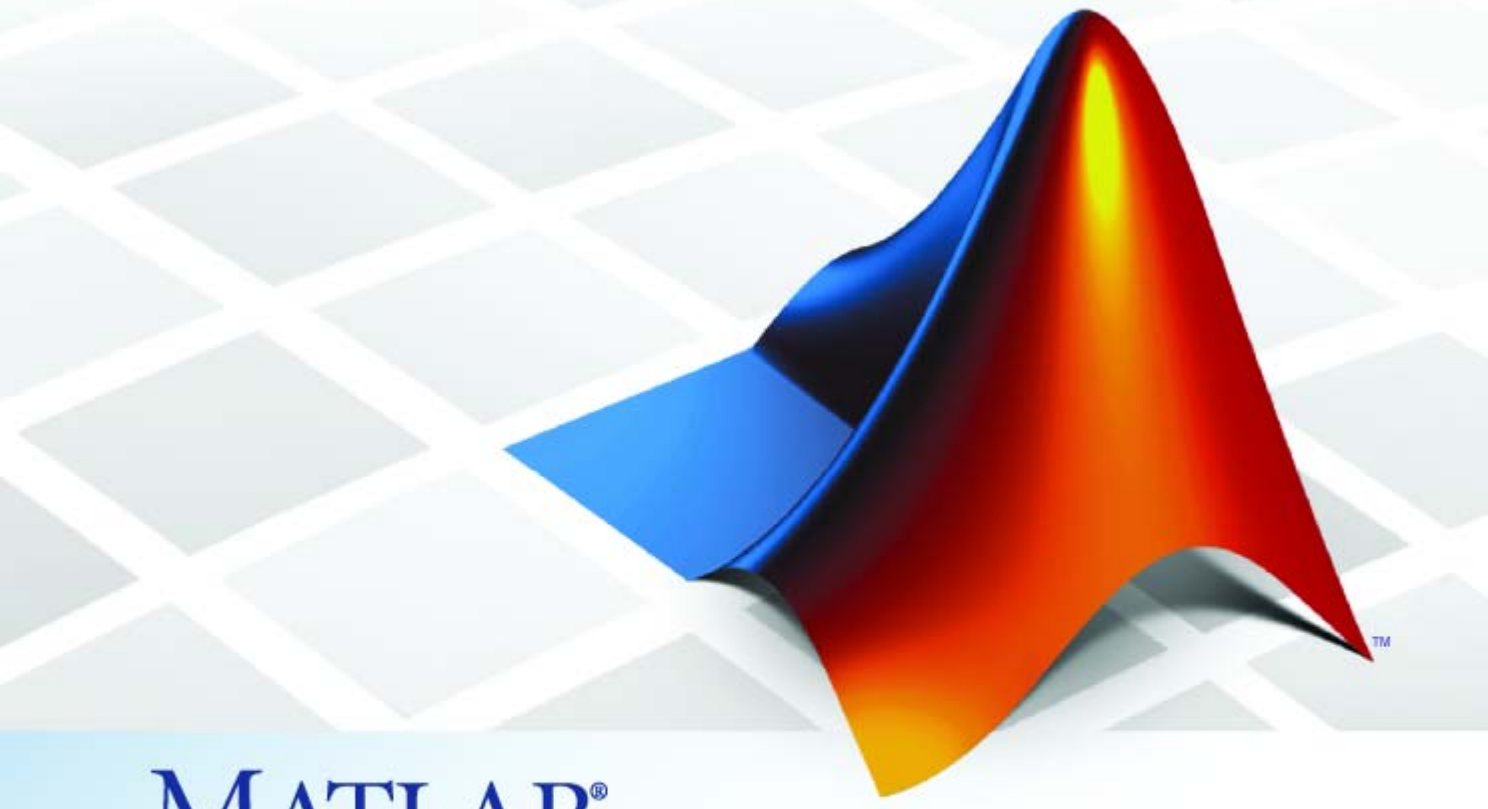
Signal Processing Toolbox™ 6

Getting Started Guide



MATLAB®

The MathWorks™

Accelerating the pace of engineering and science

How to Contact The MathWorks

`www.mathworks.com`	Web
`comp.soft-sys.matlab`	Newsgroup
`www.mathworks.com/contact_TS.html`	Technical Support

`suggest@mathworks.com`	Product enhancement suggestions
`bugs@mathworks.com`	Bug reports
`doc@mathworks.com`	Documentation error reports
`service@mathworks.com`	Order status, license renewals, passcodes
`info@mathworks.com`	Sales, pricing, and general information

508-647-7000 (Phone)

508-647-7001 (Fax)

The MathWorks, Inc.
3 Apple Hill Drive
Natick, MA 01760-2098

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

Signal Processing Toolbox™ Getting Started Guide

© COPYRIGHT 2006–2008 by The MathWorks, Inc.

The software described in this document is furnished under a license agreement. The software may be used or copied only under the terms of the license agreement. No part of this manual may be photocopied or reproduced in any form without prior written consent from The MathWorks, Inc.

FEDERAL ACQUISITION: This provision applies to all acquisitions of the Program and Documentation by, for, or through the federal government of the United States. By accepting delivery of the Program or Documentation, the government hereby agrees that this software or documentation qualifies as commercial computer software or commercial computer software documentation as such terms are used or defined in FAR 12.212, DFARS Part 227.72, and DFARS 252.227-7014. Accordingly, the terms and conditions of this Agreement and only those rights specified in this Agreement, shall pertain to and govern the use, modification, reproduction, release, performance, display, and disclosure of the Program and Documentation by the federal government (or other entity acquiring for or through the federal government) and shall supersede any conflicting contractual terms or conditions. If this License fails to meet the government's needs or is inconsistent in any respect with federal procurement law, the government agrees to return the Program and Documentation, unused, to The MathWorks, Inc.

Trademarks

MATLAB and Simulink are registered trademarks of The MathWorks, Inc. See `www.mathworks.com/trademarks` for a list of additional trademarks. Other product or brand names may be trademarks or registered trademarks of their respective holders.

Patents

The MathWorks products are protected by one or more U.S. patents. Please see `www.mathworks.com/patents` for more information.

Revision History

September 2006	First printing	New for Version 6.6 (Release 2006b)
March 2007	Online only	Revised for Version 6.7 (Release 2007a)
September 2007	Online only	Revised for Version 6.8 (Release 2007b)
March 2008	Online only	Revised for Version 6.9 (Release 2008a)
October 2008	Online only	Revised for Version 6.10 (Release 2008b)

Contents

1 Overview

Product Overview 1-2
Product Summary 1-2
Command-Line Functions and Objects 1-2
Graphical User Interfaces 1-3
Supported Data Types 1-3

Central Features 1-4
Signal Processing Toolbox Functions 1-4
Signals and Systems 1-4
Filter Design, Analysis, and Implementation 1-5
Linear System Transformations 1-5
Windowing Functions 1-5
Spectral Analysis 1-5
Transforms 1-5
Statistical Signal Processing 1-5
Parametric Modeling 1-6
Linear Prediction 1-6
Multirate Signal Processing 1-6
Waveform Generation 1-6
Other Operations 1-6

Interactive Tools 1-7

Extensibility 1-8
Expanding the Toolbox with M-Files 1-8
Expanding the Toolbox with Other Toolboxes 1-8

Finding More Information 1-9

2 Basic Signal Processing Concepts

Representing Signals 2-2
Numeric Arrays 2-2
Vector Representation 2-2

Waveform Generation: Time Vectors and Sinusoids . . . 2-4
Time Vectors 2-4
Common Sequences: Unit Impulse, Unit Step, and Unit Ramp 2-5
Multichannel Signals 2-5
Common Periodic Waveforms 2-6
Common Aperiodic Waveforms 2-7
The pulstran Function 2-8
The Sinc Function 2-9
The Dirichlet Function 2-10

Working with Data 2-12
Importing Data from Within the MATLAB Environment . . 2-12
Importing Data from Outside the MATLAB Environment 2-12
Converting Data into a MAT-File 2-12
Exporting Data 2-13
Data Precision 2-13

Selected Bibliography 2-14

3 Filter Design with the FDATool GUI

Introduction 3-2

Designing the Filter 3-3

Analyzing the Filter 3-8

Designing Additional Filters 3-10

Viewing and Annotating the Filter 3-11
Viewing the Filter in FVTool 3-11
Using FVTool for Annotation 3-15

Exporting Filters from FDATool 3-17

Using Discrete Filter Objects (dfilts) 3-19
Using dfilt Objects 3-19
Filtering with dfilt 3-19

Where to Find More Information 3-22

4 Spectral Analysis

Introduction 4-2
Spectral Analysis Examples 4-2
Spectral Estimators 4-2
Spectral Analysis Algorithms 4-2
Spectral Analysis Objects 4-3

Creating a Spectral Analysis Object 4-4

Generating a Spectrum 4-6

Changing Spectral Analysis Object Property Values . . 4-9
Using the set command to Set Property Values 4-9
Using Options Objects to Set Property Values 4-10

Measuring Signal Power 4-11

Where to Find More Information 4-15

Index

1

Overview

- "Product Overview" on page 1-2
- "Central Features" on page 1-4
- "Interactive Tools" on page 1-7
- "Extensibility" on page 1-8
- "Finding More Information" on page 1-9

Product Overview

In this section...
"Product Summary" on page 1-2
"Command-Line Functions and Objects" on page 1-2
"Graphical User Interfaces" on page 1-3
"Supported Data Types" on page 1-3

Product Summary

Signal Processing Toolbox™ software is a collection of tools based on the MATLAB® environment. The toolbox supports a wide range of signal processing operations, from waveform generation to filter design and implementation, parametric modeling, and spectral analysis. The toolbox provides two categories of tools, command-line functions/objects and graphical user interfaces:

Command-Line Functions and Objects

Command-line functions and objects are available in the following categories:

- Discrete-time filter design, analysis, and implementation
- Analog filter design, analysis, and implementation
- Linear system transformations
- Windowing functions
- Spectral analysis and cepstral analysis
- Transforms
- Statistical signal processing
- Parametric modeling
- Linear prediction
- Multirate signal processing
- Waveform generation

Note For information on objects, see *Object-Oriented Programming*.

Graphical User Interfaces

A suite of interactive graphical user interfaces are available for

- Filter design and analysis
- Window design and analysis
- Signal plotting and analysis, spectral analysis, and filtering

Supported Data Types

The Signal Processing Toolbox software supports only double-precision inputs. If you input single-precision floating-point or integer data, you should not expect to receive correct results, and in many cases, an error will occur. The Filter Design Toolbox™ product, in conjunction with the Fixed-Point Toolbox™ product, enables single-precision floating-point and fixed-point support for filtering and filter design.

Central Features

In this section...
"Signal Processing Toolbox Functions" on page 1-4
"Signals and Systems" on page 1-4
"Filter Design, Analysis, and Implementation" on page 1-5
"Linear System Transformations" on page 1-5
"Windowing Functions" on page 1-5
"Spectral Analysis" on page 1-5
"Transforms" on page 1-5
"Statistical Signal Processing" on page 1-5
"Parametric Modeling" on page 1-6
"Linear Prediction" on page 1-6
"Multirate Signal Processing" on page 1-6
"Waveform Generation" on page 1-6
"Other Operations" on page 1-6

Signal Processing Toolbox Functions

Signal Processing Toolbox functions are algorithms, expressed mostly in M-files, that implement a variety of signal processing tasks. These toolbox functions are a specialized extension of the MATLAB computational and graphical environment.

