Filter Design Package
for
Audio Precision

Version 1.1 for Windows 95,
Windows 98, and Windows NT

Momentum Data Systems, Inc.

Copyright Information

Copyright © 1989–2000
Momentum Data Systems
17330 Brookhurst Street, Suite 140
Fountain Valley, CA 92708
USA

Copyright © 2000
Audio Precision, Inc.
PO Box 2209
Beaverton, OR 97075-2209
USA

World rights reserved. No part of this publication may be stored in a retrieval system, transmitted, or reproduced in any way, including, but not limited to, photocopy, photograph, magnetic or other record, without permission in writing from the publisher.

Information in this manual is subject to change without notice, and does not represent a commitment on the part of Audio Precision, Inc.

System Two Cascade™ and APWIN™ are trademarks of Audio Precision, Inc.

Windows is a registered trademark of Microsoft Corporation.

LIMITED WARRANTY

Momentum Data Systems, Inc. specifically disclaims all warranties, expressed or implied, including but not limited to, implied warranties of merchantability and fitness for a particular purpose with respect to defects in the software and documentation and the program license granted herein, in particular, and without limiting operation of the program license with respect to any particular application, use, or purpose. In no event shall MDS be liable for any loss or profit or any other commercial damage including but not limited to special, incidental, consequential or other damages.

CUSTOMER SERVICE

Momentum Data Systems Technical Support

Phone: (714) 378-5805

Fax: (714) 378-5985

E-mail: `tech@mds.com`

Filter Design Package Table of Contents

1 INTRODUCTION

The Filter Design Package for Audio Precision (FDP) has been produced by Momentum Data Systems, Inc., to allow the user to create filters for downloading to Audio Precision System Two Cascade hardware.

This manual explains the operation of FDP. It is not intended to be a tutorial on digital signal processing, since there are many excellent texts on the subject, and it is assumed that the user has had a certain amount of academic or professional exposure to the subject. A list of references can be found in Section 5.1 on page 5-1.

More advanced versions of this software are available, including versions enabling the design of finite impulse response (FIR) filters. These versions also support more powerful filter design techniques. Please call Momentum Data Systems for further information.

1.1 Design Capabilities of FDP

Infinite Impulse Response Filter Design

- Lowpass, highpass, bandpass, and bandstop filters
- Orders up to 6 for lowpass filters
- Orders up to 4 for highpass filters
- Orders up to 8 for bandpass and bandstop filters
- Choice of five analog prototype filters:
    - Butterworth
    - Tschebyscheff
    - Inverse Tschebyscheff
    - Elliptic
    - Bessel
- Digital transformation performed by bilinear transformation

Graphical Output

- Magnitude Response vs. Frequency
- Log Magnitude vs. Frequency

- Phase Response vs. Frequency
- Group Delay vs. Frequency
- Impulse Response vs. Time
- Step Response vs. Time
- Pole and zero locations

1.2 FDP Features

- **Recycling of input for comparative analysis**—input for a filter is retained until a new specification is called for. This allows various IIR designs to be compared.
- **High mathematical precision**—64-bit floating point precision is used for all design calculations to ensure maximum accuracy. Certain critical calculations are carried out in 128-bit precision.
- **Filter specification files**—Allow the retention and retrieval of filter specifications.
- **Context-sensitive help**—Help screens are available for all data entry fields.
- **Cursor display on frequency domain plots**—The *x*- and *y*-coordinates of the cursor are continuously updated when the left mouse button is held down.
- **Graphical zooming**—All plots can be zoomed with the mouse.

1.3 Installation

FDP is provided on the Audio Precision APWIN CD-ROM. During APWIN installation, FDP is installed by checking the 'AP Filter Design Package' checkbox in the 'Select Components' dialog box. This causes the executable `apfdp.exe` and its support files to be copied to the folder `\Program Files\Audio Precision\apfdp\`. It also adds the 'AP Filter Design Package' shortcut to the 'Audio Precision APWIN 2.10' group in the start menu.

1.4 Design Method

A normalized lowpass analog transfer function is generated from the filter specifications. This normalized analog transfer function is then transformed via the analog transform formulas with the values suitably chosen for prewarping.

The unnormalized transfer function is then transformed to the digital domain via the bilinear transformation. The filter characteristics, including the impulse response, are simulated via cascaded second order sections. Poles and zeros are grouped using the L. B. Jackson algorithm to minimize stability problems. Details of the transformations are appended to the output file `SFIL.OUT`. An example is given in Section 5.2 on page 5-3.

1.5 System Operation

FDP uses a simple, menu-driven interface. It is designed to allow standard filter designs to be produced with the minimum of effort. Filter specifications, coefficients, and data files can be saved, so that the design can be retrieved, downloaded to Audio Precision hardware, and examined in detail in other applications.

All dialog boxes have a 'Help' feature which provides context-sensitive help at any time. All dialog boxes also have a 'Cancel' button, which causes FDP to revert to the previous stage of the filter design, without losing data.

FDP understands both integer and scientific (floating-point) number notation. For instance, the number 5 123 can be entered as '5123', '5123.00', or '5.123e03'.

FDP can be started in the following ways:

- From the 'Start' menu, choose 'Programs > Audio Precision > APWIN 2.10 > AP Filter Design Package.'
- From the 'Start' menu, choose 'Run' and type `apfdp` in the box. This assumes that the folder containing `apfdp.exe` is in your path.

2 USING FDP

FDP is controlled via a user interface of a main menu bar with pull-down menus and dialog boxes. In addition, a toolbar allows quick access to the most common functions. The menu bar consists of the following entries:

- **File**—Load previously stored filter and analysis input specifications
- **View**—View design data and filter files
- **Filter**—Select filter type to design
- **Output**—Select plotting options and output files
- **Window**—Select output and graphical options

After the system starts up, the following screen is displayed:

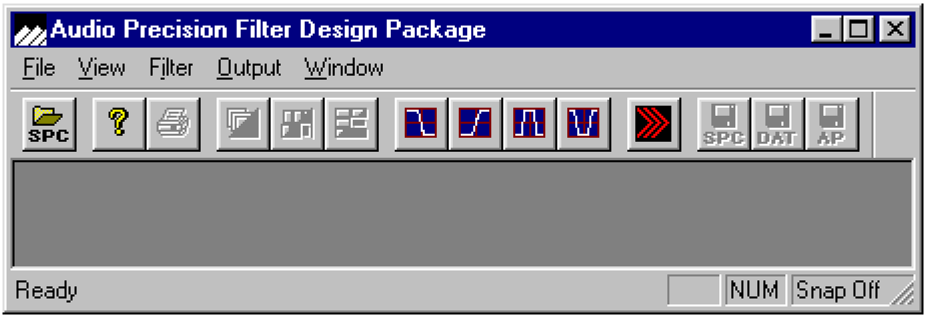


Figure 2-1: Start-up screen

The File, View, Output, and Window menus are covered in this chapter. The Filter menu is covered in Section 3.2 on page 3-2.

