` | `wintool` | `wvtool`

Purpose

Decode 2^n-level quantized integer inputs to floating-point outputs

Syntax

```
y = udecode(u,n)
y = udecode(u,n,v)
y = udecode(u,n,v,'SaturateMode')
```

Description

`y = udecode(u,n)` inverts the operation of `uencode` and reconstructs quantized floating-point values from an encoded multidimensional array of integers `u`. The input argument `n` must be an integer between 2 and 32. The integer `n` specifies that there are 2^n quantization levels for the inputs, so that entries in `u` must be either:

- Signed integers in the range $[-2^n/2, (2^n/2) - 1]$
- Unsigned integers in the range $[0, 2^n-1]$

Inputs can be real or complex values of any integer data type (`uint8`, `uint16`, `uint32`, `int8`, `int16`, `int32`). Overflows (entries in `u` outside of the ranges specified above) are saturated to the endpoints of the range interval. The output `y` has the same dimensions as `u`. Its entries have values in the range `[-1,1]`.

`y = udecode(u,n,v)` decodes `u` such that the output `y` has values in the range `[-v,v]`, where the default value for `v` is 1.

`y = udecode(u,n,v,'SaturateMode')` decodes `u` and treats input overflows (entries in `u` outside of `[-v,v]`) according to the string `'saturatemode'`, which can be one of the following:

- `'saturate'`: Saturate overflows. This is the default method for treating overflows.
  - Entries in signed inputs `u` whose values are outside of the range $[-2^n/2, (2^n/2) - 1]$ are assigned the value determined by the closest endpoint of this interval.
  - Entries in unsigned inputs `u` whose values are outside of the range $[0, 2^n-1]$ are assigned the value determined by the closest endpoint of this interval.
- `'wrap'`: Wrap all overflows according to the following:

- Entries in signed inputs `u` whose values are outside of the range [$-2^n/2$, ($2^n/2$) - 1] are wrapped back into that range using modulo 2^n arithmetic (calculated using `u = mod(u+2^n/2,2^n)-(2^n/2)`).
- Entries in unsigned inputs `u` whose values are outside of the range [0, 2^n-1] are wrapped back into the required range before decoding using modulo 2^n arithmetic (calculated using `u = mod(u,2^n)`).

Examples

```
% Create signed 8-bit integer string
u = int8([-1 1 2 -5]);
% Decode with 3 bits
ysat = udecode(u,3)
ysat =
   -0.2500    0.2500    0.5000   -1.0000
```

Notice the last entry in `u` saturates to 1, the default peak input magnitude. Change the peak input magnitude:

```
ysatv = udecode(u,3,6)   % Set peak input magnitude to 6
ysatv =
   -1.5000    1.5000    3.0000   -6.0000
```

The last input entry still saturates. Try wrapping the overflows:

```
ywrap = udecode(u,3,6,'wrap')
ywrap =
   -1.5000    1.5000    3.0000    4.5000
```

Try adding more quantization levels:

```
yprec = udecode(u,5)
yprec =
   -0.0625    0.0625    0.1250   -0.3125
```

Algorithms

The algorithm adheres to the definition for uniform decoding specified in ITU-T Recommendation G.701. Integer input values are uniquely mapped (decoded) from one of 2^n uniformly spaced integer values to quantized floating-point values in the range `[-v,v]`. The smallest

integer input value allowed is mapped to -v and the largest integer input value allowed is mapped to v. Values outside of the allowable input range are either saturated or wrapped, according to specification.

The real and imaginary components of complex inputs are decoded independently.

References

[1] *General Aspects of Digital Transmission Systems: Vocabulary of Digital Transmission and Multiplexing, and Pulse Code Modulation (PCM) Terms*, International Telecommunication Union, ITU-T Recommendation G.701, March, 1993.

See Also

uencode

uencode

Purpose

Quantize and encode floating-point inputs to integer outputs

Syntax

```
y = uencode(u,n)
y = uencode(u,n,v)
y = uencode(u,n,v,'SignFlag')
```

Description

`y = uencode(u,n)` quantizes the entries in a multidimensional array of floating-point numbers `u` and encodes them as integers using 2^n-level quantization. `n` must be an integer between 2 and 32 (inclusive). Inputs can be real or complex, double- or single-precision. The output y and the input `u` are arrays of the same size. The elements of the output y are unsigned integers with magnitudes in the range [0, 2^n-1]. Elements of the input `u` outside of the range `[-1,1]` are treated as overflows and are saturated.

- For entries in the input `u` that are less than -1, the value of the output of `uencode` is 0.
- For entries in the input `u` that are greater than 1, the value of the output of `uencode` is 2^n-1.

`y = uencode(u,n,v)` allows the input `u` to have entries with floating-point values in the range `[-v,v]` before saturating them (the default value for `v` is 1). Elements of the input `u` outside of the range `[-v,v]` are treated as overflows and are saturated:

- For input entries less than `-v`, the value of the output of `uencode` is 0.
- For input entries greater than `v`, the value of the output of `uencode` is 2^n-1.

`y = uencode(u,n,v,'SignFlag')` maps entries in a multidimensional array of floating-point numbers `u` whose entries have values in the range [-v,v] to an integer output `y`. Input entries outside this range are saturated. The integer type of the output depends on the string `'SignFlag'` and the number of quantization levels 2^n. The string `'SignFlag'` can be one of the following:

- `'signed'`: Outputs are signed integers with magnitudes in the range [$-2^n/2$, ($2^n/2$) - 1].

- `'unsigned'` (default): Outputs are unsigned integers with magnitudes in the range [0, 2^n-1].

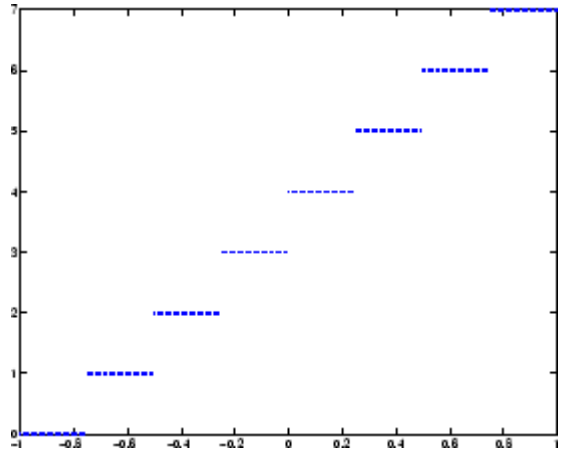
The output data types are optimized for the number of bits as shown in the table below.

n	Unsigned Integer	Signed Integer
2 to 8	`uint8`	`int8`
9 to 16	`uint16`	`int16`
17 to 32	`uint32`	`int32`

Examples

Map floating-point scalars in [-1, 1] to `uint8` (unsigned) integers, and produce a staircase plot. Note that the horizontal axis plots from -1 to 1 and the vertical axis plots from 0 to 7 (`2^3-1`):

```
u = [-1:0.01:1];
y = uencode(u,3);
plot(u,y,'.')
```



Now look at saturation effects when you under specify the peak value for the input:

```
u = [-2:0.5:2];
y = uencode(u,5,1)
y =
     0     0     0     8    16    24    31    31    31
```

Now look at the output for

```
u = [-2:0.5:2];
y = uencode(u,5,2,'signed')
y =
    -16   -12    -8    -4     0     4     8    12    15
```

Algorithms

`uencode` maps the floating-point input value to an integer value determined by the requirement for 2^n levels of quantization. This encoding adheres to the definition for uniform encoding specified in ITU-T Recommendation G.701. The input range `[-v,v]` is divided into 2^n evenly spaced intervals. Input entries in the range `[-v,v]` are first quantized according to this subdivision of the input range, and then mapped to one of 2^n integers. The range of the output depends on whether or not you specify that you want signed integers.

References

[1] *General Aspects of Digital Transmission Systems: Vocabulary of Digital Transmission and Multiplexing, and Pulse Code Modulation (PCM) Term*s, International Telecommunication Union, ITU-T Recommendation G.701, March, 1993.

See Also

`udecode`

Purpose

Inverse of `shiftdata`

Syntax

```
y = unshiftdata(x,perm,nshifts)
```

Description

`y = unshiftdata(x,perm,nshifts)` restores the orientation of the data that was shifted with `shiftdata`. The permutation vector is given by `perm`, and `nshifts` is the number of shifts that was returned from `shiftdata`.

`unshiftdata` is meant to be used in tandem with `shiftdata`. These functions are useful for creating functions that work along a certain dimension, like `filter`, `goertzel`, `sgolayfilt`, and `sosfilt`.

Examples

Example 1

This example shifts `x`, a `3-by-3` magic square, permuting dimension `2` to the first column. `unshiftdata` shifts `x` back to its original shape.

1. Create a `3-by-3` magic square:

```
x = fi(magic(3))
```

```
x =

     8     1     6
     3     5     7
     4     9     2
```

2. Shift the matrix `x` to work along the second dimension:

```
[x,perm,nshifts] = shiftdata(x,2)
```

This command returns the permutation vector, `perm`, and the number of shifts, `nshifts`, are returned along with the shifted matrix, `x`:

```
x =
```

```
     8     3     4
     1     5     9
     6     7     2


perm =

     2     1


nshifts =

     []
```

3. Shift the matrix back to its original shape:

```
y = unshiftdata(x,perm,nshifts)

y =

     8     1     6
     3     5     7
     4     9     2
```

Example 2

This example shows how `shiftdata` and `unshiftdata` work when you define `dim` as empty.

1. Define `x` as a row vector:

```
x = 1:5

x =

     1     2     3     4     5
```

2. Define `dim` as empty to shift the first non-singleton dimension of `x` to the first column:

```
[x,perm,nshifts] = shiftdata(x,[])
```

This command returns `x` as a column vector, along with `perm`, the permutation vector, and `nshifts`, the number of shifts:

```
x =

     1
     2
     3
     4
     5


perm =

     []


nshifts =

     1
```

3. Using `unshiftdata`, restore `x` to its original shape:

```
y = unshiftdata(x,perm,nshifts)

y =

     1     2     3     4     5
```

See Also

`ipermute` | `shiftdata` | `shiftdim`

upfirdn

Purpose

Upsample, apply FIR filter, and downsample

Syntax

```
yout = upfirdn(xin,h)
yout = upfirdn(xin,h,p)
yout = upfirdn(xin,h,p,q)
```

Description

`upfirdn` performs a cascade of three operations:

1. Upsampling the input data in the matrix `xin` by a factor of the integer `p` (inserting zeros)
2. FIR filtering the upsampled signal data with the impulse response sequence given in the vector or matrix `h`
3. Downsampling the result by a factor of the integer `q` (throwing away samples)

`upfirdn` has been implemented as a MEX-file for maximum speed, so only the outputs actually needed are computed. The FIR filter is usually a lowpass filter, which you must design using another function such as `firpm` or `fir1`.

Note The function `resample` performs an FIR design using `firls`, followed by rate changing implemented with `upfirdn`.

`yout = upfirdn(xin,h)` filters the input signal `xin` with the FIR filter having impulse response `h`. If `xin` is a row or column vector, then it represents a single signal. If `xin` is a matrix, then each column is filtered independently. If `h` is a row or column vector, then it represents one FIR filter. If `h` is a matrix, then each column is a separate FIR impulse response sequence. If `yout` is a row or column vector, then it represents one signal. If `yout` is a matrix, then each column is a separate output. No upsampling or downsampling is implemented with this syntax.

yout = upfirdn(*xin*,*h*,*p*) specifies the integer upsampling factor p, where p has a default value of 1.

yout = upfirdn(*xin*,*h*,*p*,*q*) specifies the integer downsampling factor q, where q has a default value of 1. The length of the output, *yout*, is ceil(((length(xin)-1)*p+length(h))/q)

Note Since upfirdn performs convolution and rate changing, the yout signals have a different length than xin. The number of rows of yout is approximately p/q times the number of rows of xin.

Tips

Usually the inputs xin and the filter h are vectors, in which case only one output signal is produced. However, when these arguments are arrays, each column is treated as a separate signal or filter. Valid combinations are:

1. xin is a vector and h is a vector.

   There is one filter and one signal, so the function convolves xin with h. The output signal yout is a row vector if xin is a row; otherwise, yout is a column vector.

2. xin is a matrix and h is a vector.

   There is one filter and many signals, so the function convolves h with each column of xin. The resulting yout will be an matrix with the same number of columns as xin.

3. xin is a vector and h is a matrix.

   There are many filters and one signal, so the function convolves each column of h with xin. The resulting yout will be an matrix with the same number of columns as h.

4. xin is a matrix and h is a matrix, both with the same number of columns.

There are many filters and many signals, so the function convolves corresponding columns of `xin` and `h`. The resulting `yout` is an matrix with the same number of columns as `xin` and `h`.

