LabVIEW™
BridgeVIEW™
LabWindows/CVI™

Digital Filter Design Toolkit
Reference Manual

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National Instruments Corporate Headquarters

6504 Bridge Point Parkway   Austin, Texas 78730-5039   USA   Tel: 512 794 0100

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About This Manual

The *Digital Filter Design Toolkit Reference Manual* contains the information you need to get started with digital filter design within BridgeVIEW, LabVIEW, and LabWindows/CVI. This manual explains digital filter design, describes the application you use to design infinite impulse resoponse (IIR) and finite impulse response (FIR) digital filters, the filter implementation equations for IIR and FIR filtering, and the DFD utilities you use for BridgeVIEW, LabVIEW, LabWindows/CVI, and Windows filtering applications. This manual assumes you know how to operate your computer and that you are familiar with its operating system.

Organization of This Manual

The *Digital Filter Design Toolkit Reference Manual* is organized as follows:

- Chapter 1, *Overview of the Digital Filter Design Toolkit*, describes 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 you use to design infinite impulse response (IIR) and finite impulse response (FIR) digital filters.

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

- Chapter 4, *Using Your Coefficient Designs with DFD Utilities*, describes the DFD Utilities that you use for BridgeVIEW, LabVIEW, LabWindows/CVI, and Windows filtering applications.

- Appendix A, *References*, lists the reference material used to produce the Digital Filter Design Toolkit.

- 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:

»	The **»** symbol leads you through nested 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** options from the last dialog box.
bold	Bold text denotes the names of menus, menu items, parameters, dialog boxes, dialog box buttons or options, icons, windows, Windows 95 tabs, panels, controls, or LEDs.
bold italic	Bold italic text denotes an activity objective, note, caution, or warning.
italic	Italic text denotes variables, emphasis, a cross reference, or an introduction to a key concept. This font also denotes text from which you supply the appropriate word or value, as in Windows 3.*x*.
`monospace`	Text in this font denotes text or characters that you should literally enter 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, operations, variables, filenames and extensions, and for statements and comments taken from programs.

Related Documentation

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

- *BridgeVIEW Online Reference Manual*
- *BridgeVIEW User Manual*
- *G Programming Reference Manual*
- *LabVIEW Online Reference Manual*
- *LabVIEW User Manual*
- *LabWindows/CVI User Manual*

Additional reference material can be found in Appendix A, *References*.

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.

1

Overview of the Digital Filter Design Toolkit

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

Installation

The following sections provide installation instructions for the Digital Filter Design Toolkit on Macintosh or Windows.

Macintosh and Power Macintosh

Complete the following steps to install the toolkit.

1. Insert the CD of the Digital Filter Design Toolkit into your CD-ROM drive, open the `Mac (68k & PowerMacs):DFD Disks` folder, and double-click the DFD Toolkit Installer icon.

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. Insert the CD of the Digital Filter Design Toolkit into your CD-ROM drive.

2. If you are using LabVIEW 5.*x* or BridgeVIEW 2.*x*, run the `Windows\DFD5.0\setup.exe`. If you are using LabVIEW 4.*x* or BridgeVIEW 1.*x*, run `Windows\DFD4.1\setup.exe`.

3. 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 include 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 enable you to replace analog filters with digital filters in applications requiring 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.

Sampling Theory

This section describes 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.

For example, suppose you can sample the time signal of interest at Δt equally spaced intervals without losing information. The Δt parameter is the sampling interval.

You obtain the *sampling rate* or sampling frequency fs from the sampling interval:

$$fs \; = \; \frac{1}{\Delta t} \, .$$

This equation indicates that, according to the sampling theorem, the highest frequency the digital system can process is

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

This frequency, the highest frequency a digital system can process, is called the Nyquist frequency.

These equations also apply to digital filters. For example, if your sampling interval is

$$\Delta t = 0.001 \ sec ,$$

then the sampling frequency is

$$fs = 1000 \ Hz ,$$

and the highest frequency that the system can process is

$$f_{Nyq} = 500 \ Hz .$$

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

IIR and FIR Filters

The digital filters designed using the Digital Filter Design (DFD) application are either infinite impulse response (IIR) or finite impulse response (FIR) filters. IIR filters process both input and output samples. These filters use past output samples in feedback structures. This feedback results in sharp transitions in frequency response characteristics. Sharp transitions are useful in many applications, but the nonlinear phase distortion associated with IIR filters sometimes limits their selection in many phase-sensitive systems.

FIR filters, however, process only input samples and can be designed to have only linear phase distortion. You can compensate for the linear phase distortion by using simple integral delays in the application. All FIR filters designed using the DFD application are linear phase FIR filters. FIR filters might require more coefficients than IIR filters to meet similar design specifications.

2

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.

The Digital Filter Design (DFD) application provides complete filter design and analysis tools you can use to design digital filters to meet your precise filter specifications. You can design your IIR and FIR filters graphically, review filter responses interactively, 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 in the DFD application. You can view the time waveforms or the spectra of both the input signal and the filtered signal while simultaneously redesigning your digital filters.

Figure 2-1 illustrates the interaction between the Digital Filter Design Toolkit and related applications.
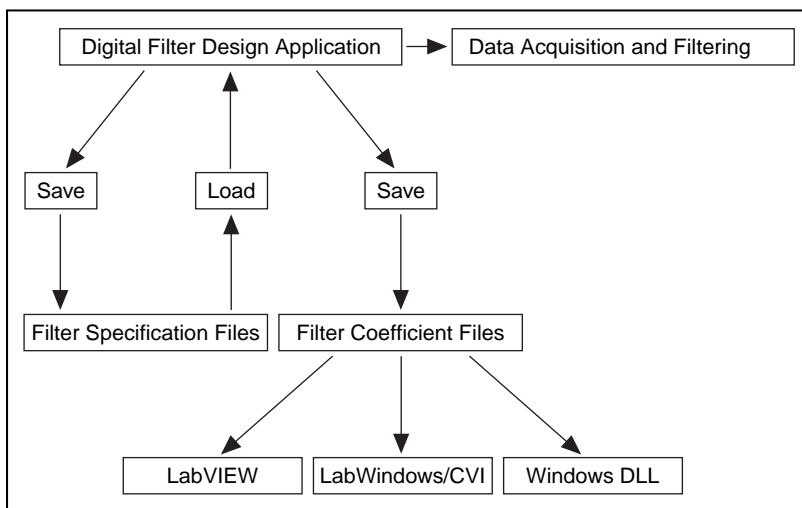

Figure 2-1. Conceptual Overview of the Digital Filter Design Toolkit

Main Menu

When you launch the DFD application, a panel displays the main available options. Figure 2-2 shows the **Main Menu** panel.
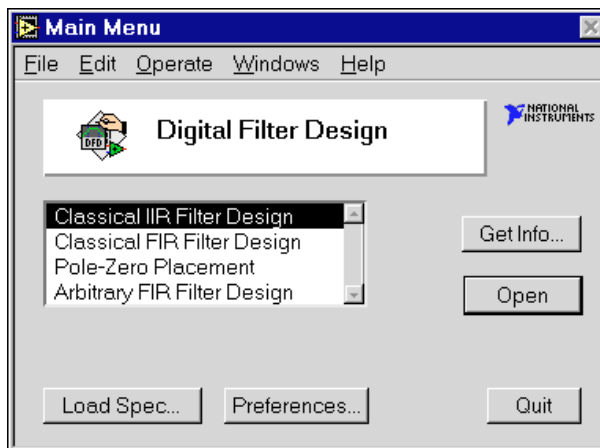

Figure 2-2. Filter Design Main Menu Panel

Opening the Filter Design Panels

From the **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 *Digital Filter Design Panels* section describes each design panel.

Directly Loading a Filter Specification File

You also can load a previously designed filter specification file directly from the **Main Menu** panel. When you click the **Load Spec…** button, the DFD application prompts you to select the filter specification file that you saved during previous design work. After you select the file, you can open the appropriate design panel for that specification file. Then you can resume work on an ongoing design project.