2.1 Toolbar and Status Bar

The toolbar allows the user to select the most commonly used features:

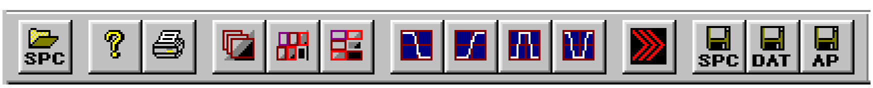


Figure 2-2: Toolbar

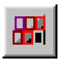
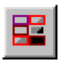
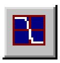
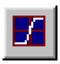
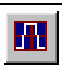
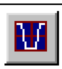
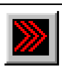
Icon	Function Name	Equivalent Menu Option	Page
	Load filter file	**File**/Load specifications	**2-3**
	About	**File**/About APFDP	**2-5**
	Print	**File**/Print	**2-3**
	Cascade windows	**Window**/Cascade	**2-11**
	Tile windows vertically	**Window**/Tile vertically	**2-11**
	Tile windows horizontally	**Window**/Tile horizontally	**2-11**
	Lowpass	**Filter**/Lowpass	**3-2**
	Highpass	**Filter**/Highpass	**3-3**
	Bandpass	**Filter**/Bandpass	**3-4**
	Bandstop	**Filter**/Bandstop	**3-4**
	Start design	**Filter**/Start design	**3-5**
	Write specification file	**Output**/Create specification file	**2-8**
	Write Matlab file	**Output**/Create Matlab file	**2-9**
	Write Audio Precision file	**Output**/Create AP filter file	**2-9**

Table 1: Toolbar functions

The status bar at the bottom of the screen displays information about the current menu selection. At the right hand side, it shows the current state of the Caps Lock and Num Lock keys, and whether plot selections snap to the grid lines. For more information on Snap to Grid, see Section 2.3.2 on page 2-8.

2.2 File Menu

To select a menu, place the cursor over the menu title and click the left mouse button. Move the cursor over the menu choices and click the left mouse button again to select the choice. The enabled options are listed in black, while disabled options appear gray. The File menu appears as follows:

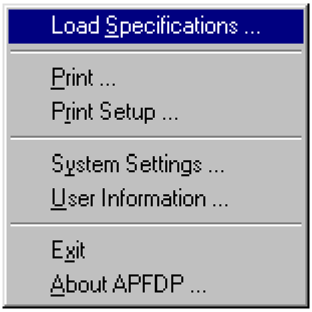


Figure 2-3: File menu

2.2.1 Load Specifications

When this option is selected, a file browser appears. All filter specification files in the current directory with extension `.SPC` are displayed. After a specification file is selected, a dialog box asks for verification of the filter type. After confirmation, a specification dialog box for the chosen filter type appears, allowing the design to be modified if required.

A filter specification file can be created by selecting the Create Specification File command in the Output menu after a filter design is complete. See Section 2.3.3 on page 2-8.

2.2.2 Print

On completion of a design, plots can be printed. The Print Selection dialog box appears:

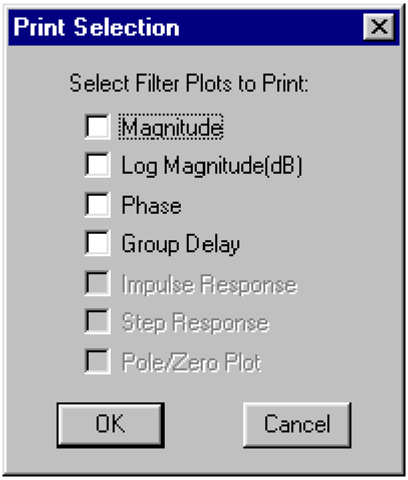


Figure 2-4: Print Selection dialog box

Choose the plots that you want to print by clicking the checkboxes. Any of the plots on the screen can be printed. For information on plotting, see Section 2.3.1 on page 2-6.

Note: Hardcopies of plots print in monochrome. The print menu does not apply to text windows, such as filter specification files. These files must be printed outside FDP.

2.2.3 Printer Setup

Select the desired printer setup. The printer setup dialog box varies according to the type of printer. These parameters are not saved after the system exits.

2.2.4 System Settings

The System Settings dialog box controls global aspects of FDP operations:

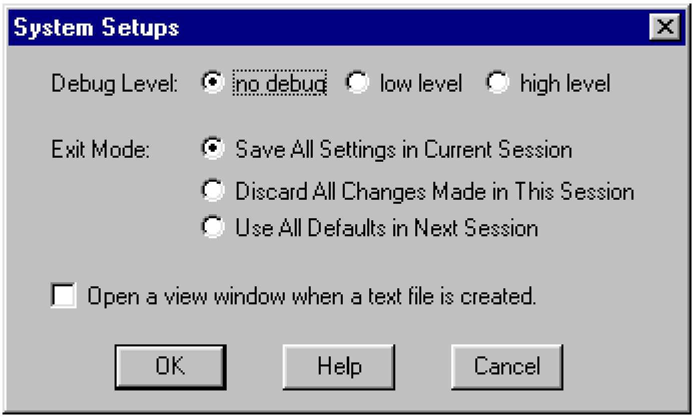


Figure 2-5: System Settings dialog box

- **Debug Level**—If an error occurs in FDP, debug information is written to the file `SFIL.STA`. The debug level determines how much information is exported, from none to the maximum amount.
- **Exit Mode**—Determines which changes are saved from one FDP session to the next.
- **Open a view window when a text file is created**—If this option is selected, a window is opened showing the contents of newly created text files. This applies to all output text files (`.SPC`, `.DAT`, `.AFL`, `.AFH`, and `.AFW`).

2.2.5 User Information

The text in the user information fields will be displayed at the foot of plots and printouts.

2.2.6 Exit

Exits the application.

2.2.7 About APFDP

Displays the current version number and technical support information. Telephone, fax, and e-mail contact information can also be found at the beginning of this manual.

2.2.8 View Menu

The View menu controls the appearance of FDP:

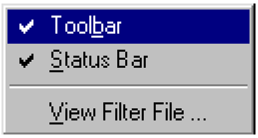


Figure 2-6: View menu

2.2.9 Toolbar and Status Bar

These options toggle the toolbar and status bar displays. Checkmarks (✓) indicate which displays are active.

2.2.10 View Filter File

This option displays a file browser in which filter specification files (files with extensions `.SPC`, `.DAT`, `.AFL`, `.AFH`, and `.AFW`) can be selected. These text files can then be viewed inside FDP.

2.3 Output Menu

The Output menu contains controls data output:

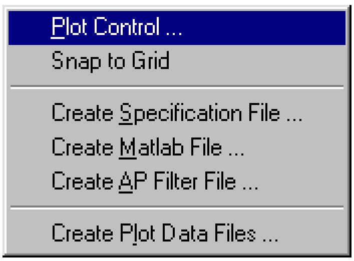


Figure 2-7: Output menu

2.3.1 Plot Control

The Plot Control dialog box controls the appearance of FDP plots:

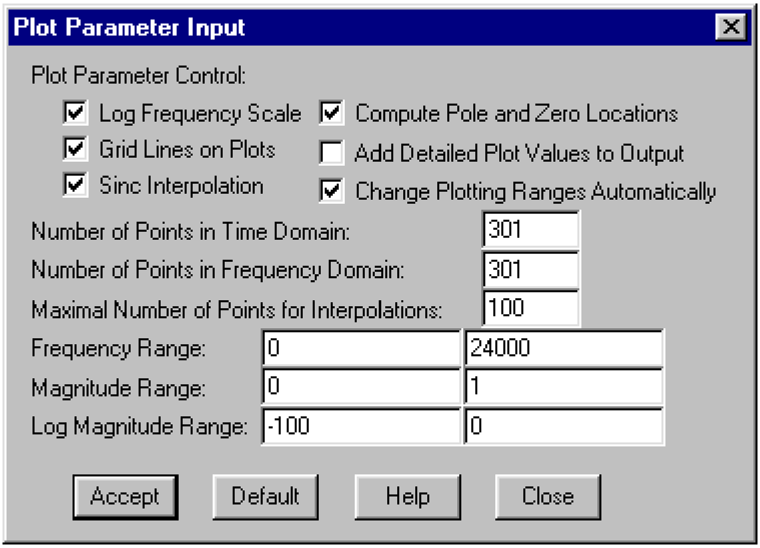


Figure 2-8: Plot Control dialog box

All checkbox options are global system settings. That is, each option remains in effect until explicitly reset via this dialog box. The settings are not activated until the next replot. This means, for example, that to see the change from linear frequency scale to logarithmic frequency scale, the ‘Accept’ button must be clicked to force a replot. The following options are available:

- **Log Frequency Scale**—Check this box for a logarithmic frequency scale. The default number of decades is 5. The lower limit of the frequency range will be reset to reflect this range. Other ranges can be selected by explicitly setting frequency range values.
- **Grid Lines on Plots**—Grid lines will appear on all plots if this option is selected. Otherwise, only tick marks on the axes will be shown.
- **Sinc Interpolation**—If this option is selected, time domain plots (impulse and step responses) will be interpolated between samples to give a smoother appearance.
- **Compute Pole and Zero Locations**—This option is ignored in FDP.
- **Add Detailed Plot Values to Output**—Detailed reports are available for magnitude, log magnitude, phase, and group delay plots. These reports consist of the data values used in the plots. They are appended to the file `SFIL.OUT` if this option is active.

- **Change Plotting Ranges Automatically**—When this option is active, the plotting ranges will automatically be set to default values whenever a new filter design is started. When this option is inactive, the previous values in the plotting ranges will be used for all subsequent plots, allowing comparison of different filters.
- **Number of Points in Time Domain**—By default, FDP will compute the impulse and step responses over 301 time points. The number of points can be changed using this field. A maximum number of 32 000 points is allowed. However, computing this many points may take as much as five minutes, depending on the speed of the computer.
- **Maximal Number of Points for Interpolations**—The number of samples used in interpolating time domain plots. Values between 21 and 150 are allowed.
- **Frequency Range**—The default *x*-axis range is $[0, \frac{1}{2} f_{\text{sampling}}]$. If another range is desired (for examining transition bands, for instance), it should be entered in these two fields.
- **Magnitude Range**—The default *y*-axis range for magnitude plots is 0 to 1. This can be changed to any value between 0 and 10. Typically, smaller ranges are used to examine ripple characteristics.
- **Log Magnitude Range**—The default *y*-axis range for log magnitude plots is –100 dB to 0 dB. This can be changed to any value between –200 dB and +10 dB.

2.3.2 Snap to Grid

Graphical zooms are achieved by clicking the right mouse button in the graph and dragging to form a rectangle. The plot is zoomed to the rectangle when the button is released. If the Snap to Grid option is active, the plot is zoomed to the largest rectangle formed by the gridlines which completely encloses the rectangle.

2.3.3 Create Specification File

FDP allows a filter specification to be saved in a text file. The file is given a `.SPC` extension. These files can later be loaded into FDP using the Load Specifications command in the File menu. See Section 2.2.1 on page 2-3.

2.3.4 Create Matlab File

The designed filter coefficients can be saved in a text file as second-order sections for loading into Matlab or other mathematical applications. The file consists of 6 columns by *n* rows, where *n* is the number of second-order sections. The gain of the filter is absorbed into the first section. The file is given a `.DAT` extension.

2.3.5 Create AP Filter File

The filter coefficients can be saved in a text file as second-order sections for downloading directly into Audio Precision System Two Cascade hardware. See the APWIN System Two Cascade User Manual for more information on downloadable filters. After a filename has been chosen, a comment string can be entered for inclusion in the file. This string is read by APWIN and displayed when the 'Filter Info' button in the User Downloadable Filters dialog box is clicked. An Audio Precision filter file is given a `.AFL`, `.AFH`, or `.AFW` extension, depending on whether the filter is lowpass, highpass, or weighting (bandpass or bandstop).

2.3.6 Create Plot Data Files

The data displayed in an FDP plot can be exported to a file for processing in other tools. The Create Plot Data Files dialog box determines which plots are exported:

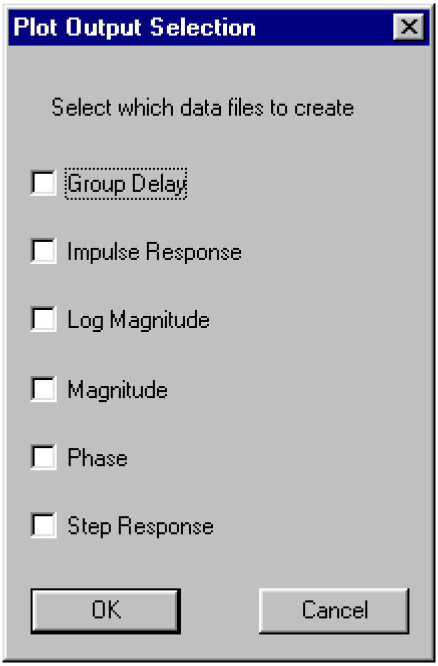


Figure 2-9: Create Plot Data Files dialog box

FDP saves all plots under the same filename, with extensions `.GDL` (group delay), `.IMP` (impulse response), `.LMN` (log magnitude), `.MAG` (magnitude), `.PHA` (phase), and `.STP` (step response). The Create Plot Data Files menu determines which of these files will be created.

2.4 Window Menu

The Window menu controls the graphical displays:

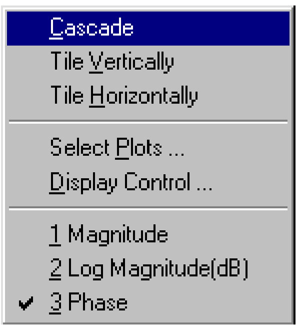


Figure 2-10: Typical Window menu

2.4.1 Cascade, Tile Vertically, and Tile Horizontally

These options arrange the plot windows on the screen in different ways. 'Cascade' places them on top of one another, offset to the right and down. 'Tile Vertically' places the windows side-by-side. 'Tile Horizontally' places them one above the other.

2.4.2 Select Plots

The Select Plots option determines which plots are displayed inside FDP:

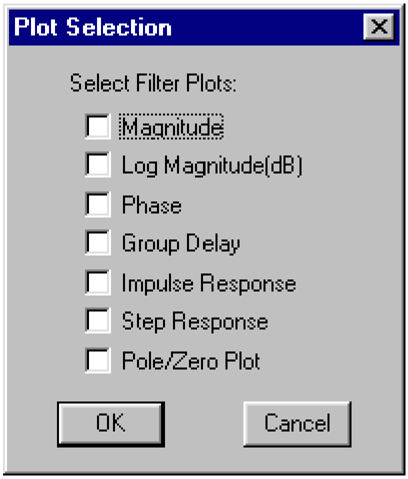


Figure 2-11: Select Plots dialog box

Any of seven plot types can be selected. A window for each plot will be displayed on the screen. A check (✓) indicates which plots are active.

2.4.3 Display Control

This dialog box controls the color, font, and line style of plots and prints. Please note that all hardcopy is monochrome. However, font sizes and line styles are preserved.

2.4.4 Choose Window

A list of all active windows appears at the bottom of the Window menu. When a window is chosen from the list, it is brought to the foreground inside FDP.