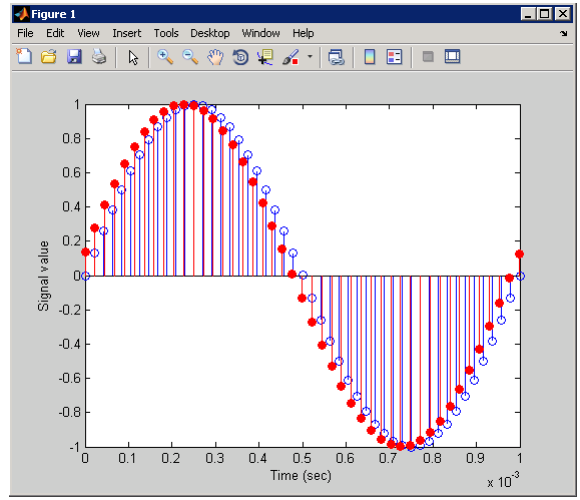
Examples

Change the sampling rate by a factor of 147/160. This factor is used to convert from 48kHz (DAT rate) to 44.1kHz (CD sampling rate).

```
L = 147; M = 160;     % Interpolation/decimation factors.
N = 24*L;
h = fir1(N-1,1/M,kaiser(N,7.8562));
h = L*h; % Passband gain = L
Fs = 48e3;            % Original sampling frequency-48kHz
n = 0:10239;          % 10240 samples, 0.213 seconds long
x  = sin(2*pi*1e3/Fs*n); % Original signal, sinusoid @ 1kHz
y = upfirdn(x,h,L,M);    % 9430 samples, still .213 seconds

% Overlay original (48kHz) with resampled
% signal (44.1kHz) in red.

stem(n(1:49)/Fs,x(1:49)); hold on
stem(n(1:45)/(Fs*L/M),y(13:57),'r','filled');
xlabel('Time (sec)');ylabel('Signal value');
```



Algorithms

`upfirdn` uses a polyphase interpolation structure. The number of multiply-add operations in the polyphase structure is approximately $(L_h L_x\text{-}pL_x)/q$ where L_h and L_x are the lengths of $h[n]$ and $x[n]$, respectively.

A more accurate flops count is computed in the program, but the actual count is still approximate. For long signals $x[n]$, the formula is often exact.

Diagnostics

If `p` and `q` are large and do not have many common factors, you may see this message:

```
Filter length is too large - reduce problem complexity.
```

Instead, you should use an interpolation function, such as `interp1`, to perform the resampling and then filter the input.

References

[1] Crochiere, R.E., and L.R. Rabiner, *Multi-Rate Signal Processing*, Prentice-Hall, Englewood Cliffs, NJ, 1983, pp. 88-91.

[2] Crochiere, R.E., "A General Program to Perform Sampling Rate Conversion of Data by Rational Ratios," *Programs for Digital Signal Processing*, IEEE Press, New York, 1979, pp. 8.2-1 to 8.2-7.

See Also

`conv` | `decimate` | `downsample` | `filter` | `interp` | `intfilt` | `resample` | `upsample`

Purpose

Increase sampling rate by integer factor

Syntax

```
y = upsample(x,n)
y = upsample(x,n,phase)
```

Description

`y = upsample(x,n)` increases the sampling rate of `x` by inserting `n`-1 zeros between samples. `x` can be a vector or a matrix. If `x` is a matrix, each column is considered a separate sequence. The upsampled `y` has `x*n` samples.

`y = upsample(x,n,phase)` specifies the number of samples by which to offset the upsampled sequence. `phase` must be an integer from 0 to `n`-1.

Examples

Increase the sampling rate of a sequence by 3:

```
x = [1 2 3 4];
y = upsample(x,3);
x,y
x =
    1  2  3  4
y =
    1  0  0  2  0  0  3  0  0  4  0  0
```

Increase the sampling rate of the sequence by 3 and add a phase offset of 2:

```
x = [1 2 3 4];
y = upsample(x,3,2);
x,y
x =
    1  2  3  4
y =
    0  0  1  0  0  2  0  0  3  0  0  4
```

Increase the sampling rate of a matrix by 3:

```
x = [1 2; 3 4; 5 6;];
y = upsample(x,3);
```

```
x,y
x =
     1     2
     3     4
     5     6
y =
     1     2
     0     0
     0     0
     3     4
     0     0
     0     0
     5     6
     0     0
     0     0
```

See Also

decimate | downsample | interp | interp1 | resample | spline | upfirdn

Purpose

Undershoot metrics of bilevel waveform transitions

Syntax

```
US = undershoot(X)
US = undershoot(X,FS)
US = undershoot(X,T)
[US,USLEV,USINST] = undershoot(...)
[...] = undershoot(...,Name,Value)
undershoot(...)
```

Description

`US = undershoot(X)` returns the greatest deviations below the final state levels of each transition in the bilevel waveform, `X`. The undershoots, `US`, are expressed as a percentage of the difference between the state levels. See "Undershoot" on page 1-1256. The length of `US` corresponds to the number of transitions detected in the input signal. The sample instants in `X` correspond to the vector indices. To determine the transitions, `undershoot` estimates the state levels of the input waveform by a histogram method. `undershoot` identifies all regions that cross the upper-state boundary of the low state and the lower-state boundary of the high state. The low-state and high-state boundaries are expressed as the state level plus or minus a multiple of the difference between the state levels. See "State-Level Tolerances" on page 1-1258.

`US = undershoot(X,FS)` specifies the sampling frequency, `FS`, in hertz. The sampling frequency determines the sample instants corresponding to the elements in `X`. The first sample instant in `X` corresponds to t=0.

`US = undershoot(X,T)` specifies the sample instants, `T`, as a vector with the same number of elements as `X`.

`[US,USLEV,USINST] = undershoot(...)` returns the levels, `USLEV`, and sample instants, `USINST`, of the undershoots for each transition.

`[...] = undershoot(...,Name,Value)` returns the greatest deviations below the final state level with additional options specified by one or more `Name,Value` pair arguments.

`undershoot(...)` plots the bilevel waveform and marks the location of the undershoot of each transition as well as the lower- and upper reference-level instants and the associated reference levels. `undershoot`

also plots the state levels and associated lower- and upper-state boundaries.

Input Arguments

X

Bilevel waveform. `X` is a real-valued row or column vector.

FS

Sample rate in hertz.

T

Vector of sample instants. The length of `T` must equal the length of the bilevel waveform, `X`.

Name-Value Pair Arguments

'PctRefLevels'

Reference levels as a percentage of the waveform amplitude. The lower-state level is defined to be 0 percent. The upper-state level is defined to be 100 percent. The value of `'PCTREFLEVELS'` is a 2-element real row vector whose elements correspond to the lower and upper percent reference levels.

Default: `[10 90]`

'Region'

Specify the region over which to perform the undershoot computation. Valid values for `'Region'` are `'Preshoot'` or `'Postshoot'`. If you specify `'Preshoot'`, the end of the pretransition aberration region is defined as the last instant when the signal exits the first state. If you specify `'Postshoot'`, the start of the posttransition aberration region is defined as the instant when the signal enters the second state.

Default: `'Postshoot'`

'SeekFactor'

Aberration region duration. Specifies the duration of the region over which to compute the undershoot for each transition as a multiple of the corresponding transition duration. The edge of the waveform may be reached, or a complete intervening transition may be detected, before the duration aberration region duration elapses. In such cases, the duration is truncated to the edge of the waveform or the start of the intervening transition.

Default: 3

'StateLevels'

Lower- and upper-state levels. Specify the levels to use for the lower- and upper-state levels as a 2-element real row vector whose first and second elements correspond to the lower- and upper-state levels of the input waveform.

'Tolerance'

Specify the tolerance that the initial and final levels of each transition must be within the respective state levels. The `'Tolerance'` value is a scalar expressing a percentage of the difference between the upper- and lower-state levels. See "State-Level Tolerances" on page 1-1258.

Default: 2

Output Arguments

US

Undershoots expressed as a percentage of the state levels. The undershoot percentages are computed based on the greatest deviation from the final state level in each transition. By default undershoots are computed for posttransition aberration regions. See "Undershoot" on page 1-1256.

USLEV

Level of the pretransition or posttransition undershoot.

USINST

Sample instants of pretransition or posttransition undershoots. If you specify the sampling frequency or sampling instants, the undershoot instants are in seconds. If you do not specify the sampling frequency or sampling instants, the undershoot instants are the indices of the input vector.

Definitions

Undershoot

For a positive-going (positive-polarity) pulse, undershoot expressed as a percentage is

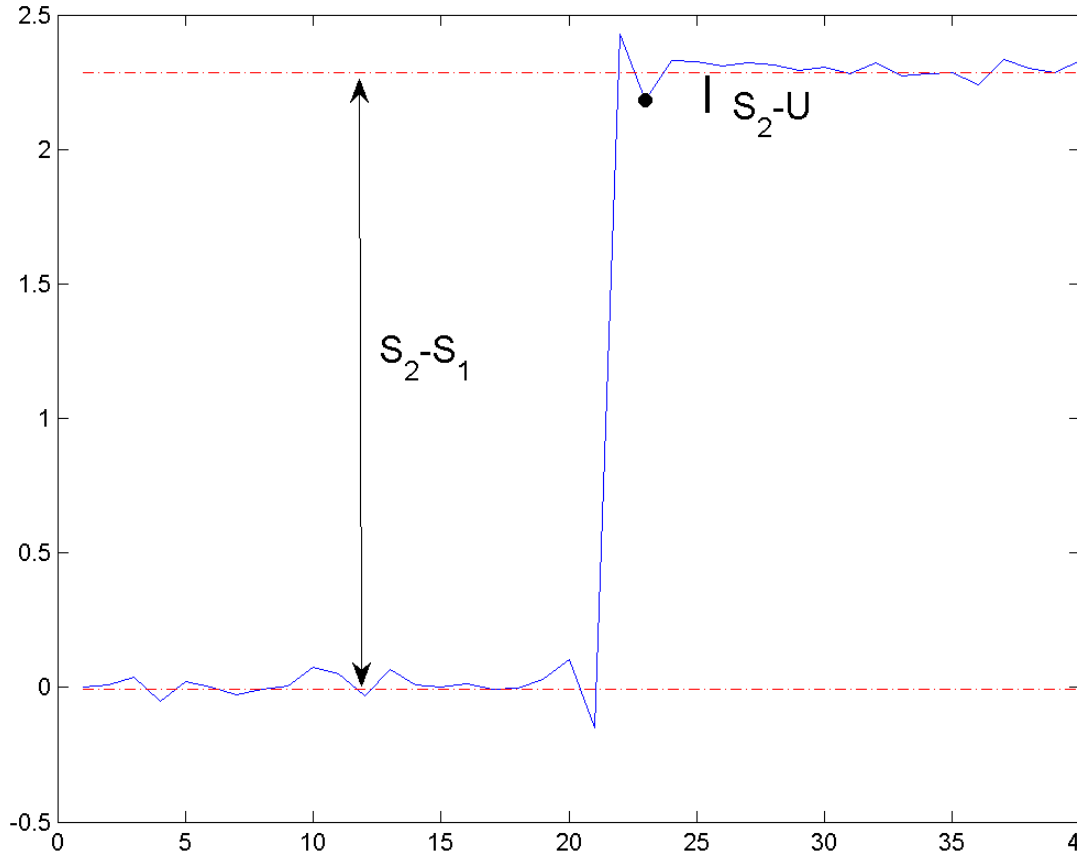
$$100\frac{(S_2 - U)}{(S_2 - S_1)}$$

where U is the greatest deviation below the high-state level, S_2 is the high state, and S_1 is the low state.

For a negative-going (negative-polarity) pulse, undershoot expressed as a percentage is

$$100\frac{(S_1 - U)}{(S_2 - S_1)}$$

The following figure illustrates the calculation of undershoot for a positive-going transition.



The red dashed lines indicate the estimated state levels. The double-sided black arrow depicts the difference between the high- and low-state levels. The solid black line indicates the difference between the high-state level and the undershoot value.

