LabVIEW

Digital Filter Design
Toolkit Reference Manual

Internet Support

GPIB: `gpib.support@natinst.com`
DAQ: `daq.support@natinst.com`
VXI: `vxi.support@natinst.com`
LabVIEW: `lv.support@natinst.com`
LabWindows: `lw.support@natinst.com`
HiQ: `hiq.support@natinst.com`

FTP Site: `ftp.natinst.com`
Web Address: `www.natinst.com`

Bulletin Board Support

BBS United States: (512) 794-5422 or (800) 327-3077
BBS United Kingdom: 01635 551422
BBS France: 1 48 65 15 59

FaxBack Support

(512) 418-1111 or (800) 329-7177

Telephone Support (U.S.)

Tel: (512) 795-8248
Fax: (512) 794-5678 or (800) 328-2203

International Offices

Australia 03 9 879 9422, Austria 0662 45 79 90 0, Belgium 02 757 00 20,
Canada (Ontario) 519 622 9310, Canada (Québec) 514 694 8521, Denmark 45 76 26 00,
Finland 90 527 2321, France 1 48 14 24 24, Germany 089 741 31 30, Hong Kong 2645 3186,
Italy 02 48301892, Japan 03 5472 2970, Korea 02 596 7456, Mexico 5 202 2544,
Netherlands 03480 33466, Norway 32 84 84 00, Singapore 2265886, Spain 91 640 0085,
Sweden 08 730 49 70, Switzerland 056 20 51 51, Taiwan 02 377 1200, U.K. 01635 523545

National Instruments Corporate Headquarters

6504 Bridge Point Parkway   Austin, TX 78730-5039 Tel: (512) 794-0100

Important Information

Warranty

Copyright

Trademarks

WARNING REGARDING MEDICAL AND CLINICAL USE OF NATIONAL INSTRUMENTS PRODUCTS

*Table
of
Contents*

About This Manual

Chapter 1
Overview of the Digital Filter Design Toolkit

Chapter 2
Digital Filter Design Application

Chapter 3
IIR and FIR Implementation

Chapter 4
Using Your Coefficient Designs: DFD Utilities

Appendix A
References

Appendix B
Customer Communication

Glossary

Figures

Tables

This manual describes the Digital Filter Design Toolkit package. You can use this program to design digital filters.

Organization of This Manual

The manual is organized as follows:

- Chapter 1, *Overview of the Digital Filter Design Toolkit*, lists the contents of the Digital Filter Design (DFD) Toolkit, contains installation instructions, and introduces you to digital filtering and design.

- Chapter 2, *Digital Filter Design Application,* describes the Digital Filter Design application that you use to design infinite impulse response (IIR) and finite impulse response (FIR) digital filters.

- Chapter 3, *IIR and FIR Implementation*, contains a description of the filter implementation equations for IIR and FIR filtering, and describes the format of the IIR and FIR filter coefficient files.

- Chapter 4, *Using Your Coefficient Designs: DFD Utilities*, contains a description of the DFD Utilities that you use for your LabVIEW®, LabWindows®/CVI, and Windows filtering applications.

- Appendix A, *References*, lists the reference material used to produce the Digital Filter Design Toolkit. For more information about the theories and algorithms implemented in the Digital Filter Design Toolkit, refer to the documents referenced in this section.

- Appendix B, *Customer Communication*, contains forms you can use to request help from National Instruments or to comment on our products and manuals.

- The *Glossary* contains an alphabetical list and description of terms used in this manual, including abbreviations, acronyms, metric prefixes, mnemonics, and symbols.

Conventions Used in This Manual

The following conventions are used in this manual:

bold	Bold text denotes menus, menu items, or dialog box buttons or options.
italic	Italic text denotes emphasis, a cross reference, or an introduction to a key concept.
bold italic	Bold italic text denotes a note, caution, or warning.
monospace	Lowercase text in this font denotes text or characters that are to be literally input from the keyboard, sections of code, programming examples, and syntax examples. This font is also used for the proper names of disk drives, paths, directories, programs, subprograms, subroutines, device names, functions, variables, filenames, and extensions, and for statements and comments taken from program code.

Warning: ***This icon to the left of bold italicized text denotes a warning, which alerts you to the possibility of damage to you or your equipment.***

Caution: ***This icon to the left of bold italicized text denotes a caution, which alerts you to the possibility of data loss or a system crash.***

Note: ***This icon to the left of bold italicized text denotes a note, which alerts you to important information.***

» *The » symbol leads you through ne*sted menu items and dialog box options to a final action. The sequence
File»**Page Setup**»**Options**»**Substitute Fonts**
directs you to pull down the **File** menu, select the **Page Setup** item, select **Options**, and finally select the **Substitute Fonts** option from the last dialog box.

paths Paths in this manual are denoted using backslashes (\) to separate drive names, directories, and files, as in drivename\ dir1name\dir2name\myfile

Abbreviations, acronyms, metric prefixes, mnemonics, symbols, and terms are listed in the *Glossary*.

Related Documentation

The following documents contain information that you may find helpful as you read this manual:

• *LabVIEW Analysis VI Reference Manual*

• *LabVIEW Data Acquisition VI Reference Manual*

• *LabVIEW Tutorial*

• *LabVIEW User Manual*

• *LabWindows/CVI Advanced Analysis Library Reference Manual*

• *Getting Started with LabWindows/CVI*

• *LabWindows/CVI User Manual*

Customer Communication

National Instruments wants to receive your comments on our products and manuals. We are interested in the applications you develop with our products, and we want to help if you have problems with them. To make it easy for you to contact us, this manual contains comment and configuration forms for you to complete. These forms are in Appendix B, *Customer Communication*, at the end of this manual.

Overview of the Digital Filter Design Toolkit
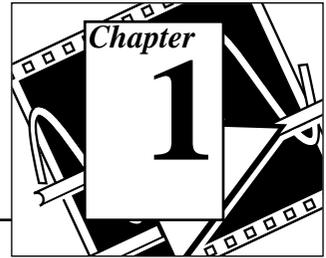
This chapter lists the contents of the Digital Filter Design (DFD) Toolkit, contains installation instructions, and introduces you to digital filtering and design.

Package Contents

Your Digital Filter Design Toolkit should contain the following materials.

- The Digital Filter Design Toolkit diskettes.

- *LabVIEW Digital Filter Design Toolkit Reference Manual*, part number 320953A-01.

Installation

The following sections contain instructions for installing the Digital Filter Design Toolkit on the Macintosh, Windows, Sun SPARCstation, and HP-UX platforms.

Macintosh and Power Macintosh

Complete the following steps to install the toolkit.

1. Insert disk 1 of the Digital Filter Design Toolkit into your 3.5 in. disk drive and double-click on the DFD Toolkit Installer icon when it appears on your desktop.

2. Follow the instructions on your screen.

Once you have completed the on-screen installation instructions, you are ready to run the Digital Filter Design Toolkit.

Windows

Complete the following steps to install the toolkit.

1. Launch Windows.

2. Insert disk 1 of the Digital Filter Design Toolkit into your 3.5 in. disk drive.

3. From the File Manager, run `setup.exe`.

4. Follow the instructions on your screen.

Once you have completed the on-screen installation instructions, you are ready to run the Digital Filter Design Toolkit.

Sun and HP-UX

Complete the following steps to install the toolkit onto your hard drive.

1. Insert the Digital Filter Design Toolkit disk into your 3.5 in. disk drive.

2. In the UNIX shell, enter the following line from a directory for which you have write permission.

3. `tar xvf /dev/rfd0a INSTALL`

4. This entry will extract the installation script file, `INSTALL`.

5. Enter the following command to start the installation process:

6. `./INSTALL`

7. Follow the instructions on your screen.

Once you have completed the on-screen installation instructions, you are ready to run the Digital Filter Design Toolkit.

Introduction to Digital Filtering

Analog filter design is one of the most important areas of electronic design. Although some analog filter design books feature simple, well tested filter designs, filter design is often reserved for specialists because it requires advanced mathematical knowledge and understanding of the processes involved in the system affecting the filter.

Modern sampling and digital signal processing tools have made it possible to replace analog filters with digital filters in applications that require flexibility and programmability. These applications include audio, telecommunications, geophysics, and medical monitoring.

Digital filters have the following advantages over their analog counterparts.

- They are software programmable.

- They are stable and predictable.

- They do not drift with temperature or humidity, or require precision components.

- They have a superior performance-to-cost ratio.

The following discussion of sampling theory is intended to give you a better understanding of the filter parameters and how they relate to the input parameters.

The sampling theorem states that you can reconstruct a continuous-time signal from discrete, equally spaced samples if the sampling frequency is at least twice that of the highest frequency in the time signal. Assume you can sample the time signal of interest at Δt equally spaced intervals without losing information. The Δt parameter is the sampling interval.

You can obtain the sampling rate or sampling frequency f_s from the sampling interval

$$f_s = \frac{1}{\Delta t} \ ,$$

which means that, according to the sampling theorem, the highest frequency that the digital system can process is

$$f_{Nyq} = \frac{f_s}{2} \ .$$

The highest frequency a digital system can process is known as the Nyquist frequency. This also applies to digital filters. For example, if your sampling interval is

$\Delta t = 0.001$ sec,

then the sampling frequency is

$f_s = 1000$ Hz,

and the highest frequency that the system can process is

$f_{Nyq} = 500$ Hz.

All frequency response designs in the DFD applications are therefore limited to half the sampling rate.

IIR and FIR Filters

The digital filters designed using the DFD applications are either Infinite Impulse Response (IIR) or Finite Impulse Response (FIR) filters. IIR filters process both input and output samples. This allows IIR filters to have very sharp transitions in their frequency response characteristics, which is very useful in many applications. The nonlinear phase distortion associated with IIR filters sometimes can limit their selection in many phase sensitive systems.

FIR filters, however, process only input samples and can be designed to have only linear phase distortion. The linear phase distortion can be compensated by simple delays in the system. FIR filters may require more coefficients than IIR filters to meet similar design specifications.

Digital Filter Design Application
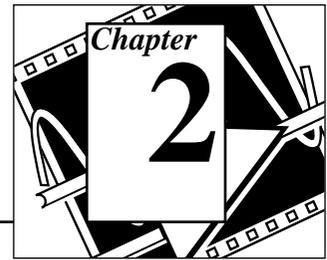
This chapter describes the Digital Filter Design application you use to design infinite impulse response (IIR) and finite impulse response (FIR) digital filters.

Introduction

The Digital Filter Design (DFD) application is a complete filter design and analysis tool you can use to design digital filters to meet your precise filter specifications. You can graphically design your IIR and FIR filters, interactively review filter responses, save your filter design work, and load your design work from previous sessions.

You can save digital filter coefficients for later implementation from within LabVIEW and LabWindows CVI. Also, you can call Windows DFD Dynamic Link Libraries (DLLs) from other applications, or other applications can load the filter coefficient files directly. This manual includes all required filter coefficient forms and implementation equations.

If you have a National Instruments data acquisition (DAQ) device, you can perform real-world filter testing from within the DFD application. You can view the time waveforms or the spectra of both the input signal and the filtered output signal while simultaneously redesigning your digital filters. Figure 2-1 shows a conceptual overview of the Digital Filter Design Toolkit and related applications.
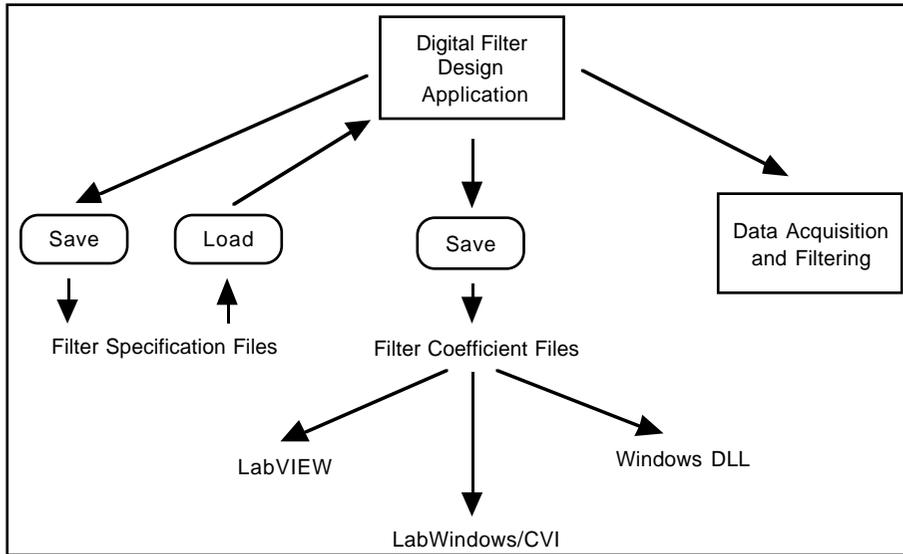
Figure 2-1. Conceptual Overview of the Digital Filter Design Toolkit

Operation: the Main Menu

When you launch the DFD application, a panel displays the mainavailableoptions. Figure 2-2 shows the Filter Design Main Menu panel.
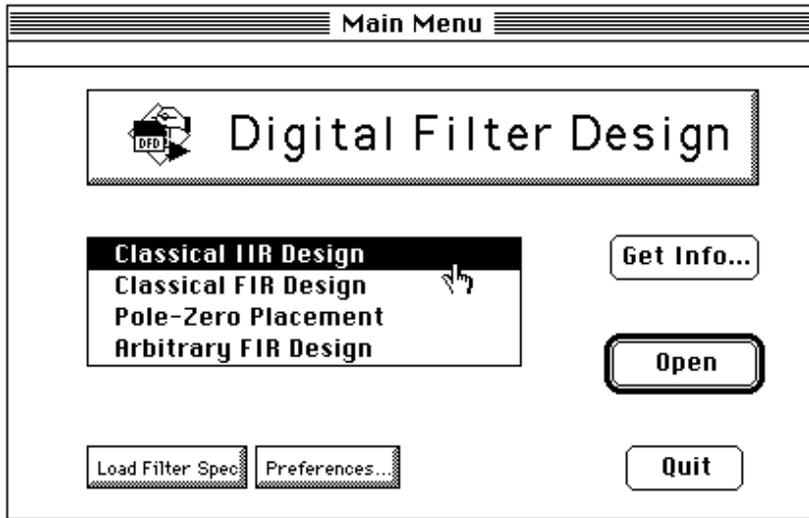
Figure 2-2. Filter Design Main Menu Panel

Opening the Filter Design Panels

From the Filter Design Main Menu Panel, you can open any of the four digital filter design panels: **Classical IIR Design**, **Classical FIR Design**, **Pole-Zero Placement**, and **Arbitrary FIR Design**. The next section of this manual discusses each design panel.