Signals and Systems

The basic entities that toolbox functions work with are signals and systems. The functions emphasize digital (or discrete) signals and filters, as opposed to analog (or continuous) signals. The principal filter type the toolbox supports is the linear, time-invariant digital filter with a single input and a single output. You can represent linear time-invariant systems using one of several models (such as transfer function, state-space, zero-pole-gain, and second-order section), and you can convert between representations.

Filter Design, Analysis, and Implementation

Signal Processing Toolbox software provides customizable support for filter design. The major filter design functions included in the toolbox are FIR and IIR filter design, analysis, and implementation, filter order estimation, and analog filter prototyping and transformations.

Linear System Transformations

The toolbox has a number of transformation functions, including conversions to and from second-order sections, state-space, pole-zero, lattice or ladder, and transfer functions.

Windowing Functions

The toolbox provides many commonly used window functions as well as graphical user interfaces to view and compare windows and design filters using these windows.

Spectral Analysis

Toolbox functions are available for estimating the power spectral density, mean-square spectral estimate, and pseudo spectrum, using parametric and nonparametric techniques. Some of the spectral analysis methods included in the toolbox are Burg, covariance, eigenvector, Thomson multitaper, periodogram, Welch, and Yule-Walker. Other functions are available for computing the average power of a power spectral density, computing a one-sided spectrum, and shifting the DC component to the center of a spectrum.

Transforms

The toolbox includes a variety of transforms and inverse transforms, including Fourier, chirp-Z, discrete cosine, Goertzel, and Hilbert.

Statistical Signal Processing

The toolbox has functions for computing correlation, cross-correlation, covariance, and autocorrelation.

Parametric Modeling

The toolbox includes these methods for autoregressive parametric modeling: Burg, covariance, Yule-Walker, and Steiglitz-McBride (for ARMA modeling). The toolbox also has functions for fitting a frequency response to an analog or discrete-time filter.

Linear Prediction

The toolbox has functions for computing linear prediction coefficients and for converting between autorcorrelations and prediction polynomials, reflection coefficients, and line spectral frequencies.

Multirate Signal Processing

The toolbox includes a number of functions for multirate signal processing, including decimation, up- and downsampling, resampling, and spline interpolation.

Waveform Generation

The toolbox has functions to generate many types of periodic and aperiodic waveforms, including chirp, Dirichlet function, Gaussian RF pulse, Gaussian monopulse, pulse train, rectangle, sawtooth, sinc, square wave, triangle, and voltage-controlled oscillator. See “Waveform Generation: Time Vectors and Sinusoids” on page 2-4 for more information.

Other Operations

A number of other operations are also available in the toolbox. Some of these are cepstral analysis, modulation, demodulation, Slepian sequences, and various plotting methods.

Interactive Tools

The power of Signal Processing Toolbox software is greatly enhanced by its easy-to-use interactive tools.

- The Filter Design and Analysis Tool (`fdatool`) provides a more comprehensive collection of features for addressing filter design. FDATool offers seamless access to the additional filter design methods, quantization features, C-code generation and other enhanced filtering features of the Filter Design Toolbox product when that product is installed. If you have the Filter Design HDL Coder™ product installed, you can also generate HDL code from FDATool.
- The Filter Visualization Tool (`fvtool`) provides a graphical environment for viewing, annotating, and printing filter response plots.
- The Window Design and Analysis Tool (`wintool`) provides an environment for designing and comparing spectral windows.
- The Window Visualization Tool (`wvtool`) provides a graphical environment for viewing, annotating, and printing window plots.
- The Signal Processing Tool (`sptool`) provides a rich graphical environment for signal viewing, filter design, and spectral analysis.

Extensibility

In this section...
"Expanding the Toolbox with M-Files" on page 1-8
"Expanding the Toolbox with Other Toolboxes" on page 1-8

Expanding the Toolbox with M-Files

One of the most important features of the MATLAB environment is that it is extensible. MATLAB lets you create your own M-files to meet numeric computation needs for research, design, or engineering of signal processing systems. Simply copy the M-files provided with the Signal Processing Toolbox product and modify them as needed, or create new functions to expand the functionality of the toolbox.

Expanding the Toolbox with Other Toolboxes

A number of other MATLAB products expand and enhance the Signal Processing Toolbox product. These include:

- Filter Design Toolbox — Toolbox for advanced filter design
- Signal Processing Blockset™ — Product based on Simulink® for creating and analyzing signal processing models
- Fixed-Point Toolbox — Toolbox for using fixed-point arithmetic
- Filter Design HDL Coder — Toolbox for creating and exporting HDL code
- Embedded IDE Link™ CC — Toolbox for working with embedded software on Texas Instruments™ DSPs

Finding More Information

This Getting Started guide provides an introduction to Signal Processing Toolbox software and examples, which give you a quick start at using some of the commands and graphical user interfaces. It is assumed that you have basic knowledge and understanding of signals and systems, including such topics as filter and linear system theory and basic Fourier analysis.

More detailed and advanced information on using the toolbox is available in the online help system by typing `doc` at the MATLAB command line or by viewing the documentation on the MathWorks Web site (`www.mathworks.com`). The toolbox also includes a number of introductory and advanced demos which you can access by typing `demos` at the MATLAB command line.

2

Basic Signal Processing Concepts

- "Representing Signals" on page 2-2
- "Waveform Generation: Time Vectors and Sinusoids" on page 2-4
- "Working with Data" on page 2-12
- "Selected Bibliography" on page 2-14

Representing Signals

In this section...

"Numeric Arrays" on page 2-2

"Vector Representation" on page 2-2

Numeric Arrays

The central data construct in the MATLAB environment is the *numeric array*, an ordered collection of real or complex numeric data with two or more dimensions. The basic data objects of signal processing (one-dimensional signals or sequences, multichannel signals, and two-dimensional signals) are all naturally suited to array representation.

Vector Representation

MATLAB represents ordinary one-dimensional sampled data signals, or sequences, as *vectors*. Vectors are 1-by-n or n-by-1 arrays, where n is the number of samples in the sequence. One way to introduce a sequence is to enter it as a list of elements at the command prompt. The statement

```
x = [4 3 7 -9 1]
```

creates a simple five-element real sequence in a row vector. Transposition turns the sequence into a column vector

```
x = x'
```

resulting in

```
x =
     4
     3
     7
    -9
     1
```

Column orientation is preferable for single channel signals because it extends naturally to the multichannel case. For multichannel data, each column of a

matrix represents one channel. Each row of such a matrix then corresponds to a sample point. A three-channel signal that consists of `x`, `2x`, and `x/π` is

```
y = [x 2*x x/pi]
```

This results in

```
y =
    4.0000    8.0000    1.2732
    3.0000    6.0000    0.9549
    7.0000   14.0000    2.2282
   -9.0000  -18.0000   -2.8648
    1.0000    2.0000    0.3183
```

Waveform Generation: Time Vectors and Sinusoids

In this section...

"Time Vectors" on page 2-4
"Common Sequences: Unit Impulse, Unit Step, and Unit Ramp" on page 2-5
"Multichannel Signals" on page 2-5
"Common Periodic Waveforms" on page 2-6
"Common Aperiodic Waveforms" on page 2-7
"The pulstran Function" on page 2-8
"The Sinc Function" on page 2-9
"The Dirichlet Function" on page 2-10

Time Vectors

Most toolbox functions require you to begin with a vector representing a time base. Consider generating data with a 1000 Hz sample frequency, for example. An appropriate time vector is

```
t = (0:0.001:1)';
```

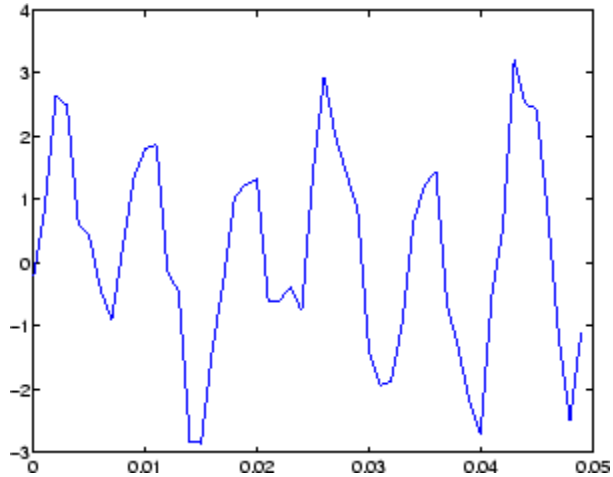
where the MATLAB colon operator creates a 1001-element row vector that represents time running from 0 to 1 s in steps of 1 ms. The transpose operator (`'`) changes the row vector into a column; the semicolon (`;`) tells MATLAB to compute, but not display the result.

Given `t`, you can create a sample signal `y` consisting of two sinusoids, one at 50 Hz and one at 120 Hz with twice the amplitude.

```
y = sin(2*pi*50*t) + 2*sin(2*pi*120*t);
```

The new variable `y`, formed from vector `t`, is also 1001 elements long. You can add normally distributed white noise to the signal and plot the first 50 points using

```
randn('state',0);
yn = y + 0.5*randn(size(t));
plot(t(1:50),yn(1:50))
```



Common Sequences: Unit Impulse, Unit Step, and Unit Ramp

Since MATLAB is a programming language, an endless variety of different signals is possible. Here are some statements that generate several commonly used sequences, including the unit impulse, unit step, and unit ramp functions:

```
t = (0:0.001:1)';
y = [1; zeros(99,1)];  % impulse
y = ones(100,1);       % step (filter assumes 0 initial cond.)
y = t;                 % ramp
y = t.^2;
y = square(4*t);
```

All of these sequences are column vectors. The last three inherit their shapes from `t`.

Multichannel Signals

Use standard MATLAB array syntax to work with multichannel signals. For example, a multichannel signal consisting of the last three signals generated above is

```
z = [t t.^2 square(4*t)];
```

You can generate a multichannel unit sample function using the outer product operator. For example, a six-element column vector whose first element is one, and whose remaining five elements are zeros, is

```
a = [1 zeros(1,5)]';
```

To duplicate column vector `a` into a matrix without performing any multiplication, use the MATLAB `colon` operator and the `ones` function:

```
c = a(:,ones(1,3));
```

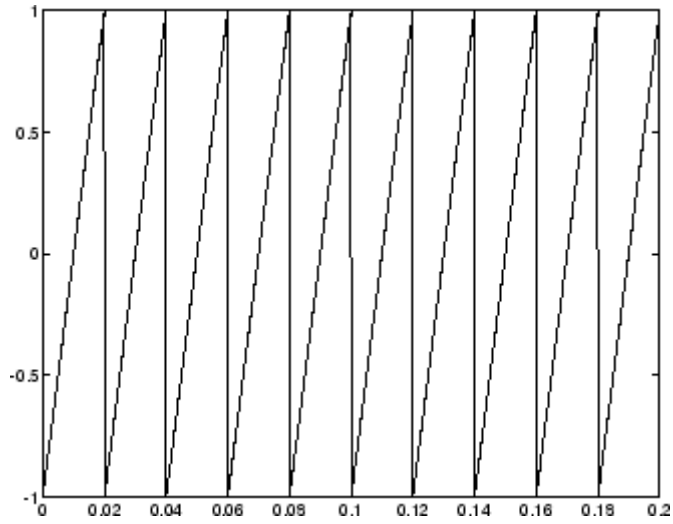
Common Periodic Waveforms

The toolbox provides functions for generating widely used periodic waveforms:

- `sawtooth` generates a sawtooth wave with peaks at ±1 and a period of 2π. An optional `width` parameter specifies a fractional multiple of 2π at which the signal's maximum occurs.
- `square` generates a square wave with a period of 2π. An optional parameter specifies *duty cycle*, the percent of the period for which the signal is positive.