3 FILTER DESIGN

3.1 Overview

A typical filter magnitude specification looks like this:

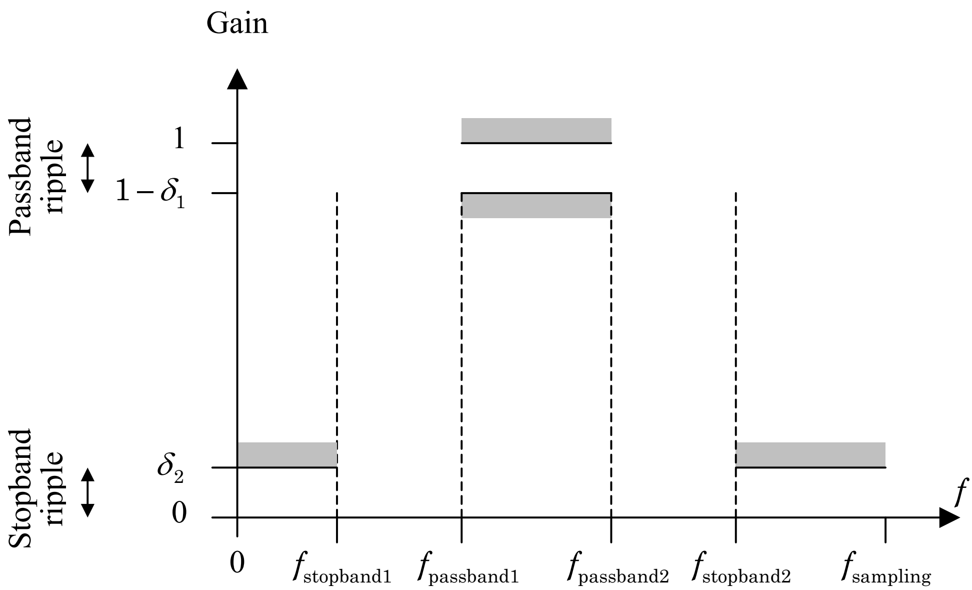


Figure 3-1: Bandpass filter magnitude specification

The filter consists of *passbands* and *stopbands* separated by *transition bands*. The *break frequencies* define the edges of the bands. The gain of the filter is specified in the passbands and stopbands, but is not specified in the transition bands. Furthermore, the passband and stopband gains are allowed to vary over a range (shown between the shaded regions in the diagram). This variation is termed *ripple*. For instance, the filter is specified to have a gain in the range $[1-\delta_1,1]$ between frequencies $f_{\text{passband1}}$ and $f_{\text{passband2}}$ in the diagram above.

FDP accepts ripple values in dB. These are defined as

$$\text{passband ripple (dB)} = -20\log_{10}(1-\delta_1)$$
$$\text{stopband ripple (dB)} = -20\log_{10}(\delta_2)\ .$$

A typical range of values for passband ripple is 0.01 dB to 3 dB. A typical range of values for stopband ripple is 20 dB to 100 dB. FDP requires the passband ripple to be less than the stopband ripple.

FDP uses specifications of the form in Figure 3-1 to define the filter. After requesting break frequencies and ripple values, FDP determines the *filter order* (number of filter stages) needed to meet the specification for each of five *filter prototypes*. Once the prototype has been chosen, the digital filter is designed. The choice of prototype will be described in Section 3.2.5 on page 3-5.

3.2 Filter Menu

The Filter menu and equivalent toolbar buttons offer the choice of four filter types:

Figure 3-2: Filter menu

When one of the filter types is chosen, FDP displays a dialog box containing fields for entering the specifications of the filter. Frequencies are assumed to be in Hz, while ripples are assumed to be in dB. Each field must contain a valid entry. FDP traps invalid entries.

3.2.1 Lowpass

The following dialog box appears if Lowpass is selected:

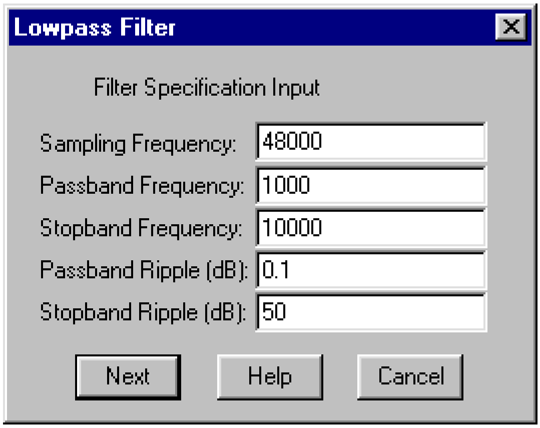


Figure 3-3: Lowpass dialog box

Enter frequencies and ripple values in the fields. If you are designing a single filter for several sample rates, refer to Section 3.4 on page 3-7. The following equation must hold for the break frequencies:

$$f_{\text{passband}} < f_{\text{stopband}} < \tfrac{1}{2} f_{\text{sampling}} .$$

3.2.2 Highpass

The following dialog box appears if Highpass is selected:

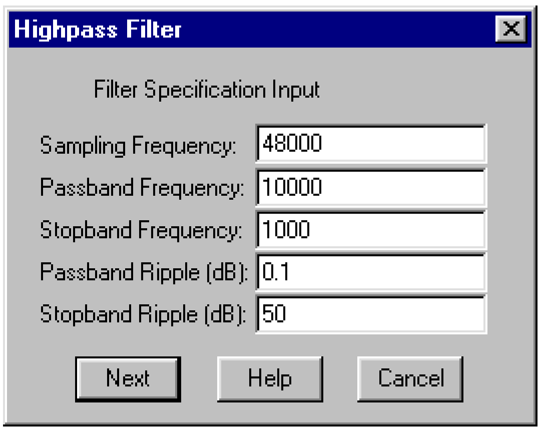


Figure 3-4: Highpass dialog box

The following equation must hold for the break frequencies:

$$f_{\text{stopband}} < f_{\text{passband}} < \tfrac{1}{2} f_{\text{sampling}} .$$

3.2.3 Bandpass

The following dialog box appears if Bandpass is selected:

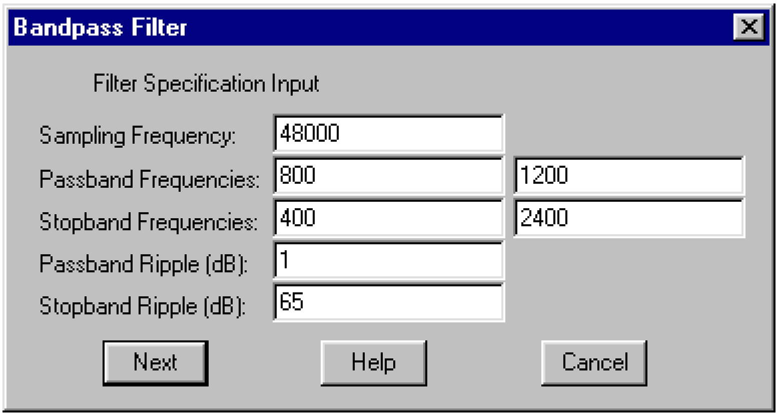


Figure 3-5: Bandpass dialog box

The following equation must hold for the break frequencies:

$$f_{\text{stopband1}} < f_{\text{passband1}} < f_{\text{passband2}} < f_{\text{stopband2}} < \tfrac{1}{2} f_{\text{sampling}} .$$

For an equiripple design, the transition regions are set internally such that the transition regions are symmetric. This is required to design an acceptable filter.