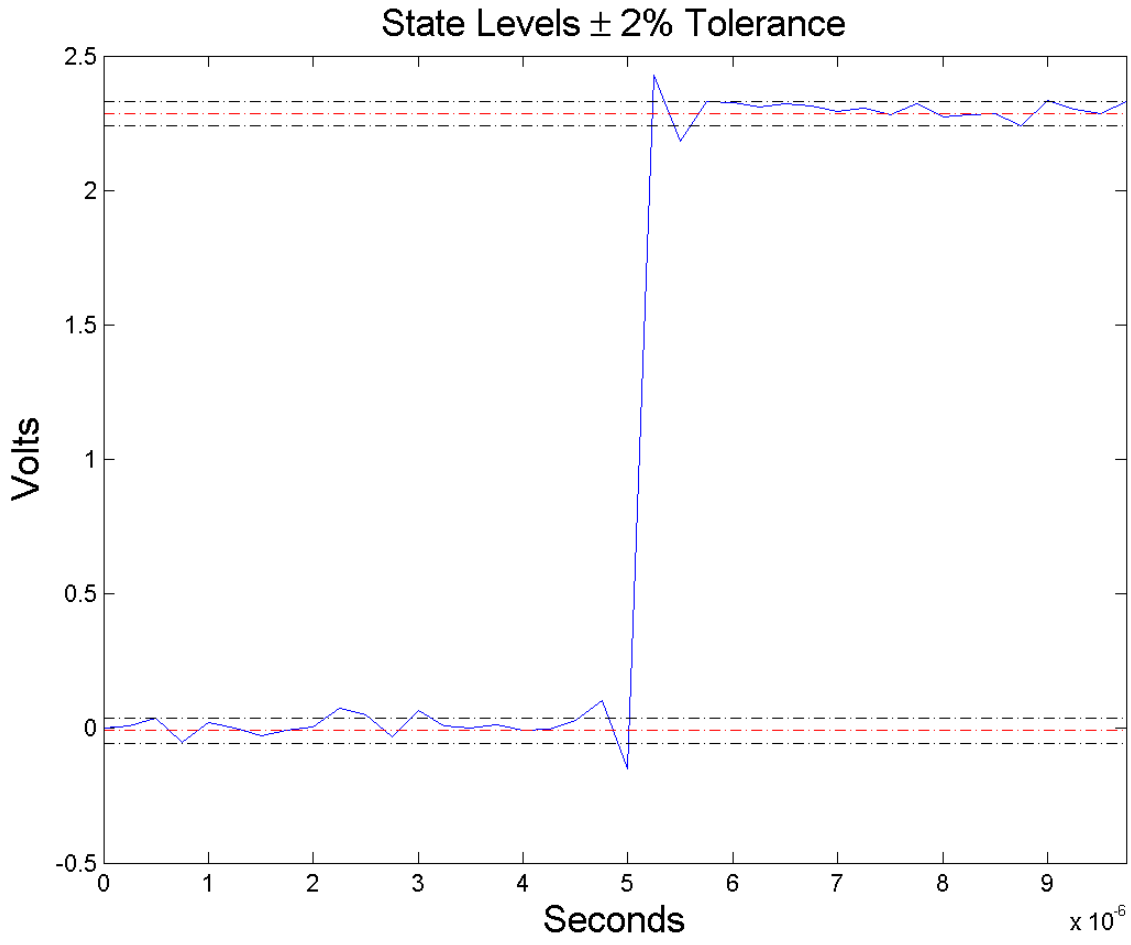
State-Level Tolerances

Each state level can have associated lower- and upper-state boundaries. These state boundaries are defined as the state level plus or minus a scalar multiple of the difference between the high state and low state. To provide a useful tolerance region, the scalar is typically a small number such as 2/100 or 3/100. In general, the α% tolerance region for the low state is defined as

$$S_1 \pm \frac{\alpha}{100}(S_2 - S_1)$$

where S_1 is the low-state level and S_2 is the high-state level. Replace the first term in the equation with S_2 to obtain the α% tolerance region for the high state.

The following figure illustrates lower and upper 2% state boundaries (tolerance regions) for a positive-polarity bilevel waveform. The red dashed lines indicate the estimated state levels.



Examples

Undershoot Percentage in Posttransition Aberration Region

Determine the maximum percent undershoot relative to the high-state level in a 2.3 V clock waveform.

Load the 2.3 V clock data. Plot the waveform. In this example, you see that the maximum undershoot in the posttransition region occurs near index 23.

```
load('transitionex.mat', 'x');
plot(x);
set(gca,'xtick',[1 5 12 19 23 30 40]);
grid on;
```

Determine the maximum percent undershoot.

```
us = undershoot(x);
```

Undershoot Percentage, Levels, and Sample Instant in Posttransition Aberration Region

Determine the maximum percent undershoot relative to the high-state level, the level of the undershoot, and the sample instant in a 2.3 V clock waveform.

Load the 2.3 V clock data with sampling instants. Plot the waveform. The clock data is sampled at 4 MHz.

```
load('transitionex.mat', 'x','t');
plot(t,x);
```

Determine the maximum percent undershoot, the level of the undershoot in volts, and the sampling instant where the maximum undershoot occurs. Plot the result.

```
[us,uslev,usinst] = undershoot(x,t);
plot(t.*1e6,x); xlabel('Microseconds');
hold on; grid on;
plot(usinst*1e6,uslev,'ro','markerfacecolor',[1 0 0]);
```

Undershoot Percentage, Levels, and Sample Instant in Pretransition Aberration Region

Determine the maximum percent undershoot relative to the low-state level, the level of the undershoot, and the sample instant in a 2.3

V clock waveform. Specify the `'Region'` as `'Preshoot'` to output pre-transition metrics.

Load the 2.3 V clock data with sampling instants. Plot the waveform. The clock data is sampled at 4 MHz.

```
load('transitionex.mat', 'x','t');
plot(t,x);
```

Determine the maximum percent undershoot, the level of the undershoot in volts, and the sampling instant where the maximum undershoot occurs. Plot the result.

```
load('transitionex.mat', 'x','t');
[us,uslev,usinst] = undershoot(x,t,'Region','Preshoot');
plot(t.*1e6,x); xlabel('Microseconds');
hold on; grid on;
plot(usinst*1e6,uslev,'ro','markerfacecolor',[1 0 0]);
```

References

[1] *IEEE Standard on Transitions, Pulses, and Related Waveforms*, IEEE Standard 181, 2003, pp. 15–17.

See Also

`overshoot` | `settlingtime` | `statelevels`

validstructures

Purpose

Structures for specification object with design method

Syntax

```
filtstruct = validstructures(D)
C = validstructures(D,METHOD)
Cs = validstructures(D,...,'SystemObject',sysobjflag)
```

Description

`filtstruct = validstructures(D)` returns a structure array containing all valid filter structures for the filter specification object, `D`, organized by design method. Each design method is a field in the structure array, `filtstruct`. The fields contain a cell array of strings.

`C = validstructures(D,METHOD)` returns the valid structures for the filter specification object, `D`, and the design method, `METHOD`, in a cell array of strings.

`Cs = validstructures(D,...,'SystemObject',sysobjflag)` returns the valid structures for designing a filter System object when *`sysobjflag`* is `true`. To use System objects, you must have the DSP System Toolbox product installed. When *`sysobjflag`* is `false`, the function returns valid structures for designing `dfilt` and `mfilt` objects, as described previously. Design methods and design options for filter System objects are not necessarily the same as those for `dfilt` and `mfilt` objects.

Examples

Design a default lowpass filter specification object. Return all valid design methods and structures in a structure array. Display the fieldnames to see all valid design methods. Display the valid filter structures for the `equiripple` field.

```
D = fdesign.lowpass;
filtstruct = validstructures(D);
fieldnames(filtstruct)
filtstruct.equiripple
```

Create a highpass filter of order 50 with a 3-dB frequency of 0.2. Obtain the available structures for a Butterworth design.

```
D = fdesign.highpass('N,F3dB',50,0.2);
C = validstructures(D,'butter');
```

If you have DSP System Toolbox software installed, use the `'SystemObject',sysobjflag` syntax to return valid structures for a filter System object:

```
Cs = validstructures(D,'butter','SystemObject',true);
```

See Also

`design` | `designmethods` | `designopts` | `fdesign`

Purpose

Voltage controlled oscillator

Syntax

```
y = vco(x,fc,fs)
y = vco(x,[Fmin Fmax],fs)
```

Description

`y = vco(x,fc,fs)` creates a signal that oscillates at a frequency determined by the real input vector or array `x` with sampling frequency `fs`. `fc` is the carrier or reference frequency; when `x` is 0, `y` is an `fc` Hz cosine with amplitude 1 sampled at `fs` Hz. `x` ranges from `-1` to `1`, where `x = -1` corresponds to 0 frequency output, `x = 0` corresponds to `fc`, and `x = 1` corresponds to `2*fc`. Output `y` is the same size as `x`.

`y = vco(x,[Fmin Fmax],fs)` scales the frequency modulation range so that ±1 values of `x` yield oscillations of `Fmin` Hz and `Fmax` Hz respectively. For best results, `Fmin` and `Fmax` should be in the range 0 to `fs/2`.

By default, `fs` is 1 and `fc` is `fs/4`.

If `x` is a matrix, `vco` produces a matrix whose columns oscillate according to the columns of `x`.

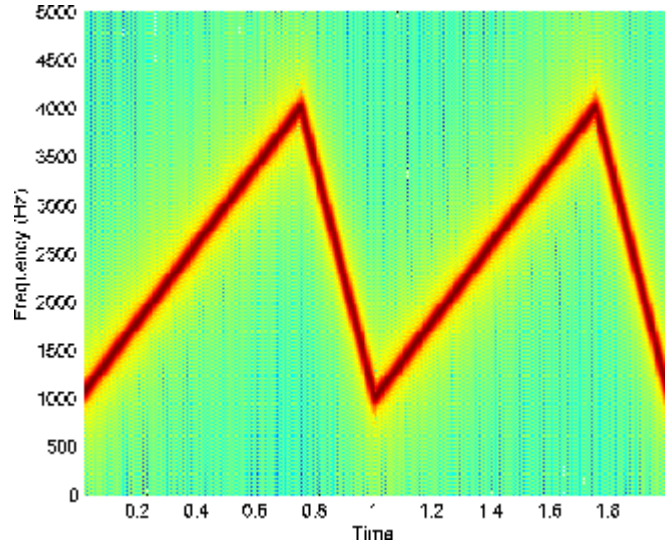
Examples

Generate two seconds of a signal sampled at 10,000 samples/second whose instantaneous frequency is a triangle function of time:

```
fs = 10000;
t = 0:1/fs:2;
x = vco(sawtooth(2*pi*t,0.75),[0.1 0.4]*fs,fs);
```

Plot the spectrogram of the generated signal:

```
spectrogram(x,kaiser(256,5),220,512,fs,'yaxis')
```



Algorithms

`vco` performs FM modulation using the `modulate` function.

Diagnostics

If any values of `x` lie outside [-1, 1], `vco` gives the following error message.

```
X outside of range [-1,1].
```

See Also

`demod` | `modulate`

window

Purpose

Window function gateway

Syntax

```
window
w = window(fhandle,n)
w = window(fhandle,n,winopt)
```

Description

`window` opens the Window Design and Analysis Tool (`wintool`).

`w = window(fhandle,n)` returns the `n`-point window, specified by its function handle, `fhandle`, in column vector `w`. Function handles are window function names preceded by an @.

```
@barthannwin
@bartlett
@blackman
@blackmanharris
@bohmanwin
@chebwin
@flattopwin
@gausswin
@hamming
@hann
@kaiser
@nuttallwin
@parzenwin
@rectwin
@taylorwin
@triang
@tukeywin
```

Note For `chebwin`, `kaiser`, and `tukeywin`, you must use include a window parameter using the syntax below.

For more information on each window function and its option(s), refer to its reference page.

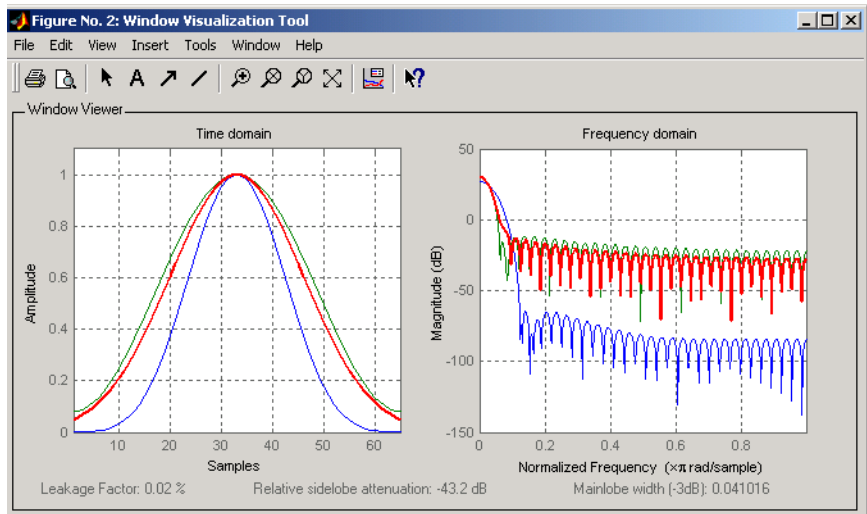
`w = window(fhandle,n,winopt)` returns the window specified by its function handle, `fhandle`, and its *`winopt`* value or sampling flag string. For `chebwin`, `kaiser`, and `tukeywin`, you must enter a *`winopt`* value. For the other windows listed below, *`winopt`* values are optional.