To generate 1.5 s of a 50 Hz sawtooth wave with a sample rate of 10 kHz and plot 0.2 s of the generated waveform, use

```
fs = 10000;
t = 0:1/fs:1.5;
x = sawtooth(2*pi*50*t);
plot(t,x), axis([0 0.2 -1 1])
```



Common Aperiodic Waveforms

The toolbox also provides functions for generating several widely used aperiodic waveforms:

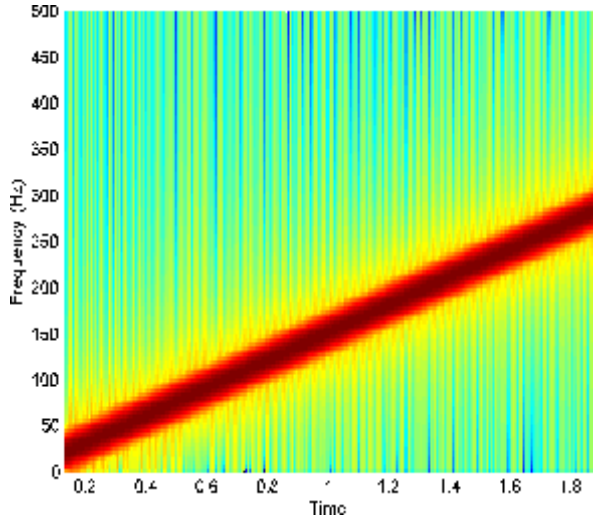
- `gauspuls` generates a Gaussian-modulated sinusoidal pulse with a specified time, center frequency, and fractional bandwidth. Optional parameters return in-phase and quadrature pulses, the RF signal envelope, and the cutoff time for the trailing pulse envelope.
- `chirp` generates a linear, log, or quadratic swept-frequency cosine signal. An optional parameter specifies alternative sweep methods. An optional parameter `phi` allows initial phase to be specified in degrees.

To compute 2 s of a linear chirp signal with a sample rate of 1 kHz, that starts at DC and crosses 150 Hz at 1 s, use

```
t = 0:1/1000:2;
y = chirp(t,0,1,150);
```

To plot the spectrogram, use

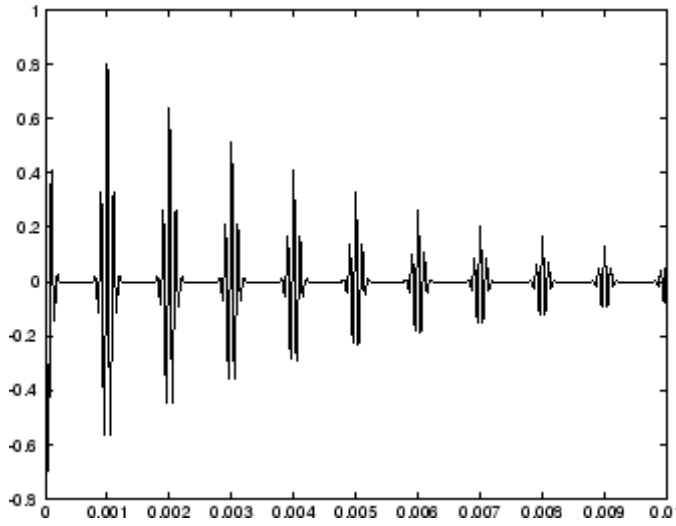
```
spectrogram(y,256,250,256,1000,'yaxis')
```



The pulstran Function

The `pulstran` function generates pulse trains from either continuous or sampled prototype pulses. The following example generates a pulse train consisting of the sum of multiple delayed interpolations of a Gaussian pulse. The pulse train is defined to have a sample rate of 50 kHz, a pulse train length of 10 ms, and a pulse repetition rate of 1 kHz; `D` specifies the delay to each pulse repetition in column 1 and an optional attenuation for each repetition in column 2. The pulse train is constructed by passing the name of the `gauspuls` function to `pulstran`, along with additional parameters that specify a 10 kHz Gaussian pulse with 50% bandwidth:

```
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)
```



The Sinc Function

The `sinc` function computes the mathematical sinc function for an input vector or matrix `x`. The sinc function is the continuous inverse Fourier transform of the rectangular pulse of width 2π and height 1.

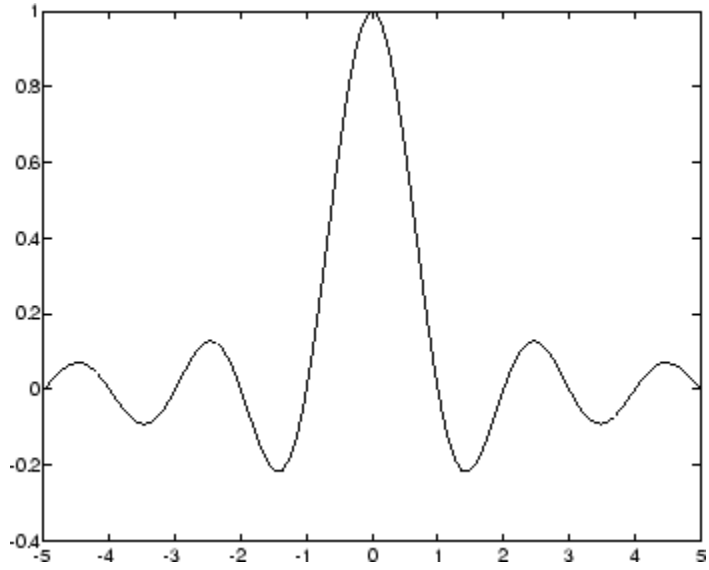
The sinc function has a value of 1 where `x` is zero, and a value of

$$\frac{\sin(\pi x)}{\pi x}$$

for all other elements of `x`.

To plot the sinc function for a linearly spaced vector with values ranging from -5 to 5, use the following commands:

```
x = linspace(-5,5);
y = sinc(x);
plot(x,y)
```



The Dirichlet Function

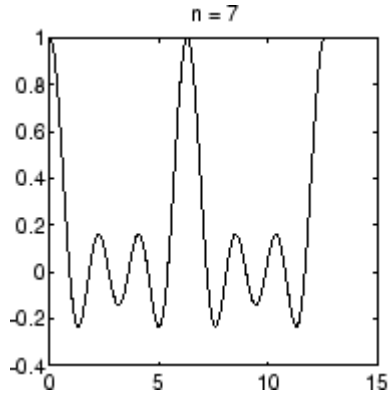
The toolbox function `diric` computes the Dirichlet function, sometimes called the *periodic sinc* or *aliased sinc* function, for an input vector or matrix `x`. The Dirichlet function is

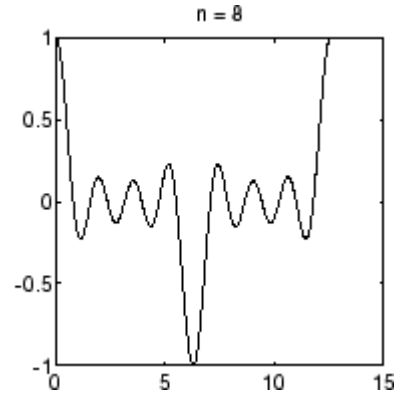
$$\text{diric}(x) = \begin{cases} -1^{k(n-1)} & x = 2\pi k, k = 0, \pm 1, \pm 2, \ldots \\ \dfrac{\sin(nx/2)}{n\sin(x/2)} & \text{otherwise} \end{cases}$$

where n is a user-specified positive integer. For n odd, the Dirichlet function has a period of 2π; for n even, its period is 4π. The magnitude of this function is $(1/n)$ times the magnitude of the discrete-time Fourier transform of the n-point rectangular window.

To plot the Dirichlet function over the range 0 to 4π for `n = 7` and `n = 8`, use

```
x = linspace(0,4*pi,300);
plot(x,diric(x,7)); axis tight;
plot(x,diric(x,8)); axis tight;
```


n = 7
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
0
5
10
15


n = 8
1
0.5
0
-0.5
-1
0
5
10
15

Working with Data

In this section...
"Importing Data from Within the MATLAB Environment" on page 2-12
"Importing Data from Outside the MATLAB Environment" on page 2-12
"Converting Data into a MAT-File" on page 2-12
"Exporting Data" on page 2-13
"Data Precision" on page 2-13

Importing Data from Within the MATLAB Environment

The examples in the preceding sections obtain data in one of two ways:

- By direct input, that is, entering the data manually at the keyboard
- By using a MATLAB or toolbox function, such as `sin`, `cos`, `sawtooth`, `square`, or `sinc`

Importing Data from Outside the MATLAB Environment

Some applications, however, may need to import data from outside MATLAB. Depending on your data format, you can do this in the following ways:

- Load data from an ASCII file or MAT-file with the MATLAB `load` command.
- Read the data into MATLAB with a low-level file I/O function, such as `fopen`, `fread`, and `fscanf`.
- Develop a MEX-file to read the data.

Converting Data into a MAT-File

Other resources are also useful, such as a high-level language program (in Fortran or C, for example) that converts your data into MAT-file format. See the "Manually Converting Data Passed to Functions" documentation for details. MATLAB reads such files using the `load` command.

Exporting Data

Similar techniques are available for exporting data generated within MATLAB. See the “Importing and Exporting Data” documentation for more details.

Data Precision

All Signal Processing Toolbox functions accept double-precision inputs. If you input single-precision floating-point or integer data types, you should not expect to receive correct results and in many cases, an error will occur. Filter Design Toolbox and Fixed-Point Toolbox products enable single-precision floating-point and fixed-point support for most `dfilt` structures.

Selected Bibliography

Algorithm development for Signal Processing Toolbox functions has drawn heavily upon the references listed below. All are recommended to the interested reader who needs to know more about signal processing than is covered in this manual.

[1] Crochiere, R.E., and L.R. Rabiner. *Multi-Rate Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1983. pp. 88-91.

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

[3] Jackson, L.B. *Digital Filters and Signal Processing*. Third Ed. Boston: Kluwer Academic Publishers, 1989.

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

[5] Oppenheim, A.V., and R.W. Schafer. *Discrete-Time Signal Processing*. Englewood Cliffs, NJ: Prentice Hall, 1989.

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

[7] Percival, D.B., and A.T. Walden. *Spectral Analysis for Physical Applications: Multitaper and Conventional Univariate Techniques*. Cambridge: Cambridge University Press, 1993.

[8] Pratt,W.K. *Digital Image Processing*. New York: John Wiley & Sons, 1991.

[9] Proakis, J.G., and D.G. Manolakis. *Digital Signal Processing: Principles, Algorithms, and Applications*. Upper Saddle River, NJ: Prentice Hall, 1996.