3.2.4 Bandstop

The following dialog box appears if Bandstop is selected:

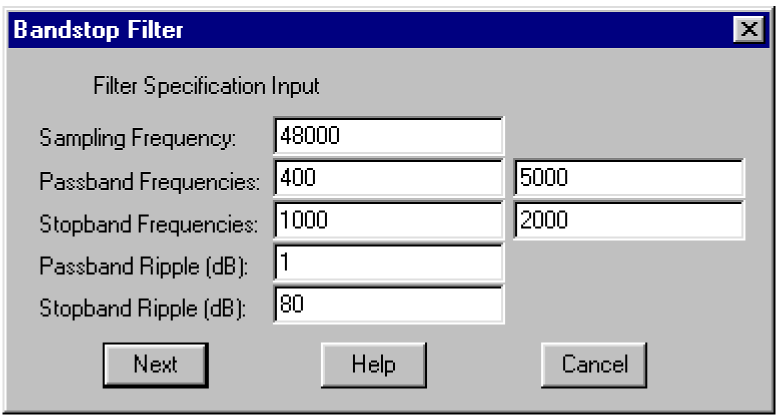


Figure 3-6: Bandstop dialog box

The following equation must hold for the break frequencies:

$$f_{\text{passband1}} < f_{\text{stopband1}} < f_{\text{stopband2}} < f_{\text{passband2}} < \tfrac{1}{2} f_{\text{sampling}}.$$

3.2.5 Start Design

This menu choice resumes a filter design at the point it left off when a 'Cancel' button was pressed. If no filter has been designed, it starts a new filter design.

3.3 Filter Prototypes

FDP offers five classical analog filter designs to use as a prototype for the digital filter:

Filter Type	Passband	Stopband	Rolloff Rate
Butterworth	Maximally flat	Monotonic	Average
Tschebyscheff	Ripples	Monotonic	High
Inverse Tschebyscheff	Monotonic	Ripples	High
Elliptic	Ripples	Ripples	Highest
Bessel	Monotonic	Monotonic	Low

Table 2: Analog prototype filter characteristics

An additional strength of the analog Bessel prototype is that its group delay is maximally flat. However, this property is not necessarily preserved in the digital filter.

The elliptic filter provides the sharpest rolloff rate. That is, for a given filter order, passband ripple, and stopband attenuation, the elliptic will have the narrowest transition bands of any filter. However, low frequency elliptic designs may require precise placement of the zeros, which may not be possible. If problems occur, an inverse Tschebyscheff should be considered.

The number of filter orders required to meet the specifications increases as the rolloff rate of the filter decreases. Thus a specification which requires 4 orders if implemented with an elliptic filter may require 6 orders if implemented with a Butterworth filter. Since the Audio Precision System

Two Cascade has a restriction on the maximum number of filter orders, the choice of analog prototype may be restricted to the sharper rolloff filter types. FDP will respond with an error message if the following limits are exceeded:

Filter Type	Maximum order
Lowpass	6
Highpass	4
Weighting	8

Table 3: Maximum filter size on Audio Precision System Two Cascade

3.3.1 Choosing a filter prototype

After filling in the fields in the filter specification dialog box, click ‘Next’. FDP checks that the break frequencies satisfy the equations for the chosen filter type. If so, the following dialog box appears:

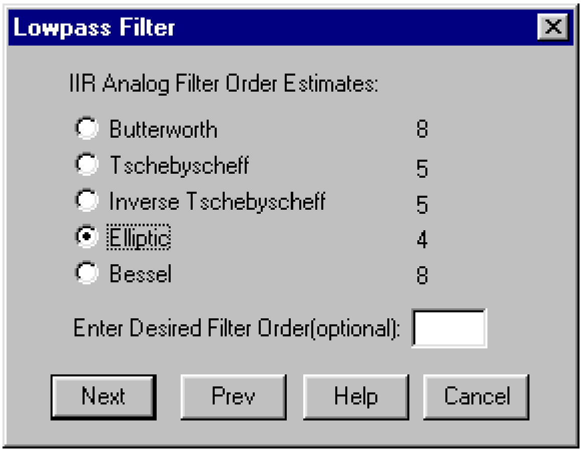


Figure 3-7: Prototype Selection dialog box

Here we show a typical result for a lowpass filter. Since the number of orders is restricted to 6 for lowpass filters, neither the Butterworth nor the Bessel filters can be used. The choice among the remaining three filter types depends on other criteria, such as the amount of phase distortion that can be tolerated.

3.3.2 Obtaining graphical output

After choosing the filter prototype, click ‘Next’. FDP designs the digital filter and displays the plots chosen in the Plot Selection dialog box, described in Section 2.4.2 on page 2-11.

3.4 Multiple Sample Rates

Unlike analog (continuous-time) filters, a digital filter is designed for a specific sample rate. If the sample rate changes, the filter response will also change. Often, a filter is desired whose response is independent of sample rate. It is then necessary to design a filter for each sample rate that will be encountered. FDP reduces the work required in two ways:

- After the first filter has been designed, clicking on the same filter button in the toolbar will pull up the filter specifications. Only the sample rate need be changed to design a filter for the new sample rate.
- Filters designed for different sample rates can be appended to existing Audio Precision filter files. For instance, a single file named `BUTTER_1000.AFL` might contain a Butterworth lowpass filter with a cutoff frequency of 1 kHz at sample rates of 32 000 Hz, 44 100 Hz, and 48 000 Hz.

To design a file with filters at multiple sample rates, follow this procedure:

- Choose a filter type from the Filter menu, or from the buttons on the toolbar.
- Enter the specifications of the filter.
- Enter one of the desired sample rates and click ‘Next’.
- Choose a filter prototype and click ‘Next’. FDP designs the filter.
- Save the filter to an Audio Precision filter file by clicking on the ‘Save AP Filter File’ button on the toolbar. FDP prompts for an information string and creates the file.
- Choose the same filter type again from the Filter menu or the toolbar buttons. FDP displays the original filter specifications.
- Enter another of the desired sample rates and click ‘Next’.
- Choose another filter prototype—it need not be the same one, or of the same order, as the first—and click ‘Next’. FDP designs the filter.

- Append this filter to the original file by clicking on the ‘Save AP Filter File’ button on the toolbar, choosing the original filename, and clicking ‘Append’ at the prompt.

The last four operations can be repeated as many times as needed. APWIN does not require the sample rates to be in numerical order. If a sample rate is used that does not correspond to any of the sample rates in the file, APWIN chooses the filter with the closest sample rate.

4 EXAMPLE DESIGNS

This chapter presents four example filter designs.

4.1 1 000 Hz lowpass filter, 65 536 Hz sample rate

We use the following specifications:

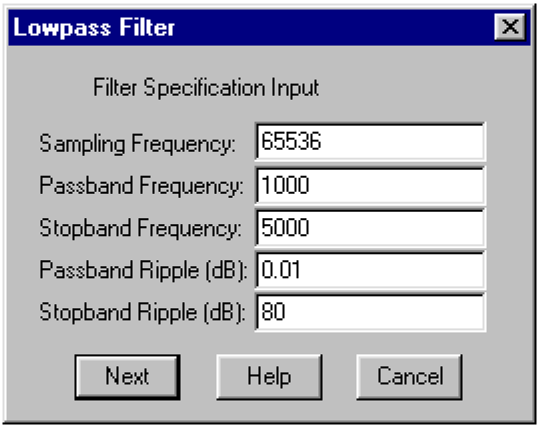


Figure 4-1: Lowpass filter specifications

This filter has tight restrictions on the passband ripple and stopband attenuation. FDP indicates that a 6-pole Tschebyscheff filter will suffice. The resulting magnitude response is:

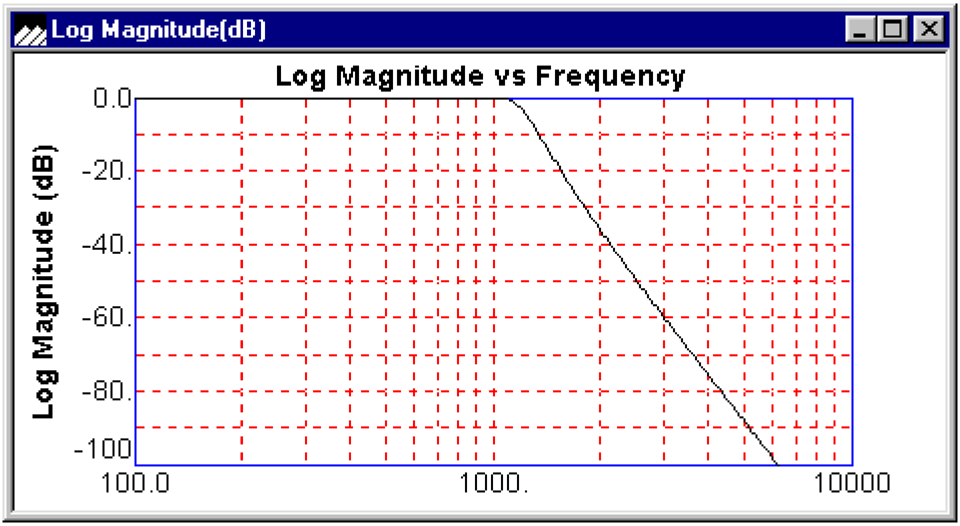


Figure 4-2: Lowpass filter log magnitude response

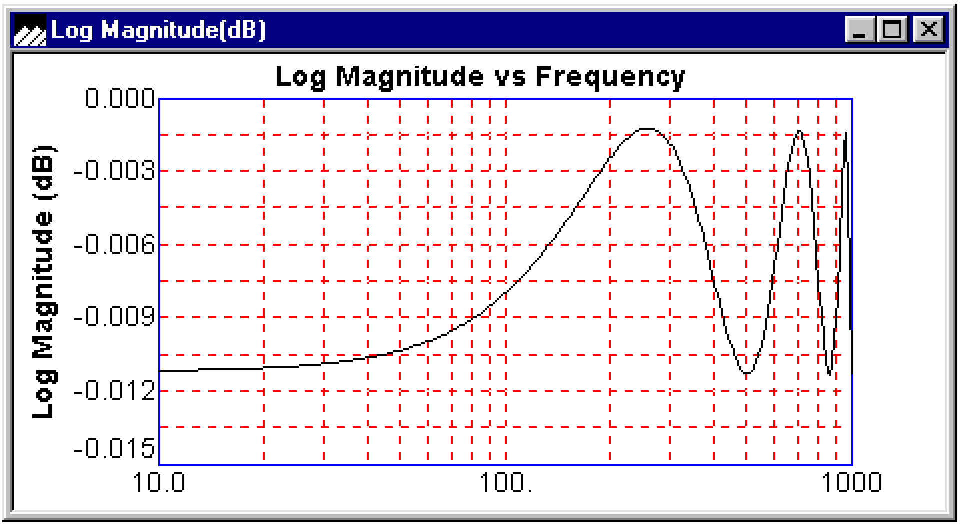


Figure 4-3: Ripple characteristics of lowpass filter

The design exceeds the specification in the stopband: the –80 dB point is at around 4 500 Hz. (The specification can be met exactly by reducing the ripple in the passband to the point where any further reduction in ripple would require an extra filter order. For this design, the optimal point is when the passband ripple is 0.00157 dB.) A close-up of the ripple characteristics of the original filter shows that the ripple is indeed 0.01 dB. The slight loss of 0.0012 dB is due to coefficient quantization.

FDP produces the following `.AFL` (Audio Precision lowpass filter) file when the 'Save AP Filter File' button is clicked. See Section 2.3.5 on page 2-9 for more details.

```
#AP Filter File
#Filter Type: Low Pass
#Analog Filter Type: Tschebyscheff
#Passband Ripple: 0.1000E-01 dB
#Stopband Attenuation: 0.8000E+02 dB
#Passband Cutoff Frequency: 0.100000E+04 Hz
#Stopband Cutoff Frequency: 0.500000E+04 Hz

info: 1000 Hz Tschebyscheff lowpass

###*****************************************************************
***********
###@ Filters for 65536.00 Hz Sample Rate
###*****************************************************************
***********

sample_rate: 65536.00
biquad: -9.400252103805542000000e-001 4.422166347503662100000e-001
1.095652580261230500000e-003 5.477666854858398400000e-004
5.477666854858398400000e-004
biquad: -9.529479742050170900000e-001 4.570752382278442400000e-001
2.063512802124023400000e-003 1.031756401062011700000e-003
1.031756401062011700000e-003
biquad: -9.776712656021118200000e-001 4.838646650314331100000e-001
3.093004226684570300000e-003 1.546502113342285200000e-003
1.546502113342285200000e-003
```

4.2 80 Hz highpass filter, 44 100 Hz sample rate

The specifications and magnitude response of this filter are shown below. FDP indicates that a 4-pole Butterworth filter will do the job.

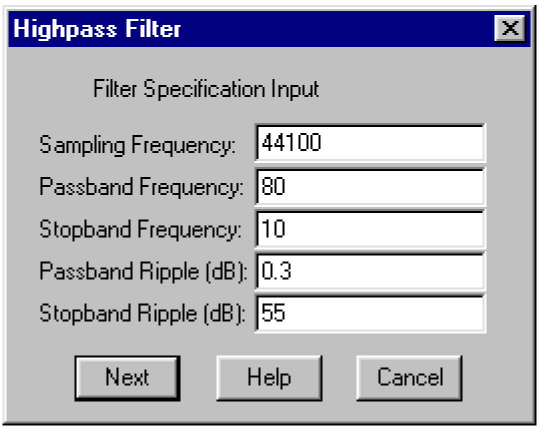


Figure 4-4: Highpass filter specifications

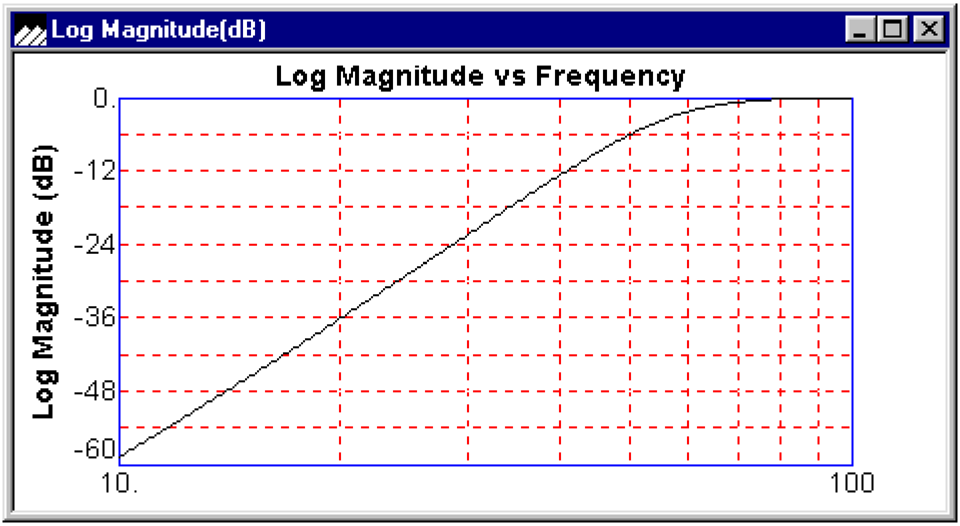


Figure 4-5: Highpass filter log magnitude response

4.3 300 Hz–4 000 Hz bandpass filter, 11 025 Hz sample rate

The specifications and magnitude response of this filter are shown below. An 8-pole elliptic design satisfies the constraint on maximum filter order (see Table 3 on page 3-6).