Editing the DFD Preferences

To customize your DFD application preferences, click the **Preferences...** button on the **Main Menu** panel. Then you can edit your DFD application preferences for future design sessions.

Quitting the DFD Application

Click the **Quit** button on the **Main Menu** panel to quit the DFD application.

Digital Filter Design Panels

When you double-click one of the four design selections in the **Main Menu** panel, 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 also can 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

The following sections describe the controls and features of the
DFD application.

Using the DFD Menu

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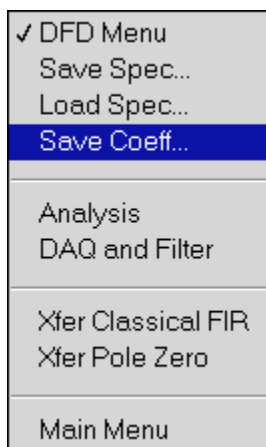
Figure 2-3. DFD Pop-up Menu

Saving Filter Specifications

To save all your specifications for the present filter design panel, select
DFD Menu»Save Spec.... The DFD application prompts you for the name
of the filter specification file to save. It is suggested that you name your
spec files appropriately for a given filter design. For example, if you design
a lowpass IIR filter, name the file `lowpass.iir`, or `lowp1.iir` 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 have no
effect on how the DFD 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 Menu»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 analysis.

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 in 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

Select **DFD Menu»Save Coeff...** to save your filter coefficients to a file. The DFD application first prompts you for the format of the coefficient file. You can select **text** format or **log** format. Select **text** format to view or print the coefficient file, or to use the coefficients in other non-LabVIEW filtering applications. Select **log** format for LabVIEW-only filtering applications. However, LabVIEW filtering utilities read both text-formatted 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. Name your coefficient files appropriately for a given filter design. For example, if you save bandpass IIR filter coefficients, name the file bpiir.txt or bpiir.log, depending on the coefficient file type.

Analyzing Filter Designs

To analyze your filter design, choose **DFD Menu»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 also can 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 test the present filter design on real-world signals, choose **DFD Menu»DAQ and Filter**. The DFD application loads and runs the **DAQ and Filter** panel. From this panel, you can configure your DAQ device and then acquire real signals. The acquired data passes through the currently designed filter, and the DFD application plots the input and output waveforms and spectrums.

Simulated DAQ and Filter Testing

You also can test your filter designs using a built-in simulated function generator. Choose **DFD Menu»DAQ and Filter** and configure the DAQ source to simulated DAQ. You then can click the **Function Generator** button on the **DAQ and Filter** panel to view and edit settings including signal type, frequency, amplitude, and noise level.

Transferring Filter Designs

You can transfer some filter design specifications from one design panel to another. For example, you can configure your passband and stopband requirements in designing an FIR filter and find the IIR filter that meets your design specifications. Not all design panels can share specifications. Table 2-2 shows the transfers you can perform 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 **Main Menu** panel, choose **DFD Menu»Main Menu**.

Panning and Zooming Options

Any graph you drop onto the front panel includes the graph palette.
This palette has controls for panning (scrolling the display area of a graph)
and for zooming in and out of sections of the graph. Many DFD graphs
include the graph palette. The following figure illustrates a graph with its
accompanying graph palette.
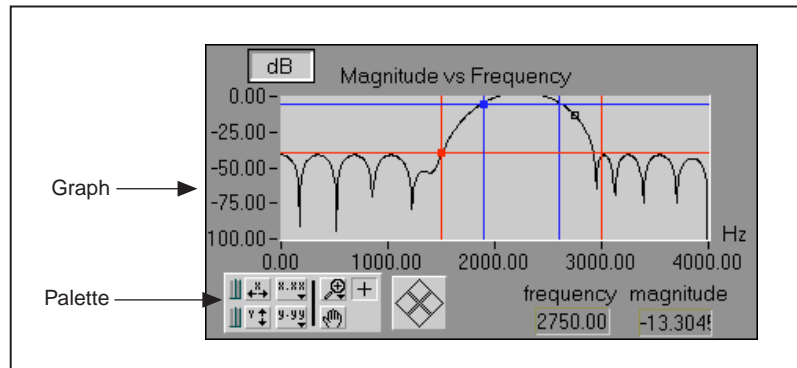
When you click the **Autoscale X** button, the DFD application autoscales
the *x*-data of the graph.

When you click the **Autoscale Y** button, the DFD application autoscales
the *y*-data of the graph.

If you want the graph to autoscale either of the scales continuously,
click the lock switch to lock autoscaling.

The scale format buttons give you run-time control over the format of the
x-scale and *y*-scale markers.

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

The plus or crosshatch indicates that you are in standard operate mode.
In operate mode, you can click in the graph to move cursors around.

When you click the panning tool, you switch to a mode in which you can
scroll the visible data by clicking and dragging sections of the graph.

When you click the zoom tool, you can zoom in on a section of the graph by dragging a selection rectangle around that section. If you click the zoom tool, a pop-up menu opens in which you can choose other methods of zooming. The following illustration shows this menu.
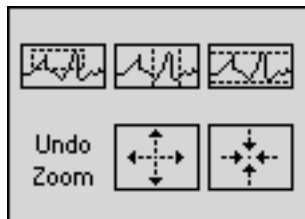
This menu contains the following buttons.

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 click and hold the mouse button on a specific point, the graph continuously zooms in until you release the mouse button.

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

☞ **Note**   *For the previous two modes, you can zoom in and zoom out about a point. Shift-click to zoom in the opposite direction.*

Graph Cursors

The following illustrations show two cursors and the cursor movement control.
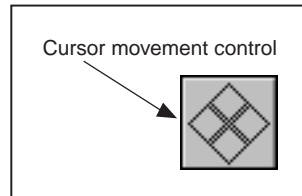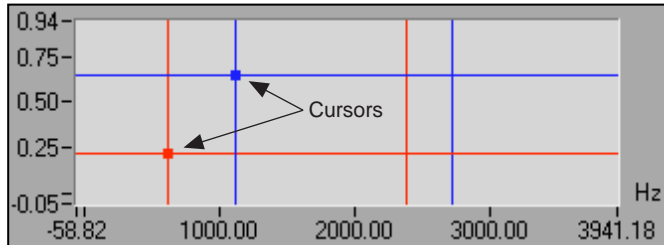
Move a cursor on a graph or chart by dragging it with the **Operating** tool or by using the cursor movement control. Click the arrows on the cursor movement control to move all cursors selected in the specified direction. You select cursors by moving them on the graph with the **Operating** tool.

Classical IIR Filter Design

Figure 2-4 shows the **Classical IIR Filter Design** panel. This 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

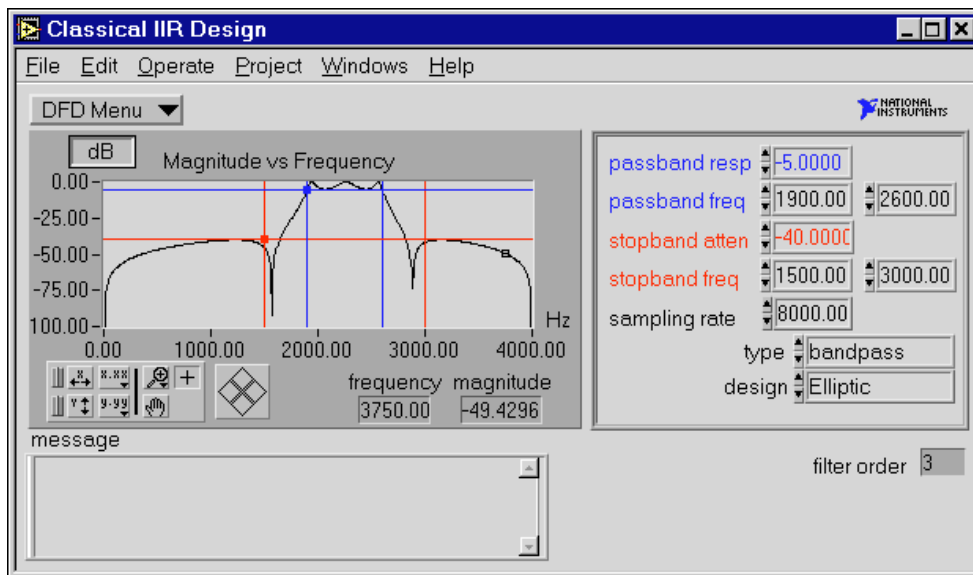Use this panel to design classical IIR digital filters. These filters include the classic types (lowpass, highpass, bandpass, and bandstop) and the classic designs (Butterworth, Chebyshev, Inverse Chebyshev, and Elliptic).