Directly Loading a Filter Specification File

You can also load a previously designed filter specification file directly from the **Main** menu. If you choose **Main**»**Load Filter Specification**, the DFD application prompts you to select the filter specification file that you saved during previous design work. After you select the file, the application asks you if you want to open the appropriate design panel for that specification file. This way, you can easily resume work on an ongoing design project.

Editing the DFD Preferences

To customize your DFD application preferences, choose **Main»Preferences...**. Then you can edit your DFD application preferences for future design sessions.

Quitting the DFD Application

Choose **Main**»**QUIT** to quit the DFD application.

Digital Filter Design Panels

If you double-click on one of the four design selections in the **Main** menu, the DFD application loads and runs the selected design panel. You can use these design panels to design IIR or FIR filters, save your design work and filter coefficients, or load previous filter designs.

After designing your filter, you can move from the design panels to the Analysis of Filter Design panel to view various frequency domain and time domain filter responses. Then you can save these responses to text files for use in other applications. You can also perform real-world testing of your filter designs by moving to the DAQ and Filter panel, which performs data acquisition and filtering in parallel with your filter designing.

Common Controls and Features

Using the DFD Menu

All four filter design panels, the Analysis of Filter Design panel, and the DAQ and Filter panel have a DFD Menu from which you can select a number of options. Figure 2-3 shows the DFD Menu popup for theClassical IIR Design panel. This section discusses each DFD Menu option.

Figure 2-3. DFD Menu Popup

Saving Filter Specifications

To save all your specifications for the present filter design panel, select
DFD»**Save Spec...**. The DFD application prompts you for the name of
the filter specification file to save. You should name your spec files
appropriately for a given filter design. For example, if you design a
lowpass IIR filter, a good name is lowpass.iir, or lowp1 if this
design is the first of many lowpass IIR designs. Table 2-1 lists
suggested filename extensions for the four filter design panels. These
names are simply suggested extensions and have no effect on how
theDFD application interprets the file contents.

Table 2-1.   Suggested Specification Filename Extensions

Design Panel	Filename
Classical IIR Design	*filename*.iir
Classical IIR Design	*filename*.fir
Pole-Zero Placement	*filename*.pz
Arbitrary FIR Design	*filename*.arb

Loading Filter Specifications

To load a filter specification file into the present filter design panel, select **DFD**»**Load Spec....** the DFD application prompts you for the location of the filter specification file to load. If the selected spec file is the same type design as the present design panel, the DFD application loads the specification from the selected file into the present design panel for viewing, editing, or analyzing.

If you designed the selected spec file in a different design panel than the present panel, the DFD application prompts you to open the appropriate design panel for that specification file. For example, if you are using the Pole-Zero Placement panel and you load a specification file saved from within the Classical FIR Design panel, the DFD application prompts you to open the Classical FIR Design panel to resume work on the loaded filter specifications.

Saving Filter Coefficients

To save all your filter coefficients to a file, select **DFD**»**Save Coeff....**. The DFD application first prompts you for the format of the coefficient file. You can select **text format** or **log format**. Use text format to view or print the coefficient file, or to use the coefficients in other non-LabVIEW filtering applications. Use log format for LabVIEW-only filtering applications, although LabVIEW filtering utilities can read both text and log formatted coefficient files.

After you select the format of the coefficient file, the DFD application prompts you for the name of the filter coefficient file to save. You should name your coefficient files appropriately for a given filter design. For example, if you save bandpass IIR filter coefficients, a

good name is "bpiir.txt" or "bpiir.log," depending on the coefficient file type.

Analyzing Filter Designs

To begin analyzing your filter design, choose **DFD**»**Analysis**. The DFD application loads and runs the Analysis of Filter Design panel. From this analysis panel, you can view the filter magnitude response, phase response, impulse response, step response, and pole-zero plot. You can also view and print full-screen plots of each response. From the full-screen views, you can save the analysis results to a text file.

DAQ and Filter: Real-World Testing

If you have a National Instruments DAQ device and you want to see how the present filter design performs on real-world signals, choose **DFD**»**DAQ and Filter**. The DFD application loads and runs the DAQ and Filter panel. From this panel, you can set up your DAQ device and then acquire real signals. The acquired data then passes through the currently designed filter, and the DFD application plots the input and output waveforms and spectrums.

Transferring Filter Designs

You can transfer some filter design specifications from one design panel to another. For example, you can set up your passband and stopband requirements in designing an FIR filter, and see what kind of IIR filter meets your design specifications. Not all design panels can share specifications. Table 2-2, Filter Specification Transfers, shows the allowable transfers and their corresponding **DFD** menu options.

Table 2-2.   Filter Specification Transfers

Design Transfer	DFD Menu Option
Filter specs from the Classical IIR to Classical FIR	Xfer Classical FIR
Filter specs from the Classical FIR to Classical IIR	Xfer Classical IIR
Poles and zeros from Classical IIR to Pole-Zero Placement	Xfer Pole Zero

Returning to the Main Menu

To return to the DFD Main menu, choose **DFD Menu**»**Main Menu** in each panel.

Panning and Zooming Options

The graph palette is included with any graph you drop onto the front panel. This palette has controls for panning (scrolling the display area of a graph) and for zooming in and out of sections of the graph. The graph palette is included with many DFD graphs. A graph with its accompanying graph palette is shown in the following illustration.
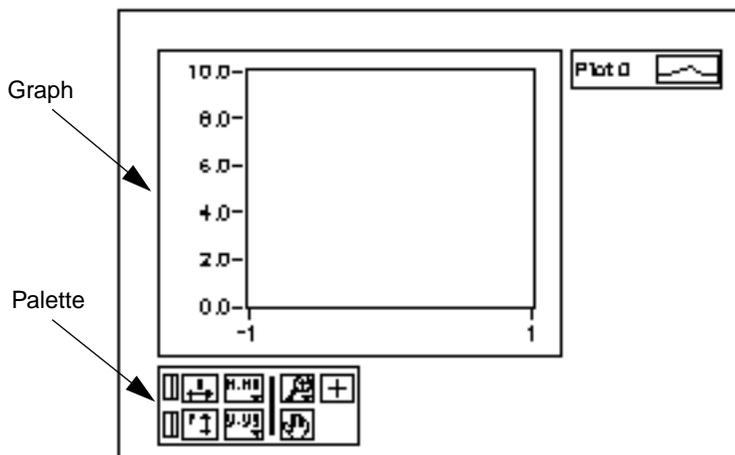
If you press the x autoscale button, shown at the left, the DFD application autoscales the X data of the graph. If you press the y autoscale button, shown at the left, the DFD application autoscales the Y data of the graph. If you want the graph to autoscale either of the scales continuously, click on the lock switch, shown at the left, to lock autoscaling on.

The scale format buttons, shown left, give you run-time control over the format of the X and Y scale markers respectively.

You use the remaining three buttons to control the operation mode for the graph.

Normally, you are in standard operate mode, indicated by the plus or crosshatch. In operate mode, you can click in the graph to move cursors around.

If you press the panning tool, shown to the left, you switch to a mode in which you can scroll the visible data by clicking and dragging sections of the graph.

If you press the zoom tool, shown at the left, you can zoom in on a section of the graph by dragging a selection rectangle around that section. If you click on the zoom tool, you get a pop-up menu you can use to choose some other methods of zooming. This menu is shown in the following illustration.
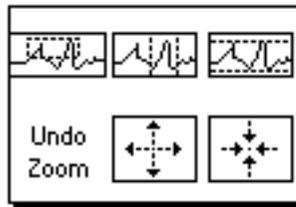
A description of each of these options follows.
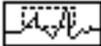
Zoom by rectangle.

Zoom by rectangle, with zooming restricted to x data (the y scale remains unchanged.

Zoom by rectangle, with zooming restricted to y data (the z scale remains unchanged).

Undo last zoom. Resets the graph to its previous setting.

Zoom in about a point. If you hold down the mouse on a specific point, the graph continuously zooms in until you release the mouse button.

Zoom out about a point. If you hold down the mouse on a specific point, the graph continuously zooms out until you release the mouse button.

☞ *Note:*   ***For the last two modes, you can zoom in and zoom out about a point . Shift-clicking zooms in the other direction.***

Graph Cursors

Following are illustrations of a waveform graph showing two cursors and the cursor movement control.
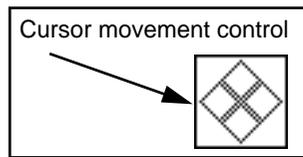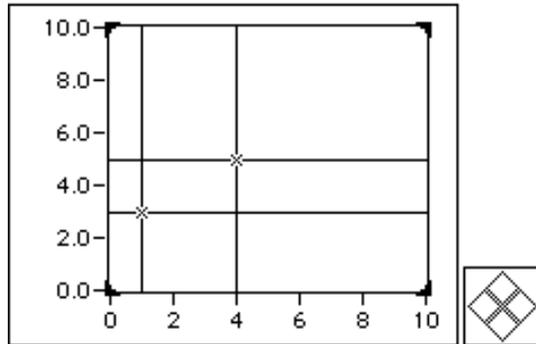
You can move a cursor on a graph or chart by dragging it with the Operating tool, or by using the cursor movement control. Clicking the arrows on the cursor movement control causes all cursors selected to move in the specified direction. You select cursors by moving them on the graph with the Operating tool.

Classical IIR Filter Design

Figure 2-4 shows the Classical IIR Filter Design panel. The panel includes a graphical interface with the Magnitude vs. Frequency cursors and plot on the left side and a text-based interface with digital controls on the right side.
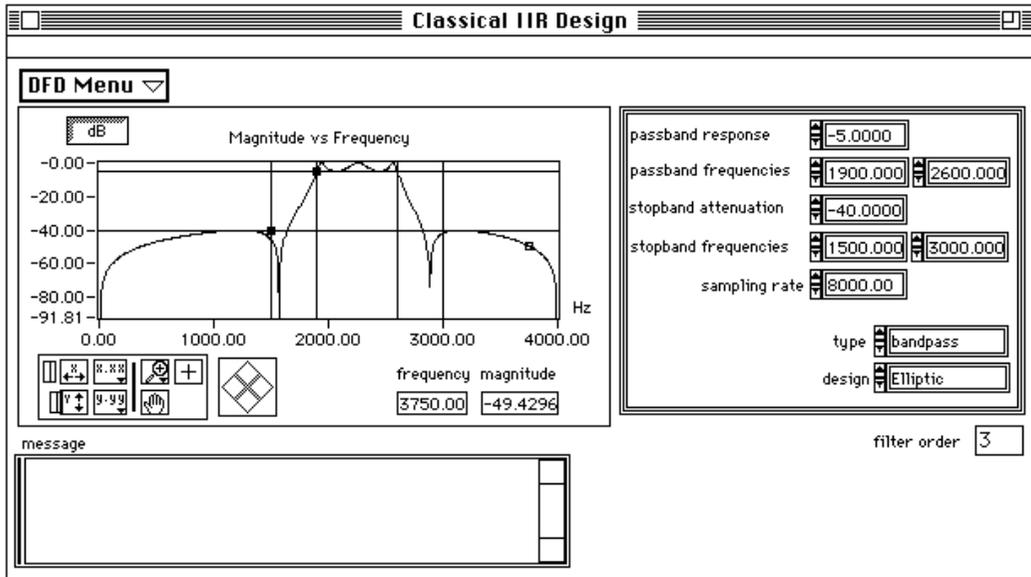
Figure 2-4. Classical IIR Filter Design Panel

Use this panel to design classical IIR digital filters. These filters include the classic types (lowpass, highpass, bandpass, and bandstop) and the classic designs (Butterworth, Chebyshev, Inverse Chebyshev, and Elliptic).

To design classical IIR filters, adjust the filter specifications. The passband and stopband requirements define a filter specification. You can fully define these requirements by using either text entry or the cursors in the Magnitude vs. Frequency graph. As you use the mouse to click and drag the cursors, the text entries update. Likewise, as you enter new specifications in the text entries, the cursors update.

The lower passband frequency fp1, and upper passband frequency fp2, and the passband response Gp, define the passband specification. For the bandpass filter, the passband ranges from fp1 to fp2. The passband is the region in the frequency domain with a response near 1.0. Gp is the minimum allowable passband gain or filter magnitude response. In Figure 2-4, the passband is specified as having a minimum gain of -5 dB between the frequencies of fp1 = 1900 Hz and fp2 = 2600 Hz.