Window	winopt Description	winopt Value
`blackman`	sampling flag string	`'periodic'` or `'symmetric'`
`chebwin`	sidelobe attenuation relative to mainlobe	numeric
`flattopwin`	sampling flag string	`'periodic'` or `'symmetric'`
`gausswin`	alpha value (reciprocal of standard deviation)	numeric
`hamming`	sampling flag string	`'periodic'` or `'symmetric'`
`hann`	sampling flag string	`'periodic'` or `'symmetric'`
`kaiser`	beta value	numeric
`taylorwin`	1. number of sidelobes 2. maximum sidelobe level in dB relative to mainlobe peak	1. integer greater than or equal to 1 2. negative value
`tukeywin`	ratio of taper to constant sections	numeric

Examples

Create Blackman Harris, Hamming, and Gaussian windows and plot them in the same WVTool.

```
N = 65;
w = window(@blackmanharris,N);
w1 = window(@hamming,N);
w2 = window(@gausswin,N,2.5);
wvtool(w,w1,w2)
```



See Also

barthannwin | bartlett | blackman | blackmanharris | bohmanwin | chebwin | flattopwin | gausswin | hamming | hann | kaiser | nuttallwin | parzenwin | rectwin | triang | taylorwin | tukeywin

Purpose

FIR filter using windowed impulse response

Syntax

```
h = window(d,'window',fcnhndl)
h = window(d,win)
```

description

Note This is a description of the overloaded method used in conjunction with `fdesign` to design a filter from a filter specification object. To access the window function gateway see `window`.

`h = window(d,'window',fcnhndl)` designs an FIR filter using the specifications in filter specification 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`.

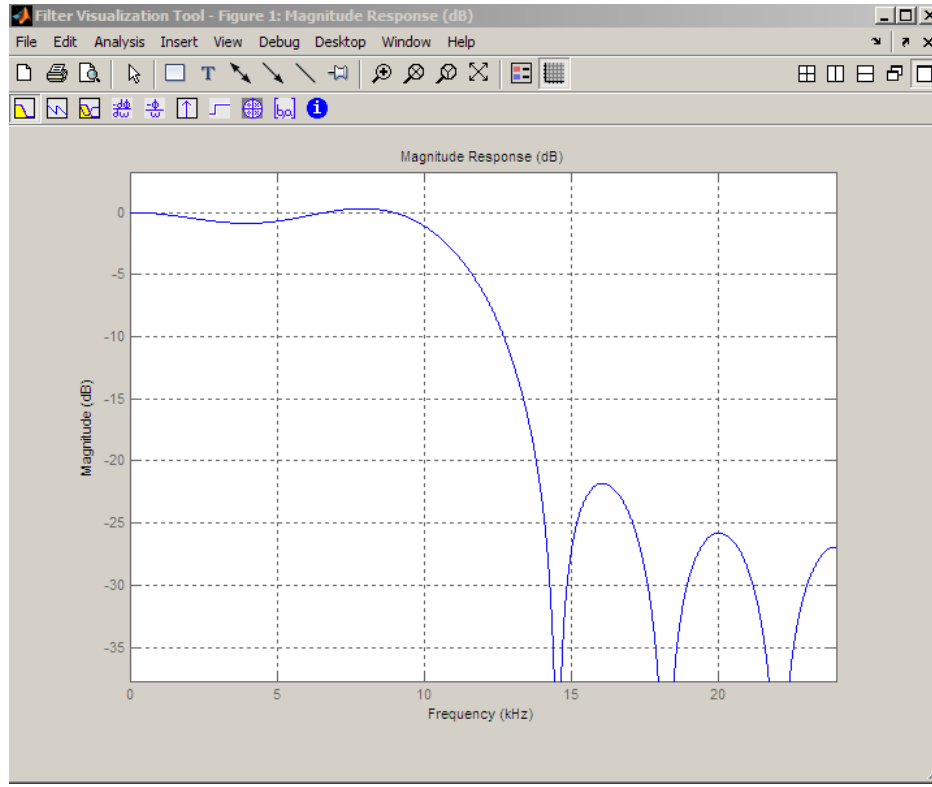
`fcnhndl` is a handle to a filter design function that returns a window vector, such as the `hamming` or `blackman` functions. `fcnarg` is an optional argument that returns a window. You pass the function to `window`. Refer to example 1 in the following section to see the function argument used to design 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.

Examples

Construct a lowpass filter specification object of order 10 with a cutoff frequency of 12 kilohertz. We use a sampling frequency of 48 kilohertz. Next we use a function handle to the function Kaiser to provide the window.

```
d=fdesign.lowpass('n,fc',10,12000,48000);
Hd=window(d,'window',@kaiser);
fvtool(Hd);
```



See Also

`design` | `designmethods` | `fdesign`

Purpose

Open Window Design and Analysis Tool

Syntax

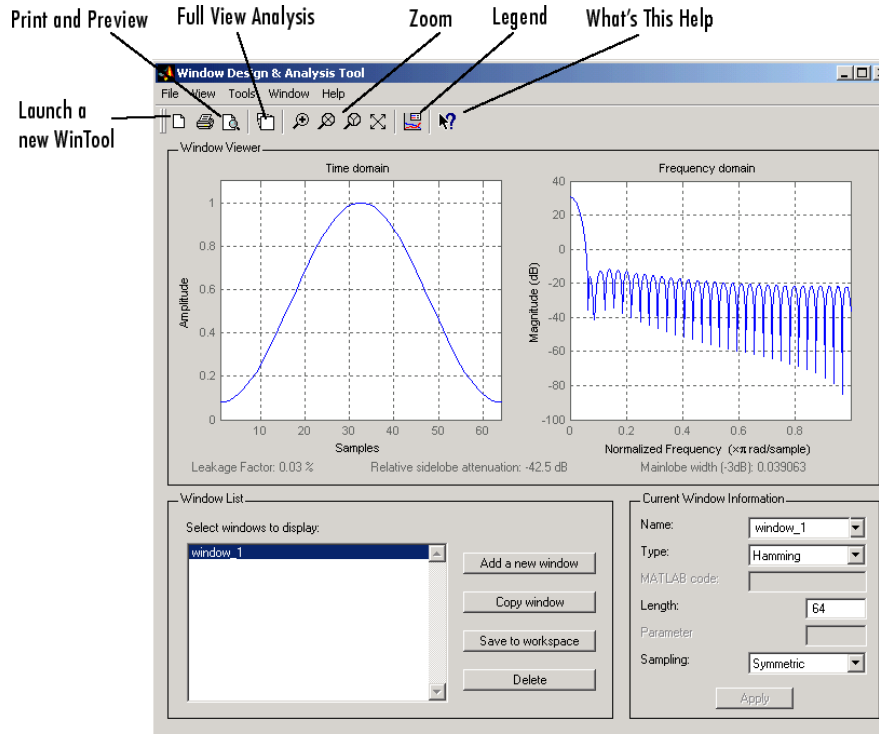
```
wintool
wintool(obj1,obj2,...)
```

Description

`wintool` opens the Window Design and Analysis Tool (WinTool), a graphical user interface (GUI) for designing and analyzing spectral windows. It opens with a default 64-point Hamming window.

`wintool(obj1,obj2,...)` opens WinTool with the `sigwin` window object(s) specified in `obj1`, `obj2`, etc.

Note A related tool, `wvtool`, is available for displaying, annotating, or printing windows.



`wintool` has three panels:

- Window Viewer displays the time domain and frequency domain representations of the selected window(s). The currently active window is shown in bold. Three window measurements are shown below the plots.
  - Leakage factor — ratio of power in the sidelobes to the total window power
  - Relative sidelobe attenuation — difference in height from the mainlobe peak to the highest sidelobe peak
  - Mainlobe width (-3dB) — width of the mainlobe at 3 dB below the mainlobe peak

- Window List lists the windows available for display in the Window Viewer. Highlight one or more windows to display them. The Window List buttons are:
  - **Add a new window** — Adds a default Hamming window with length 64 and symmetric sampling. You can change the information for this window by applying changes made in the **Current Window Information** panel.
  - **Copy window** — Copies the selected window(s).
  - **Save to workspace** — Saves the selected window(s) as vector(s) to the MATLAB workspace. The name of the window in `wintool` is used as the vector name.
  - **Delete** — Removes the selected window(s) from the window list.
- *Current Window Information* displays information about the currently active window. The active window name is shown in the `Name` field. To make another window active, select its name from the **Name** menu.

Window Parameters

Each window is defined by the parameters in the Current Window Information panel. You can change the current window's characteristics by changing its parameters and clicking **Apply**. The parameters of the current window are

- **Name** — Name of the window. The name is used for the legend in the Window Viewer, in the Window List, and for the vector saved to the workspace. You can either select a name from the menu or type the desired name in the edit box.
- **Type** — Algorithm for the window. Select the type from the menu. All Signal Processing Toolbox windows are available.
- **MATLAB code** — Any valid MATLAB expression that returns a vector defining the window if `Type = User Defined`.
- **Length** — Number of samples.

- **Parameter** — Additional parameter for windows that require it, such as Chebyshev, which requires you to specify the sidelobe attenuation. Note that the title "Parameter" changes to the appropriate parameter name.
- **Sampling** — Type of sampling to use for generalized cosine windows (Hamming, Hann, and Blackman) — `Periodic` or `Symmetric`. `Periodic` computes a length `n+1` window and returns the first `n` points, and `Symmetric` computes and returns the `n` points specified in `Length`.

WinTool Menus

In addition to the usual menus items, `wintool` contains these `wintool`-specific menu commands:

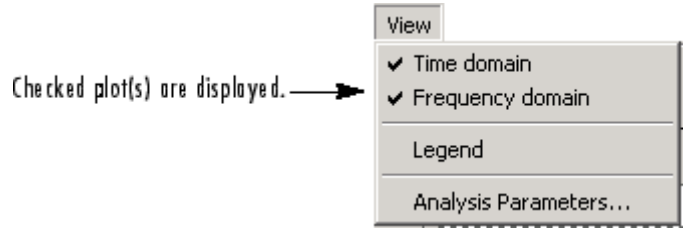
File menu:

- **Export** — Exports window coefficient vectors to the MATLAB workspace, a text file, or a MAT-file.

  In the **Window List** in WinTool, highlight the window(s) you want to export and then select **File > Export**. For exporting to the workspace or a MAT-file, specify the variable name for each window coefficient or object. To overwrite variables in the workspace, select the Overwrite variables check box.
- **Full View Analysis** — Copies the windows shown in both plots to a separate `wvtool` figure window. This is useful for printing and annotating. This option is also available with the Full View Analysis toolbar button.

View menu:

- **Time domain** — Select to show the time domain plot in the Window Viewer panel.
- **Frequency domain** — Select to show the frequency domain plot in the Window Viewer panel.



- **Legend** — Toggles the window name legend on and off. This option is also available with the Legend toolbar button.
- **Analysis Parameters** — Controls the response plot parameters, including number of points, range, *x*- and *y*-axis units, sampling frequency, and normalized magnitude.

  You can also access the Analysis Parameters by right-clicking the *x*-axis label of a plot in the Window Viewer panel. The *x*-axis units for the time domain plot depend on the selected `Sampling Frequency` units.

Frequency Domain	Time Domain
Hz	sec
kHz	ms
MHz	µs
GHz	picosec

Tools menu:

- **Zoom In** — Zooms in along both *x*- and *y*-axes.
- **Zoom X** — Zooms in along the *x*-axis only. Drag the mouse in the *x* direction to select the zoom area.
- **Zoom Y** — Zooms in along the y-axis only. Drag the mouse in the *y* direction to select the zoom area.
- **Full View** — Returns to full view.