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

[11] 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

Filter Design with the FDATool GUI

- "Introduction" on page 3-2
- "Designing the Filter" on page 3-3
- "Analyzing the Filter" on page 3-8
- "Designing Additional Filters" on page 3-10
- "Viewing and Annotating the Filter" on page 3-11
- "Exporting Filters from FDATool" on page 3-17
- "Using Discrete Filter Objects (dfilts)" on page 3-19
- "Where to Find More Information" on page 3-22

Introduction

This section describes how to graphically design and implement digital filters using the Signal Processing Toolbox FDATool GUI. Filter design is the process of creating the filter coefficients to meet specific frequency specifications. Filter implementation involves choosing and applying a particular filter structure to those coefficients. Only after both design and implementation have been performed can your data be filtered.

Although many methods exist for designing the filter coefficients, this chapter focuses on using the basic features of the Filter Design and Analysis Tool (FDATool) GUI. For filter implementation, this chapter uses `dfilt`.

This section includes a brief discussion of applying the completed filter design and filter implementation using MATLAB command line functions, such as `filter`.

For an interactive FDATool demo, type `demos` at the MATLAB command line, and select **Toolboxes**. Expand the tree, scroll down, and select **Signal Processing Toolbox**. Under Filter Design and Analysis, click **Introduction to Filter Design and Analysis Tool**.

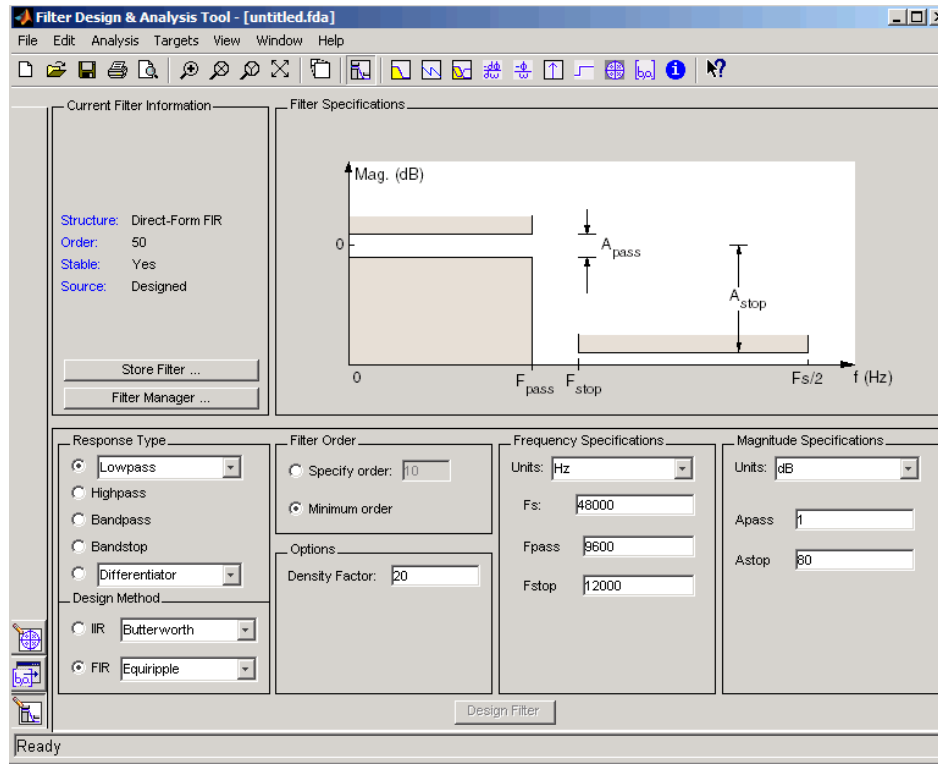
Designing the Filter

This section is a step-by-step introduction to using the Filter Design and Analysis Tool (FDATool) to design an octave-band filter. An octave is the interval between two frequencies having a ratio of 2:1. An octave-band filter is a bandpass filter with high cutoff frequency approximately twice that of the low cutoff frequency. The class of an octave filter is determined by its allowable passband ripple and its stopband attenuation. Refer to the ANSI S1.11–2004 standard for more information. For more information on designing filters, see "FDATool: A Filter Design and Analysis GUI" in Signal Processing Toolbox User's Guide. (Note that you can also access FDATool from SPTool).

1 Start FDATool from the MATLAB command line.

```
fdatool
```

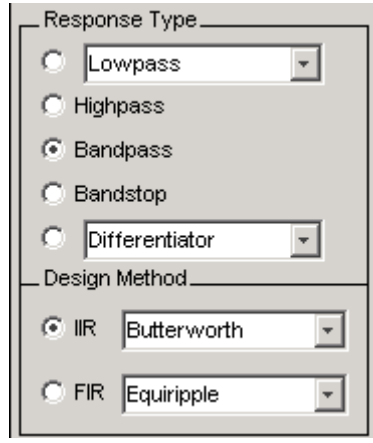
The FDATool dialog opens with a default filter. Its filter information is summarized in the upper left (**Current Filter Information**) and its filter specifications are depicted in the upper right. In addition to displaying filter specification, this upper right pane displays filter responses and filter coefficients.



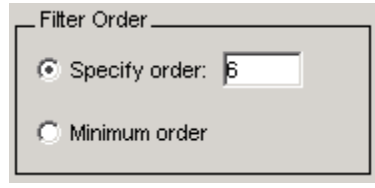
The bottom half of FDATool shows the Filter Design panel, where you specify the filter parameters. Other panels, such as Import filter from workspace and Pole/Zero Editor, which you access with the buttons on the lower left, are also displayed in this area. If you have other products installed, you may see additional buttons.

2 In the **Response Type** pane, select **Bandpass**.

3 In the **Design Method** pane, select **IIR**, and then select `Butterworth` from the selection list.

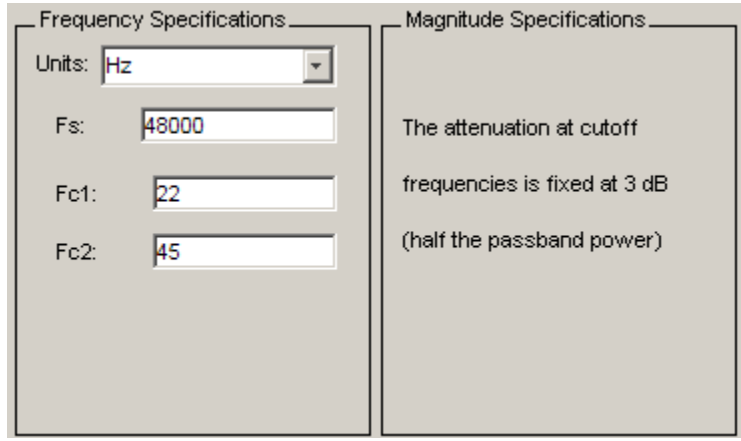


4 For the Filter Order, select **Specify order**, and then enter 6.

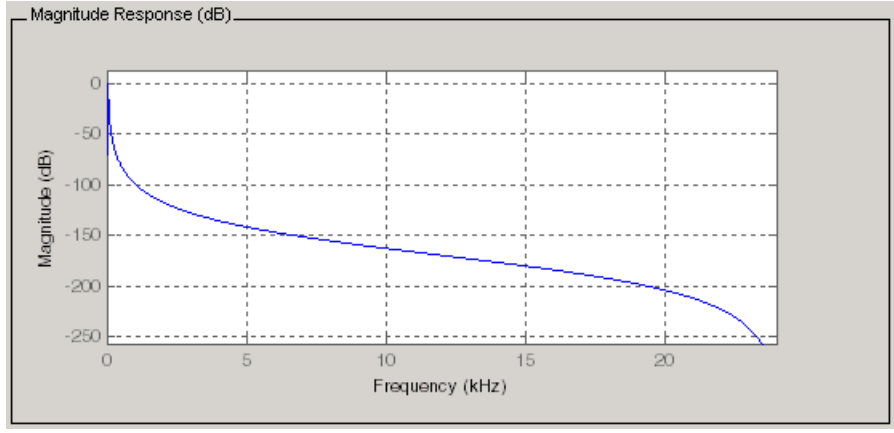


5 Set the Frequency Specifications as follows:

Parameter	Setting	Description
Units	Hz	Units for the parameters
Fs	48000	Sampling frequency
Fc1	22	First cutoff frequency (i.e., the frequency preceding the passband at which the magnitude response is 3 dB below the passband gain)
Fc2	45	Second cutoff frequency (i.e., the frequency following the passband at which the magnitude response is 3 dB below the passband gain)



6 After specifying the filter design parameters, click the **Design Filter** button at the bottom of the design panel to compute the filter coefficients. The display updates to show the magnitude response of the designed filter.



Notice that the **Design Filter** button is disabled after you compute the coefficients for your filter design. This button is enabled again if you make any changes to the filter specifications.

7 Click the **Store Filter** button.

8 In the Store Filter dialog, change the filter name to `Bandpass Butterworth-1` and click **OK** to save the filter in the Filter Manager.

‘

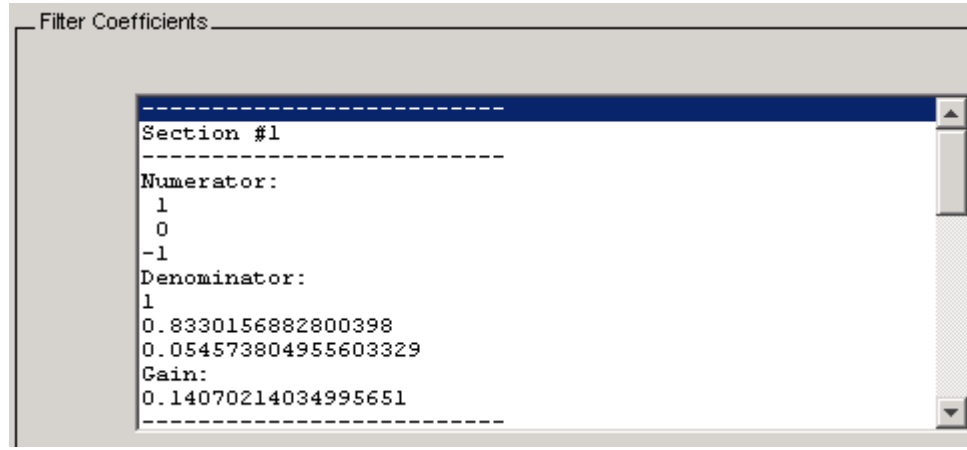
Analyzing the Filter

After designing the filter, you can view the following filter responses in the display region by clicking on the associated toolbar button or by selecting the desired response from the **Analysis** menu.

Response	Toolbar Button Image
Magnitude response	
Phase response	
Magnitude and Phase responses	
Group delay	
Phase delay	
Impulse response	
Step response	
Pole-zero plot	
Filter coefficients	
Filter information	
Zero-phase response–only available from a context menu. Right-click on the *y*-axis of a Magnitude or Magnitude and Phase response plot or select **Analysis > Analysis Parameters**.	

Note Other analyses are available if you have the Filter Design Toolbox product installed.

1. Examine the displayed magnitude response of the filter.
2. Display other responses, as desired. Click the appropriate buttons, shown in the table above or select the desired response from the **Analysis** menu.
3. Click the **Filter coefficients** button to display the filter coefficients.



Designing Additional Filters

You have designed one of the bands of an octave filter bank. This section shows you how to design and save the other nine bands. The following table defines the parameters for the remaining bands. Note that all of the bands use these parameters: **Bandpass**, **IIR – Butterworth** , **order** = 6, **Fs** = 48000 Hz .