The passband is defined as:

lowpass ($0 \leq f \leq fp1$)

highpass ($fp1 \leq f \leq fsamp/2$)

bandpass ($fp1 \leq f \leq fp2$)

bandstop ($0 \leq f \leq fp1$, $fp2 \leq f \leq fsamp/2$)

where:

fp1 = passband frequency 1

fp2 = passband frequency 2

fsamp = sampling rate

The lower stopband frequency, fs1 and the upper stopband frequency fs2, and the stopband attenuation, Gs, define the stopband specification. For the bandpass filter, the stopband ranges from 0.0 (DC) to the lower stopband frequency, fs1, and from the upper stopband frequency, fs2, to half the sampling rate (Nyquist). The stopband is that region in the frequency domain with a response near 0.0. Gs is the minimum acceptable stopband attenuation or filter magnitude response. In Figure 2-4, Classical IIR Filter Design Panel, the stopband specification is a minimum attenuation of -40 dB between the frequencies of 0 and
fs1 = 1500 Hz and between the frequencies of
fs2 = 2600 Hz and 4000 Hz.

The stopband is defined as:

lowpass ($fs1 \leq f \leq fsamp/2$)

highpass ($0 \leq f \leq fs1$)

bandpass ($0 \leq f \leq fs1$, $fs2 \leq f \leq fsamp/2$)

bandstop ($fs1 \leq f \leq fs2$)

where:

fs1 = stopband frequency 1

fs2 = stopband frequency 2

fsamp = sampling rate

In this panel, the DFD application estimates the minimum filter order that the selected type and design require to meet or exceed the modified filter specifications. The DFD application computes other appropriate filter parameters and designs and plots the IIR filter automatically. You

see immediate graphical feedback to help you determine whether the filter meets your specifications.

Classical IIR Design Panel Controls and Displays

DFD Menu ▽

Use the design panel DFD menu to save your filter specifications and coefficients, load filter designs from previous work, open the Analysis or the DAQ and Filter panels, transfer the IIR design specifications to the FIR design panel, transfer the poles and zeros to the Pole-Zero placement panel, or return to the Filter Design Main Menu.

The graph in Figure 2-5, Magnitude vs. Frequency, plots the frequency response H(f) magnitude of the designed digital filter.
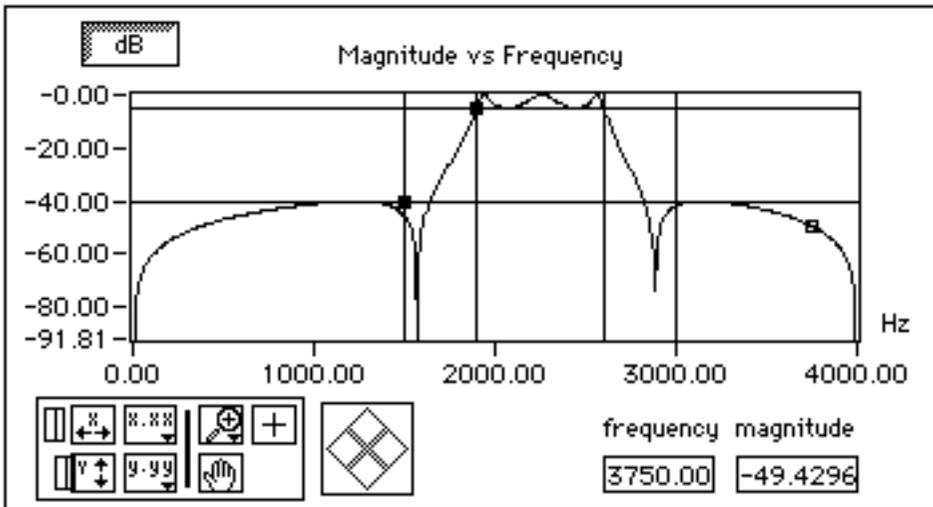
Figure 2-5. Magnitude vs. Frequency

The y-axis is in linear or decibel units, depending on how you set the button in the upper left corner of the graph.

The x-axis is in units of hertz. The full scale ranges from 0.0 to Nyquist (sampling rate/2).

By moving the blue cursor lines or crosshairs, you control the passband response (horizontal lines) and the passband frequencies (vertical lines).

By moving the red cursor lines, you control the stopband attenuation (horizontal lines) and the stopband frequencies (vertical lines).

These cursors represent the filter design specifications for the selected classical IIR filter. In the passband, the filter has a gain of ≥ specified passband response. In the stopband, the filter has a gain of ≤ specified stopband attenuation.

Use the linear/dB button to control the display units (linear or dB) of all magnitude and gain controls and displays. These controls and displays include Magnitude vs. Frequency plot (y-axis), passband response, stopband attenuation, and tracking cursor magnitude.

The frequency and magnitude indicators display the location of the tracking transparent square cursor.This cursor is locked to the frequency response H(f), so moving this cursor updates the frequency and magnitude digital displays with data points from H(f), as shown below.

The complete filter specifications can be entered using the text entry portion at the upper right-hand side of the design panel.

The passband response is the minimum gain in the passband. The horizontal blue cursor line represents this response in the Magnitude vs. Frequency plot.

In the passband, the filter gain is guaranteed to be at least as high as the specified passband response, Gp:

$|H(f)| \geq Gp.$

The first passband frequency defines one frequency edge of the passband. The vertical blue cursor lines represent this frequency in the Magnitude vs. Frequency plot.

The second passband frequency defines the second frequency edge of the passband. The vertical blue cursor lines represent this frequency in the Magnitude vs. Frequency plot.

The stopband attenuation is the minimum attenuation in the stopband. The horizontal red cursor line represents this attenuation in the Magnitude vs. Frequency plot.

In the stopband, the filter gain is guaranteed to be no higher than the specified stopband attenuation, Gs:

$|H(f)| \leq Gs.$

The first stopband frequency defines one frequency edge of the stopband. The vertical red cursor lines represent this frequency in the Magnitude vs. Frequency plot.

The second stopband frequency defines the second frequency edge of the stopband. The vertical red cursor lines represent this frequency in the Magnitude vs. Frequency plot.

The sampling rate control specifies the sampling rate in samples per second (hertz).

The type control specifies one of four classical filter types according to the following values:

0:lowpass
1:highpass
2:bandpass
3:bandstop

The design control specifies one of four classical filter design algorithms according to the following values:

0: Butterworth
1: Chebyshev
2: Inverse Chebyshev
3: Elliptic

The filter order indicator, shown below, displays the estimated filter order of the classical IIR filter. The DFD application automatically estimates the filter order as the lowest possible order that meets or exceeds the desired filter specifications.

filter order   3
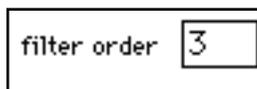
The message window displays errors that occur during the IIR design procedure. Usually these errors reflect filter specifications which are inconsistent with the chosen filter type.

message
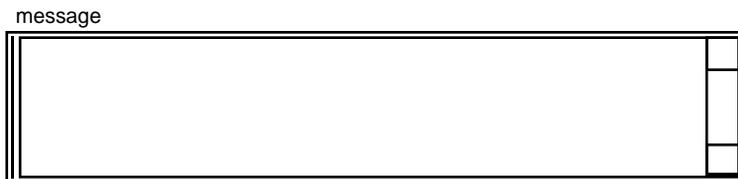
Classical FIR Design

Figure 2-6 shows the Classical FIR Design panel. This panel is very similar to the Classical IIR Design panel and operates in much the same way. The panel includes a graphical interface with the Magnitude vs. Frequency cursors and plot on the left side, and a text-based interface with digital controls on the right side.
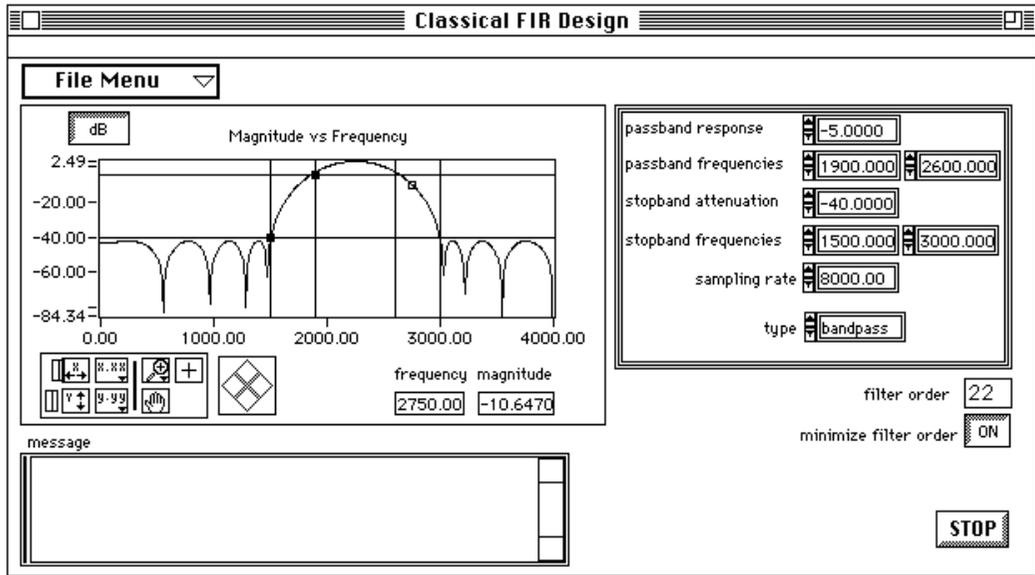
Figure 2-6. Classical FIR Design Panel

Use the Classical FIR Design panel to design classical FIR digital
filters. These filters include the classic types (lowpass, highpass,
bandpass, and bandstop) and use the Parks-McClellan equiripple FIR
filter design algorithm.

To design classical FIR filters, adjust the desired filter specifications.
The desired passband and stopband requirements define a filter
specification. You can fully define these requirements by using either
text entry or the cursors in the Magnitude vs. Frequency graph. As you
move the cursors, the text entries update. Likewise, as you enter new
specifications into the text entries, the cursors update.

The lower passband frequency, fp1 and upper passband frequency fp2,
and the passband response, Gp, define the passband specification. For
the bandpass filter, the passband ranges from fp1 to fp2. The passband
is the region in the frequency domain with a response near 1.0. Gp is
the minimum acceptable passband gain or filter magnitude response. In
Figure 2-6, the passband specification is a minimum gain of -5 dB
between the frequencies of fp1 = 1900 Hz and fp2 = 2600 Hz.

The passband is defined as:

lowpass ($0 \leq f \leq fp1$)
highpass ($fp1 \leq f \leq fsamp/2$)
bandpass ($fp1 \leq f \leq fp2$)
bandstop ($0 \leq f \leq fp1$, $fp2 \leq f \leq fsamp/2$)

where:

fp1 = passband frequency 1
fp2 = passband frequency 2
fsamp = sampling rate

The stopband frequencies, fs1 and fs2, and the stopband attenuation, Gs, define the stopband specification. For the bandpass filter, the stopband ranges from 0.0 (DC) to the lower stopband frequency, fs1, and from the upper stopband frequency, fs2, to half the sampling rate (Nyquist). The stopband is the region in the frequency domain with a response near 0.0. Gs is the minimum acceptable stopband attenuation or filter magnitude response. In Figure 2-6, the stopband specification is a minimum attenuation of -40 dB between the frequencies of 0 and fs1 = 1500 Hz and between the frequencies of fs2 = 2600 Hz and 4000 Hz.

The stopband is defined as:

lowpass ($fs1 \leq f \leq fsamp/2$)
highpass ($0 \leq f \leq fs1$)
bandpass ($0 \leq f \leq fs1$, $fs2 \leq f \leq fsamp/2$)
bandstop ($fs1 \leq f \leq fs2$)

where:

fs1 = stopband frequency 1
fs2 = stopband frequency 2
fsamp = sampling rate

The Classical FIR Design panel estimates the minimal filter order that the selected type and design require to meet or exceed the modified filter specifications. The DFD application automatically computes other appropriate filter parameters and designs and plots the FIR filter. You see immediate graphical feedback to help you determine whether the filter meets your specifications.

Classical FIR Design Panel Controls and Displays

These controls are virtually identical to those in the Classical IIR Design panel, with a few additions (a minimize filter order button and an error message display box).

DFD Menu ▽

Use the design panel DFD Menu to save your filter specifications and coefficients, load filter designs from previous work, open the Analysis or the DAQ and Filter panels, transfer the FIR design specifications to the IIR design panel, or return to the Filter Design Main Menu.

The graph in Figure 2-7 plots the frequency response H(f) magnitude of the designed digital filter.
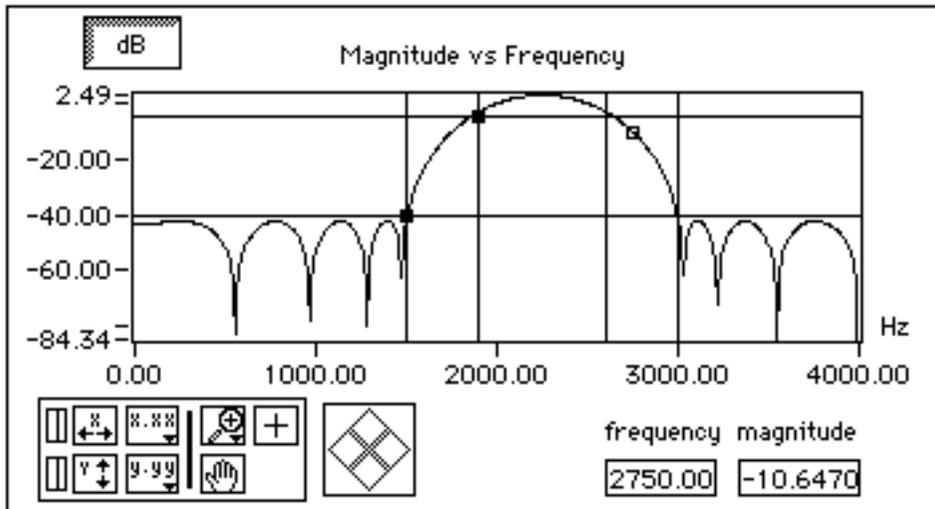
Figure 2-7. Frequency Response Magnitude

The y-axis is in linear or decibel units, depending on how you set the button in the upper left corner of this graph.