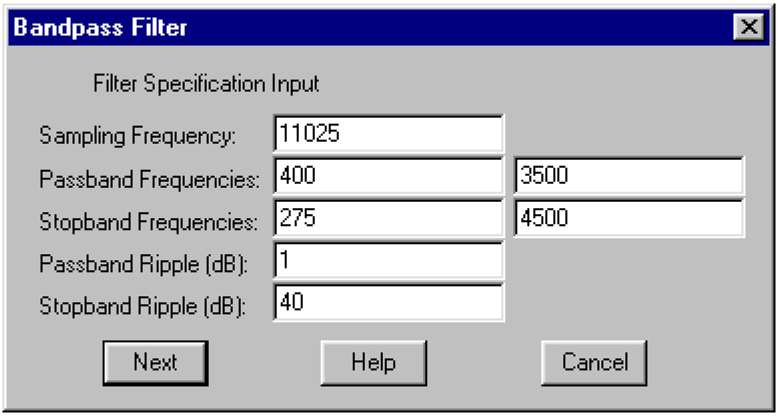


Figure 4-6: Bandpass filter specifications

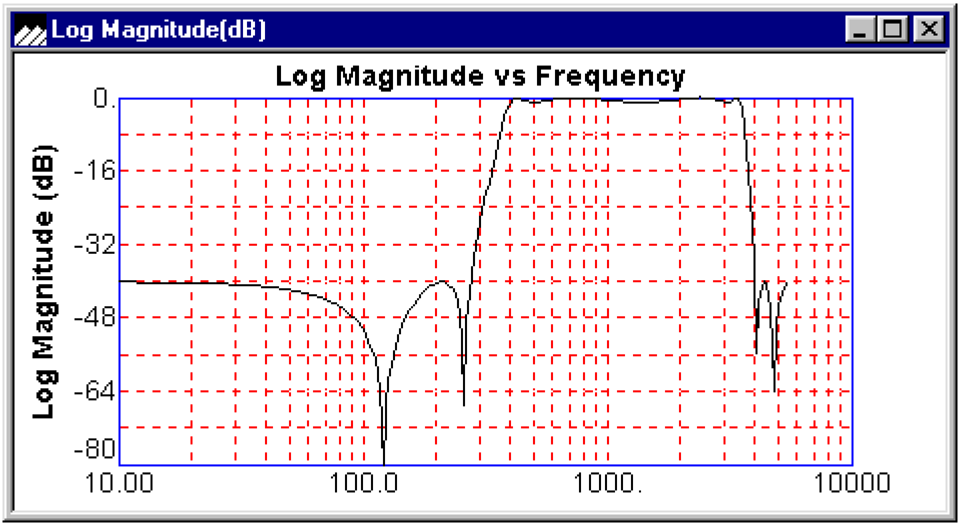


Figure 4-7: Bandpass filter log magnitude response

4.4 60 Hz notch filter, 32 000 Hz sample rate

The specifications and magnitude response of this filter are shown below. Again, an 8-pole elliptic design is used.

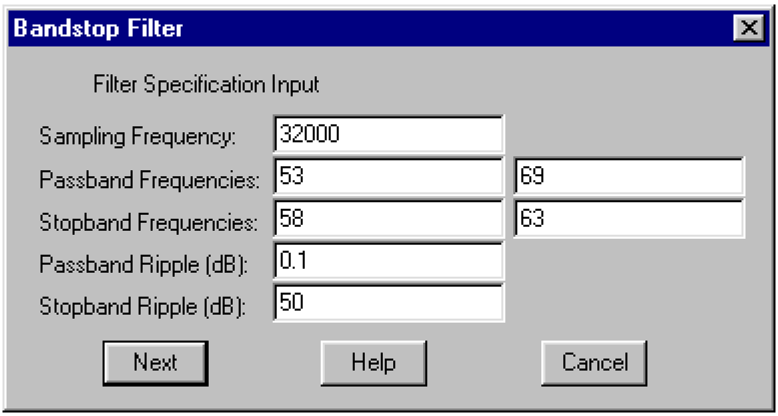


Figure 4-8: Bandstop filter specifications

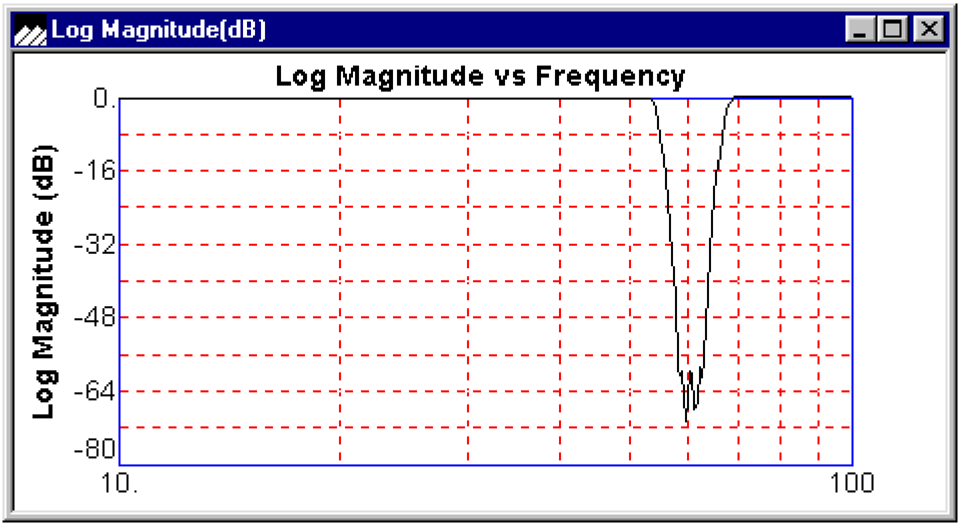


Figure 4-9: Bandstop filter log magnitude response

5 APPENDIX

5.1 Bibliography

A. Antoniou, *Digital Filters: Analysis, Design and Applications*. McGraw Hill, 1993.

P. Bernhardt, “Simplified Design of High Order Recursive Group Delay Filters”, *IEEE Trans. Acoustics, Speech, and Signal Processing*, vol. 29, no. 5, October 1980.

R. Crochiere and L. Rabiner, *Multirate Digital Signal Processing*. Prentice Hall, 1983.

D. Elliott, *Handbook of Digital Signal Processing Engineering Applications*. Academic Press, 1987.

M. El-Sharkawy, *Digital Signal Processing Applications with Motorola's DSP56002 Processor*. Prentice Hall, 1996.

F. Harris, “On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform”, *Proc. IEEE*, vol. 66, no. 1, pp. 51–83, January 1978.

E. Hogenauer, “An Economical Class of Digital Filters for Decimation and Interpolation”; *IEEE Trans. Acoustics, Speech, and Signal Processing*, vol. 29, no. 2, April 1981.

L. Jackson, *Digital Filters and Signal Processing: With Matlab Exercises*. Kluwer, 1996.

N. Jayant and P. Noll, *Digital Coding of Waveforms: Principles and Applications to Speech and Video*. Prentice Hall, 1984.

J. Kaiser, “Nonrecursive Digital Filter Design using the I_0–sinh Window Function”, *Proc. IEEE Int. Symp. Circuits and Systems*, 1984.

A. Nuttal, “Some Windows with Very Good Sidelobe Behavior”, *IEEE Trans. Acoustics, Speech, and Signal Processing*, vol. 29, no. 1, pp 84–91, February 1981.