Fc1	Fc2	Filter Name
45	89	Bandpass Butterworth-2
89	178	Bandpass Butterworth-3
178	355	Bandpass Butterworth-4
355	708	Bandpass Butterworth-5
708	1413	Bandpass Butterworth-6
1413	2818	Bandpass Butterworth-7
2818	5623	Bandpass Butterworth-8
5623	11220	Bandpass Butterworth-9
11220	22387	Bandpass Butterworth-10

1. Using the parameters listed in the table above, for each table row, set the appropriate the **Fc1** and **Fc2** values.
2. Design the filter by clicking the **Design Filter** button.
3. Click **Store Filter** to save the filter.
4. Change the name to the appropriate filter name shown in the table above.
5. Repeat these steps until all 10 filters are designed and stored.

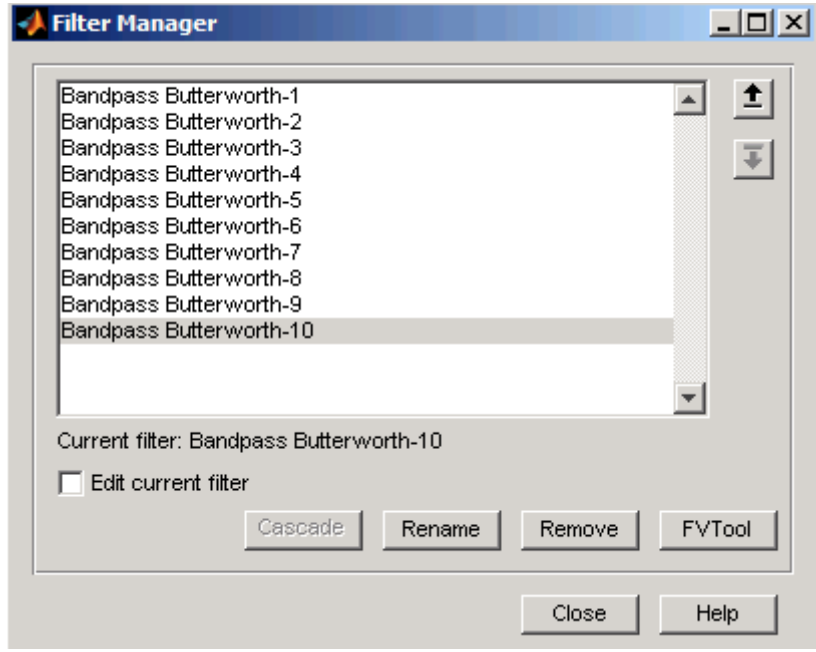
Viewing and Annotating the Filter

In this section...
"Viewing the Filter in FVTool" on page 3-11
"Using FVTool for Annotation" on page 3-15

Viewing the Filter in FVTool

This section teaches you how to use the Filter Visualization Tool (FVTool) to view the octave-band filter. It also describes how to annotate your filter.

1. Click the **Filter Manager** button to display the Filter Manager, which lists your saved filters.

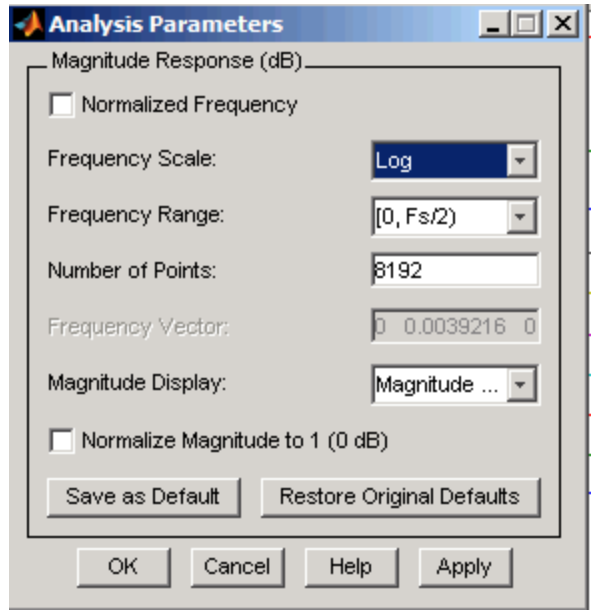


2. Press Ctrl+click on each filter name to select all the filters, and then click **FVTool**. FVTool opens with the filter responses overlaid for easy comparison. (If you want to view a single filter in FVTool, click the **Full**

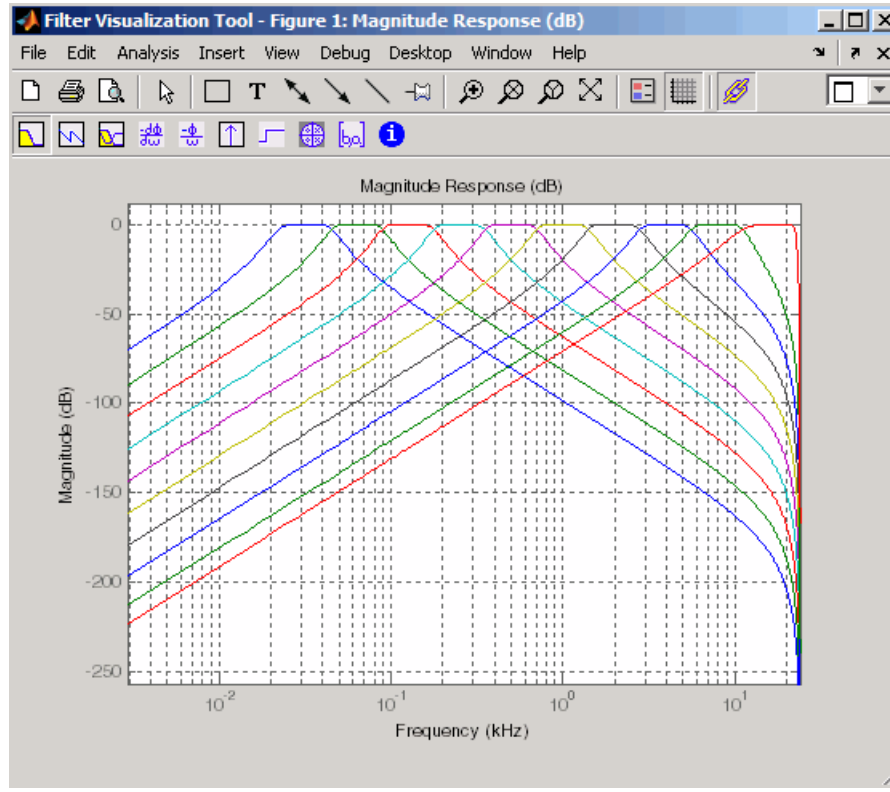
View Analysis button when that filter is shown in the FDATool display panel or select **View > Filter Visualization Tool**).

3 Change the x-axis scale to logarithmic by selecting **Analysis > Analysis Parameters** to display the Analysis Parameters dialog.

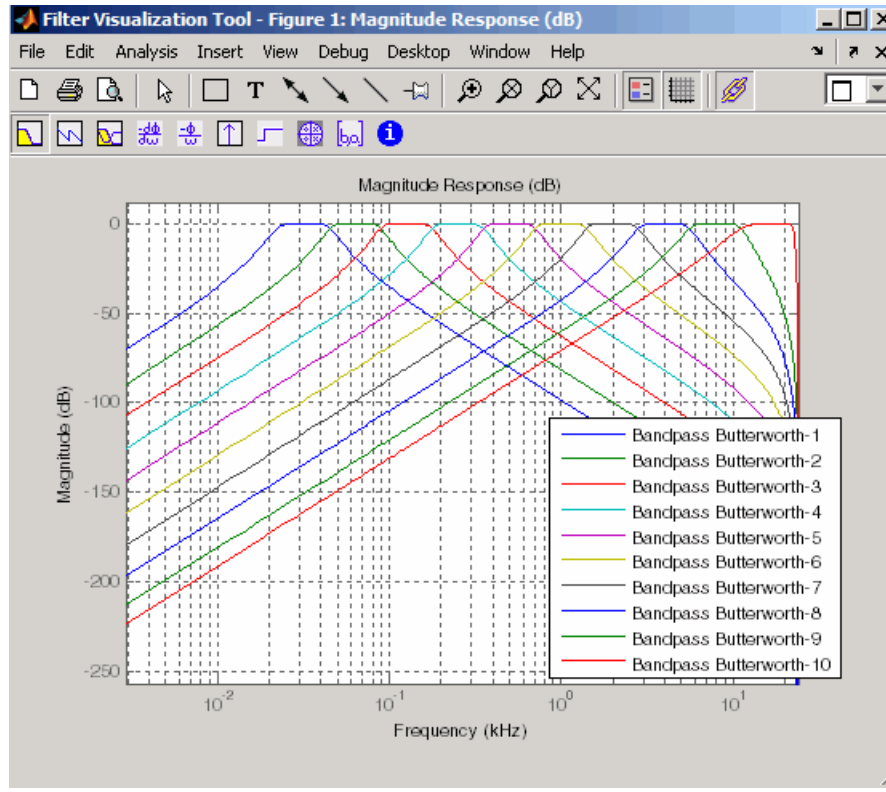
4 Change the **Frequency Scale** to `Log`.



5 Click **OK**.

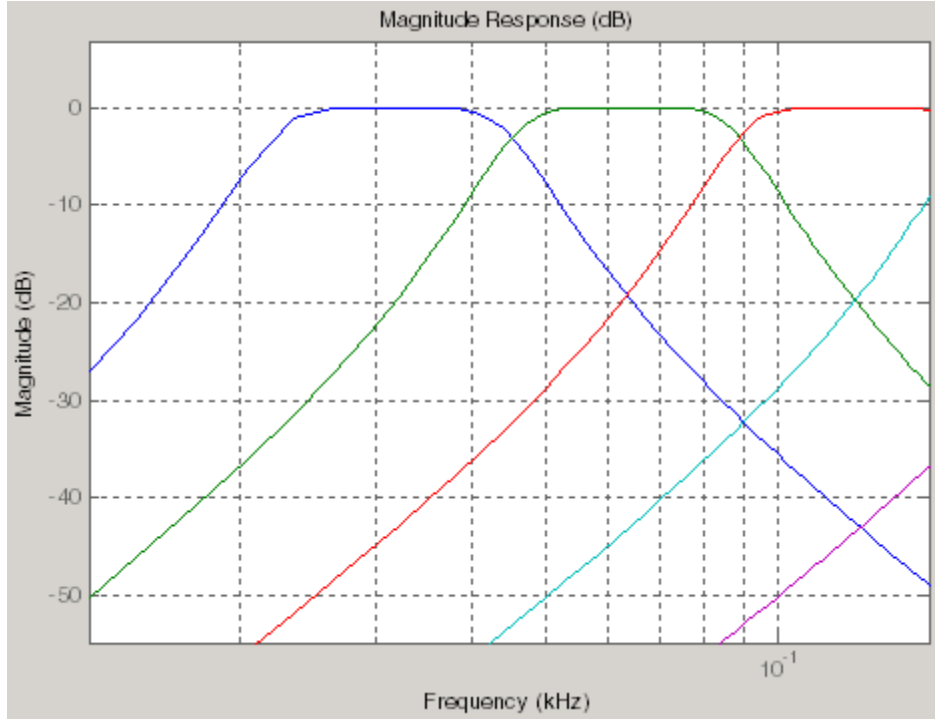


6 Click the **Legend** button to turn on the legend, which you can drag to the desired location.



7 Click the **Legend** button again to turn off the legend.

Use the **Zoom** button and drag a rectangle around the first few passbands to zoom in.