The x-axis is in hertz. The full scale ranges from 0.0 to Nyquist (sampling rate/2).

The blue cursor lines control the passband response (gain in the passband and horizontal lines) and the passband frequencies (vertical lines).

The red cursor lines control the stopband attenuation (gain in the stopband and horizontal lines) and the stopband frequencies (vertical lines).

These cursors represent the filter design specifications for the selected classical IIR filter. In the passband, the filter has a gain of ≥ specified passband response. In the stopband, the filter has a gain of ≤ specified stopband attenuation.

The dB button controls the display units (linear or decibel) of all magnitude and gain controls and displays. These controls and displays include Magnitude vs. Frequency plot (y-axis), passband response, stopband attenuation, and tracking cursor magnitude.

The frequency and magnitude indicators display the location of the tracking transparent square cursor. This cursor is locked to the frequency response H(f), so moving this cursor updates the frequency and magnitude digital displays with data points from H(f), as shown below.

The complete filter specifications can be entered using the text entry portion at the upper right-hand side of the design panel.

The passband response is the minimum gain in the passband. The horizontal blue cursor line represents this response in the Magnitude vs. Frequency plot.

In the passband, the filter gain is guaranteed to be at least as high as the specified passband response, Gp:

$$|\text{H(f)}| \geq \text{Gp}.$$

The first passband frequency defines one frequency edge of the passband. The vertical blue cursor lines represent this frequency in the Magnitude vs. Frequency plot.

The second passband frequency defines the second frequency edge of the passband. The vertical blue cursor lines represent this frequency in the Magnitude vs. Frequency plot.

The stopband attenuation is the minimum attenuation in the stopband. The horizontal red cursor line represents this attenuation in the Magnitude vs. Frequency plot.

In the stopband, the filter gain is guaranteed to be no higher than the specified stopband attenuation, Gs:

$$|\text{H(f)}| \leq \text{Gs}.$$

The first stopband frequency defines one frequency edge of the stopband. The vertical red cursor lines represents this frequency in the Magnitude vs. Frequency plot.

The second stopband frequency defines the second frequency edge of the stopband. The vertical red cursor lines represent this frequency in the Magnitude vs. Frequency plot.

The sampling rate control specifies the sampling rate in samples per second (hertz).

The type control specifies one of four classical filter types according to the following values:

    0: lowpass
    1: highpass
    2: bandpass
    3: bandstop

The filter order indicator displays the estimated filter order of the classical FIR filter.
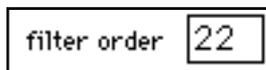
filter order 22

The DFD application automatically estimates the filter order as the lowest possible order that meets or exceeds the desired filter specifications.

The minimize filter order button controls whether the DFD application minimizes the estimated filter order. If this button is off, the DFD application uses a fast formula to estimate the filter order to meet or exceed the desired filter specifications. If this button is on, the DFD application iteratively adjusts the filter order until it finds the minimum order that meets or exceeds the filter specifications.

minimize filter order ON
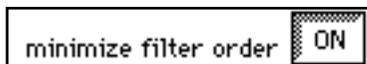
The message window displays errors that occur during the FIR design procedure. These errors occur when the filter specifications are inconsistent with the chosen filter type.

message
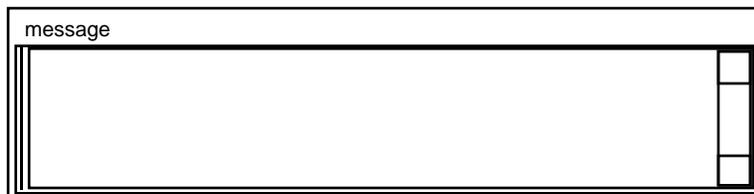
Pole-Zero Placement Filter Design

Figure 2-8 shows the Pole-Zero Placement filter design panel. The panel includes a graphical interface with the z-plane pole and zero cursors on the left side and a text-based interface with digital controls on the right side.
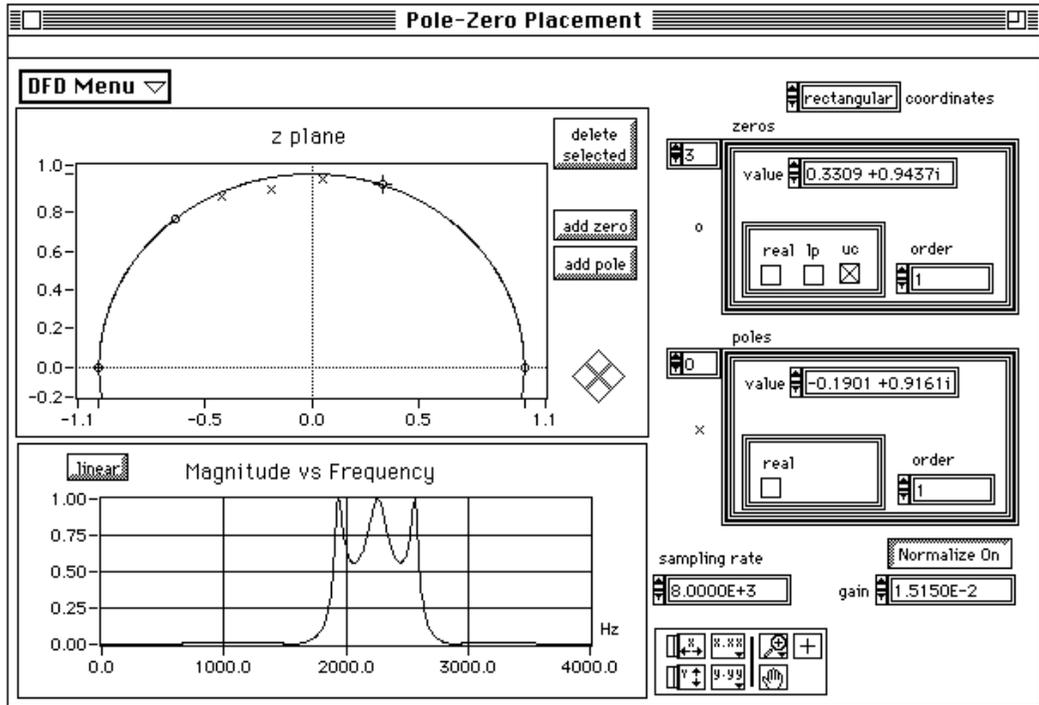
Figure 2-8. Pole-Zero Placement Filter Design Panel

Use the Pole-Zero Placement Filter Design panel to design IIR digital filters by manipulating the filter poles and zeros in the z plane. The poles and zeros initially may have come from classical IIR designs. Use this panel to move existing poles and zeros directly on the z plane plot. You can add and delete poles and zeros and accurately control their important characteristics.

You can fully describe the poles and zeros by using either the text entry or the cursors in the z-plane plot. As you change the cursors, the text entries update automatically. Likewise, as you modify the text entries, the pole/zero cursors update automatically.

The pole and zero locations in the z-plane, the characteristics of each pole and zero, the gain, and the sampling rate fully describe pole-zero filter designs. Any change in these parameters corresponds to a change in the filter coefficients. The DFD application matches the poles and

zeros and creates stable second-order stages for IIR filter coefficients. The DFD application then uses these coefficients to compute the filter magnitude response. The Magnitude vs. Frequency plot updates automatically whenever you change the poles or zeros, for immediate graphical feedback to your pole-zero filter designs.

Pole-Zero Placement Panel Controls and Displays

DFD Menu ▽

Use the design panel DFD menu to save your filter specifications and coefficients, load filter designs from previous work, open the Analysis or DAQ and Filter panels, or return to the **Filter Design Main** menu.

Figure 2-9 shows the z-plane plot of the filter poles and zeros. You can move each pole (red "x") anywhere within the unit circle, along and above the x-axis. You can move each zero (blue "o") anywhere along and above the x-axis.
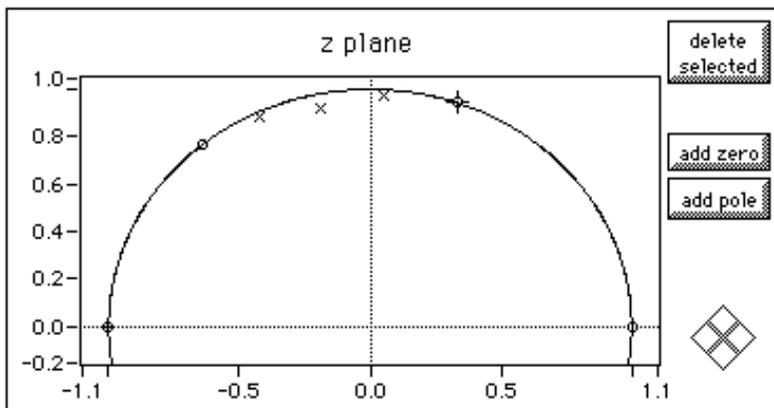
Figure 2-9. Z-Plane plot of Filter Poles and Zeros
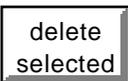
delete
selected

Click on the delete selected button to delete the selected pole or zero. Click on poles and zeros to select them.
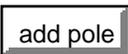
add pole

Click on the add pole button to add a pole to the z-plane. The new pole is located at the origin.
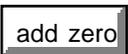
add zero

Click on the add zero button to add a zero to the z-plane. The new zero is located at the origin.

The coordinates control specifies how the DFD application displays the poles and zeros, either in rectangular or polar coordinates.
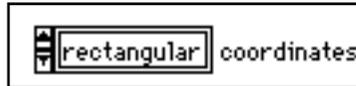
The following figure shows the array of zeros in rectangular coordinates. The complex value of each zero represents its rectangular position on the z-plane. The integer 3 in the left box is the displayed zero index. By changing this index value, you can display a particular zero in the array of zeros. When you select a particular zero in the z-plane plot, the DFD application sets the index value of the array to the selected zero.
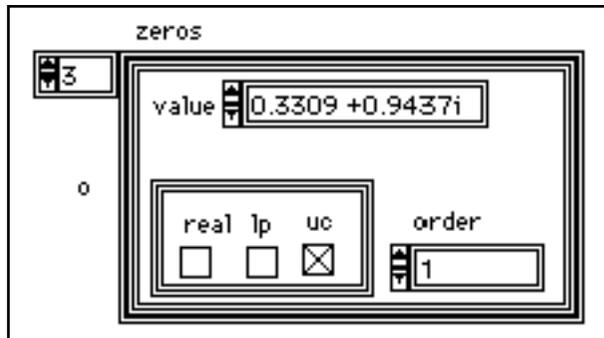
If you click in the real check box, the zero becomes purely real and is limited to real-axis movement.

If you click in the lp check box, the zero has linear phase. That is, if the zero is not real or on the unit circle, the DFD application matches it with another zero at a radius of $1/r$, where r is the radius of the original zero (the radius is the distance from the origin). Linear phase zeros are important in linear phase FIR filters. If your z-plane plot contains only zeros, and all the zeros have linear phase, then the FIR filter you designed has an overall linear phase response.

If you click on the uc check box, the zero is forced to be located on the unit circle (radius of 1.0) and is limited to movement along the unit circle.

The order text entry is the order of the zero, or the number of actual zeros at this location in the z plane.

An Mth-order zero at z = b has a z-transform of

```
H(z)  =  (z-b)^M
```

The following figure shows the array of poles in rectangular coordinates. The complex value of each pole represents its rectangular position on the z-plane. The integer 0 in the left box is the displayed pole index. By changing this index value, you can specify which pole in the array of poles displays. When you select a particular pole in the z-plane plot, the DFD application sets the index value of the array to the selected pole.
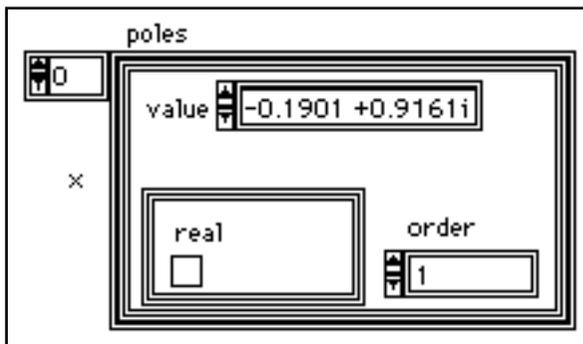
Only one special characteristic applies to poles—whether they are real. If you click in the real check box, the pole becomes purely real and is limited to real-axis movement.

The order text entry specifies the pole order, or the number of actual poles at this location in the z plane.

An Mth-order pole at z = a has a z-transform of

```
H(z)  =  (z-a)^{-M}
```

If you change the coordinates to polar coordinates, the DFD application displays the poles and zeros in polar coordinates.
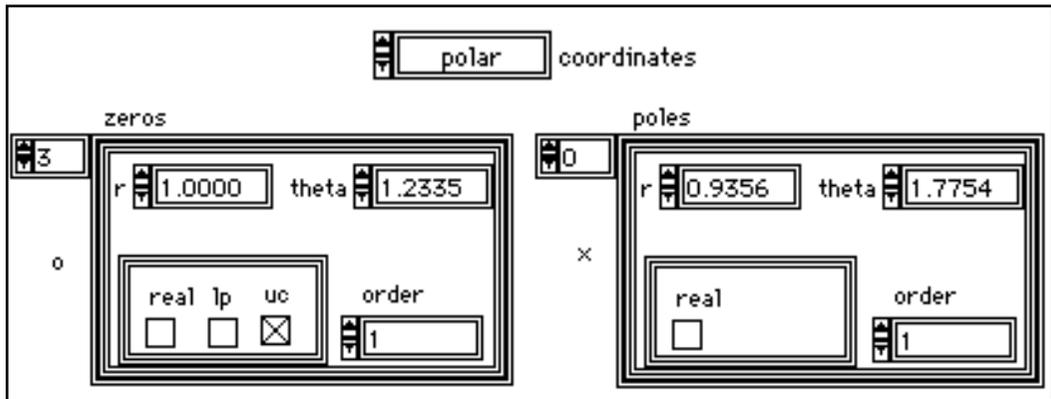
The following graph plots the frequency response H(f) magnitude of the designed digital filter.
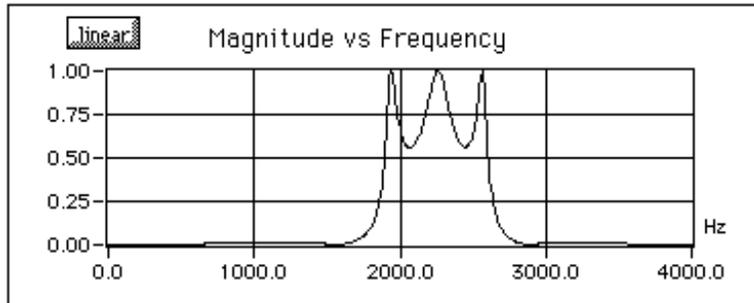
The y-axis is in linear or decibel units, depending on how you set the button in the upper left corner of this graph.