To design classical IIR filters, adjust the filter specifications. The passband and stopband requirements define a filter specification. You can 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 (fp_1), upper passband frequency (fp_2), and the passband response (Gp), define the passband specification. For the bandpass filter, the passband ranges from fp_1 to fp_2. 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 $fp_1 = 1900$ Hz and $fp_2 = 2600$ Hz.

The following ranges define the passband:

lowpass $0 \leq f \leq fp_1$

highpass $fp_1 \leq f \leq f_{samp}/2$

bandpass $fp_1 \leq f \leq fp_2$

bandstop $0 \leq f \leq fp_1 , fp_2 \leq f \leq f_{samp}/2$

where fp_1 is passband frequency 1,

                 fp_2 is passband frequency 2, and

                 f_{samp} is the sampling rate.

The lower stopband frequency (fs_1), the upper stopband frequency (fs_2), 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 (fs_1), and from the upper stopband frequency (fs_2) to half of the sampling rate (*Nyquist rate*). 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 2-4, the stopband specification has a minimum attenuation of –40 dB between the frequencies of 0 and $fs_1 = 1500$ Hz and between the frequencies of $fs_2 = 2600$ Hz and 4000 Hz.

The following ranges define the stopband:

lowpass $fs_1 \leq f \leq f_{samp}/2$

highpass $0 \leq f \leq fs_1$

bandpass $0 \leq f \leq fs_1 , fs_2 \leq f \leq f_{samp}/2$

bandstop $fs_1 \leq f \leq 2$

where fs_1 is passband frequency 1,

                 fs_2 is passband frequency 2, and

                 f_{samp} is the sampling rate.

In this panel, the DFD application estimates the minimum filter order required for the selected type and design 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

Use the design panel **DFD Menu** to complete the following tasks:

- 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 **Classical FIR Design** panel.

- Transfer the poles and zeros to the **Pole-Zero Placement** panel.

- Return to the **Main Menu** panel.

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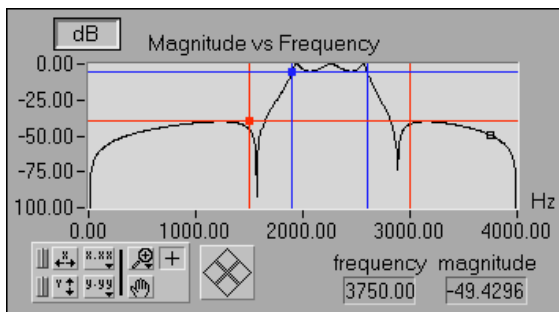
Figure 2-5. Magnitude vs Frequency

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

The frequency (*x*-axis) is in units of hertz. The full scale ranges from 0.0 to Nyquist (half the sampling rate).

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 greater than or equal to the specified passband response. In the stopband, the filter has a gain less than or equal to the specified stopband attenuation.

dB

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.

frequency	magnitude
3750.00	-49.4296

You can enter the complete filter specifications using the text entry portion at the upper-right side of the design panel.

passband resp	-5.0000	
passband freq	1900.00	2600.00
stopband atten	-40.0000	
stopband freq	1500.00	3000.00
sampling rate	8000.00	
type	bandpass	
design	Elliptic	

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
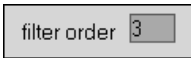
filter order 3

The **filter order** indicator, shown in the illustration at left, 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.

The **message** window displays errors that occur during the IIR design procedure. These errors usually reflect filter specifications that are inconsistent with the chosen filter type.
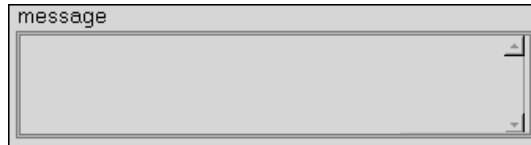
Classical FIR Design

Figure 2-6 shows the **Classical FIR Design** panel. This panel functions similarly to the **Classical IIR 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.
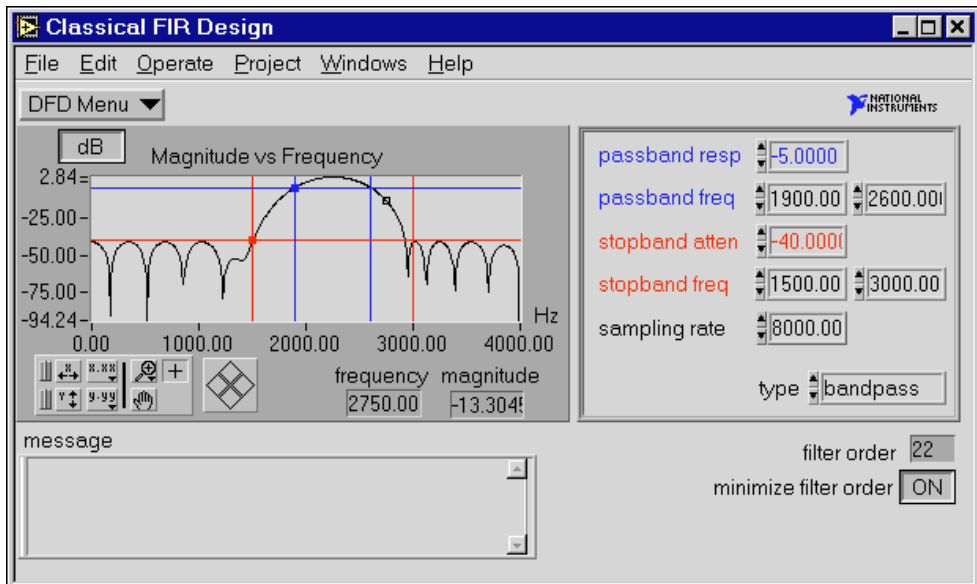

Figure 2-6. Classical FIR Design Panel

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 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 (fp_1), upper passband frequency (fp_2), and the passband response (Gp) define the passband specification. For the bandpass filter, the passband ranges from fp_1 to fp_2. 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 $fp_1 = 1900$ Hz and $fp_2 = 2600$ Hz.

The following ranges define the passband:

lowpass	$0 \le f \le fp_1$
highpass	$fp_1 \le f \le f_{samp}/2$
bandpass	$fp_1 \le f \le fp_2$
bandstop	$0 \le f \le fp_1$, $fp_2 \le f \le f_{samp}/2$

where	fp_1 is passband frequency 1,
	fp_2 is passband frequency 2, and
	f_{samp} is the sampling rate.

The stopband frequencies (fs_1 and fs_2) 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 (fs_1) and from the upper stopband frequency (fs_2), to half of 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 has a minimum attenuation of –40 dB between the frequencies of 0 and $fs_1 = 1500$ Hz and between the frequencies of $fs_2 = 2600$ Hz and 4000 Hz.

The following ranges define the stopband:

lowpass $fs_1 \leq f \leq f_{samp}/2$

highpass $0 \leq f \leq fs_1$

bandpass $0 \leq f \leq fs_1$, $fs_2 \leq f \leq f_{samp}/2$

bandstop $fs_1 \leq f \leq 2$

where fs_1 is passband frequency 1,

fs_2 is passband frequency 2, and

fp_{samp} is the sampling rate.

The **Classical FIR Design** panel estimates the minimal filter order required for the selected type and design 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 similar 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 complete the following tasks:

•   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 **Classical IIR Design** panel.