8 Click the **Restore Default View** button to return to the full view.

9 Display other responses, as desired. (The FVTool Analysis toolbar buttons and **Analysis** menu are the same as in FDATool. See "Analyzing the Filter" on page 3-8 for descriptions of the buttons.

Using FVTool for Annotation

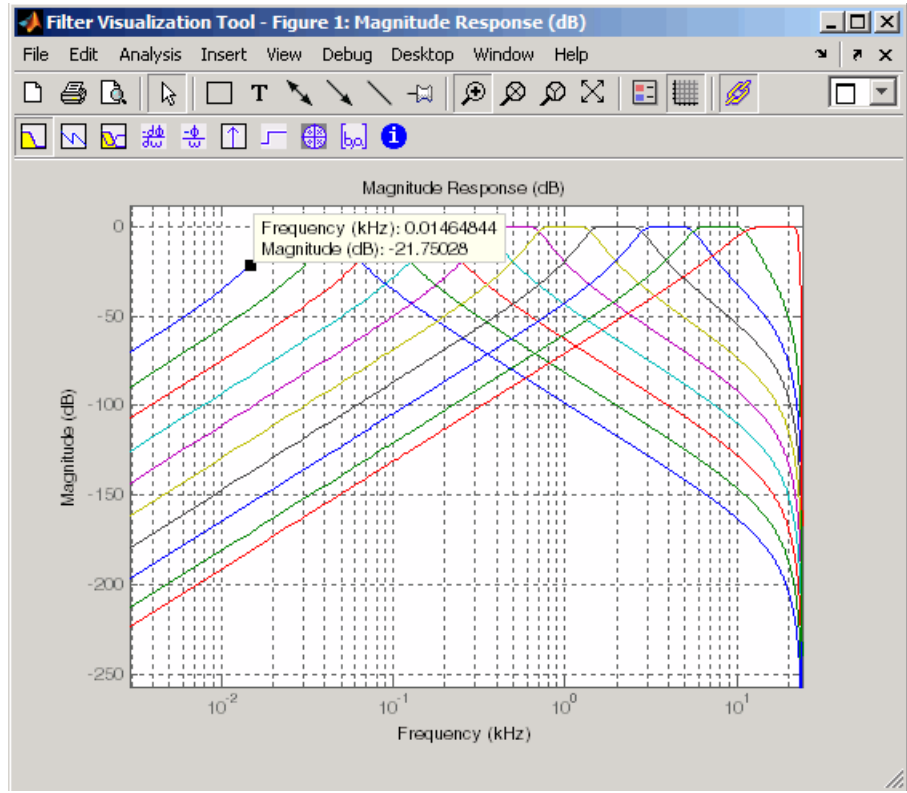
FVTool is also useful for doing further analysis, adding annotations, and printing. Available annotations include adding rectangles, text boxes, arrows and lines, and adding data markers.

For a demo about FVTool, type `demos` at the MATLAB command line, and select **Toolboxes**. Expand the tree, scroll down, and select **Signal Processing Toolbox**. Under Filter Design and Analysis, click **Filter Analysis using FVTool and its API**.

Note Do not close FDATool at this time. You will use it in future sections.

1. Use the toolbar buttons to annotate your response plot. Add a line by clicking one of the line buttons, and then use your mouse to draw the line on your plot.

2. Add a data maker by clicking on a plot at the desired point. The data marker shows the frequency and magnitude at that point.



3. Close FVTool by selecting **File > Close**.

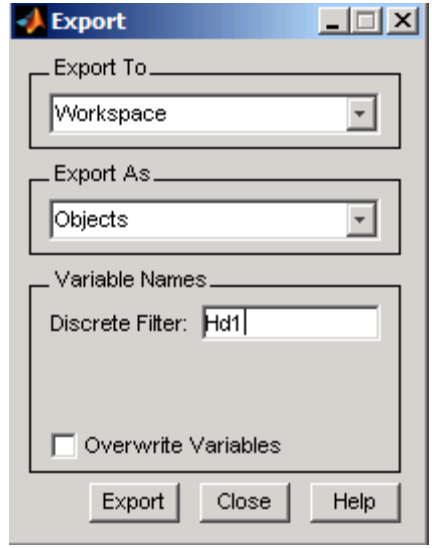
Exporting Filters from FDATool

FDATool provides a simple way to create filter objects (`dfilts`) from your filter designs. This is particularly useful for saving your filter design to the MATLAB workspace for use with command line functions. You can also save your filters to M-files using **File > Generate M-file** to run in scripts or batch files.

1. In FDATool, click **Filter Manager** and highlight only the `Bandpass Butterworth-1` filter.

2. Select **File > Export**.

3. Set **Export to** to `Workspace`. Set **Export as** to `Objects`. In **Discrete Filter** type `Hd1`. Click **Export** to export the first filter in your filter bank to an `Hd1 dfilt` object in the workspace.



4. Repeat steps 1 through 3 for each of the remaining nine filters. Highlight each filter individually to make it the active filter and change the **Discrete Filter** name to match the filter number. When you finish you will have 10 `dfilt` objects in the workspace.

5 Close FDATool by selecting **File > Close**.

6 On the MATLAB command line, verify that your objects were exported by using the `whos` command.

```
whos
  Name      Size             Bytes  Class           Attributes

  Hd1       1x1                     dfilt.df2sos
  Hd10      1x1                     dfilt.df2sos
  Hd2       1x1                     dfilt.df2sos
  Hd3       1x1                     dfilt.df2sos
  Hd4       1x1                     dfilt.df2sos
  Hd5       1x1                     dfilt.df2sos
  Hd6       1x1                     dfilt.df2sos
  Hd7       1x1                     dfilt.df2sos
  Hd8       1x1                     dfilt.df2sos
  Hd9       1x1                     dfilt.df2sos
```

Using Discrete Filter Objects (dfilts)

In this section...

"Using dfilt Objects" on page 3-19

"Filtering with dfilt" on page 3-19

Using dfilt Objects

In the previous section, you created `dfilt` objects containing your filters. In this section, you explore some `dfilt` command line methods. See the `dfilt` reference page for descriptions of all available methods. `dfilt`s are also used in Filter Design Toolbox and Filter Design HDL Coder products, which extend the capabilities of Signal Processing Toolbox functions and GUIs.

For an interactive demo about `dfilt`s, type `demos` at the MATLAB command line, and select **Toolboxes**. Expand the tree, scroll down, and select **Signal Processing Toolbox**. Under Filter Design and Analysis, click **Getting Started with Discrete-Time Filter Objects**.

Filtering with dfilt

1 Type the following on the MATLAB command line to concatenate your filter bank filter objects into a single `dfilt` object.

```
Hd = [Hd1 Hd2 Hd3 Hd4 Hd5 Hd6 Hd7 Hd8 Hd9 Hd10];
```

2 To view the first filter, type `Hd(1)`.

```
Hd(1)

ans =
 FilterStructure: 'Direct-Form II, Second-Order Sections'
       sosMatrix: [3x6 double]
     ScaleValues: [3.40097054256801e-009;1;1;1]
PersistentMemory: false
```

3 A number of methods can be used to view and manipulate the `Hd1` dfilt object. Try the `info` command:

```
info(Hd1)           % Displays filter information

Discrete-Time IIR Filter (real)
-------------------------------
Filter Structure    : Direct-Form II, Second-Order Sections
Number of Sections  : 3
Stable              : Yes
Linear Phase        : No
```

4 You can open FVTool from the MATLAB command line and specify display parameters as follows.

```
F = fvtool(Hd,'Analysis','magnitude')   % Open FVTool with
                                        % magnitude display
set(F,'FrequencyScale','Log')           % Change to log scale
```

This produces the same display as step 5 of "Viewing the Filter in FVTool" on page 3-11 above.

5 Now using the MATLAB command line, create some discrete white Gaussian noise data, which you can then filter using the filter bank.

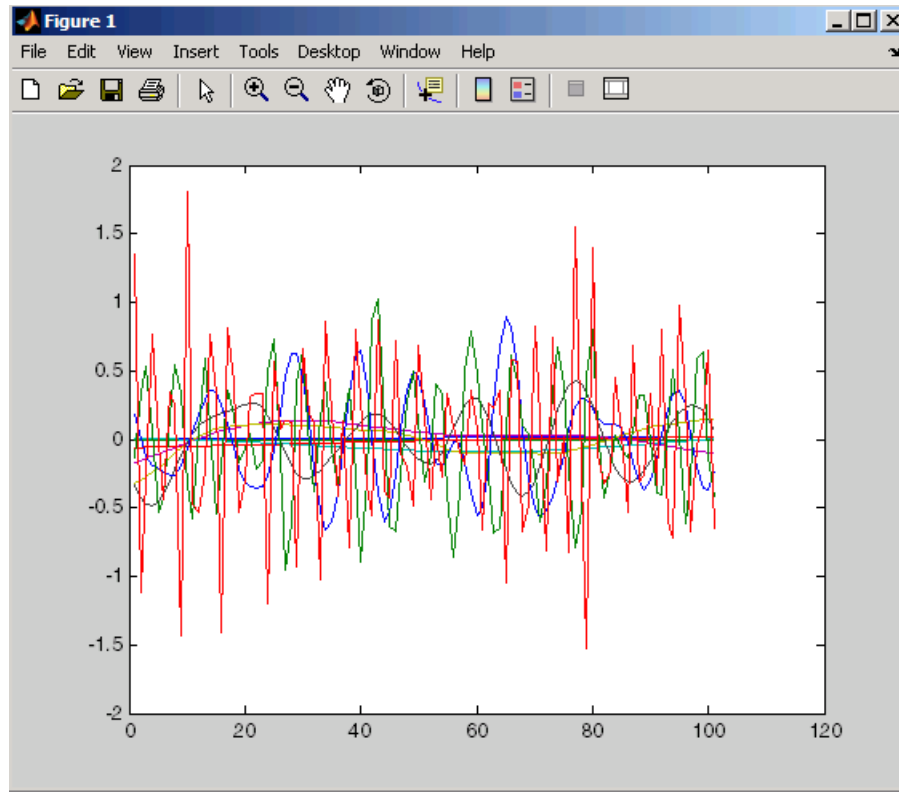
```
rand('state',0);        % Initialize random number generator
Nx = 100000;            % Number of noise data points
xw = randn(Nx,1);       % Create whtie noise
for i=1:10,
  yw(:,i)=filter(Hd(i),xw); % Filter the white noise through
end                         % the entire filter bank.
                            % (:,i) means all rows of column i
```

Note Do not delete this filtered data. You will use it in the Spectral Analysis section.

---

6 Plot the filtered data.

```
plot(yw)
```



The next section discusses spectral analysis, where you analyze this data.

Where to Find More Information

The previous sections described how to use the fundamental features of FDATool, FVTool, and the `dfilt` command. For more advanced information and more complex examples, refer to other sections of Signal Processing Toolbox online help or Signal Processing Toolbox documentation available on The MathWorks Web site (`www.mathworks.com`).