The x-axis is in hertz. The full scale ranges from 0.0 to Nyquist (sampling rate/2).

The sampling rate control specifies the sampling rate in samples per second (hertz).

```
┌──────────────────┐
│  sampling  rate  │
│ ▲┌──────────────┐│
│ ▼│8.0000E+3     ││
│  └──────────────┘│
└──────────────────┘
```
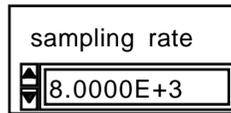
The gain control specifies the gain constant for the designed filter. Increasing this gain increases the overall gain of the designed filter.

```
┌─────────────────────────┐
│       ┌───────────────┐ │
│       │ Normalize On  │ │
│       └───────────────┘ │
│      ▲┌───────────────┐ │
│ gain ▼│1.5150E-2      │ │
│       └───────────────┘ │
└─────────────────────────┘
```

Setting the normalize button to Normalize On adjusts the filter gain so that the maximum response is 1.0 (0 dB). If you set this button set to "Normalize On," you cannot manually adjust the gain control.

Arbitrary FIR Design

Figure 2-10 shows the Arbitrary FIR Design panel. The panel includes a graphical interface with the Magnitude vs. Frequency cursors on the left side and a text-based interface with digital controls on the right side.
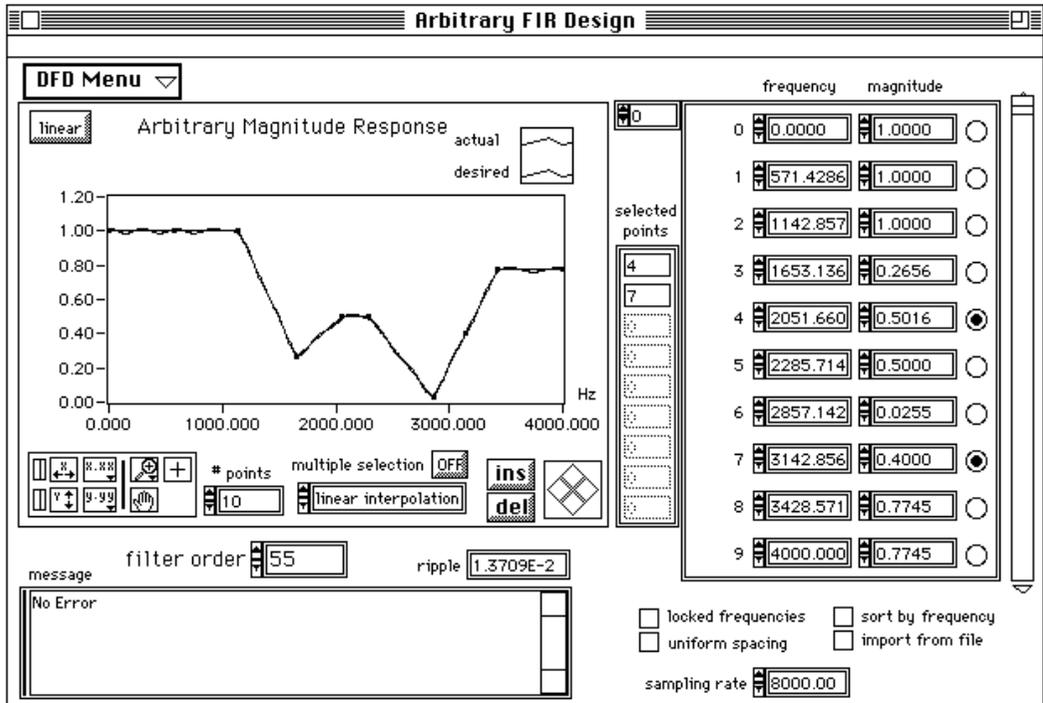
Figure 2-10. Arbitrary FIR Design Panel

Use the Arbitrary FIR Design panel to design arbitrary-magnitude FIR digital filters. Enter or modify the array magnitude response points (frequency and magnitude). From these points, the DFD application forms a desired magnitude response that covers the entire frequency range from 0.0 to half the sampling rate. the DFD application then takes this desired response, along with the filter order, and uses the Parks-McClellan algorithm to design an optimal equiripple FIR filter. The Parks-McClellan algorithm minimizes the difference between the desired and actual filter response across the entire frequency range.

To design arbitrary-magnitude FIR filters, enter or modify the desired frequency-magnitude points and choose an interpolation type to

generate the desired response between your specified points. The DFD application automatically designs and plots the equiripple FIR filter. You get immediate graphical feedback to help determine whether the filter meets your specifications.

Arbitrary FIR Filter Design Panel Controls and Displays

DFD Menu ▽

Use the design panel DFD Menu to save your filter specifications and coefficients, load filter designs from previous work, open the Analysis or DAQ and Filter panels, or return to the Filter Design Main Menu.

The graph in Figure 2-11 plots the desired and actual magnitude response of the designed FIR filter.
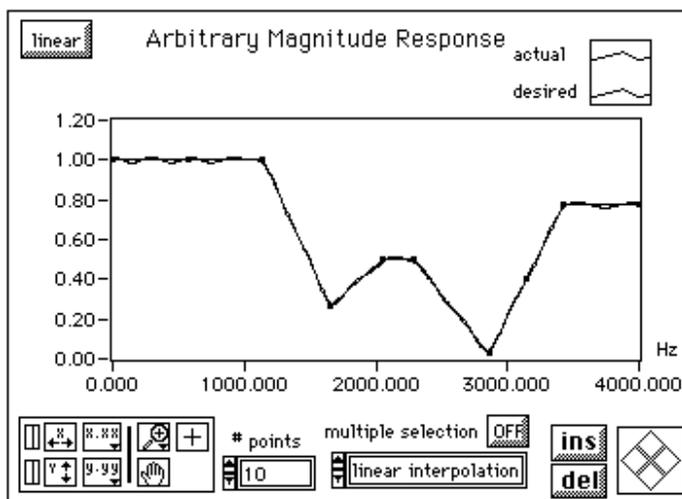
Figure 2-11. Desired and Actual Magnitude Response

The y-axis is in linear or decibel units, depending on how you set the button in the upper left corner of the graph. The x-axis is in hertz. The full scale ranges from 0.0 to Nyquist (sampling rate/2).

dB
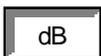
The dB button controls the display units (linear or decibel) of all the magnitude controls and displays.

These controls and displays include the Magnitude vs. Frequency plot (y-axis) and the magnitudes in the array of frequency-magnitude points.

The **# points** control specifies the total number of frequency-magnitude points the DFD application uses to create the desired filter magnitude response.

```
# points
▲
▼  1 0
```
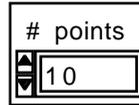
Reducing this number deletes points from the end of the frequency-magnitude array, while increasing this number inserts the additional number of points to the right of the selected point.

Set the multiple selection button to on to select more than one frequency-magnitude point on the response graph. Clicking on a point you already selected removes that point from the selection list.

```
multiple selection  OFF
```
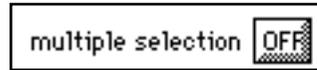
The interpolation control selects the type of interpolation the DFD application uses to generate the desired response from the array of frequency-magnitude points.

```
▲
▼  linear  interpolation
```

Choose linear interpolation to create "flat" filters (lowpass, highpass, bandpass, and bandstop). Choose spline interpolation to create smoothly-varying filters.

ins

Click on the insert button to insert a frequency-magnitude point between the selected point and the next point.

If the selected point is the last point in the frequency-magnitude array, the DFD application inserts the new point between the last two points of the array.

The DFD application inserts new points at halfway along the line connecting the two outer points.

del

Click on the delete button to delete the selected frequency-magnitude points. The DFD application deletes all selected points.

These points are the selected frequency-magnitude points. You can select points on the Arbitrary Magnitude Response graph by clicking on the point, or directly from the frequency-magnitude array shown at right by clicking on the circle to the right of each point.
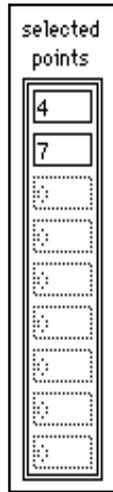
The following array is the array of frequency-magnitude points the DFD application uses to construct the desired filter magnitude response. The DFD application forms the desired filter response by interpolating between these points.
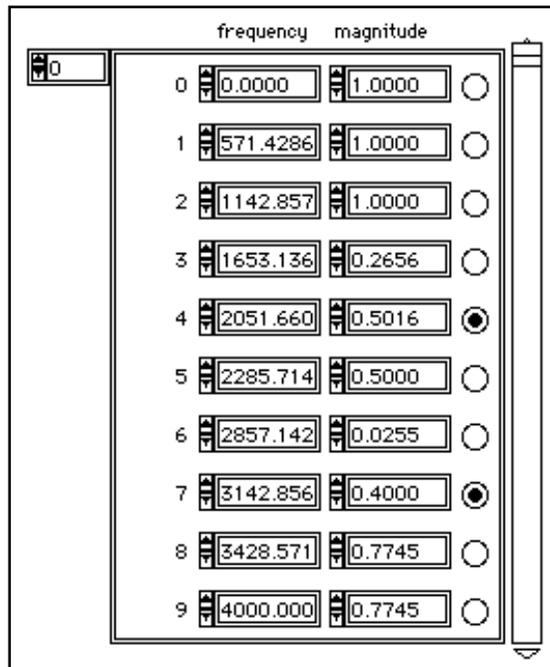
The frequency of each point is in hertz and the magnitude is in linear
or decibel units of gain, depending on the setting of the button in the
upper left corner of the Arbitrary Magnitude Response graph.

You can select points in this array by clicking in the circle to the right
of each point. You can then delete the selected points by clicking on
the delete button, or move them by clicking on the desired direction
diamond in the lower right corner of the Arbitrary Magnitude Response
graph.

The **filter order** control specifies the total number of coefficients in the
digital FIR filter.
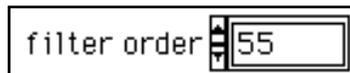
The **ripple** indicator displays the largest absolute error (linear) between the desired and actual filter responses.

ripple   1.3709E-2
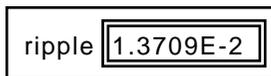
The Message window displays errors that occurred during the FIR design procedure.

message
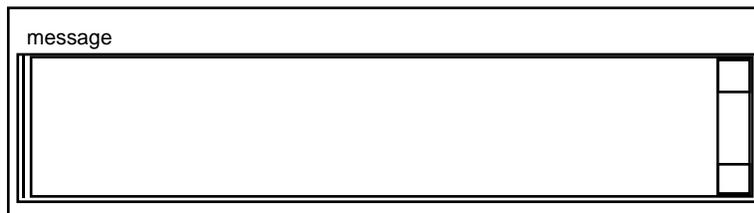
Click in the locked frequencies box to lock the present frequency values of the frequency-magnitude points. If you click in this box, you can alter only the magnitude or y-value of the frequency-magnitude points.

☐ locked frequencies   ☐ sort by frequency
☐ uniform spacing   ☐ import from file

Click in the uniform spacing box to space the frequency values of the frequency-magnitude points. The DFD application spaces the frequency-magnitude points uniformly from 0.0 to sampling rate/2, inclusive.

Click in the sort by frequency box to sort the frequency-magnitude points in both the response graph and the array according to ascending frequency.

The value of each frequency-magnitude point remains unchanged; however, the point order may change.

The sampling rate control specifies the sampling rate in samples per second (hertz).

sampling rate ⬦ 8000.00

Analysis of Filter Design Panel

Figure 2-12 shows the Analysis of Filter Design panel. Use this panel to view the filter magnitude response, phase response, impulse response, step response, and pole-zero plot. You can also view and print full-screen plots of each response. From the full-screen views, you can save the analysis results to text files.
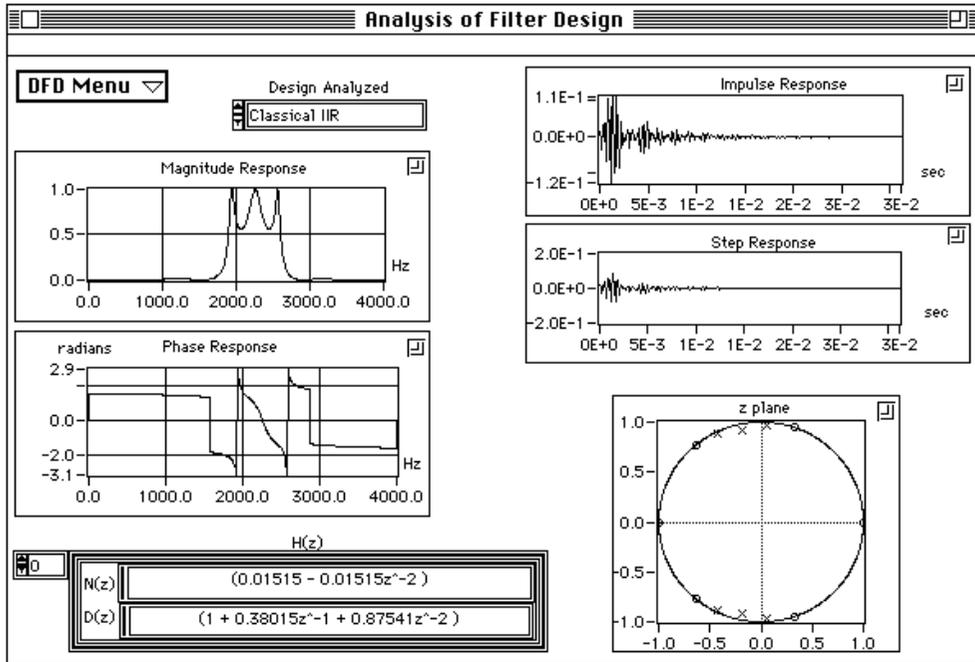
Figure 2-12. Analysis of Filter Design Panel

If you selected **DFD Menu»Analysis** from a filter design panel, the Analysis of Filter Design panel uses that particular filter design to compute the various filter responses. You can also analyze any of the four filter designs from the Design Analyzed ring selector; the Analysis of Filter Design panel uses the filter parameters from the selected filter design.