•   Return to the **Main Menu** panel.

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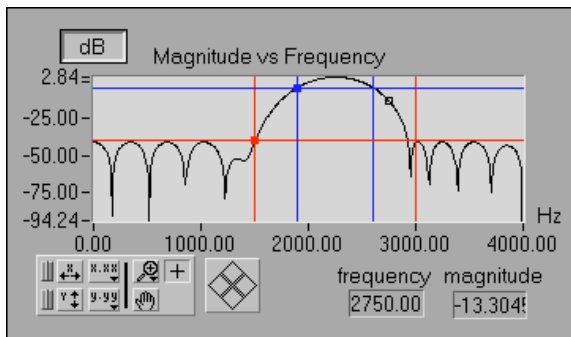
Figure 2-7. Frequency Response Magnitude

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 (half the sampling rate).

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 greater than or equal to the specified passband response. In the stopband, the filter has a gain less than or equal to the 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 in the following illustration.

frequency	magnitude
2750.00	-13.3045

You can enter the complete filter specifications using the text entry portion at the upper-right side of the design panel.

02Ebdpss.bmp

passband resp	-5.0000	
passband freq	1900.00	2600.00
stopband atten	-40.0000	
stopband freq	1500.00	3000.00
sampling rate	8000.00	
type	bandpass	

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 panel. 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
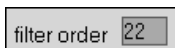
filter order 22

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

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

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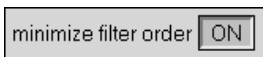
The **minimize filter order** button controls whether the DFD application minimizes the estimated filter order. If this button is OFF, the DFD application uses a fast formula to estimate the filter order to meet or exceed the desired filter specifications. If this button is ON, the DFD application iteratively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

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
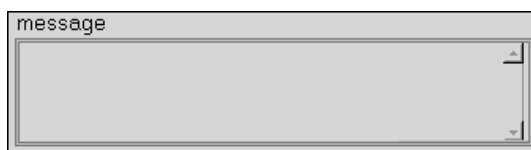
Pole-Zero Placement Filter Design

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.
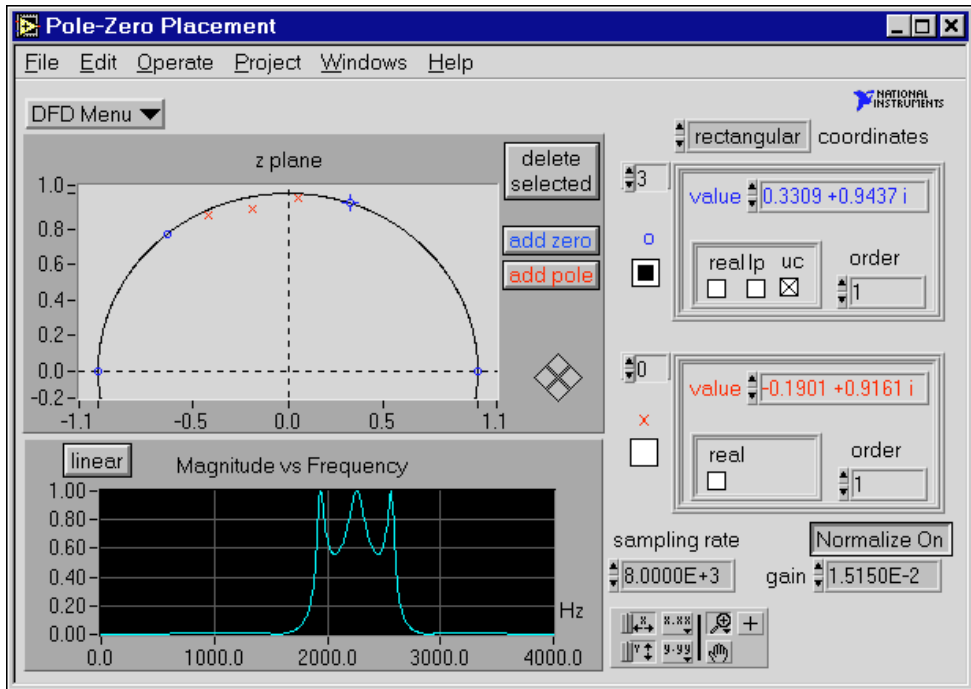
Figure 2-8. Pole-Zero Placement Filter Design Panel

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 might have originated 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 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 and zero cursors update automatically.

The following specifications describe pole–zero filter designs:

*   Pole and zero locations in the *z*-plane

*   Characteristics of each pole and zero

*   Gain

*   Sampling rate

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. For immediate graphical feedback to your pole-zero filter designs, the Magnitude vs Frequency plot updates automatically when you change the poles or zeros.

Pole-Zero Placement Panel Controls and Displays

DFD Menu ▼

Use the design panel DFD menu to complete the following tasks:

*   Save your filter specifications and coefficients.

*   Load filter designs from previous work.

*   Open the **Analysis** or **DAQ and Filter** panels.

*   Return to the **Main Menu** panel.

Figure 2-9 shows the *z*-plane plot of the filtered poles and zeros. You can move each pole (red ×) 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.
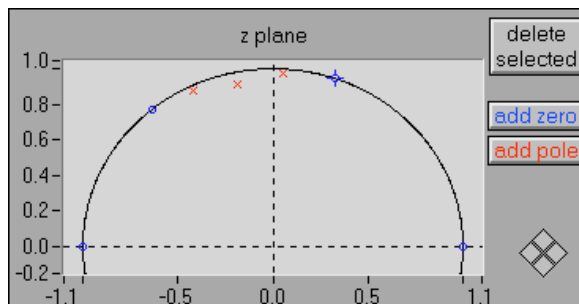
Figure 2-9. Z-Plane Plot of Filter Poles and Zeros

delete
selected

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

|	
add pole	Click the **add pole** button to add a pole to the *z*-plane. The new pole is located at the origin.
add zero	Click the **add zero** button to add a zero to the *z*-plane. The new zero is located at the origin.
rectangular coordinates	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 index of the displayed zero. By changing this index value, you can display a particular zero of 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.
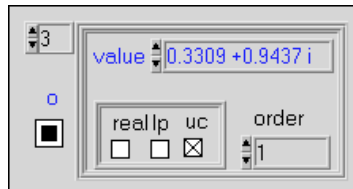
If you select the **real** check box, the zero becomes purely real and is limited to real-axis movement.

When you select the **lp** check box, the zero has linear phase. 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 select 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 index of the displayed pole. 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.
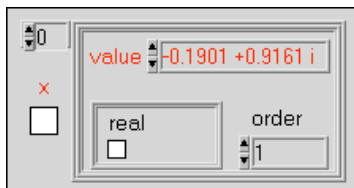

Only one special characteristic applies to poles—whether they are real. If you select the **real** check box, the pole becomes purely real and is limited to real-axis movement.

The **order** text entry specifies the pole order, or the number of actual poles at this location in the *z*-plane.
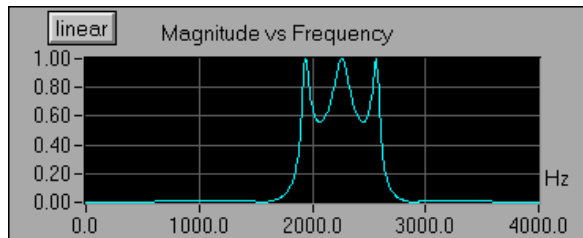
An *Mth*-order pole at $z = a$ has a *z*-transform of

$$H(z) \; = \; (z - a)^{-M}$$

If you change the coordinates to polar coordinates, the DFD application displays the poles and zeros in polar coordinates.
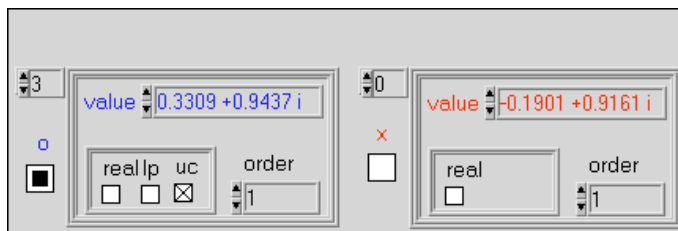
The following graph plots the frequency response $H(f)$ magnitude of the designed digital filter.