4

Spectral Analysis

- "Introduction" on page 4-2
- "Creating a Spectral Analysis Object" on page 4-4
- "Generating a Spectrum" on page 4-6
- "Changing Spectral Analysis Object Property Values" on page 4-9
- "Measuring Signal Power" on page 4-11
- "Where to Find More Information" on page 4-15

Introduction

In this section...
"Spectral Analysis Examples" on page 4-2
"Spectral Estimators" on page 4-2
"Spectral Analysis Algorithms" on page 4-2
"Spectral Analysis Objects" on page 4-3

Spectral Analysis Examples

This section has two major examples. The first example shows you how to compute and display a mean-square spectrum. The second example shows you how to measure the average power of a signal.

Note The examples in this section use filters and data you created in Chapter 3, "Filter Design with the FDATool GUI".

Spectral Estimators

Spectral analysis includes three types of spectral estimators — power spectral density (PSD), mean-square spectrum (MSS) and pseudo spectrum.

- Power spectral density (`psd`) measures power per unit of frequency and has power/frequency units.
- Mean-square (power) spectrum (`msspectrum`) measures power at a specific frequency.
- Pseudospectrum (`pseudospectrum`) returns a pseudo spectrum that does not have any units.

Spectral Analysis Algorithms

Signal Processing Toolbox software provides algorithms to compute the above spectral estimators. The following table indicates which algorithms are available to compute each type of estimator and produce a spectrum object.

For general information on objects, see *Object-Oriented Programming*. For details on spectrum objects, see the `spectrum` reference page.

Spectral Estimator	Algorithms
Power spectral density (`psd`)	Burg (`spectrum.burg`), Covariance (`spectrum.cov`), Modified covariance (`spectrum.mcov`), Thomson multitaper method (MTM) (`spectrum.mtm`), Periodogram (`spectrum.periodogram`), Welch (`spectrum.welch`), Yule-Walker autoregressive (`spectrum.yulear`) See also `dspdata.psd`
Mean-square spectrum (`msspectrum`)	Periodogram (`spectrum.periodogram`), Welch (`spectrum.welch`) See also `dspdata.msspectrum`
Pseudo spectrum (`pseudospectrum`)	Eigenvector (`spectrum.eigenvector`), MUSIC (Multiple Signal Classification) (`spectrum.music`) See also `dspdata.pseudospectrum`

Spectral Analysis Objects

Spectral analysis objects contain property values for the particular algorithm. To calculate a spectrum estimate, you first create an estimator object using one of the algorithms (e.g., `h = spectrum.burg`), and then pass it to one of the spectrum estimator objects with your data (e.g., `Hpsd = psd(h,x)`). In this example, `h` is a Burg spectrum object, `x` is the original input data, and `Hpsd` is the resulting PSD.

For more information and examples, see the Getting Started with Spectral Analysis Objects demo.

Creating a Spectral Analysis Object

Continuing the example in Chapter 3, “Filter Design with the FDATool GUI”, you will create a spectral analysis object that uses default property values, and then, use it to generate a mean-square spectrum of the data filtered through the octave-band filter bank.

Create a Welch’s method spectrum object using default values by typing the following at the MATLAB command line.

```
h = spectrum.welch

h =
    EstimationMethod: 'Welch'
       SegmentLength: 64
      OverlapPercent: 50
          WindowName: 'Hamming'
        SamplingFlag: 'symmetric'
```

If you want to specify parameters instead of using default values, you use syntax similar to the following, which uses a Kaiser window and a segment length of 66. Note that the Kaiser window has an additional parameter (`Beta`). See the `spectrum.welch` reference page for more information.

```
h = spectrum.welch('kaiser',66,50)

h =
    EstimationMethod: 'Welch'
       SegmentLength: 66
      OverlapPercent: 50
          WindowName: 'Kaiser'
                Beta: 0.5000
```

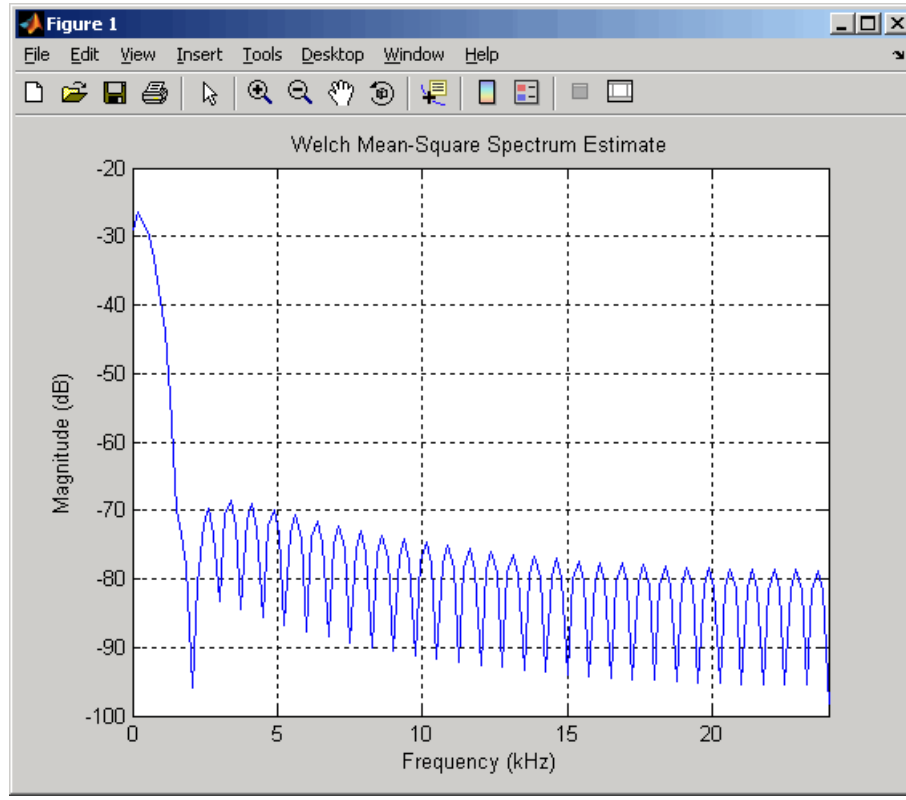
Note For the rest of this example, use the default `h = spectrum.welch` object.

You can also change some of the property values of a created spectrum object. See “Changing Spectral Analysis Object Property Values” on page 4-9 for more information.

Generating a Spectrum

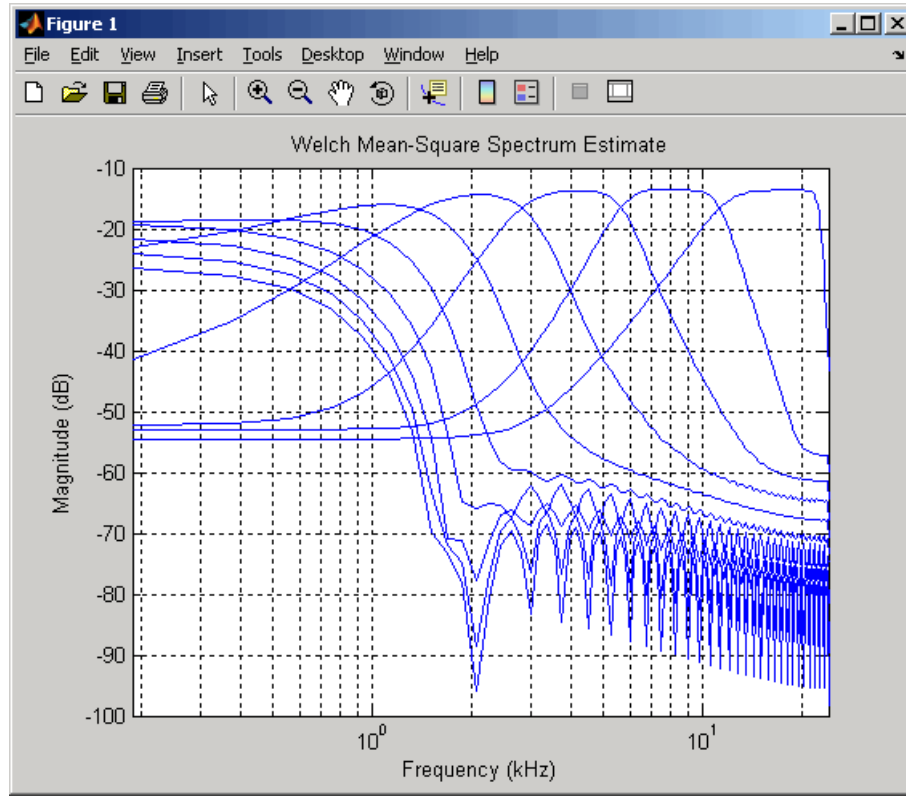
To generate a spectrum, you use a spectral estimation method on your spectrum object and data. Using the data, `yw`, created in Chapter 3, "Filter Design with the FDATool GUI", create a mean-square spectrum, by typing

```
Fs = 48000;      % Set the sampling frequency

h = spectrum.welch;  % Create spectrum object
msPyyw = msspectrum(h,yw(:,1),'Fs',Fs)

msPyyw =
                    Name: 'Mean-Square Spectrum'
                    Data: [129x1 double]
            SpectrumType: 'Onesided'
    NormalizedFrequency: false
                     Fs: 48000
             Frequencies: [129x1 double]

plot(msPyyw)
```



To see the spectra for all the filters, type

```
for i=1:10
  msspectrum(h,yw(:,i),'Fs',Fs);
  hold on;                  % Plot each spectrum in the same
end                         %   figure window
set(gca,'Xscale','log')     % Change to log scale
```



The syntax for using `psd`, `msspectrum`, or `pseudospectrum` is the same. As shown in the example above, the first input is the spectrum object (`h`) and the second input (`yw`) is the signal to measure, followed by any settable properties using property-value pairs (e.g., `'Fs', Fs`). To set property-value pairs, you list the property first and then the value for that property.

Changing Spectral Analysis Object Property Values

In this section...

"Using the set command to Set Property Values" on page 4-9

"Using Options Objects to Set Property Values" on page 4-10

Using the set command to Set Property Values

After you have created a spectrum object, you can use the `set` method or dot notation to change any of its properties, except its `EstimationMethod` property. (Since `EstimationMethod` is the central property of a particular spectrum object, you cannot change it.) To change the window from Hamming (used in the Welch object created above) to Chebyshev, use dot notation as follows:

```
h.WindowName = 'Chebyshev'

h =
    EstimationMethod: 'Welch'
       SegmentLength: 66
      OverlapPercent: 50
          WindowName: 'Chebyshev'
       SidelobeAtten: 100
```

or use the `set` method,

```
set(h,'WindowName','Chebyshev')
```

As illustrated above, Chebyshev windows have a sidelobe attenuation parameter that automatically appears in the list of properties. To change a window parameter, you must use a cell array containing the window name and parameter value, such as,

```
h = spectrum.welch({'Chebyshev',80})

h =
    EstimationMethod: 'Welch'
       SegmentLength: 64
      OverlapPercent: 50
```

```
          WindowName: 'Chebyshev'
       SidelobeAtten: 80
```

To view the amplitude and magnitude responses of various windows, type `wintool` to open the Window Design and Analysis Tool.