Use the DFD Menu to load filter designs from previous work, open the DAQ and Filter panel, go to the selected filter design panel, or return to the Filter Design Main Menu.

Use the Design Analyzed control to select which filter control to analyze. If you continue to modify the same filter design that is presently being analyzed, the DFD will recompute all filter responses.

```
         Design Analyzed
  ┌─────────────────────────┐
  │ Classical IIR           │
  └─────────────────────────┘
```
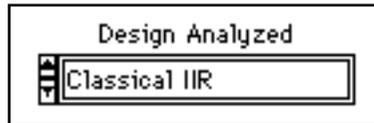
Analysis Displays

zoom box

Each of the five filter plots has a zoom box in the upper right corner. Clicking in this box brings up a full-screen version of that plot. In the full-screen versions of these plots, you can change the units from linear to decibel (magnitude response), from radians to degrees (phase response) or from seconds to samples (impulse and step responses). From each full screen view, you can save the response data to text files.

Magnitude Response

The magnitude response is the magnitude of the filter's response H(f) as frequency caries from zero to half the sampling rate.

The following figure illustrates the magnitude response of the selected filter design.
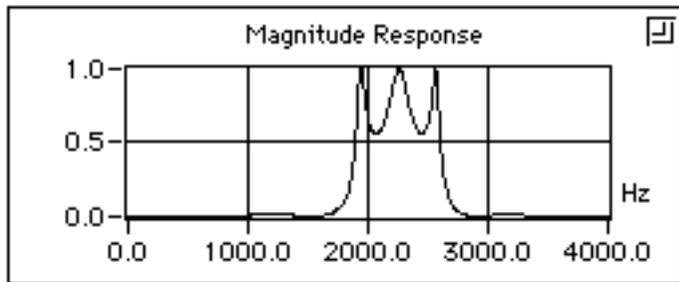
Phase Response

The phase response is the phase of the filter's response H(f) as frequency varies from zero to the sampling rate.

The following figure illustrates the phase response of the selected filter design.
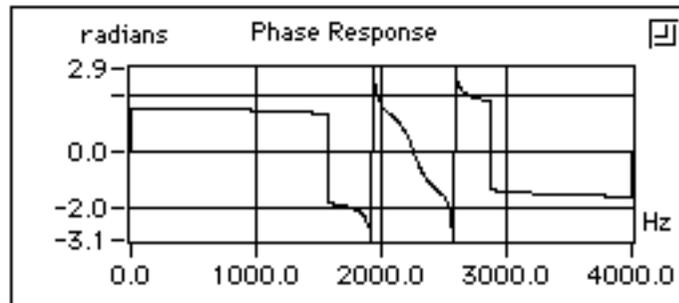
Impulse Response

The impulse response of a digital filter is the filter's output when the input is a unit sample sequence (1, 0, 0,...). The input before the unity sample is also zero.

The following figure illustrates the phase response of the selected filter design.
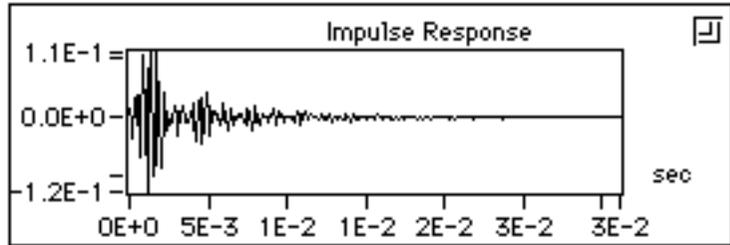
Step Response

The step response of a digital filter is the filter's output when the input is a unit step sequence (1, 1, 1, ...). The input samples before the step sequence is defined as zero.

The following figure illustrates the step response of the designed filter.
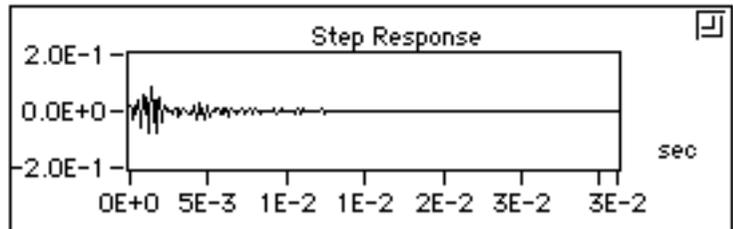
Z-Plane Plot

The following figure illustrates the z-plane plot of the filter poles and zeros.
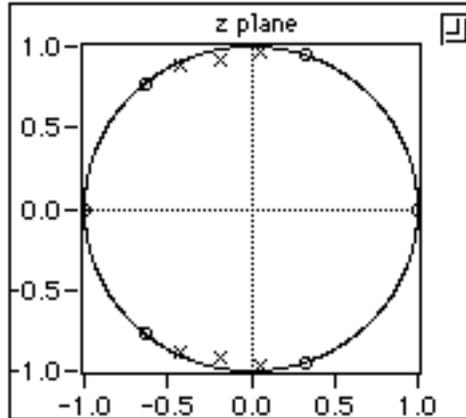
Each pole is represented by a red "x". Each zero is represented by a blue "o".

H(z) for IIR Filters

H(z) is the z-transform of the designed digital filter.
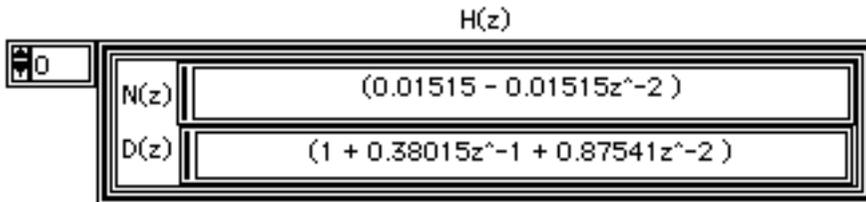
For an IIR filter, H(z) can be represented by a product of fractions of second order z polynomials.

$$H(z) = \prod_{k=1}^{N_s} \frac{N_k(z)}{D_k(z)}$$

$N_k(z)$ = numerator for stage k

$D_k(z)$ = denominator for stage k

N_s = number of second-order stages

You can view the N(z) and D(z) polynomials for other stages by incrementing the index shown in the upper left hand side of the H(z) display.

H(z) for FIR Filters

H(z) is the z-transform of the designed digital filter, as illustrated by the following figure. You can scroll through H(z) using the scroll bar at the right.

H(z)

```
(0.00635 - 0.00883z^-1 - 0.02848z^-2 + 0.04627z^-3 +
0.04104z^-4 - 0.11146z^-5 - 0.01413z^-6 + 0.18108z^-7 -
0.05985z^-8 - 0.20023z^-9 + 0.15162z^-10 + 0.15162z^-11
- 0.20023z^-12 - 0.05985z^-13 + 0.18108z^-14 -
0.01413z^-15 - 0.11146z^-16 + 0.04104z^-17 +
```
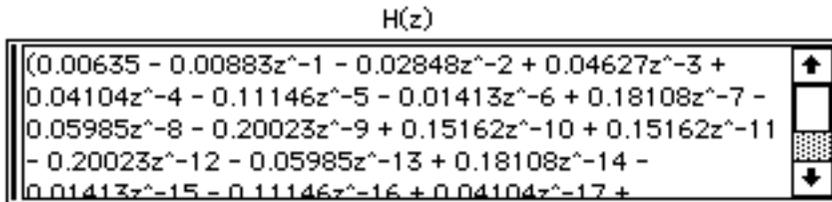
For an FIR filter, H(z) can be represented as a polynomial in z^-1.

$$H(z) \ = \ \sum_{j \, = \, 0}^{order \, - \, 1} h_j z^{-j}$$

where
j=0, 1, ... order -1
h_j= FIR filter coefficients
order = number of FIR coefficients

DAQ and Filter Panel

Figure 2-13 shows the DAQ and Filter panel. You can use this panel if you have a National Instruments DAQ device and you want to see how the present filter design performs on real-world signals. In this panel, you can set up your DAQ device and then acquire real signals. The

acquired data then passes through the currently designed filter, and the DFD application plots the input and output waveforms and spectrums.
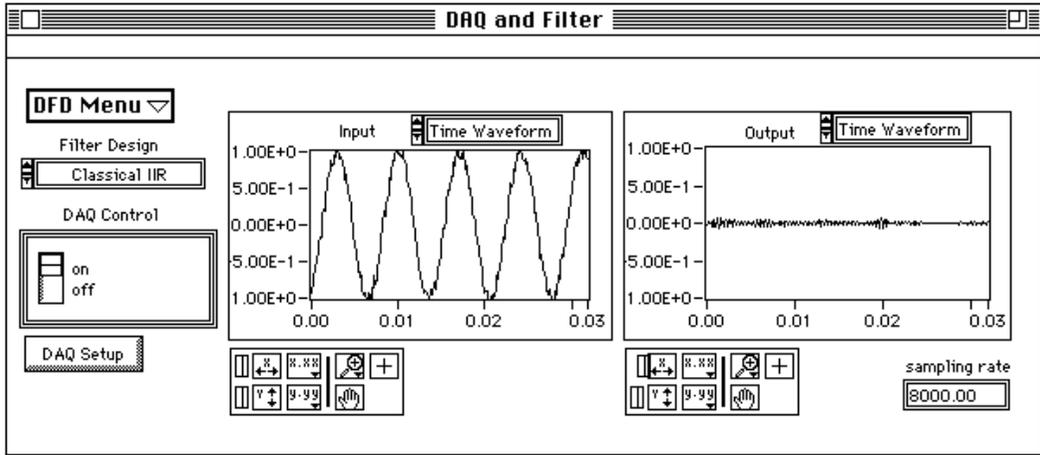
Figure 2-13. The DAQ and Filter Panel

If you select **DFD Menu»DAQ and Filter** from a filter design panel, the DAQ and Filter panel uses that particular set of filter coefficients when filtering the acquired signals. You can also use any of the four filter designs from the Filter Design ring selector; the DAQ and Filter panel uses the filter parameters from the selected design specifications.

Use the DFD Menu to load and test filter designs from previous work, open Analysis panel, goto the selected filter design panel, or return to the Filter Design Main Menu.

Use the Filter Design control to set the Filter Design to use in filtering of the acquired signal. From the DFD menu select "Goto Design" to load and run the corresponding filter design panel.
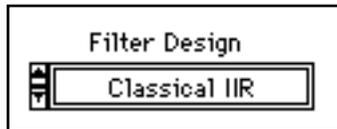
Use the on/off switch to control whether you want the DFD to acquire blocks continuously or on demand.
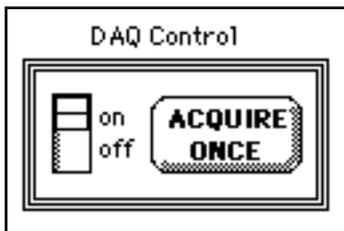
Set the switch to "on" to continuously acquire blocks of data.

Set the switch to "off" to acquire when "ACQUIRE ONCE" button is clicked.
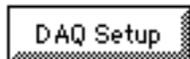
DAQ Setup   Click on the DAQ Setup button to change the data acquisition settings such as the device number, number of samples to acquire, triggering parameters, or sampling rate.

Select between viewing the Time Waveform or the Spectrum of the input acquired signal or the filtered signal, using the ring selector, shown just above each response plot. Figure 2-14 shows an example of
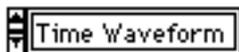
switching displays for the spectrum of both the input and filtered signals:
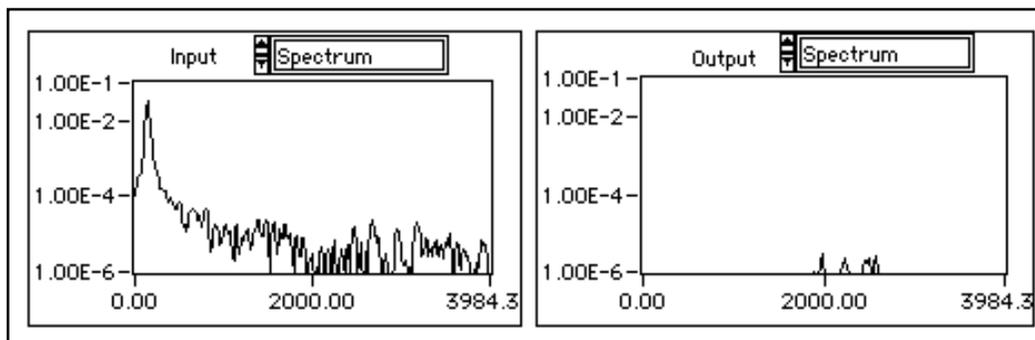
Figure 2-14. Switching Displays

The actual sampling rate is shown in an indicator at the lower left-hand side of the DAQ and Filter Panel, as in the following figure.
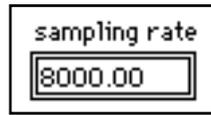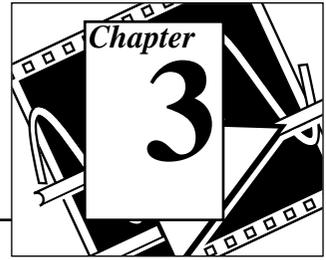
IIR and FIR Implementation

This chapter contains a description of the filter implementation equations for IIR and FIR filtering, and describes the format of the IIR and FIR filter coefficient files.