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 (half the sampling rate).

The **sampling rate** control specifies the sampling rate in samples per second (hertz).
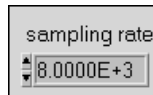
The **gain** control specifies the gain constant for the designed filter. Increasing this gain increases the overall gain of the designed filter.
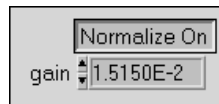
Setting the normalize button to **Normalize On** adjusts the filter gain so that the maximum response is 1.0 (0 dB). If you set this button to **Normalize On**, you cannot adjust the gain control manually.

Arbitrary FIR Design

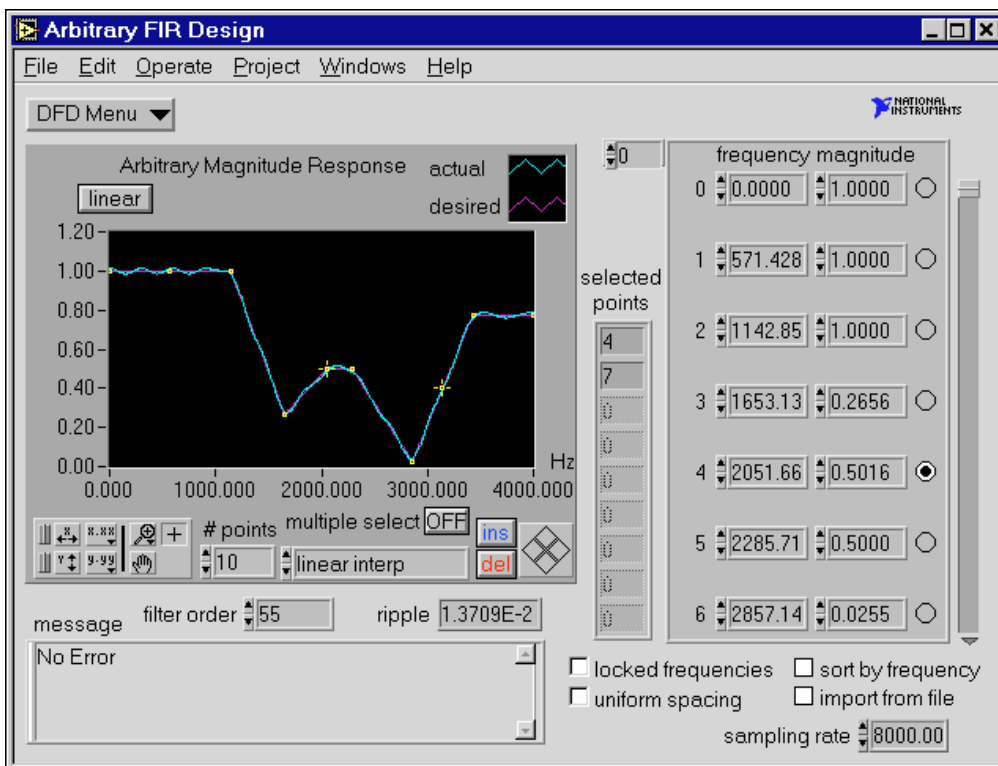Figure 2-10 shows the **Arbitrary FIR Design** panel. The panel includes a graphical interface with the Magnitude vs Frequency cursors on the left side and a text-based interface with digital controls on the right side.

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 Nyquist (half the sampling rate). The DFD application then processes 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 complete the following tasks:

• Save your filter specifications and coefficients.

• Load filter designs from previous work.

• Open the **Analysis** or **DAQ and Filter** panels.

• Return to the **Main Menu** panel.

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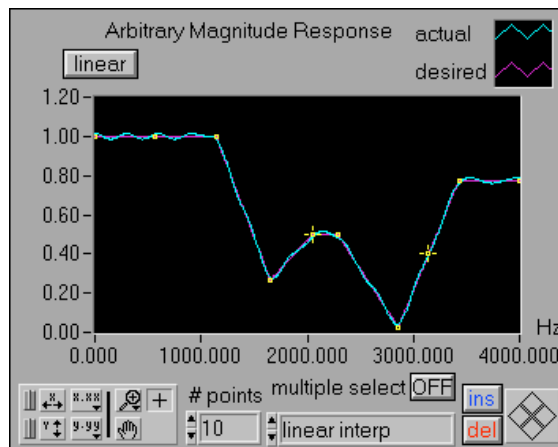
Figure 2-11. Desired and Actual Magnitude Response

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 (half the sampling rate).

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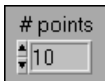
The **# points** control specifies the number of frequency-magnitude points the DFD application uses to create the desired filter magnitude response.

Reducing this number deletes points from the end of the frequency-magnitude array, whereas increasing this number inserts the additional number of points to the right of the selected point.

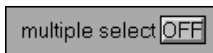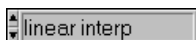Set the **multiple select** button to ON to select more than one frequency-magnitude point on the response graph. Clicking a selected point removes that point from the selection list.

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

Choose **linear interp** to create flat filters (lowpass, highpass, bandpass, and bandstop). Choose **spline interp** to create smoothly varying filters.

Click 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.

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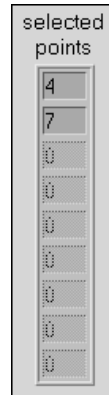
The following array is the array of frequency-magnitude points the DFD application uses to construct the desired filter magnitude response. The DFD application forms the desired filter response by interpolating between these points.
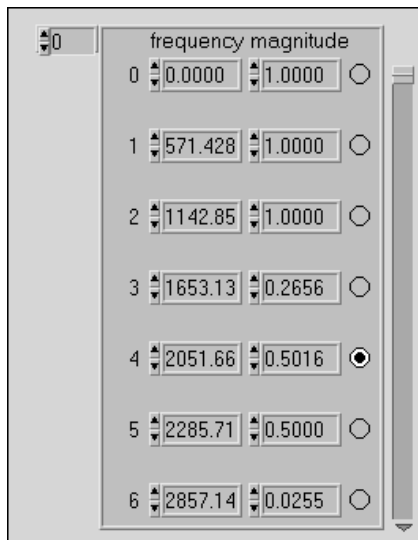
The frequency of each point is in hertz and the magnitude is in linear or decibel units of gain, depending on the setting of the button in the upper-left corner of the Arbitrary Magnitude Response graph.

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

filter order ‡ 55

The **filter order** control specifies the total number of coefficients in the digital FIR filter.

ripple 1.3709E-2

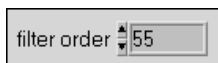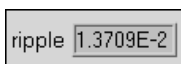The **ripple** indicator displays the largest absolute error (linear) between the desired and actual filter responses.

The **message** window displays errors that occurred during the FIR design procedure.

message
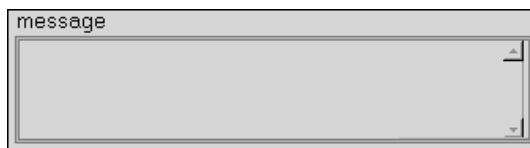
Select the **locked frequencies** check box to lock the present frequency values of the frequency-magnitude points. If you select this check box, you can alter only the magnitude or *y* value of the frequency-magnitude points.

□ locked frequencies □ sort by frequency
□ uniform spacing   □ import from file
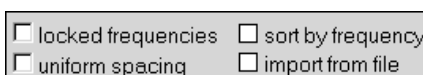
Select the **uniform spacing** check box to space the frequency values frequency-magnitude points. The DFD application spaces the frequency-magnitude points uniformly from 0.0 to half the sampling rate, inclusive.

Select the **sort by frequency** check 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; only the order of the points can change.