See Also

window | wvtool

Purpose

Open Window Visualization Tool

Syntax

```
wvtool(WindowVector)
wvtool(WinObj)
wvtool(WindowVector1,WinObj1,...,WinObjN,WindowVectorN)
H = wvtool(...)
```

Description

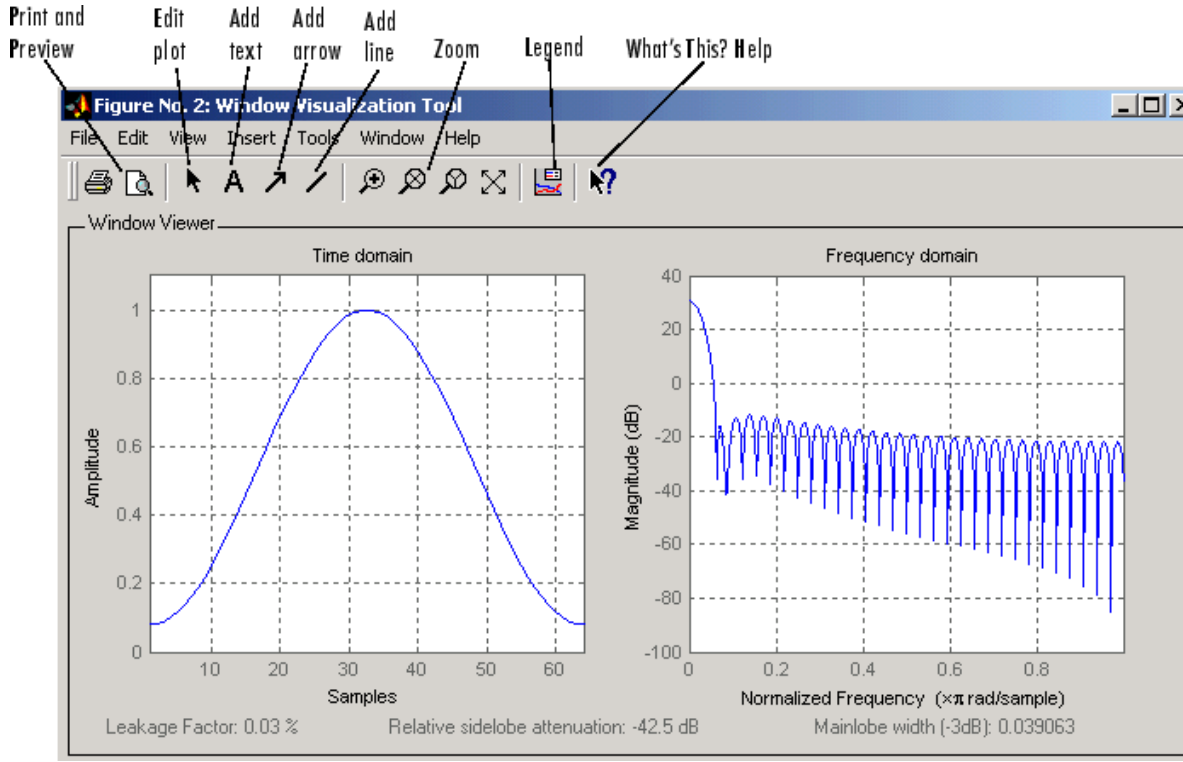
`wvtool(WindowVector)` opens the Window Visualization Tool (WVTool) with time and frequency domain plots of the window vector specified in `WindowVector`. `WindowVector` must be a real-valued row or column vector. By default, the frequency domain plot is the magnitude squared of the Fourier transform of the window vector in decibels (dB). You can generate window vectors for a number of common window functions using the Signal Processing Toolbox software. See `window` for a list of supported window functions.

`wvtool(WinObj)` opens WVTool with time and frequency domain plots of the `sigwin` object `WinObj`. See `sigwin` for a list of supported signal processing window objects.

`wvtool(WindowVector1,WinObj1,...,WinObjN,WindowVectorN)` opens WVTool with time and frequency domain plots of the window vectors or window objects specified in `WindowVector1`,`WinObj1`,...,`WinObjN`, `WindowVectorN`.

`H = wvtool(...)` returns the figure handle `H`.

Note A related tool, `wintool`, is available for designing and analyzing windows.



Note If you launch WVTool from FDATool, an **Add/Replace** icon, which controls how new windows are added from FDATool, appears on the toolbar.

WVTool Menus

In addition to the usual menus items, `wvtool` contains these `wvtool`-specific menu commands:

File menu:

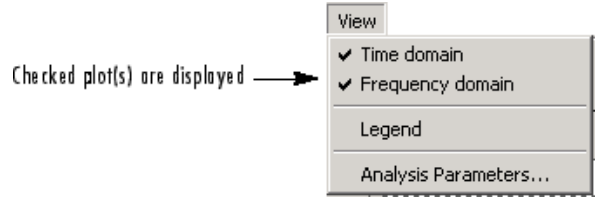
- **Export** — Exports the displayed plot(s) to a graphic file.

Edit menu:

- **Copy figure** — Copies the displayed plot(s) to the clipboard (available only on Windows platforms).
- **Copy options** — Displays the Preferences dialog box (available only on Windows platforms).
- **Figure**, **Axes**, and **Current Object Properties** — Displays the Property Editor.

View menu:

- **Time domain** — Check to show the time domain plot.
- **Frequency domain** — Check to show the frequency domain plot.



- **Legend** — Toggles the window name legend on and off. This option is also available with the **Legend** toolbar button.
- *Analysis Parameters* — Controls the response plot parameters, including number of points, range, *x*- and *y*-axis units, sampling frequency, and normalized magnitude.

  You can also access the Analysis Parameters by right-clicking the *x*-axis label of a plot in the Window Viewer panel.
- **Insert** menu:

You use the **Insert** menu to add labels, titles, arrows, lines, text, and axes to your plots.

Tools menu:

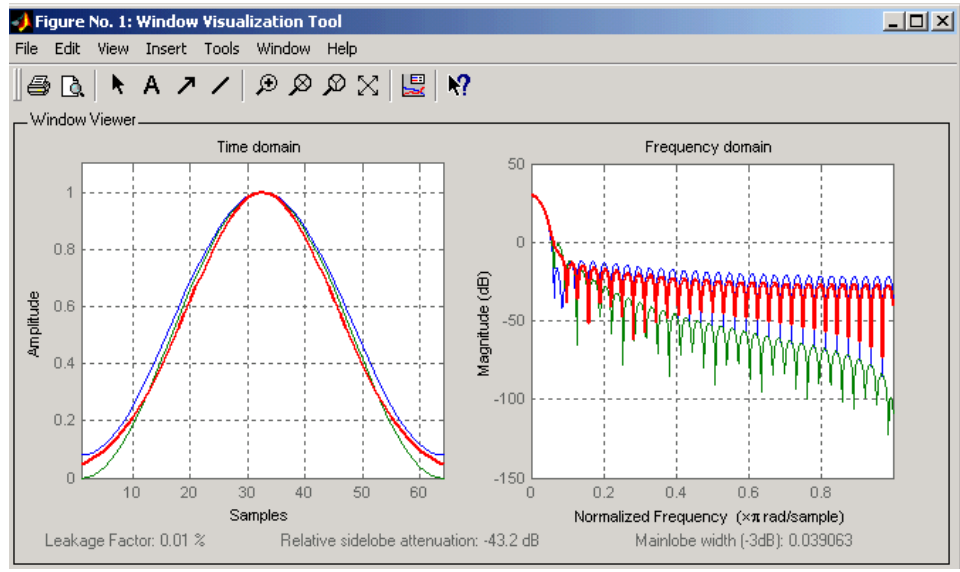
- **Edit Plot** — Turns on plot editing mode
- **Zoom In** — Zooms in along both *x*- and *y*-axes.

- **Zoom X** — Zooms in along the *x*-axis only. Drag the mouse in the *x* direction to select the zoom area.
- **Zoom Y** — Zooms in along the y-axis only. Drag the mouse in the *y* direction to select the zoom area.
- **Full View** — Returns to full view.

Examples

Compare Hamming, Hann, and Gaussian windows:

```
wvtool(hamming(64),hann(64),gausswin(64))
```



Compare Kaiser window objects with different beta values:

```
H = sigwin.kaiser(128,1.5);
% Kaiser window with beta=4.5
H1 = sigwin.kaiser(128,4.5);
wvtool(H,H1)
```

See Also

`fdatool` | `sigwin` | `window` | `wintool`

Purpose

Cross-correlation

Syntax

```
c = xcorr(x,y)
c = xcorr(x)
c = xcorr(x,y,'option')
c = xcorr(x,'option')
c = xcorr(x,y,maxlags)
c = xcorr(x,maxlags)
c = xcorr(x,y,maxlags,'option')
c = xcorr(x,maxlags,'option')
[c,lags] = xcorr(...)
[c,lags] = xcorr(gpuArrayX,gpuArrayY,maxlags,'option')
```

Description

`xcorr` estimates the cross-correlation sequence of a random process. Autocorrelation is handled as a special case.

The true cross-correlation sequence is

$$R_{xy}(m) = E\{x_{n+m}y_n^*\} = E\{x_n y_{n-m}^*\}$$

where x_n and y_n are jointly stationary random processes, $-\infty < n < \infty$, and $E\{\cdot\}$ is the expected value operator. `xcorr` must estimate the sequence because, in practice, only a finite segment of one realization of the infinite-length random process is available.

`c = xcorr(x,y)` returns the cross-correlation sequence in a length $2*N-1$ vector, where `x` and `y` are length N vectors ($N>1$). If `x` and `y` are not the same length, the shorter vector is zero-padded to the length of the longer vector.

By default, `xcorr` computes raw correlations with no normalization.

$$\hat{R}xy(m) = \begin{cases} \sum_{n=0}^{N-m-1} x_{n+m}y_n^* & m \geq 0 \\ \hat{R}y\,x(-m)^* & m < 0 \end{cases}$$

The output vector c has elements given by:

$$c(m) = \hat{R}_{xy}(m - N) \quad m = 1, 2, \ldots 2N - 1$$

In general, the correlation function requires normalization to produce an accurate estimate (see below).

`c = xcorr(x)` is the autocorrelation sequence for the vector `x`. If `x` is an N-by-P matrix, c is a matrix with $2N$-1 rows whose P^2 columns contain the cross-correlation sequences for all combinations of the columns of `x`. For more information on matrix processing with `xcorr`, see "Multiple Channels".

`xcorr` produces correlations identically equal to 1.0 at zero lag only when you perform an autocorrelation and only when you set the `'coeff'` option. For example,

```
x=0:0.01:10;
X = sin(x);
[r,lags]=xcorr(X,'coeff');
max(r)
```

`c = xcorr(x,y,'option')` specifies a normalization option for the cross-correlation, where `'option'` is

- `'biased'`: Biased estimate of the cross-correlation function

$$\hat{R}_{xy,biased}(m) = \frac{1}{N}\hat{R}_{xy}(m)$$

- `'unbiased'`: Unbiased estimate of the cross-correlation function

$$\hat{R}_{xy,unbiased}(m) = \frac{1}{N - |m|}\hat{R}_{xy}(m)$$

- `'coeff'`: Normalizes the sequence so the autocorrelations at zero lag are identically 1.0.

- `'none'`, to use the raw, unscaled cross-correlations (default)

See [1] for more information on the properties of biased and unbiased correlation estimates.

`c = xcorr(x,'option')` specifies one of the above normalization options for the autocorrelation.

`c = xcorr(x,y,maxlags)` returns the cross-correlation sequence over the lag range `[-maxlags:maxlags]`. Output `c` has length `2*maxlags+1`.

`c = xcorr(x,maxlags)` returns the autocorrelation sequence over the lag range `[-maxlags:maxlags]`. Output `c` has length `2*maxlags+1`. If `x` is an *N*-by-*P* matrix, `c` is a matrix with `2*maxlags+1` rows whose P^2 columns contain the autocorrelation sequences for all combinations of the columns of `x`.

`c = xcorr(x,y,maxlags,'option')` specifies both a maximum number of lags and a scaling option for the cross-correlation.

`c = xcorr(x,maxlags,'option')` specifies both a maximum number of lags and a scaling option for the autocorrelation.

`[c,lags] = xcorr(...)` returns a vector of the lag indices at which `c` was estimated, with the range `[-maxlags:maxlags]`. When `maxlags` is not specified, the range of `lags` is `[-N+1:N-1]`.

In all cases, the cross-correlation or autocorrelation computed by `xcorr` has the zeroth lag in the middle of the sequence, at element or row `maxlags+1` (element or row `N` if `maxlags` is not specified).

`[c,lags] = xcorr(gpuArrayX,gpuArrayY,maxlags,'option')` returns the autocorrelation or cross-correlation sequence for input objects of class `gpuArray`. See "Use gpuArray Data" for details on gpuArray objects. Using `xcorr` with `gpuArray` objects requires Parallel Computing Toolbox software and a CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details. The returned autocorrelation or cross-correlation sequence, `c`, is a gpuArray object.

"GPU Acceleration for Autocorrelation Sequence Estimation" on page 1-1285 shows you how to compute the autocorrelation sequence on the GPU.