Infinite Impulse Response Filters

Infinite impulse response filters (IIR) filters are digital filters with impulse responses that can theoretically be infinite in length (duration). The general difference equation characterizing IIR filters is

$$y_i = \frac{1}{a_0}\left(\sum_{j=0}^{N_b - 1} b_j x_{i-j} - \sum_{k=1}^{N_a - 1} a_k y_{i-k} \right) \tag{3-1}$$

where N_b is the number of *forward* coefficients (b_j) and N_a is the number of *reverse* coefficients (a_k).

In most IIR filter designs, coefficient a_0 is 1. The output sample at the present sample index i consists of the sum of scaled present and past inputs (x_i and x_{i-j} when $_j \neq 0$) and scaled past outputs (y_{i-k}).

The response of the general IIR filter to an impulse ($x_0 = 1$ and $x_i = 0$ for all $i \neq 0$) is called the impulse response of the filter. The impulse response of the filter described by equation (3-1) is indeed of infinite length for nonzero coefficients. In practical filter applications, however, the impulse response of stable IIR filters decays to near zero in a finite number of samples.

The advantage of digital IIR filters over finite impulse response (FIR) filters is that IIR filters usually require fewer coefficients to perform similar filtering operations. Thus, IIR filters execute much faster and do not require extra memory, because they execute in place.

The disadvantage of IIR filters is that the phase response is nonlinear. If the application does not require phase information, such as simple

signal monitoring, IIR filters may be appropriate. You should use FIR filters for those applications requiring linear phase responses.

IIR filters are also known as recursive filters or autoregressive moving-average (ARMA) filters. See Appendix A, *References*, for material that can give you more information on this topic.

Cascade Form IIR Filtering

Filters implemented using the structure defined by equation 3-1 directly are known as *direct form* IIR filters. Direct form implementations are often sensitive to errors introduced by coefficient quantization and by computational precision limits. Additionally, a filter designed to be stable can become unstable with increasing coefficient length, which is proportional to filter order.

A less sensitive structure can be obtained by breaking up the direct form transfer function into lower order sections, or filter stages. The direct form transfer function of the filter given by equation 3-1 (with $a_0 = 1$) can be written as a ratio of z transforms, as follows:

$$H(z) \ = \ \frac{b_0 + b_1 z^{-1} + \ldots + b_{N_b - 1} z^{-(N_b - 1)}}{1 + a_1 z^{-1} + \ldots + a_{N_a - 1} z^{-(N_a - 1)}} \tag{3-2}$$

By factoring equation 3-2 into second-order sections, the filter's transfer function becomes a product of second-order filter functions

$$H(z) \ = \ \prod_{k \ = \ 1}^{N_s} \frac{b_{0k} + b_{1k} z^{-1} + b_{2k} z^{-2}}{1 + a_{1k} z^{-1} + a_{2k} z^{-2}} \tag{3-3}$$

where $N_s = \lfloor N_a/2 \rfloor$ is the largest integer $\leq N_a/2$, and $N_a \geq N_b$. This new filter structure can be described as a *cascade* of second-order filters.

Cascaded Filter Stages
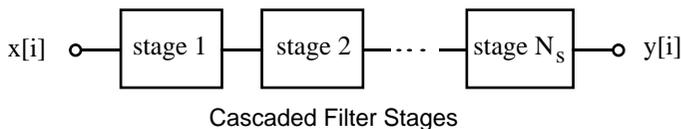
Each individual second-order stage can be implemented using the direct form filter equations:

$$y[i] = b_0 x[i] + b_1 x[i-1] + b_2 x[i-2] - a_1 y[i-1] - a_2 y[i-2]$$

These *direct form* equations can be represented graphically as follows.
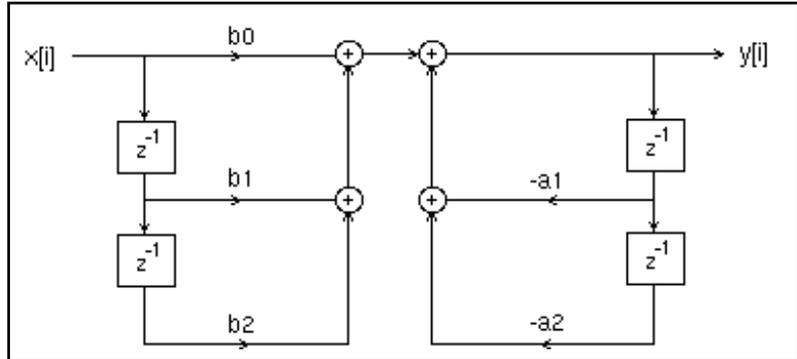
For each stage, you must maintain two past inputs ($x[i-1]$, $x[i-2]$) and two past outputs ($y[i-1]$, $y[i-2]$)

A more efficient implementation of each second-order stage is known at the direct form II. Each individual second-order stage can be implemented using the direct form II filter equations:

$$s[i] = x[i] - a_1 s[i-1] - a_2 s[i-2]$$

$$y[i] = b_0 s[i] + b_1 s[i-1] + b_2 s[i-2]$$

These *direct form II* equations can be represented graphically as follows.
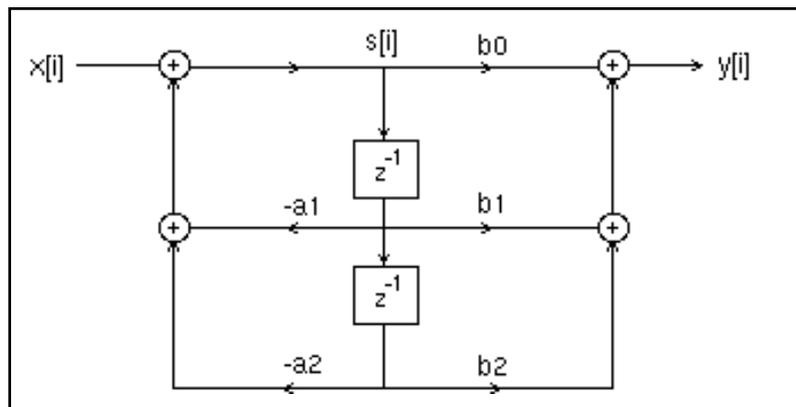
Finite Impulse Response Filters

Finite impulse response (FIR) filters are digital filters whose impulse response is finite. FIR filters are also known as nonrecursive filters, convolution filters, or moving-average (MA) filters because you can express the output of an FIR filter as a finite convolution

$$y_i = \sum_{k=0}^{n-1} h_k x_{i-k} \qquad (3\text{-}4)$$

where *X* represents the input sequence to be filtered, *Y* represents the output filtered sequence, and *H* represents the FIR filter coefficients.

The following list gives the most important characteristics of FIR filters.

- They can achieve linear phase due to filter coefficient symmetry in the realization.

- They are always stable.

- You can perform the filtering function using the convolution and, as such, a delay is generally associated with the output sequence

$$\text{delay} = \frac{n-1}{2}$$

where *n* is the number of FIR filter coefficients.

You design FIR filters by approximating a specified desired frequency response of a discrete-time system. The most common techniques approximate the desired magnitude response while maintaining a linear phase response.

Format of the Filter Coefficient Text Files

When you save your filter coefficients to a text file, the DFD application generates a readable text file containing all the information you need to implement the designed FIR or IIR digitial filter. This section details the format for both FIR and IIR filter coefficient files.

FIR Coefficient File Format

An example FIR coefficient text file and description is given in the following table:

coefficient file example	description
FIR Filter Coefficients	type of file
Sampling Rate	sampling rate label
8.000000E+3	sampling rate in Hz
N	filter order label
22	filter order
h[0..21]	coefficients label
6.350871E-3	1st coefficient, h[o]
-8.833535E-3	2nd coefficient, h[1]
-2.847674E-2	.
4.626607E-2	.
4.103986E-2	.
-1.114579E-1	
-1.412791E-2	
1.810791E-1	
-5.984635E-2	
-2.002337E-1	

coefficient file example	description
1.516199E-1	
1.516199E-1	
-2.002337E-1	
-5.984635E-2	
1.810791E-1	
-1.412791E-2	
-1.114579E-1	
4.103986E-2	
4.626607E-2	.
-2.847674E-2	.
-8.833535E-3	.
6.350871E-3	last coefficient, h[N-1]

You can implement the FIR filter using equation 3-4 directly.

IIR Coefficient File Format

IIR coefficient files are slightly more complex than FIR coefficient files. IIR filters are usually described by two sets of coefficients, *a* and *b* coefficients. There are a total of M * S *a* coefficients and (M+1)*S *b* coefficients, where M is the stage order (usually 2) and S is the number of stages. An IIR filter with three second-order stages has two *a* coefficients per stage for a total of six *a* coefficients, and three *b* coefficients per stage for a total of nine *b* coefficients.

An example IIR coefficient text file and description is given in the following table:

coefficient file example	description
IIR Filter Coefficients	coefficient type
Sampling Rate	sampling rate label
8.000000E+3	sampling rate in Hz
Stage Order	stage order label

coefficient file example	description
2	order of each stage
Number of Stages	number of stages label
3	number of stages
a Coefficients	a coefficients label
6	number of a coefficients
3.801467E-1	a1 for stage 1
8.754090E-1	a2 for stage 1
-1.021050E-1	a1 for stage 2
9.492741E-1	a2 for stage 2
8.460304E-1	a1 for stage 3
9.540986E-1	a2 for stage 3
b Coefficients	b coefficients label
9	number of b coefficients
1.514603E-2	b0 for stage 1
0.000000E+0	b1 for stage 1
1.514603E-2	b2 for stage 1
1.000000E+0	b0 for stage 2
6.618322E-1	b1 for stage 2
1.000000E+0	b2 for stage 2
1.000000E+0	b0 for stage 3
1.276187E+0	b1 for stage 3
1.000000E+0	b2 for stage 3

You can implement the IIR filter in cascade stages by using equation 3-1 (maintaining two past inputs and two past outputs for each stage), or by using the direct form II equations (maintaining two past internal states).

Using Your Coefficient Designs: DFD Utilities
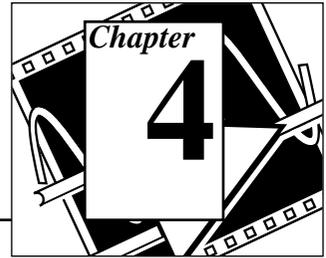
This chapter contains a description of the DFD Utilities that you use for LabVIEW®, LabWindows®/CVI, and Windows filtering applications.

LabVIEW DFD Utilities

This section contains descriptions of the DFD utilities that you can use from within your LabVIEW applications to read DFD filter coefficient files and to filter your data using the coefficients.

There are two DFD utility VIs, Read DFD Coefficients and DFD Filter. To use these VIs, connect the file path of your coefficient file to Read DFD Coefficients. The output Coefficient Cluster is then connected to DFD Filter, along with your input signal. Once this sequence is followed and the VIs have executed, your filtered data will be available at the DFD Filter output Filtered X.

Read DFD Coefficients

The following VI reads the Digital Filter Design (DFD) filter coefficient files and returns the coefficient data in a DFD coefficient cluster. You can use the DFD Filter VI to filter your signals using the DFD coefficient.
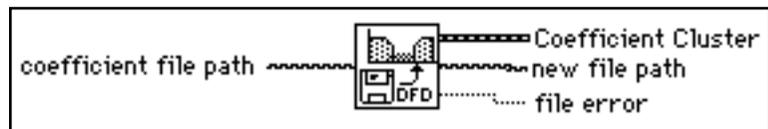
coefficient file path is the LabVIEW path to the DFD coefficient file. This file can either be in log or text file format. If coefficient file path is empty, you will be able to select a coefficient file from an open file dialog.

Coefficient Cluster is the cluster of coefficient information read from the coefficient file. The Coefficient Cluster is composed of the following.

coefficient type: either 0 (IIR) or 1 (FIR)
sampling rate: sampling rate in Hz
      IIR Filter Cluster: cascade IIR Filter Cluster
      h(n): FIR filter coefficients

new file path is the file path to the coefficient file read. If coefficient file path was empty, then the new file path contains the path to the file selected from the open file dialog.

file error is set to true if an error occurred while reading or interpreting the coefficient file.

DFD Filter

The DFD Filter VI filters the input array X using the Digital Filter Design (DFD) coefficient cluster. You should use the Read DFD Coefficients VI to read your DFD coefficient files and properly initialize the input Coefficient Cluster.
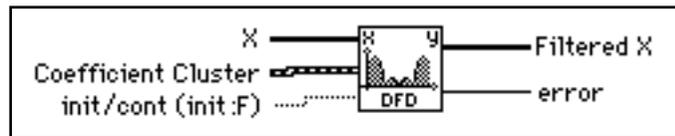
X contains the array of input samples to filter.