Select the **import from file** check box to import frequency-magnitude points from a text file.

The imported file format consists of the following tab-delimited columns:

1st line:	sampling rate	dBLinear setting (0 for **linear**, 1 for **dB**)
2nd line:	frequency 1	magnitude 1
3rd line:	frequency 2	magnitude 2
4th line:	frequency 3	magnitude 3
.	.	.
.	.	.
.	.	.
last line:	last frequency	last magnitude

For example, a file with 5 frequency-magnitude points appears as below.

```
8000.0                        1
0.0                           −60.0
1000.0                        −40.0
2000.0                        −20.0
3000.0                        0.0
4000.0                        −60.0
```

sampling rate ⇕ 8000.00   The **sampling rate** control specifies the sampling rate in samples per second (hertz).

Analysis of Filter Design Panel

Figure 2-12 shows the **Analysis of Filter Design** panel. Use this panel to complete the following tasks:

- View the filter magnitude response, phase response, impulse response, step response, and pole-zero plot.

- View and print full-screen plots of each response.

- In the full-screen views, save the analysis results to text files.
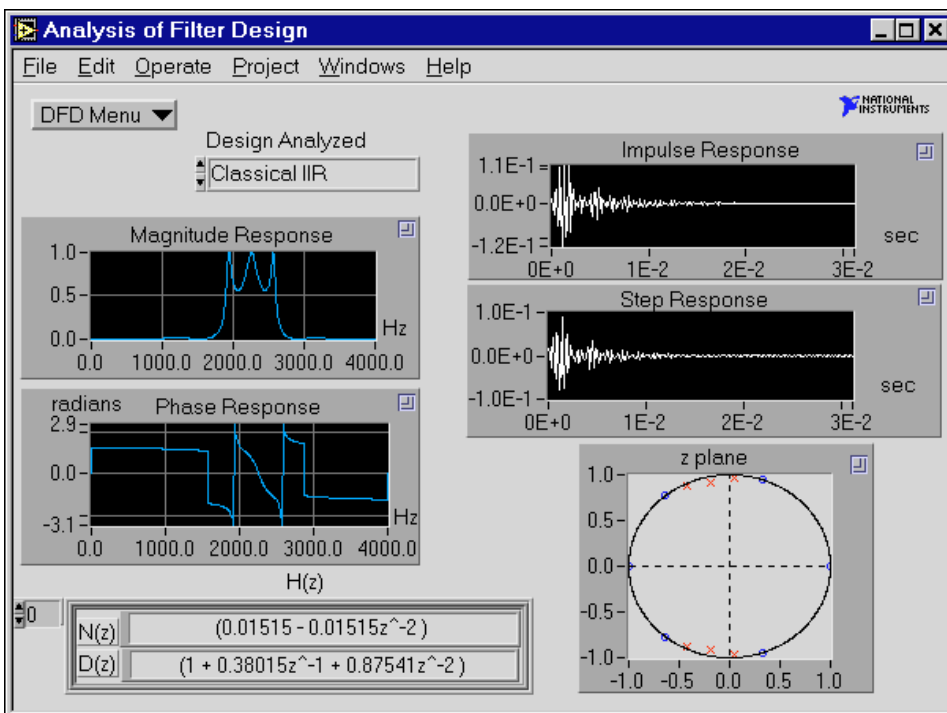
Figure 2-12. Analysis of Filter Design Panel

If you select **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 also can analyze any of the four filter designs from the **Design Analyzed** ring control; the **Analysis of Filter Design** panel uses the filter parameters from the selected filter design.

DFD Menu ▼

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 **Main Menu** panel.

Design Analyzed
Classical IIR
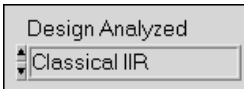
Use the **Design Analyzed** control to select the filter control to analyze. If you continue to modify the same filter design that the DFD application is analyzing, the application recomputes all filter responses.

Analysis Displays

Each of the five filter plots has a zoom box in the upper-right corner. Click in this box to display a full-screen version of the 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 response $H(f)$ as frequency varies from zero to half the sampling rate.

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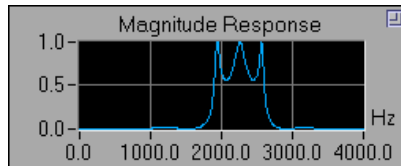
Phase Response

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

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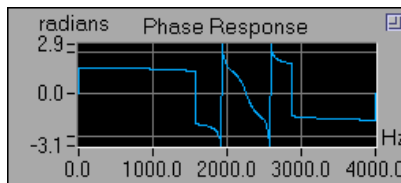
Impulse Response

The Impulse Response of a digital filter is the output of the filter 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 impulse response of the selected filter design.
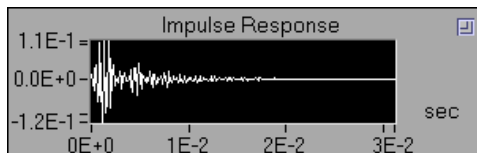
Step Response

The Step Response of a digital filter is the output of the filter when the input is a unit step sequence (1, 1, 1, …). The input samples before the step sequence are defined as zero.

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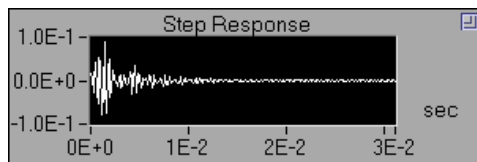
Z-Plane Plot

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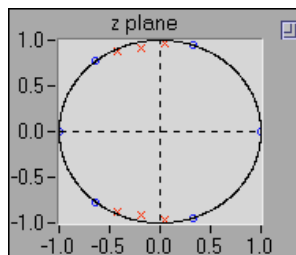
Each pole is represented by a red ×. Each zero is represented by a blue o.

H(z) for IIR Filters

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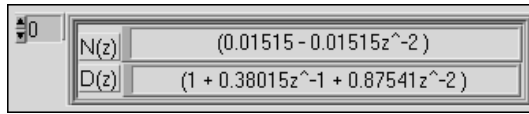
For an IIR filter, $H(z)$ can be represented by a product of fractions of second-order *z*-polynomials:

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

where $N_k(z)$ is the numerator for stage *k*,

$D_k(z)$ is the denominator for stage *k*, and

N_s is the 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 side of the $H(z)$ display.

H(z) for FIR Filters

$H(z)$ is the *z*-transform of the designed digital filter, as illustrated in the following figure. You can scroll through **H(z)** using the scroll bar.
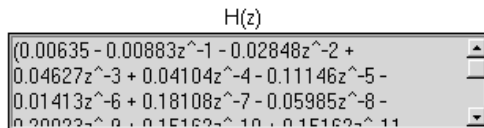
For an FIR filter, $H(z)$ can be represented as a polynomial in z^{-1}.

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

where $j = 0, 1, \ldots, order - 1$,

h_j represents the FIR filter coefficients, and

order is the number of FIR coefficients.