A. Oppenheim and R. Schafer, *Digital Signal Processing*. Prentice Hall, 1975.

T. Parks and C. Burrus, *Digital Filter Design (Topics in Digital Signal Processing)*. John Wiley & Sons, 1987.

J. Proakis and D. Manolakis, *Digital Signal Processing: Principles, Algorithms and Applications*. Prentice Hall, 3rd ed., 1995.

R. Roberts and C. Mullis, *Digital Signal Processing*. Addison-Wesley, 1987.

R. Webster, "On Qualifying Windows for FIR Filter Design", *IEEE Trans. Acoustics, Speech, and Signal Processing*, vol. 31, no. 1, pp 237–240, February 1983.

5.2 Example Output

The following is an example of the output written to `SFIL.OUT` during a filter design. We use the Butterworth highpass filter from Section 4.2.

Filter specification

The specifications and analog prototype as chosen by the user.

```
FILTER TYPE:HIGH PASS                      2H
PASSBAND RIPPLE IN -dB        -.3000E+00
STOPBAND RIPPLE IN -dB        -.5500E+02
PASSBAND CUTOFF FREQUENCY   0.800000E+02 HERTZ
STOPBAND CUTOFF FREQUENCY   0.100000E+02 HERTZ
SAMPLING FREQUENCY          0.441000E+05 HERTZ
FILTER ORDER:     4
FILTER DESIGN METHOD:      Bilinear Transformation
FILTER REALIZATION METHOD: Cascaded Second Order Sections
ANALOG FILTER TYPE:        Butterworth
```

Normalized lowpass transfer function

The normalized lowpass transfer function is given by

$$H(s) = H_a \prod_{i=1}^{N} \left(\frac{B_{i0}s^2 + B_{i1}s + B_{i2}}{s^2 + A_{i1}s + A_{i2}} \right)$$

where N is the number of second-order sections. This filter has a cutoff frequency of 1 Hz.

```
NORMALIZED ANALOG TRANSFER FUNCTION T(s) - PRODUCT OF SECTIONS
NUMERATOR COEFFICIENTS                        DENOMINATOR COEFFICIENTS
S**2 TERM     S TERM        CONST TERM    S**2 TERM     S TERM
CONST TERM
0.000000e+000 0.000000e+000 1.000000e+000 1.000000e+000
1.847759e+000 1.000000e+000
0.000000e+000 0.000000e+000 1.000000e+000 1.000000e+000 7.653669e-
001 1.000000e+000
INITIAL GAIN  1.000000000e+000
```

Unnormalized highpass transfer function

The unnormalized analog transfer function is formed by frequency transformation from lowpass to highpass:

$$s \to \left(\frac{\omega_0}{s} \right)$$

where ω_0 is the prewarped cutoff frequency of the highpass filter.

```
UNNORMALIZED ANALOG TRANSFER FUNCTION T(s) - PRODUCT OF SECTIONS
NUMERATOR COEFFICIENTS                    DENOMINATOR COEFFICIENTS
S**2 TERM     S TERM        CONST TERM    S**2 TERM     S TERM
CONST TERM
1.000000e+000 0.000000e+000 0.000000e+000 1.000000e+000
6.679175e+002 1.306637e+005
1.000000e+000 0.000000e+000 0.000000e+000 1.000000e+000
2.766605e+002 1.306637e+005
INITIAL GAIN  1.000000000e+000
```

Digital transfer function

The digital transfer function is formed by applying the bilinear transformation to the unnormalized analog transfer function:

$$s \to \left(\frac{1 - z^{-1}}{1 + z^{-1}} \right)$$

$$\Rightarrow H(z) = H_d \prod_{i=1}^{N} \left(\frac{b_{i0} z^2 + b_{i1} + b_{i2}}{z^2 + a_{i1} + a_{i2}} \right) = H_d \prod_{i=1}^{N} \left(\frac{b_{i0} + b_{i1} z^{-1} + b_{i2} z^{-2}}{1 + a_{i1} z^{-1} + a_{i2} z^{-2}} \right) .$$

```
DIGITAL TRANSFER FUNCTION Hd(z) - PRODUCT OF SECTIONS
NUMERATOR COEFFICIENTS                     DENOMINATOR COEFFICIENTS
Z**2 TERM      Z TERM        CONST TERM    Z**2 TERM      Z TERM
CONST TERM
1.000000e+000 -2.000000e+000 1.000000e+000 1.000000e+000 -
1.984902e+000 9.849686e-001
1.000000e+000 -2.000000e+000 1.000000e+000 1.000000e+000 -
1.993679e+000 9.937462e-001
INITIAL GAIN  9.893476664e-001
```

Zeros and poles of *H*(*z*)

These are computed by factorizing the second-order sections.

```
ZEROS OF THIS TRANSFER FUNCTION
REAL PART         IMAGINARY PART     RADIUS            ANGLE
1.000490251e+000  0.000000000e+000   1.000490251e+000
0.000000000e+000
9.995099894e-001  0.000000000e+000   9.995099894e-001
0.000000000e+000
1.000489170e+000  0.000000000e+000   1.000489170e+000
0.000000000e+000
9.995110690e-001  0.000000000e+000   9.995110690e-001
0.000000000e+000

POLES OF THIS TRANSFER FUNCTION
REAL PART         IMAGINARY PART     RADIUS            ANGLE
9.924508333e-001  3.125265524e-003   9.924557541e-001
3.149027729e-003
9.924508333e-001 -3.125265524e-003   9.924557541e-001 -
3.149027729e-003
9.968395233e-001  7.550298527e-003   9.968681168e-001
7.574091889e-003
9.968395233e-001 -7.550298527e-003   9.968681168e-001 -
7.574091889e-003
```

Quantized transfer function, zeros, and poles of *H*(*z*)

The digital filter coefficients are quantized to the wordlength used in the Audio Precision System Two Cascade hardware. This quantization may change the position of the zeros and poles of the filter. The new transfer function and its zeros and poles are computed.

```
DIGITAL TRANSFER FUNCTION Hd(z) - PRODUCT OF SECTIONS
NUMERATOR COEFFICIENTS                    DENOMINATOR COEFFICIENTS
Z**2 TERM      Z TERM        CONST TERM    Z**2 TERM      Z TERM
CONST TERM
4.962337e-001 -9.924675e-001 4.962337e-001 5.000000e-001 -
9.924508e-001 4.924842e-001
4.984281e-001 -9.968563e-001 4.984281e-001 5.000000e-001 -
9.968395e-001 4.968730e-001
INITIAL GAIN  1.000000000e+000

ZEROS OF THIS TRANSFER FUNCTION
REAL PART         IMAGINARY PART     RADIUS            ANGLE
1.000490251e+000  0.000000000e+000   1.000490251e+000
0.000000000e+000
9.995099894e-001  0.000000000e+000   9.995099894e-001
0.000000000e+000
1.000489170e+000  0.000000000e+000   1.000489170e+000
0.000000000e+000
9.995110690e-001  0.000000000e+000   9.995110690e-001
0.000000000e+000

POLES OF THIS TRANSFER FUNCTION
REAL PART         IMAGINARY PART     RADIUS            ANGLE
9.924508333e-001  3.125265524e-003   9.924557541e-001
3.149027729e-003
9.924508333e-001 -3.125265524e-003   9.924557541e-001 -
3.149027729e-003
9.968395233e-001  7.550298527e-003   9.968681168e-001
7.574091889e-003
9.968395233e-001 -7.550298527e-003   9.968681168e-001 -
7.574091889e-003
```