Coefficient Cluster is composed of the following.

coefficient type: either 0(IIR) or 1(FIR)
sampling rate: sampling rate in Hz
IIR Filter Cluster: cascade IIR Filter Cluster
h(n): FIR filter coefficients

init/cont (init:F) controls the initialization of the internal filter states. When init/cont (init:F) is FALSE (default), the internal states are initialized to zero. When init/cont (init:F) is TRUE, the internal filter states are initialized to the final filter states from the previous call to this instance of this VI. To filter a large data sequence that has been split into smaller blocks, set this control to FALSE for the first block, and to TRUE for continuous filtering of all remaining blocks.

Filtered X is the array of filtered output samples.

error is the error code returned from the filtering VIs. You can wire this output to the Find First Error VI to produce an error cluster. This cluster can then be wired to the Simple Error Handler VI or the General Error Handler VI for an immediate report on any errors. You can also look up error codes in the Error Codes Appendix in the LabVIEW Cross Reference Manual.

LabWindows/CVI Utilities

This section contains descriptions of the DFD utilities that you can use from within your LabWindows/CVI applications to read DFD filter coefficient files and to filter your data using the coefficients.

The DFD Instrument Driver

The Digital Filter Design Toolkit provides a LabWindows/CVI instrument driver file called *DFDUTILS.FP*. This file can be found in the path *DFDUTILS\CVISRC\INSTR* of your installation directory.

The DFD utility functions contained in the instrument driver DFDUTILS.FP use a filter coefficient structure that holds the filter coefficients. Here is a listing of header file DFDUTILS.H which contains this filter structure as well as the four DFD utility function prototypes.

```
#define intnum long
#define floatnum double
typedef struct {
    intnum type;/* type of filter (lp,hp,bp,bs) */
    intnum order;/* order of filter */
    intnum reset;/* 0 - don't reset, 1 - reset */
    intnumn a;/* number of a coefficients */
    floatnum *a;/* pointer to a coefficients */
    intnum nb;/* number of b coefficients */
    floatnum *b;/* pointer to b coefficients */
    intnum ns;/* number of internal states */
    floatnum *s;/*pntr to internal state array */
}FilterStruct, *FilterPtr;
FilterPtr AllocCoeffDFD (void);
long ReadCoeffDFD (char coeffPath[],FilterPtr
filterCoefficients, double *samplingrate);
long FreeCoeffDFD (FilterPtr filterCoefficients);
```

```
long FilterDFD (double inputArray[], long n, FilterPtr
filterCoefficients, double outputArray[]);
```

AllocCoeffDFD

FilterPtr **fptr** = **AllocCoeffDFD** (void);

Use AllocCoeffDFD to allocate and clear the DFD filter coefficient
structure. You must always call this once to properly allocate the DFD
filter coefficient structure.

Return Value

fptr	FilterPtr	pointer to allocated filter structure

ReadCoeffDFD

long **err** = **ReadCoeffDFD** (char **coeffPath**[],
          FilterPtr **filterCoefficients**, double ***samplingRate**);

Use ReadCoeffDFD to read your DFD filter coefficient file. You must always call AllocCoeffDFD once before calling this function.

Parameters

Input	**coeffPath**	character array	pathname of DFD coefficient file
Output	**filterCoefficients**	FilterPtr	pointer to filter coefficient structure
	samplingRate	double	pointer to sampling rate

Return Value

err	long integer	error code

FreeCoeffDFD

long **err** = long **FreeCoeffDFD** (FilterPtr **filterCoefficients**);

Use FreeCoeffDFD to free the DFD filter coefficient structure and all of its coefficient arrays.

Parameters

Input	**filterCoefficients**	FilterPtr	pointer to filter coefficient structure

Return Value

err	long integer	error code

FilterDFD

long **err** = **FilterDFD** (double **inputArray**[], long **n**,
        FilterPtr **filterCoefficients**, double **outputArray**[]);

Use FilterDFD to filter the input samples using the DFD filter coefficients. You must always call AllocCoeffDFD and ReadCoeffDFD once before calling this function.

This function can be used to filter blocks of one continuous sequence of input samples. The input state of the filter is maintained using the DFD filter coefficient structure. The number of output samples will always equal the number of input samples (n).

Parameters

Input	**inputArray**	double array	input array of unfiltered samples
	n	long integer	number of elements in input array
	filterCoefficients	FilterPtr	pointer to filter coefficient structure
Output	**outputArray**	double array	output array of filtered samples which must be at least as large as inputArray

Return Value

err	long integer	error code

Using the DFD Instrument Driver

Add the DFDUTILS.FP to your project and DFDUTILS.H to your source code. Now you can call the DFD utility functions in your C code. An example called DFDXMPL.PRJ under subdirectory of DFDUTILS\CVISRC\EXAMPLE shows you how to call the DFD utility functions.

Windows DLL DFD Utilities

This section contains descriptions of the DFD utilities that you can use from within your LabWindows/CVI applications to read DFD filter coefficient files and to filter your data using the coefficients.

The Digital Filter Design Toolkit provides a 16-bit DLL called DFD16.DLL and a 32-bit DLL called DFD32.DLL for Windows' users. These two DLLs are located in the subdirectory DFDUTILS\WINSRC\WINDLL of your installation directory.

The function prototypes in the DFD DLL are:

```
FilterPtr FAR PASCAL AllocCoeffDFD (void);
long FAR PASCAL ReadCoeffDFD (char
    coeffPath[],FilterPtr filterCoefficients,
    double *samplingrate);
long FAR PASCAL FreeCoeffDFD (FilterPtr
    filterCoefficients);
long FAR PASCAL FilterDFD (double inputArray[],
    long n, FilterPtr filterCoefficients,
    double outputArray[]);
```

See the descriptions for each function and its parameters in the previous section LabWindows/CVI Utilities . Call these functions the same way in your code as you call other DLL functions.

The Digital Filter Design Toolkit also provides an example for Visual Basic 3.0 showing you how to call the DFD utility functions. The source code is located under the subdirectory DFDUTILS\WINSRC\EXAMPLE\VB of your installation directory.

Before you run the Visual Basic example, copy DFD16.DLL to your Windows directory.
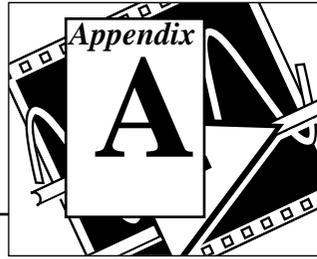
References

This appendix lists the reference material used to produce the Digital Filter Design Toolkit. For more information about the theories and algorithms implemented in the Digital Filter Design Toolkit, refer to the following documents.

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

2. T. W. Parks and C. S. Burrus, *Digital Filter Design*, John Wiley & Sons, Inc., 1987

3. L. B. Jackson, *Digital Filters and Signal Processing*, Boston: Kluwer, 1986

4. A. B. Williams and F. J. Taylor, *Electronic Filter Design Handbook*, New York: McGraw-Hill, 1988

5. T. W. Parks and J. H. McClellan, "Chebyshev Approximation for Nonrecursive Digital Filters with Linear Phase", *IEEE Trans. Curcuit Theory*, Vol. CT-19, pp 189-194, Mar. 1972a

6. T. W. Parks and J. H. McClellan, "A Program for the Design of Linear Phase Finite Impulse Response Filters", *IEEE Trans. Audio Electroacoustics*, Vol. AU-20, No. 3, pp.195-199, Aug. 1972a
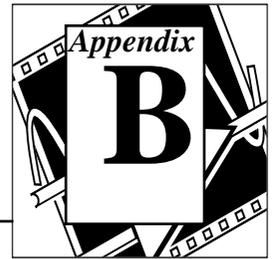
Customer Communication

For your convenience, this appendix contains forms to help you gather the information necessary to help us solve your technical problems and a form you can use to comment on the product documentation. When you contact us, we need the information on the Technical Support Form and the configuration form, if your manual contains one, about your system configuration to answer your questions as quickly as possible.

National Instruments has technical assistance through electronic, fax, and telephone systems to quickly provide the information you need. Our electronic services include a bulletin board service, an FTP site, a FaxBack system, and e-mail support. If you have a hardware or software problem, first try the electronic support systems. If the information available on these systems does not answer your questions, we offer fax and telephone support through our technical support centers, which are staffed by applications engineers.

Electronic Services

Bulletin Board Support

National Instruments has BBS and FTP sites dedicated for 24-hour support with a collection of files and documents to answer most common customer questions. From these sites, you can also download the latest instrument drivers, updates, and example programs. For recorded instructions on how to use the bulletin board and FTP services and for BBS automated information, call (512) 795-6990. You can access these services at:

United States: (512) 794-5422 or (800) 327-3077
  Up to 14,400 baud, 8 data bits, 1 stop bit, no parity

United Kingdom: 01635 551422
  Up to 9,600 baud, 8 data bits, 1 stop bit, no parity

France: 1 48 65 15 59
  Up to 9,600 baud, 8 data bits, 1 stop bit, no parity

FTP Support

To access our FTP site, log on to our Internet host, `ftp.natinst.com`, as anonymous and use your Internet address, such as `joesmith@anywhere.com`, as your password. The support files and documents are located in the `/support` directories.

FaxBack Support

FaxBack is a 24-hour information retrieval system containing a library of documents on a wide range of technical information. You can access FaxBack from a touch-tone telephone at the following numbers:

(512) 418-1111 or (800) 329-7177

E-Mail Support (currently U.S. only)

You can submit technical support questions to the appropriate applications engineering team through e-mail at the Internet addresses listed below. Remember to include your name, address, and phone number so we can contact you with solutions and suggestions.

GPIB: gpib.support@natinst.com
DAQ: daq.support@natinst.com
VXI: vxi.support@natinst.com
LabVIEW: lv.support@natinst.com
LabWindows: lw.support@natinst.com
HiQ: hiq.support@natinst.com

Fax and Telephone Support

National Instruments has branch offices all over the world. Use the list below to find the technical support number for your country. If there is no National Instruments office in your country, contact the source from which you purchased your software to obtain support.

	Fax	Telephone
Australia	03 9 879 9422	03 9 879 9179
Austria	0662 45 79 90 0	0662 45 79 90 19
Belgium	02 757 00 20	02 757 03 11
Canada (Ontario)	519 622 9310	519 622 9311
Canada (Quebec)	514 694 8521	514 694 4399
Denmark	45 76 26 00	45 76 71 11
Finland	90 527 2321	90 502 2930
France	1 48 14 24 24	1 48 14 24 14
Germany	089 741 31 30	089 714 60 35
Hong Kong	2645 3186	2686 8505
Italy	02 48301892	02 48301915
Japan	03 5472 2970	03 5472 2977
Korea	02 596 7456	02 596 7455
Mexico	5 202 2544	5 520 3282
Netherlands	03480 33466	03480 30673
Norway	32 84 84 00	32 84 86 00
Singapore	2265886	2265887
Spain	91 640 0085	91 640 0533
Sweden	08 730 49 70	08 730 43 70
Switzerland	056 20 51 51	056 20 51 55
Taiwan	02 377 1200	02 737 4644
U.K.	01635 523545	01635 523154

Technical Support Form

Photocopy this form and update it each time you make changes to your software or hardware, and use the completed copy of this form as a reference for your current configuration. Completing this form accurately before contacting National Instruments for technical support helps our applications engineers answer your questions more efficiently.

If you are using any National Instruments hardware or software products related to this problem, include the configuration forms from their user manuals. Include additional pages if necessary.

Name _____

Company _____

Address _____

_____

Fax (___)_____ Phone (___) _____

Computer brand _____ Model _____ Processor _____

Operating system (include version number) _____

Clock speed _____MHz RAM _____MB   Display adapter _____

Mouse ___yes ___no   Other adapters installed _____

Hard disk capacity _____MB   Brand _____

Instruments used _____

_____

National Instruments hardware product model _____ Revision_____

Configuration_____

National Instruments software product _____ Version_____

Configuration_____

The problem is: _____

_____

_____

_____

_____

List any error messages: _____

_____

_____

The following steps reproduce the problem: _____

_____

_____

_____

_____

Documentation Comment Form

National Instruments encourages you to comment on the documentation supplied with our products. This information helps us provide quality products to meet your needs.

Title: Digital Filter Design Toolkit Reference

Edition Date: August 1995

Part Number: 320953A-01

Please comment on the completeness, clarity, and organization of the manual.

_____

_____

_____

_____

_____

_____

_____

If you find errors in the manual, please record the page numbers and describe the errors.

_____

_____

_____

_____

_____

_____

_____

Thank you for your help.

Name _____

Title _____

Company _____

Address _____

_____

Phone (   ) _____

Mail to: Technical Publications   **Fax to:** Technical Publications
National Instruments Corporation   National Instruments Corporation
6504 Bridge Point Parkway, MS 53-02   MS 53-02
Austin, TX 78730-5039   (512) 794-5678
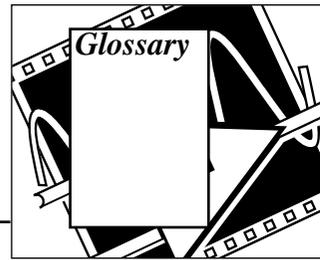
Prefix	Meaning	Value
m-	milli-	10^{-3}
μ-	micro-	10^{-6}
n-	nano-	10^{-9}

Numbers

1D	One-dimensional.
2D	Two-dimensional.

A

ASCII	American Standard Code for Information Interchange.

D

DAQ	Data acquisition.
datalog file	A file for saving LabVIEW data in binary format.
DFD	Digital Filter Design
DFT	Discrete Fourier transform.

F

FFT	Fast Fourier transform.
FIR	Finite Impulse Response
frame	A segment of time domain data.

I

IIR   Infinite Impulse Response.

M

MB   Megabytes of memory.

N

Nyquist Rate   Half the sampling rate.

O

oversampling rate   The ratio between the number of Gabor coefficients and the number of test samples.

S

sampling rate   The rate at which a continuous waveform is digitized.

spectral changes   Changes in the frequency content of a signal.

V

VI   Virtual instrument.