Examples

The second output, `lags`, is useful for plotting the cross-correlation or autocorrelation. For example, the estimated autocorrelation of zero-mean Gaussian white noise $c_{ww}(m)$ can be displayed for $-10 \le m \le 10$ using:

```
ww = randn(1000,1);
[c_ww,lags] = xcorr(ww,10,'coeff');
stem(lags,c_ww)
```

Swapping the `x` and `y` input arguments reverses (and conjugates) the output correlation sequence. For row vectors, the resulting sequences are reversed left to right; for column vectors, up and down. The following example illustrates this property (`mat2str` is used for a compact display of complex numbers):

```
x = [1,2i,3]; y = [4,5,6];
[c1,lags] = xcorr(x,y);
c1 = mat2str(c1,2), lags
c2 = conj(fliplr(xcorr(y,x)));
c2 = mat2str(c2,2)
```

For the case where input argument `x` is a matrix, the output columns are arranged so that extracting a row and rearranging it into a square array produces the cross-correlation matrix corresponding to the lag of the chosen row. For example, the cross-correlation at zero lag can be retrieved by:

```
X = randn(2,2);
[M,P] = size(X);
c = xcorr(X);
c0 = zeros(P); c0(:) = c(M,:)    % Extract zero-lag row
```

You can calculate the matrix of correlation coefficients that the MATLAB function `corrcoef` generates by substituting:

```
c = xcov(X,'coef')
```

in the last example. The function `xcov` subtracts the mean and then calls `xcorr`.

Use `fftshift` to move the second half of the sequence starting at the zeroth lag to the front of the sequence. `fftshift` swaps the first and second halves of a sequence.

GPU Acceleration for Autocorrelation Sequence Estimation

The following example requires Parallel Computing Toolbox software and a CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details.

Create a signal consisting of a 10-Hz sine wave in additive noise. Use `gpuArray` to create a gpuArray object stored on your computer's GPU.

```
t = 0:0.001:10-0.001;
x = cos(2*pi*10*t)+randn(size(t));
X = gpuArray(x);
```

Compute the normalized autocorrelation sequence to lag 200.

```
[xc,lags] = xcorr(X,200,'coeff');
```

The output, `xc`, is a gpuArray object. Use `gather` to transfer the data from the GPU to the MATLAB workspace as a double-precision vector.

```
xc = gather(xc);
```

Algorithms

For more information on estimating covariance and correlation functions, see [1].

References

[1] Orfanidis, S.J., *Optimum Signal Processing. An Introduction. 2nd Edition*, Prentice-Hall, Englewood Cliffs, NJ, 1996.

See Also

`conv` | `corrcoef` | `cov` | `xcorr2` | `xcov`

Purpose

2–D cross-correlation

Syntax

```
C = xcorr2(A,B)
A = xcorr2(A)
C = xcorr2(gpuArrayA,gpuArrayB)
```

Description

`C = xcorr2(A,B)` returns the cross-correlation of matrices `A` and `B` with no scaling. `xcorr2` is the two-dimensional version of `xcorr`. It has its maximum value when the two matrices are aligned so that they are shaped as similarly as possible.

If matrix *A* has dimensions (*Ma*, *Na*) and matrix *B* has dimensions (*Mb*, *Nb*), The equation for the two-dimensional discrete cross-correlation is

$$C(i,j) = \sum_{m=0}^{(Ma-1)} \sum_{n=0}^{(Na-1)} A(m,n) \cdot conj(B(m+i,n+j))$$

where $0 \le i < Ma + Mb - 1$ and $0 \le j < Na + Nb - 1$.

`A = xcorr2(A)` is the autocorrelation matrix of input matrix `A`. It is identical to `xcorr2(A,A)`.

`C = xcorr2(gpuArrayA,gpuArrayB)` returns the cross-correlation of the input matrices of class `gpuArray`. See "Use gpuArray Data" for details on gpuArray objects. Using `xcorr2` with `gpuArray` objects requires Parallel Computing Toolbox software and a CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details. The output cross-correlation matrix, `C`, is a gpuArray object. See "GPU Acceleration for Cross-Correlation Matrix Computation" on page 1-1292 for an example of using the GPU to compute the cross-correlation.

Examples

Output Matrix Size

If matrix I1 has dimensions (4,3) and matrix I2 has dimensions (2,2), the following equations determine the number of rows and columns of the output matrix:

$$C_{\text{full}_{\text{rows}}} = I1_{\text{rows}} + I2_{\text{rows}} - 1 = 4 + 2 - 1 = 5$$
$$C_{\text{full}_{\text{columns}}} = I1_{\text{columns}} + I2_{\text{columns}} - 1 = 3 + 2 - 1 = 4$$

The resulting matrix is

$$C_{\text{full}} = \begin{bmatrix} c_{00} & c_{01} & c_{02} & c_{03} \\ c_{10} & c_{11} & c_{12} & c_{13} \\ c_{20} & c_{21} & c_{22} & c_{23} \\ c_{30} & c_{31} & c_{32} & c_{33} \\ c_{40} & c_{41} & c_{42} & c_{43} \end{bmatrix}$$

Computing a Specific Element

$$C_{\text{valid}_{\text{columns}}} = I1_{\text{columns}} - I2_{\text{columns}} + 1 = 2$$

In cross-correlation, the value of an output element is computed as a weighted sum of neighboring elements. For example, suppose the first input matrix represents an image and is defined as

```
I1 = [17  24   1   8  15
      23   5   7  14  16
       4   6  13  20  22
      10  12  19  21   3
      11  18  25   2   9]
```

The second input matrix also represents an image and is defined as

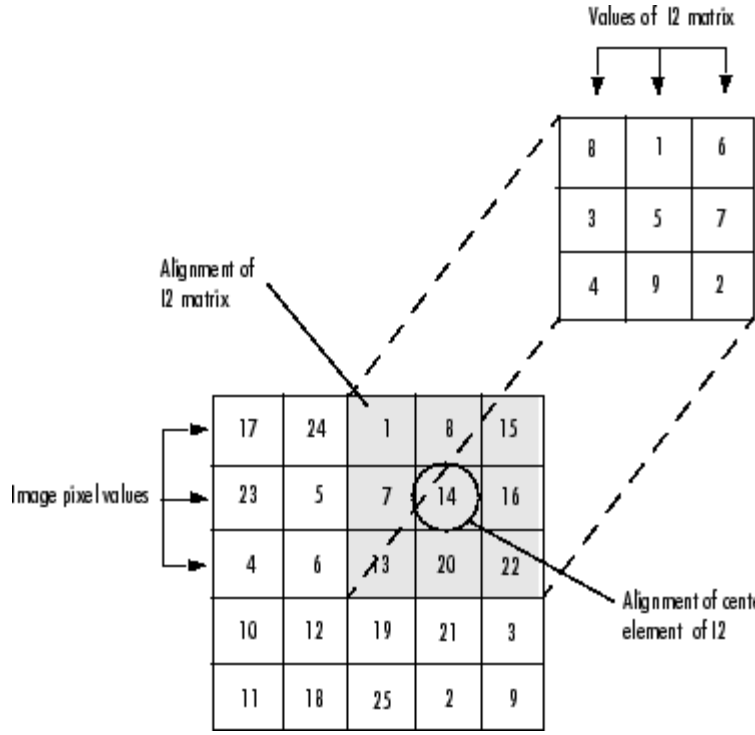
```
I2 = [8   1   6
      3   5   7
      4   9   2]
```

The following figure shows how to compute the (2,4) output element (zero-based indexing) using these steps:

1. Slide the center element of I2 so that lies on top of the (1,3) element of I1.
2. Multiply each weight in I2 by the element of I1 underneath.
3. Sum the individual products from step 2.

The (2,4) output element from the cross-correlation is

$$1 \cdot 8 + 8 \cdot 1 + 15 \cdot 6 + 7 \cdot 3 + 14 \cdot 5 + 16 \cdot 7 + 13 \cdot 4 + 20 \cdot 9 + 22 \cdot 2 = 585$$



The normalized cross-correlation of the (2,4) output element is

585/sqrt(sum(dot(I1p,I1p))*sum(dot(I2,I2))) = 0.8070

where I1p = [1 8 15; 7 14 16; 13 20 22].

Recovery of Template Shift with Cross-Correlation

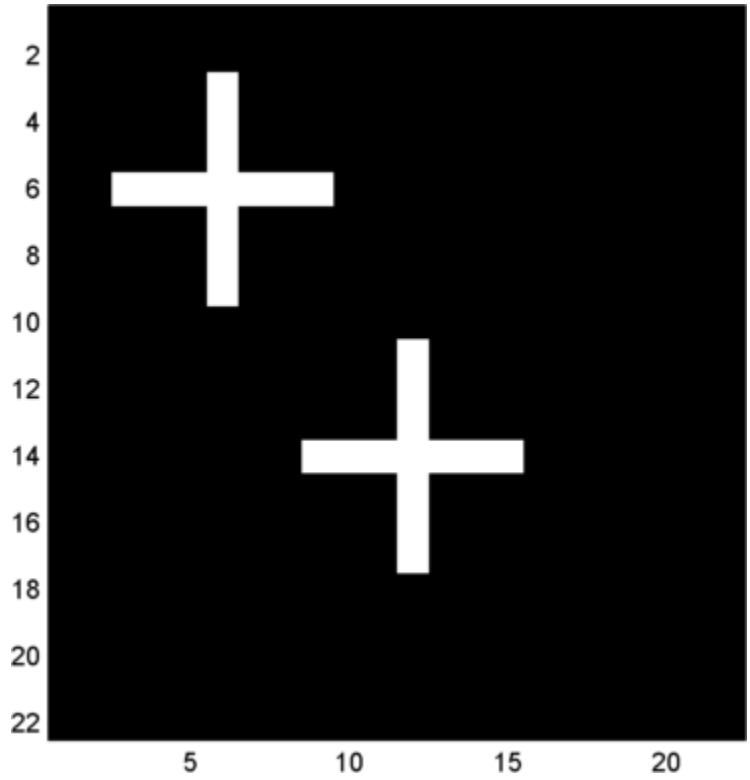
Shift a template by a known amount and recover the shift using cross-correlation.

Create a template in an 11-by-11 matrix. Create a 22-by-22 matrix and shift the original template by 8 along the row dimension and 6 along the column dimension.

```
template = .2*ones(11);
template(6,3:9) = .6;
template(3:9,6) = .6;
offsetTemplate = .2*ones(22);
offset = [8 6];
offsetTemplate( (1:size(template,1))+offset(1),...
                   (1:size(template,2))+offset(2) ) = template;
```

Plot the original and shifted templates.

```
imagesc(offsetTemplate); colormap gray;
hold on;
imagesc(template);
```



Cross-correlate the two matrices and find the maximum absolute value of the cross-correlation. Use the position of the maximum absolute value to determine the shift in the template. Check the result against the known shift.

```
cc = xcorr2(offsetTemplate,template);
[max_cc, imax] = max(abs(cc(:)));
[ypeak, xpeak] = ind2sub(size(cc),imax(1));
corr_offset = [ (ypeak-size(template,1)) (xpeak-size(template,2)) ];
isequal(corr_offset,offset)
```

The returned 1 indicates that the shift obtained the cross-correlation equals the known the template shift in both the row and column dimension.

GPU Acceleration for Cross-Correlation Matrix Computation

The following example requires Parallel Computing Toolbox software and a CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details.

Repeat the example "Recovery of Template Shift with Cross-Correlation" on page 1-1290. For convenience, the code to create the original and shifted templates is repeated.

```
template = .2*ones(11);
template(6,3:9) = .6;
template(3:9,6) = .6;
offsetTemplate = .2*ones(22);
offset = [8 6];
offsetTemplate( (1:size(template,1))+offset(1),...
                    (1:size(template,2))+offset(2) ) = template;
```

Put the original and shifted template matrices on your GPU using `gpuArray` objects.

```
template = gpuArray(template);
offsetTemplate = gpuArray(offsetTemplate);
```

Compute the cross-correlation on the GPU.

```
cc = xcorr2(offsetTemplate,template);
```

Return the result to the MATLAB workspace using `gather`, use the maximum absolute value of the cross-correlation to determine the shift, and compare the result with the known shift.

```
cc = gather(cc);
[max_cc, imax] = max(abs(cc(:)));
```

```
[ypeak, xpeak] = ind2sub(size(cc),imax(1));
corr_offset = [ (ypeak-size(template,1)) (xpeak-size(template,2)) ];
isequal(corr_offset,offset)
```

See Also

conv2 | filter2 | xcorr

Purpose

Cross-covariance

Syntax

```
c = xcov(x,y)
c = xcov(x)
c = xcov(x,'option')
[c,lags] = xcov(x,y,maxlags)
[c,lags] = xcov(x,maxlags)
[c,lags] = xcov(x,maxlags)
[c,lags] = xcov(x,y,maxlags,'option')
[c,lags] = xcov(gpuArrayX,gpuArrayY,maxlags,'option')
```

Description

`xcov` estimates the cross-covariance sequence of random processes. Autocovariance is handled as a special case.

The true cross-covariance sequence is the cross-correlation of mean-removed sequences

$$\phi_{xy}(m) = E\{(x_{n+m} - \mu_x)(y_n - \mu_y)^*)\}$$

where μ_x and μ_y are the mean values of the two stationary random processes, $*$ denotes the complex conjugate, and $E\{\cdot\}$ is the expected value operator. `xcov` estimates the sequence because, in practice, access is available to only a finite segment of the infinite-length random process.