DAQ and Filter Panel

Figure 2-13 shows the **DAQ and Filter** panel. Use this panel if you have a National Instruments DAQ device and you want to see how the current filter design performs on real-world signals, or if you want to check the performance of your filter with a simulated signal. In this panel, you can configure your DAQ device and then acquire real signals. The acquired data passes through the designed filter, and the DFD application plots the input and output waveforms and spectrums.
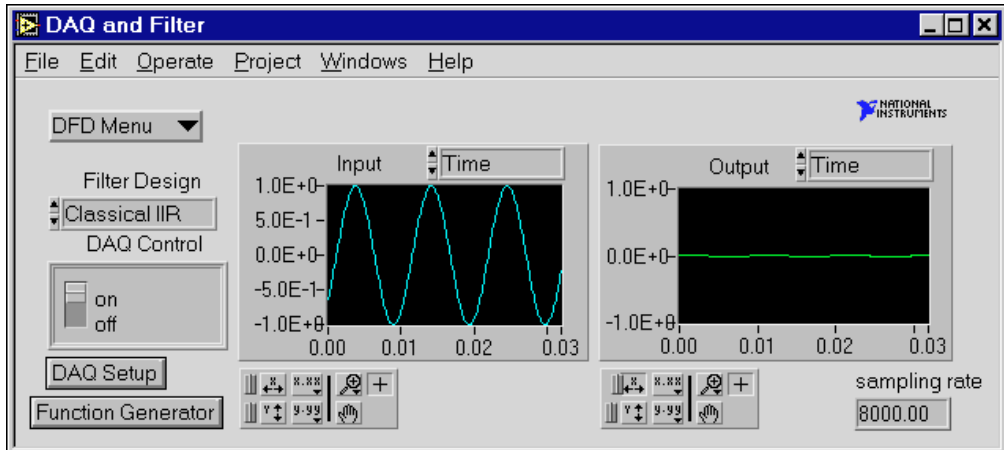
Figure 2-13. DAQ and Filter Panel

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 also can use any of the four filter designs from the **Filter Design** ring control. The **DAQ and Filter** panel uses the filter parameters from the selected design specifications.
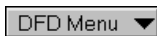
Use the **DFD Menu** to load and test filter designs from previous work, open Analysis panel, go to the selected filter design panel, or return to the **Main Menu** panel.

Use the **Filter Design** control to designate the filter design to use in filtering of the acquired signal. From the **DFD Menu** select **Go to 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 the **Acquire Once** button is clicked.
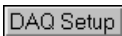
DAQ Setup

Click 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. You also can set the source to either **DAQ Device** or **Simulated DAQ**.

If you configure the source to Simulated DAQ, a built-in simulated function generator provides signals to the DAQ and Filter test panel. From the **DAQ and Filter** panel, click the **Function Generator** button to view and edit settings including signal type, frequency, amplitude, and noise level.

To change the view of a response plot, use the ring control above the plot. Select either **Time Waveform** or **Spectrum** for the input acquired signal or the filtered signal.

Figure 2-14 shows an example of switching displays for the spectrum of both the input and filtered signals.
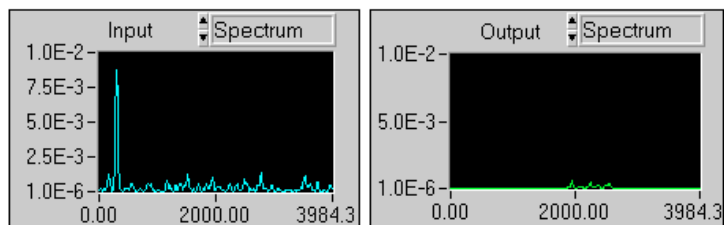
Figure 2-14. Switching Displays

sampling rate
8000.00
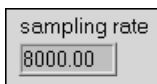
The actual sampling rate appears in an indicator at the lower-left side of the DAQ and Filter Panel, as illustrated in the figure at left.

3

IIR and FIR Implementation

This chapter describes the filter implementation equations for IIR and FIR filtering as well as the format of the IIR and FIR filter coefficient files.

Infinite Impulse Response Filters

Infinite impulse response (IIR) filters are digital filters with impulse responses that theoretically can 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 has an 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. Therefore, 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 might be appropriate. Use FIR filters for applications requiring linear phase responses.

IIR filters are also known as recursive filters or autoregressive moving-average (ARMA) filters. See Appendix A, *References*, for additional references for information about this topic.

Cascade-Form IIR Filtering

Filters implemented using the structure defined directly by Equation 3-1 are known as direct-form IIR filters. Direct-form implementations often are 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.

You can obtain a less-sensitive structure by dividing 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:

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

By factoring Equation 3-2 into second-order sections, the transfer function of the filter 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 less than or equal to $N_a / 2$, and $N_a \geq N_b$. This new filter structure can be described as a cascade of second-order filters.

Figure 3-1. Cascaded Filter Stages

You can implement each individual second-order stage using the direct-form filter equations:

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

The following illustration shows the graphical representation of these direct-form equations.
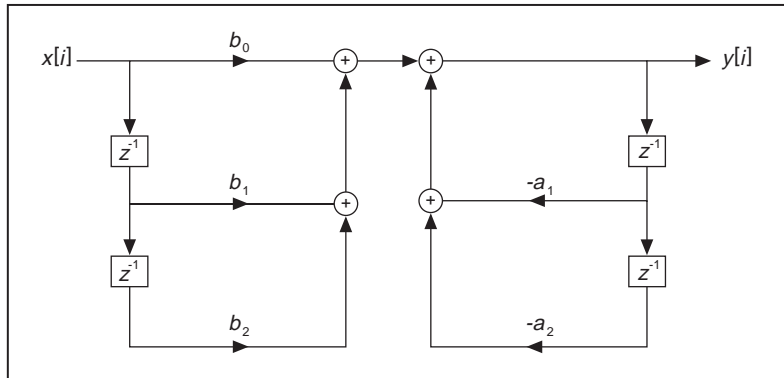
For each stage, you must maintain two past inputs ($x[i-1]$, $x[i-2]$) and two past outputs ($y[i-1]$, $y[i-2]$).

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

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

The following illustration shows the graphical representation of these direct-form II equations.
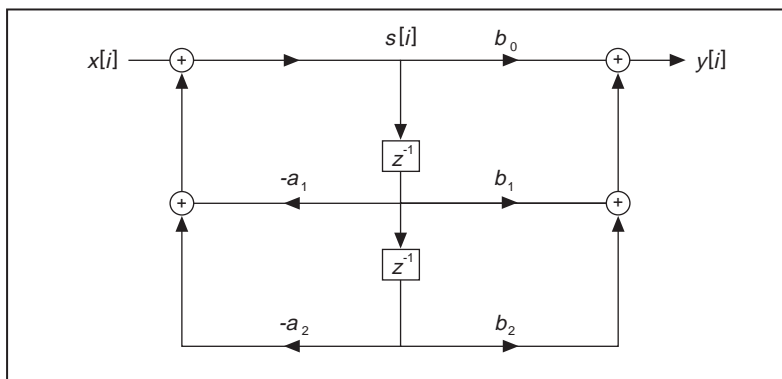
Finite Impulse Response Filters

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:

$$y_i = \sum_{k=0}^{n-1} h_k x_{i-k}, \tag{3-4}$$

where x_i represents the input sequence to be filtered, y_i represents the output filtered sequence, and h_k represents the FIR filter coefficients.

FIR filters have the following characteristics:

*   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. A delay generally is associated with the output sequence

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

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 digital filter. This section describes the format for both FIR and IIR filter-coefficient files.

FIR-Coefficient File Format

The following table provides example FIR-coefficient text files and descriptions.

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, $h[0]$
–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	

Coefficient File Example	Description
–2.002337E–1	
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 usually are described by two sets of coefficients, a and b coefficients. A total of $M \times S$ a coefficients and $(M + 1) \times S$ b coefficients exist, 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.

The following table provides example IIR-coefficient text files and descriptions.

Coefficient File Example	Description
IIR Filter Coefficients	coefficient type
Sampling Rate	sampling rate label
8.000000E+3	sampling rate in Hz

Coefficient File Example	Description
Stage Order	stage order label
2	order of each stage
Number of Stages	number of stages label
3	number of stages
a Coefficients	*a* coefficients label
6	number of coefficients
3.801467E–1	a_1 for stage 1
8.754090E–1	a_2 for stage 1
–1.021050E–1	a_1 for stage 2
9.492741E–1	a_2 for stage 2
8.460304E–1	a_1 for stage 3
9.450986E–1	a_2 for stage 3
b Coefficients	*b* coefficients label
9	number of *b* coefficients
1.514603E–2	b_0 for stage 1
0.000000E+0	b_1 for stage 1
1.514603E–2	b_2 for stage 1
1.000000E+0	b_0 for stage 2
6.618322E–1	b_1 for stage 2
1.000000E+0	b_2 for stage 2
1.000000E+0	b_0 for stage 3
1.276187E+0	b_1 for stage 3
1.000000E+0	b_2 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).