Using Options Objects to Set Property Values

Another way to set properties for a particular estimation method is to use the options object associated with that method. For example, use the following syntax to create an options object from the spectrum object for use with the mean-square spectrum:

```
Hopts  = msspectrumopts(h)   % Create options object

Hopts =
                    NFFT: 'Nextpow2'
    NormalizedFrequency: true
                     Fs: 'Normalized'
           SpectrumType: 'Onesided'
               CenterDC: false
```

You can then change any of the options properties using `set`, followed by the options object and property-value pair(s), such as

```
set(Hopts,'Fs',48000)
```

To pass the options object to the first `msspectrum` in the filtered data, use

```
msspectrum(h,yw(:,1),Hopts)
```

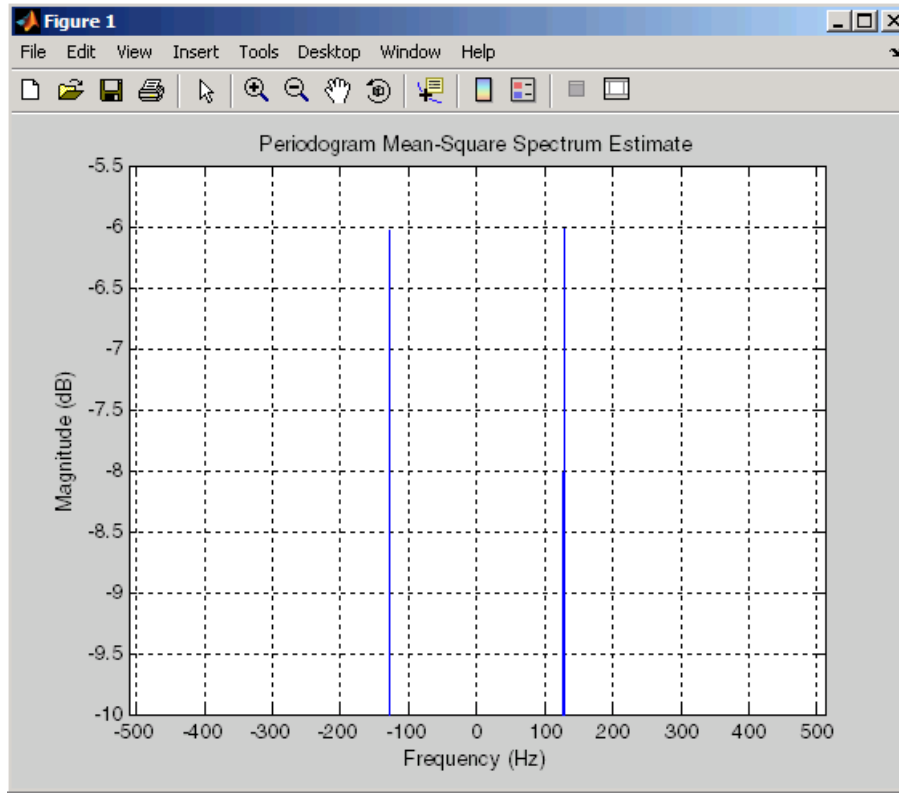
Options objects that correspond to `psd` and `pseudospectrum` are `psdopts` and `pseudospectrumopts`, respectively.

Measuring Signal Power

This section shows you how to measure the average power of a deterministic periodic signal. This type of signal is continuous in time, but produces a discrete power spectra. A signal made up of sinusoids is an example of a power signal that has infinite energy, but finite average power . The example below shows how to measure and estimate the average power of a single sinusoidal signal with a peak amplitude of 1 v and a frequency component at 128 Hz.

First, you measure the signal's average power using a periodogram spectrum object, and then calculate and plot the mean-square (power) spectrum.

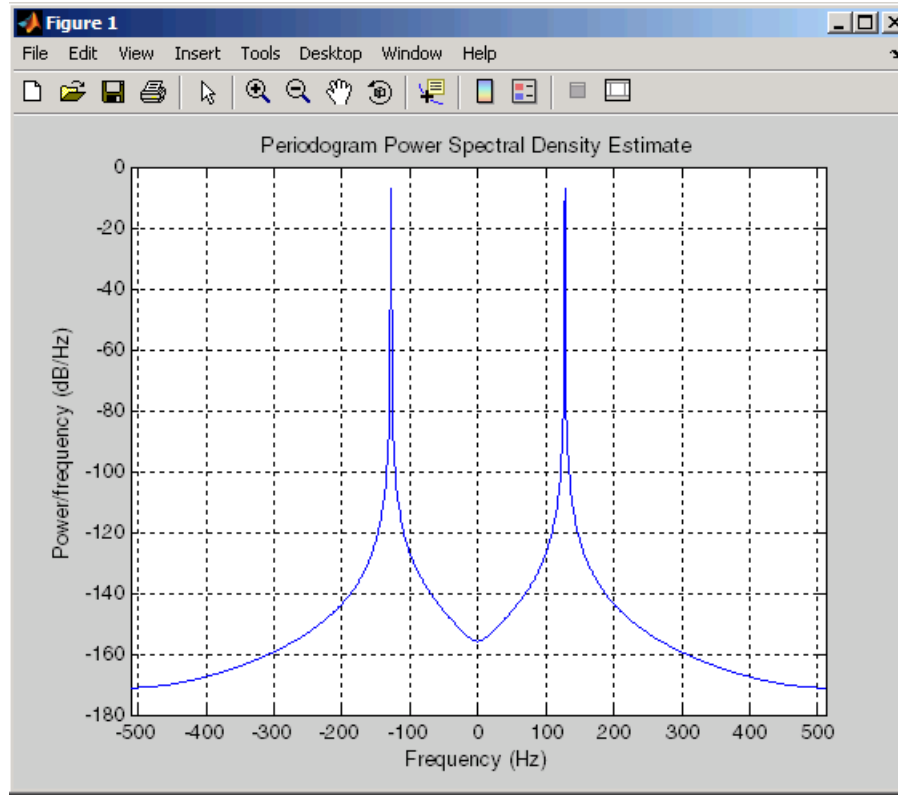
```
Fs = 1024;                      % Sampling frequency
t  = 0:1/Fs:1-(1/Fs);           % Time vector
A  = 1;                         % Peak amplitude
F1 = 128;                       % Hz
x  = A*sin(2*pi*t*F1);          % Sinusoidal signal

hp = spectrum.periodogram('hamming');  % Create periodogram

% Create options object and set properties
hpopts = psdopts(hp,x);
set(hpopts,'Fs',Fs,'SpectrumType','twosided','centerdc',true);

msspectrum(hp,x,hpopts);
v = axis; axis([v(1) v(2) -10 -5.5]);   % Zoom in Y.
```



The average power of each complex sinusoid is approximately -6 dB.

Another way to calculate a signal's average power is by "integrating" the area under the power spectral density (PSD) curve. To do this, you use the `psd` method on the spectrum object created above (`hp`) to get a PSD data object, and then use the `avgpower` method.

```
hpsd = psd(hp,x,hpopts);
plot(hpsd);
```



Notice that the peaks of this plot are not the same height as the mean-square spectrum peaks plotted above. The area under the PSD curve is the measure of the average power, not the peak heights. By using the `avgpower` method and converting the result to dB, you can see that the average power is the same for both of them.

```
power_freqdomain = avgpower(hpsd)

power_freqdomain =

    0.5000
```

According to Parseval's theorem, the total average power in a sinusoid is the same, whether it's computed in the time domain or the frequency domain.

You can verify the signal's estimated total average power by summing the signal in the time domain.

```
power_timedomain = sum(abs(x).^2)/length(x)

power_timedomain =

    0.5000
```

Now, converting this linear value to a logarithmic value (and taking only the positive frequencies), you see that the average power is the same as that shown in the mean-square spectrum plot above.

```
10*log10(power_freqdomain/2)

ans =

   -6.0206
```

Where to Find More Information

The previous sections described how to use the fundamental features of spectral analysis. For more advanced information and more complex examples, refer to other sections of Signal Processing Toolbox online help or Signal Processing Toolbox documentation available on The MathWorks Web site (`www.mathworks.com`).

Index

A

aliasing
    sinc functions 2-10
analog signals. *See* signals
annotation 3-15
ASCII files
    importing 2-12
average power 4-11

B

bandpass filters 3-3
buttons
    filter analysis 3-8

C

cell array 4-9
chirp signals 2-7
continuous signals. *See* signals
cycles
    duty 2-6

D

data
    importing 2-12
    matrices 2-2
    multichannel matrix 2-2
    multichannel signals 2-6
    supported types 1-3
    time vectors 2-4
    vectors 2-2
demos 1-9
    dfilt 3-19
    FDATool 3-2
    FVTool 3-15
    spectral analysis 4-3
`dfilt` 3-19
digital signals 1-4
Dirichlet functions
    definition 2-10
dot notation 4-9
double-precision inputs 1-3
duty cycles 2-6

E

export 3-17
extensibility 1-8

F

`fdatool` GUI
    analysis buttons 3-8
    group delay 3-8
    impulse response 3-8
    magnitude response 3-8
    phase delay 3-8
    phase response 3-8
    pole-zero plots 3-8
    step response 3-8
files
    M 2-12
    MAT 2-12
    MEX 2-12
filter coefficients
    icon 3-8
filter information
    icon 3-8
filters
    linear time-invariant digital 1-4
    octave-band 3-3
`fopen` function 2-12
`fread` function 2-12
functions
    Dirichlet 2-10
    sinc 2-9

G

gauspuls function
    pulse trains 2-8
graphical user interface (GUI) 1-7
    See also sptool GUI, fdatool GUI, wintool GUI, fvtool GUI, wvtool GUI
group delay
    icon 3-8

I

import
    data 2-12
impulse 2-5
impulse response
    icon 3-8
inverse Fourier transforms
    See sinc function 2-9

M

M-files
    expanding a toolbox 1-8
magnitude and phase response
    icon 3-8
magnitude response
    icon 3-8
MAT-files
    converting to 2-12
    importing 2-12
matrices
    data 2-3
matrix 2-2
MEX-files 2-12
multichannel data 2-6

N

noise data 3-20

O

octave-band filters 3-3
options object 4-10

P

P-V pairs 4-8
periodogram 4-11
phase delay
    icon 3-8
phase response
    icon 3-8
pole-zero plot
    icon 3-8
power spectral density 4-2
power spectrum 4-2
property values
    changing 4-9
property-value pairs 4-8
pulse trains
    example 2-8
    pulstran function 2-8

R

ramp 2-5
references
    DSP 2-14

S

sawtooth function
    example 2-6
sawtooth wave 2-6
set 4-9
signal
    average power 4-11

signals
adding noise 2-4
aperiodic 2-7
chirp 2-7
continuous (analog) 1-4
`diric` function 2-10
discrete (digital) 1-4
generating 2-5
multichannel 2-6
periodic 2-6
plotting 2-4
`pulstran` function 2-8
representing 2-2
sawtooth 2-6
`sinc` 2-9
sinusoidal 2-4
square wave 2-6
single-precision inputs 1-3
spectrum
creating 4-6
mean-square spectrum 4-2
options object 4-10
PSD 4-2
pseudo spectrum 4-2
spectrum object
creating 4-4
Welch 4-4
`sptool` GUI
data entering 2-12
`square` function
example 2-6
square wave
See `square` function 2-6
step 2-5
step response
icon 3-8

T

time vectors 2-4
tools
interactive GUIs 1-7

U

unit impulse function 2-5
unit ramp function 2-5
unit sample multichannel 2-6
unit step function 2-5

V

vectors
data representation 2-2
waveform generation 2-4

W

waveforms. *See* signals
Welch spectrum object
example 4-4
white noise 2-4

Z

zero-phase response 3-8