`c = xcov(x,y)` returns the cross-covariance sequence in a length $2N$-1 vector, where `x` and `y` are length N vectors. For information on how arrays are processed with xcov, see "Multiple Channels".

`c = xcov(x)` is the autocovariance sequence for the vector `x`. Where `x` is an N-by-P array, v = xcov(x) returns an array with $2N$-1 rows whose P^2 columns contain the cross-covariance sequences for all combinations of the columns of `x`.

By default, `xcov` computes raw covariances with no normalization. For a length N vector

$$c_{xy}(m) = \begin{cases} \sum_{n=0}^{N-|m|-1} \left(x(n+m) - \frac{1}{N}\sum_{i=0}^{N-1} x_i \right)\left(y_n^* - \frac{1}{N}\sum_{i=0}^{N-1} y_i^* \right) & m \geq 0 \\ c_{yx}^*(-m) & m < 0 \end{cases}$$

The output vector `c` has elements given by `c(m)` = $c_{xy}(m\text{-}N)$, $m = 1, \ldots, 2N\text{-}1$.

The covariance function requires normalization to estimate the function properly.

`c = xcov(x,'option')` specifies a scaling option, where `'option'` is

- `'biased'`, for biased estimates of the cross-covariance function
- `'unbiased'`, for unbiased estimates of the cross-covariance function
- `'coeff'`, to normalize the sequence so the auto-covariances at zero lag are identically 1.0
- `'none'`, to use the raw, unscaled cross-covariances (default)

See [1] for more information on the properties of biased and unbiased correlation and covariance estimates.

`[c,lags] = xcov(x,y,maxlags)` where `x` and `y` are length `m` vectors, returns the cross-covariance sequence in a length `2*maxlags+1` vector `c`. `lags` is a vector of the lag indices where `c` was estimated, that is, `[-maxlags:maxlags]`.

`[c,lags] = xcov(x,maxlags)` is the autocovariance sequence over the range of lags `[-maxlags:maxlags]`.

`[c,lags] = xcov(x,maxlags)` where `x` is an `m`-by-`p` array, returns array `c` with `2*maxlags+1` rows whose P^2 columns contain the cross-covariance sequences for all combinations of the columns of `x`.

`[c,lags] = xcov(x,y,maxlags,'option')` specifies a scaling option, where `'option'` is the last input argument.

`[c,lags] = xcov(gpuArrayX,gpuArrayY,maxlags,'option')` returns the autocovariance or cross-covariance sequence for input objects

of class gpuArray. See "Use gpuArray Data" for details on gpuArray objects. Using xcov with gpuArray objects requires Parallel Computing Toolbox software and a CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details. The returned autocovariance or cross-covariance sequence, c, is a gpuArray object.

"Autocovariance using the GPU" on page 1-1296 shows you how to compute the autocovariance sequence on the GPU.

In all cases, xcov gives an output such that the zeroth lag of the covariance vector is in the middle of the sequence, at element or row maxlag+1 or at m.

Examples

The second output lags is useful when plotting. For example, the estimated autocovariance of white Gaussian noise $c_{ww}(m)$ can be displayed for $-10 \le m \le 10$ using:

```
ww = randn(1000,1);     % White Gaussian noise
[cov_ww,lags] = xcov(ww,10,'coeff');
stem(lags,cov_ww)
```

Autocovariance using the GPU

The following example requires Parallel Computing Toolbox software and a CUDA-enabled NVIDIA GPU with compute capability 1.3 or above. See http://www.mathworks.com/products/parallel-computing/requirements.html for details.

Create a signal consisting of a 10-Hz sine wave in additive noise. Use gpuArray to create a gpuArray object stored on your computer's GPU.

```
t = 0:0.001:10-0.001;
x = cos(2*pi*10*t)+randn(size(t));
X = gpuArray(x);
```

Compute the autocovariance sequence to lag 200.

```
[xc,lags] = xcov(X,200);
```

The output, `xc`, is a gpuArray object. Use `gather` to transfer the data from the GPU to the MATLAB workspace as a double-precision vector.

```
xc = gather(xc);
```

Algorithms

`xcov` computes the mean of its inputs, subtracts the mean, and then calls `xcorr`. For more information on estimating covariance and correlation functions, see [1].

Diagnostics

`xcov` does not check for any errors other than the correct number of input arguments. Instead, it relies on the error checking in `xcorr`.

References

[1] Orfanidis, S.J., *Optimum Signal Processing. An Introduction. 2nd Edition*, Prentice-Hall, Englewood Cliffs, NJ, 1996.

See Also

`conv | corrcoef | cov | xcorr | xcorr2`

yulewalk

Purpose

Recursive digital filter design

Syntax

```
[b,a] = yulewalk(n,f,m)
```

Description

`yulewalk` designs recursive IIR digital filters using a least-squares fit to a specified frequency response.

`[b,a] = yulewalk(n,f,m)` returns row vectors `b` and `a` containing the `n+1` coefficients of the order `n` IIR filter whose frequency-magnitude characteristics approximately match those given in vectors `f` and `m`:

- `f` is a vector of frequency points, specified in the range between 0 and 1, where 1 corresponds to half the sample frequency (the Nyquist frequency). The first point of `f` must be 0 and the last point 1, with all intermediate points in increasing order. Duplicate frequency points are allowed, corresponding to steps in the frequency response.
- `m` is a vector containing the desired magnitude response at the points specified in `f`.
- `f` and `m` must be the same length.
- `plot(f,m)` displays the filter shape.

The output filter coefficients are ordered in descending powers of z.

$$\frac{B(z)}{A(z)} = \frac{b(1) + b(2)z^{-1} + \cdots + b(n+1)z^{-n}}{a(1) + a(2)z^{-1} + \cdots + a(n+1)z^{-n}}$$

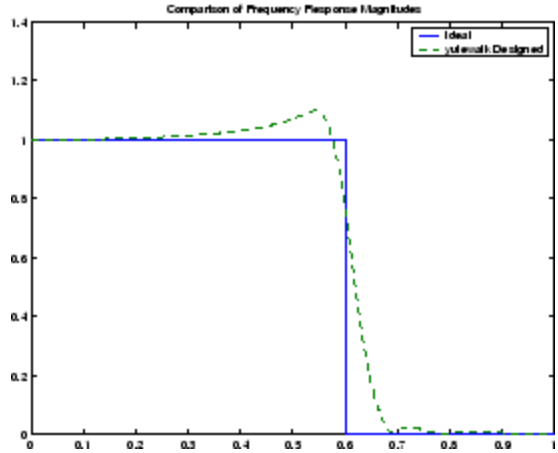
When specifying the frequency response, avoid excessively sharp transitions from passband to stopband. You may need to experiment with the slope of the transition region to get the best filter design.

Examples

Design an 8th-order lowpass filter and overplot the desired frequency response with the actual frequency response:

```
f = [0 0.6 0.6 1];
m = [1 1 0 0];
[b,a] = yulewalk(8,f,m);
```

```
[h,w] = freqz(b,a,128);
plot(f,m,w/pi,abs(h),'--')
legend('Ideal','yulewalk Designed')
title('Comparison of Frequency Response Magnitudes')
```



Algorithms

`yulewalk` performs a least-squares fit in the time domain. It computes the denominator coefficients using modified Yule-Walker equations, with correlation coefficients computed by inverse Fourier transformation of the specified frequency response. To compute the numerator, `yulewalk` takes the following steps:

1. Computes a numerator polynomial corresponding to an additive decomposition of the power frequency response.
2. Evaluates the complete frequency response corresponding to the numerator and denominator polynomials.
3. Uses a spectral factorization technique to obtain the impulse response of the filter.
4. Obtains the numerator polynomial by a least-squares fit to this impulse response.

References

[1] Friedlander, B., and B. Porat, "The Modified Yule-Walker Method of ARMA Spectral Estimation," *IEEE Transactions on Aerospace Electronic Systems, AES-20, No. 2* (March 1984), pp. 158-173.

See Also

`butter` | `cheby1` | `cheby2` | `ellip` | `fir2` | `firls` | `maxflat` | `firpm`

Purpose

Zero-phase response of digital filter

Syntax

```
[Hr,w] = zerophase(b,a)
[Hr,w] = zerophase(sos)
[Hr,w] = zerophase(Hd)
[Hr,w] = zerophase(...,nfft)
[Hr,w] = zerophase(...,nfft,'whole')
[Hr,w] = zerophase(...,w)
[Hr,f] = zerophase(...,f,fs)
[Hr,w,phi] = zerophase(...)
zerophase(...)
```

Description

`[Hr,w] = zerophase(b,a)` returns the zero-phase response `Hr`, and the frequency vector `w` (in radians/sample) at which `Hr` is computed, given a filter defined by numerator `b` and denominator `a`. For FIR filters where `a=1`, you can omit the value `a` from the command. The zero-phase response is evaluated at `512` equally spaced points on the upper half of the unit circle.

The zero-phase response, `Hr(w)`, is related to the frequency response, `H(w)` by

$$H(e^{j\omega}) = Hr(\omega)e^{j\varphi(\omega)}$$

where $H(e^{j\omega})$ is the frequency response, $Hr(\omega)$ is the zero-phase response and [[PHI1]](w) is the continuous phase.

Note The zero-phase response is always real, but it is not the equivalent of the magnitude response. The former can be negative while the latter cannot be negative.

`[Hr,w] = zerophase(sos)` returns the zero-phase response for the second order sections matrix, `sos`. `sos` is a K-by-6 matrix, where the number of sections, K, must be greater than or equal to 2. If the number of sections is less than 2, `zerophase` considers the input to be the

numerator vector, `b`. Each row of `sos` corresponds to the coefficients of a second order (biquad) filter. The i-th row of the `sos` matrix corresponds to `[bi(1) bi(2) bi(3) ai(1) ai(2) ai(3)]`.

`[Hr,w] = zerophase(Hd)` returns the zero-phase response for the `dfilt` filter object, `Hd`, or the array of `dfilt` filter objects. If `Hd` is an array of `dfilt` objects, each column of `Hr` is the zero-phase response of the corresponding `dfilt` object.

`[Hr,w] = zerophase(...,nfft)` returns the zero-phase response `Hr` and frequency vector `w` (radians/sample), using `nfft` frequency points on the upper half of the unit circle.

`[Hr,w] = zerophase(...,nfft,'whole')` returns the zero-phase response `Hr` and frequency vector `w` (radians/sample), using `nfft` frequency points around the whole unit circle.

`[Hr,w] = zerophase(...,w)` returns the zero-phase response `Hr` and frequency vector `w` (radians/sample) at frequencies in vector `w`. The vector `w` must have at least two elements.

`[Hr,f] = zerophase(...,f,fs)` returns the zero-phase response `Hr` and frequency vector `f` (Hz), using the sampling frequency `fs` (in Hz), to determine the frequency vector `f` (in Hz) at which `Hr` is computed. The vector `f` must have at least two elements.

`[Hr,w,phi] = zerophase(...)` returns the zero-phase response `Hr`, frequency vector `w` (rad/sample), and the continuous phase component, `phi`. (Note that this quantity is not equivalent to the phase response of the filter when the zero-phase response is negative.)

`zerophase(...)` plots the zero-phase response versus frequency. The plot is displayed in the current figure window. If the input is the numerator and denominator coefficients, a second order sections matrix, or a single `dfilt` object, the zero–phase response of the single filter is displayed. If the input is an array of `dfilt` objects, the zero–phase responses of all filters in the array are displayed.

Note If the input to `zerophase` is single precision, the zero-phase response is calculated using single-precision arithmetic. The output, `Hr`, is single precision.

Examples

Example 1

Plot the zero-phase response of a constrained least squares FIR filter:

```
b=fircls1(54,.3,.02,.008);
zerophase(b);
```

Example 2

Plot the zero-phase response of an elliptic filter:

```
[b,a]=ellip(10,.5,20,.4);
zerophase(b,a,512,'whole');
```

See Also

`freqs` | `freqz` | `fvtool` | `grpdelay` | `invfreqz` | `phasedelay` | `phasez`

Purpose

Convert zero-pole-gain filter parameters to second-order sections form

Syntax

```
[sos,g] = zp2sos(z,p,k)
[sos,g] = zp2sos(z,p,k,'order')
[sos,g] = zp2sos(z,p,k,'order','scale')
[sos,g] = zp2sos(z,p,k,'order','scale',zeroflag)
sos = zp2sos(...)
```

Description

`zp2sos` converts a discrete-time zero-pole-gain representation of a given digital filter to an equivalent second-order section representation.