4

Using Your Coefficient Designs with DFD Utilities

This chapter describes the DFD Utilities that you use for BridgeVIEW, LabVIEW, LabWindows/CVI, and Windows filtering applications.

LabVIEW DFD Utilities

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

The two DFD utility *virtual instruments (VIs)* are Read DFD Coefficients and DFD Filter. To use these VIs, connect the file path of your coefficient file to Read DFD Coefficients. Then connect the output **Coefficient Cluster** to DFD Filter, along with your input signal. Once this sequence is followed and the VIs have executed, your filtered data is available at the DFD Filter output **Filtered X**.

Read DFD Coefficients

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.
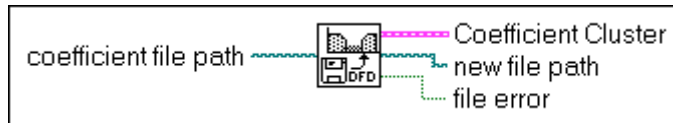
coefficient file path is the LabVIEW path to the DFD coefficient file. This file can either be in log or text-file format. If **coefficient file path** is empty, you can select a coefficient file from an open file dialog.

Coefficient Cluster is the cluster of coefficient information read from the coefficient file. The **Coefficient Cluster** contains the following parameters:

coefficient type is either 0 (IIR) or 1 (FIR).

sampling rate is the sampling rate in Hz.

IIR Filter Cluster is the cascade IIR filter cluster.

h(n) contains the FIR filter coefficients.

new file path is the file path to the coefficient file read. If **coefficient file path** is 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 has occurred while reading or interpreting the coefficient file.

DFD Filter

Filters the input array **X** using the Digital Filter Design (DFD) coefficient cluster. Use the Read DFD Coefficients VI to read your DFD coefficient files and properly initialize the input **Coefficient Cluster**.
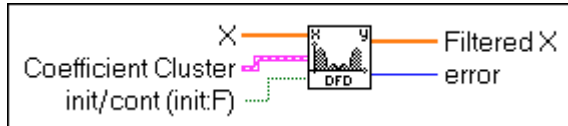
X contains the array of input samples to filter.

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

coefficient type is either 0 (IIR) or 1 (FIR).

sampling rate is the sampling rate in Hz.

IIR Filter Cluster is the cascade IIR Filter Cluster.

h(n) specifies the 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. Then you can wire this cluster to the Simple Error Handler VI or the General Error Handler VI for an immediate report on any errors. You can find descriptions of error codes in Appendix B, *Error Codes*, of the *Function and VI Reference Manual*.

LabWindows/CVI Utilities

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

The DFD Instrument Driver

The Digital Filter Design Toolkit provides a LabWindows/CVI instrument driver file named `DFDUTILS.FP`. You can find this file in the `DFDUTILS\CVISRC\INSTR` subdirectory 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. The header file `DFDUTILS.H` 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 (1p,hp,bp,bs) */
    intnum order; /* order of filter */
    intnum reset; /* 0 - don't reset, 1- reset */
    intnum 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 FilterDFD (double inputArray[], long n, FilterPtr
filterCoefficients, double outputArray[];
```

AllocCoeffDFD

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

Purpose

Allocates and clears the DFD filter coefficient structure. You must call this function once to allocate the DFD filter coefficient structure properly.

Return Value

Name	Type	Description
fptr	FilterPtr	Pointer allocated to filter structure

ReadCoeffDFD

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

Purpose

Reads your DFD filter coefficient file. You must call AllocCoeffDFD once before calling this function.

Parameters

Input

Name	Type	Description
coeffPath	character array	Pathname of DFD-coefficient file

Output

Name	Type	Description
filterCoefficients	FilterPtr	Pointer to filter-coefficient structure
samplingRate	double	Pointer to sampling rate

Return Value

Name	Type	Description
err	long integer	Error code

FreeCoeffDFD

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

Purpose

Frees the DFD filter coefficient structure and all of its coefficient arrays.

Parameters

Input

Name	Type	Description
filterCoefficients	FilterPtr	Pointer to filter-coefficient structure

Return Value

Name	Type	Description
err	long integer	Error code

FilterDFD

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

Purpose

Filters the input samples using the DFD filter coefficients. You must call `AllocCoeffDFD` and `ReadCoeffDFD` once before calling this function.

You can use this function 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 equals the number of input samples (*n*).

Parameters

Input

Name	Type	Description
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

Name	Type	Description
outputArray	double array	Output array of filtered samples that must be at least as large as **inputArray**

Return Value

Name	Type	Description
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` in the `DFDUTILS\CVISRC\EXAMPLE` subdirectory 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 installs a 32-bit DLL named DFD32.DLL for Windows 95/NT users. This DLL is located in the DFDUTILS\WINSRC\WINDLL subdirectory of your installation directory, along with a header file, DFDUTILS.H.

The DFD DLL and header file have the following function prototypes:

```
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[]);
```

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 4.0 showing you how to call the DFD utility functions. The source code is in the DFDUTILS\WINSRC\EXAMPLE\VB subdirectory of your installation directory.

A

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.

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

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

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

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

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

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.

B

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 fax-on-demand 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
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: 01 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.

Fax-on-Demand Support

Fax-on-Demand is a 24-hour information retrieval system containing a library of documents on a wide range of technical information. You can access Fax-on-Demand from a touch-tone telephone at 512 418 1111.

E-Mail Support (Currently USA Only)

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

`support@natinst.com`

Telephone and Fax 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.

Country	Telephone	Fax
Australia	03 9879 5166	03 9879 6277
Austria	0662 45 79 90 0	0662 45 79 90 19
Belgium	02 757 00 20	02 757 03 11
Brazil	011 288 3336	011 288 8528
Canada (Ontario)	905 785 0085	905 785 0086
Canada (Québec)	514 694 8521	514 694 4399
Denmark	45 76 26 00	45 76 26 02
Finland	09 725 725 11	09 725 725 55
France	01 48 14 24 24	01 48 14 24 14
Germany	089 741 31 30	089 714 60 35
Hong Kong	2645 3186	2686 8505
Israel	03 6120092	03 6120095
Italy	02 413091	02 41309215
Japan	03 5472 2970	03 5472 2977
Korea	02 596 7456	02 596 7455
Mexico	5 520 2635	5 520 3282
Netherlands	0348 433466	0348 430673
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	056 200 51 51	056 200 51 55
Taiwan	02 377 1200	02 737 4644
United Kingdom	01635 523545	01635 523154
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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: _____

_____

_____

_____

_____

Digital Filter Design Toolkit
Hardware and Software Configuration Form

Record the settings and revisions of your hardware and software on the line to the right of each item. Complete a new copy of this form each time you revise your software or hardware configuration, and use 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.

National Instruments Products

Hardware revision _____

Interrupt level of hardware _____

DMA channels of hardware _____

Base I/O address of hardware _____

Programming choice _____

National Instruments software _____

Other boards in system _____

Base I/O address of other boards _____

DMA channels of other boards _____

Interrupt level of other boards _____

Other Products

Computer make and model _____

Microprocessor _____

Clock frequency or speed _____

Type of video board installed _____

Operating system version _____

Operating system mode _____

Programming language _____

Programming language version _____

Other boards in system _____

Base I/O address of other boards _____

DMA channels of other boards _____

Interrupt level of other boards _____

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 Manual*

Edition Date: March 1998

Part Number: 320953B-01

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

_____

_____

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

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Thank you for your help.

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Glossary

Prefix	Meanings	Value
n-	nano-	10^{-9}
μ-	micro-	10^{-6}
m-	milli-	10^{-3}

data acquisition (DAQ)   The process of acquiring data, typically from A/D or digital input plug-in boards.

DFD   Digital Filter Design.

FIR   Finite impulse response.

IIR   Infinite impulse response.

Nyquist rate   Half the sampling rate.

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

virtual instruments (VIs)   Programs in the graphical programming language called G; so called because they model the appearance and function of a physical instrument.