`[sos,g] = zp2sos(z,p,k)` creates a matrix `sos` in second-order section form with gain `g` equivalent to the discrete-time zero-pole-gain filter represented by input arguments `z`, `p`, and `k`. Vectors `z` and `p` contain the zeros and poles of the filter's transfer function $H(z)$, not necessarily in any particular order.

$$H(z) = k\frac{(z-z_1)(z-z_2)\cdots(z-z_n)}{(z-p_1)(z-p_2)\cdots(z-p_m)}$$

where n and m are the lengths of `z` and `p`, respectively, and `k` is a scalar gain. The zeros and poles must be real or complex conjugate pairs. `sos` is an L-by-6 matrix

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

whose rows contain the numerator and denominator coefficients b_{ik} and a_{ik} of the second-order sections of $H(z)$.

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

The number L of rows of the matrix `sos` is the closest integer greater than or equal to the maximum of $n/2$ and $m/2$.

`[sos,g] = zp2sos(z,p,k,'order')` specifies the order of the rows in `sos`, where `'order'` is

- `'down'`, to order the sections so the first row of `sos` contains the poles closest to the unit circle
- `'up'`, to order the sections so the first row of `sos` contains the poles farthest from the unit circle (default)

`[sos,g] = zp2sos(z,p,k,'order','scale')` specifies the desired scaling of the gain and the numerator coefficients of all second-order sections, where `'scale'` is

- `'none'`, to apply no scaling (default)
- `'inf'`, to apply infinity-norm scaling
- `'two'`, to apply 2-norm scaling

Using infinity-norm scaling in conjunction with `up`-ordering minimizes the probability of overflow in the realization. Using 2-norm scaling in conjunction with `down`-ordering minimizes the peak round-off noise.

Note Infinity-norm and 2-norm scaling are appropriate only for direct-form II implementations.

`[sos,g] = zp2sos(z,p,k,'order','scale',zeroflag)` specifies whether to keep together real zeros that are the negatives of each other instead of ordering them according to proximity to poles. Setting `zeroflag` to `true` keeps the zeros together and results in a numerator with a middle coefficient equal to zero. The default for `zeroflag` is `false`.

`sos = zp2sos(...)` embeds the overall system gain, `g`, in the first section, $H_1(z)$, so that

$$H(z) = \prod_{k=1}^{L} H_k(z)$$

Note Embedding the gain in the first section when scaling a direct-form II structure is not recommended and may result in erratic scaling. To avoid embedding the gain, use `ss2sos` with two outputs.

Examples

Find a second-order section form of a Butterworth lowpass filter:

```
[z,p,k] = butter(5,0.2);
sos = zp2sos(z,p,k);
```

Algorithms

`zp2sos` uses a four-step algorithm to determine the second-order section representation for an input zero-pole-gain system:

1. It groups the zeros and poles into complex conjugate pairs using the `cplxpair` function.
2. It forms the second-order section by matching the pole and zero pairs according to the following rules:
   a. Match the poles closest to the unit circle with the zeros closest to those poles.
   b. Match the poles next closest to the unit circle with the zeros closest to those poles.
   c. Continue until all of the poles and zeros are matched.

   `zp2sos` groups real poles into sections with the real poles closest to them in absolute value. The same rule holds for real zeros.
3. It orders the sections according to the proximity of the pole pairs to the unit circle. `zp2sos` normally orders the sections with poles closest to the unit circle last in the cascade. You can tell `zp2sos` to order the sections in the reverse order by specifying the `down` flag.

4 `zp2sos` scales the sections by the norm specified in the `'scale'` argument. For arbitrary $H(\omega)$, the scaling is defined by

$$\|H\|_p = \left[\frac{1}{2\pi} \int_0^{2\pi} |H(\omega)|^p \, d\omega \right]^{\frac{1}{p}}$$

where p can be either ∞ or 2. See the references for details on the scaling. This scaling is an attempt to minimize overflow or peak round-off noise in fixed point filter implementations.

References

[1] Jackson, L.B., *Digital Filters and Signal Processing*, 3rd ed., Kluwer Academic Publishers, Boston, 1996, Chapter 11.

[2] Mitra, S.K., *Digital Signal Processing: A Computer-Based Approach*, McGraw-Hill, New York, 1998, Chapter 9.

[3] Vaidyanathan, P.P., "Robust Digital Filter Structures," *Handbook for Digital Signal Processing*, S.K. Mitra and J.F. Kaiser, ed., John Wiley & Sons, New York, 1993, Chapter 7.

See Also

`cplxpair` | `filternorm` | `sos2zp` | `ss2sos` | `tf2sos` | `zp2ss` | `zp2tf`

zp2ss

Purpose

Convert zero-pole-gain filter parameters to state-space form

Syntax

```
[A,B,C,D] = zp2ss(z,p,k)
```

Description

`zp2ss` converts a zero-pole-gain representation of a given system to an equivalent state-space representation.

`[A,B,C,D] = zp2ss(z,p,k)` finds a single input, multiple output, state-space representation

$$\dot{x} = Ax + Bu$$
$$y = Cx + Du$$

given a system in factored transfer function form.

$$H(s) = \frac{Z(s)}{P(s)} = k\frac{(s - z_1)(s - z_2)\cdots(s - z_n)}{(s - p_1)(s - p_2)\cdots(s - p_n)}$$

Column vector `p` specifies the pole locations, and matrix `z` the zero locations with as many columns as there are outputs. The gains for each numerator transfer function are in vector `k`. The `A`, `B`, `C`, and `D` matrices are returned in controller canonical form.

`Inf` values may be used as place holders in `z` if some columns have fewer zeros than others.

Algorithms

`zp2ss`, for single-input systems, groups complex pairs together into two-by-two blocks down the diagonal of the `A` matrix. This requires the zeros and poles to be real or complex conjugate pairs.

See Also

`sos2ss` | `ss2zp` | `tf2ss` | `zp2sos` | `zp2tf`

Purpose

Convert zero-pole-gain filter parameters to transfer function form

Syntax

```
[b,a] = zp2tf(z,p,k)
```

Description

`zp2tf` forms transfer function polynomials from the zeros, poles, and gains of a system in factored form.

`[b,a] = zp2tf(z,p,k)` finds a rational transfer function

$$\frac{B(s)}{A(s)} = \frac{b_1 s^{(n-1)} + \cdots + b_{(n-1)}s + b_n}{a_1 s^{(m-1)} + \cdots + a_{(m-1)}s + a_m}$$

given a system in factored transfer function form

$$H(s) = \frac{Z(s)}{P(s)} = k\frac{(s - z_1)(s - z_2)\cdots(s - z_m)}{(s - p_1)(s - p_2)\cdots(s - p_n)}$$

Column vector `p` specifies the pole locations, and matrix `z` specifies the zero locations, with as many columns as there are outputs. The gains for each numerator transfer function are in vector `k`. The zeros and poles must be real or come in complex conjugate pairs. The polynomial denominator coefficients are returned in row vector `a` and the polynomial numerator coefficients are returned in matrix `b`, which has as many rows as there are columns of `z`.

`Inf` values can be used as place holders in `z` if some columns have fewer zeros than others.

Algorithms

The system is converted to transfer function form using `poly` with `p` and the columns of `z`.

See Also

`sos2tf` | `ss2tf` | `tf2zp` | `tf2zpk` | `zp2sos` | `zp2ss`

zplane

Purpose

Zero-pole plot

Syntax

```
zplane(z,p)
zplane(b,a)
zplane(Hd)
[hz,hp,ht] = zplane(z,p)
```

Description

This function displays the poles and zeros of discrete-time systems.

`zplane(z,p)` plots the zeros specified in column vector `z` and the poles specified in column vector `p` in the current figure window. The symbol `'o'` represents a zero and the symbol `'x'` represents a pole. The plot includes the unit circle for reference. If `z` and `p` are arrays, `zplane` plots the poles and zeros in the columns of `z` and `p` in different colors.

You can override the automatic scaling of `zplane` using

```
axis([xmin xmax ymin ymax])
```

or

```
set(gca,'ylim',[ymin ymax])
```

or

```
set(gca,'xlim',[xmin xmax])
```

after calling `zplane`. This is useful in the case where one or a few of the zeros or poles have such a large magnitude that the others are grouped tightly around the origin and are hard to distinguish.

`zplane(b,a)` where `b` and `a` are row vectors, first uses `roots` to find the zeros and poles of the transfer function represented by numerator coefficients `b` and denominator coefficients `a`. The transfer function is defined in terms of z^{-1}:

$$H(z) = \frac{B(z)}{A(z)} = \frac{b(1) + b(2)z^{-1} + \cdots + b(n+1)z^{-n}}{a(1) + a(2)z^{-1} + \cdots + a(m+1)z^{-m}}$$

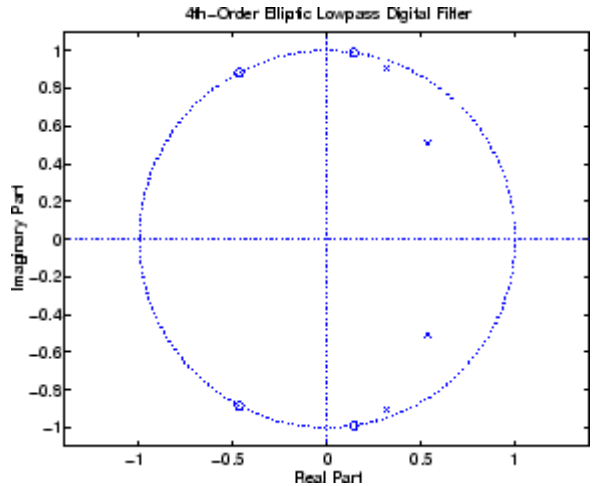
`zplane(Hd)` finds the zeros and poles of the transfer function represented by the `dfilt` filter object `Hd`. The pole-zero plot is displayed in `fvtool`.

`[hz,hp,ht] = zplane(z,p)` returns vectors of handles to the zero lines, `hz`, and the pole lines, `hp`. `ht` is a vector of handles to the axes/unit circle line and to text objects, which are present when there are multiple zeros or poles. If there are no zeros or no poles, `hz` or `hp` is the empty matrix `[]`.

Examples

For data sampled at 1000 Hz, plot the poles and zeros of a 4th-order elliptic lowpass digital filter with cutoff frequency of 200 Hz, 3 dB of ripple in the passband, and 30 dB of attenuation in the stopband:

```
[z,p,k] = ellip(4,3,30,200/500);
zplane(z,p);
title('4th-Order Elliptic Lowpass Digital Filter');
```



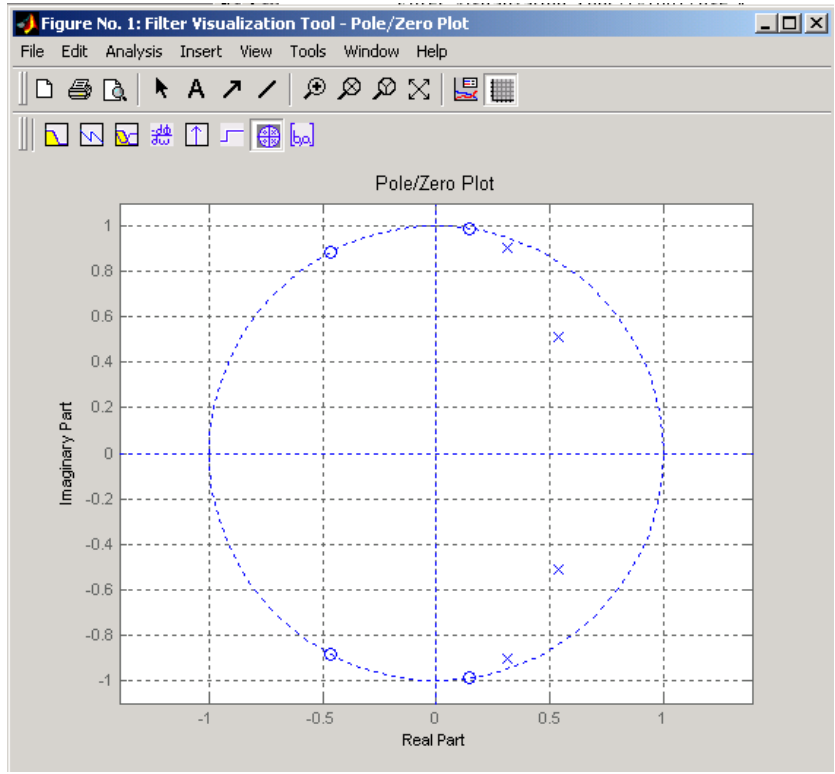
To generate the same plot with a transfer function representation of the filter, use:

```
[b,a] = ellip(4,3,30,200/500);    % Transfer function
```

```
zplane(b,a)
```

To generate the same plot using a `dfilt` object and displaying the result in the Filter Visualization Tool (`fvtool`) use:

```
[b,a] = ellip(4,3,30,200/500);
Hd=dfilt.df1(b,a);
zplane(Hd)
```



See Also

`freqz`

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